Consideration a Novel Manner for Data Sending Quality in Heterogeneous Radio Networks

Mohammadreza Amini, Omid Moradtalab, and Ebadollah Zohrevandi

Abstract—In real-time networks a large number of application programs are relying on video data and heterogeneous data transmission techniques. The aim of this research is presenting a method for end-to-end vouch quality service in surface application-layer for sending video data in comparison form in wireless heterogeneous networks. This method tries to improve the video sending over the wireless heterogeneous networks with used techniques in surface layer, link and application. The offered method is showing a considerable improvement in quality observing by user. In addition to this, other specifications such as shortage of data load that had require to resending and limited the relation period length to require time for second data sending, help to be used the offered method in the wireless devices that have a limited energy. The presented method and the achieved improvement is simulated and presented in the NS-2 software.

Keywords—Heterogeneous wireless networks, adaptation mechanism, multi-level, Handoff, stop mechanism, graceful degrades, application layer.

I. INTRODUCTION

TWO properties of future network are that they are wireless and they have the movement ability. the wireless network have different types such as satellite networks, cellular communication networks, local area network (LAN) and personal area networks including Bluetooth [16]. The public specifications of wireless heterogeneous networks are described as follow.

a) High error rate, because of geographical natural obstacles, signal multi-path weakening, and moving persons in the local networks have high error rate [17]-[18].

b) limited bandwidth and low sending speed in act, there are no local wireless network with high bandwidth and high sending rate in cellular and local wireless networks with satellite communication network, there are problems such as high bandwidth different as between two networks and also the problems of unsymmetrical that prevent of using a unite protocol [8]–[9] – [15]–[17] – [27].

c) Connection cutting off spans: at cell changing time, the connection is cutting off for a moment, which is called handoff. For lessening the handoff, the wireless networks should have co wearing with all technologies [23]. In the wireless networks, we cannot use the TPC for sending the video data. Because this protocol accounted the protocol of losing the package to be the result of congestion in the networks whereas it maybe the result of wireless link error or handoff for sending the video data [43].

We use the UDP protocol and since this protocol has not any control on numbering, time-label and data flow types it is necessary that works be done by mechanisms at application layer. It seems that for solving the problems of wireless networks for sending the video data, we should make changes in several layers and it is not enough the change in only one layer [41]–[42].

II. CONGESTION MULTI-LEVEL EXPLICIT FEEDBACK

The TPC congestion controlling is base on the idea that the congestion at network is the only cause of getting lost of the packages. Therefore, with any package that got lost, the window decreases its own congestion. At the wireless environment, in which the error rate is very high, and the connection are alternatively cutting off and joining, the decrease in congestion window causes a decrease in the use of bandwidth, after losing the packages, which have not got lost because of congestion. Later, the congestion explicit feedback mechanism or ECN has been present to confront these problems.

For achieving an exact feedback of network, we used a mechanism which we called application layer explicit multilevel feedback for congestion [32]. This mechanism has used two allocated bits at Ip fillet (bits 7 and 8 at eight bits a "service type" at IPV4 and eight bits a traffic class at IPV6) for network feedback representation [30]. This mechanism uses two allocated bits for congestion explicit declaration, declaring four different congestion feedbacks. « 00 » for declaring that the connection cannot send the congestion explicit declaration feedback, « 01 » for declaring without-congestion state, « 10 » for declaring slight-congestion state and « 11 » for declaring the intense-congestion state.

Three different limits for RED queue have been used for separating these different levels [4]. These three limits are minimum limit, middle limit and maximum limit. When the

Mohammadreza Amini is with Department of Electrical Engineering, Islamic Azad University, Boroujerd branch, Iran (phone: +98 9124583992; e-mail: iman.amini@gmail.com).

Omid Moradtalab is with Department of Electrical Engineering, Islamic Azad University, Boroujerd branch, Iran (phone: +98 851 3341183; e-mail: omid2006gold@yahoo.com).

Ebadollah Zohrevandi is with Department of Computer Engineering, Islamic Azad University, Malayer branch, Iran (phone: +989188522643; email: ezohrevand2004@yahoo.com).

queue size is lower than the minimum limit, no package is marked, because no congestion happened in the network .when the queue size is between minimum and middle limits, the packages with maximum probability of $P_{mark(10)}$ is marked with « 10 ». The other packages that are of same « 01 » quality, remains without mark. When package size is between maximum and middle limits, the packages are marked with maximum probability of $P_{mark(11)}$ with « 11 » mark and the rest of packages are marked with « 10 ». After the maximum limit, all the packages mark with « 11 ».

All of these quantities take quantity at network line-finders is two IP fillet bits. After reaching this feedback to the receiver, the receiver informed this quantity to application layer, and without sending the IP layer feedback for congestion explicit declaration to sender, awaits alone for adaptation operation at application layer.

The only help of the use of network occupation-state feedback transportation layer is the increase or decreases of the congestion window at sender for controlling sending rate. This method will cause the prevention of congestion, but the important issue that exists that the intense increasing of window in size has a bad effect on video sending rate, which is sensitive to intense changes of sending rate. This issue showed the benefits of using feedback at application layer level.

In network feedback state and made decisions, whit regard to the increase of decision-making parameters, solves problems of making a mistake in the two different scenarios of losing the package either because of error of wireless link or for congestion. The used feedback mechanism has good synchronic use ability with other traffics. In addition, this mechanism has the use ability with old line-finders that put different quantities in IP fillet. This issue helps us to use this mechanism in the network in which old line-finders exist yet.

Here, three basic changes have been applied on the presented mechanisms to be implemented either for the using risk or for its cost. This is possible because firstly the line finders in the wireless network are the same communicational station. Secondly, we can make the necessary changes for this mechanism only by changing the queue mechanism, which usually is implemented in the line-finders, is so software from. Thirdly, because the offered algorithm is able to adapt to the old line-finders, it is not possible for the scenarios of congestion declaration error to be emerged.

A. Marking Probability Calculation

Here, we use probability calculation as fallow. At $\min_{th} < QueueSize < \min d_{th}$ limit, the package marking probability $P_{mark(10)}$ will be equal to:

$$P_{mark(10)} = \frac{QueueSize - \min_{th}}{mid_{th}} \cdot P_{\max} \qquad (1)$$

Where queue size is the buffer size, $\min_{th}$ is the minimum limit and $\min_{th}$ is the middle congestion limit. Similarly, calculation of the marking probability of $P_{mark(11)}$ and $P_{mark(10)}$ at $mid_{th} < Queue\ Size < \max_{th}$ limit, will be calculate as fallow

$$P_{mark(11)} = \frac{QueueSize - \min_{th}}{mid_{th}} \cdot P_{\max} \qquad (2)$$

$$P_{mark} = 1 - P_{mark(11)} \qquad (3)$$

With regard to the fast that these calculation bear little cost and can be executed easily and simultaneously with marking mechanism, applied operation difficulty on the routers is one very much.

B. Justifiability Examination of Congestion Multi-Level Feedback Mechanism

The necessary change for implementing explicit multi-level congestion declaration is different in two units. The first one is applying a change in packages marking mechanism that happens in line-finders. The second one is the necessary changes in link receiver direction, which should be able to transfer the data, which are relevant to feedback, to the application layer [30].First one is that the obtained feedback is not used for matching the transportation layer fillet quantity and contrary to the before mechanisms, it has been transported to application layer. The second is that the multi-level feedback mechanism in network layer level or transportation layer level is not transferred to the sender and only the made decisions in application layer will be declared to sender in fillet application layer and the third basic difference is in the manner of probability calculation. The probability in this method is calculated as follow.

III. DISTINGUISHING THE HANDOFF AND CUTTING OFF SPANS

In this unit, we use the ability of mobile devises to distinguish the occurrence or the occurrence probability of connection cutting off spans in wireless networks. In mechanisms such as freeze TCP [1]-[40], it has been referred to using the signal weakness for distinguishing the handoff and cutting off spans at the wireless networks. We use this mobile host ability for transferring it to application layer adaptation and use it as a decision-making parameter in the application layer adaptation mechanism. To do this, we imagine that a mobile host is at minimum limit of a wireless network. At cell changing time or changing the wireless service, before doing handoff and cutting off, the signal weakens in mobile host. In addition, in the case that the connection cutting off is because of interference and wireless link problems, this connection cutting off can forecast by mobile host of receive signal weakening.

IV. BANDWIDTH STATE, SENDING SPEED AND DISPLAY BUFFER

As referred before, the accessible bandwidth is the least unused bandwidth of all links that are between sender and receiver throughout the route, if the sending rate of data flow is less than accessible bandwidth or equal to it, the receiving rate is receiver is equal to sender sending rate. Otherwise, the receiving rate is less than sender is. With regard to what has been said, the receiving rate is a function of sending rate, link capacity and accessible bandwidth among links between basis and destination. No pharynx in the system can determine the receiving rate by itself.

For measuring end-to-end quality of accessible bandwidth, a set of periodic flows is sent between two ends of flow. For each flow, the receiver analyzes and gains the unilateral delay changes, specifying whether the sending rate is more than accessible bandwidth quantity or is less than it is. We have supposed that for measuring speed quantity and accessible bandwidth, we can use a set of video packages as a package train to estimate asymptotic dispersion rate which can be imagined to be the receiving rate in sender [14]. This measurement has been done on movable window with constant size. Therefore, the number of packages in different estimation may vary. Then the estimated quantity was used for concluding a bout accessible bandwidth case. We supposed that the receivable package rate in receiver is calculated by formula (4).This formula addition the receive package number Pi, at Δt time:

$$r_t = \frac{1}{\Delta t_r} \sum_i p_i \qquad (4)$$

Then the calculated quantity improved by applying a coefficient, registered in the old calculated quantity in calculation. If this factor that has an amount between zero and one, $0 \leq \alpha \leq 1$, be much more, it gives more weight to old quantity at decision-making.

$$\hat{r}_r = (1-\alpha)r_r + \alpha r_r \qquad (5)$$

Then, the video receiving rate is compared with the sending quantity and the achieved calculation is sent to adaptation mechanism for decision-making. The achieved quantity has two states:

The accessible bandwidth is less than the necessary quantity for video data flow or the accessible with quantity is more than or equal to the necessary quantity for video data flow.

After calculating the bandwidth quantities, the sending and receiving rates, receiving multi-level feedback quantity, testing whether the handoff and communication cutting off has happened or is happening. Finally, users display buffer and the occupation degree at the next going and returning time are calculated as a scattered video quality signal for user. The received package number will be calculated in the next going and returning time as indicated in formula (6) where, $\hat{r}_r(i)$ is the receivers receiving rate quantity at i time.

$$R(RTT) = \sum_{i=t+1}^{t+RTT} \hat{r}(i) \qquad (6)$$

The number of used packages will be calculated by user by formula (7) where u_i is users using rate quantity of display buffer packages at i time.

$$U(RTT) = \sum_{i=t+1}^{t+RTT} u_i(i)$$

In this case, the display buffer size will be as fallow:

$$buf(t+RTT) = buf(t) + \sum_{i=t+1}^{t+RTT} \hat{r}(i) - \sum_{i=t+1}^{t+RTT} u_i(i) \qquad (7)$$

V. ADAPTATION MECHANISM

After gaining all of the necessary information, the decision will be made and executed. In our offered mechanism, firstly the quality and speed are selected so that they are used with each other. Secondly, it is not necessary that after the time of lowering the sending speed, we must have compensation time, because the decisions would be made so that the receivers display buffer does not empty.

A. Quality Adaptation

There are different methods for quality adaptation their most important ones are: changing the congestion rate of an online encoder, switching between several pre-encoded video types with different quality, eliminating a layer of congestion hierarchy layers at encoder and changing the compressing of one pre-compressed type.

In our offered method, we imagine that quality adaptation is done at sender side and adaptation type is also requested by switching between different types with different video data quality. This sending rate adaptation is made on the basis of the receivers' request.

B. Stop Mechanism

In designing this mechanism, we suppose that the UDP will be used for transportation layer. The numbering and putting the time label on packages are done by application layer mechanisms.

We use the stop mechanism, used in application layer, for confronting to losing the packages and their resending at the time of hand off. For using application layer stop method we, at first, use a method like freeze-TCP in transportation layer. This mechanism, called ATCP, is presented for confronting to hand offs and cutting off of the long connections at wireless networks. This mechanism requires only the changes at connections mobile side and network layer feedback. For declaring the connection position in this algorithm, it is supposed that at the time of starting and cuttings off the connections, the link layer send an accidental declaration signal for mobile TCP. At the end, this algorithm increases the efficiency and giving passage to the transmission control protocol on the wireless links.

We changed the above algorithm and used at our application layer adaptation so that we assume that the signals weakness and its distinction in link layer are used by mobile user as a sign for anticipating the probability of connection cutting off. In this case, if the sender side becomes aware of probable connection cutting off, it notifies the application layer of this probability by a signal.

Notice that at this method, the probability of connection cutting off is sent to application layer for anticipation the connection state, before the connection cutting off declaration of cutting signal. Now, if the sender becomes aware of this probability before the receiver, it tries to, before the connection cutting off, send the connection cutting off declaration to the receiver, and the receiver, by a signal, declares this probability to application layer, where the adaptation mechanism is applied with a signal.

The stop mechanism at application layer cause to more giving passage to connection, because the mechanisms like slow start and resending will never be executed at transportation layer and the congestion window problems and untimely decreasing will not be occurred.

When the connection sender, unable to distinguish its probability before the connection cutting off, or when it get from the receiver an end feedback based on stop , occurrence signal of the connection cutting off arrives from link layer to application layer and until receiving the next package of receiver, which is the sign of ending the connection cutting off, it stays at the same state and after the of connection cutting off, it starts sending with the same previous rate or new rate of receivers request. All of decisions are made on application layer and in receiver side.

Usually the supposition that the sender of a video data flow of a server, which is in a fixed wiry network, is more probable, but for future applications, it is possible that a method be needed for the case in which the sender is wireless or mobile. In our offered mechanism, it has been imagined that both side of sender and receiver connection can be wireless, mobile and in connection cutting off state, distinguished by the sender side the sender notifies the receiver of occurred connection cutting off and without being await for decision makers, it itself zeroes its own sending rate temporarily. After solving the connection cutting off problem, the sender checks again the position based on the arriving certified package and restart the connection according to that and the receiver's request.

C. Graceful Degrade

With notice to the properties of wireless links, their excessive error and long connection off spans at wireless heterogeneous networks, we conclude that sometimes it may be impossible to prevent the display from becoming empty by changing the sending rate and quality and etc.

At this state, it seems to be useful to offer a mechanism as the last solution for preventing the display buffer from becoming empty (Coming down under middle limit).

We try to do speed-quality adaptation so that the display buffer has video frame more than a certain limit for displaying. However, at certain case that happened, it is better to have a mechanism for slow decreasing the observed picture quality by user in order to compensate the fast emptying of display buffer by achieved time spans at this slow quality decrease. This mechanism is used in controlled case, in order to have the least possible effect on displaying video data. For this purpose, it is supposed that when the display buffer is emptying of video frames and at the same time connection handoff and cutting off in the system happened or is happening, the offered algorithm determines the maximum users using rate for display buffer. This action is implemented at display buffer by determining a delay between using two consecutive frames.

We can do the calculations simultaneously with adaptation mechanism at the application layer until the processing time arrives to minimum. Notice that this mechanism will cause the increase in the display delay quantity at displaying start of any video data frame and because this quantity in approaching to delay time quantity between using packages displaying 12 frames on second, the created delay quantity is insignificant from supervisor's vision.

The other issue that seems necessary to be stated is implementing the place of this mechanism in application layer.

This mechanism will be applied in displaying queen and at time of its delivering to application program. It is notable that the offered mechanism is not an powerful enough mechanism for doing this work and only was presented with considering the display buffer conditions and packages with fixed size for displaying and very easy hypothesis about video contents and network parameters.

VI. SIMULATION AND CONCLUSION

For simulation of the offered algorithm, we used of NS-2 network simulator [31]. One of the network simulator's desired properties is possibility for increasing the protocols, mechanism, traffics, applications and totally the things that are in the simulator itself, but have different hypothesis and parameters.

A. The used Header for Application Layer

For implementing this application, we try to assemble it on the UDP protocol and we ourselves increase the controlling process to it.

Multime dia	Seq No	MEC N	Rate	Qual	ACK	Time	Disconn ect

Fig. 1 The application layer designed package for transferring multi – media information

B. Measurement Parameters of Performance Algorithm

- Implementing cost
- One of the parameters assigned to implementing cost is the necessary change rate of an algorithm. In offered algorithm, the necessary change degree for implementing has been only in application layer and congestion explicit declaration mechanism. Since the application layer adaptation

mechanism is in software form, the applying cost of this change is not much and in comparison with improvement that it will give to system, it can be overlooked.

• The offered congestion explicit declaration in this algorithm, observing the ECN designing principles, which are offered by Mr. Fluid, [35] can be used with old line-finders.

• The display buffer size and it's not being empty as the quality seen by user: This parameter is the size of this buffer that its being empty shows that there is no package to display. Therefore, it is not being empty at the time of occurrences such as hand off and congestion for network is the main purpose of this algorithm.

C. Conclusion

In offered simulating mechanism, the following scenario was used for examining the different parameters, by using the topology indicated in Fig. 2 and by using video traffic an actual test was done on network. In this simulation, we suppose that the type of used traffic has been CBR and it has had a primary sending rate equal to one tenth megabit on second. The packages displaying rate is supposed to be 25 frames on second. In this unit, we used simulation of display butters occupation degree as a parameter that indicates the quality observed by the user.
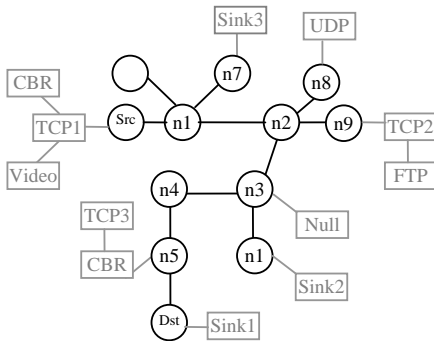

Fig. 2 Used topology in scenario for testing the offered algorithm

For simplicity, frame size has been fixed and equal to 500 bytes. The maximum size of display buffer at destination is 50 frames that will be 2.5-kilo bytes for 600 bytes frames, is not a large quantity, and can be use in mobile wireless systems. Figs. 3, 4 can show it [46].
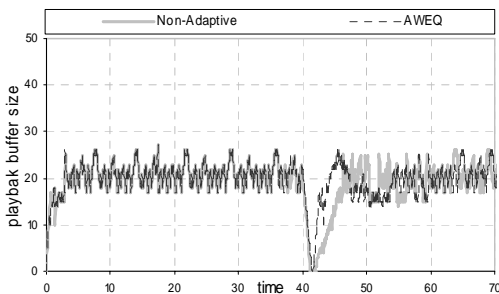

Fig. 3 Display buffer size for a state that connection cutting off has been occurs in network for a period about one second
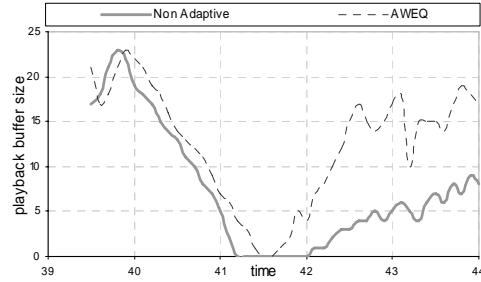

Fig. 4 Display buffer size for a state that connection cutting off has been occurs in network for a period about one second

The above diagrams show that at the time of happening the intense handoff in network, the offered algorithm prevents from fast emptying of buffer by using graceful degrade mechanism and then after ending the probable period of handoff, it fills faster the buffer by fast ending the stop state and starting with a fast receivable mechanism. Therefore, the quality of displayed video to user in this state is better than the inadaptable method finally. We considered the congestion happening scenario in network. In offered algorithm, at the time of congestion emergence in network graceful degrade mechanism begin working when the buffer size approaches to its minimum limit, in the other hand by decreasing the sending rate and also video quality, it helps to decreasing the congestion position. In network in Figs. (5, 6), we see that in spite of congestion emergence at network; the display buffer has not been emptied. Therefore, the quality of display video to user in this state is better than the inadaptable method. That is, inadaptable method means sending video data on network infrastructure and ensuring of the same quality without application layer adaptation.
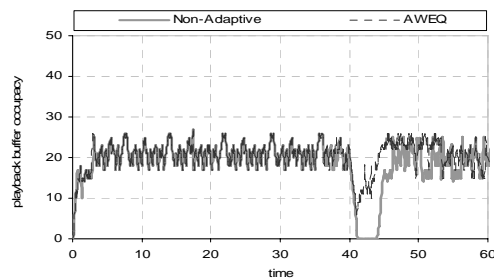

Fig. 5 Displays buffer size for the state that the congestion has happened at network
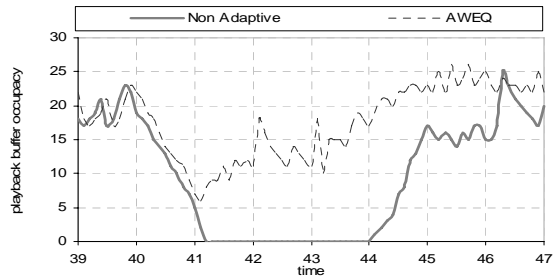

Fig. 6 Displays buffer size for the state that the congestion has happened at network (the zooming case)

REFERENCES

[1] Ajay Kr. Singh, Sridhar Iyer, "ATCP: Improving TCP performance over mobile wireless environments", IEEE 4th international workshop on mobile and communication networks, IEEE, 2002, pp.239-243.

[2] Akan, O.B.; Akyildiz, I.F.; "ATL: An Adaptive Transport Layer Suite for Next-Generation Wireless Internet", Selected Areas in Communications, IEEE Journal on, vol. 22, Issue: 5, pp 802 – 817, June 2004.

[3] Aravind Velayutham, Hung-Yun Hsieh, Raghupathy Sivakumar; "On Transport Layer Adaptation in Heterogeneous Wireless Data Networks", IWQoS 2005: 69-80.

[4] Arjan Durresi, Mukundan Sridharan, Chunlei Liu, Mukul Goyal, "Traffic Management using Multilevel Explicit Congestion Notification", Proceedings of the 5th World Multiconference on Systemics, Cybernetics and Informatics SCI'2001, ABR over the Internet, Orlando, FL, July 22-25, 2001, pp. 12-17.

[5] Bakre, A.; Badrinath, B.R.;"I-TCP: indirect TCP for mobile hosts", Distributed Computing Systems, 1995., Proceedings of the 15th International Conference on , 30 May-2 June 1995, pp 136 – 143

[6] Bakre, A.V.; Badrinath, B.R.; "Implementation and performance evaluation of Indirect TCP", Computers, IEEE Transactions, Volume: 46, Issue: 3, March 1997, Pages: 260 – 278.

[7] Bakshi, B.S.; Krishna, P.; Vaidya, N.H.; Pradhan, D.K.;" Improving performance of TCP over wireless networks", Distributed Computing Systems, 1997., Proceedings of the 17th International Conference on , May 1997, pp 365 – 373.

[8] Bregni, S.; Caratti, D.; Martignon, F.; "Enhanced loss differentiation algorithms for use in TCP sources over heterogeneous wireless networks", Global Telecommunications Conference, 2003. GLOBECOM '03. IEEE, Vol. 2, 1-5 Dec.2003, P.P. 666 – 670.

[9] Chadi Barakat; Eitan Altman; Walid Dabbous; "On TCP Performance in a Heterogeneous Network: A Survay", 2004.

[10] Chang-Jung Kao; Wanjiun Liao; Chin-Hei Chien; Jen-Chi Liu; "Improving TCP performance in heterogeneous mobile networks", Vehicular Technology Conference, 2003. VTC 2003-Spring. The 57th IEEE Semiannual, Vol. 3, P.P. 1735 - 38, 22-25 April 2003.

[11] Chinta, M.; Helal, A.; Lee, C.; "ILC-TCP: an interlayer collaboration protocol for TCP performance improvement in mobile and wireless environments", Wireless Communications and Networking, 2003. WCNC 2003. 2003 IEEE , Volume: 2, 16-20 March 2003 Pages:1004 - 1010 vol.2

[12] Dongwook Lee, Jong Won Kim;" Delayed-duplicated ACK (DDA) algorithm for TCP performance enhancement to overcome packet sequence disruption in fast-handoff of Mobile IPv6", Distributed Computing Systems Workshops, 2004. Proceedings of 24th International Conference on, March 2004, pp 98 – 103.

[13] Eshak, N.; Baba, M.D.; "Design a new transport protocol (wireless TCP) to support mobility for mobile ad hoc networks", 4th National Conference on Telecommunication Technology, NCTT 2003 Proceedings, 14-15 Jan. 2003, pp 144 – 147.

[14] Ewa Kusmierek, David H.C. Du, "Streaming video delivery over Internet with adaptive end-to-end QoS", The Journal of Systems and Software, ELSEVIER, Vol.75, Issue 3, 2004, pp 237-252.

[15] Fei Peng; Leung, V.C.M.; "Enhancing fairness and throughput of TCP in heterogeneous wireless networks", Personal, Indoor and Mobile Radio Communications, 2003. PIMRC 2003. 14th IEEE Proceedings on, Vol. 1, P.P.94-98, 7-10 Sept. 2003.

[16] Findlay, D.; Flygare, H.; Hancock, R.; Haslestad, T.; Hepworth, E.; Higgins, D.; McCann, S.;"3G Interworking with wireless LANs", 3G Mobile Communication Technologies, 2002. Third International Conference on (Conf. Publ. No. 489), P.P. 394-9, 8-10 May 2002.

[17] Francis Jing Jye Yang, "Micro-Mobility Management and Freeze-TCP Implementation in a Wireless Network", MS Thesis, University of Canterbury, Christchurch, New Zealand.

[18] George Xylomens, "TCP Performance Issues over Wireless Links", IEEE Communications Magazine, Vol.39, No.4, April 2001.

[19] Goff, T.; Moronski, J.; Phatak, D.S.; Gupta, V.; "Freeze-TCP: a true end-to-end TCP enhancement mechanism for mobile environments", INFOCOM 2000, Proceedings Nineteenth Annual Joint Conference of the IEEE Computer and Communications Societies, IEEE, vol:3, March 2000, pp1537 – 1545.

[20] Haas, Z.J.; Agrawal, P.; "Mobile-TCP: an asymmetric transport protocol design for mobile systems", ICC 97 IEEE International Conference on communications, IEEE, vol. 2, 8-12 June 1997, pp 1054 – 1058.

[21] Huston, G.; "TCP in a wireless world", Internet Computing, IEEE, Vol. 5, Issue: 2, March-April 2001, pp 82 – 84.

[22] Iu Yu; Ye Min-hua; Zhang Hui-min; "Improve TCP performance over wireless link" 14th IEEE conference on Personal, Indoor and Mobile Radio Communications, 2003. PIMRC, IEEE, vol. 1, Sept. 2003, pp110 – 113.

[23] Jae-Woo Kwon; Hee-Dong Park; You-Ze Cho;" An efficient TCP mechanism for Mobile IP handoffs", Electrical and Electronic Technology, 2001. TENCON. Proceedings of IEEE Region 10 International Conference on, vol. 1, Aug. 2001, pp 278 – 281.

[24] Jain, A.; Pruthi, A.; Thakur, R.C.; Bhatia, M.P.S.;"TCP analysis over wireless mobile ad hoc networks", Personal Wireless Communications, 2002 IEEE International Conference on, 15-17 Dec. 2002 Pages: 95 – 99.

[25] Jijun Luo, Xuemin Huang, "Planning Future Heterogeneous Wireless Networks", Progress in Electromagnetics Research Symposium 2005, Hangzhou, China, August 22-26.

[26] K. Ramakrishnan, S. Floyd, D. Black, "The Addition of Explicit Congestion Notification (ECN) to IP", RFC 3168 (2001).

[27] Linying Xu, Manyun Liu, Lianfang Zhang, "TCP Performance Evaluation over Wireless Networks", Canadian Conference on Electrical and Computer Engineering, IEEE, Vol.2, May 2004, pp. 983-986.

[28] Liu Yu; Ye Min-hua; Zhang Hui-min;" The handoff schemes in mobile IP", Vehicular Technology Conference, 2003. VTC 2003-Spring. The 57th IEEE Semiannual, Vol. 1, 22-25 April 2003, pp 485 – 489.

[29] MJ Blangé., IP Karkowski, and BCB Vermeulen, "Service discovery in heterogeneous wireless networks", International Workshop on Wireless Ad-hoc Networks (IWWAN), Oulu, Finland 2004.

[30] Mukundan Sridharan, Arjan Durresi, Chunlei Liu, Raj Jain, " Wireless TCP Enhancements using Multi-level ECN", Proceedings of SPIE Internet Quality of Service, ITCOMM 2003 , Vol. 5245, Orlando Florida (2003) 180-187.

[31] NS-2 distributed code web page at http://www.isi.edu/nsnam/ns/ns-contributed.html

[32] Ouldooz Baghban Karimi, Mahmood Fathy, Saleh yousefi, "Application level Wireless Multi-level ECN for Video and Real-time Data," accepted in ICN2006, unpublished.

[33] Pablo Vidales, Leo Patanapongpibul, Glenford Mapp, Andy Hopper, "Experiences with Heterogeneous Wireless Networks, Unveiling the Challenges"

[34] Rhee, I.; Balaguru, N.; Rouskas, G.N.; "MTCP: scalable TCP-like congestion control for reliable multicast" INFOCOM '99 Proceedings, IEEE, vol: 3, March 1999, pp 1265 – 73.

[35] S.Floyd, "Specifying Alternate Semantics for the Explicit Congestion Notification (ECN) bit", Internet Draft (2005).

[36] Seong-Soo Park; Dong-Ho Cho; Sung Kim; "Performance improvements of TCP protocol for mobile data service", Global Telecommunications Conference, 1997. GLOBECOM '97, IEEE, Vol. 3, P.P.:1871 - 75, 3-8 Nov. 1997.

[37] Singh, A.K.; Iyer, S.; "ATCP: Improving TCP performance over mobile wireless environments", Mobile and Wireless Communications Network, 2002. 4th International Workshop on, Sept. 2002, pp 239 – 243.

[38] Srisankar Kunniyur, R. Srikant, "End-to-End Congestion Control Schemes: Utility Functions, Random Losses and ECN Marks," IEEE Transactions on Networking, Vol.11, No.5, October 2003, pp.689-702.

[39] Sung-Eun Kim; Copeland, J.A.;" Interworking between WLANs and 3G networks: TCP challenges", Wireless Communications and Networking Conference, 2004. WCNC. 2004 IEEE, Vol. 2, 21-25, P.P. 1252 – 57, March 2004.

[40] Tom Goff, James Moronski, D.S Phatak, Freeze-TCP: A true end-to-end enhancement mechanism for mobile environments," IEEE Infocom, 2000, pp. 1537 – 1545.

[41] V. Venugopal Reddy, Vindo Sharama, M.B, Suma, "Providing QoS to TCP and Real Time Connections in the Internet," Queuing Systems, Kluwer Academic Publishers, No. 46, 2004, pp. 461- 480.

[42] Veselin Rakocevic, "Congestion Control for multimedia applications in the wireless Internet", International Journal of Communication Systems, Vol.17, No.7, 2004, pp. 723-734.

[43] Vulupala, V.; Kumar, V.;" Dynamic mobile IP and Nice-TCP for improving TCP/IP performance", Mobile Data Management, 2002. Proceedings. Third International Conference on, 8-11 Jan. 2002, pp163 – 164.

[44] Wenqing Ding, Abbas Jamalipour, "A New explicit loss notification with acknowledgement for wireless TCP", 12th IEEE Symposium on PIMRC, IEEE, Vol.1, 2001, pp. 65-69.

[45] Wu, E.H.-K.; Mei-Zhen Chen; "JTCP: jitter-based TCP for heterogeneous wireless networks", Selected Areas in Communications, IEEE Journal on, Vol. 22, Issue: 4, pp. 757 – 66, May 2004.

[46] E.Zohrevandi, R.Roustaei, O.Moradtalab, New Scheme for Improving the Quality of Service in Heterogeneous Wireless Network for Data Stream Sending, International Conference on Communication System and Computer Networks", CSCN 2008, WCSET" Volume 33, PP 200-205 September 2008.

[47] Yavatkar, R., Bhagawat, N.;" Improving end-to-end performance of TCP over mobile internetworks", Mobile Computing Systems and Applications, 1994. Proceedings, Workshop on, Dec. 1994, pp146 – 152.

[48] Xiaojing He, Linying Xu, Manyun Liu, Lianfang Zhang, "TCP Performance Evaluation over Wireless Networks, Canadian Conference on Electrical and Computer Engiering, 2004, Vol.2, pp.983-986.

[49] Yang, "Streaming Video over the Internet", Presentation in Signal Compression Lab, ECE Department, University of California Santa Barbara.

[50] Zhenghua Fu; Xiaoqiao Meng; Songwu Lu;" How bad TCP can perform in mobile ad hoc networks", Computers and Communications, 2002. Proceedings. ISCC 2002. Seventh International Symposium on, July 2002, pp 298 – 303.

Mohammadreza Amini is born at 1981. He is an Iranian telecommunication researcher. He graduated in electronic course from Isfahan University of technology and he has MS grade in telecommunication of Malek Ashtar University. Now he teaches in Islamic Azad University of Boroujerd branch.

Omid Moradtalab was born in Malayer at 1986. He is an Iranian electrical and computer researcher, and a student of Broujerd Azad University in Iran. To now, he has different invention about electronic. He has 3 degree and 3 international papers at electronic field.

Ebadollah Zohrevandi was born in Malayer at 1979. He studied at Malayer schools. He went to Iran University of Science and Technology (IUST) in 1999 at computer engineering field (hardware) and he has MS grade in computer engineering. He has four papers in computer and mystical. He is specialist at computer networks. He teaches in Islamic Azad.