http://ansinet.com/itj

ITJ

ISSN 1812-5638

INFORMATION TECHNOLOGY JOURNAL

ANSInet
Asian Network for Scientific Information
308 Lasani Town, Sargodha Road, Faisalabad - Pakistan

Information Technology Journal 12 (1): 168-173, 2013
ISSN 1812-5638 / DOI: 10.3923/itj.2013.168.173

An Improved TFRC Scheme for Wired/Wireless Hybrid Networks

Zhi-Jiang Xu, Shui-Qin Chu, Yuan Wu and Li-Min Meng
College of Information Engineering, Zhejiang University of Technology, Hangzhou, 310023, China

Abstract: TFRC (TCP (Transmission Control Protocol) Friendly Rate Control) has been widely used in wired networks for its enhanced friendliness and fairness. However, TFRC cannot distinguish between packet losses due to network congestion and those due to wireless link error in wireless networks. Thus, in this study an improved TFRC scheme that is able to differentiate between congestion losses and wireless link error losses for wired/wireless hybrid network is proposed. Specifically, the improved TFRC scheme utilizes the information of the one-way delay to regulate the transmission rate at the sender and simulation experimental results show this strategy is effective and performs much better than the traditional TFRC.

Key words: TFRC, congestion control, hybrid networks, transmission control

INTRODUCTION

With the rapid development of Internet technology, various kinds of real-time streaming services have appeared recently in heterogeneous wired and wireless networks (Taleb *et al*., 2008).

Because of its advantages over TCP, User Datagram Protocol (UDP) has been widely adopted by most of real-time streaming services at present (Floyd *et al*., 2000). However, UDP has some intrinsic drawbacks and cannot meet the requirements of bulk data transfer perfectly (Ren *et al*., 2009). Specifically, UDP does not have the congestion control mechanism which results in a high rate of packet loss due to network congestion and an aggressive bandwidth occupation compared to TCP. Those disadvantages may lead to the breakdown of whole networks under serious conditions. In addition, the ever-increasing real-time multimedia services yield a rapid growth in network traffic, making the network congestion even more serious.

Thus, it affects the transmission of UDP own application in turn and further exacerbates network congestion (Peng and Zheng, 2010; Mayadas *et al*., 2009).

Therefore, real-time streaming services require a proper congestion control mechanism which not only avoid each node transiting packets blindly and aggressively but also is TCP-friendly, where a mechanism is "TCP-friendly" if it is reasonably fair when competing with other TCP flows for bandwidth (Bruno *et al*., 2008).

TCP-friendly congestion control is a well-known congestion control mechanism. Roughly speaking, it can be divided into two categories, namely, the windows-based approach and the rate-based approach (Widmer *et al*., 2001). The windows-based TCP-friendly congestion control mechanisms may yield a sudden flow and result in a large delay jitter. Hence, this approach is not suitable for real-time streaming media applications. In comparison, the rate-based TCP friendly congestion control mechanisms make use of congestion information of network to adjust transmission rates. TFRC is the representative of such kind of congestion control mechanisms (Floyd *et al*., 2003).

TFRC was designed originally for the wired networks. Unfortunately, TFRC cannot distinguish between packet losses due to congestion and those due to wireless link error in wireless environment. To distinguish those two different types of packet losses, many previous works have been carried out.

For instance, Lin and Long (2010) used the average delay jitter to determine the causes of packet losses in wireless network and to control the packet sending rate accordingly. In comparison, Li *et al*. (2006) proposed to use the jitter ratio to determine the causes of packet losses. Zhou *et al*. (2007) proposed an enhanced TFRC scheme trying to distinguish the two types of losses based on the differentiating method used in TCP Veno. Pyun *et al*. (2003) proposed a WM-TFRC scheme that adopts the Access Point (AP) in wireless Local Area Network (LAN) to measure the rate of loss events and feeds back its value to the sender. Meanwhile, the receiver also provides feedback about the rate of total loss events to the sender. Thus, the sender can calculate the rate of congestion loss events.

Different from the previous works that used Round-Trip Time (RTT) to distinguish two types of packet losses, an improved TFRC scheme that only uses the one-way delay to differentiate two types of packet losses is proposed.

Corresponding Author: Zhi-Jiang Xu, College of Information Engineering, Zhejiang University of Technology, Hangzhou, 310023, China Tel: +86-571-85290372

When the receiver receives a packet from the sender, it costs some time for the receiver to process and generate an Acknowledgment (ACK) to the sender, denotes it as T_r. Then the measured RTT is defined as:

$$RTT = RTT^0 + T_r \quad (1)$$

where, RTT^0 is the actual time of transmission of the packet in the networks, here, RTT is corrupted by T_r (Li *et al*., 2010).

Specifically, the one-way delay is a measure of duration for packet delivery from the sender node to the receiver node in the networks (Choi and Yoo, 2005). Since the one-way delay does not include the processing time, thus, it is more accurate and simple than the RTT. Furthermore, video transmission is basically regarded as one-way transmission except for some control information. That is, the sender node is only responsible for sending the video packets and the receiver node receives those. Therefore, the one-way delay which is from the sender to the receiver, does need to considered while the reverse one is ignored. For the above reason, only part of information of the RTT is needed (Al-Omari *et al*., 2009). In this study, an improved TFRC scheme adopts the information on the one-way delay to differentiate between congestion losses and wireless link error losses for wired/wireless hybrid network is proposed.

MECHANISM OF TFRC

TFRC was proposed originally to ensure fairness in bandwidth sharing among different TCP flows. The key advantage of TFRC is that it is able to maintain a relatively smooth packet-sending rate and reduces the fluctuation in throughput over time in comparison with TCP. Therefore, TFRC is more suitable for applications such as telephony and streaming services that usually have relatively smooth sending rates.

Specifically, a stable sending-rate mathematical model of TCP Reno (Padhye *et al*., 2000) was given by:

$$X = \frac{S}{R\sqrt{\frac{2b\times P}{3}} + t_{RTO}\sqrt{\frac{3b\times P}{8}}P(1+32P^2)} \quad (2)$$

where, X denotes the sending-rate in bytes/second. S represents the packet-size in bytes. R is the round trip time in seconds. $P\in[0,1]$ stands for the loss event rate, i.e., the ratio between the total number of packets lost and the total number of packets transmitted. t_{RTO} is the timeout value for TCP retransmission in seconds and b is the number of packets acknowledged by a single TCP acknowledgment. t_{RTO} and b are normally set to $t_{RTO} = 4R$, $b = 1$, respectively.

The detailed workflows of TFRC are as follows:

- A receiver measures the loss event rate and returns an ACK with this information to the sender
- The sender uses these messages to calculate the round-trip time
- The sender puts the loss event rate and round-trip time into the throughput Eq. 2 and obtains an acceptable sending-rate
- The sender uses the obtained sending-rate to send packets to the corresponding receiver

CHALLENGES OF TFRC IN WIRELESS ENVIRONMENT

TFRC has been widely adopted for rate control and it works pretty well in wired networks. However, the performance of TFRC degrades sharply in wireless networks, where the packet losses may derive from wireless link errors including the transmit bit errors, fading and handoffs (Walse and Dhotre, 2007). Pervious work pointed out that TFRC designed originally for the wired networks and hence it cannot distinguish between packet losses due to congestion and those due to wireless link error (Jung *et al*., 2006). As a result, TFRC can only be used to deal with the buffer overflow in the wired network, since it blindly interprets any kind of packet losses as an indicator for congestion. However, as mentioned earlier, the packet losses may also derive from a short-term wireless link error, such as the frequency-selective fading due to multi-path transmission, co-channel interference etc. Taking any packet loss as an indication for congestion results in a conservative rate control and hence TFRC usually suffers from throughput loss in wireless networks. In this study, an improved TFRC scheme based on the one-way delay for receiver to distinguish between packet losses due to network congestions and those losses due to wireless link error is proposed. The proposed improved TFRC scheme achieves better performance and is more simply than the conventional schemes in wireless environment.

THE PROPOSED SCHEME

To focus on the objective in this study, two kinds of packet losses are considered, namely, the losses due to congestion and the losses due to wireless link error. As mentioned before, the key task of the improved TFRC mechanism is to distinguish these two kinds of packet losses accurately. It is well known that network congestion occurs when the packet sent exceed the capacity of network, resulting in that the packets are accumulated at intermediate nodes, e.g., routers (Wu *et al*., 2009; Sasipraba and Srivatsa, 2006). Congestion may lead to a long delay. The more serious the network congestion is, the longer the packets queue in the router and the greater the one-way delay is (Al-Nabhan *et al*., 2006).

Since one-way delay time caused by wireless link error usually is much shorter than that caused by network congestion, this phenomena can be used to improve the TFRC scheme such that the aforementioned two different causes of packet losses can be differentiated.

Let S_i be the time to send the i-th data packet and let R_i be the time to receive the i-th data packet. Then, R_i-S_i denotes the one-way delay for delivery of the i-th packet. Specifically, let $SLD_i = R_i - S_i$ denote this one-way delay and D denotes the delay jitter. In further, the delay jitter is defined as the difference between two consecutive one-way delay, i.e., it is given by:

$$D(i, i-1) = SLD_i - SLD_{i-1} \quad (3)$$

Specifically, the delay jitter D>0 denotes a longer delay suffered by the new packet. If the delay jitter D is greater than a threshold K, i.e., $SLD_i - SLD_{i-1} > K$, then the network is assumed to be congested and the packet loss is due to the network congestion loss. Otherwise, the packet loss is considered to be caused by the wireless network link error. To avoid the occurrence of a large delay jitter due to a sudden change in network condition, e.g., topology, the weighted average one-way delay is introduced, that is:

$$\overline{SLD_i} = \alpha\overline{SLD_{i-1}} + (1-\alpha)SLD_i \quad (4)$$

where, α is the weight.

When the network is congested, D is greater than the threshold K, i.e.:

$$\overline{SLD_i} - \overline{SLD_{i-1}} > K \quad (5)$$

Take Eq. 4 into Eq. 5 to get:

$$\alpha\overline{SLD_{i-1}} + (1-\alpha)SLD_i - \overline{SLD_{i-1}} > K \quad (6)$$

In addition, Eq. 6 can be equivalently changed into:

$$(1-\alpha)\overline{SLD_{i-1}} + K < (1-\alpha)SLD_i \quad (7)$$

or:

$$SLD_i - \overline{SLD_{i-1}} > \frac{K}{1-\alpha} \quad (8)$$

Since $\overline{SLD_{i-1}} > 0$, a new threshold can be obtained as follows:

$$\frac{SLD_i}{\overline{SLD_{i-1}}} > \frac{K}{\overline{SLD_{i-1}}(1-\alpha)} + 1 \quad (9)$$

For normal real-time multimedia applications, when the variation of the one-way delay is greater than 10%, the network is normally considered to be or has been congested (Chen *et al*., 2003). Therefore, K is generally set to be $0.1\times\overline{SLD_{i-1}}$.

Let:

$$\eta = \frac{SLD_i}{\overline{SLD_{i-1}}}$$

If packet loss are detected and then η needs to be calculated to determine whether the network packet loss is caused by congestion or wireless link error. $\eta > \frac{0.1}{1-\alpha} + 1$ means that the one-way delay of the network is increasing and the network tends to be congested. In this case, the current packet loss caused by congestion happens. Therefore, the packet-sending rate should be reduced to avoid congestion. On the other hand, $\eta < \frac{0.1}{1-\alpha} + 1$ represents that the one-way delay of the network is reducing and the packet loss is due to the wireless link error. In this case, the current packet loss caused by wireless link error happens. Therefore, the packet-sending rate does not need to be reduced.

The value of η mainly depends on the choice of α which means that the chosen of α affects the performance of the algorithm significantly. Since α is normally set to be a value from 0.6-1.0, thus many experimental tests have been taken to determine the value of α in this interval. The result of the tests shows that 0.95 is the best choice for α.

The mechanism of the improved TFRC is shown as follows:

- When the receiver receives a packet, the one-way delay of the packet is firstly calculated and then if packet losses are detected, the reason of the losses needs to be judged. Hence, the packet loss caused by wireless link error is ignored by the calculation of event loss rate. After calculation of the event loss rate, the receiver feeds back an ACK to the sender

- The sender uses these messages to calculate the round-trip time
- The sender puts the loss event rate and round-trip time into the throughput Eq. 2 and obtains an acceptable sending-rate
- The sender uses the obtained sending-rate to send packets to the corresponding receiver

SIMULATIONS AND ANALYSIS

Here, simulations are carried out in order to measure the performance of the improved TFRC compared to TCP and the traditional TFRC. All the simulations are using Network Simulator 2 tool (NS-2). A dumbbell network topology in the simulations is shown in Fig. 1. S1 and S2 stand for bottleneck in the wired link. They are the intermediate points in the networks. WN1 and WN0 stand for wired source node that send TCP flow and TFRC flow respectively. N1 and N0 are both wireless destination node that receive the TCP flow and TFRC flow respectively. In other words, WN1 and WN0 are the senders and N1 and N0 are the receivers. The link between WN1 and S1, WN0 and S1, S1 and S2 are wired link. The link between S1 and S2 also is bottleneck link. The link between S2 and N1, S2 and N0 are wireless link. The network related parameters are list in Table 1. The packet size of TCP flow, TFRC flow and the improved TFRC flow are set to 1000 Bytes.

Simulation scene 1: A TCP flow is established between WN1 and N1 while the data flow between WN0 and N0 is the traditional TFRC flow and the total time of this simulation is set to 120 sec. Figure 2 shows the throughput of both TCP and traditional TFRC acquired in this simulation. The average throughput of original TFRC in Fig. 2 is 2917 kb sec^{-1} and the average throughput of TCP is 1679 kb sec^{-1}. The throughput ratio of the traditional TFRC and the TCP flow is 1.73. The results demonstrate the friendliness of TFRC to TCP.

Table 1: Parameters-setting in simulation scene

Parameter	Value
Bottleneck link bandwidth	5 Mb sec^{-1}
Bottleneck link delay	5 msec
Wired link bandwidth	100 Mb sec^{-1}
Wired link delay	1 msec
The way of packet loss	Drop tail
Wireless link bandwidth	11 Mb sec^{-1}
Packet loss rate of wireless link	0.05
Test duration	120 sec

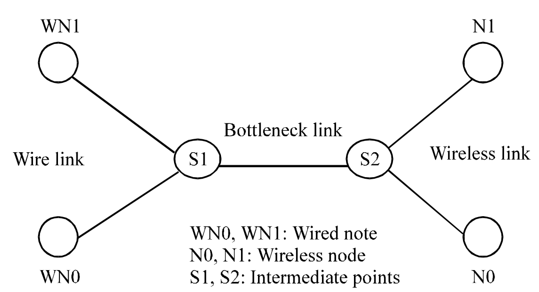

Fig. 1: Wired/wireless hybrid network topology

Simulation scene 2: A TCP flow is established between WN1 and N1, the data flow between WN0 and N0 is the improved TFRC flow and the total time of this scene is set to 120 sec. Figure 3 shows the throughput of both TCP and improved TFRC acquired in this scene. The average throughput of improved TFRC in Fig. 3 is 3048 kb sec^{-1} and the average throughput of TCP is 1614 kb sec^{-1}. The throughput ratio of the traditional TFRC and the TCP flow

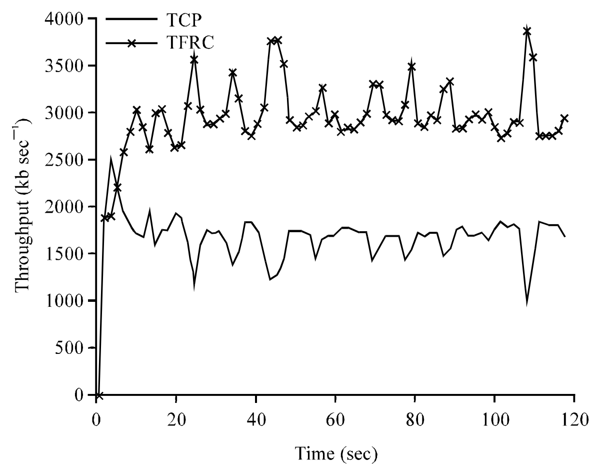

Fig. 2: Throughput comparison between transmission control protocol (TCP) connection and TCP friendly rate control (TFRC) connection

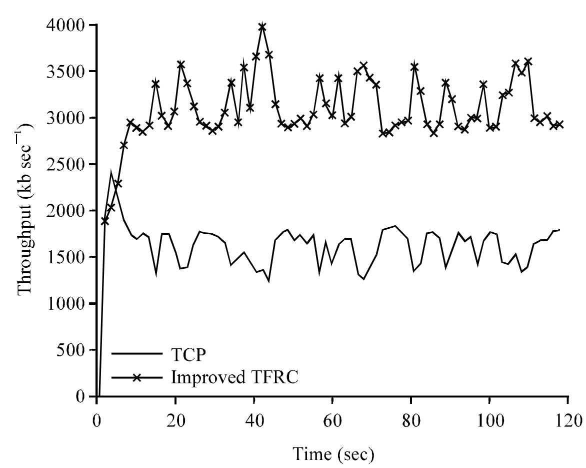

Fig. 3: Throughput comparison between TCP connection and improved TFRC connection

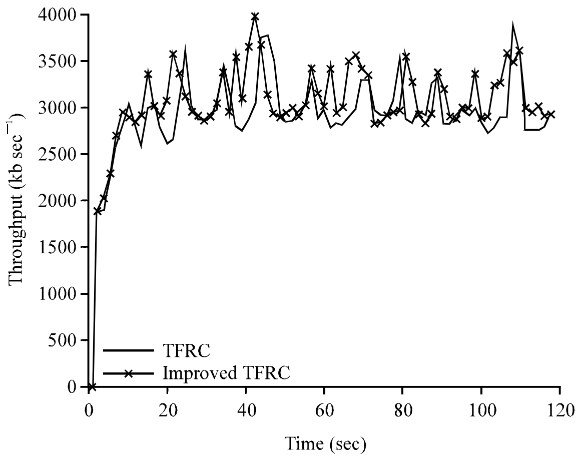

Fig. 4: Throughput comparison between original TFRC connection and improved TFRC connection

Table 2: The statistical comparison table of simulation results

Algorithm	Average throughput (kb sec^{-1})	Average throughput of TCP (kb sec^{-1})	Total throughput of Network (kb sec^{-1})
Original TFRC	2917	1679	4596
Improved TFRC	3048	1614	4662

is 1.88. The results show that the improved TFRC can fairly share the wireless bandwidth with TCP which means the improved TFRC supports TCP-friendliness.

Figure 4 is the performance comparison of throughput between the traditional TFRC and the improved TFRC. Table 2 shows the comparison lists of the traditional TFRC and the improved TFRC in detail.

The amplitude fluctuation of the improved TFRC's curve is less than that of the traditional TFRC in Fig. 4. It illustrates that the proposed scheme can improve loss packet discrimination. Hence, it can increase the reliability of network congestion control and the stability of the network performance. Figure 4 demonstrates that the improved TFRC performs better than the traditional TFRC in throughput. The average throughput of improved TFRC is 131 kb sec^{-1} larger than the traditional TFRC. However, the average throughput of TCP only reduced 65 kb sec^{-1} which indicates that the total throughput of the network has increased.

In addition, the throughput of TFRC increased 3.07% by using the mechanism in Lin and Long (2010) while the proposed improved TFRC scheme in this study has enhanced the throughput by 4.5%. In addition, the curve of the proposed improved TFRC is more flat than the one in Lin's. The throughput ratio of the improved TFRC and TCP is 1.88 while this value is 2.46 referenced to Lin's. In other words, the proposed improved TFRC is more TCP-friendly than that appeared in Lin's.

CONCLUSION

In this study, an improved TFRC scheme is proposed which is based on the one-way delay to distinguish packet losses due to network congestion and those losses due to wireless link error for wired/wireless hybrid networks. The results of the simulations demonstrate that the improved TFRC scheme achieves a better performance than the original TFRC. It has not only enhanced the throughput of the whole network but is also more TCP-friendly.

ACKNOWLEDGMENTS

This material is based upon work funded by Zhejiang Provincial Natural Science Foundation of China under Grant No. Y1101077, Zhejiang Provincial Key Science and Technology Innovative Team (2010R10011-03) and Zhejiang Provincial Key Lab. of Communication Networks and Applications.

REFERENCES

Al-Nabhan, M., S. Yousef and J. Al-Saraireh, 2006. TCP protocol and red gateway supporting the QoS of multimedia transmission over wireless networks. Inform. Technol. J., 5: 689-697.

Al-Omari, H., F. Wolff, C. Papachristou and D. McIntyre, 2009. Avoiding delay jitter in cyber-physical systems using one way delay variations model. Proceedings of the International Conference on Computational Science and Engineering, Volume 2, August 29-31, 2009, Vancouver, USA pp: 295-302.

Bruno, R., M. Conti and E. Gregori, 2008. Throughput analysis and measurements in IEEE 802.11 WLANs with TCP and UDP traffic flows. IEEE Trans. Mobile Comput., 7: 171-186.

Chen, Y., C. Qiao, M. Hamdi and D.H.K. Tsang, 2003. Proportional differentiation: A scalable QoS approach. IEEE Commun. Magaz., 41: 52-58.

Choi, J.H. and C. Yoo, 2005. Analytic end-to-end estimation for the one-way delay and its variation. Proceedings of the IEEE 2nd Consumer Communications and Networking Conference, January 3-6, 2005, Las Vegas, USA., pp: 527-532.

Floyd, S., M. Handley, J. Padhye and J. Widmer, 2003. TCP friendly rate control (TFRC): Protocol specification. Network Working Group, RFC: 3448. http://www.ietf.org/rfc/rfc3448.txt

Floyd, S., M. Handley and J. Padhye, 2000. A comparison of equation-based and AIMD congestion control. ACIRI Technical Report. http://www.icir.org/tfrc/aimd.pdf

Jung, I.M., N.B. Karayiannis and S. Pei, 2006. Improving TCP-friendly rate control in wired and wireless networks by a scheme based on wireless signal strength. Proceedings of the International Conference on Networking and Services, July 16-18, 2006, Slicon Valley, CA., pp: 20-20.

Li, H., N. Xiong, J.H. Park and Q. Cao, 2010. Predictive control for Vehicular sensor networks based on round-trip time-delay prediction. IET Commun, 4: 807-809.

Li, Q., D. Chen, Y.C. Liu, L.N. Zheng, 2006. Jitter ratio based TFRC scheme in wireless-wired hybrid network. Proceedings of the International Conference on Digital Telecommunications, August 29-31, 2006, Cap Esterel, Cote d'Azur, France, pp: 38-38.

Lin, Y.H. and Z.H. Long, 2010. Improved algorithm of TFRC aiming at controlling real-time transmission in wireless network. Comp. Eng. Design, 31: 1898-1900.

Mayadas, A.F., J. Bourne and P. Bacsich, 2009. Online education today. Science, 323: 85-89.

Padhye, J., V. Firoiu, D.F. Towsley and J.F. Kurose, 2000. Modeling TCP reno performance: A simple model and its empirical validation. IEEE Transac. Network., 8: 133-145.

Peng, T. and Q. Zheng, 2010. Resource occupation of peer-to-peer multicasting. Inform. Technol. J., 9: 438-445.

Pyun, J.Y., Y. Kim, K.H. Jang, J.A. park and S.J. Ko, 2003. Wireless measurement based resource allocation for QoS provisioning over IEEE 802.11 wireless LAN. IEEE Transac. Consum. Elect., 49: 614-620.

Ren, Y., H. Tang, J. Li and H. Qian, 2009. Performance comparison of UDP-based protocols over fast long distance network. Inform. Technol. J., 8: 600-604.

Sasipraba, T. and S.K. Srivatsa, 2006. Network border patrol, a novel congestion avoidance mechanism for improving QOS in wireless networks. Inform. Technol. J., 5: 427-432.

Taleb, T., K. Kashibuchi, A. Leonardi, S. Palazzo, K. Hashimoto, N. Kato and Y. Nemoto, 2008. A cross-layer approach for an efficient delivery of TCP/RTP-based multimedia applications in heterogeneous wireless networks. IEEE Trans. Veh. Technol., 57: 3801-3814.

Walse, K.H. and D.R. Dhotre, 2007. Wireless network: Performance analysis of TCP. Inform. Technol. J., 6: 363-369.

Widmer, J., R. Denda and M. Mauve, 2001. A survey on TCP-friendly congestion control. IEEE Network Magaz., 15: 28-37.

Wu, W., Z. Zhang, X. Sha and C. He, 2009. Auto rate MAC protocol based on congestion detection for wireless Ad Hoc networks. Inform. Technol. J., 8: 1205-1212.

Zhou, B., C.P. Fu, C.T. Lau, C.H. Foh, 2007. An enhancement of TFRC over wireless networks. Proceedings of the IEEE Wireless Communications and Networking Conference, March 11-15, 2007, Hong Kong, China, pp: 3019-3024.