ISSN: 2278-0181
ICACT - 2016 Conference Proceedings

Distortion Aware Concurrent Multipath Transfer for Mobile Video Streaming using Network

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***Abstract*— Stream Control Transmission Protocol (SCTP) is the standard transport-layer solution to enable CMT in multi-homed communication environments. However, delivering high-quality streaming video with the existing CMT solutions still remains problematic due to the stringent QoS (Quality of Service) requirements and path asymmetry in heterogeneous wireless networks. In this paper, we advance the state of the art by introducing video distortion into the decision process of multipath data transfer. The proposed Distortion-Aware Concurrent Multipath Transfer (CMT-DA) solution includes three phases: 1) per-path status estimation and congestion control; 2) quality-optimal video flow rate allocation; 3) delay and loss controlled data retransmission. The term 'flow rate allocation' indicates dynamically picking appropriate access networks and assigning the transmission rates. We analytically formulate the data distribution over multiple communication paths to minimize the end-to-end video distortion and derive the solution based on the utility maximization theory. The performance of the proposed CMT-DA is evaluated through extensive semi-physical emulations in Exata involving H.264 video streaming. Experimental results show that CMT-DA outperforms the reference schemes in terms of video PSNR (Peak Signal-to-Noise Ratio), good put, and inter-packet delay.**

Key words—Distortion awareness, concurrent multipath transfer, mobile video streaming, heterogeneous wireless networks, SCTP, multihoming.

I. INTRODUCTION

During the past few years, mobile video streaming service (e.g., YouTube [1], Hulu [2], online gaming, etc.) has become one of the "killer applications" and
the video traffic headed for hand-held devices has experienced explosive growth.

On one hand, the Wi-Fi networks are limited in radio coverage and mobility support for individual users. On the other hand, the cellular networks can well sustain the user mobility but their bandwidth is often inadequate to support the throughput-demanding video applications. Although the 4G LTE and WiMAX can provide higher peak data rate and extended coverage, the available capacity will still be insufficient compared to the ever-growing video data traffic.The latest market research conducted by Cisco company indicates that video streaming accounts for 53% of the mobile Internet traffic in 2013 and will reach 69% by the year 2018.In parallel, global mobile data is expected to increase 11-fold in the next five years. Another on-going trend feeding this tremendous growth is the popularity of powerful mobileTerminals (e.g., smart phones and iPad), which facilitate individual users to access the internet and watch videofrom everywhere.Despite the rapid advancements in network infrastructures, it is still challenging to deliver high-quality streaming video over wireless platforms.

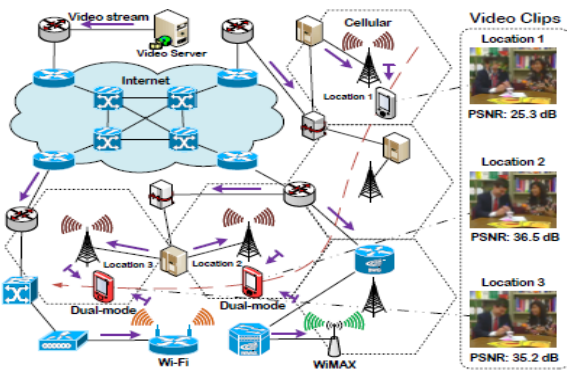


Fig-1 Exploiting concurrent multipath transfer for enhanced streaming video quality in heterogeneous WirelessNetworks with multi-homed clients.

II. RELATED WORKS

The related work to this paper can be generally classified into two categories:
(1) Concurrent multipath transfer,
(2) Cooperative video delivery in heterogeneous
Wireless networks.

A. Concurrent multipath transfer

It has three negative effects are:

1) Unnecessaryfast retransmissions; 2) overly conservative congestionwindowgrowth;3)increased acknowledgment traffic.A CMTsolution with a potentiallyfailed.A path that experiences a single timeout is marked asa "potentially failed" (PF), indicating the degrees of its communication reliability.Although the path status is an important factor that affects the scheduling policy, theapplication requirements should also be considered to guarantee the QoS. Basically, the proposed CMT-DA is different from the CMTQA as we take the video distortion as the

ISSN: 2278-0181
ICACT - 2016 Conference Proceedings

benchmark. Still, the proposed solutions (path status estimation, flow rate allocation, and retransmission control) in CMT-DA are significantly different from those in CMT-QA.

B. Cooperative Video Delivery in Heterogeneous Wireless Networks

The existing studies in this field can be divided into:
1) rate allocation policies2) packet scheduling approaches 3) FEC coding Schemes study the distributed rate allocation policies for multihomed video streaming over heterogeneous wireless networks and conclude that the media aware rate allocation policies outperform the heuristic AIMD-based schemes in improving video quality. The Earliest Delivery Path First (EDPF) algorithm takes into account the available bandwidth, propagation delay and video frame size for estimating the arrival time and aims to find an earliest path for delivering the video packet. We propose a novel Sub-Frame Level (SFL) scheduling approach, which deliberately splits the large-size video frames to optimize the delay performance of high definition video streaming.

III. MODEL AND PROBLEM STATEMENT

The transmission of a single video flow using the SCTP association forms a source node to a multi-homed client as shown in the figure 2.The encoded video data is divided into several chunks and dispatched onto different paths. The major components at the sender side are the parameter control unit, flow rate allocator, and retransmission controller. The retransmission controller leverages the Explicit Congestion Notification (ECN) to differentiate between network congestion and wireless channel errors.The flow rate allocation is the critical step in the scheduling procedure. This problem involves the models of communication path and video distortion.

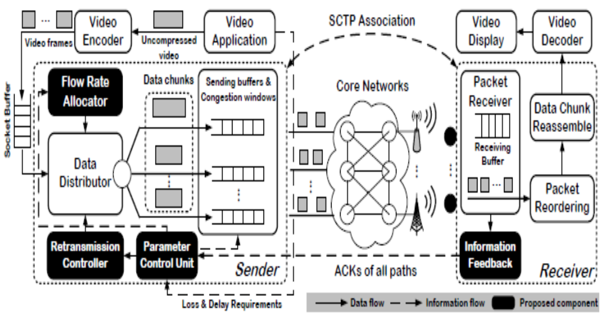


Fig-2 System overview of the proposed CMT-DA solution.

A. Communication Path Model

We consider a heterogeneous wireless network integrating P communication paths between twotransmission ends.The end-to-end connection can be constructed by binding a pair of IP addresses from the source and destination nodes, respectively. It is well-known that wireless access link is most likely to be the bottleneck for end-to-end transmission due to the limited bandwidth and time-varying channel status. Each communication path p 2 P is considered to be an independent transport link uncorrelated with others and is characterized by theFollowing properties.
1)The available bandwidth μp. This metric does not indicate the raw per-path capacity, but the time varying share of that bandwidth as perceived by the end-to-end flow.
2).The round trip time RTTp, which represents the length of time it takes for a data packet to be sent plus the delay it takes for an acknowledgment of that packet to be received.
3).The path loss rate, assumed to be an independent and identically distributed(i.i.d) process, uncorrelated with the input video streaming rate.

B. Video Distortion Model

The end-to-end distortion(Dtotal) perceived by the end user can be generally computed as the sum of the source distortion (Dsrc) and the channel distortion (Dchl). Overall, the end-to-end distortion can thus be written as

Dtotal = Dsrc + Dchl

The video quality depends on both the distortion due to a lossy encoding of the media information, and the distortion due to losses experienced in the network. Dsrcis mostly determined by the video source rate (R) and the video sequence parameters (e.g., for the same encoding bit rate, the more complex the sequence, the higher the source distortion). The source distortion decays with increasing encoding rate. The decay is quite steep for low bit rate values, but it becomes very slow at high bit rate. The channel distortion is dependent on the effective loss rate, which is caused by the transmission loss and expired arrivals of video packets. It can be computed as the average of the loss probabilities of all the communicationpaths. Hence, we can explicitly formulate Dtotal (expressed in units of Mean Square Error, MSE) as:constants for a specificvideo codec and video sequence. These parameters can be estimated from three or more trial encodings using nonlinear regression techniques. To allow fast adaptationof the flow rate allocation to abrupt changes in the video content, these parameters can be updated for each Groupof Pictures (GoP) in the encoded video sequence.The encoding parameters (e.g., the frame structure, GoPsize, etc.) also affect the source and channel distortion.
But they are not used as control parameters as the proposedCMT-DA is a transport-layer protocol. Since thismodel takes into account the effects of intra coding andspatial loop filtering, it provides accurate estimations forend-to-end distortion. Terminals (e.g., smart phones and iPad), which facilitate Individual users to access the Internet and watch video from everywhere.
Despite the rapid advancements in network infrastructures,it is still challenging to deliver high-quality streaming video over wireless platforms .

$$D_{\text{total}} = D_0 + \frac{\alpha}{R - R_0} + \beta \cdot \Pi,$$

C. Derivation of Effective Loss Rate

The effective loss rate represents the combined probability of transmission and overdue losses.
We provide derivations based on continuous time Markov chain and Gilbert loss model.

a. Transmission Loss Rate

The chunks will be fragmented into packets when transmitted over each communication path.

b. Overdue Loss Rate

The overdue loss rate represents the ratio of packets arriving at the destination out of the deadline and is determined by the end-to-end delay.

IV PROPOSED CMT-DA SOLUTIONS

A. Path Status Estimation and Congestion Control

The communication path model includes the status of available bandwidth, round trip time, and path loss rate. The path loss rates are updated once an ACK is received on any path. It represents the ratio of successfully delivered packets at the destination to the total number of packets dispatched onto path p. The per-path congestion control is that it allows the ACK packets to be sent back through any uplink path, i.e., not necessarily the path on which the last packet is received by the destination. The receiver sends the ACK on a most reliable uplink path and this reduces the probability of dropped/overdue feedback packets. The purpose for picking the most reliable uplink channel is also to reduce the round trip time. A path congestion controller wills timeout if no feedback is received for a period RTOp.Therefore, the timeouts are handled separately for each path.

B. Flow Rate Allocation

The piecewise linear (PWL) approximation to derive the potential rate allocation vectors tominimize. The end-to-end video distortion. Then, we use the utility based rate allocation function to minimize the video distortion while alleviating load imbalance. As Dtotal is dependent on the sum of the transmission and overdue loss rate over each communication path, a PWL approximation can be obtained based on a univariate function. It can be achieved by dividing the interest region of each univariate function into a sufficient number of non-overlapped small intervals. To minimize the end-to-end video distortion, the rate allocation algorithm is inclined to assign loads to the communication paths with higher quality.

$$L_p = \frac{\mu_p \cdot (1 - \pi_p) - R_p}{\left(\sum_{p \in \mathcal{P}} \mu_p \cdot (1 - \pi_p) - \sum_{p \in \mathcal{P}} R_p\right) / P}$$

Algorithm 1 Utility maximization based flow rate allocation

Require: $\{RTT_p, \mu_p, \pi_p\}_{p \in \mathcal{P}}, \mathcal{T}, \Delta$;
Ensure: $\mathcal{R} = \{R_p\}_{p \in \mathcal{P}}$;
for each path p in $\mathcal{P}$ **do**
  $U_p \Leftarrow \frac{\varphi(R_p + \Delta R_p) - \varphi(R_p)}{\Delta R_p}$;
  $L_p \Leftarrow \frac{\mu_p \cdot (1 - \pi_p) - \sum_{f \in \mathcal{F}} R_p}{(\sum_{p \in \mathcal{P}} \mu_p \cdot (1 - \pi_p) - \sum_{p \in \mathcal{P}} R_p)/P}$;
  $\Delta R_p \Leftarrow \Delta R_p / U_p$;
  $R_p \Leftarrow R_p + \Delta R_p$;
  Update the approximate function $\varphi(R_p)$;
end for
$\overline{\mathcal{U}} \Leftarrow \arg\max_{\mathcal{R}} \{\mathcal{U}\}$;
if $L_p \leq$ TLV **then**
  $R_p \Leftarrow R_p + \Delta R_p$ //Intra-path allocation;
  Update free resources of path p;
else
  find other path that can transfer part of its assigned rate to path $p' \neq p \in \mathcal{P}$ with maximum transition utility improvement $\Delta\mathcal{U}$ //Inter-path allocation;
  if $\Delta\mathcal{U} > 0$ **then**
    $R_p \Leftarrow R_p + \Delta R_p$;
    Update free resources of path p and p';
  end if
end if

C. Data Retransmission Control

The data retransmission is necessary to satisfy the imposed loss requirements for achieving acceptable video quality and SCTP standard defines two retransmission schemes: fast retransmission and timeout retransmission.The retransmission process is activated when packet loss occurs in one path, either detected by the SACKs on gap report or after a retransmission timeout without acknowledgment. The standard CMT retransmission policy has no mechanism to distinguish random losses from congestion, and therefore treats all losses as congestion induced. In the context of heterogeneous wireless networks, thePacket losses can be classified into three categories:1) congestion loss caused by bandwidth limitations or router buffer overflows; 2) packet errors due to external interference or wireless channel fading; and 3) link failure or handover loss.

Algorithm 2 Delay and loss controlled data retransmission

```
if ECN_p == true then
  if the congestion controller expires after RTO_p then
    ssthresh_p = max(cwnd_p/2, 3 × MTU);
    cwnd_p = MTU;
  end if
  if received 3 dup-SACKs then
    ssthresh_p = max(cwnd_p/2, 3 × MTU);
    cwnd_p = ssthresh_p;
  end if
  if recorded Π' > Δ then
    p' = arg min_{p∈P} {E{D_p(S_p)}};
    if E{D_p'(S_p)} < T then
      Retransmit the lost data chunk through p';
    end if
  end if
end if
```

V. PROBLEM

The expected effective loss rate of all the paths can be estimated with and the channel distortion can be expressed as

$$D_{\text{chl}} = \beta \cdot \frac{\sum_{p\in\mathcal{P}} R_p \cdot \{\pi_p^* + (1-\pi_p^*) \cdot \mathbb{P}\{D_p > \mathcal{T}\}\}}{\sum_{p\in\mathcal{P}} R_p}.$$

In conjunction with the source distortion, we can obtain the end-to-end video distortion as follows

$$D_{\text{total}}(\mathcal{R}) = D_0 + \frac{\alpha}{R - R_o} + \beta \cdot \frac{\sum_{p\in\mathcal{P}} R_p \cdot \Pi_p}{\sum_{p\in\mathcal{P}} R_p}$$

To reduce the probability of out-of-order packet arrivals, the scheduling policy aims at minimizing the delay jitters of different paths.

For each data distribution interval,
Determine the values of

$$\mathcal{R} = \{R_p\}_{p\in\mathcal{P}}$$

To minimize:

$$D_{\text{total}}(\mathcal{R}) = D_0 + \frac{\alpha}{R - R_o} + \beta \cdot \frac{\sum_{p\in\mathcal{P}} R_p \cdot \Pi_p}{\sum_{p\in\mathcal{P}} R_p}$$

$$\text{s. t.} \begin{cases} \Pi_p = \pi_p^* + (1-\pi_p^*) \cdot \mathbb{P}\{D_p > \mathcal{T}\}, p \in \mathcal{P}, \\ \pi_p^* = \text{Equation (11)}, \\ \mathbb{P}\{D_p > \mathcal{T}\} = \text{Equation (17)}, \\ \mathbb{E}\{D_p\} + \left(\left\lceil \frac{S_p}{MTU} \right\rceil - 1\right) \cdot \omega_p \leq \mathcal{T}, p \in \mathcal{P}, \\ R_p \leq \mu_p, p \in \mathcal{P}, \\ \mathbb{E}\{D_p\} = \mathbb{E}\{D_{p'}\}, p' \neq p, \{p, p'\} \in \mathcal{P}. \end{cases}$$

To obtain the close-to-optimal result with fast convergence for efficient online operation, we propose a progressive flow rate allocation algorithm to solve the optimization problem based on the utility maximization theory.

Table 1: Basic notations used throughout this paper

Symbol	Definition
$\mathbb{P}, \mathbb{E}$	the probability value, expectation value.
$\mathcal{P}, p$	the set of available paths, a path element.
P	the number of available paths.
$\mathcal{R}, R_p$	the flow rate allocation vector, an element.
$\mathcal{T}, \Delta$	the delay, loss requirement.
RTT_p	the round trip time of p.
μ_p	the available bandwidth of p.
ν_p	the residual bandwidth of p.
π_p^B	the path loss rate of p.
Π_p^*	the effective loss rate over p.
π_p^*	the transmission loss rate over p.
G/B	the Good/Bad state of p.
ξ_p^G	the state transition probability of p from B to G.
n, n_p	the total number of packets, dispatched onto p.
D_p	the end-to-end delay over path p.
$\mathcal{U}, U_p$	the system utility matrix, an element.

Table 2: Parameter configurations of wireless network

Cellular parameter	Value
Target SIR value	10 dB
Orthogonality factor	0.4
Common control channel power	33 dB
Maximum power of BS	43 dB
Total cell bandwidth	3.84 Mc/s
Inter/intra cell interference ratio	0.55
Background noise power	-106 dB
available capacity	300 Kbps
average loss rate	2%
average burst length	10 ms
WiMAX parameter	**Value**
System bandwidth	7 MHz
Number of carriers	256
Sampling factor	8/7
Average SNR	15 dB
Symbol duration	2048
available capacity	1200 Kbps
average loss rate	4%
average burst length	15 ms
WLAN parameter	**Value**
Average channel bit rate	2 Mbps
Slot time	10 μs
Maximum contention window	32
available capacity	500 Kbps
average loss rate	6%
average burst length	20 ms

VI. PERFORMANCE EVALUATIONS

The evaluation methodology that includesthe emulation setup, performancemetrics, reference schemes, and emulation scenario.

A. *Evaluation Methodology*

a. Emulation Setup

The architecture of evaluation system is presented in Fig and the main configurations are set as follows.

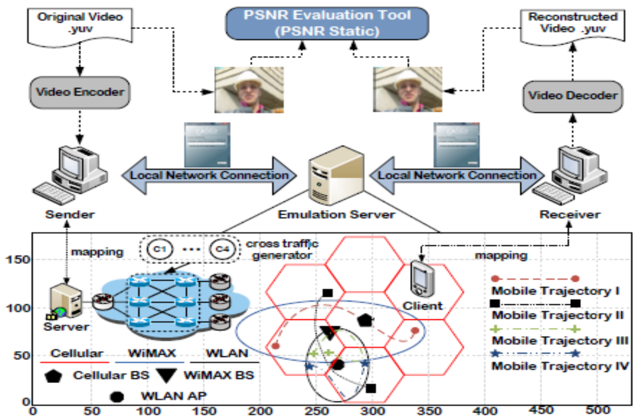


Fig- 3 System architecture for performance evaluation.

Exata is an advanced edition of QualNetin which we can perform semi-physical emulations. In order to implement the real-video-streaming based emulations, we integrate the source code of JSVM1 [as Objective File Library (.LIB)] with Exata and develop an application layer protocol of “Video Transmission”. The detailed descriptions of the development steps could be referred to Exata Programmer’s Guide.

b. Reference Schemes

The path quality is estimated based on the ratio of chunk delivery time to sending buffersize. CMT-QA distinguishes the network congestionsfrom wireless channel errors with the RTTp cwnd production. The path congestion window sizeis updated once consecutive losses are detected.

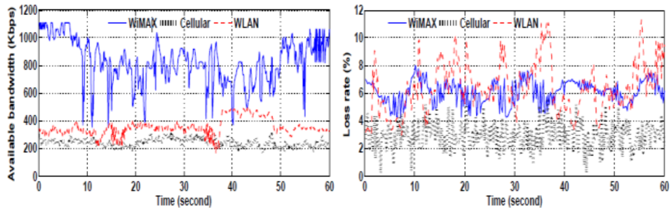
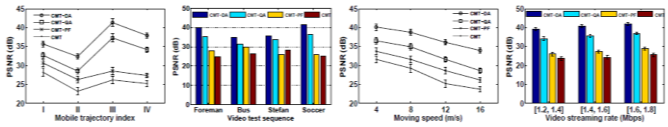


Fig 3:Profile of channel status information:
(a) available bandwidth, (b) path loss rate



(a) different trajectories (b) different sequences (c) different speeds (d) VBR streaming

c. Performance Metrics

- PSNR (Peak Signal-to-Noise Ratio) is the standard metric of objective video quality and is a function of the mean square error between the original and the received video frames.
- Inter-packet delay-High jitter values between packets cause bad visual quality (e.g., video glitches and stalls during the display)
- Good put is an application-level throughput ,i.e.,the number of useful information bitssuccessfully received by the destination within the imposed deadline.
- Effective loss rate, theeffective loss rate includes both the transmission andoverdue losses.

d. Emulation Scenarios

The four mobile trajectories represent the different access options for the mobile user in the integrated heterogeneous wireless networks. The mobile client requests to the server through a wireless interface and constructs the connection whenever it moves into the coverage. The initial moving speed of the client is set to be 2 m/s.

B. *Evaluation Results*

It is well known that cellular networks exhibit better performance in sustaining user mobility than WiMAX and WLAN but provide a lower peak data rate.

a. PSNR

The proposed CMT-DA takes into account the delay constraint imposed by the video applications while the

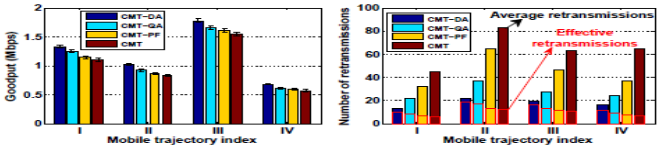
ISSN: 2278-0181

reference schemes are unaware of the video traffic information.

b. Inter-packet Delay

The average inter-packet delay of the four competingschemes with respect to different mobile trajectories.As a rule of thumb, largerend-to-end delays result in lower video quality in real-time applications. CMT-DA achieves significantly lower delays than the reference schemes and Fig. 8b shows the resulting ratio of overdue packets in all the simulation scenarios.

c. Goodput



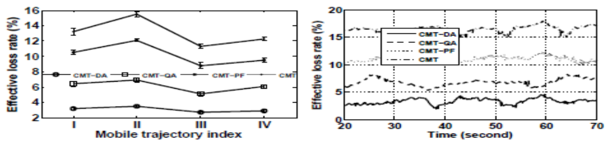
(a) average goodput (b) retransmissions

Fig. 4 (a) Average goodput (b) retransmissions

The video PSNR or subjective quality is not only correlated with the good-put, but also depends on the weight of the lost video frames, statistics of scene contents, etc. The unnecessary retransmissions may increase the effective loss rate and degrade the good-put performance of input video streaming.

d. Effective Loss Rate

The more packets the multihomed terminalreceives within the deadline, the lower effective loss rate is observed. As is expected, CMT-DA out performs the competing schemes due to its superiority in both delay and loss performance. CMT-PF outperforms the CMTbecause it is able to pick out the paths with lower quality and mark it as a ‘path failure’.



(a) average values (b) instantaneous values

Fig 5: Average Values and Instantaneous Values

VII CONCLUSION

The future wireless environment is expected to be a converged system that incorporates different access networks with diverse transmission features and capabilities. The increasing powerfulness and popularity of multi-homed mobile terminals facilitate the bandwidth aggregation for enhanced transmission reliability and data throughput.

Optimizing the SCTP is a critical step towards integrating heterogeneous wireless networks for efficient video delivery. This paper proposes a novel distortion-aware concurrent multipath transfer (CMT-DA) scheme to support high-quality video streaming over heterogeneous wireless networks. Through modeling and analysis, we have developed solutions for per-path status estimation, congestion window adaption, flow rate allocation, and data retransmission. As future work, we will study the cost minimization problem of utilizing CMT for mobile video delivery in heterogeneous wireless networks.

REFERENCES

[1] YouTube, http://www.youtube.com/.
[2] Hulu, http://www.hulu.com/.
[3] Cisco company, “Cisco Visual Networking Index: Global Mobile Data Traffic Forecast Update, 2013-2018,” Feb. 2014.
[4] Z. Li, J. Lin, M. I. Akodjenou, et al., “Watching videos from everywhere: a study of the PPTV mobile VoD system,” in Proc. of ACM IMC, 2012.
[5] C. Xu, T. Liu, J. Guan, et al., “CMT-QA: Quality-Aware Adaptive Concurrent Multipath Data Transfer in Heterogeneous Wireless Networks,” IEEE Trans. Mobile Computing, vol. 12, no. 11, pp. 2193-2205, 2013.
[6] W. Song and W. Zhuang, “Performance Analysis of Probabilistic Multipath Transmission of Video Streaming Traffic over Multi-Radio Wireless Devices,” IEEE Trans. Wireless Commun., vol. 11, no. 4, pp. 1554-1564, 2012.
[7] S. Han, H. Joo, D. Lee and H. Song, “An End-to-End Virtual Path Construction System for Stable Live Video Streaming over Heterogeneous Wireless Networks,” IEEE Journal of Selected Areas in Communications, vol. 29, no. 5, pp. 1032-1041, 2011.
[8] K. Chebrolu and R. Rao, “Bandwidth aggregation for real-time applications in heterogeneous wireless networks,” IEEE Trans. Mobile Computing, vol. 5, no. 4, pp. 388-403, 2006.
[9] J. Yooon, H. Zhang, S. Banerjee and S. Rangarajan, “MuVi: AMulticast Video Delivery Scheme for 4G Cellular Networks,” in Proc. of ACM MobiCom, 2012.
[10] N. Freris, C. Hsu, J. Singh, and X. Zhu, “Distortion-Aware Scalable Video Streaming to Multinetwork Clients,” IEEE/ACM Trans. Networking, vol. 21, no. 2, pp. 469-481, 2013.
[11] Mushroom Networks Inc., Wireless broadband bonding network appliance, http://www.mushroomnetworks.com
[12] R. Stewart, “Stream Control Transmission Protocol (SCTP),” IETF RFC 4960, 2007.
[13] J. R. Iyengar, P. Amer, and R. Stewart, “Concurrent Multipath Transfer Using SCTP Multihoming over Independent End-to-End Paths,” IEEE/ACM Trans.Networking, vol. 14, no. 5, pp. 951-964,2006.
[14] P. Natarajan, N. Ekiz, P.D. Amer, and R. Stewart, “Concurrent Multipath Transfer during Path Failure,” Computer Communications, vol. 32, no. 15, pp. 1577-1587, 2009.
[15] T. Dreibholz, E. P. Rathgeb, I. Rungeler, R. Seggelmann, M.Tuxen, and R. Stewartm, “Stream Control Transmission Protocol:Past, Current and Future Standardization Activities,” IEEE Comm. Magazine, vol. 49, no. 4, pp. 82-88, 2011.
[16] C. Xu, E. Fallon, Y. Qiao, L. Zhong, and G.-M.Muntean, “Performance Evaluation of Multimedia Content Distribution over Multihomed Wireless Networks,” IEEE Trans. Broadcasting, vol. 57, no. 2, pp. 204-215, 2011.
[17] J. Liao, J. Wang, T. Li, and X. Zhu, “Introducing Multipath Selection for Concurrent Multipath Transfer in the Future Internet,” Computer Networks, vol. 55, no. 4, pp. 1024-1035, 2011.
[18] J. Wang, J. Liao, T. Li, et al., “OSIA: Out-of-Order Scheduling for In-Order Arriving in Concurrent Multi-path Transfer,” Journal of Network and Computer Applications, vol. 35, no. 2, pp. 633-643, 2012.
[19] X. Zhu, P. Agrawal, J. P. Singh, et al.,“Distributed Rate Allocation Policies for Multihomed Video Streaming Over Heterogeneous Access Networks,” IEEE Trans. Multimedia, vol. 11, no. 4, pp. 752-764, 2009.
[20] T. Alpcan, J. P. Singh, and T. Basar, “Robust Rate Control for Heterogeneous Network Access in MultihomedEnvironments,”IEEE Trans. Mobile Computing, vol. 8, no. 1, pp. 41-51, 2009.
[21] J. Wu, J. Yang, X. Wu, and J. Chen, “A low latency scheduling approach for high definition video streaming over heterogeneous wireless networks,” in Proc. of IEEE GLOBECOM, 2013.

[22] A. Chow, H. Yang, C. Xia, M. Kim, Z. Liu, and H. Lei, "EMS:Encoded Multipath Streaming for Real-time Live Streaming Applications,"in Proc. of IEEE ICNP, 2010. IEEE TRANSACTIONS ON MOBILE COMPUTING, VOL:14, APRIL:201514
[23] J. Lee, S. Bahk, "On the MDP-Based Cost Minimization for Video on-Demand Services in a Heterogeneous Wireless Network with Multihomed Terminals," IEEE Trans. Mobile Computing, vol. 12, no.9, pp. 1737-1749, 2013.
[24] A. H. Zahran and C. J. Sreenan, "Threshold-Based Media Streaming Optimization for Heterogeneous Wireless Networks," IEEE Trans. Mobile Computing, vol. 9, no. 6, pp. 753-764, 2010.
[25] K. Ramakrishnan, S. Floyd, and D. Black, "The addition of explicit congestion notification (ECN) to IP," RFC 3168, Sep. 2001.