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Quality-Oriented Multiple-Source Multimedia Delivery Over Heterogeneous Wireless Networks

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***Abstract*—This paper proposes a novel quality-oriented algorithm for multiple-source delivery of multimedia over heterogeneous wireless networks, which enables maintaining high levels of user-perceived quality. Unlike existing solutions, which perform delivery adaptation by adjusting the original multimedia quality to varying network conditions, this solution is based on dynamically balancing the multimedia content among multiple streams in order to achieve its goal. The proposed scheme and three other known approaches are compared in terms of estimated user-perceived quality. Simulation results show how this scheme outperforms the other solutions, including cases where the number of simultaneous receivers increases significantly. In addition, real environment tests show how the proposed scheme provides higher quality in terms of both objective and subjective metrics than competing approaches.**

***Index Terms*—Buffered multimedia delivery, multiple-source streaming, user-perceived quality.**

I. Introduction

WITH the almost ubiquitous availability of the Internet and advances in personal computer and mobile phone technologies, everyday life seems to be always connected to the network. In addition, we continue to experience technology advances with new devices, higher bandwidths, integration of existing networks and the emergence of new services. These kinds of efforts are often labeled as *Beyond 3G* [1], [2] or *Next Generation Networks* (NGNs) [3]–[5]. Fig. 1 shows an example of such a NGN, where ITU-T provides the NGN vision [4], merging fixed and mobile networks, integrating these with an IP Multimedia Subsystem (IMS) for various services, and then extending the connection through a Home Gateway (HG) [6] into the consumer electronics area of the Digital Living Network Alliance (DLNA) [7]. However, these advances in NGNs usually focus on issues below the transport layer and how it manages issues caused by heterogeneous networks. Due to this limitation in the networking, determining and maintaining high *quality of service* (QoS) levels during multimedia delivery for specific applications is an important research area.

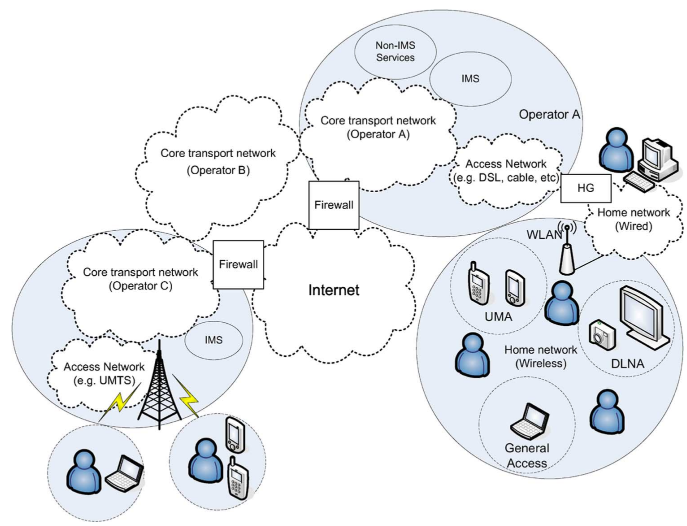

Fig. 1. Simplified architecture of a next generation network with IMS, HG, and DLNA.

Compared to single source streaming [5], [8], multiple-sender based approaches show good performance when dealing with variable network conditions during actual multimedia streaming. Some of these multiple-source approaches make use of application level multicast (ALM) [9]. However, for simplicity these schemes use Constant Bit Rate (CBR) when streaming video [10], unlike real-life high quality video encoding and delivery which in general makes use of Variable Bit Rate (VBR). In addition, they do not generally maintain high QoS levels. In terms of techniques for increasing QoS, play buffer adaptation is becoming a popular approach because using data buffering provides more flexibility to applications, but often affects the streaming quality because of buffer underflow. Generally, good initial buffer estimation or adjustment of frame duration enables better user-perceived quality in the resultant stream.

The Smart Personal Information Network (Smart PIN) [11] is introduced here in order to support novel user-centric data delivery approaches which consider a user's personal interest for data exchange and buffer management for high quality multimedia streaming. Specifically, a novel Quality-oriented Algorithm for Multiple-source Multimedia Delivery (QAMMD) is proposed for exchanging data among distributed devices which reacts to network condition changes in wireless heterogeneous networks. QAMMD includes a novel Buffer

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Underflow Avoidance Scheme (BUAS) [12]. This paper describes BUAS in details including a double buffer architecture, play buffer underflow estimation, buffer underflow avoidance scheme and segmented data streaming. In addition, it shows how QAMMD obtains good performance during multimedia streaming evaluation, including both simulation and actual prototyping tests.

In the next section, a literature review of related work is presented. Section III presents Smart PIN and our multimedia data replication scheme (MDRS). Section IV describes the multiple-source streaming scheme in detail. Simulation setup and testing results, which involve comparison with existing schemes, are shown in Section V. Conclusions are presented in the final section.

II. Related Work

The underlying principle behind multimedia streaming assumes a client-server architecture with a server serving several clients. *Single-source streaming approaches* involve one server streaming multimedia to any client. In Application Layer Multicast (ALM), most of the tree-based ALM approaches are single-source streaming, since each receiver usually gets content from one server except for some schemes which use patching [13], [14], multi-trees [15], or Multi-Description Coding (MDC) [16].

Compared to single source streaming, *multiple-source based approaches* show good performance when trying to overcome issues such as varying network conditions. In addition to some of the tree-based approaches [13]–[16], the mesh-based ALM solutions [8], [9], [17], [18] also use multiple streaming sources. However, a disadvantage of these approaches is that they use multicast which does not support high QoS provisioning which is obviously undesirable [10].

There are broadly two kinds of approaches in terms of division and assembly of content for delivery: *Interlaced Packet Assembly* (IPA) [17], [19], [20] and *Multiple-Description Code* (MDC) [18]. IPA benefits from efficient usage of network resources reducing replicated transmission of data with a reasonable saving in overhead. However, it does not treat data, which is more important to users in any way different to other, less-important data. MDC approaches are very good for lossy environments, but there is too much overhead to deliver duplicated information which is considered important.

In terms of QoS of delivered multimedia, a significant problem in the streaming is the mismatch between the available network bandwidth and the media encoding/sending rates. There are two major avenues to solving this issue: encoded media adaptation [21]–[23] and play buffer adaptation [24]–[26]. Media adaptation approaches adjust multimedia quality to the available network resources while approaches using data buffering provide more flexibility for applications, but often are affected in their streaming quality by buffer underflow. In general good buffer underflow management enables better user-perceived quality of the delivered multimedia.

There are several *encoding multimedia adaptation approaches* which make use of application and transport layers information to adjust the rate of multimedia stream to the estimated available network bandwidth [21]–[23]. These approaches could be categorized regarding how they handle the mismatch in bit rate between the application and network.

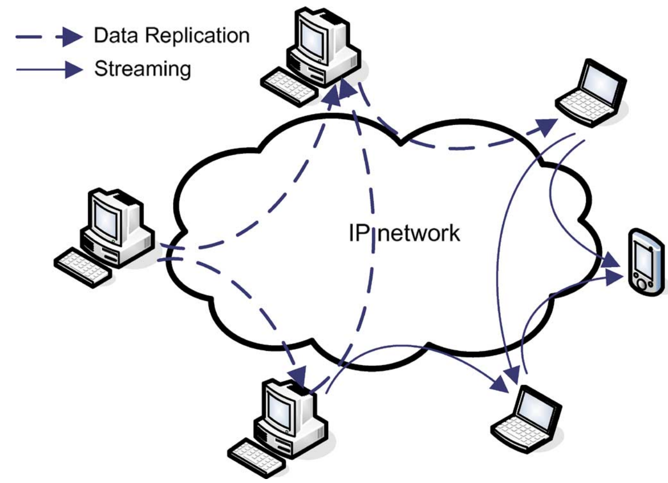


Fig. 2. Smart PIN overview.

These are good for adapting to network conditions but they sacrifice multimedia quality to compensate for this.

The approaches described in [24], [25] use an initial play buffer adaptation, and provide more flexibility to the application, but can suffer from unexpected stops in playing, caused by a lack of buffered data. The major reason for this buffer underflow is a mismatch between receiving rate and decoding data rate due to irregular network conditions. In order to resolve this issue, the following approaches provide ways to estimate the initial buffer, which is enough to avoid buffer underflow. However, the downside to this is that it introduces delay. As another play buffer adaptation approach, Adaptive Media Playout (AMP) [26] uses delay adjustment between frames according to network conditions such as delay and bandwidth. This approach uses variation instead of mean of bandwidth for frame duration adaptation.

III. Smart Personal Information Network

A. Overview

Smart PIN (performance- and cost-oriented personal information network) is a context-aware solution which focuses on efficient user access to information located on remotely distributed devices in a heterogeneous network environment. Smart PIN assumes its data will be stored in context-content data pairs and considers annotated context as part of metadata. In order to address both information overload, and the heterogeneity of devices and network connectivity, Smart PIN supports the kind of utility function-based data replication and multiple-source streaming scheme presented in Fig. 2.

The Smart PIN system architecture introduces network, service and management components as presented in Fig. 3. The service component shows that Smart PIN focuses on content sharing as a context-aware system.

The network components suppose existing approaches such as IEEE 802.21.[1] Media Independent Handover (MIH), Auto-configuration and Self-management of Personal Area Networks (ASPAN) [27] or other always-connected networks are used. The service component also includes many assumptions including that overlay network management is provided, a

[1]IEEE 802.21, http://www.ieee802.org/21/

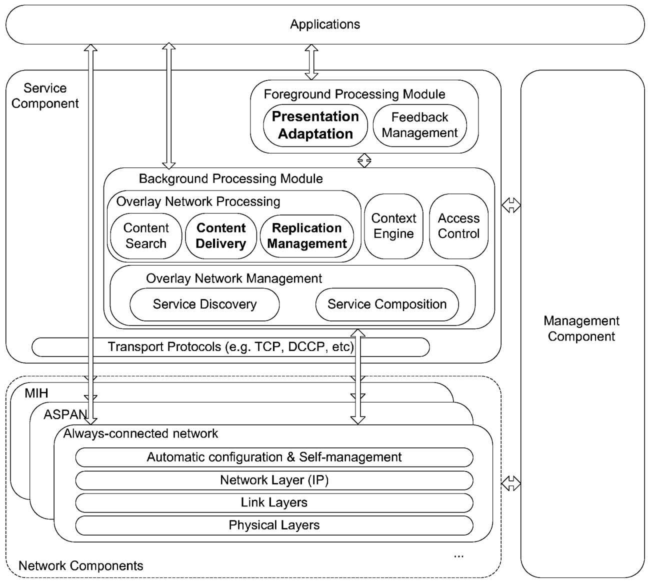


Fig. 3. Detailed architecture of Smart PIN.

content search function is supported, context-annotated content is used, access control is in place, which supports private content and shared content based on user-defined permissions. In addition, basic feedback management is provided.

The service component includes the main contributions of Smart PIN, which are data replication decision-making, content delivery and content presentation. Specifically, novel data replication and quality adaptive multiple-source delivery schemes will be introduced in the sections to follow.

B. Multimedia Data Replication Scheme

Data replication systems intentionally create multiple replicas to achieve improved performance using metrics such as data availability and these depend on replica allocation. Smart PIN uses a utility function that considers both user interest and popularity of data. Data allocation and delivery are also involved in this utility function.

Smart PIN supports a utility function-based data replication scheme in order to address both information overload, and the heterogeneity of devices and network connectivity. In this context, Smart PIN employs a novel *Multimedia Data Replication Scheme (MDRS)*, which is divided into two steps, data selection and data delivery.

In order to cope with large-sized multimedia content, Smart PIN employs data segmentation into fixed length segments (FIX_SEG) and variable length segments (VAR_SEG). Small-sized data is not segmented and is labeled NO_SEG. VAR_SEG data is usually data that uses real-time delivery such as a movie, and FIX_SEG is for other types of content.

During data selection, data is classified into three categories based on two thresholds depending on their utility for users. This categorization determines which data will be replicated to other devices. In order to decide how many data sets are needed among the devices, a minimum data set requirement (G^k) is also applied. A more detailed description of MDRS is discussed in [11].

IV. Quality-Oriented Algorithm for Multiple-Source Multimedia Delivery (QAMMD)

In order to overcome varying network conditions, *the Quality-oriented Algorithm for Multiple-source Multimedia Delivery (QAMMD)* uses an innovative double buffering architecture which includes multiple virtual buffers associated with multiple network connections and a play buffer. As mentioned earlier, QAMMD makes use of a novel Buffer Underflow Avoidance Scheme (BUAS) [12] which is described in the following sections, covering double buffer architecture, play buffer underflow estimation, a buffer underflow avoidance scheme and segmented data streaming.

A. Double Buffering Architecture

In order to support the delivery of high quality multimedia streams, the proposed unicast-based multiple-source streaming approach, QAMMD adopts a novel double buffering architecture. It includes n *senders*, n *connections* from each streaming sender to a receiver and *two levels of buffers* at the receiver as shown in Fig. 4: *multiple virtual receiving buffers* and *a play buffer*. Additionally, a novel *Buffer Coordination Module* (BCM) balances the functionality of this two-level buffer structure. Currently, all the buffers on the receiver side are assumed to be unbounded in size and the senders are assumed to share the same amount of multimedia data scheduled to be streamed to a receiver.

The multiple virtual receiving buffers are managed as *Individual Storage Spaces* (ISS). Each ISS stores multimedia data received via one of the n connections established between the multiple sources and the receiver. Although other protocols could also be used for this purpose, QAMMD makes use of the TCP Friendly Rate Control (TFRC) protocol [28], in order to best balance the aggressiveness of the multimedia delivery with friendliness towards other traffic. Each ISS i receives data via the network at a rate, R_{R_i} and provides data to the play buffer with rate R_{S_i}. R_{R_i} is estimated using TFRC throughput [28], whereas R_{S_i} is determined based on dividing the maximum media encoding rate (R_{max}) by the number of nodes. Although ISSs do not really store the data (the play buffer stores it for efficiency), they enable BCM control of the data flow to the play buffer for buffer underflow avoidance.

The *play buffer* uses the MPEG Video Buffering Verifier (VBV) mechanism [29] with an unbounded buffer size. When the number of packets in the play buffer reaches the initial number of packets set for efficient buffering (S_{init}), the data is fed to the decoder. This MPEG VBV operation guarantees that encoding-related factors do not cause buffer underflow in the play buffer given certain VBV buffer sizes, VBV delay and maximum media encoding rate R_{max}, as required by local playback [30]. Under these conditions, the play buffer will receive data at the R_{max} rate, which can be determined at encoding time. Consequently, R_{max} is used as the aggregated target value for the overall ISS sending rate which is $\sum_{i=1}^{n} R_{S_i}$. However, when setting S_{init}, current network conditions are considered, as remote delivery is often very different from local playback.

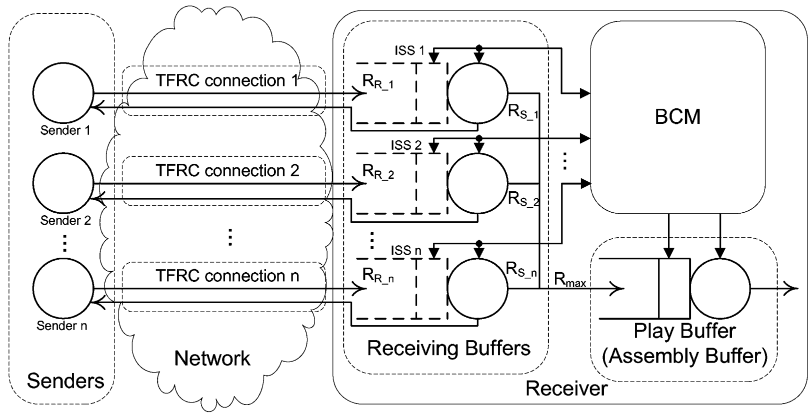


Fig. 4. Example of a QAMMD-based multimedia delivery system.

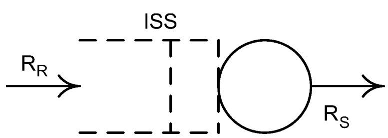


Fig. 5. A single buffer from QAMMD-based multimedia delivery system.

The *Buffer Coordination Module* (BCM) controls data flow between the TFRC connections, ISS and play buffer. BCM involves packet partition and rate allocation mechanisms, which are discussed later. To manipulate buffer parameters based on the information from the buffers and multimedia data in order to balance the receiving buffers and play buffer, BCM retrieves media-related information such as the VBV buffer size, VBV delay, and media rate (R_{max}). In addition, it determines receiving buffer parameters such as R_{S_i}, S_{init}, etc. In doing so, BCM uses an innovative *buffer underflow avoidance scheme* (BUAS) which is described in the next subsection.

B. Playing Buffer Underflow Estimation

Adopting a play buffer in the streaming application benefits quality of service as network conditions vary. However, this approach may suffer from buffer underflow. As shown in Fig. 5, a single buffer receives data at the rate R_R, and consumes data at the rate, R_S.

Xu and Helzer [24] model a single TFRC traffic which is similar to a Markov Modulated Deterministic Process (MMDP), which is a popularly used ATM traffic model. They provide *buffer underflow probability* (BUP) functions as both a closed form and an iterative form, which means the total duration of all buffer underflow events is greater than 0 sec. The closed-form BUP, $\gamma(x)$ is presented in (1), where x is the number of initial buffering packets.

$$\gamma(x) \approx \left(F_{\hat{\theta}_n}(\hat{\theta}_{\text{media}}) + \left(1 - F_{\hat{\theta}_n}(\hat{\theta}_{\text{media}})\right)\frac{1-e^{-\beta}}{1-e^{-\alpha}}\right) \cdot e^{-(\alpha-\beta)x}, \quad x > 0 \quad (1)$$

As shown in (2) [24], $F_{\hat{\theta}_n}(k)$ is the exponential random variable cumulative distribution function of $\hat{\theta}_n$ (i.e. the number of packets successfully sent between loss events) which is presented with loss event rate (p).

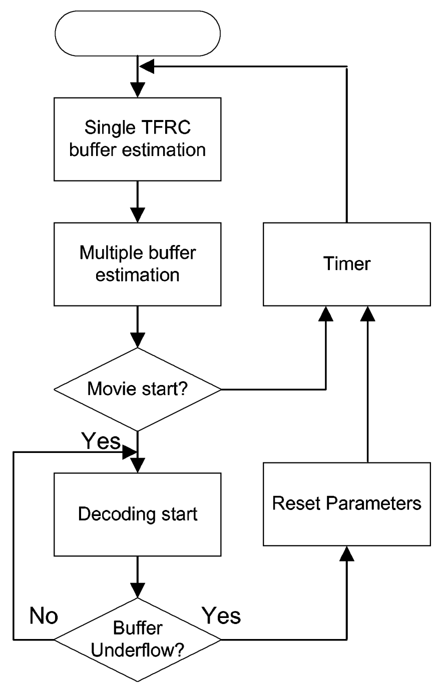


Fig. 6. BUAS flowchart.

$$F_{\hat{\theta}_n}(k) \approx 1 - \sum_{h=0}^{6} \frac{(7pk)^h}{h!} \cdot e^{-7pk} \quad (2)$$

Eq. (3) is throughput function of TFRC, which is defined with loss event rate (p) and round trip time (rtt).

$$R(p, rtt) = \frac{1}{rtt \cdot \sqrt{\frac{2p}{3}} + 12 \cdot rtt \cdot \left(\sqrt{\frac{3p}{8}}\right) \cdot p \cdot (1 + 32p^2)} \quad (3)$$

As defined in [24], $\hat{\theta}_{\text{media}}$ is presented in (4), where R_S is the consuming rate of delivered multimedia and $R(\cdot)$ is (3). Using (2) and (4), $F_{\hat{\theta}_n}$ can be presented as a function with R_S and rtt.

$$\hat{\theta}_{\text{media}} = \frac{1}{R^{-1}(R_S, rtt)} \quad (4)$$

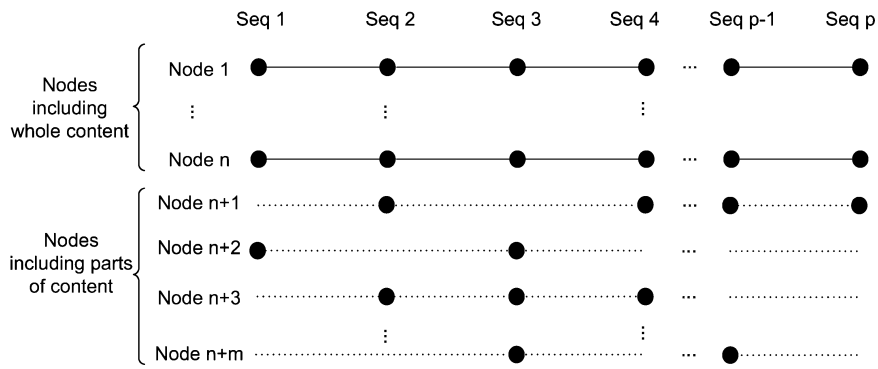


Fig. 7. Example of data replication of MDRS for streaming.

In addition, when the achieved bitrates of single TFRC, $R_R$is provided, p is also retrieved using (5).

$$p = R^{-1}(R_R, rtt) \tag{5}$$

α/β are the inverse of the expected value of changing buffered packets in decreasing/increasing states of the play buffer which are presented with p, $\hat{\theta}_{\mathrm{media}}$ and rtt. The detailed derivation of this is not discussed in this paper.

In conclusion, the closed-form BUP [24] can be presented as a function of buffer size (x), round trip time (rtt), TFRC receiving rate (R_R) and consuming rate of delivered multimedia (R_S) which is presented as (6).

$$\gamma(x) = F(x, rtt, R_R, R_S), \quad x_i > 0 \tag{6}$$

The inverse of $\gamma(x)$, $\gamma^{-1}(P)$ can be computed where P is the buffer underflow probability of a single buffer with given buffer size (x), round trip time (rtt), TFRC receiving rate (R_R) and consuming rate of delivered multimedia (R_S) as shown in (7).

$$\gamma^{-1}(P) \approx \frac{log\left(\frac{P}{F_{\hat{\theta}_n}(\hat{\theta}_{\mathrm{media}})+\left(1-F_{\hat{\theta}_n}(\hat{\theta}_{\mathrm{media}})\right)\frac{1-e^{-\beta}}{1-e^{-\alpha}}}\right)}{\beta-\alpha} \tag{7}$$

C. BCM Buffer Underflow Avoidance Scheme

The proposed BCM employs a novel *buffer underflow avoidance scheme (BUAS)* which is described in Fig. 6. The BUAS determines initial buffer estimation S_{init} for play buffer used in QAMMD. S_{init} can be the VBV buffer trigger (S_{VBV}) or assembly buffer trigger (S_{ab}). S_{VBV} is easily calculated at encoding time [29]. In case of VBR, a verified VBV buffer size is given during encoding time. For CBR, S_{VBV} can be determined with VBV delay and data rate instead of a given value. S_{ab} is the initial size estimation of the assembly buffer which is chosen as buffer size estimations by ISSs. BUAS proposes an approach to estimate the initial buffer size of the assembly buffer, S_{ab} using a single TFRC (BUP) analytic model [24] for each ISS. Simply, BUAS selects S_{init} as the biggest value from S_{VBV} and S_{ab}.

Following the results of Section IV-B, The BUP of i, $\gamma_i(x_i)$ in closed-form makes use of several parameters including round trip time(rtt), R_{R_i} and R_{S_i} since TFRC uses an equation based on loss and round trip time to determine bandwidth. Consequently, $\gamma_i(x_i)$ could be described as in (8).

$$\gamma_i(x_i) = F(x_i, rtt_i, R_{R_i}, R_{S_i}), \quad x_i > 0 \tag{8}$$

BUAS considers that assembly buffer underflow occurs when all ISSs have reached underflow. Using this assumption, the overall BUP (P) is the product of BUPs of all n ISS's as in Eq. (9)

$$P = \prod_{i=0}^{n} \gamma_i(x_i) \tag{9}$$

In order to achieve the given target BUP, P_{target} in the assembly buffer, BUAS estimates the initial buffer size with (10). P_{target} is dependent on the number of users to be supported. The higher the number of users, the smaller the probability of buffer underflow is required. For example, a good target value for 200 users is 0.005. A smaller value of P_{target} supports more users with a certain service level.

$$S_{ab} = \sum_{i=0}^{n} \gamma_i^{-1}\left((P_{target})^{\frac{1}{n}}\right) \tag{10}$$

In summary, BUAS determines the initial buffer size S_{init} before providing frame data to the decoder using S_{VBV} determined during encoding time and S_{ab} which is estimated periodically using target assembly buffer underflow probability (P_{target}), received data rate (R_{R_i}), data rate to the decoder (R_{S_i}) and inverse function of ISS BUP function $F^{-1}(\cdot)$.

D. Segmented Data Streaming Scheme

In order to use data replicated with MDRS, QAMMD needs to search segments and to deliver those from the multiple-servers. It is assumed that the search function is supported in overlay network processing. However, some of the segmented content for delivery (e.g. VAR_SEG) is not fully replicated to a specific node, presented in Fig. 7, as data replication scheduling for data delivery.

Streaming with nodes including the whole content is the ideal case of QAMMD since the initial buffer prediction using bandwidth estimation is more precise with a fixed number of nodes during the streaming service. However, data replication requires performing anytime, and it is also useful to use more nodes in order to have more bandwidth although they do not include all parts of the content. In addition, the buffer for received data supports using partly replicated content, since there is previously received data.

There are several assumptions for rate allocation and packet partitions to the source. As mentioned previously, both of them

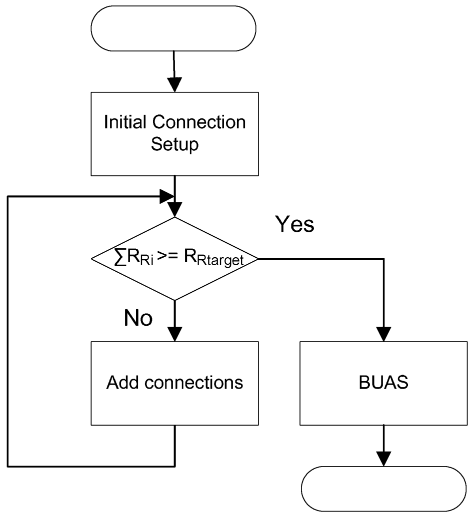


Fig. 8. Flowchart for rate allocation in BCM.

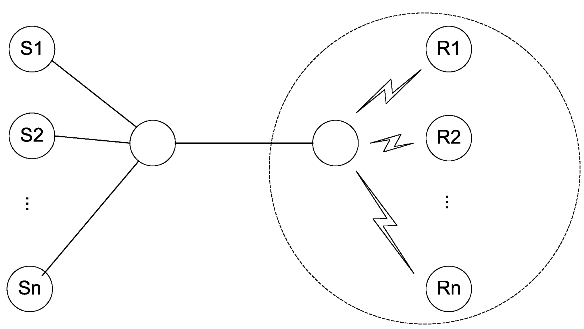


Fig. 9. WLAN dumbbell topology.

start only with nodes which have whole content. The allocated rate is proportional to the rate which the node can transmit. When there is not enough bandwidth for $R_{R_{target}}$, the number of connections can be increased. A simple flow chart for this procedure is presented in Fig. 8. $R_{R_{target}}$ is assumed to be that little bit higher (e.g. 10%) than the multimedia encoding rate, R_{max}. The packet partition can be a server-based approach similar to [17] or a receiver-based approach similar to [18]. Smart PIN can use either, but assumes that the second one is used.

V. Modeling and Simulation

Our proposed scheme was evaluated via network simulation using Network Simulator 2 (NS-2)[2] and this is now described.

A. Network Topology and Scenario

The "*dumbbell*" topology is a typical model of the Internet [31] and is a popular topology for streaming applications [32]–[34]. There are two dumbbell topologies which are used in the paper. These are the WLAN dumbbell topology and WLAN-WiMAX dumbbell topology as presented in Figs. 9 and 10(a), respectively. The dumbbell topologies used in the simulations have a middle section with 200 Mbps bandwidth and 5 msecs delay. Therefore, the wireless links are the actual bottlenecks since they have less bandwidth than the middle link

[2]Network Simulator 2, http://www.isi.edu/nsnam/ns/

TABLE I
Used NS-2 IEEE 802.11g Physical Layer Parameters

Parameter	Value
Freq_	2.4GHz
Pt_	2.81838×10^{-1}
RXThresh_	7.74636×10^{-9}
CSThresh_	7.74636×10^{-9}

TABLE II
Used NS-2 IEEE 802.11g MAC Layer Parameters

Parameter	Value
SlotTime_	9usec
SIFS_	16usec
PreambleLength_	96 bits
PLCPHeaderLength_	40 bits
PLCPDataRate_	6 Mbps
dataRate_	54 Mbps
basicRate_	6 Mbps

as illustrated in Figs. 9 and 10(a). Based on simple movement of mobile nodes, MIH is used for handover as illustrated in Fig. 10(b). As mentioned, these topologies are used for multiple-source streaming. Data replication traffic is simulated by background TCP traffic.

An IEEE 802.11g WLAN was used for simulation based on the NS-2 default implementation. The detailed parameters for the NS-2 IEEE 802.11g model for physical layer and MAC layer are presented in Tables I and II. The No Ad-Hoc (NOAH)[3] extension is also used in order to simulate the realistic environment which is adopted for conventional WLAN access points.

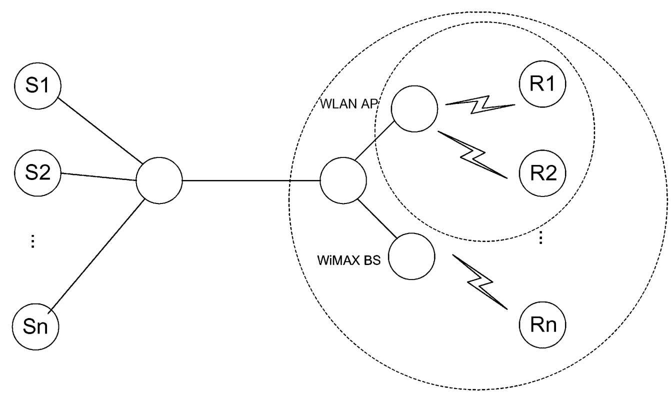
In Table I, Feq_means the frequency which is used, Pt_is the transmit power, RXThresh_represents the signal strength of one frame received by the receiver and CSThresh_is carrier sense threshold to determine whether one frame is detected by the receiver.

In Table II, SlotTime_is a unit of back-off delay, SIFS_represents short interframe space, PreableLength_means the length of Physical Layer Convergence Procedure (PLCP) preamble, PLCPHeaderLength_is the length of the PLCP message header, PLCPDataRate_is the rate for control frame, but this is related to Extended IFS (EIFS). DataRate_is the rate for data frames and finally, BasicRate_is the rate for control frames.

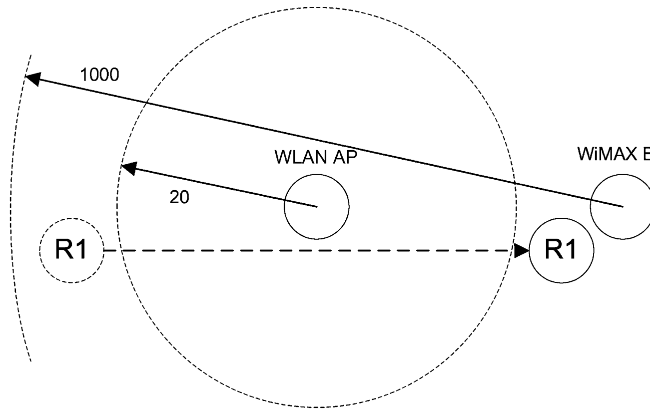
The NIST IEEE 802.16 module[4] is used as the WiMAX extension for NS-2 in this paper. This is based on the IEEE 802.16 standard [35] and the mobility extension 80216e-2005 [36]. This model is extended as a subclass of the NS-2 802.11 model including the physical layer (Phy/WirelessPhy/OFDM) and MAC (e.g. Mac/802_16). As the MAC operation of IEEE 802.16 is different from that of 802.11, MAC configuration is required before the simulation starts, including address classifier (i.e. SDUClassifier), MAC interfacing (i.e. WimaxScheduler), channel, etc. A subscriber station and a base station have different features in WimaxScheduler, so they are implemented as separate classes (i.e. WimaxScheduler/SS and WimaxScheduler/BS). The detailed parameters for the NS-2

[3]NO Ad-Hoc Routing Agent (NOAH): http://icapeople.epfl.ch/widmer/uwb/ns-2/noah/

[4]EMNTG Seamless and Secure Mobility: http://w3.antd.nist.gov/seamlessandsecure/download.html, last accessed 18 Nov. 2009



(a)



(b)

Fig. 10. WLAN-WiMAX simulation test scenario. (a) Network topology. (b) Mobility scenario.

TABLE III
NS-2 IEEE 802.16 PHYSICAL LAYER PARAMETERS

Parameter	Value
Freq_	2.4GHz
Pt_	0.025
RXThresh_	1.26562×10^{-13}
CSThresh_	1.012496×10^{-13}

TABLE IV
NS-2 IEEE 802.16 MAC LAYER PARAMETERS

Parameter	Value
dcd_interval_	5 secs
ucd_interval_	5 secs
Default modulation	OFDM_16QAM_3_4
t21_timeout_	0.02 secs
client_timeout_	50 secs

IEEE 802.16 model for physical layer and MAC layer are presented in Tables III and IV.

Table III includes the same parameters which are used in Table I. In Table IV, dcd_inteveral_is the Downlink Channel Descriptor massage interval, ucd_interval_represents the Uplink Channel Descriptor message interval, Default_modulation means the modulation method used, T21_timeout_is the DL-MAP message (WiMAX management message) timeout value and Client_timeout_is a timer value for removing client which does not communicate with base station.

B. Simulation Models, Setup, and Video Sequences

Modeling and simulations employ models for QAMMD, Predictive Buffering Algorithm (PBA) [25], a MSDVS-like [17] multiple TFRC connections-based approach (mTFRC) and a PROMISE-like [9] UDP-based multiple streaming solution (mUDP). QAMMD and PBA adopt a buffer estimation algorithm but they use different solutions. PBA uses a statistical approach which assumes the connections are not correlated. mTFRC and mUDP do not use buffer prediction. mTFRC uses adaptive data delivery based on TFRC protocol. mUDP uses equalized bandwidth allocation at the start of the streaming instead of dynamic bandwidth allocation which is used for the other solutions. In all approaches, the receiver requests the same packets to be delivered from the multiple senders. In addition, all approaches adopt static peer selection and initially connect to three nodes only.

In QAMMD, P_{target} is set to 0.005 and S_{VBV} is set to 224 kbytes which is determined at encoding time. The same S_{VBV} is used by mTFRC and mUPD, too. All approaches use 3.2 Mbps as the target bandwidth which is higher than the encoding rate of 3 Mbps in order to cover network delivery overhead.

Five five-minute long VBR encoded video sequences were selected from movies with different degrees of motion content: *"Die Hard 1"* - high, *"Jurassic Park 3"* - average, *"Don't Say A Word"* - average/low, *"Family Man"* - low and *"The Road To El Dorado"* (animated) - average/high. The clips were MPEG-2 encoded at 3 Mbps using the same frame rate (25 frames/sec) and the same IBBP frame pattern (12 frames/GOP). Traces were collected from these clips and used during simulations. In order to provide strong statistical reliability for the simulation tests, the results include several runs (up to 10 times for each test with different starting points in a movies).

C. Simulation Results and Analysis

1) WLAN Dumbbell Topology: The WLAN dumbbell topology as depicted in Fig. 9, is used for the tests in this section. The number of receiving nodes is limited to 4 since 17.4 Mbps and 13.2 Mbps are the maximum achievable throughputs using UDP and TCP over the WLAN dumbbell topology. The test includes simulation results when 10 different starting points are selected from each movie in order to achieve stronger statistical results. PSNR, buffer underflow and average waiting time for the WLAN dumbbell topology will be discussed in this section.

Fig. 11(a) shows a comparison between schemes in terms of estimated PSNR with increasing numbers of users. On average, when using QAMMD, PSNR is 60.9 dB, whereas when PBA, mTFRC and mUDP are employed PSNR is 51.0 dB, 38.1 dB and 44.8 dB, respectively. It can be seen how QAMMD behaves with 19.4% better than PBA, with 59.8% better than mTFRC and with 35.9% better than mUDP. Specifically, when the wireless bottleneck channel is crowded with 4 nodes, QAMMD offers 42.3% better perceived quality than PBA, 90.7% better per-

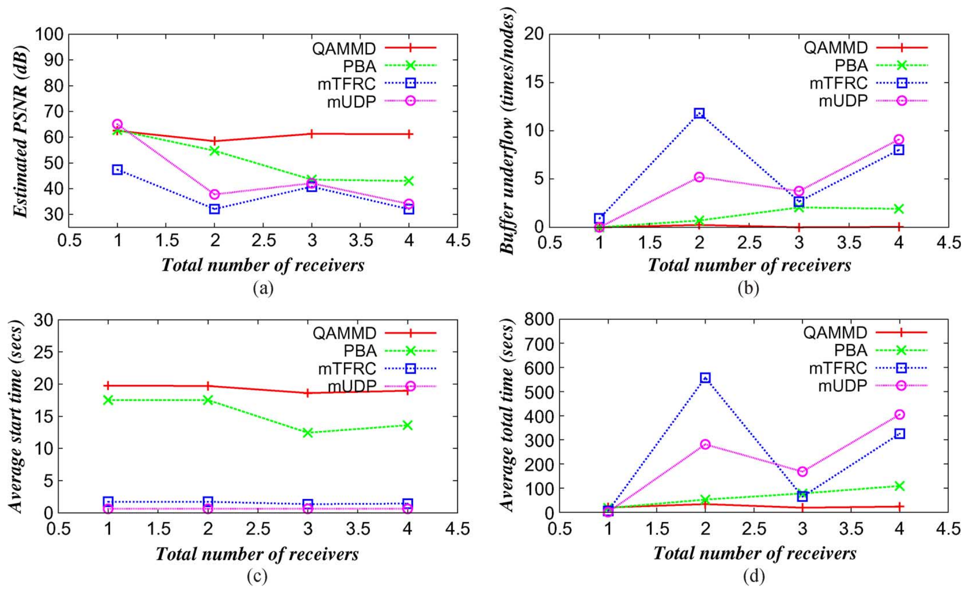


Fig. 11. WLAN test results comparison with various solutions. (a) Estimated PSNR. (b) Buffer underflow. (c) Average initial waiting time. (d) Average total waiting time.

ceived quality than mTFRC and 79.5% better than mUDP expressed in terms of PSNR.

Buffer underflow is compared in Fig. 11(b) between the schemes. On average, when using QAMMD, buffer underflow is 0.07, whereas when PBA, mTFRC and mUDP are employed buffer underflow is 1.17, 5.86 and 4.52, respectively. It can be seen how QAMMD has 94.0% less buffer underflow events than PBA. In addition, mTFRC and mUDP show almost 83.7 and 65.6 times more buffer underflow events than QAMMD, respectively.

Average initial waiting time and average total waiting time as overheads are presented in Fig. 11(c) and (d), respectively. On average, when using QAMMD, initial waiting time is 19.3 secs, whereas when PBA, mTFRC and mUDP are employed initial waiting time is 15.3 secs, 1.6 secs and 0.6 secs, respectively. However, total waiting time dramatically changes. When using QAMMD, the total waiting time is 23.8 secs, whereas when PBA, mTFRC and mUDP are employed total waiting time is 64.3 secs, 238.3 secs and 213.8 secs, respectively.

2) WLAN-WiMAX Dumbbell Topology: The WLAN-WiMAX dumbbell topology (See Fig. 10(a)) is used for the tests in this section. There are only up to 3 receiving nodes available because of lower bandwidth from WiMAX. The test includes simulation results of 10 different start points of each movie similar to wireless topology in order to have realistic statistics. In addition, the test scenario includes a simple mobility scenario which is presented in Fig. 10(b). Receivers start streaming within the coverage of WiMAX, move into the coverage of WLAN and go out to the WiMAX coverage again. The start time of the receiver movement varies from 12 secs to 22 secs in a uniform distribution. Similar to the wired dumbbell topology, PSNR, buffer underflow and average waiting time will be discussed in this section.

Fig. 12(a) shows a comparison between schemes in terms of estimated PSNR with increasing numbers of users. On average, when using QAMMD, PSNR is 84.2 dB, whereas when PBA, mTFRC and mUDP are employed PSNR is 70.8 dB, 56.2 dB and 57.0 dB, respectively. It can be seen how QAMMD behaves with 18.9% better than PBA, with 49.8% better than mTFRC and with 47.7% better than mUDP. Specifically, when the wireless bottleneck channel is crowded with 3 nodes, QAMMD offers 68.5% better perceived quality than PBA, 112.3% better perceived quality than mTFRC and 111.3% better than mUDP, expressed in terms of PSNR.

Buffer underflow is compared between the schemes in Fig. 12(b). On average, when using QAMMD, buffer underflow events are 0.03, whereas when PBA, mTFRC and mUDP are employed buffer underflow metric is 0.23, 2.57 and 2.99, respectively. It can be seen how QAMMD behaves with 87.0% better than PBA. In addition, mTFRC and mUDP show almost 85.7 and 99.7 times more buffer underflow events than QAMMD, respectively.

Average initial waiting time and average total waiting time as overheads are presented in Fig. 12(c) and (d), respectively. On average, when using QAMMD, initial waiting time is 17.3 secs, whereas when PBA, mTFRC and mUDP are employed initial waiting time is 6.9 secs, 0.7 secs and 0.6 secs, respectively. However, total waiting time dramatically changes again. When using QAMMD, total waiting time is 18.4 secs, whereas when mTFRC and mUDP are employed total waiting time is 123.4 secs and 150.0 secs, respectively. Only the case of PBA is shorter than QAMMD with an average of total waiting time of 17.0 secs. However, QAMMD shows shorter total waiting time for two and three nodes.

VI. Prototyping Architecture, Implementation Details, and User-Based Perceptual Testing

A. VLC Modification for Multiple-Source Streaming

The Video LAN Client (VLC) is an open-source multimedia player, freely available from the internet. It has been ported onto various platforms including Microsoft Windows, Linux,

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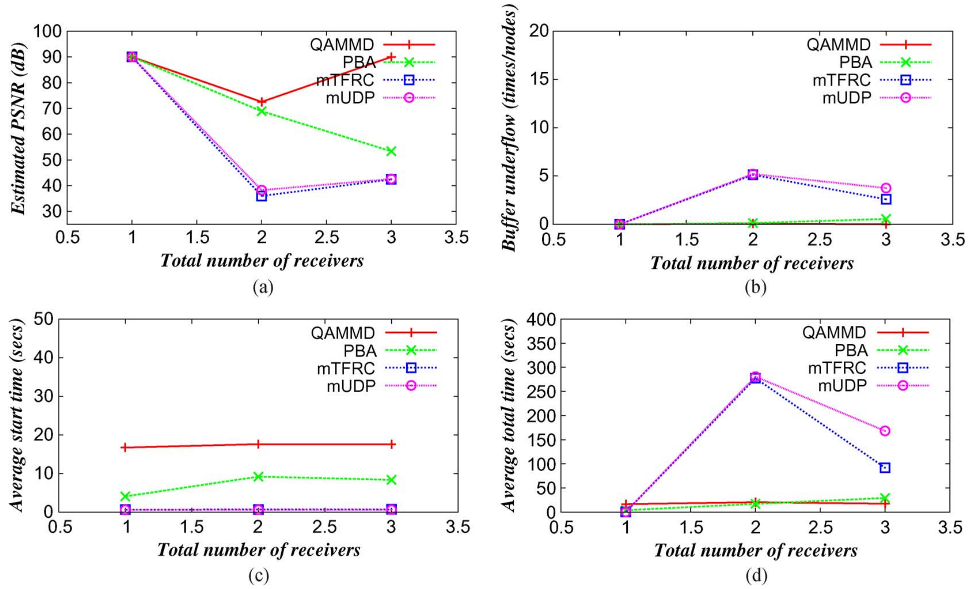

Fig. 12. WLAN-WiMAX test results comparison with various solutions. (a) Estimated PSNR. (b) Buffer underflow. (c) Average initial waiting time. (d) Average total waiting time.

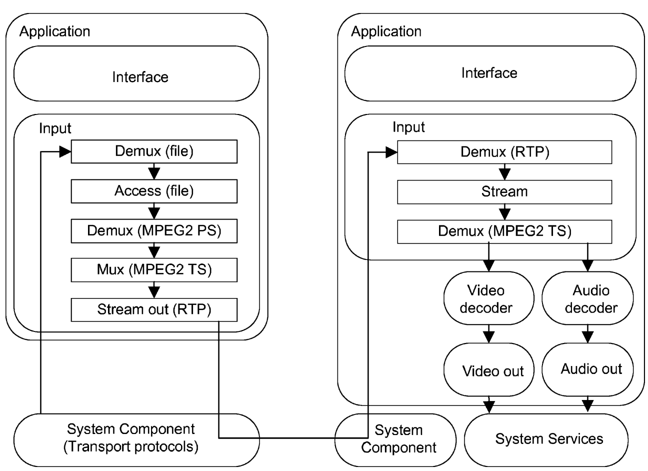

Fig. 13. VLC module chain for MPEG2 movie streaming.

etc., supporting various multimedia formats and streaming. The Smart PIN prototyping system uses VLC for servers and a client running on Linux. Currently, the VLC 0.9.8a version is used on a Debian variant Linux, Ubuntu[5] 8.10 (Linux kernel 2.6.27-11) for overall testing on Pentium 4 processor computers.

On the Linux platform, VLC supports DCCP as a transport protocol which includes the TFRC option (i.e. DCCP CCID-3) for congestion control. At the application level, VLC uses RTP in order to deliver multimedia data when DCCP is used. RTP could be delivered over UDP but the test procedures are different from DCCP.

VLC modules for streaming are presented in Fig. 13. When it is used for streaming, the MPEG2 file (MPEG2 PS format) is converted into MPEG2 Transport Stream (MPEG2 TS) format and is delivered over RTP to the client. When data reaches the client, the *RTP demux* extracts the MPEG stream and passes it to the *Stream* module.

[5]Ubunutu, http://www.ubuntu.com/, last accessed 18 Nov. 2009

The *Stream* module is implemented as a thread and communicates with the *RTP demux* through a FIFO queue. In the *Stream* thread, the *MPEG2 TS demux* module is used for demultiplexing of elementary streams such as MPEG video and audio. The proposed *Buffer Underflow Avoidance Scheme* (BUAS) is used in *RTP demux* and *Stream* thread since they have a queue between them.

Since VLC does not support multiple-source streaming, structure modifications include connection establishment, packet scheduling, and sender synchronization. These modifications are applied mainly in *Mux* in senders, *Demux* in the receiver, and the *Stream* module in senders and a receiver as depicted in Fig. 14. Although they mention that VLC supports MPEG2 PS streaming, the receiver implementation does not detect MPEG2 PS streaming through the network. In order to support this, the Stream module in the receiver also needs to be modified. In addition, a bandwidth limitation is also applied on the server side in order to achieve similar conditions for our simulation.

The modifications for *connection establishment* include multiple *Mux* creation based on the number of servers. In addition, receiver-based connection setup is used for multiple-source streaming. VLC establishes the connection between a sender and a receiver in different ways depending on the transport protocol. If a connection-based protocol such as TCP or DCCP is used, the receiver initiates a connection and starts the sender's data transmission. If a connection-less protocol (i.e. UDP) is used, the sender transmits data before the receiver starts to receive data. In case of multiple-source streaming, the receiver has to make sure the servers have started in order to synchronize data transmission from the multiple senders.

After the connection is set up, *data scheduling* is required for each of the servers which transmit data. In this paper, data is equally allocated among the servers. Packet numbering is required in order to support reassembling of streamed data, since there is no reordering mechanism between multiple connec-

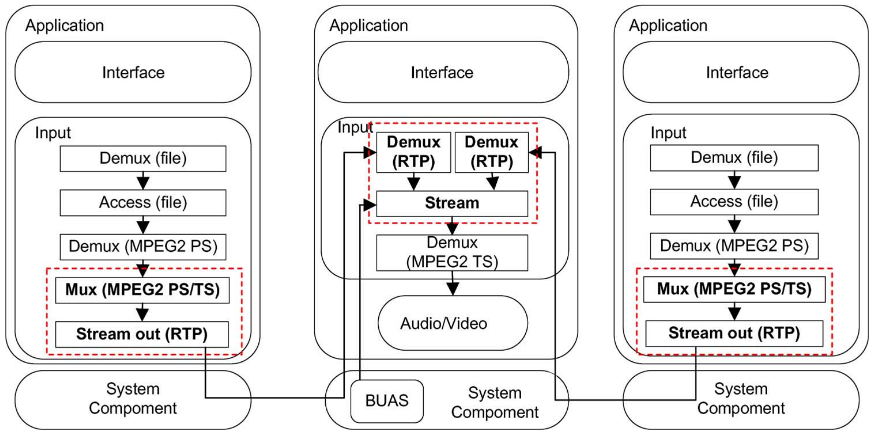

Fig. 14. VLC module chain for multiple-source based MPEG2 movie streaming.

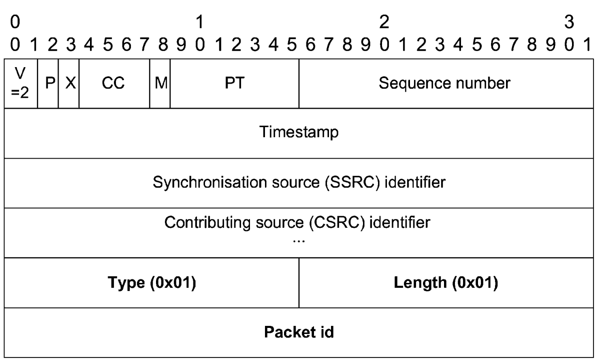

Fig. 15. Extension enabled RTP header $(X = 1)$.

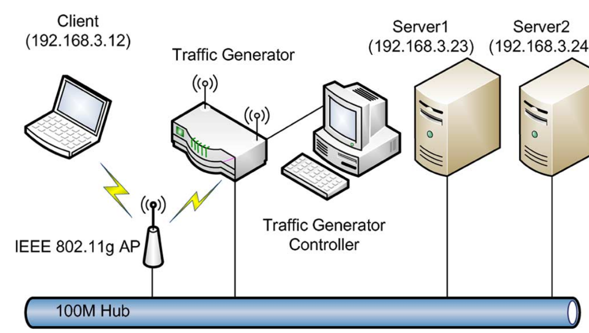

Fig. 16. Smart PIN prototyping test environment.

tions in standard RTP. The RTP header extension [37] is used for packet numbering. Since the extension bit is set, the fixed header is followed by exactly one header extension as presented in Fig. 15. There is a sequence number field in the RTP header, but that is only for single RTP connection in order to support detection of duplicated or lost packets. The header extension has a simple type, length and value (TLV) structure as shown below. There is no standardized type, so 0x01 is used for distinguishing the packet id. Since the length of packet id field is 4 bytes, the length is set as 1 word.

Sender synchronization is required due to the potential timing skew between multiple senders for multiplexing MPEG 2 audio and video components as MPEG2 does not support multiple-source situations. MPEG2 TS, which is designed for network delivery, is very dependent on timing; every packet includes a timestamp set by each sender according to its own clock. These timestamps are used in decoding. In order to reduce the time difference between senders and enable correct decoding, a solution is to have the receiver initiate synchronization between the multiple senders when the all connections have been established. However, as MPEG2 PS does not have the issues of MPEG2 TS, since multimedia component delivery can be considered as any normal file transfer from multiple senders, testing in this paper are based on MPEG2 PS. MPEG2 PS does not include an automatic solution to cope with delay jitter and out-of-order arrival of data. Buffering of data [38] and reordering based on sequence numbers solve these issues.

B. Test Setup Description

The test setup consists of two different configurations for network tests and user perceptual tests, respectively. The network test configuration is based on a wireless network emulation and includes a traffic generator. The user perceptual test configuration uses test settings for user perceived quality and delivers the movie clips recorded from the network tests.

1) Network Test Setup: The network test configuration includes a conventional network environment connected through WLAN and wired technologies. A LANforge Traffic Generator[6] is used for emulating a real network environment. The test topology is illustrated in Fig. 16 and it is emulating the same topology as used in the NS-2 simulation which is based on WLAN. The traffic generator delivers background traffic through wired and wireless environments. The background traffic consists of a number of TCP and UDP flows which change bitrates randomly from 800 Kbps to 1 Mbps. The wireless client PC is equipped with a NETGEAR WG311T wireless card[7] which supports IEEE 802.11g.

Using the configuration from Fig. 16, servers stream to the client through RTP using different transport protocols such as UDP and DCCP. For our real-life testing, three out of five movies from the simulations are used: "*Die Hard 1*" (DH) - with very high motion content, "*Don't Say A Word*" (DS), with average - low

[6]LANforge traffic generator, http://www.candelatech.com/, last accessed 18 Nov. 2009

[7]http://www.netgear.com/Products/Adapters/SuperGWirelessAdapters/WG311T.aspx, last accessed 18 Nov. 2009

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TABLE V
QUALITY SCALE FOR SUBJECTIVE TESTING (ITU-T R P.911)

Rating	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible, not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

motion content and "*The Road To El Dorado*" (RT) - high motion animated video. For each movie clip we use a clip of 3 minutes duration for multimedia streaming. The transmitted videos are saved for user testing with different testing environments such as streaming approaches and network conditions.

2) User Perceived Test Setup: Although quite well established, PSNR is not a standard method for measuring the assessment of user-perceived quality. Subjective testing is normally used in order to confirm the results of objective testing expressed in terms of PSNR. User perceived quality can be level of information assimilation and user satisfaction [39]. Specifically, the ITU standards ITU-T P.911 [40], P.910 [41], ITU-R BT-500 [42], etc., are commonly used for measuring subjective quality as a part of user satisfaction. However, ITU-T P.911 is best for assessing quality in multimedia systems dealing with both video and audio.

Subjective user testing was performed in order to assess multimedia perceptual quality based on the ITU-T R. P.911 [40] recommendations. This standard uses the Mean Opinion Score (MOS) as quality metric for the answers to the questions the subjects are asked to respond to. The scores are expressed on the 5-point quality scale as shown in Table V.

There are several criticisms of the ITU-T recommendation for perceived user quality tests that can be found in the literature [43]. In order to comment on those points, the user tests in this paper include following features:

- ITU P.911 tests require users to watch short sequences of approximately 10 seconds in duration. This is too short for a user to assess the quality of multimedia so in this paper, 30 second clips are used instead.
- The judgments from users are mainly based on the picture quality, but in reality, both audio and video are related to the multimedia environment tested. In order to overcome this limitation, the tests include clips with synchronized video and audio.
- The test includes only delivered sequences which are degraded. However, the results based on the difference between the degraded and original sequences could be significant. Therefore the tests include the original clips.
- Perceptual tests in general do not capture change of perception about the quality that users may have during communication under varying network conditions. In order to improve this, our tests include clips delivered through different network conditions.

VLC for Microsoft Windows is the player used in the perceptual tests. Test scripts were written using the Perl script language.[8] The user test room setting included four PC sets with the same monitor types and resolutions for subjective testing. All the monitors were calibrated using PANTONE huey.[9] The test room is shielded from natural light in order to control noise level and to maintain a constant luminance level. A maximum of four people could attend the test at the same time. They were informed that they cannot have any discussions or move equipment and they should keep a fixed distance from the monitor. This test setup conforms to the standard recommendations as indicated in [40].

One session of user tests consisted of 23 phases which included short instructions for the flow of the test phase and video for the blind assessment test. Each phase of the test included three different displays as shown below. The total duration of one phase was about 50 seconds.

- Phase title: 5 seconds (e.g. Die Hard 1. Please note clip code (DH) on the questionnaire sheet. Video starts in 5 seconds)
- Movie playing: 30 seconds
- Assessment direction: 15 seconds. 3 questions for each clip. (e.g. please note down your answer on the questionnaire sheet.)

As shown in Table VI, 30 different movie clips were recorded over the network testbed in Fig. 16. These clips included 3 different movies which were DH, DS and RT. In addition, 5 streaming schemes were applied. Two different network conditions included low network load and high network load. The low network load included 3 TCPs and 1 UDP connection and the high network load includes 4 TCPs and 2 UDPs connections. For fair comparison of results, the user perceptual tests make use of the same solutions the simulations tests considered, namely Quality Adaptive Multiple-source Multimedia Delivery (QAMMD), Predictive Buffer Algorithm (PBA) [25], a MSDVS-like [17] multiple TFRC connections-based approach (mTFRC) and a PROMISE-like [9] UDP-based multiple streaming solution (mUDP). A single-source based approach (Single) was also used over the TFRC protocol, in order to compare with multiple-source streaming approaches. More details about these schemes are provided in Section V-B.

A session of user perceptual tests included only 23 movie clips in order to keep the test time to less than 30 minutes. These movie clips included three original movie clips as a preamble. Before starting to view the delivered movie clips, these provide users with some idea of the quality of the original movie clips. The other clips are 20 movie clips from Table VI. The total test duration was about 25 minutes. There were three different versions of the test set where each started from a different clip which differs in terms of content and type and included 20 movie clips. For example, test set 1 involved the clips with IDs from 01 to 20, test set 2 - clips with IDs from 11 to 31, and test set 3 - clips with IDs 21 to 31 and 01 to 10.

C. Network Performance Test and Result Analysis

Network test results include Peak Signal to Noise Ratio (PSNR) and initial waiting time. In order to measure PSNR, the MSU Video Quality Measurement Tool[10] is used. In order

[8]The Perl Directory - perl.org, http://www.perl.org/, last accessed 18 Nov. 2009

[9]Monitor & Printer Profiling - PANTONE huey, http://www.pantone.com/pages/products/product.aspx?pid=79, last accessed 18 Nov. 2009

[10]MSU Video Quality Measurement Tool (PSNR, MSE, VQM, SSIM), http://compression.ru/video/quality_measure/video_measurement_tool_en.html, last accessed 18 Nov. 2009

TABLE VI
MOVIE CLIP NUMBERING

Clip id	Approach	Movie	Traffic	Clip id	Approach	Movie	Traffic
01	Single	DH	Low	16	Single	DH	High
02	mUDP	DS	High	17	mUDP	DS	Low
03	mTFRC	RT	Low	18	mTFRC	RT	High
04	PBA	DH	High	19	PBA	DH	Low
05	QAMMD	DS	Low	20	QAMMD	DS	High
06	Single	RT	High	21	Single	RT	Low
07	mUDP	DH	Low	22	mUDP	DH	High
08	mTFRC	DS	High	23	mTFRC	DS	Low
09	PBA	RT	Low	24	PBA	RT	High
10	QAMMD	DH	High	25	QAMMD	DH	Low
11	Single	DS	Low	26	Single	DS	High
12	mUDP	RT	High	27	mUDP	RT	Low
13	mTFRC	DH	Low	28	mTFRC	DH	High
14	PBA	DS	High	29	PBA	DS	Low
15	QAMMD	RT	Low	30	QAMMD	RT	High

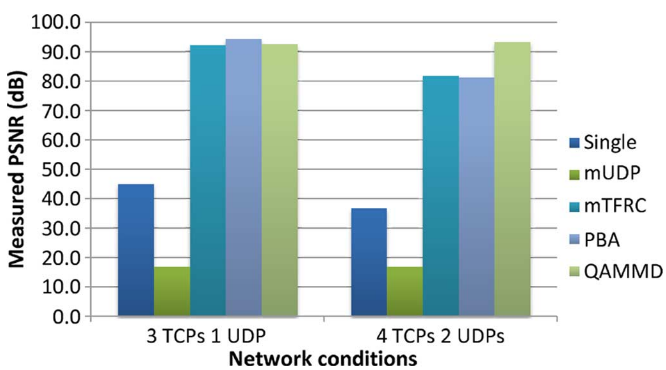

Fig. 17. Measured PSNR statistics chart.

TABLE VII
INITIAL WAITING TIME STATISTICS (IN SECONDS)

		PBA	QAMMD
3 TCPs 1 UDP	DH	40.8	46.1
	DS	42.5	46.2
	RT	38.3	44.3
	Average	40.6	45.6
4 TCPs 2 UDPs	DH	42.3	46.1
	DS	41.0	43.8
	RT	37.7	41.9
	Average	41.0	44.7
Total average		40.5	45.1

to reduce synchronization issue, the first 10 seconds of PSNR measurements are used in this paper. The total waiting time is not considered for prototyping tests in this paper since VLC does not support re-buffering. In total, 10 tests were performed for every scheme and movie clip.

Fig. 17 shows a comparison between schemes in terms of PSNR under different network conditions. The low traffic condition has 4 background connections including 3 TCPs and 1 UDP. The high traffic condition has 6 background connections including 4 TCPs and 2 UDPs. Typical bitrates for TCP and UDP flows are between 800 Kbps and 1 Mbps.

On average, the TFRC-based approaches such as QAMMD, mTFRC and PBA show high PSNR values. QAMMD usage results in 92.9 dB whereas mTFRC and PBA - in 87.0 dB and 87.8 dB, respectively. It can be seen how QAMMD behaves with 5.8% better than PBA, with 6.8% better than mTFRC. However, very low PSNR values are obtained (41.0 dB and 16.8 dB) when Single and mUDP are employed. When high traffic is found, QAMMD shows greater benefit than other approaches. Using QAMMD results in 93.3 dB whereas employing mTFRC and PBA determines 81.3 dB and 81.8 dB, respectively. It can be seen how QAMMD behaves with 14.1% better than PBA, with 14.8% better than mTFRC. In the case of low traffic, TFRC-based approaches such as QAMMD, mTFRC and PBA show similar PSNR values. QAMMD has 92.6 dB whereas mTFRC and PBA have 94.3 dB and 92.3 dB, respectively.

The average initial waiting time is presented in Table VII. PBA and QAMMD are compared since only those adopt some kind of buffering algorithms. On average, when using QAMMD, the initial waiting time is 44.7 seconds, whereas when PBA is employed the initial waiting time is 41.0 seconds.

D. User Perceived Quality Test and Result Analysis

The user perceptual test includes three questions which are related to quality, continuity and synchronization between the video and audio from a multimedia clip. The continuity of the video is adopted for another aspect of the quality of the movie. Since there are multiple-source approaches, user perception of synchronization is also included as one of the questions. Based on ITU-T R. P.911 [40], Mean Opinion Score (MOS) is used. The following tables present the results from 30 users who are on average 27.5 year old. The test results include a long duration for the assessment since each phase has 30 seconds of playing time.

Fig. 18 shows a comparison between schemes in terms of MOS on the quality of multimedia clips with different network conditions. On average, when using QAMMD, MOS is 3.7, whereas when Single, mUDP, mTFRC and PBA are employed, MOS values are 3.1, 1.4, 2.6 and 3.3, respectively. It can be seen how QAMMD behaves with 12.1% better performance than PBA, with 42.3% better performance than mTFRC and with 19.4% better performance than Single. When compared to mUDP, QAMMD has 1.6 times higher MOS. In addition, QAMMD has higher MOS than PBA as star t-test ($t = 1.76, d.f. = 5, p < 0.07$). Specifically, when the wireless

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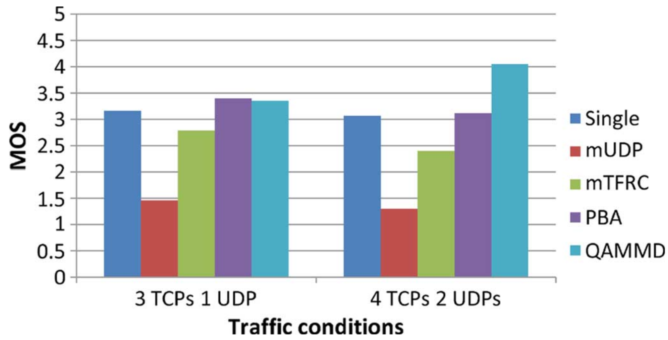

Fig. 18. MOS for quality of multimedia clip.

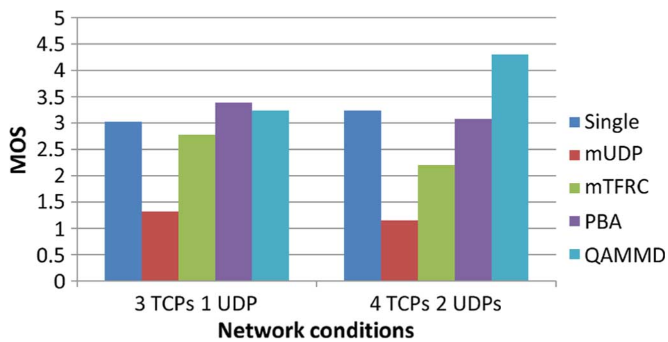

Fig. 19. MOS for continuity of video.

bottleneck channel becomes crowded with 4 TCPs and 2 UDPs, QAMMD offers 32.3% better perceived quality than PBA, 70.8% better perceived quality than mTFRC and 32.3% better than Single, as expressed in terms of MOS. In comparison with mUDP, QAMMD shows about 2.2 times higher MOS.

Fig. 19 shows a comparison between schemes in terms of MOS on the video continuity of the multimedia clips under different network conditions. The overall results are similar to MOS for quality. On average, when using QAMMD, MOS is 3.8, whereas when Single, mUDP, mTFRC and PBA are employed, MOS is 3.1, 1.2, 2.5 and 3.2, respectively. It can be seen that QAMMD behaves with 18.8% better than PBA, with 52.0% better than mTFRC and with 22.6% better than Single in terms of performance. In comparison with mUDP, QAMMD results in about 2.2 times better MOS. In addition, QAMMD has higher MOS than PBA, result confirmed by a t-test with 99% confidence $(t = 1.55, d.f. = 5, p < 0.1)$. Specifically, when the wireless bottleneck channel is loaded with 4 TCPs and 2 UDPs, QAMMD offers 57.1% better perceived quality than PBA, 95.5% better perceived quality than mTFRC and 34.4% better than Single, as expressed in terms of MOS for quality. In comparison with mUDP, QAMMD shows about 2.6 times better MOS.

Fig. 20 compares various schemes in terms of MOS on the synchronization between video and audio components of the multimedia clips delivered under different network conditions. The overall result is yet again similar to that for MOS for quality, although multiple-source streaming is involved. On average, when using QAMMD, MOS is 4.0, whereas when Single, mUDP, mTFRC and PBA are employed, MOS is 3.7, 1.9, 3.2 and 3.8, respectively. Once again it can be seen that QAMMD behaves with 5.3% better than PBA, with 25.0%

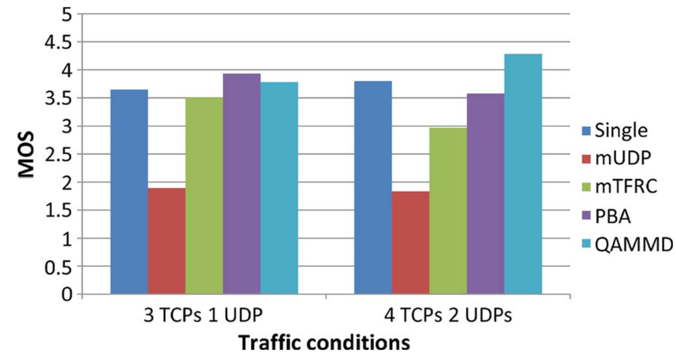

Fig. 20. MOS for synchronization between video and audio.

better than mTFRC and with 8.1% better than Single. In comparison with mUDP, QAMMD behaves about 1.1 times better. In addition, statistically it can be said that QAMMD has higher MOS than PBA as a t-test confirmed with 87% confidence $(t = 1.30, d.f. = 5, p < 0.13)$. Specifically, when the wireless bottleneck channel is loaded with 4 TCPs and 2 UDPs, QAMMD offers 19.4% better perceived quality than PBA, 43.3% better perceived quality than mTFRC and 13.2% better than Single expressed in terms of MOS for quality. In comparison with mUDP, QAMMD shows about 1.4 times better MOS.

VII. Conclusion

This paper introduced a novel *Quality-oriented Algorithm for Multiple-source Multimedia Delivery* (QAMMD) which employs a double virtual receiving buffer architecture to maintain quality at high levels in highly-loaded network conditions during multimedia delivery. QAMMD also employs a buffer underflow avoidance scheme (BUAS) which optimally balances the flow of data between the multiple connections and a play buffer in order to achieve high multimedia quality without content adaptation to network conditions.

In terms of simulation, QAMMD is compared with a multiple TFRC connection-based scheme (mTFRC) and a multiple UDP connection-based scheme (mUDP). In addition, a similar buffer estimation-oriented algorithm, the Predictive Buffer Algorithm (PBA) is used in simulations. Performance is evaluated for multiple-source streaming approaches in terms of PSNR and overhead measurements. QAMMD shows better performance in terms of estimated PSNR, buffer underflow and total waiting time using different wireless network topologies. The prototyping test is mainly focused on QAMMD using comparison-based tests. The metrics used include Peak Signal to Noise Ratio (PSNR) as an objective metric and Mean Opinion Score (MOS) as a subjective metric for quality of multimedia delivered to the user. In terms of PSNR, QAMMD shows better performance in comparison to other schemes, especially for cases where there is more traffic over the wireless network. Similar to the simulation tests, QAMMD also shows better performance in user perceptual tests when compared to other similar approaches.

An extended work could focus on the optimization of QAMMD. QAMMD includes an initial delay for buffer estimation. After starting the playout, adaptive control of playback can improve the quality of service by adjustment of time duration between frames [26]. Since standard MIH is assumed, some handover approaches [44] can further improve the performance. Last, but not least, testing with more complex topologies which more closely resemble the real Internet and subjective assessment in a real-life environment [45] are considered as future works.

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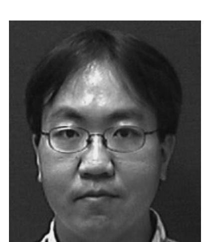

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