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Further Explore on Performance of Various Traffic Combination based on Network Calculus

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Abstract—The end-to-end transport principle has been one of the building blocks since the Internet born. However, with the explosion of Internet scale and various services, all of these pound on the poor scalability and strictness of the end-to-end model. With the Internet switch from host-center network to services/information/content centric network. In this paper, we investigate the two traditional end-to-end transport protocols thoroughly as a guidance for designing an appropriate substitute transport principle for the new architecture. In detail, our work explores various network traffic combination performance from the perspective of the transport layer. Firstly, we model the end-to-end transport process and analyze the delivery possibility against the transmission attempts. Then, we deduce the service-curve of the composite TCP and UDP traffic. Further, we simulate the topology taken from our campus network to evaluate the performance of two cases: 1. different traffic amount with the same traffic combination; 2. various traffic combinations under the same traffic flow amount. The simulation results show that UDP take 30%-75% of the network traffic could benefit the throughput performance, which offers an indicator for adjusting the node traffic combination in order to improve the network performance. Moreover, we consider delay, packet drop rate and resource utilization efficiency. Based on all these, the composite traffic combination could be modified to ensure the quality of service more comprehensively.

Index Terms—end-to-end transport model, performance evaluation, delay, drop rate, network calculus, NS-2, composite traffic combination

I. INTRODUCTION

Since the early days of the Internet in the 1980s, the end-to-end principle has been one of the building blocks in the transport layer. However, with the rapid explosion of Internet scale and various services emerging, all of these pound on the poor scalability and strictness of the original end-to-end principle. A suitable new transport model within better Internet architecture should be considered to meet newly emerging stricter requirements.

One milestone of the Internet researches is the distributed data caches such as web caching, content delivery network and the newly born in-network caching networks. The web caching and the content delivery network both emphasize where to place the content to make customers' experience better. On the other hand, in-network caching does not care where to locate content in the context of the network nodes are all caching nodes if necessary. The focus of in-network caching is what to cache and how to adjust the cache content to better support customers' obtainment of their required content. It is common that there are caching nodes into the current Internet. With the intermediate caching nodes through the communication process, it obviously tells that the original end-to-end principle is facing huge challenges. Therefore, it calls for novel transport model for the revolutionized network architecture.

In addition, concluded from the previous researches [1]-[4], it is easy to see that the focus of Internet has been changed from where the corresponding end points are located to how to efficiently fetch service/information/content in need. In line with the development phases of future Internet architecture, it is noted that service/information/content centric network has stuck out a mile to substitute the initial host-centric Internet. At the same time, in the new Internet architecture, the end-to-end model is not suitable.

Therefore, in order to design feasible new transport models, we must firstly reference the network performance with the massive increase in traffic and network accessing users by virtue of the end-to-end principle. In this paper, we contribute to investigate the network performance with two traditional end-to-end transport control mechanisms of transmission control protocol (TCP) and user datagram protocol (UDP). With the global traffic explosion, how TCP and UDP traffic combination impacts on the network performance would play a significant role in directing new transport model design. Various traffic combinations would have different effect on network performance. Detail analysis of those would provide a reference for adjusting the resources achievement processes in the service/ information/content centric network to guarantee users' experience.

As we can see, in the recent years, more and more people have participated in the Internet and more and more applications are launched in the Internet. From the

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year of 2005 to 2012, the amount of network users has increased to 564,000,000 and the popular increase rate is 42.1%. In addition, we could draw from the Chinese Internet data report that the audio and video applications take the primary traffic percentage compared with the data transfer applications. The traffic combination has also become varied in the specific network topology. Investigation of various traffic combination influence on network performance is one contribution of the work in this paper.

This following paper is arranged as follows. We model the end-to-end transport layer vs. in-network caching transport model and analyze the service-curve of the TCP and UDP traffic by virtue of the network calculus as shown in section II. The combined TCP and UDP traffic performance evaluation is developed in section III. In detail we simulate our campus network topology in two scenarios. One scenario is that we use the different traffic flows amount over the same traffic combination. The other scenario is we employ different traffic combination type with the same traffic flows amount. The simulation results provide the references of adjusting the traffic combination in the network node to guarantee better throughput and low delay. Finally, we conclude the paper in section IV.

II. Mathematical Analysis

A. End-to-End Model

Our model is based on the following assumptions. A source node sends packets to a destination node over several intermediate nodes [5]. These nodes are connected by wired links that have a given packet drop scheme. In our model, the source node retransmits a packet if it is lost at the transport process. We assume that the number of transmission attempts is limited by a parameter N, which refers to the end-to-end transmission attempts from the source. In our model, we use the following set of definitions:

H : number of hops
N : maximum number of transmission attempts
p : probability of the link packet drop
P_s : successful transmission probability
P_f: failed transmission probability
A : number of link-level transmissions

We first determine the delivery probability over H hops with at most N attempts, denoted by P_s^H. We then derive $E(A)$, the expected number of transmissions expended on the delivery of a single packet. Then $E(A|Y=y)$, for $y \in [1, N]$ could determine the expected transmissions number given that there are Y transmission attempts. Suppose P_n^H be the probability that a packet is successfully transmitted with a maximum transmission limit of n attempts.

$$P_1^H = (1-p)^H .$$

$$P_s^H = 1-(1-P_1^H)^N = 1-(1-(1-p)^H)^N \tag{1}$$

In order to find $E(A|Y=y)$, we first derive P(Y=y)

$$P(Y=y) = \begin{cases} (1-(1-p)^H)^{y-1}(1-p)^H, 1 \le y < N \\ (1-(1-p)^H)^{N-1}, y = N \end{cases} \tag{2}$$

The number of transmissions in the successful case is equal to the number of the hops of the path: $E(A_y | Y=y) = H$

$$E(A) = \sum_{y=1}^{N} P(Y=y) E(A|Y=y) \tag{3}$$

We set the parameter p as 1%, 10%, 20%, 50% and 80%for the link packet drop probability to simulate the delivery ratio against the expected number of transmissions, as shown in Fig. 1. The number of path hops is set 5. The number of the most transmission attempts is 250. We could see how the packet delivery possibility varies with the number of attempts to transmit the data packets.

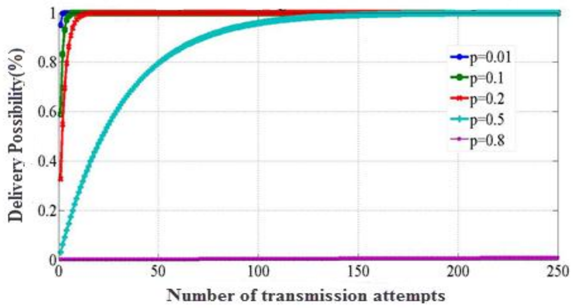
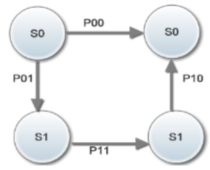


Figure 1. Delivery possibility against number of attempts.

B. In-network Cache-based Transport Model

As for the specific requested resource, within the in-network cache scenario [6]-[7], the transport model of a specific resource could satisfy the Markov property, which could make predictions for the future of the process based solely on its present resource conserve state and the previous one transport process of the specific resource, its future and past are independent.

We could build the Markov model as follows. Assume the request client named Creq, a client requests a source named Req_File, it must send interest packets to search the proper route for transporting Req_File. In this case, there are two general state of the intermediate nodes including conserved-state S1 and unsaved-state S0. Define the hop count of finding the first conserved-state S1 as H_Num, the previous transport process leading to state interchange can have direct effect on the resource achievement performance.



$$P = \begin{bmatrix} P_{00} & P_{01} \\ P_{10} & P_{11} \end{bmatrix}$$

Figure 2. Markov state transition.

The state transition is shown in the Fig. 2.

We employ the following set of definitions:

Creq	client which initial a request for a source
P_{suc}	the probability of successfully achieving the requested file
P_{si0}	the ith node does not have the copy of the requested file
P_{si1}	the ith node has the copy
H_Num	the hop number of the successful route
Avai_Num	the available routes number

$$P_{route} = 1/Avai\_Num$$

$$P_{suc} = P_{sH\_Num1} \prod_{i=0}^{H\_Num-1} P_{si0} * P_{route}$$

We simulate the attempts needed for successful transport process, as shown in Fig. 3.

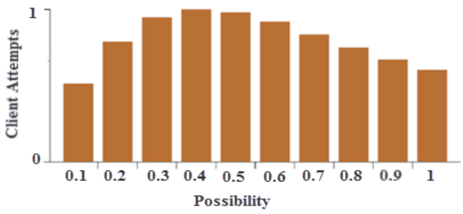


Figure 3. Clients attempts for achieving files.

C. Service-curve of TCP together with UDP

In the network calculus, there are three performance bounds about network consisting of data backlog bound, delay bound and output flow. As shown in Fig. 4, the $d(t)$ shows the delay bound and $R(t)-R^*(t)$ indicates the data backlog bound. The output flow depends on the arrival-curve and the service-curve [8]-[9].

Theorem 1. If a flow's arrival curve is α, the system service curve it crosses over is β, then, at any time t,

$$R(t)-R^*(t) \le \sup_{s\ge 0}\{\alpha(s)-\beta(s)\}$$

Theorem 2. If a flow's arrival curve is α, the system service curve it crosses over is β, then, at any time t,

$$d(t) \le h(\alpha,\beta)$$

Theorem 3. If a flow's arrival curve is α, the system service curve it crosses over is β, then output flow is limited as follows:

$$\alpha^* = \alpha \oslash \beta$$

According to these three theorem, we could deduce that the end-to-end transport model could be conceived as the concatenation of nodes. Which is defined in Theorem 4.

Theorem 4. If a flow crosses the systems S_1 and S_2, assume the service-curve of S_i is β_i, then a concatenation of these two systems will provide the service curve as follows:

$$\beta_1 \otimes \beta_2$$

According to theorem 4, we could know more about the model that if a packet starts from the source node, ends in the destination node. The path hop number is denoted by h, the first intermediate node provides the service curve as β_1, and then the ith intermediate node provides the service curve as β_i, $i \in [1, h-1]$. Therefore, the end-to-end transport model provides the formula (4).

$$\beta_1 \otimes \beta_2 \otimes \ldots \otimes \beta_i, 1 \le i \le h-1 \tag{4}$$

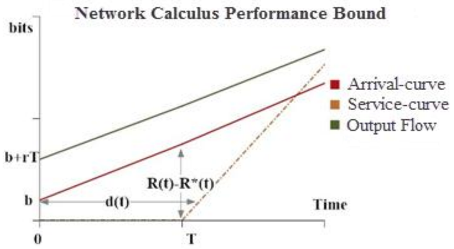


Figure 4. Arrival-curve, service-curve, output flow.

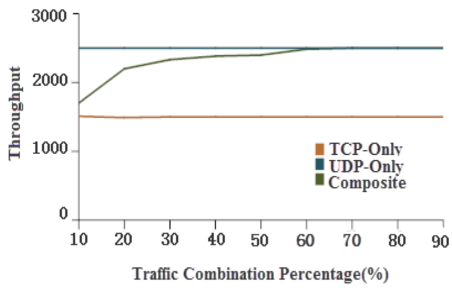
TCP and UDP are both end-to-end transport protocols and follows theorem of concatenation of nodes. TCP establishes definite connections while UDP directly transmit data packet without establishing connection. They have quite different characteristics in the operation. The composite TCP and UDP traffic will lead to different network performance. Various traffic combinations could be applied in various network resource condition scenarios [10]-[11]. We use the following set of definitions:

B:	the maximum bandwidth of the current network
Be:	current used bandwidth
CW:	congestion window size
RW:	receive window size
QL:	node queue length limitation

We could model the TCP together with UDP transport as the following formulas (5), (6) and (7). As shown in Fig. 5, we provide the throughput and delay bound for them.

$$TCP = \begin{cases} \beta_s^t = \beta_1^t \otimes \beta_2^t \otimes \ldots \otimes \beta_i^t, \quad 1 \le i \le h-1 \\ R(t)-R^*(t) \le \sup_{s\ge 0}\{\alpha(s)-\beta^t{}_s(s)\} \\ \qquad \le Min(CW,RW)+B-Be \\ d(t) \le h(\alpha,\beta^t{}_s) \le Timeout1 \\ \alpha^* = \alpha \oslash \beta^t{}_s \end{cases} \tag{5}$$

$$UDP = \begin{cases} \beta_s = \beta_1 \otimes \beta_2 \otimes \ldots \otimes \beta_i, 1 \le i \le h-1 \\ R(t)-R^*(t) \le \sup_{s\ge 0}\{\alpha(s)-\beta_s(s)\} \\ \qquad \le \sum_{n=1}^{h-1} QL_n + B - Be \\ d(t) \le h(\alpha,\beta_s) \le Timeout2 \\ \alpha^* = \alpha \oslash \beta_s \end{cases} \tag{6}$$



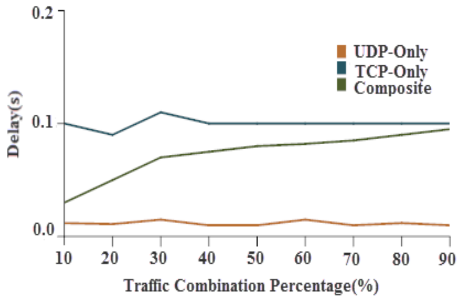


Figure 5. Throughput and delay bounds for TCP, UDP, composite TCP and UDP.

$$TCP+UDP=\begin{cases}\beta_s^m=\beta_1^m\otimes\beta_2^m\otimes\ldots\otimes\beta_i^m,\ \ 1\le i\le h-1\\ MB=Min(Min(CW,RW)+B-\\ \qquad Be\sum_{n=1}^{h-1}QL_n+B-Be)\\ R(t)-R^*(t)\le\sup_{s\ge 0}\left\{\alpha(s)-\beta_s^m(s)\right\}\le MB \ (7)\\ d(t)\le h(\alpha,\beta_s^m)\\ \qquad \le \min(Timeout1, Timeout2)\\ \alpha^*=\alpha\oslash\beta_s^m\end{cases}$$

III. PERFORMANCE EVALUATION

In this section, we select one typical network topology to evaluate the performance of the various traffic combination. Besides, we study various traffic quantities impact on the performance metrics. Through the simulation, we could study the network performance of various composite TCP together with UDP traffic flows more completely.

Each data transport starts from the source node, and forwards it by several intermediate nodes, then reaches the destination nodes. Each intermediate node could be seen as a system, which receives data packet and forwards it to the next hop. The end-to-end transport model could be seen as the concatenation of the path nodes during the communication process.

A. *Simulation Scenario*

As shown in Fig. 6, in this simulation scenario, we use two stub networks and one core network, one destination node. In the simulation run, the two stub networks respectively send TCP and UDP flows as illustrated in Fig.6. The core network is responsible for forwarding the flows to the destination node. We set the node number in the core network as 20, which is considered as a reasonable setting compared with our campus network topology [12]-[14].

TABLE I. LINKS METRICS

Items	Data Links			
	Stub1-n1 Stub2-n2	n1-core n2-core	Core-n3	n3-Destination
Bandwidth	2.5Mb	2Mb	2.5Mb	2.5Mb
Delay	10ms	10ms	10ms	10ms
DropScheme	DropTail	DropTail	DropTail	DropTail

The links metrics are listed in the Table I.

According to Table II, we change the traffic combination one by one and obtains the trace files of the various traffic combination. We evaluate the traffic characteristics from the aspect of throughput. The evaluation results are shown in Fig. 7, Fig. 8 and Fig. 9.

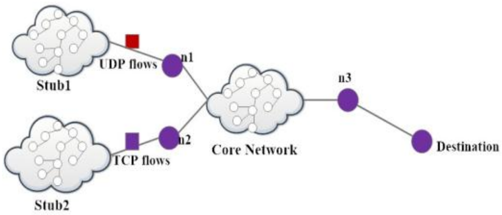


Figure 6. Simulation scenario.

TABLE II. TRAFFIC AMOUNT AND PERCENTAGE

TCP percentage		UDP percentage	
100%	45%	0%	55%
95%	40%	5%	60%
90%	35%	10%	65%
85%	30%	15%	70%
80%	25%	20%	75%
75%	20%	25%	80%
70%	15%	30%	85%
65%	10%	35%	90%
60%	5%	40%	95%
55%	0%	45%	100%
50%		50%	
Traffic flow amount		200	

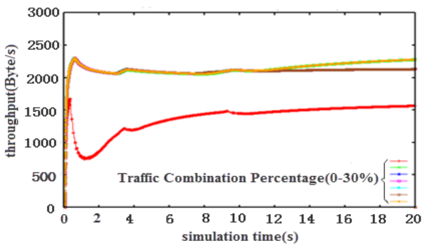


Figure 7. Throughput vs. percentage 0-30%.

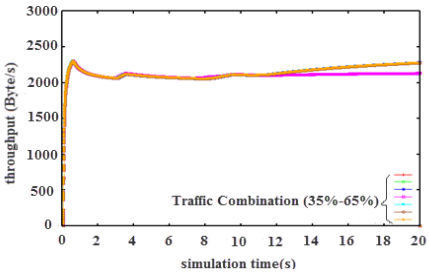


Figure 8. Throughput vs. percentage 35%-65%.

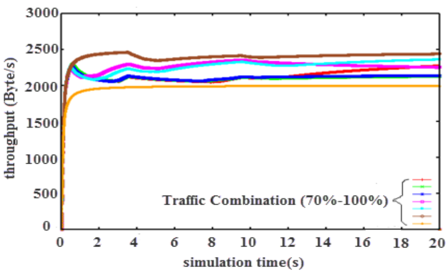


Figure 9. Throughput vs. percentage 70%-100%.

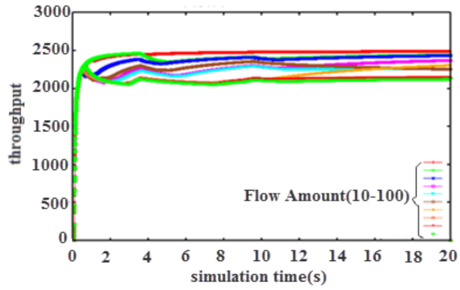


Figure 10. Throughput vs. flow amount (10-100).

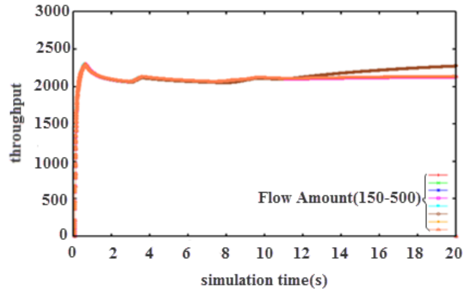


Figure 11. Throughput vs. flow amount (150-500).

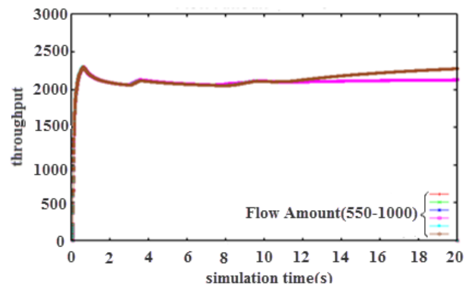


Figure 12. Throughput vs. flow amount (550-1000).

B. *Various Traffic Combination*

The simulation results of throughput against various traffic combination inform us of the fact that when there is no UDP traffic flow in the network, the throughput is the least in all the simulation results. When the network traffic eliminate the TCP traffic flow, we could find that the curve is the most stable and smooth in all the results. Apart from these two limitation cases, when UDP takes up 5%-65%, the curve changes are almost unified. When UDP flows takes up 70%-95%, the throughput changes much more. As we can see from the characteristics of throughput-single simulation results, we should keep the traffic combination of 30%-75% in order to maintain a stable, smooth and efficient throughput metrics.

C. *Various Traffic Flow Amount*

In the subsequent simulation, we change the traffic flow amount to test the throughput variation of the same traffic combination with 50% TCP flows and 50% UDP flows, which is one value feasible from the above simulation results. The amount of traffic flows range from 10 to 1000. As shown in Fig. 10, Fig. 11 and Fig. 12, the throughput varies with the traffic flow amount increases.

According to Table III, we change the traffic amount range from 10-1000 with the stable ratio 1:1 of TCP and UDP traffic. We evaluate the traffic characteristics from the aspect of throughput. The evaluation results are shown in Fig. 10, Fig. 11 and Fig. 12.

TABLE III. TRAFFIC AMOUNT INCREASE RANGE

Item	TCP:UDP	Total Flow Amount
Light weight traffic	1 : 1 50% : 50%	10
		20
		30
		40
		50
		60
		70
		80
		90
		100
Middle weight traffic	1 : 1 50% : 50%	150
		200
		250
		300
		350
		400
		450
		500
Large weight traffic	1 : 1 50% : 50%	550
		600
		700
		800
		900
		1000

The simulation of throughput against various flow amount acknowledges when the flow amount is very light-weight, we could see that the throughput changes a lot and it is not very stable, especially with enough network resource, it just depends on the running applications. With the increase of flow amount, the throughput becomes stable and the cures are unified from 150 flows to 1000 flows. Although there would unavoidably lead to some packet drop and congestion control leading to the delivery ratio changes, as we can preview, however in this paper, we just discuss the throughput performance for various traffic combination.

D. *End-to-End Delay vs. Percentage*

In addition to the throughput evaluation, we then analyze the end-to-end delay. Similarly, we first evaluate the end-to-end delay distribution against various traffic percentage, as shown in Fig. 13-15. The simulation results demonstrate that the end-to-end delays are acceptable between 40%-70%.

The delay distribution could be used to decide the performance level. As shown in Fig. 16, the simulation of average end-to-end delay against various percentage acknowledges when the percentage remains 70%-95%, the delay nearly stays the same level. The acceptable range could be 20%-65%.

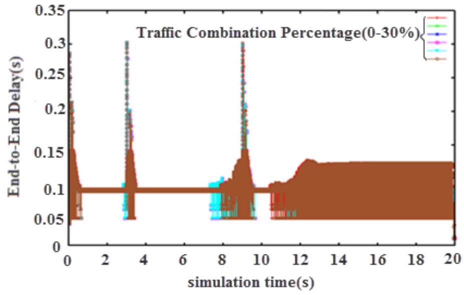


Figure 13. End-to-end delay vs. percentage (0-30%).

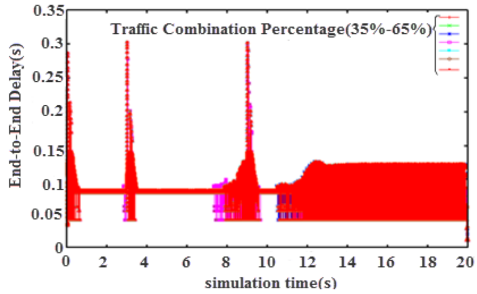


Figure 14. End-to-end delay vs. percentage (35%-65%).

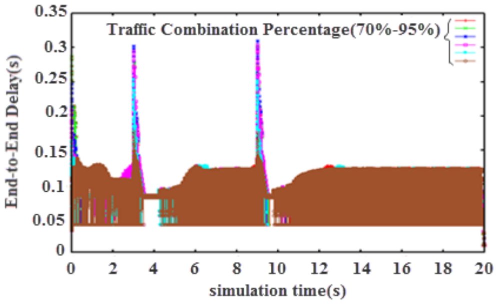


Figure 15. End-to-end delay vs. percentage (70-95%).

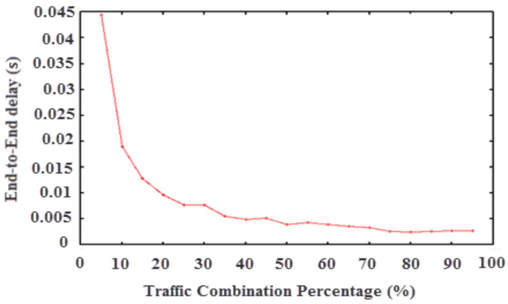


Figure 16. Average end-to-end delay vs. percentage (5%-95%).

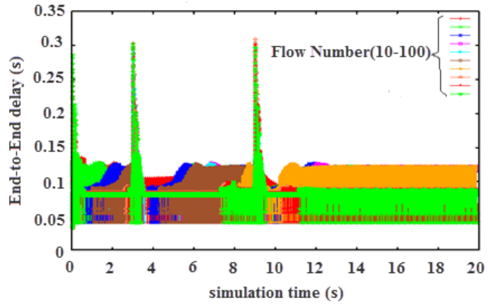


Figure 17. Delay distribution vs. flow number (10-100).

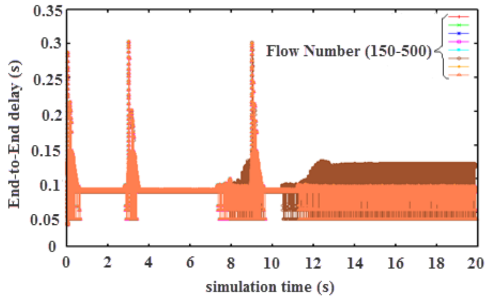


Figure 18. Delay distribution vs. flow number (150-500).

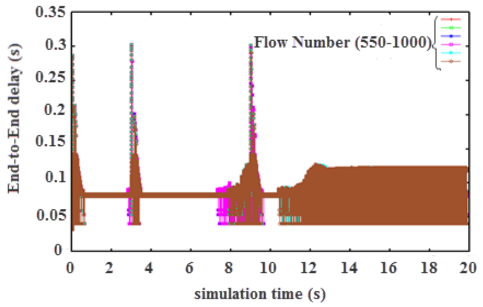


Figure 19. Delay distribution vs. flow number (550-1000).

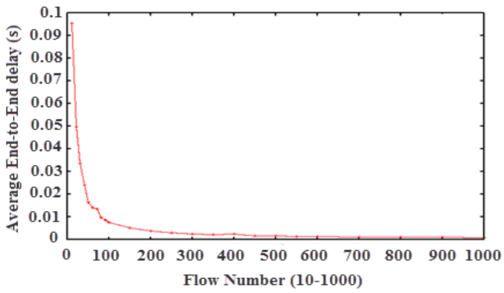


Figure 20. Average end-to-end delay vs. flow number.

E. End-to-End Delay vs. Flow Number

We evaluate the end-to-end delay distribution against various flow number, as shown in Fig. 17-19.

The simulation results demonstrate that the end-to-end delay almost remains the same since flow number is larger than 200. And the delays are acceptable between 200-1000. The delay distribution could be used to decide the performance level. In Fig. 20, the simulation of average end-to-end delay against various flow number acknowledges when the flows increase, the average delay is quite acceptable since the flow number is larger than 500.

F. Drop Rate vs. Percentage

As for packet drop rate, we evaluate its performance against percentage, which are shown in Fig. 21-23. The simulation results demonstrate that the packet drop rate increases with the percentage rises. When the percentage is above 65%, the drop rate is similar and stays well at the same level.

In Fig. 24, the simulation of the drop rate against the percentage again illustrates that the drop rate rises with the increase of percentage. And the increment becomes less and less. Therefore, the drop rate could be adjusted as low as possible for achieving low drop rate.

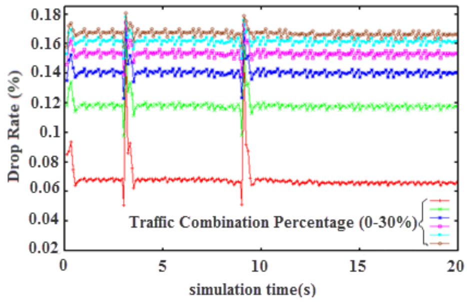


Figure 21. Drop rate vs. percentage (0-30%).

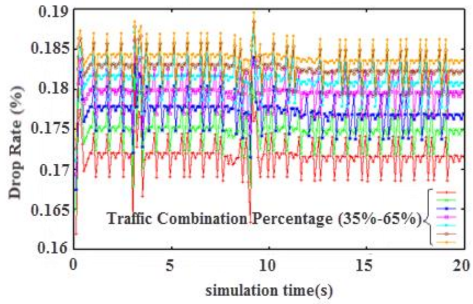


Figure 22. Drop rate vs. percentage (35-65%).

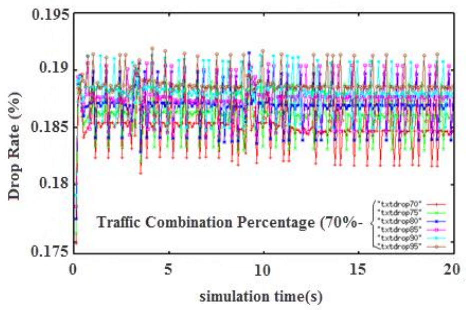


Figure 23. Drop rate vs. percentage (70-95%).

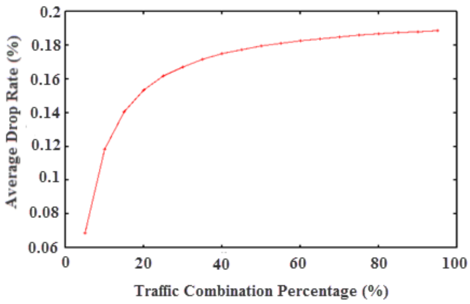


Figure 24. Drop rate vs. percentage (5-95%).

G. *Drop Rate vs. Flow Number*

In Fig. 25-27, they inform us that with the increase of the flow number, the drop rate increases and the drop rate jitter becomes larger as well. As a result of the flow number, the network condition becomes unstable.

In Fig. 28, the simulation of the drop rate against flow number again illustrates that the drop rate rises with the increase of the flow number. And the increment becomes less and less. Nevertheless, the total drop rate stays below 18% in the simulation result. Therefore, given the network simulation parameter set, the flow number should be controlled under 150.

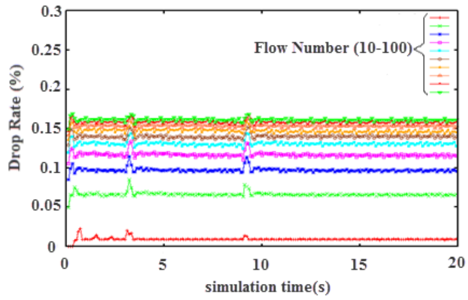


Figure 25. Drop rate vs. flow number (10-100).

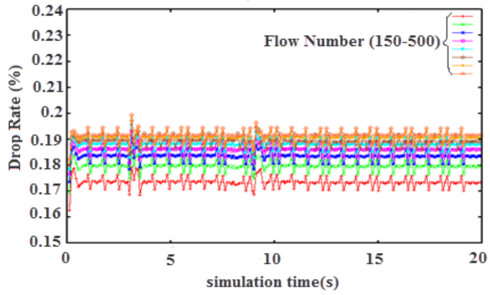


Figure 26. Drop rate vs. flow number (150-500).

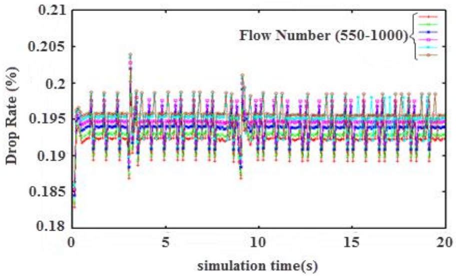


Figure 27. Drop rate vs. flow number (550-1000).

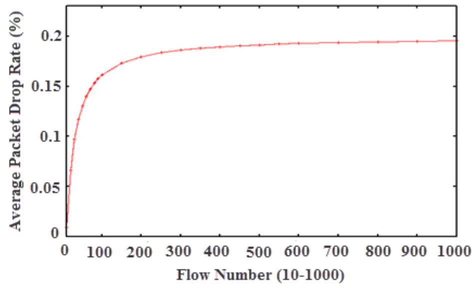


Figure 28. Average drop rate vs. flow number.

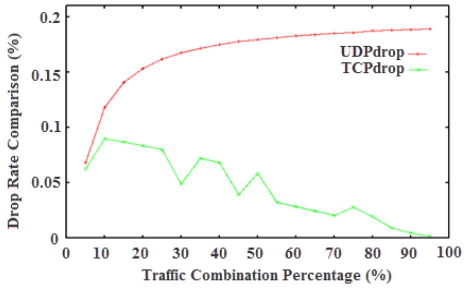


Figure 29. Drop rate comparison vs. percentage.

Apart from the above simulation of the packet drop rate, consequently, we explore how the TCP and UDP traffic perform respectively in the composite scenario. The Fig. 29 and Fig. 30 show the comparison results against percentage and flow number.

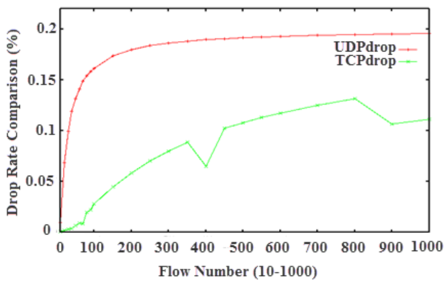


Figure 30. Drop rate comparison vs. flow number.

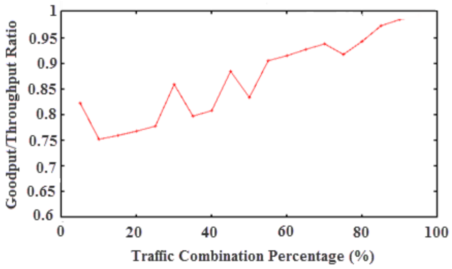


Figure 31. Goodput/throughput ratio vs. percentage.

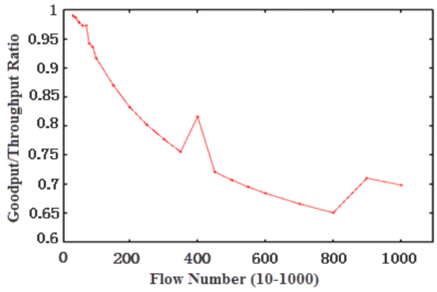


Figure 32. Goodput/throughput ratio vs. flow number.

H. GoodPut vs. Throughput

In Fig. 31 and Fig. 32, we analyze the gap between the throughput and goodput. The ratio of goodput to throughput could provide us the important characteristic of the network efficiency. Considered the results, we could learn that with the percentage rising, the ratio of goodput to throughput becomes larger which previews the network utilization is better. When the flow number becomes larger, the network performance could not stay well. And with the statistical results, we could adjust the flow number below 200 for the specific scenario.

IV. Conclusions

In this paper, we investigate the features of various composite TCP and UDP traffic combination. We model the end-to-end transport vs. in-network caching transport model and analyze the delivery probability against the transmission attempts. Then, we introduce the network calculus performance bounds to deduce the TCP and UDP traffic combination bounds model. Based on them, we deploy our simulation in two groups: (1) throughput, end-to-end delay, packet drop rate and resource efficiency against various traffic combination; (2) throughput, end-to-end delay, packet drop rate and resource efficiency under different flow amount over the same traffic combination. The simulation results further provide the reference for adjusting the node traffic combination to better guarantee the users' experience.

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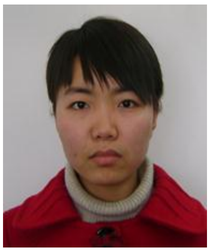

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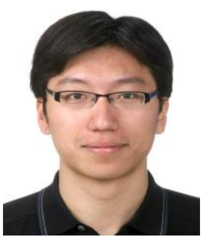

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