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퍼지이론과 예증을 이용한 WebRTC환경의 로컬 네트워크 속도 조정

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요 약

WebRTC는 내부 및 외부 플러그인 없이 브라우저간의 음성 전화, 비디오 채팅, 및 p2p파일 공유를 지원하는 최신 기술 중 하나이다. 그러나 아직 다뤄져야 할 많은 문제가 있다. 이 논문에서는 그 중 대역폭 이라는 작은 필드에 초점을 맞췄다. 다운로드 및 업로드를 위한 대역폭은 서비스 제공자에 의해 고정되어있지만, 특정 지역에 있는 사용자의 수는 시간에 따라 크게 증가되고 있다. 본 논문에서는 대역폭의 한계를 극복하기 위하여 퍼지 컨트롤을 기반으로 클라이언트 자체에서 프레임 비율 및 스트리밍 비디오의 해상도 변경하는 모델을 제안한다.

키워드 : 퍼지 이론, 대역폭 공유, 대역폭 제어, 스트리밍 비디오

Adjusting Local Network Speed by Using Fuzzy Theory with An Illustration in WebRTC Environment

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Abstract

WebRTC is one of the most recent technologies that supports browser-to-browser for voice calling, video chat, and P2P file sharing without the need of either internal or external plugins. However, there has some limitation which lets our development deal with many problems. This research will focus just on a small field of that problem which is bandwidth. While the bandwidth for downloading and uploading is fixed by service providers, but the number of users in a certain area is increasing largely by the time. In this paper, we propose a model to overcome the limitation of the bandwidth based on fuzzy control to adjust utilized bandwidth by changing frame rate and resolution of streaming video in the client itself.

Keywords : Fuzzy theory, Sharing bandwidth, Control bandwidth, Streaming video

1. Introduction

Along with the development of the world, technologies are playing in an important part, especially Network communication technology. As a past, we develop many technology for better video streaming, interactive human

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system. From that, we have more and more successive steps on real time communication which is very useful in tele-presence, tele-health, tele-conference, etc... In a recent year, Google has announced their new technology for Real Time Communication on the web browser without any installation and plug-in, meanwhile desktop and mobile clients are under development. Because of its convenience, many research involved in exploratory, descriptive, and explanatory the framework. At first, we had the analyzed of receive-side real-time congestion control in WebRTC [1] for showing the performant of the congestion control. In addition, the implementing of live video streaming protocols into web applications with the use of WebRTC [2], the P2P-MCU approach to multi-party video conference with WebRTC [3], and the analyzed of challenges of video conferencing services in WebRTC [4] showed that WebRTC is really a powerful framework for developing real time application. Thus, many companies are developing and building many applications based on WebRTC architecture. For example, tele-service is commonly used among far distance end-users with high quality and interactive system. By the limitation of network infrastructure, we cannot develop beyond the network obstruction. Suppose the situation that, in a local area network only has a 5x5 internet connection. We had two persons with 2 subscribed streams coming in just fine (perfect clarity) and the 3rd one was coming in audio-only because of bandwidth issues. But everybody was publishing and subscribing while physically sitting in the same room on the same network. The problem was that the first 2 publishers and subscribers had already completely saturated the network by the time the 3rd person came on. If we were able to set an absolute bandwidth use cap on all the subscribed streams, this situation would not happen. However, this solution isn't flexible and it might cause redundant network bandwidth, the bandwidth isn't maximized usable. Furthermore, control bandwidth while the connection is established by season description protocol in webRTC, isn't easy. Currently, we have no API to do so. To solve the problem, we have a several ways for controlling bandwidth. In this paper, we try to control bandwidth by reducing the frame rate and resolution of video. While we can reduce device's CPU usage by reducing the frame rate, upon the result, bandwidth for transferring data will decrease correspondingly.

In order to improve the quality of service [5] showed that if we can handle the congestion control, media codec, we can reduce the lateness of arrival package. It is very meaningful for low bandwidth with the low rate of sending and receiving packages. Additionally, [6] analyzed the Bundle technology in the browser based WebRTC, in which, Bundle groups multiple transport connections between peers into a single connection and reduces the data transmission, upon on that, bandwidth utilization will be reduced too. But these methods use much computing resource and cause package loss.

Recently, fuzzy control is somehow used in the Intelligence Control System. It's based on the human mind for making a decision. Ideally, the problem is solved based on the feeling of our sense, whenever the utilized of each stream is greater than each other or the upcoming stream, its bandwidth will be shared by an adjustment of resolution or frame rate with applyConstraints() method in MediaStreamTrack WebRTC API.

In the next section, we will discuss some researches which relate to our research, such as fuzzy control, bandwidth control and WebRTC technology. Furthermore, at the section 3, an overview system will give you in detail of our system design which includes a negotiating algorithm. Finally, an experiment using OPNET demonstrates how system balances the network resources used.

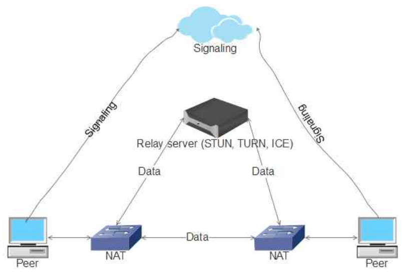
2. Related Research

A vertical handover decision algorithm based on fuzzy control theory in [7], took into consider the factor of cost, power level, bandwidth in wireless communication networks. They calculate the membership degree of the input factors and normalize bandwidth for each Base Station. They result showed that the algorithm realizes the optimized vertical handover by evaluating and analyzing various input parameters. Regarding the control of fuzzy logic system [8], presented a connection admission control. It provides an interval decision, so that a soft-decision can be made based on a design tradeoff between cell loss ratio and bandwidth utilization, which is impossible for the hard decision boundary. In addition, Scheduling research [9] in a cloud environment with many features such as large scale, diversity, and heterogeneity. The user requirements for cloud computing resources are commonly characterized by uncertainty and imprecision. Also Many users generally cannot give a precise requirement in accordance with the attitude of the tasks. They commonly generate an imprecise requirement according to the character of the task. The ability to satisfy these fuzzy requirements is an important manifestation of the friendliness of cloud computing. Therefore, they propose a dynamic resource scheduling method based on fuzzy control theory with resource requirements prediction, and the relationships between resource availability and the resource requirements. Furthermore, Leveraging WebRTC for P2P Content Distribution in Web Browsers [10], present a distributed content sharing facility using WebRTC Data Channels as well as an emulation component for test and measurement purposes. In the performance of a network like the Internet handling so-called elastic traffic where the rate of flows adjusts to fill available bandwidth [11], [12] assume traffic consists of point to point transfers of individual documents of finite size arriving according to a Poisson process. Notable results are that weighted sharing has limited impact on perceived quality of service and that discrimination in favour of short documents leads to considerably better performance than fair sharing.

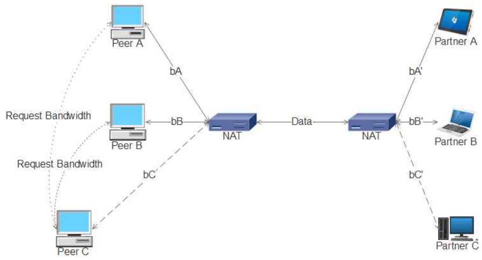
3. System Overview and Algorithm

In the case of webRTC, every participant will join as a peer. Thus, each participant has an equality to take network resources. Based on the quality of the network, WebRTC will automatically deliver resources to each peer. In (Figure 1), whenever a peer connection is established, other factors excepts those peers, can't be re-configured (Signaling in WebRTC is just a machine by which peers send control message to each other to establishing a communication protocol). Therefore, we propose a model where peers negotiated bandwidth with each other rather than using server for limit the transportation package. In the figure (Figure 2) shows that, device A and B are using network by using WebRTC to share their data (video, audio) with their partners and their utilized bandwidth is bA and bB respectively, by the time device C join into the network. It only gets the audio signal which is described the dash arrow with bandwidth bC, it has no streaming video because of bandwidth limitation. Thus, device C will communicate with A and B for negotiating network resources.

(Figure 1) Connection between peers in WebRTC overview



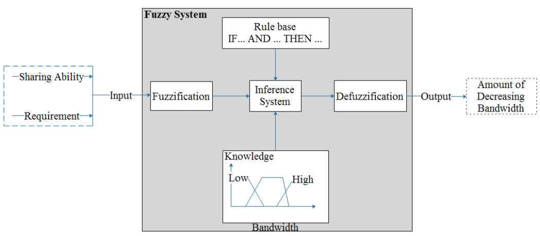
(Figure 2) Requesting bandwidth overview



3.1 Fuzzy Control Systems

Generally, the system has two input variables and one output variable which is shown in (Figure 3). After receiving the input the system will change those scalar values into a fuzzy value, the process is so called Fuzzification. Then, the human knowledge and rule based are used as a constraint of the system to produce output in Deffuzification process. In the next section we will discuss in details of the system using Mamdani Fuzzy Models.

(Figure 3) A design of fuzzy system for controlling bandwidth utilization



3.1.1 Fuzzification

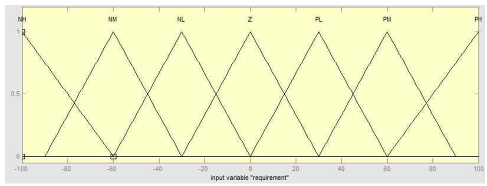
Assuming that $_{sage}$ is a current utilized network resource of the paired device, R_{usage} is a current utilized network resource of the new coming peered (the peered can be in video streaming or only voice), and $M\ n_{usage}$ is minimum bandwidth or number of bytes per second can be sent, to get video streaming. We will form scalar values for the input system by using two equations below.

$$S = \frac{Min_{usage} - S_{usage}}{S_{usage}} \times 100. \quad (1)$$

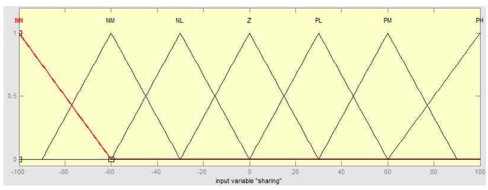
$$R = \frac{Min_{usage} - R_{usage}}{Min_{usage}} \times 100. \quad (2)$$

The sharing ability variable in (1) and requirement variable in (2), range from -100 to 100, the positive value means it need more bandwidth and can't share resources to others. Otherwise, it can reduce its bandwidth usage. The Triangularshaped membership functions of the system input are defined in (Figure 4, 5).

(Figure 4) Requirement membership



(Figure 5) Sharing membership function



3.1.2 Fuzzy Inference Model

Referring to the levels of the peers which have utilized bandwidth and the required network resource of new upcoming peers. We define the symbol as positive or negative in accordance with the increasing or decreasing network resources. The input variables are the requirements for resources and the sharing ability resources of the established peers, the output variables are the resources that the new peer need to be increased or current peers need to be decreased frame rate or resolution of streaming video. The fuzzy values of the input and output variables are described as Positive High, Positive Medium, Positive Low, Zero, Negative Low, Negative Medium, Negative High or their abbreviation words are PH, PM, PL, N, NL, NM, NH by descending order.

Naming the bandwidth requirement for the new peer is bR and the sharing ability bandwidth of established peer is bS, which are the input of the system, and the output of system is named as bandwidth output bO which is the ability the pairs can decrease frame rate or video streaming resolution. Therefore, we can express the set of rule as follows:

If bR is PH and bS is NH, then bO is NH

(Figure 6) Request bandwidth based on fuzzy rules

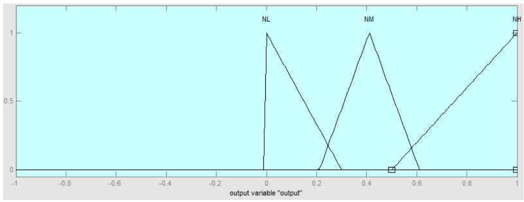
bR/bS	NH	NM	NL	N	PL	PM	PH
NH	N	N	N	N	N	N	N
NM	N	N	N	N	N	N	N
NL	N	N	N	N	N	N	N
N	**NL**	**NL**	N	N	N	N	N
PL	**NL**	**NL**	N	N	N	N	N
PM	**NM**	**NM**	**NL**	**NL**	N	N	N
PH	**NH**	**NH**	**NM**	**NL**	N	N	N

If the peer need high bandwidth and the established peer doesn't need spending much bandwidth, then it will reduce the frame with high rate. Probably, we do nothing if bR is kind of Negative because we don't want to decrease the bandwidth of upcoming peer. Also in the case bS is kind of Positive which means the established peers need more bandwidth, they can't share bandwidth with others. Thus, the output is N (do nothing). The discussion of those rules between input and output are shown in (Figure 6).

3.1.3 Defuzzification

The system will calculate based on human knowledge which is set of rules then produce output variable with the triangular shaped membership function of the system output is defined in (Figure 7), in the interval [0, 1]. But it still a fuzzy variable, therefore the real value of decreasing value is calculated by (3) with *Dec* is real output of the system.

(Figure 7) Output membership function



$$c = (S_{\ sage} - Min_{usage}) \times output. \qquad (3)$$

In the system, the inference process uses max-min inference method to produce output variable, it can be formed as

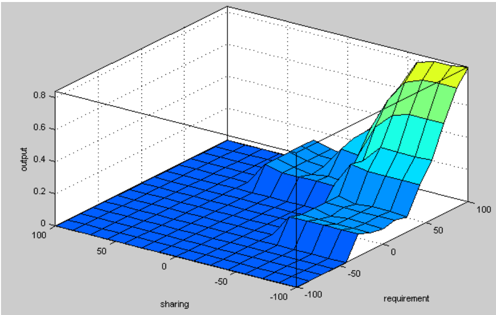
$$\mu^i_{output} = \max\ \min\left[\mu^i, \mu^i_R\right]\right]. \qquad (4)$$

,where μ^i_{output}, μ^i_S, μ^i_R is a truth value of result membership function corresponding to input value i. After getting the value of μ^i_{output}, we use centroid defuzzification method to calculate output variable. The relationship between input and output is shown in (Figure 8).

We will prove that our formulas aren't going wrong in the worst cases. In (1) if $S_{usage} < Min_{usage}$ which means paired devices can't reach the bandwidth requirement of video streaming, the variable S has a positive value, also in (2) if

$n\ _{sage} < R_{usage}$, the variable R has a negative value. We will do nothing in the both of cases as shown in the previous section.

(Figure 8) Relationship between input and output



Furthermore, if sharing peer can decrease its bandwidth (frame rate or video solution) with value is calculated in (3), it can't decrease to the below of video streaming requirement Min_{usage}. Assuming that $_{sage}\ Dec < Min_{usage}$, replacing value in (3) we have $S_{usage} - (S_{usage} - Min_{usage}) \times output < Min_{usage}$ $\Leftrightarrow S_{usage} - Min_{usage} < (S_{usage} - Min_{usage}) \times output$. The pairs devices can share their resources, thus $_{sage}\ \ Min_{usage} \Rightarrow S_{usage} - Min_{usage} > 0$, and our assumption becomes $output > 1$. This contradicts our definition that $output\ \ [0,1]$.

3.2 Negotiating algorithm

The process for negotiating bandwidth resource is a dynamic algorithm based on the request of new coming peers and the present peers, which is described in (Figure 9) with following steps,

Step 1: Starting video streaming via WebRTC and make fuzzy estimation of bandwidth resource.

Step 2: Sending request resource to other peer in the same network if need.

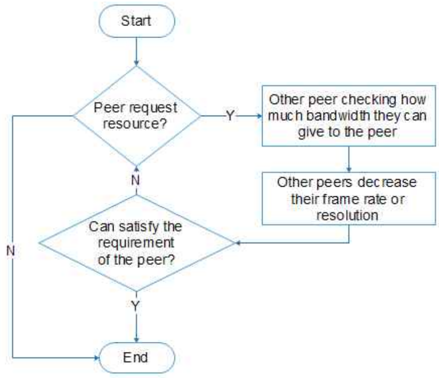
Step 3: Other peers checking their current usage resource and how much they can share by fuzzy variable.

Step 4: Other peers decease their frame or resolution based on previous step.

Step 5: Others peers send responding message to tell that they have been shared bandwidth, the peer checks the satisfy bandwidth condition aiming whether need to send request again or not.

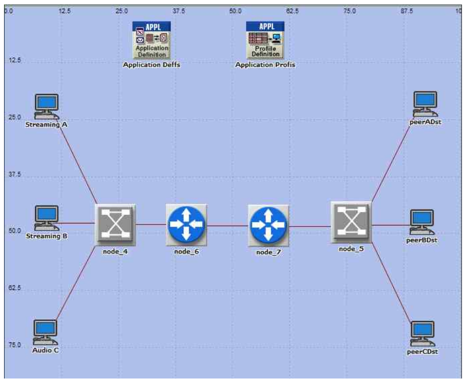
Step 6: Ending the negotiating process.

(Figure 9) Negotiating algorithm diagram between peers in Local Network



4. Simulation experiments

(Figure 10) A design of simulation network in OPNET



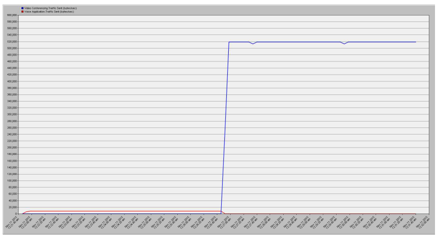
We will use OPNET as a tool in the experiment. (Figure 10) shows that, Streaming A streams with peerADst with Video Conferencing (Frame interarrival time: 30 frames/sec, Frame Size Information: 352*240), Streaming B streams with peerBDst with Video Conferencing (Frame interarrival time: 30 frames/sec, Frame Size Information: 352*240), Audio C streams with peerCDst with Voice (PCM Quality Speech).

The simulation processing is shown in the Table 1, in which Audio C will have Video Streaming after two steps for both A and B reduces their resolutions and frame rates.

<Table 1> Simulation step processing

Step	Stream A	Stream B	Stream C
Origin	Video Conferencing (30 frames/sec, 352*240 pixels)	Video Conferencing (30 frames/sec, 352*240 pixels)	Voice (PCM Quality Speech
Step 1	Video Conferencing (15 frames/sec, 128*240 pixels)	Not Change	Not Change
Step 2	Video Conferencing (10 frames/sec, 128*120 pixels)	Video Conferencing (30 frames/sec, 128*120 pixels)	Video Conferencing (30 frames/sec, 128*240 pixels)

(Figure 11) Packages sending rate in streaming of Device C



(Figure 12) Packages sending rate in streaming of Device A, B and C

In the experiment result which is shown in (Figure 11), package sending rate is became greater after a negotiating time between Streaming A and B. Also in (Figure 12), the sending package from A and B, are reduced due to the reduced of frame rate and video resolution.

Generally, the result of this experiment shows that the fuzzy system control can evaluate amount of both sharing and requesting bandwidth, available resource always shares to a corresponding request to improve the quality service of the group which shows in Table 1 by the differences between the original and the final step 2. Our method just focuses on the client itself, whenever one client needs, it sends a request to others in the local, this action does not need any involved actions from the server or a management network control system which makes the sample of the network where we already have a complex system.

5. Conclusion

In this paper, we proposed a Balancing bandwidth method by using fuzzy control in WebRTC technology to overcome the inherent of network infrastructure with a bandwidth limitation which is fixed by providers, in the local area such as an office or school. Based on human senses, the amount of bandwidth for those took much resources need to be decreased as much as they can. By using resources usage negotiated algorithm for local users, we can have a flexible network with a better quality service.

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