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Study of QoE protection mechanism based on network coding

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ABSTRACT

With the advent of information age, a large number of new multifarious information services are emerging, and requirements for information services are increasing. Hence, users' experience quality has become an important academic research topic. The appearance of network coding provides a new method for improving QoE. This paper pays much attention on how to utilize network coding technology for improving transmission efficiency of media streaming and heterogeneous network, and how to establish QoE protection mechanism of TCP. We compared TCP/NC with TCP Reno and TCP Vegas using network simulator NS-2, the experimental results show that the TCP/NC is effective and network coding is useful to improve QoE.

KEYWORDS

Network coding; QoE; Flow media; Heterogeneous networks; TCP.

INTRODUCTION

With the advent of the information age, new services of high diversity emerge continuously. The demand of people for services is growing higher. In order to obtain users' recognition for services, service providers must ensure a good quality of experience (QoE)[1-3], which is a kind of service evaluation method using users' recognition degree as criteria. Combines influence factors from service level, user level and environmental level, QoE directly reflects recognition degree of user for service. Ahlswede proposed the thought of network coding[4-6] based on the concept of network information flow, and from the viewpoint of information theory, they strictly proved that the maximum capacity of the communication network could be reached through network coding, and the existing network resources could be maximized. The proposition of network coding provides new solutions for how to improve users' QoE[7-10].

In the traditional communication network model, the only functions of router are storage and forwarding, but network coding breaks this routine, which combines several packets through algebraic method and sends them, and which decodes through utilizing relevance between packets. Hence, transmission rate is improved, and the upper limit of transmittable information of max-flow min-cut theorem could be realized. Many research results have shown that network coding could effectively resolve wireless network transmission problems, improve network throughput, decrease number of network transmission, save bandwidth, and increase information transmission security. Hence users' QoE could be effectively guaranteed.

Paper[10]show the random linear network coding applied to blocks of video frames can significantly simplify the packet requests at the network layer and save resources by avoiding duplicate packet reception and improve the QoE for video streaming applications. Paper[11] present novel network approaches based on Network Coding (NC) to address the main network impairments to user satisfaction in multimedia networks: slow delivery over wireless networks, QoE metrics such as buffering length and service interruptions. The experiment results show that NC provides resilience and efficiency in the network. Paper[12] introduce a reliable transport protocol using network coding to provide robustness against losses. Paper[13] analyze the trade-off between the usage cost and the QoE. Blocks are coded using random linear codes to alleviate the duplicate packet reception problem.

This paper carries out review on how to improve users' QoE through using network coding technology from the following three points: TCP performance improvement based on network coding, flow media transmission, and transmission technology of Heterogeneous Networks. The rest of this paper is organized as follows. In section II, we present the model of flow media QoE protection mechanism. In section III, we introduce the network coding of heterogeneous networks and the QoE model. In section IV, we give an overview of TCP/NC mechanism with network coding. We compared TCP/NC with TCP Reno and TCP Vegas in simulation to verify the effectiveness of TCP/NC. Section V, we conclude this paper.

FLOW MEDIA QOE PROTECTION MECHANISM BASED ON NETWORK CODING

P2P media streaming has very high requirements on the timeliness of data transmission, and excessive transmission delay will lead to serious decline of system overall performance and service quality. System packet loss rate, business throughput, delay and jitter have a certain impact on the customer experience. Delay mainly results in long-time video pause phenomenon, and jitter mainly results in video and audio pause and mosaic phenomenon. Network coding technology could fully utilize the computation ability of communication node for improving network efficiency.

In reference [14][15], the method of network coding application in streaming media is proposed, and Lava and R2 are designed. Network coding is first-timely used in streaming media system by Lava application, and it is added into streaming system as a plug-in. Experiment results show that network coding technology could effectively reduce the start playback delay of new-added nodes and improves the performance of streaming media live system. Adoptive random network coding algorithm is applied by ARLNCStream[16] system, new coding is dynamically adjusted and sent by each nodes according to the condition of source packet and encoding packet. Simulation results show that data block could be randomly and equally distributed among each node by this algorithm, which could reduce average delay and improve system robustness. Packet loss will occur if delay and jitter is big enough, but the buffer mechanism on client side could weaken delay and jitter. Data delay and jitter have been eclectically considered in reference[10], and corresponding QoE mechanism is proposed.

Paper[10]takes an analytical approach to study Quality of user Experience (QoE)for video streaming applications. They show that random linear network coding applied to blocks of video frames can significantly simplify the packet requests at the network layer and save resources by avoiding duplicate packet reception. At the application layer, the user requests blocks from the server. The application layer at the server feeds the requested block to the network layer at which the block is divided to multiple packets and sent to the user. Figure 1 illustrates this process.

There is a trade-off between the initial waiting time and likelihood of playback interruptions. There are the two of the key QoE metrics in paper[10]. The waiting time captures the delay aspect of the user experience, and the interruption probability captures the reliability aspect of the experience.

The first metric is the initial waiting time before the playback starts. The problem of streaming a finite media file to a single receiver over unreliable channel, modelled via a Poisson process of rate R. The dynamics of the receiver's buffer can be described as follows:

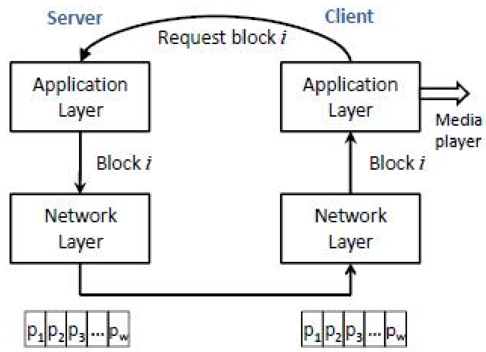

Figure 1: The structure of network coding layer

$$Q(t) = \max\{D + A(t) - t, 0\} \tag{1}$$

Where D is the initial buffer size and A (t) is a Poisson process of rate R. Hence, they declared an interruption in streaming when the buffer size reaches zero before reaching the end of the file. More precisely, let

$$T_e = \inf\{t: Q(t) \leq 0\} \tag{2}$$

$$T_f = \inf\{t: Q(t) \geq T - t$$

The video streaming is interrupted if and only if $T_e < T_f$.

The second metric that affects QoE is the probability of experiencing an interruption during the playback, which is denoted by

$$p(D) = P_r\{T_e < T_f\} \tag{3}$$

There is a fundamental trade-off between the interruption probability and the initial buffer size. For example, in order to have zero probability of interruption it is necessary to fully download the file, i.e., D=T.

Paper[10] is the first step in analytical characterization of QoE trade-offs in wireless media streaming applications. Many of the insights obtained from the presented simple model may be extended to a general class of arrival processes.

QOE METRICS IN HETEROGENEOUS NETWORKS BASED ON NETWORK CODING

A large variety of wireless technologies such as third-generation (3G), pre-4G known as Long Term Evolution (LTE) cellular and Wi-Fi/WLAN are being widely deployed with the success of wireless and mobile communication. Mobile devices with multiple wireless interfaces such as cellular and Wi-Fi are widely available in the market. Therefore, when such user equipment has access to multiple networks, it must take decisions on associating with one or more such access networks. A distributed random linear network coding[17][18] will remove the need of coordination between these heterogeneous networks and will eliminate the intelligence required in routing methods to avoid reception of duplicate packets from these networks. Add a network coding layer between the transport layer and the network layer of the protocol stack that provides a clean interface of network coding layer with transport layer. Figure 2 illustrates the network coding in heterogeneous networks.

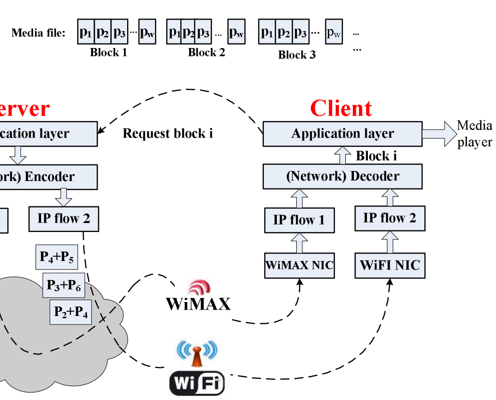

Figure 2 : Network coding in heterogeneous networks

Paper[13] analyses the trade-off between the usage cost and the Quality of user Experience (QoE). It models each access network as a server that provides packets to the user according to a Poisson process with a certain rate and cost. Blocks are coded using random linear codes to alleviate the duplicate packet reception problem.

They consider a media streaming system. There are two types of servers in the system: free servers and the costly ones. They combine all the free servers into one free server form which packets arrive according to a Poisson process of rate R_0, and merge all of the costly servers into one costly server with effective rate of R_c. The user's action at time t is denoted by $u_t \in \{0,1\}$, where u_t=0 if only the free server is used at time t, while $u_t = 1$ if both free and costly servers are used. Figure 3 illustrates the two classes of servers.

The dynamics of the receiver's buffer size x_t can be described as follows:

$$x_t = D + N_t + \int_0^t u_t \, dN_t^c - t \tag{4}$$

Where D is the initial buffer size, N_t Poisson processes of rate R_0 and N_t^c is a Poisson counter of rate R_c which is independent of the process N_t. The term t is unit rate of media playback.

Control Policy: Let $h_t = \{x_s: 0 \leq s \leq t\} \cup \{u_s: 0 \leq s < t\}$ denote the history of the buffer sizes and actions up to time t, and H be the set of all histories for all t. π is a mapping$\pi: H \rightarrow \{0,1\}$, where at any time t

$$\pi(h_t) = \begin{cases} 0, if\ only\ the\ free\ server\ is\ chosen \\ 1, if\ both\ servers\ are\ chosen. \end{cases} \tag{5}$$

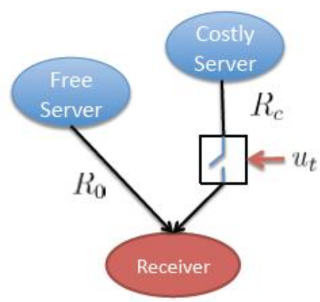

Figure 3 : Two classes of servers of the system

The first metric of QoE is the initial waiting time before the playback starts. This is directly captured by the initial buffer size D. Another metric that affects QoE is the probability of interruption during the playback for a particular control policy π denoted by

$$p^\pi(D) = P\{T_e < T_f \tag{6}$$

Where $T_e$and $T_f$are define in (2). The policy π is defined to be (D, ε), feasible if $p^\pi(D) \leq \varepsilon$
The third metric is the expected cost of using the costly server which is proportional to the expected usage time of the costly server. The usage cost of a (D, ε)-feasible policy π is given by

$$J^\pi(D, \epsilon) = E\left[\int_0^f u_t dt\right] \tag{7}$$

The value function or optimal cost function V is define as

$$V(D, \epsilon) = \min_{\pi \in \prod(D,\epsilon)} J^\pi\ (D, \epsilon) \tag{8}$$

In this model, there is a fundamental trade-off between the interruption probability ϵ, the initial buffer size D, and the usage cost. These trade-offs depend on the association policy as well as the system parameters $R_0$and R_c.

TCP USING NETWORK CODING

In wireless network, because of unstable network and limited resource, congestion losses and random losses of packet often occur, but TCP protocol could not distinguish the exact reason of packet losses. For packet losses not caused by congestion, TCP protocol still treats them through adopting strategy of reducing congestion window to reduce sending rate, but this method will decrease throughput and increase delay. In fact, for non-congestion losses caused by wireless transmission problems, the right method that should be TCP protocol adopted is improvement of sending rate instead of decrease of congestion window.

J.K.Sundararajan[19][20] first proposed TCP/NC protocol based on network coding, whose main thought is that a network coding (NC) level is added between TCP level and IP level but trying not to change the original TCP protocol stack, and encoding packet is formed after encoding of data packet utilizing network coding characteristics. Since information from many data packet is include into encoding packet, for any form of random encoding packet losses, the brought effect could be ignored. The simulation experiments shown that some packet losses could not cause overtime and decrease congestion window, which avoid of decrease of TCP performance. The structural diagram of network coding level is shown in Figure 4.

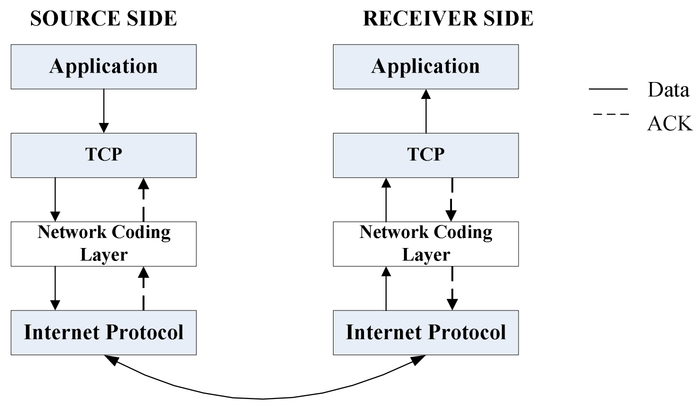

Figure 4 : Structural diagram of network coding level

Transmission terminal on NC level

NC level transmission terminal is composed of coding buffer module and coding window module. Coding buffer is used for preserving original data packet from TCP level transmission terminal, the length of data packet involved in encoding is adjusted to equal with the longest among them by adding zeros to their end. The main function of coding window is carrying out random linear encoding for data packet in coding buffer, and then transmits encoding packet to IP level.

NC level receiving terminal

The main function modules on NC level receiving terminal are decoding buffer and decoding matrix modules. Decoding buffer is used for preserving new received encoding packet and decoded data packets that have not been transmitted to TCP level receiving terminal. Gaussian elimination method is adopted by decoding matrix for decoding calculation of received encoding packets, and the original data could be acquired after receiving enough linearly independent encoding packets. The encoding header information of each new receiving encoding packet is extracted and analyzed, and the coefficient vector is saved in decoding matrix as a new row. When decoding matrix is transformed into its simplest model through Gaussian elimination method, the data packet corresponding to the matrix principle element column is seen packet, and obviously, the number of Seen packets is equal to the rank of decoding matrix as shown in Figure 5.

It is effective if new received encoding packet is linearly independent with existing encoding packet in current decoding buffer. The original packet could only be acquired if receiving terminal receives enough effective encoding packets or decoding matrix is full.

ACK acknowledgement mechanism and buffer management

Acknowledgement could not be carried out according to the byte number in the original packet after TCP/NC encoding, but the production of Seen packet is ordered and is consistent with the original packet sequence.

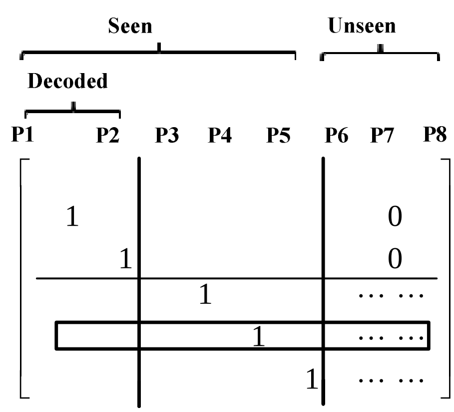

Figure 5: Schematic Diagram of Decoding Matrix

Hence, acknowledgement could be carried out according to the production of Seen packet. Gaussian elimination method will be executed for judging whether the packet is effective when TCP/NC receives an encoding packet. An ACK will be returned if this packet is effective, otherwise it is discarded. In a big enough finite field, the possibility of producing the next seen packet for each encoding packet is very high. Even if an encoding packet is lost because of channel error, the next arriving encoding packet will produce new seen packet on receiving terminal. The repeated ACK phenomena will not occur, and the incorrect invocation of congestion control mechanism will be avoided. ACK acknowledgement mechanism is shown in Figure 6.

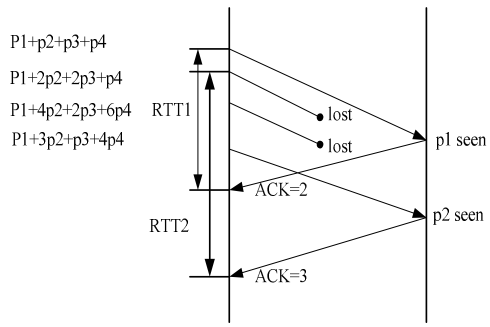

Figure 6: ACK acknowledgement mechanisms

The network coding function expansion is carried out based on the existing TCP protocol. Under condition of maintaining independence between each levels of protocol stack and maintaining TCP end-to-end characteristics, even if non-congestion discard occurs, the incorrect invocation of congestion control mechanism by TCP protocol could be avoided. The benefit of network coding of masking the losses in the network to achieve fast transmission of data and to avoid retransmissions of lost data gives another choice for implementing network coding at the link layer. With the approach of implementation the LTE network and the WLAN network can also take part in network coding.

Performance analysis for TCP/NC

1). Throughput model

The receiving terminal of TCP/NC sends confirmation message to source terminal according to its receiving efficient encoding packet. As long as the selection of encoding coefficient is under a large enough finite fields, there is high possibility for production of the next visible packet for each encoding packet. Even if the packet loss occurs because of channel error, the new visible packet will be produced by receiving terminal when arrives the next encoding packet. Hence, the repeated acknowledgement phenomenon will not occur, and the mistakenly calling of congestion control mechanism could be avoided. According to analysis in reference[21], when terminal-to-terminal TCP/NC protocol is used as transmission control protocol, its average throughput could be calculated by the following formula:

$$T = \frac{1-p}{R * SRTT} W_{\max} \quad \textbf{(9)}$$

Where, P is the packet loss rate, R is the redundant factor of TCP/NC， SRTT is the effective round-trip time, and $W_{\max}$ is the size of maximum window.

2). Performance simulation for TCP/NC

Simulations of throughput with the variation of packet loss rate for TCP Reno, TCP Vegas and TCP/NC are respectively carried out in NS platform [22], and the network topology is shown in Figure 7. The simulation period is 500s.

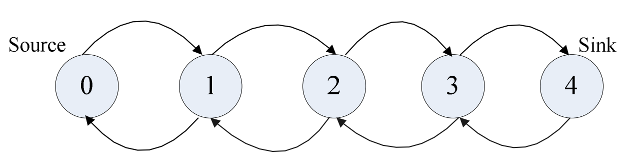

Figure 7: Structure chart of network simulation

(1). Throughput

As shown in Figure 8, when the network packet loss rate is zero, the throughput of TCP/NC is slightly lower than that of TCP Reno and TCP Vegas; when packet loss rate reaches 0.01, throughput of TCP Reno and TCP Vegas rapidly decrease; when packet loss rate further increase, there throughput is almost zero. In comparison, instead of the rapidly decrease with the increase of packet loss rate, throughput of TCP/NC protocol declines with moderate trend, and hence, its throughput is significantly better than that of TCP Reno and TCP Vegas[22].

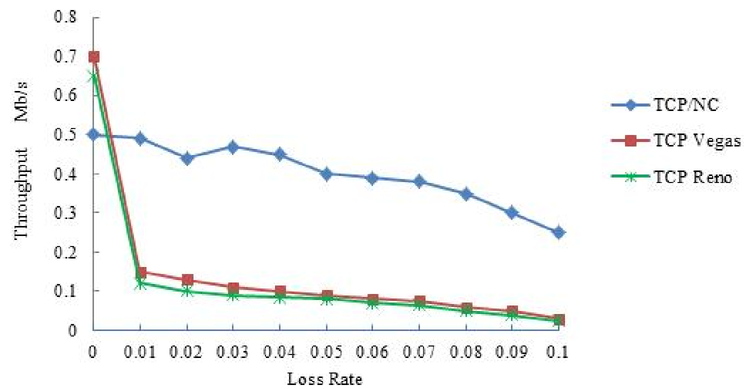

Figure 8: TCP throughput simulation value

(2). Redundant factor

As pointed out in reference[21], the coding and decoding could be successfully carried out if R>=1/(1-p), but how to choose the value of R for optimum performance of TCP? We will explore the impact of redundant factor on network coding performance through simulation[22]. Assuming that parameters of TCP level are chosen as follow: W_{max} =100, RTT=0.8s, and 1packet=1000B, changes of TCP/NC throughput during the variation process of redundant factor from 1 to 2 will be studied respectively under three different situations: B=1Mb/s, p=10%; B=1Mb/s, p=25%; and B=2Mb/s, p=25%. Simulation results are shown in Figure 9.

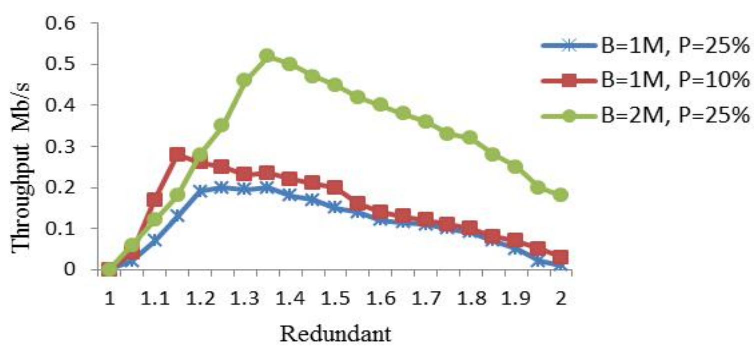

Figure 9: Redundant factors on network coding performance

As shown in Figure 9, if B=1Mb/s and p=10%, the maximum value of TCP/NC throughput could be acquired when R=1.15; if B=1Mb/s and p=25%, the maximum value of TCP/NC throughput could be acquired when R=1.25; and if B=2Mb/s and p=25%, the maximum value of TCP/NC throughput could be acquired when R=1.35. When redundant factor is too large, TCP/NC throughput begins to decline.

If the link bandwidth is small, the value of R should be equal of slightly larger than 1/(1-p), and if link bandwidth is large or packet loss rate is high, a conveniently larger R or W_{max} could be chosen for more efficient usage of bandwidth.

CONCLUSIONS

In this paper, we studied the problem how network coding can provide the current networks with the mechanisms necessary to guarantee end users QoE. More and more devices are becoming multimedia enabled including traditional television sets, future work will aim at creating strategies for operators and end users to provide the best experience in cloud network, mobile cloud and the Internet of Things and Content.

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