# Performance Enhancement of SIP- based VoIP Server

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**Abstract** - *The performance improvement on SIP-based VoIP server is given in this paper. The signaling server "SIP Proxy" and media streaming server "RTP-Relay" are employed. Firstly, the performance of VoIP server is analyzed and evaluated in detail. Then the sophistic method is employed to improve the performance of VoIP server.*

*In the analysis results of "SIP Proxy", the CPU time for the registration and call-setup sessions need 3.241 ms and 7.985 ms respectively. Moreover, in the "RTP-Relay" system, when the packet size from 1 to 32 bytes, the maximum throughput of packet is about 82,000 and 16,000 per second for UDP and TCP respectively. The large the packet, the less throughput is founded.*

*In the improved method, three methods are employed to improve the performance. The capacity by 31 times is achieved. The capacity with 3,524 for each "RTP-Relay" is achieved. The improved method proposed in this paper is reasonable good.*

**Keywords:** Voice over IP, SIP Proxy, RTP-Relay

## 1. INTRODUCTION

In the past, the VoIP application is growing like crazy. The VoIP protocols such as H.323 [1], SIP [2], IMS[3], and Skype [4] were designed for real applications. However, due to the simplicity and flexibility, many efforts [5,6,7] were paid on the SIP protocol and most of VoIP product followed the SIP standard. In the future trend, the SIP-based VoIP solution will become more popular in the broadband and wireless networks.

The capacity of SIP-based VoIP server is critical to the planning and operation of IP-telecom system. However, the capacity of SIP-based VoIP server is not well-studied or understood. Moreover, the improvement of SIP-based VoIP server is also not well-studied and understood. The clear analysis results on capacity will help to plan a suitable IP-telecom system. Moreover, the improved method on the VoIP server will reduce the cost of operation.

In this paper, the performance analyzed and evaluated on the SIP-based VoIP server is given in the next section. The improved method and results on the VoIP server will be proposed and described in the section III. The conclusions will be given at the last section.

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## 2. EVALUATION ON VOIP SERVER

The SIP-based VoIP application is employed to evaluate the performance of VoIP server. The VoIP server includes "SIP-Proxy" and "RTP-Relay". The "SIP Proxy" handles the signaling of call-setup. However, the "RTP-Relay" exchanges the media streaming.

In the SIP client, the regular registration process is needed to keep client always on-line. Otherwise, the call-in process will fail. The flow-chat of SIP-based registration process is given in Figure-1. The flow-chat of call-setup process is also given in the Figure-2. The CPU time for the registration and call-setup processes is evaluated in the Table-1. There are two types of registration process. The first, "Registration-I", includes the authentication process. The MD5 algorithm and password are employed to authenticate the user. If the "SIP UAC" doesn't change the IP address and repeat the registration process, the "Registration-II" is employed to bypass the authentication process. The call-setup process is also with two types. In the Table-1, the Pentium CPU with 2.4GHz is employed to evaluate the CPU time and capacity. The capacity of registration and call-setup processes is 309 and 125 times per second. It means that "SIP Proxy" can process 309 REGISTER requests or 125 INVITE requests within one second.

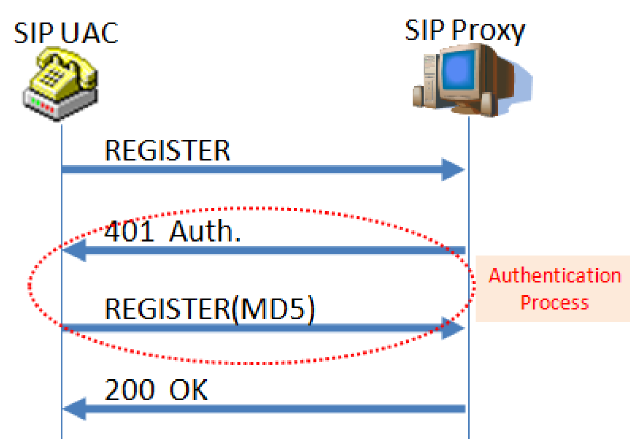


Figure-1. The flow-chart of SIP-based registration process.

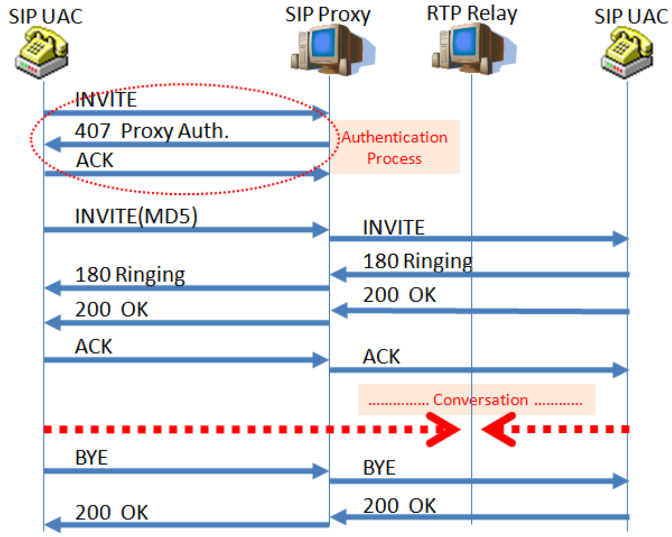


Figure-2. The flow-chart of SIP-based call-setup process.

Table-1. The CPU time and capacity for each session. (not include Internet Tx/Rx time).

| Session | Command Flow | CPU Time | Capacity: (session/ sec.) |
|---|---|---|---|
| Registration-I (with MD5) | REGISTER+401+ REGISTER+200 | 3.241 ms | 309 |
| Registration-II (without MD5) | REGISTER+200 OK | 1.838 ms | 544 |
| Call-Setup-I (with MD5) | INVITE+407+ACK+ INVITE+180+200+ ACK+BYE+200 | 7.985 ms | 125 |
| Call-Setup-II (without MD5) | INVITE+180+200+ ACK+BYE+200 OK | 5.254 ms | 190 |

In the "RTP-Relay" system, the audio and video packet is received, exchanged and re-sent to each other. The maximum throughput (MT) of packet with different size is listed in the Table-2. The MT of packet is same when the packet size is small than 32. The MT will decrease when the packet size increase. Moreover, the MT of UDP packet is more TCP packet.

Table-2 The maxima throughput and CPU times with different packet size.

| **Packet Size (Bytes)** | **Maxima Throughput (Packet No/Second)** | | **Percentage of CPU Usage (%)** | |
|---|---|---|---|---|
| | **UDP** | **TCP** | **UDP** | **TCP** |
| **1** | 82410 | 15104 | 61 | 66 |
| **2** | 82374 | 19532 | 59 | 48 |
| **4** | 82998 | 12358 | 60 | 52 |
| **8** | 83556 | 16656 | 59 | 50 |
| **16** | 82264 | 16440 | 57 | 56 |
| **32** | 66734 | 16918 | 11 | 38 |
| **64** | 44208 | 15918 | 10 | 36 |
| **128** | 35246 | 16758 | 10 | 32 |
| **256** | 23900 | 11218 | 15 | 30 |
| **512** | 17372 | 9760 | 7 | 28 |
| **1024** | 11318 | 8292 | 8 | 26 |

## 3. IMPROVEMENT ON VOIP SERVER

In this session, the improved method on the "SIP Proxy" and "RTP-Relay" is proposed and evaluated. In the "SIP Proxy" system, three methods are employed to improve the performance. These methods are described and listed as below:

1. In the registration process, the MD5-based authentication process can be bypass when the IP address and port number of SIP UAC is unchanged. The capacity of "SIP Proxy" will be improve from 309 to 544. The capacity of "SIP Proxy" in the registration process increases 76%
2. The "SIP Proxy" detects the keep-alive time interval (KATI) of NAT automatically and asks "SIP UAC" to register every KATI seconds. The capacity of "SIP Proxy" can be optimized and maximized. The KATI of different trademark™ NAT is listed in the Table-3.
3. In the NAT device, the KATI of TCP is large than UDP. The TCP-based registration process will improve the capacity of "SIP Proxy" dramatically. The capacity will be improved more than 31 times.

Table-3. The keep-alive time interval (KATI) of NAT for each trademark.

| Trademark of NAT | Keep-Alive Time Interval (unit: Seconds) | |
|---|---|---|
| | UDP | TCP |
| PCI CQW-MR500 | 75 | 1800 |
| AboCom FSM410 | 90 | >3600 |
| AboCom MH1000 | 65 | >3600 |
| Corega CG-WLR300GNH | 75 | 1800 |
| Buffalo WZR-HP-G300NH | 100 | >3600 |
| Buffalo BBR-4HG | 65 | 2700 |
| TP-Link TL-R402M | 90 | 1800 |
| D-Link DI-604 | 95 | >3600 |
| D-Link DIR-655 | 80 | >3600 |
| D-Link DIR-320 | 80 | >3600 |
| D-Link DIR-100 | 75 | >3600 |
| Lemel LM-IS6400B | 70 | 900 |
| DrayTek Vigor2104p | 100 | >3600 |
| I.O DATA NP-88RL | 80 | >3600 |
| Zonet ZSR0104B | 90 | 1800 |
| ZyXEL P-330w v2 | 175 | >3600 |
| ZyXEL Prestige 304 | 70 | 900 |
| 3Com WL-537 | 175 | 1800 |
| Asus RX-3041 | 90 | >3600 |
| SMC WBR14-G2 | 65 | >3600 |
| Average | 90.25 | 2835 |

In the "RTP-Relay" system, the bottleneck of capacity is the maximum throughput of packet. Two methods can be employed to enhance the capacity of "RTP-Relay". These methods are described and listed as below:

1. The UDP-based RTP packet can improve the capacity of "RTP-Relay" 2~4 times. For example, in the Table-2, with 16 bytes packet size, the maximum throughput is above 82264 and 16440 respectively. The performance of UDP is much better than TCP.
2. The large size of RTP packet will also enhance the capacity of "RTP-Relay". For example, in the G.729 mode, the capacity of "RTP-Relay" for each time-stamp interval is listed in Table-4. The time-stamp interval increases, the capacity of "RTP-Relay" increases also.

Table-4. The time-stamp interval vs. capacity of RTP-Relay.

| Time-Stamp Interval (Sample Points) | RTP packet size (RTP Header=12 bytes) | Packet No./ Second | Capacity of RTP-Relay |
|---|---|---|---|
| 10ms(80) | 22 bytes | 100 | 66734/100= 667 |
| 20ms(160) | 32 bytes | 50 | 66734/50=1332 |
| 30ms(240) | 42 bytes | 33.3 | 44208/33.3=1473 |
| 40ms(320) | 52 bytes | 25 | 44208/25=1768 |
| 50ms(400) | 62 bytes | 20 | 44208/20=2210 |
| 60ms(480) | 72 bytes | 16.7 | 35246/16.7=2110 |
| 70ms(560) | 82 bytes | 14.3 | 35246/14.3=2464 |
| 80ms(640) | 92 bytes | 12.5 | 35246/12.5=2819 |
| 90ms(720) | 102 bytes | 11.1 | 35246/11.1=3175 |
| 100ms(800) | 112 bytes | 10 | 35246/10=3524 |

# 4. CONCLUSION

The performance of SIP-based VoIP server is evaluated and improved in this paper. In the "SIP Proxy", three methods are employed to improve the performance of "SIP Proxy". More than 31 times on capacity is achieved. Moreover, two methods are employed to improve the performance of "RTP-Relay". The capacity with 3524 for each "RTP-Relay" can be achieved. The analyzed results and improvement method can help to plan the VoIP system and architecture.

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