SIP-based User-Oriented Handoff Control for Video Phone

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Abstract

The goal of the research is to reestablish the session quickly for moving users according to a suit of user-oriented seamless handoff control framework （USHC） the prior researches mentioned. Furthermore, the model about making use of SIP with offer/answer, which is rested on two methods consisting of the REFER with Quality Negotiation and RS-based 3PCC session mobility controls adopted for inter-domain handoff and intra-domain handoff respectively, is to provide the service of video transmitting and defines the detail SDP exchanged during handoff. Besides, we analyzed the situation that the underflow and overflow case in the procedure of RS-based 3PCC and REFER with Quality Negotiation respectively caused by network delay during the procedure of handoff and set up two kinds of thresholds to detect the problem about network delay. In the conclusion, we state the future work about audio/video synchronization to verify whether the theorem is correct or not.

1. Introduction

With the prosperity and popularization of the internet network, the exchange of information is more and more fast. The application of multimedia is more verity with the maturity of internet technology and improvement of equipment. Consequently, the request of service quality for users is getting much higher. Voice Over IP (VoIP) is a service about transmitting speech data through internet. Handley M et al. [6][9] proposed the Session Initiation Protocol (SIP) in 1999. The protocol used in setting-up, maintain and cut off of the multimedia session, extensively make uses of service about VoIP at present. SIP is a simple and elastic protocol, because the operation of SIP focuses on application layer mainly, different network types, operating systems and hardware specifications will not influence its operation.

In addition, the terminal host has characteristic of mobility in the environment of the wireless network. Therefore, when the terminal host moves in different Access Points (APs) or different domain, it will cause the disconnection between AP and the terminal host. If the time of disconnection is too long, it will influence seriously the comfortable for users. Moreover, because of un-prediction of network, the situations about packet delay and packet loss may occur in the transmitting process of multimedia data. Packet delay will make the jitter and skew. Therefore, the situation of asynchronous communication about intra-stream and inter-stream makes the multimedia to play un-smoothly and decreases the quality of the video.

Saha D et al. [10] addressed traditional mobility management and divided mobility management into two parts. The one is location management and the other is handoff management. In addition, they divided the present existing mobility protocol into three kinds, including micromobility, macromobility and global mobility, introduced the operation one by one and analyzed the comparison of the location update, the handoff latency and QoS. Kwon T.T et al. [8] came to explore different mobility managerial structures from the view of VoIP and focused on Mobile IP and SIP. Kwon T.T et al. proposed the method, shadow registration, to reduce the service delay of VoIP while carrying on macromobility. The result of research found the SIP delay time was longer among a small scale, but the delay time to handoff on a large scale was still better in MIP. Banerjee N et al. [1] discussed to support macromobily in the heterogeneous network. As the process of moving to cause handoff in the heterogeneous network, it will suffer from the unpredictable network delay and packet loss. Thus, it will debase the quality of service on application layer, such as VoIP, streaming video. This research proposed a structure called SIP Back to Back User Agent about moving in heterogeneous network, and the purpose

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was to reduce the packet loss. Hac and Xue [4]presented a completed system in controlling media unit and proposed three kinds of buffer control algorithm , including Switch, First order delay and Second order delay. It used the comparator, regulator and corrector, and proposed the method of feedback. Additionally, the research defined three kinds of states according to the degrees of threshold to control smoothly the discard and duplication of frames in buffer.

In order to solve above-mentioned handoff structure, this paper based on a suit of (USHC) which is on the view of user to reestablish sessions quickly while users moving and combines offer/answer with SIP [7]to provide the service of video transmission, including RS-based 3PCC and REFER with quality negotiation. This paper is organized as below. The part 2 details the UHCF. The part 3 analyses the UHCF in all different situations. The part 4 makes a conclusion.

2. User-oriented Handoff Control Framework (USHC) UHCF

The achievement of UHCF is divided into two stages. The first stage is RS-based Third Party Call Control method (RS-based 3PCC)[2] which is applied in intra-domain. The method which extends from third Party Call Control (3PCC) utilizes mainly relay session to transmit packets and establishes the new session at the same time. Besides, it adopts an offer/answer model with SDP[3][5][11] to achieve the purpose of multimedia communication. During the process of establishing new sessions, it will exchange related SDP. The process is detailed in the figure 1. The second stage is REFER with Quality Negotiation which is applied in inter-domain. This paper mainly utilizes refer-to header specified in REF~~RE~~ method to re-establish the new connection of the new user agent (N_UA) and the corresponding agent (C_UA) while the mobile user is moving from M_UA to N_UA. To maintain the conversation comfortable degree and reduce handoff delay since connection establishment must be fulfilled before the user arrives. The process is detailed in the figure 1.

3. The Analysis of USHC

3.1 The packet analysis of RS-based 3PCC

We consider two kinds of extreme instances. The situations are that RS-3PCC occurs underflow and overflow which details below.

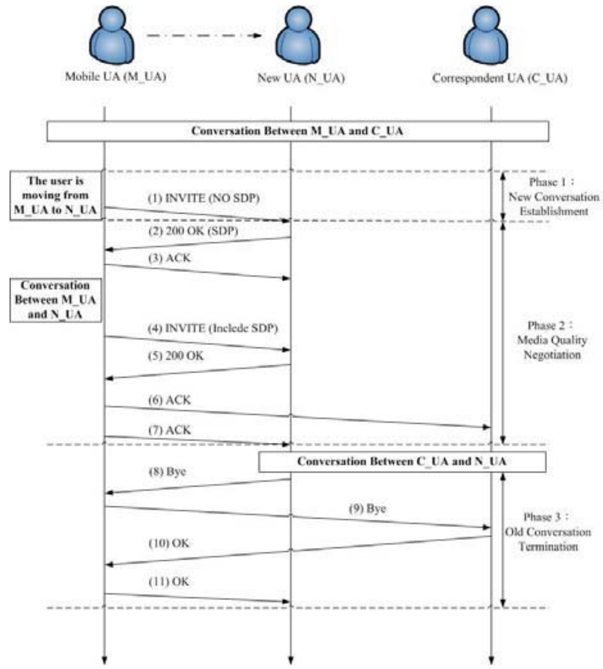

Figure 1 The procedure of RS-based 3PCC

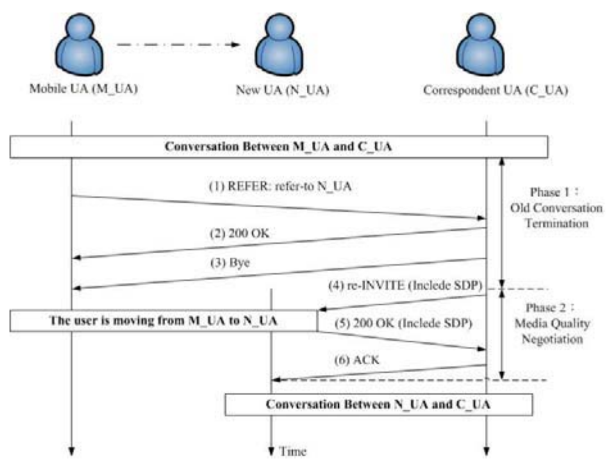

Figure 2 The procedure of REFER with Quality Negotiation

3.1.1. RS-based 3PCC occurs underflow

If the C_UA transmits packets to the N_UA through the M_UA and experiences the minimal delay time in the process of RS-based 3PCC. After establishing new sessions, the C_UA transmits packets to the N_UA directly and the packets experience the maximum delay time. In the way, the delay time is too long to cause the smooth. The figure 3 depicts the situation.

The table1 expresses all parameters about formula 1and 2. The low threshold buffer is Low Threshold minus 1.

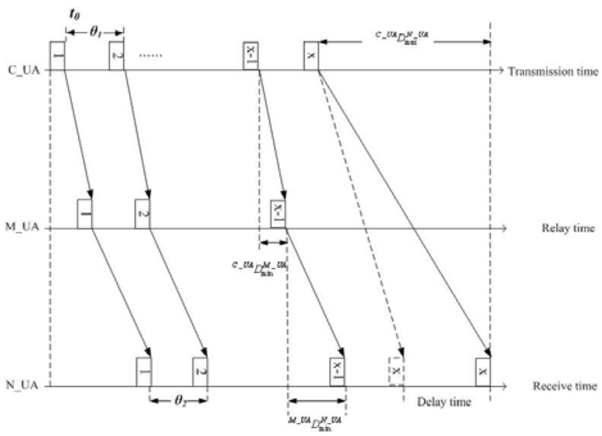

Figure 3 RS-based 3PCC occurs underflow

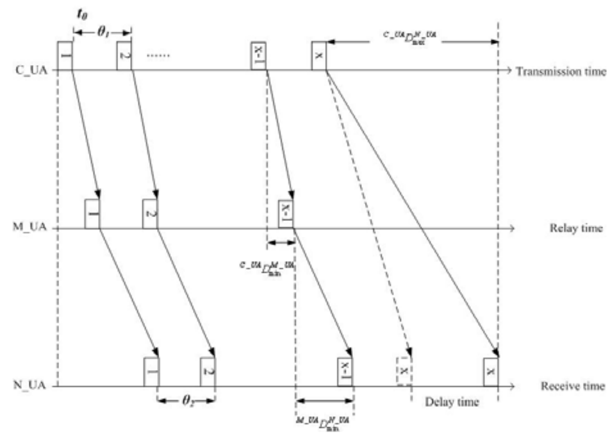

Figure 4 RS-based 3PCC occurs overflow

$$Delaytime=\left(\theta_1+{}^{C\_UA}D_{\max}^{N\_UA}\right)-\left({}^{C\_UA}D_{\min}^{M\_UA}+{}^{M\_UA}D_{\min}^{N\_UA}+\theta_2\right) \quad (1)$$

$$Low\quad Threshold = \frac{Delay\quad time}{\theta_2} \quad (2)$$

Table 1 The parameter about underflow

Parameters	Description
${}^{C\_UA}D_{\min}^{M\_UA}$	The media unit minimum delay rime from C_UA to M_UA
${}^{M\_UA}D_{\min}^{N\_UA}$	The media unit minimum delay rime from M_UA to N_UA
${}^{C\_UA}D_{\max}^{N\_UA}$	The media unit maximum delay rime from C_UA to N_UA
θ_1	the display rate in C_UA
θ_2	The display rate in N_UA
t_0	The star-up time to transmit the media unit
x	The number of media unit

3.1.2. RS-based 3PCC occurs overflow

If the C_UA transmits packets to the N_UA through the M_UA and experiences the maximum delay time in the process of RS-based 3PCC. After establishing new sessions, the C_UA transmits packets to the N_UA directly and the packets experiences the minimum delay time. In the way, the packets will occur collision to make packet loss. The figure 4 depicts the situation.

The table2 expresses all parameters about formula 3 and 4. The High threshold buffer is x minus 1.

$$t_0+(x-1)\times\theta_1+{}^{C\_UA}D_{\min}^{N\_UA}\le t_0+{}^{C\_UA}D_{man}^{M\_UA}+{}^{M\_UA}D_{\max}^{N\_UA}+\theta_2 \quad (3)$$

$$x\le\frac{\left(t_0+{}^{C\_UA}D_{man}^{M\_UA}+{}^{M\_UA}D_{\max}^{N\_UA}+\theta_2\right)-{}^{C\_UA}D_{\min}^{N\_UA}}{\theta_1}+1 \quad (4)$$

Table 2 The parameter about overflow

Parameters	Description
${}^{C\_UA}D_{\max}^{M\_UA}$	The media unit maximum delay rime from C_UA to M_UA
${}^{M\_UA}D_{\max}^{N\_UA}$	The media unit maximum delay rime from M_UA to N_UA
${}^{C\_UA}D_{\min}^{N\_UA}$	The media unit minumum delay rime from C_UA to N_UA
θ_1	the display rate in C_UA
θ_2	The display rate in N_UA
t_0	The star-up time to transmit the media unit
x	The number of media unit

3.2The packet analysis of REFER

3.2.1. REFER occurs underflow

Before the (x-1) packet, the packets transmitted to the N_UA through the M_UA and experienced the minimum delay time. After the reestablish session time, the packets experienced the maximum delay time from the C_UA to the N_UA until the x packet. When the N_UA starts to display the packet, the x packet does not arrive. The figure 5 depicts the situation. The table3 expresses all parameter about the formula 5 and 6.

$$Delaytime=\left(D_{REFER}+{}^{C\_UA}D_{\max}^{N\_UA}\right)-\left({}^{C\_UA}D_{\min}^{M\_UA}+{}^{M\_UA}D_{\min}^{N\_UA}+\theta_2\right) \quad (5)$$

$$Low\quad Threshold = \frac{Delay\quad time}{\theta_2} \quad (6)$$

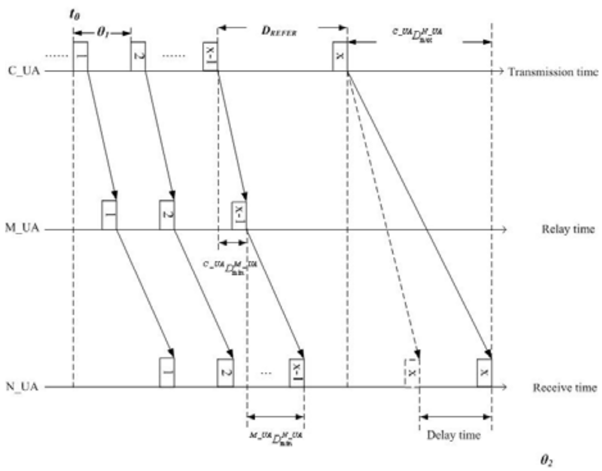

Figure 5 REFER occurs underflow

Table 3 The parameter about underflow

Parameters	Description
${}^{C\_UA}D_{\min}^{M\_UA}$	The media unit minimum delay rime from C_UA to M_UA
$B_{delay}^{M\_UA}$	The buffer delay time in M_UA
${}^{M\_UA}D_{\min}^{N\_UA}$	The media unit minimum delay rime from M_UA to N_UA
${}^{C\_UA}D_{\max}^{N\_UA}$	The media unit maximum delay rime from C_UA to N_UA
$\boldsymbol{D_{REFER}}$	The time of REFER

4. Conclusions

The proposed USHC provides a handoff framework based on user perspective. It consists of two stage strategies to support seamless user-oriented handoff. Although this research proposed a theoretical method to improve the communication, it is lack for simulation and experiment results to support it. That is the future purpose that the research must to strengthen.

5. Acknowledgement

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