A Mobile Multicast Streaming Video System with Seamless Playback on Wireless Mesh Networks

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Abstract - *In a wireless mesh network (WMN), handoff will occur when a mobile station (MS) changes its association from one mesh point (MP) to another due to poor link quality. For video streaming services, it is possible that the received video stream in the MS could be discontinued after the handoff process. Most researches on handoff schemes were concentrated on the efficiency of handoff process or the optimization of its latency rather than the service quality and continuity. This paper proposes a reasonable and practical multicast video streaming system in WMN which is suitable for the services like IPTV. In the proposed system, a novel handoff scheme is designed to achieve seamless playback during and after the handoff process. In addition, a handoff decision algorithm, instead of the traditional method using RSSI, is also proposed to reduce the handoff overhead and the time gap in playback. The simulation results illustrae the effectiveness of the proposed scheme.*

Key Words: Handoff, Streaming Video, Seamless Playback, Wireless Mesh Network

1.INTRODUCTION

In recent years, wireless networks, including LANs, MANs, and WANs, are widely deployed. More and more applications have been developed and applied in wireless networks. Especially, there is a tendency for the 'last-mile' access networks, which is closest to the users, to be wirelessly deployed. One popular structure of such kinds of networks is wireless mesh network (WMN) [2].

In a wireless mesh network, all access points (or base stations) are connected to each other via wireless RF links. Logically, WMN is a 2-tier architecture consisting of several mesh points (MPs) and mobile stations (MSs) as shown in Fig-1. The first-tier network (called tier-1 network), which is formed by all the mesh points, is used to transfer data between the mesh points. Some of the mesh points may also act as the access points for MSs. Such MP is called MAP. The network formed by an MAP as well as its associated MSs is defined as the second-tier network (called tier-2 network). The RF channels used in both tiers are certainly different. Unlike Ad Hoc network, there is surely at least one MP or MAP connected to the wired network in a WMN due to its role as an access network. This MP or MAP is called Mesh Portal (MPP).

On the other hand, streaming video technologies have been widely applied in entertainment, home care, video conference, remote teaching, etc. Particularly, broadcast and multicast IPTV service are strongly promoted currently [3]. Since IPTV is operated on Internet in which heterogeneous networks are inter-connected, each type of network, including WMNs, should provide QoS maintenance mechanisms to guarantee the desired playback quality.

In addition, the design of video coding scheme can provide additional tools for maintaining playback quality. For example, Fine Granularity Scalability (FGS) of MPEG-4 coding allows creation of flexible and scalable video bit streams that deliver higher compression efficiency [10]. FGS encodes the video into base layer (BL) stream and enhancement layer (EL) stream using different coding schemes.

This paper concentrates on the design of multicast streaming video system and the related handoff problem on WMN. A novel multicast real-time streaming video system with seamless playback is proposed. The considered handoff problem is to resolve the discontinuity of video streams on an MS caused by associating from the original MAP to a new one. MS could lose data during the handoff process. The video streams will be broken when some episodes are missing. There have been several researches proposed for the handoff schemes on WMNs or wireless ad hoc networks. Most of them are interested in decreasing the handoff time, the handoff latency, or the framework [4][5][6]. However, these schemes are unable to solve the problem of broken videos. The goal of the proposed solution in this paper is to achieve continuous playback of video stream with less extra bandwidth waste than the traditional handoff scheme by RSSI.

Mobile stations could not receive any data during handoff process. In [1], the authors proposed that the MS has to buffer complete video in its own buffer and then play out the buffered video stream after finishing handoff with the new MAP whose video stream playback time is behind the MS's. However, it can not handle the situation in which the playback time of the new MAP is ahead of the MS's. In this paper, a novel scheme, named as 'Burst Push,' is proposed to solve this problem. In Burst Push scheme, each MAPs/MPs should prepare an extra buffer for storing an amount of base-layer video stream data that has been transmitted to its subscribers. If the playback time of the new arrived MS is behind the MAP's, the buffered base layer stream will be transmitted to the MS in burst manner such that the playback in the MS can be keep ongoing. In addition, a handoff decision algorithm is also proposed in this paper. The goal of the proposed handoff decision algorithm is to save the bandwidth consumption due to the handoff process and the video stream forwarding as much as possible.

A simulation on the proposed system using IEEE 802.11 MAC and the handoff decision algorithm is implemented by NS-2 [16]. The simulation results are compared with the system using traditional RSSI handoff decision scheme. Three measurements, including playback time gap, handoff overhead, and total number of new MAPs inside the multicast tree, are used to view the performance in bandwidth waste and playback quality during handoff process. The simulation results shows that the proposed handoff decision algorithm is superior to the traditional RSSI scheme in saving bandwidth and reducing the duration of poorer-quality playback during the handoff process.

The rests of this paper are organized as follows. Section 2 presents the related works. The detailed explanation of the proposed multicast streaming video system on WMN is provided in Section 3. The solution of seamless playback in handoff process is described in Section 4. The simulation results and the effectiveness are illustrated in Section 5. Finally, the conclusions are given in Section 6.

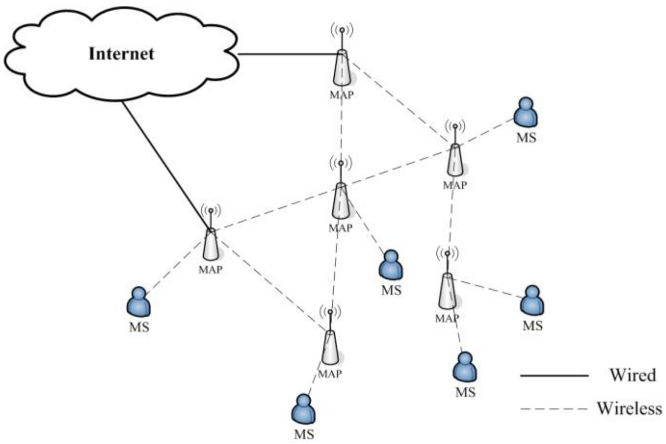

Fig-1: The architecture of wireless mesh network.

2. Related Researches

The routing path is very crucial for the performance in multicast streaming video system. There are many well-known multicast routing protocols, such as MOLSR[11], DVMRP[12], CBT[13], PIM[14], etc., have been proposed. Multicast routers construct the multicast tree with the data source as the root ot the tree. The source transmits a copy of packets along the multicast tree to all of the group members. Note that the process of building multicast tree is not involved in this paper. This paper concentrates on the design of reasonable and practical handoff scheme under the situation in which a multicast tree has been built for forwarding video stream data. That is, the proposed handoff scheme is independent of the constructing procedure of multicast routing paths.

Nevertheless, the cost function of the multicast tree still influences the performance of data transmission. In [15], the multicast trees on WMN can be classified into two kinds: Shortest Path Trees (SPTs) and Minimum Cost Trees (MCTs). It is indicated that SPTs offer better performance for multicast flows with higher packet delivery ratio and lower end-to-end delay than MCTs. Therefore, SPT is adopted in the proposed handoff scheme of this paper, That is, the multicast tree will be expanded with adding the shortest path from a newly joined MP to the tree.

Fig-2 illustrates an example of tree-based multicast architecture in a WMN. A multicast tree built in tier-1 network and some MSs in tier-2 networks are the registered member of the multicast group. Tier-1 network is responsible for forwarding video stream data between MAPs/MPs along the multicast tree. In tier-2 networks, the video stream data are then transmitted from MAP to MS in multicast manner (e.g. using multicast MAC frames for IEEE 802.11 WMNs). For example, MAP3 is the root of the multicast tree in tier-1 network. MAP3 multicasts the data to MAP7 and MAP9, and then MAP7 forwards the data to MAP11 and MAP13, and so on. These MAPs then multicast to their subscriber MSs respectively.

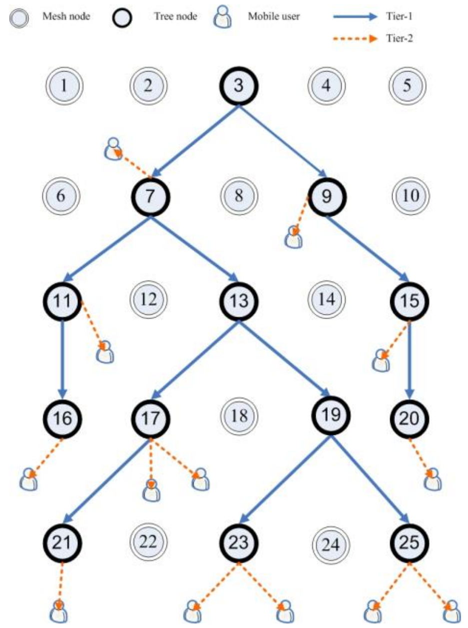

Fig-2: An example of tree-based multicast structure in WMN.

Handoff occurs when a mobile station needs to change its association from one MAP to another. The handoff process includes two parts: handoff decision and handoff procedure (e.g. re-authentication and re-association to the new AP in IEEE 802.11 WMN). There are many handoff schemes proposed for ad-hoc wireless networks. Most of them are focused on optimizing handoff latency and the framework of handoff [4][5][6]. However, the continuity of playback for multimedia streams was not concerned in these researches. It is possible that the playback will be broken during the handoff process.

In most of wireless networks, the traditional mechanism for deciding to initiate the handoff procedure is based on the inspected value of Received Signal Strength Indicator (RSSI). If the RSSI is lower than a predefined threshold, the access point with the largest RSSI is selected as the new one in the handoff procedure [8]. However, for multimedia streaming service, using RSSI alone to decide the new MAP might force the MS to select the undesired MAP such that the playback of multimedia stream might be broken.

There was a research concentrated on the handoff scheme with seamless playback of video stream on Mobile IP system called Synchronized Multimedia Multicast (SMM) [1]. In SMM mechanism, scalable video coding (SVC) [9] is exploited to achieve seamless playback of continuous media stream when the mobile stations perform handoff. This concept is also applied in the proposed WMN handoff scheme of this paper. SVC has become the trend on video coding for modern video transmission system. An example of SVC is Fine Granularity Scalability (FGS) of MPEG-4. FGS is a MPEG-4 coding tool which allows creation of flexible and scalable video bit streams that deliver higher compression efficiency [10]. FGS encodes the video into base layer (BL) stream and enhancement layer (EL) stream using different codec. The BL stream can be decoded alone to show a video with poorer quality while the EL stream is only used to combine with the BL to enhance the quality. In the illustration of the proposed seamless playback solution of this paper in the following sections, FGS is adopted as the codec of video stream. However, the proposed solution can also applied to other SVC coding schemes.

3. Multicast streaming video system on WMN

WMN is composed of several MAPs/MPs and MSs. Some of these MAPs/MPs must be the gateway connected to Internet by the backhaul network. Each MAP or MP transmits data to and receive data from its neighboring MAPs or MPs via wireless links. If the destination of the received data is at another MAP (or MP), the receiving MAP (or MP) will forward them to its neighbors according to the specific routing algorithms. If the destination of the received data is the associated MSs of the receiving MAP, the data will be forwarded to the destination MSs via the wireless link managed by the MAP.

In the proposed multicast streaming video system, it is assumed that the sources of the video streams are located in the backhaul network or located at the gateway MAPs or MPs of WMN. In order to multicasting the video stream to the subscriber MSs in the WMN, a multicast tree with the gateway MAP (or MP) as its root should be constructed in tier-1 network of the WMN. The multicast tree may be constructed by some specific multicast routing algorithm, e.g. MOLSR, discussed in Section 2. Let '*subscriber MAP*' denote the MAP with which at least one subscriber MS of the video stream associates. The data packets of the video stream are forwarded from source MAP (or MP) to the subscriber MAPs along the multicast tree. As the video stream data arrive at the subscriber MAP, it will be buffered in the MAP and then multicasted or broadcasted in the MAP's tier-2 network. Each subscriber MS receives and buffers the multicasted or broadcasted packets for playback.

To reduce the delay jitter caused in tier-1 network and simplify the jitter buffer control on subscriber MSs, the subscriber MAP is required to buffer a continuous segment of the video stream with limited length and follow the QoS requirement in multicasting or broadcasting the video stream data. The conceptual structure of the above buffering mechanism is shown as Fig-3. The incoming video stream data are buffered in Queue1 for later forwarding to next-hop neighbors. These data are also copied to Queue2 simultaneously for later multicasting or broadcasting to subscriber MSs in tier-2 network.

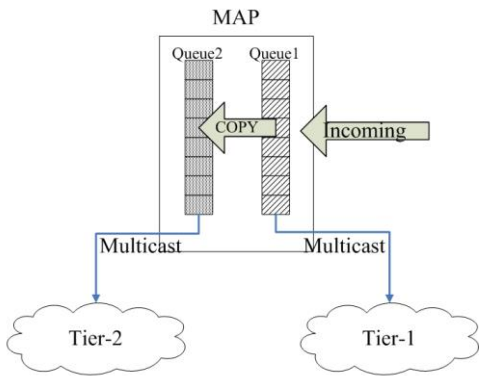

Fig-3: Conceptual structure of buffering mechanism in subscriber MAP.

And further, MS may not be able to receive packets during handoff process. This will cause discontinuity in video playback. To resolve this problem, the proposed solution is to require MS to buffer the video segment that is possibly lost for playback in the handoff duration. The detail is described later in Section 4.

In addition, there also exists synchronization problem in playback timing among subscriber MAPs. This problem is caused by the packet forwarding in tier-1 network and the proposed buffering mechanism for smooth playback. In tier-1 network, the packets are transmitted by complying with the specific MAC protocol, e.g. DCF or EDCA of IEEE 802.11. Some of the protocols may cause unpredicted transmission delay. After arriving at an MP (or MAP), the packets are buffered for further forwarding to the next-hop MPs. Since the length of routing path between two MPs is different, the end-to-end delay of any pair of MPs could be different each other. Such difference of end-to-end delay may cause unsynchronized playback timing among subscriber MAPs. Furthermore, in the proposed buffering mechanism, each subscriber MAP may tune its buffer size (i.e. Queue2 in Figure 3) according to the wireless link condition of its tier-2 network. This may also cause nonsynchronization in playback timing among subscriber MAPs. Such difference in playback timing will lead to discontinuous playback after handoff. Fig-4 illustrates an example of this problem. The playback time of the video stream in MAP1 is at 3''15 while the playback time in MAP2 is at 3"20. Once the MS perform handoff and change its association from MAP1 to MAP2, it will lose the video data of 5 seconds between 3"15 and 3"20. Thus, the continuity of video playback in MS is broken.

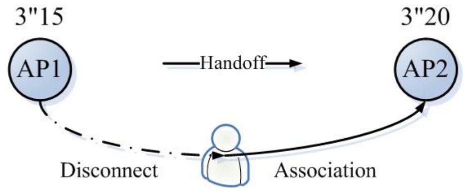

Fig-4: An example of synchronization problem in playback timing after handoff.

4. The proposed handoff mechanism

4.1 Handoff process

The handoff process in the proposed system and an example is illustrated with Fig-5 as follows. Note that the video is encoded by FGS scheme here. Actually, the proposed handoff mechanism can be also applied to the systems with other SVC codecs.

In the proposed scheme, each subscriber MAP is required to prepare a buffer (named as *handoff buffer*) for caching a segment of the BL stream that has been transmitted to its subscriber MSs. After starting the handoff process and being disassociated with its original subscriber MAP, a subscriber MS may not be able to receive the video stream data but could still continue its playback as long as there are video stream data stored in its internal buffer. After being associated to the new subscriber MAP, the subscriber MS can request the new subscriber MAP to separately transmit the missing parts from the contents cached in the handoff buffer if the playback timing of the MAP is ahead of the MS's. Thus, seamless playback can be achieved for the subscriber MS even though the data in the internal buffer of the MS is exhausted. Note that the subscriber MAP is required to only cache BL stream data in the handoff buffer in order to save the bandwidth in wireless link and the buffer size. The tradeoff such design is that the seamless playback is achieved but with poorer quality. However, the duration of playback with poorer quality is short since most handoff procedures will be finished in at most several seconds. After receiving the contents in the handoff buffer of the new subscriber MAP, the subscriber MS has all the BL stream data whose duration is from the time of internal buffer exhaustion until the current playback time of the new subscriber MAP. Then, MS may receive the normal multicast complete video stream (including BL stream and EL stream) from the new subscriber MAP and the playback quality is back to the normal.

In the example shown in Fig-5, Bn and En denote the n^{th} transmission unit of BL stream and EL stream, respectively. At the time of finishing the handoff process for the new arrival MS, the MAP has transmitted all the transmission units prior B7 and E7. In addition, the MAP also has cached partial transmitted BL stream, B4 to B7, in its handoff buffer. Meanwhile, the buffered video stream in the new arrival MS is up to 4^{th} transmission unit (i.e. B4 and E4). Thus, the new arrival MS will suffer from discontinuous playback if it directly receives the normal multicast video stream from the MAP after handoff. According the proposed handoff process, the new arrival MS will request the MAP to transmit the missing parts, i.e. B5 to B7, to it. Therefore, the playback in the new arrival MS will be continuous with poorer quality during playing 5^{th} to 7^{th} units. After obtaining B5 to B7, the new arrival MS receives video data from the normal multicast stream and its playback becomes synchronized with the other subscriber MSs.

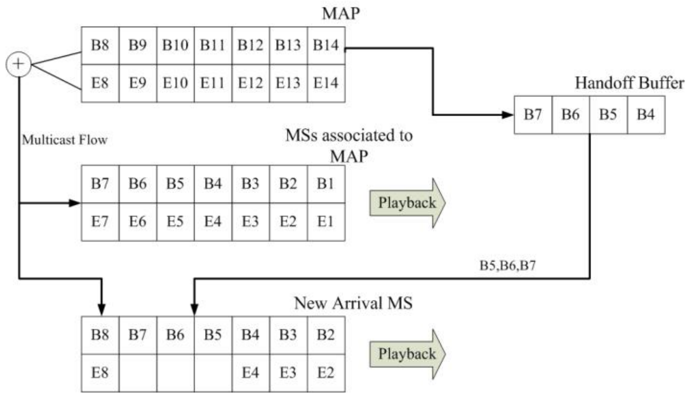

Fig-5: The proposed handoff process

On the other hand, nonsynchronous playback timing among subscriber MAPs will complicate the handoff decision and the handoff process. All the possible cases of nonsynchronous playback timing are discussed as follows:

- *Case 1 – The new MAP is outside the original multicast tree*:

In this case, the MS is associated to a new MAP that is not in the original multicast tree. Therefore, the multicast tree must be modified to transmit video stream packets to the MS. The new MAP should become a member of the tree. Therefore, a path must be determined to connect the new MAP to one of the MAPs inside the original tree. In order to simplify the handoff procedure and save the bandwidth for forwarding video stream data, this paper proposed that the MAP which is inside the original tree and connected to the new MAP may copy and transmit its contents of internal buffer for multicasting (i.e., Queue2 in Figure 3) and the handoff buffer to the new MAP along the path. Then, the new MAP can start multicasting video stream to the subscriber MS. It can be regarded as in the following Case 2 and the problem of nonsynchronous playback timing can be resolved as follows.

- *Case 2 – The new MAP is inside the multicast tree*:

In this case, it is not necessary to modify the original multicast tree and thus extra bandwidth for forwarding video stream in tier-1 network is not needed. Two sub-cases are discussed as follows.

- *Case 2.1 – The new MAP's playback timing is ahead of the MS's*:

The proposed scheme with extra handoff buffer in the MAP for transmitting BL stream to the MS is adopted to continue the playback with poorer video quality.

- *Case 2.2 – The new MAP's playback timing is behind the MS's*:

In this case, the MS still plays the video stream stored in its internal buffer after handoff. Once the buffered video stream has been exhausted, the playback in this MS will be disrupted if the MS starts play the video stream multicasted by the MAP. The reason is that the playback timing of the normal multicast video stream in the new MAP is behind of the MS's. For example, assume that the internal buffer of the new arrivals MS contains the 10^{th} to 12^{th} units of video stream after being associated to the new MAP. Meanwhile, the MAP has just transmitted the 7^{th} unit. After playing three video units, the MAP will transmit the 11^{th} unit while the expected video unit of the new arrival MS is the 13^{th}. If the MS continues playing the video stream after exhaustion of internally buffered data, the video will be looked like to be rewinded and then replayed from 11^{th} unit.

A scheme, named as '*Burst Push*', is proposed to resolve this problem. It pulls up the playback timing of the MAP with the new arrival MS's. Fig-6 illustrates how it works. Assume that the MAP has just transmitted the 7^{th} video unit and keep the 8^{th} to 14^{th} units for further multicast. At the same time, the video units stored in the internal buffer of the new arrival MS are the 10^{th} to 13^{th}. In Burst Push scheme, the MAP has to multicasts the 8th to 12^{th} video units in burst manner (instead of following original timing property of the video stream) after finishing the handoff process of the new arrival MS. After the burst transmission, MAP multicasts the following segments (from the 13^{th} unit) with the original timing again. Consequently, the playback timing in the new MAP is pulled up with the new arrival MS.

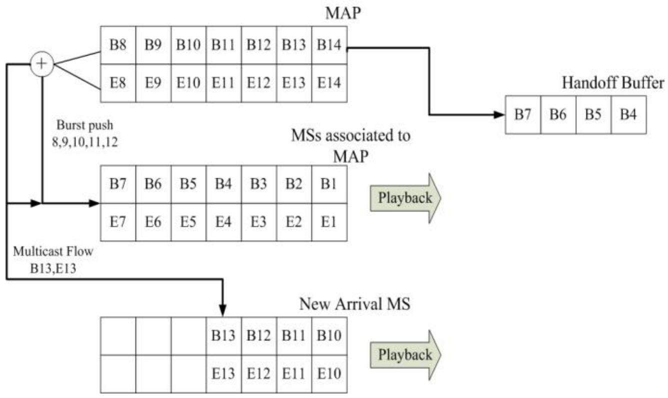

Fig-6: An example of Burst Push

Burst Push mechanism can resolve the timing synchronization problem without extra required bandwidth. The tradeoff is that all the subscribers MSs except the new arrival one must enlarge the internal buffer temporarily to cache the burst incoming video stream data. However, the enlarged internal buffer of these subscriber MSs can be shrunk if the new arrival MS disassociated with the MAP. Once the new arrival MS causing buffer enlargement moves to another MAP, the MAP which performs Burst Push may notify its subscriber MSs to reduce the buffer size and temporarily stop multicasting video stream data until the size of cached data being reduced to the desired value.

4.1 Handoff decision algorithm

To save the network bandwidth and the buffer space, it is necessary to decide the most appropriate MAP when a subscriber MS needs handoff and has searched out all the MAPs in its signal range. Note that it is assumed that the signal strength (e.g.. the value of RSSI) and the current playback time of each found MAP are also recorded by the MS at the searching time. To achieve this goal in the proposed system, three principles of choosing an MAP is proposed here. They are list as follows in the order of their priority:

Principle 1. Choose the MAP whose signal strength indicator is in a predefined reasonable range and is located in the multicast tree.

Principle 2. Choose the MAP with which the interval of poorer video quality during handoff process is minimal (for Case 2-1).

Principle 3. Choose the MAP with which the enlargement of internal buffer for the MSs originally associated to the MAP is minimal (for Case 2-2).

The proposed handoff decision algorithm according the above three principles is shown in Fig-7.

Handoff decision algorithm

Input: *MAP*,*MS_VT*, *Threshold*,

Output: The selected MAP

Begin

VTD_min ← ∞;

//Step 1: MS chooses the MAPs whose signal strength is larger than the given threshold as the candidate MAPs

for (i =1 **to** N){

  if (*RSS*(map_i) > *Threshold*)

   Insert map_i into set *CMAP*; }

//Step 2: MS calculates the cost of modifying multicast tree (for Case 1)

If (all MAPs in *CMAP* are not in multicast tree){

Calculate the hop count from $cmap_i$ to the nearest MAP in the multicast tree, $\forall 1 \le i \le M$;

Choose the MAP with minimum hop count as the selected MAP;

goto End; }

//Step 3:select the MAP with smallest difference in playback time (for Case 2)

for (i =1 **to** M) {

If ((cmapi is in the multicast tree) && |*MAP_VT*($cmap_i$)–*MS_VT*| < *VTD_min*)) {

VTD_min ← |*MAP_VT*($cmap_i$)–*MS_VT*|;

Set $cmap_i$ as the selected MAP; }}

End.

Fig-7: The handoff decision algorithm

The notations used in this algorithm are listed as follows:

- *MAP*: the set of the MAPs that has been searched out by the MS. $MAP = \{map_1, map_2, ..., map_N\}$, where N is the total number of the MAPs in the set.
- *RSS*(*map*): the signal strength of MAP *map*.
- *MAP_VT*(*map*): the current playback time of MAP *map*.
- *MS_VT*: the current playback time of the MS.
- *VTD_min*: the minimal value of the difference in playback time between the selected MAP and the MS.
- *Threshold*: the threshold of signal strength for choosing candidate MAPs.
- *CMAP*: the set of MAPs whose signal strength is larger than *Threshold*. (Such MAP is called 'Candidate MAP'.) $CMAP = \{CMAP_i \mid cmap_1, cmap_2, ..., cmap_M\}$, where M is the total number of candidate MAPs.

The steps of the algorithm are detailed as follows:

Step 1: The MS searches out all the MAPs in its transmission range for those whose signal strength is larger than the given threshold (according to Principle 1). The found MAPs are called candidate MAPs and their current playback time are also recorded.

Step 2: This step is designed to handle the case in which all candidates MAPs are not in the multicast tree, i.e., the exception of Case 2. The multicast tree must be expanded to include the selected MAP during the handoff process. The MAP with minimal cost of the expansion should be chosen to save the additional bandwidth for forwarding the video stream in the expanded multicast tree. In this algorithm, the cost is defined as the minimal hop counts from the selected MAP to one of the MAP in the original multicast tree. The value of the cost is calculated for each candidate MAP and the one with minimum cost is selected as the new MAP for handoff.

Step 3: This step is designed to handle Case 2. According to Principle 1, only the candidate MAPs located in the multicast tree are considered to become the final selection such that it is not necessary to expand or modify the multicast tree and occupy bandwidth in tier-1 network for forwarding video stream. According to Principle 2 and Principle 3, it can be easily derived that the duration of poorer-quality video in Case 2.1 or the the enlargement of internal buffer in Case 2.2 is minimized if the difference in palyback time between the new arrival MS and the selected MAP. Thus, the criterion of selecting the desired MAP in Step 3 is choosing the one which has the minimal difference in playback time from the MS.

5. Simulation Results

In this section, simulation results of the proposed multicast streaming video system and the handoff mechanism are illustrated. A simulation system using IEEE 802.11 as the MAC protocol is implemented by NS-2 [16] with several different scenarios and network topologies. The handoff overhead and the total bandwidth consumption of the proposed handoff mechanism are compared with the traditional handoff scheme in which RSSI is used to decide the new MAP. The topology of tier-1 network is randomly generated with a node of fixed coordinate as the center and then the others are randomly distributed around the node. To prevent all nodes are gathered in a small region, the maximum degree of each MAP is restricted to 3 (or 4). An MAP is randomly selected as the source of the video stream and each intermediate node in the multicast tree has at most two (or three) descendants. In addition, the number of member nodes in the multicast tree is limited to at most 70% of the MPs in the network topology. The BL and EL video streams are simulated by constant-bit-rate (CBR) traffic with 256Kbps and 768 Kbps, respectively. The transmitting range of MPs/MAPs and MSs is set as 200 meters. The channel used in tier-1 network and tier-2 network are 1 and 11, respectively. The simulation is executed 670 seconds for different network size. Initially, one subscriber MS is associated to each member MAP of the multicast tree. The random moving velocity of an MS is between 10 to 12 meter/sec.All the parameters related to the simulation system are listed in Table-1.

Table-1: Parameters of the simulation

Parameters	**Default**
Network Size	15 to 30
Maximum degree of MP	3 or 4
Maximum Tree Members	70% of all nodes
Maximum number of descendants of a tree member	2 or 3
Video Frame Rate	29.97 frames/sec
BL Stream Data Rate	256 Kbps
EL Stream Data Rate	768 Kbps
MAC_type	IEEE 802.11b
Data Rates	11Mbps, 54Mbps
Transmitting Range	200m
RF Channel	1 for tier-1, 11 for tier-2
Simulation Time	670s
MS moving velocity	10m/s to 12m/s
Maximum x-coordinate	~ 2000m
Maximum y-coordinate	~ 2000m

Three performance measurements are considered in the simulation: *playback time gap*, *handoff overhead*, and *total number of new MAPs inside the multicast tree*. These measurements are defined as follows.

1. *Playback Time Gap (PTG)*:

PTG is the difference in the number of video frames between the MS's current playback and the new MAP's when handoff occurs. Obviously, smaller PTG is desired. PTG is defined as the following equation:

$$PTG = | MSVF - MAPVF | \tag{1}$$

Where

MSVF: The video frame number that has just been played by the MS;

MAPVF: The video frame number that has just been multicasted by the MAP.

2. *Handoff overhead*:

Since this paper concentrates on the effectiveness of the proposed system and handoff scheme for seamless video playback, the considered handoff overhead is defined as the amount of extra video stream data transmitted during the handoff process in order to keep continuous playback.

The handoff overhead can be one of the following equations, dependent on which case the handoff belongs to:

$$\text{Handoff overhead for Case 2.1} = \frac{PTG}{\text{video frame rate}} \times (\text{BL data rate}) \tag{2}$$

$$\text{Handoff overhead for Case 2.2} = \frac{PTG}{\text{video frame rate}} \times (\text{EL data rate} + \text{BL data rate}) \tag{3}$$

$$\text{Handoff overhead for Case 1 with playback timing status of Case 2.1} = \left(\frac{PTG}{\text{video frame rate}} \times (\text{BL data rate})\right) + L_q \times HopCount \tag{4}$$

$$\text{Handoff overhead for Case 1 with playback timing status of Case 2.2} = L_q \times HopCount \tag{5}$$

Where

HopCount: the number of hops from the new MAP to the original multicast tree;

L_q: the amount of the data copied from the original multicast tree during the handoff process in Case 1.

Equation (2) is the handoff overhead for the case (i.e., Case 2.1) in which the new MAP is inside the multicast tree and its playback time is ahead the MS's. In this case, the required amount of extra transmitted data is the total amount of the BL stream data of which the subscriber MS is lack. Equation (3) is similar to Equation (2), but for the case (i.e., Case 2.2) in which the MAP's playback time is behind the MS's. The required amount of extra transmitted data in this case is the total amount of the complete video stream data transmitted by Burst Push scheme to the other subscribers MSs which are originally associated to the new MAP. Equation (4) and Equation (5) are for Case 1. Once the MS selects the new MAP which is outside the multicast tree, the new MAP must firstly join to the multicast tree by the path with minimum number of hops. The video stream data store in the connected MAP of the original multicast tree will be transmitted to the new MAP along the path. Note that these video stream data are forwarded hop by hop in tier-1 network. Thus, the total amount of handoff overhead caused by such video stream forwarding is $L_q \times HopCount$. After copying the buffered video stream data from the original multicast tree, the situation of the new MAP can be regarded as in either Case 2.1 or Case 2.2, depending on the playback timing status. Case 2.1 is considered in Equation (4) while Case 2.2 is considered in Equation (5). Thus, the overhead in Case 2.1, i.e., Equation (2), is added to Equation (4). However, the overhead in Case 2.2, i.e., Equation (3) is not considered in Equation (5) since there's no other associated subscriber MSs when the MAP has just joined the multicast tree, and thus Burst Push is not necessary. Note that L_q is ignored in the simulation in order to simplify the implementation.

3. *Total number of new MAPs inside the multicast tree*:

In the proposed handoff decision, the new MAP which is the member of the multicast tree will have precedence over those not in the tree. The reason of such choice is to avoid wasting extra bandwidth of tier-1 network on adding a new tree member and further forwarding video stream data. That is, the more new MAPs are not inside the multicast tree, the more bandwidth of tier-1 network is further consumed. Thus, the total number of new MAPs inside the multicast tree accumulated in the simulation interval can be used as the index of further bandwidth consumption due to handoff decision.

Fig-8 and Fig-9 show the simulation outcome of average handoff overhead versus network size. In these two figures, the blue bars are the results of proposed handoff decision algorithm and the purple bars are the results of traditional RSSI scheme. It is obvious that the handoff overhead of the proposed algorithm is much less than the RSSI scheme in all the cases with maximum number of descendants of a tree member equal to 2 and 3. The handoff overhead of the RSSI scheme is 15-time larger than the one of the proposed algorithm in the best case and at least triple in the worst case.

The amount of the extra transmitted video stream data is proportional to the amount of the difference in the playback timing between the MS and the new MAP, i.e., the value of PTG which defined in Equation (1). Thus, the required extra bandwidth can be saved more if PTG is smaller. In addition, smaller PTG leads to shorter duration of playing poorer-quality video in the handoff process. Fig-10 and Fig-11 show the simulation results of average PTG versus the network size with maximum number of descendants of a tree member equal to 2 and 3.

As shown in Fig-10 and Fig-11, the average PTG of the proposed decision algorithm is less than the traditional scheme RSSI in all cases. Namely, with the proposed mechanism, the required extra bandwidth is less than traditional RSSI scheme and the MS plays video stream with poorer quality for a relatively shorter time interval.

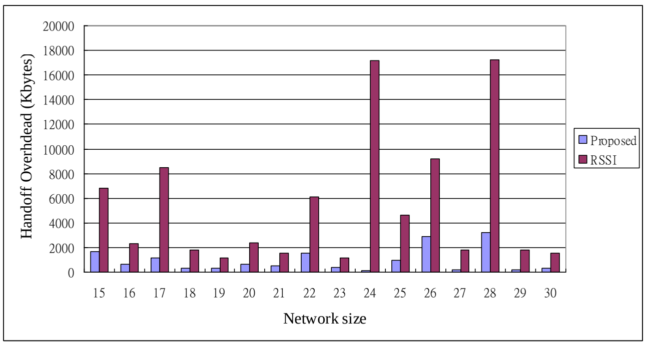

Fig-8: Total handoff overhead in each node number (maximum number of descendants of a tree member = 2).

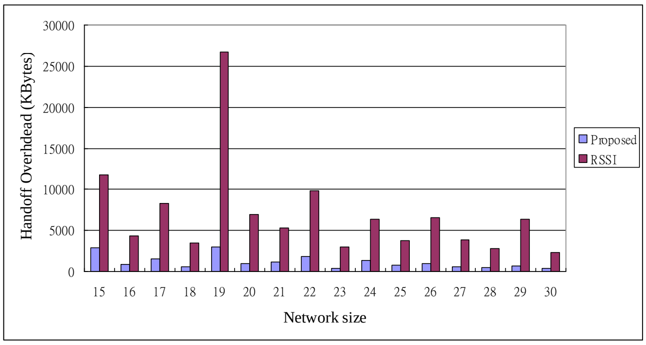

Fig-9: Total handoff overhead in each node number (maximum number of descendants of a tree member = 3).

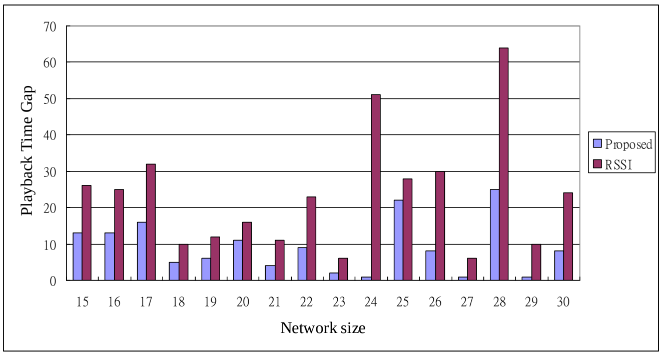

Fig-10: Average playback time gap in each handoff occurs (maximum number of descendants of a tree member = 2)

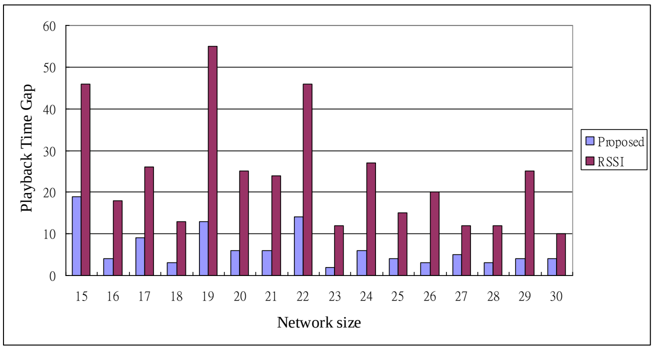

Fig-11: Average playback time gap in each handoff occurs (maximum number of descendants of a tree member = 3)

The influence of selecting a new MAP inside or outside the multicast tree in the handoff process is also discussed. The major goal of the proposed system is to ensure the continuity of video stream playback at the MS during handoff process and to save the total consumed bandwidth as much as possible. Once the new MAP is chosen from those outside the multicast tree, the new MAP must join to the multicast tree. Some extra bandwidth in tier-1 network is used to forward video stream data to the new MAP. Then the new MAP multicasts the video stream data to the MS in its tier-2 network. Further, handoff overhead is 'temporary' bandwidth consumption. It occurs just during the handoff process to keep the continuity of video stream playback. Nevertheless, the bandwidth of forwarding the video stream (along the original multicast tree to the new MAP in the tier-1 network and from the new MAP to the MS) is relatively 'long-term'. The cumulative amount of stream forwarding in tier-1 network is almost definitely larger than the handoff overhead. The reason is that it is almost not possible that the duration for which a subscriber MS stay in the signal range of an MAP is shorter than the handoff process. Thus, in order to save the bandwidth in the long term, the MS should choose the MAP for handoff that is inside the multicast tree. The simulation results on the total number of new MAPs inside the multicast tree for handoff are shown in Fig-12 and Fig-13. In all cases, the proposed handoff decision has larger number of new MAPs inside the multicast tree and less number of MAPs outside the multicast tree than traditional RSSI scheme. The simulation results shows that the proposed handoff decision can saves more extra "long-term" bandwidth than the RSSI scheme even sometimes its "short-term" handoff overhead is larger than the RSSI scheme.

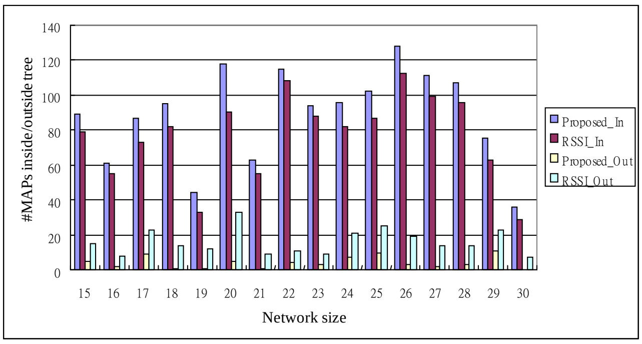

Fig-12: Number of the new MAP inside/outside the multicast tree (maximum number of descendants of a tree member = 2).

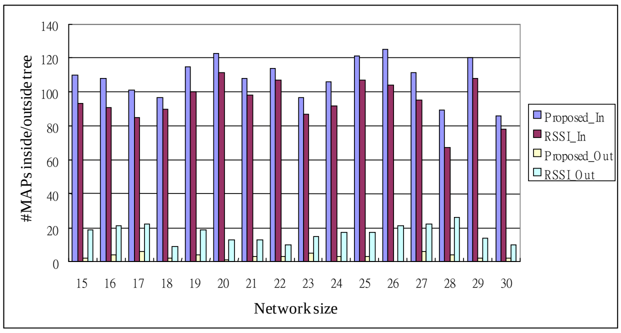

Fig-13: Number of the new MAP inside/outside the multicast tree (maximum number of descendants of a tree member = 3).

6. Conclusions

In this paper, a novel mobile multicast streaming video system with seamless playback over wireless mesh networks and a corresponding handoff decision algorithm are proposed. The major merit of this system is that it ensures continuous playback of video stream which is encoded by SVC during handoff process. With the proposed handoff decision algorithm, the mobile station selects the desired MAP that causes minimal bandwidth waste and minimal duration of poorer playback quality. The simulation of the proposed system is also implemented by using NS-2. The effectiveness of the proposed handoff decision algorithm is compared with the traditional RSSI scheme. The simulation results show that the proposed handoff decision indeed provides less bandwidth waste for both short-term handoff process and long-term video stream forwarding. In addition, it also provides shorter duration of playing video of poorer quality. Thus, the proposed handoff decision algorithm is more suitable than the traditional RSSI scheme in the proposed system.

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