RTT Prediction in Heavy Tailed Networks

L. Rizo-Dominguez, D. Munoz-Rodriguez, C. Vargas-Rosales, D. Torres-Roman, and J. Ramirez-Pacheco

***Abstract*—TCP is the most used transport protocol in the Internet and it relies on RTT (Round Trip Time) predictions for the retransmission control algorithm. Most of the algorithms reported in the literature consider memoryless traffic characteristics and do not study the performance under heavy tailed scenarios present in the Internet. In this paper, an algorithm for RTT prediction in a heavy-tailed environment is introduced, and it is shown to follow closely and accurately the actual RTT. The proposed algorithm is simple and permits online implementations. Results are compared with those obtained with other methodologies for real trace sets. It is shown that the proposed algorithm leads to a lower prediction error.**

***Index Terms*—Heavy tail jitter distribution, RTO, RTT prediction and Internet packet delay.**

I. Introduction

FORECASTING is a common practice in engineering as it permits anticipation of the best parameter settings to operate a system optimally. Nevertheless, predictions often depart from true values due to the presence of unforeseen variations that upset the system performance. A common forecasting practice is to rely on the expected value since it optimizes the mean square error criterion between the estimated and the true value. This approach is valid when the assessment is conducted over a large number of results and large time scales. However, several system operations demand a short-term prediction and must be based on recent observations of the current realization. Also, network nodes carry out prediction of diverse parameters in different protocols so that resources are used as efficient as possible. A dominant transport protocol in the Internet is TCP (Transmission Control Protocol), [1], where the packet retransmissions are directly related to the current RTT (Round Trip Time) parameter. However, as the packets travel along the route they will be exposed to noise, queueing-delays and packet losses that cause errors that render in RTT variations. Therefore, it is necessary to forecast the RTT parameter value in order to regulate the packet retransmission process. For instance, if the predicted value (also known as Retransmission Time Out -RTO) is lower than the actual RTT, a packet loss is assumed, and the packet is retransmitted reducing the effective bandwidth. On the other hand, an RTO overestimation leads to a delayed packet loss detection upsetting the congestion control process, [2], [3]. Thus, an RTT prediction with good accuracy is fundamental to avoid situations with congestion and delays. Packet delay prediction often depends on the correlation characteristics of the data. Some approaches propose simplified estimators that rely both, on past observation and on previous predictions. A common estimator proposed by Jacobson, [4], behaves adequately in environments with Gaussian distributed delay with a small correlation. However, for heavy-tailed scenarios, the predictions depart substantially from the true value. Alternative methods, [3], suggest an estimator where past observations are window weighted for a set of realizations. In this paper, we present, under a bound error criterion, a new simplified RTT predictor suitable for heavy-tailed scenarios that exhibits a reduced forecast error. It is also shown that small prediction errors are seldomly achievable for big jitter dispersion scenarios and that there is a tradeoff between a desirable small prediction error and the jitter dispersion. Feasible regions according to traffic characteristics and QoS requirement are also reported.

Manuscript received December 2, 2013. The associate editor coordinating the review of this letter and approving it for publication was B. Bellalta.

L. Rizo is with ITESO (e-mail: lrizo@iteso.mx).

David Munoz and Cesar Vargas are with Tecnologico de Monterrey, Campus Monterrey, N.L., 64849 (e-mail: {dmunoz, cvargas}@itesm.mx).

D. Torres is with CINVESTAV, Guadalajara (e-mail: dtorres@gdl.cinvestav.mx).

J. Ramirez is with the Universidad del Caribe (e-mail: jramirez@ucaribe.edu.mx).

Digital Object Identifier 10.1109/LCOMM.2014.013114.132668

II. The Proposed Predictor

In this section, we introduce an RTT prediction algorithm under an error bound probability criterion in a heavy tailed environment. Let $RTT(k)$ denote the round trip delay experienced by the $(k-1)$-th packet, and let the next packet delay prediction $RTO(k)$ be of the form

$$RTO(k) = RTT(k-1) + \xi, \tag{1}$$

where ξ stands for a travel time variation that will be set to keep probabilistically the error between the forecasted and the true value, $e(k) = RTT(k) - RTO(k)$, within specified constraints.

It has already been said that some values of $e(k)$ impact adversely network performance. For instance, a positive value of error $e(k)$ indicates an RTT underestimation, leading to unnecessary retransmissions and throughput reduction; while a negative value will slow down recovery of network congestion control.

The prediction error $e(k)$ can be considered a random variable that it is desirable to keep within certain limits. A first approach could be to use the Chebyshev inequality that takes the form

$$P\{|e(k)| > \epsilon\} \leq \frac{E\{(RTT(k) - RTO(k))^2\}}{\epsilon^2} = q. \tag{2}$$

Stating that the absolute error $|e(k)|$ is mean to exceed a value ϵ with a probability smaller than a value q. This formulation is not applicable to heavy tailed environments as the probability of $|e(k)|$ becoming large is not negligible and its variance may not be defined. This may lead to a meaningless inequality. An alternative probabilistic performance criterion can be written as

$$P\{|e(k)| \leq \epsilon\} \geq \psi, \quad (3)$$

where ψ is a definable quality of service (QoS) parameter, denoting the minimum proportion of time that the prediction error is below the allowance error ϵ. The value of this error allowance tends to be small, i.e., $\epsilon \rightarrow 0$, thus the criterion shown in (3). Now, considering the definition of error $e(k) = RTT(k) - RTO(k)$ and substituting $RTO(k)$ from (1), then (3) can be rewritten as

$$P\{-\epsilon + \xi \leq RTT(k) - RTT(k-1) \leq \epsilon + \xi\} \geq \psi, \quad (4)$$

The difference $RTT(k) - RTT(k-1) = J(k)$ is known as the Internet delay-jitter and it has been reported to exhibit a heavy tailed behavior [12], [13]. In particular $J(k)$ follows a Cauchy distribution, [10], i.e., the distribution of $J(k)$ is

$$P\{J(k) \leq x\} = \frac{1}{2} + \frac{1}{\pi} tan^{-1}\frac{x}{\gamma} \geq \psi, \quad (5)$$

Therefore, using (5) in (4), we get

$$P\{-\epsilon + \xi \leq J(k) \leq \epsilon + \xi\} = \frac{1}{\pi} tan^{-1}(\frac{\epsilon + \xi}{\gamma}) - \frac{1}{\pi} tan^{-1}(\frac{-\epsilon + \xi}{\gamma}) \geq \psi \quad (6)$$

Equation (6) is further reduce to

$$P(-\epsilon + \xi \leq J(k) \leq \epsilon + \xi) = \frac{1}{\pi} \tan^{-1}\left(\frac{2\epsilon\gamma}{\gamma^2 - \epsilon^2 + \xi^2}\right) \geq \psi \quad (7)$$

where parameter γ is the jitter dispersion induced by the Internet queueing process. Taking the tangent on both sides in inequality (7), and taking advantage of the monotonic behavior of the tangent function, for a given specified quality of service parameter ψ (QoS) and error allowance ϵ the estimated travel time variation can be given by

$$\xi^2 \leq \frac{2\gamma\epsilon}{\tan(\pi\psi)} + \epsilon^2 - \gamma^2 \quad (8)$$

Note that ξ^2 is upper bounded by a quadratic form of ϵ , and it can be verified that ξ^2 is real-valued for $\epsilon > \gamma\{\csc(\pi\psi) - \cot(\pi\psi)\}$.

Thus, for instance if the QoS is set to be of $\psi = 0.9$, i.e., at least 90% of the time the delay-jitter is within the limits desired, the error requirement ϵ cannot be smaller than 8.61γ Figure 1 shows the normalized ϵ/γ feasible allowance region for a given dispersion γ and a probability requested criterion ψ. As jitter dispersion increases, as in the case of heavy tailed scenarios, the ratio ϵ/γ decreases, compromising QoS ψ.

Finally, according to (1) and (8) the value of the RTT predictor is determined by the QoS parameter ψ, the desired allowance ϵ, and the jitter-delay γ dispersion. This is

$$RTO(k) = RTT(k-1) + \sqrt{\frac{2\gamma\epsilon}{\tan(\pi\psi)} + \epsilon^2 - \gamma^2}. \quad (9)$$

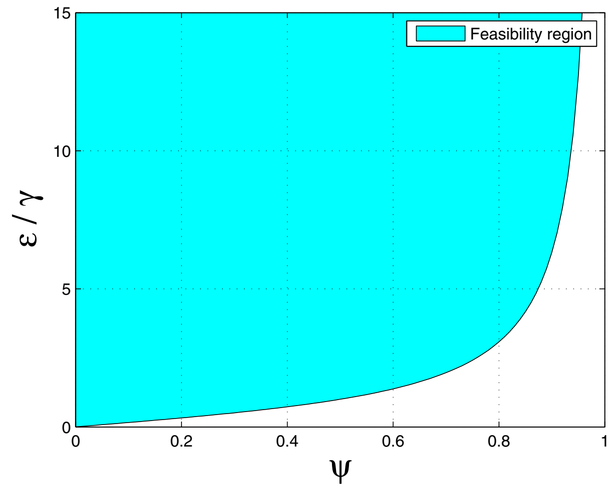

Fig. 1. ϵ/γ feasibility region for a given probability ψ.

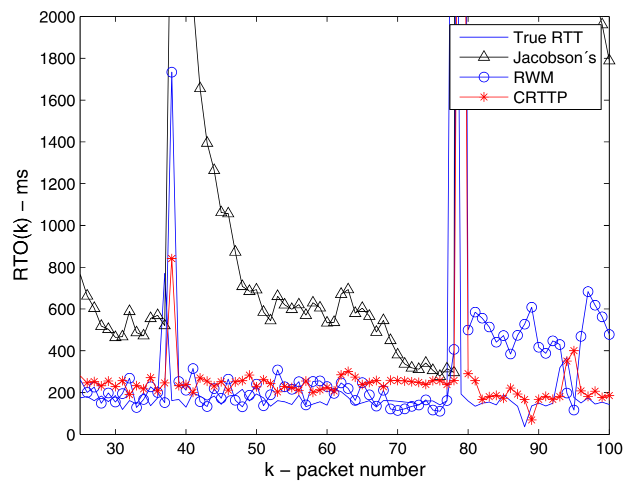

Fig. 2. $RTO(k)$ predictor comparison.

III. Results and Discussion

In order to evaluate the performance of the proposed algorithm, the predictor was assessed using reported traffic traces. We present results that have been obtained by means of the available data in those traces and using a random sampling rate stated in the corresponding references [7], [8], [9]. We present results that compare the performance of the proposed predictor to that of other methods [9] by showing values of the RTT predicted by all these methods and those true values obtained analyzing the reported traffic traces. We also show performance in terms of the error between the predicted and the true value, $e(k) = RTT(k) - RTO(k)$.

It can be seen in equations (8) and (9), that knowledge of the jitter dispersion γ parameter is required. This can be obtained either from quartiles and interactions, [6]. Since it is expected that the γ estimation be in real-time, it must be simple and swift. Thus, use of the quartile based estimation is recommended. For the purpose of this work, Mc. Culloch estimation, [5], is used to find the dispersion parameter.

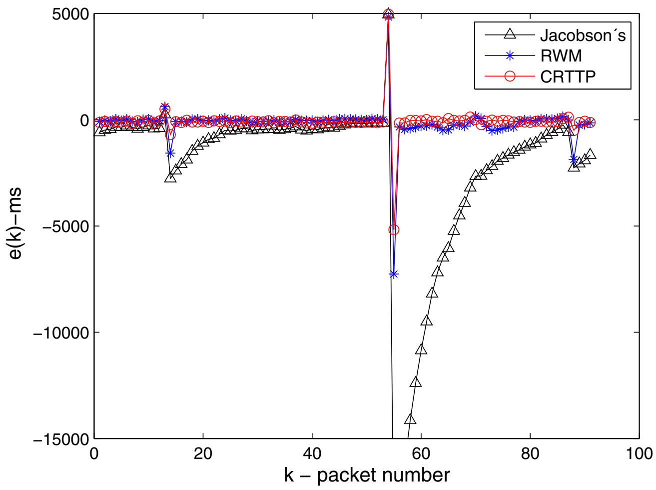

Fig. 3. Error of predictors.

Figure 2 presents the values of the predicted RTT, i.e., $RTO(k)$, obtained by using the Jacobson estimator (estimator mostly used in current TCP implementations), the Recursive Weighted Median (RWM), [3], which is the algorithm designed for general heavy tailed environment, and the proposed Cauchy Round Trip Time Predictor (CRTTP). These predictors are compared to the true RTT values obtained from the traffic trace.

The results show that Jacobsons algorithm is very sensitive to RTT sudden changes (large excursions from the mean RTT values), and exhibits quick response to changes, but a very slow recovery. We can also see that most of the time Jacobsons algorithm provides overestimates of the true RTT values. On the other hand, also in Figure 2, we can see that performance of the predictors RWM and CRTTP lead to significant accuracy and good recovery timing of the predictions of RTT which result in closer predictions to the true RTT values.

Figure 3 compares the error performance criterion for the different predictors compared already in Figure 2. It can be observed that the proposed predictor CRTTP has a smaller error than any of the other methods for a heavy tailed delay environment. It can also be seen that for some periods of time, the RWM predictor is better than the CRTTP, but for other periods of time, RWM does not have small values of error. For all the time period, it can be seen that Jacobsons algorithm gives error values larger than those of the other methods.

The root mean square error (RMSE) in the observation windows is also compared for different data sets and presented in Table I. It can be seen that the proposed algorithm predictor offers a better accuracy while keeping lower complexity for online implementation.

In Figure 4 it can be seen that the proposed predictor takes advantage with a reduced expected error and the error variance of CRTTP is the lowest. The negative error is related to overestimation, and produces a slowly packet loss detection. Otherwise, the positive error damages the TCP performance with needless retransmissions.

TABLE I
ROOT MEAN SQUARE PREDICTION ERROR

Prediction Algorithm	TIME AVERAGED SQUARE ERROR		
Jacobson	19.5ms(a)	136.21ms(b)	4680ms(c)
RWM	4.36ms(a)	67.77ms(b)	969ms(c)
CRTTP	3.16ms(a)	20.1ms(b)	768.9ms(c)

(a) RMSE value for data traces in reference [9]
(b) RMSE value for data traces in reference [7]
(c) RMSE value for data traces in reference [8]

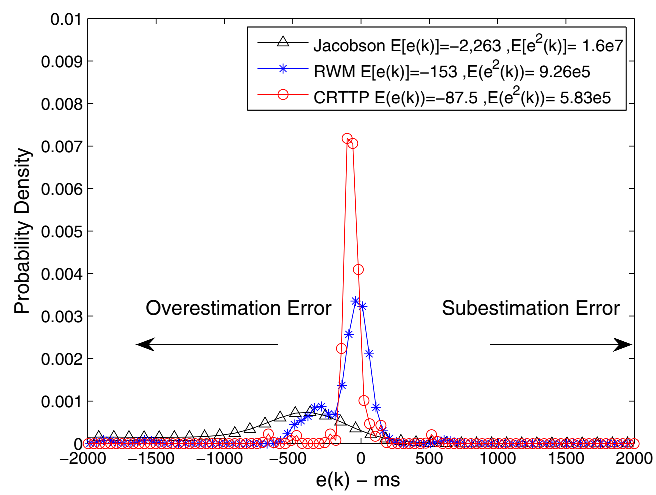

Fig. 4. Probability density of prediction error.

Another comparison of the algorithms is through their execution times in a common computer platform. All the algorithms were scripted in Matlab using an i5-Core 3.1 Ghz processor with 4 GB of RAM.

The mean execution time was $4.33\mu s$ for the Jacobson algorithm that is a reduced value due to the low computational complexity. However, a better accuracy was obtained by WM and CRTTP algorithms that present an execution time of $230\mu s$ and $207\mu s$, respectively. Hence, the algorithm proposed provides better accuracy and shorter execution times than those of the Jacobsons algorithm. It is also important to point out that the proposed predictor, CRTTP, requires only some 20 RTT observations for an adequate performance.

IV. CONCLUSIONS

In this paper, we addressed the problem of RTT prediction (Retransmission Time Out-RTO). A simple predictor suitable for a heavy tailed jitter environment has been presented and compared to other predictors in the open literature. The performance of the proposed predictor is assessed using different Internet RTT measurement sets. Results show that the Cauchy predictor offers advantages both in terms of low complexity and lower prediction error. As a consequence of the smaller values of error, CRTTP, results in less overestimation and subestimation instances than the other algorithms.

REFERENCES

[1] L. Qiong and D. Mills, "Jitter-based delay-boundary prediction of wide-area networks," *IEEE/ACM Trans. Networking*, vol. 9, pp: 578–590, 2001.

[2] L. Ma, K. Barner, and G. Arce, "Statistical analysis of TCP's retransmission timeout algorithm," *IEEE/ACM Trans. Networking*, vol. 14, no. 2, Apr. 2006.

[3] L. Ma, G. R. Arce, and K. Barner, "TCP retransmission timeout algorithm using weighted medians," *IEEE Signal Process. Lett.*, vol. 11, no. 6, pp. 569–572, June 2004.

[4] V. Jacobson, "Congestion avoidance and control," in *Proc. 1988 SIGCOMM*, pp. 314–329.

[5] J. H. McCulloch, "Simple consistent estimators of stable distribution parameters," *Commun. in Statistics – Simulation and Computation*, vol. 15, pp. 1109–1136, 1986.

[6] V. Akgiray and C. G. Lamoureux, "Estimation of stable-law parameters: a comparative study," *J. Business & Economic Statistics*, vol. 7, no 1, pp. 85–93, Jan. 1989.

[7] The Harvard Hourglass project, "Network coordinate research at Harvard," http://www.eecs.harvard.edu/syrah/nc/, Sept. 12, 2012.

[8] D. Torres-Román home page, "RTT Measurements," http://gdl.cinvestav.mx/dtorres/, Aug. 1, 2013.

[9] University of Delaware Network Time Protocol (NTP) Data Raw, http://www.eecis.udel.edu/ ntp/ntpstats/pogo/primary/, May 25, 2012.

[10] L. Rizo, D. Muñoz, D. Torres, and C. Vargas-Rosales, "Jitter in IP networks, a Cauchy approach," *IEEE Commun. Lett.*, vol. 14, pp. 190–192, 2010.

[11] W. Willinger, M. S. Taqqu, R. Sherman, and D. Wilson, "Self-similarity through high-variability: statistical analysis of Ethernet LAN traffic at the source level," *IEEE/ACM Trans. Networking*, vol. 5, pp. 71–86, 1997.

[12] V. Paxson, "End to end Internet packet dynamics," in *Proc. 1997 ACM SIGCOMM.*

[13] T. K. Bandhopadhya, M. Saxena, and A. Tiwari, "Jitter's alpha-stable distribution behaviour," *International J. Computer Technol. and Electron. Engineering*, vol 3, pp. 13–16, 2013.

[14] S. Belhaj and M. Tagina, "Modeling and prediction of the Internet end-to-end delay using recurrent neural networks," *JNW*, vol 4, pp. 528–535, 2009.