韓國 컴퓨터情報學會 論文誌
第12卷 第1號, 2007. 3.

2007-12-1-2-4

VoIP(Voice over Internet Protocol) 품질 측정을 위한 UA(User Agent) 및 서버 기능 연구

강 현 중*, 남 흥 우**

Implementation of QoS-Measuring System for Voice over IP

Hyun-Joong Kang*, Heung-Woo Nam**

요 약

유무선 통신 기술, 디지털 미디어, 그리고 영상처리 기술의 비약적인 발전과 서비스 통합화 추세는 광대역통합망(BcN: Broadband convergence Network)과 같은 초고속 네트워크를 통하여 VoIP, IPTV와 같은 여러 형태의 새로운 서비스를 창출하게 되었다. VoIP 서비스가 기존의 공중회선 교환망에서와 같은 이윤 창출을 위해서는 기존 서비스 이상의 품질을 제공하여야 한다. 따라서 실시간 품질측정 프레임워크는 VoIP 서비스를 제공하기 위한 가장 중요한 요소라고 할 수 있다. 이를 위해 IETF (Internet Engineering Task Force)에서는 RTCP (Real-Time Transport Protocol Control Protocol)를 확장한 RTCP-Extended Reports (RTCP-XR)을 정의하였다. 그러나 RTCP-XR에서는 음성품질을 측정하기 위한 항목만을 정의하였을 뿐 실제 VoIP 품질 측정을 위한 절차와 방법은 규정하지는 않았다. 본 논문은 종단간 패킷화된 음성을 효과적으로 측정하기 위한 프레임워크 제시를 목적으로 하고 있다. 이를 위해 본 논문에서는 VoIP 품질 측정개념과 더불어 제안된 프레임워크에서 측정방법을 단계적으로 기술하였다. 아울러 RTCP-XR의 개념을 확장한 새로운 형태의 포맷을 제안하였다.

Abstract

Advances in networking technology, digital media, and codecs have made it possible for the Internet evolves into a Broadband convergence Network (BcN) and provides various services including Voice over Internet Protocol (VoIP) and IPTV over their high-speed IP networks. In order for the Internet to make a profit as traditional Public Switched Telephone Network (PSTN), it must provide high quality VoIP services. Therefore, real time quality measurement framework is the most important requisite to provide VoIP service. For this, IETF (Internet Engineering Task Force) defined RTCP-Extended Reports (RTCP-XR) that extend RTCP (Real-Time Transport Protocol Control Protocol). However, procedure and method for actually VoIP quality measurement did not recommended nothing but defined item to measure voice quality. Our objective in this paper is to describes a practical measuring framework for end-to-end QoS of switched voice packet in an IP environment. It includes concepts as well as step-by-step procedures for measuring packetized voice streams. It also proposes new formats that extend RTCP-XR's concept.

▸ Keyword : VoIP, QoS, RTCP, PLC

• 제1저자 : 강현중
• 접수일 : 2006.9.18, 심사일 : 2006.10.12, 심사완료일 : 2007. 3.13.
* 서일대학 인터넷정보과 교수, ** 서일대학 정보통신과 시간강사
※ 본 논문은 2006년 서일대학 학술 연구비에 의해 연구되었음

Ⅰ. 서 론

Until a recent date all telephony connections are set up via circuit switching whereby node-to-node links in an origin/destination connection are set up via interconnects, and the connection is maintained exclusively for exchanges of information between the origin and destination until it is torn down [1].

An alternate way of setting up end-to-end connections that is widely used for transmission of data is packet switching, such as that used in the Internet, whereby origin-to-destination connections are effected by node-to-node, store-and-forward relay of small segments of data sets that are reassembled at the destination.

Advances in networking technology, digital media, and codecs have made it possible for the Internet evolves into a Broadband convergence Network (BcN) and provides various services including Internet Protocol (IP) telephony and on-demand television over their high-speed IP networks. In order for the Internet to realize a profit as traditional Public Switched Telephone Network (PSTN), it must provide competent quality of services for VoIP systems comparable to traditional PSTN systems.

A large number of malicious factors are concerned to make a high-quality VoIP service. These factors include the speech codec, encoding (compression) schemes, packet loss, delay, delay variation, and the network architecture. Other factors involved in making a successful VoIP call includes the call setup signaling protocol, call admission control, security concerns, and the ability to traverse NAT (Network Address Translation) and firewall [2][3][4].

A successful end-to-end realization of IP telephony services presumes well-defined QoS measuring framework in the service provider's and customer's networks [5]. The VoIP metrics report block of Real-Time Transport Protocol Control Protocol Extended Reports (RTCP XR) can be applied to any one-to-one or one-to-many voice application. However, RTCP XR only defines packet type to convey information that supplements the six statistics that are contained in the report blocks used by RTCP's Sender and Receiver packets [5].

Our objective in this paper is to describes a practical measuring framework for end-to-end QoS of packet switched voice in an IP environment including Packet Loss Concealment (PLC) techniques and Network Time Protocol (NTP). It

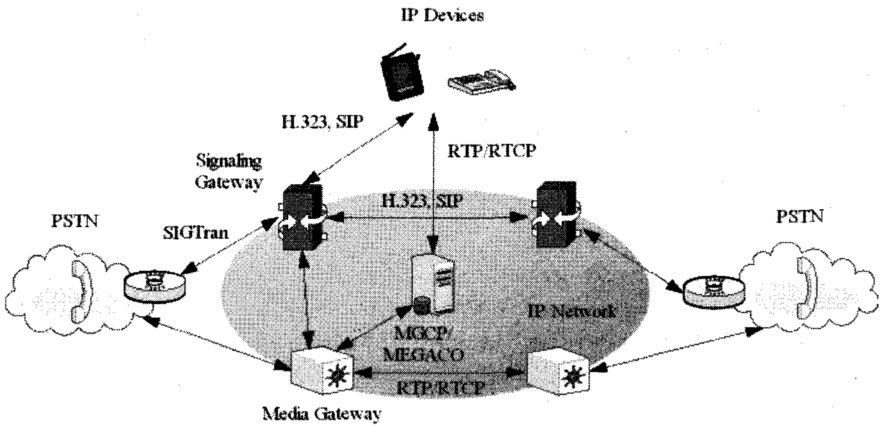


Fig. 1. Typical architecture of VoIP system

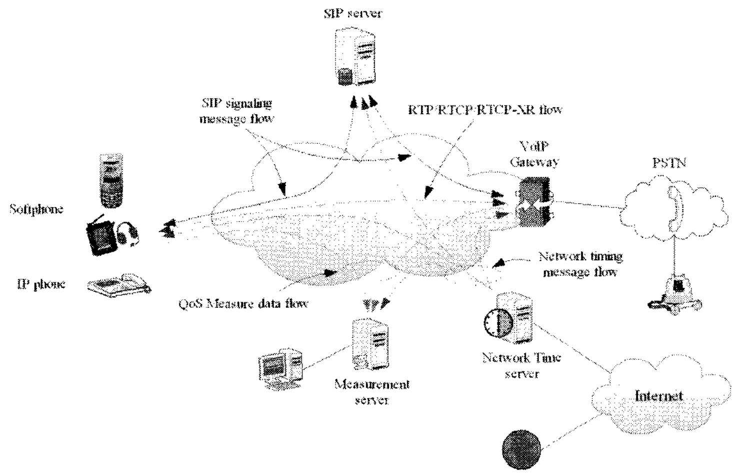


Fig. 2. Proposed VoIP measurement framework

includes concepts as well as step-by-step procedures for setting up components, creating session, measuring packetized voice streams over IP networks.

The rest of this paper is organized as follows. In section 2, we will introduce a general components and control architecture of VoIP systems. The proposed VoIP QoS measurement architecture and functional components are described in section 3. In section 3, we also will introduce their flow diagrams in detail so as to communicate with each other. Finally, Conclusions were drawn in Section 4.

II. VoIP Service and Components

In connectionless network architectures such as IP networks, the responsibility for session establishment and signaling resides in the end stations. To successfully emulate voice services across an IP network, enhancements to the signaling stacks are required.

As shown in ⟨Fig. 1⟩, several protocols exist for VoIP signaling, including H.323, Media Gateway Control Protocol (MGCP), Session Initiation Protocol (SIP), and Megaco, that enable creating, modifying, and terminating multimedia sessions with one or more participants over IP networks. The signaling protocols enable creating, modifying, and terminating multimedia sessions with one or more participants over IP networks.

Because of its simplicity, scalability, modularity, and ease with which it integrates with other applications, SIP is attractive for use in packetized voice architectures [6]. In order to provide inter-operability, a number of gateways provide for translation and call control functions between the two dissimilar network types. Encoding, protocol, and call control mappings occur in gateways between two endpoints [1][2][7].

VoIP refers to packetized voice communication over IP networks. In VoIP applications a voice signal is first packetized and then transmitted over an IP network. When the packets arrive at their destination they are reassembled and buffered in order to restore the voice data. However, at the receiving end, packets are missing or distorting due

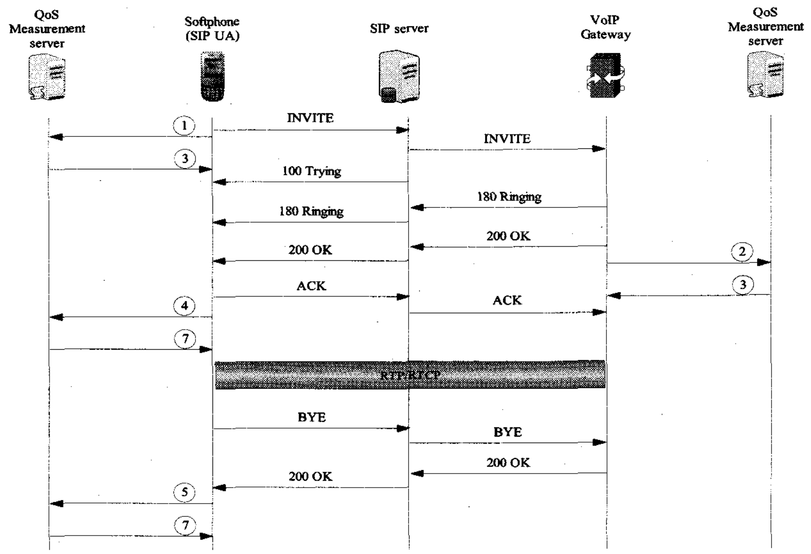


Fig. 3. VoIP QoS measurement procedure - normal session

to network delay, network congestion (jitter) and network errors. This packet loss or delay degrades the quality of speech at the receiving end.

The Mean Opinion Score (MOS) test is widely accepted as a standard for speech quality rating. However, the subjective MOS rating is time-consuming and inaccurate. In recent years, several objective MOS measures were developed, such as Perceptual Analysis Measurement System (PAMS) and Perceptual Evaluation of Speech Quality (PESQ) [3][4].

The E-model standards, the E-model started as a research by European Telecommunications Standards Institute (ETSI), also provide a formula for calculating the loss of interactivity as function of the one-way delay. The E-model expresses an overall rating of the quality of a call and can be translated into quality and MOS [6][7][8][9].

The chief requirement that real-time media places on the transport protocol is for predictable variation in network transit time. RTP provides end-to-end network transport functions suitable for applications transmitting real-time data such as audio or video stream.

It is used to transport data via UDP. RTP does not address resource reservation and does not guarantee QoS for real-time services. The data transport is augmented by a control protocol (RTCP) to allow monitoring of the data delivery in a manner scalable to large multicast networks and to provide minimal control and identification functionality [8][9].

III. Measurement Framework of Voice Stream in Packet Network

In this section, we describe the architecture of the proposed VoIP measuring framework. The framework can be integrated to the commercial VoIP system as shown in 〈Fig. 2〉.

The RTCP XR packets are useful across multiple applications, in particular, the VoIP metrics report block provides useful metrics for monitoring voice over IP (VoIP) calls. These metrics include packet loss and discard metrics, delay metrics, analog metrics, and voice quality metrics. However, little more detailed procedure need to be defined for real operation [10].

To solve these probelems, as shown in Fig 3, Fig 4, and Fig 5, 7 extended RTCP-XR format and procedure are proposed in this paper.

If a caller want to setup a session, as shown in ⟨Fig. 3⟩, it will send INVITE message to callee to join the session. At the same time, as shown in ⟨Fig. 4⟩, the caller will send type-1 initiation message to measurement server. After send 200 OK message to the caller, as shown in ⟨Fig. 5⟩, the callee will send type-2 initiation message to the measurement server. The measurement server will respond the initiation messages respectively.

When the caller receives the callee's respond, it will send ACK message to reply the respond. Immediately after that, the caller will also send type-4 start message to the measurement server. After that the caller will setup the media channels such as RTP streaming with the caller.

If the caller or the callee do not want to join the session anymore, they will send BYE message to the other participant. After that, the caller and the callee will send type-5 end message to the measurement server [11].

RTP nodes send RTCP packet each other to analyze network state and report network availability periodically. As shown in Fig 1, most RTCP-XR packets are exchanged in short connection setup time and the other packets do not give load greatly to network because RTCP-XR packets are exchanged periodically under threshold value.

Table 1 shows the relationship between (random/burst) packet loss and MOS value that is captured at the measurement server. It also shows the relationship between (random/burst) packet loss and R rate. As shown in Table 1, VoIP quality becomes different definitely according to PLC availability. It means that VoIP QoS can be elevated through terminal function such as PLC and measurement framework that proposed in this paper.

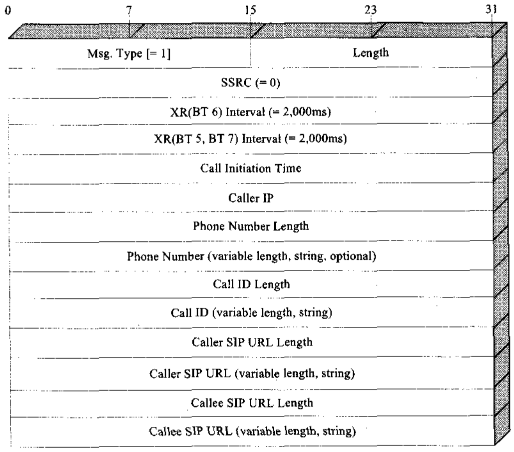


Fig. 4. Type-1 message

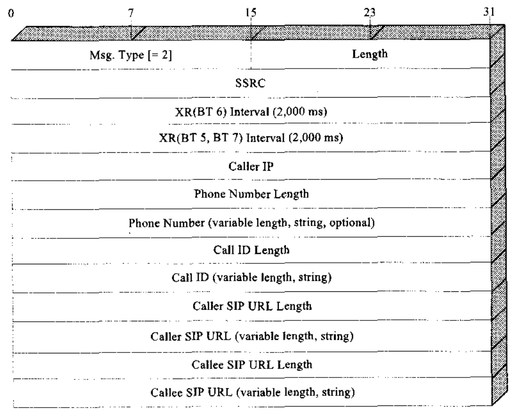


Fig. 5. Type-2 message

RTP and RTCP are designed to be independent of the underlying transport and network layers. The protocol supports the use of RTP-level translators and mixers. Real-Time Control Protocol (RTCP) provides for reliable information transfer once the audio stream has been established. A reliable session-oriented protocol such as TCP is deployed between end stations to carry the signaling channels. RTP, which is built on top of UDP, is used to transport the real-time audio stream. RTP uses UDP as a transport mechanism because it has lower delay than TCP and because actual voice traffic, unlike data traffic or signaling, tolerates low levels of loss and cannot effectively exploit retransmission.

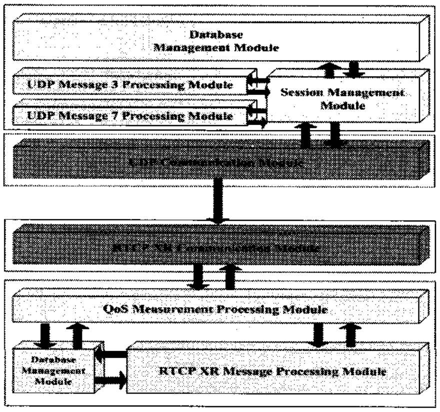


Fig. 6. Protocol suite of measurement server

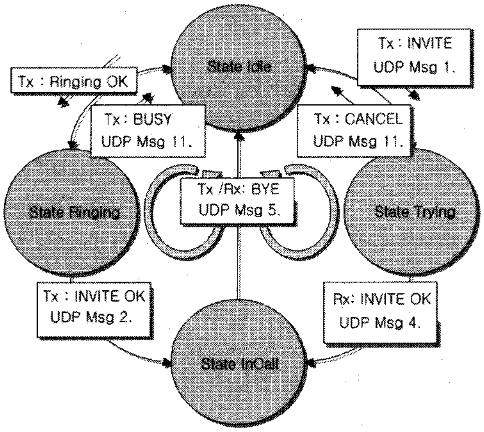


Fig. 7. State machine of measurement server

Table 1. The relationship between packet loss rate and VoIP QoS

	Measurement Time(min)		0%			3%						5%						10%					
						W/o PLC			PLC			W/o PLC			PLC			W/o PLC			PLC		
			MOS	R	Delay	MOS	R	Delay	MOS	R	Delay	MOS	R	Delay	MOS	R	Delay	MOS	R	Delay	MOS	R	Delay
Random	0.5	Tx	4.4	94	12	3.6	71	30	3.9	78	27	3	59	24	3.5	68	20	2.4	47	36	3	58	21
		Rx	4.4	93	26	3.6	71	10	3.9	78	16	3	59	15	3.5	68	26	2.5	48	6	3	58	24
	1	Tx	4.4	93	20	3.6	71	30	3.9	78	29	3	59	24	3.5	68	10	2.4	47	36	3	58	23
		Rx	4.4	93	19	3.6	71	15	3.9	78	15	3	59	16	3.5	68	28	2.5	48	7	3	58	17
	1.5	Tx	4.4	94	18	3.6	71	31	3.9	78	33	3	59	25	3.5	68	12	2.5	48	18	3	58	23
		Rx	4.4	93	19	3.6	71	16	3.9	78	9	3	59	16	3.5	68	31	2.5	48	25	3	58	16
	2	Tx	4.4	93	21	3.6	71	30	3.9	78	27	3	59	32	3.5	68	10	2.5	48	15	2.9	57	30
		Rx	4.4	93	19	3.6	71	9	3.9	78	7	3	59	9	3.5	68	31	2.5	48	18	3	58	22
	2.5	Tx	4.4	93	20	3.6	71	34	3.9	78	21	3	59	34	3.5	68	9	2.5	48	20	2.9	57	32
		Rx	4.4	94	18	3.6	71	9	3.9	78	20	3	59	7	3.5	68	34	2.5	48	22	3	58	9
	Average	Tx	4.4	93	18	3.6	70	28	3.9	78	27	3	59	25	3.5	68	12	2.4	47	28	3	58	26
		Rx	4.4	92	20	3.6	71	12	3.9	78	13	3	59	14	3.5	68	30	2.5	48	12	3	58	18
Burst	0.5	Tx				3.3	63	34	3.3	64	29	2.9	57	33	3.3	63	23	2.6	50	26	2.7	53	41
		Rx				3.3	64	5	3.3	64	13	3	58	8	3.3	63	14	2.6	50	16	2.7	53	20
	1	Tx				3.3	63	35	3.3	64	29	2.9	57	33	3.3	63	24	2.6	50	29	2.7	53	26
		Rx				3.3	64	6	3.3	64	11	3	58	6	3.3	64	15	2.6	50	16	2.7	53	16
	1.5	Tx				3.3	63	23	3.3	64	31	2.9	57	34	3.3	63	27	2.6	50	31	2.7	53	28
		Rx				3.3	64	14	3.4	65	9	3	58	1	3.3	64	11	2.6	50	9	2.7	53	16
	2	Tx				3.3	63	23	3.3	64	36	2.9	57	43	3.3	63	30	2.6	50	24	2.7	53	27
		Rx				3.3	64	16	3.4	65	9	3	58	5	3.3	64	10	2.6	50	16	2.7	53	15
	2.5	Tx				3.3	63	23	3.3	64	39	2.9	57	42	3.3	63	30	2.5	49	33	2.7	53	31
		Rx				3.3	64	5	3.4	65	1	3	58	3	3.3	63	20	2.6	50	14	2.7	53	6
	Average	Tx				3.3	63	29	3.3	64	33	2.9	56	38	3.3	63	27	2.5	49	27	2.7	53	31
		Rx				3.3	64	11	3.4	65	9	3	58	5	3.3	64	14	2.6	50	13	2.7	53	15

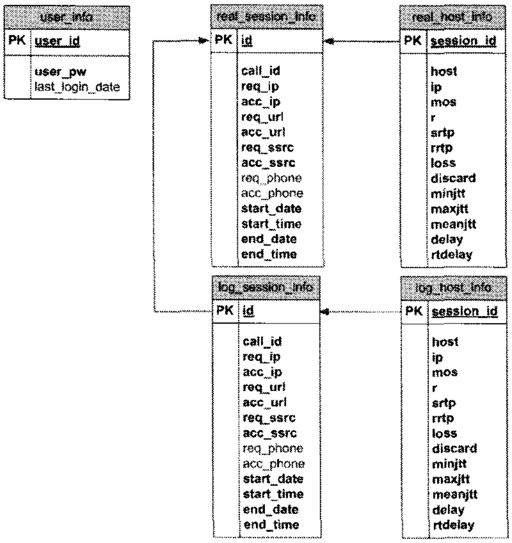


Fig. 8. Database architecture of measurement server

IV. Conclusion

Voice-over-IP (VoIP) uses packetized transmission of speech over the Internet (IP network) and has been thought as one of the killer application of BcN. In order for the Internet to realize a profit as traditional Public Switched Telephone Network (PSTN), it must provide competent quality of services for VoIP systems comparable to traditional PSTN systems.

A successful end-to-end realization of IP telephony services presumes well-defined QoS measuring framework in the service provider's and customer's networks. E-Model is regarded as prominent rating technology that can be applied most properly when estimate speech quality for VoIP service because it considering about data network characteristic such as loss, delay.

This paper described a practical measuring framework based on E-model for end-to-end QoS of packet switched voice in an IP environment including Packet Loss Concealment (PLC) techniques and Network Time Protocol (NTP). We also investigated the effects of packet loss and delay jitter on speech quality in VoIP scenarios. In addition to the block types defined RTCP XR, for VoIP monitoring, additional block types were defined in this paper by adhering to the RTCP XR framework.

참고문헌

[1] William C. Hardy: VOIP Service Quality -Measuring and Evaluating Packet Switched-Voice, McGraw-Hill, (2003)

[2] Victoria Fineberg: A Practical Architecture for Implementing End-to-End QoS in an IP Network, IEEE Communications Magazine, Jan. (2002) 122~130

[3] Definition of Categories of Speech Transmission Quality, ITU-T Recommendation G.109, (1999)

[4] The E-Model, a computational model for use in transmission planing, ITU-T Recommendation G.107, (1998)

[5] T. Friedman, R. Caceres: RTP Control Protocol Extended Reports (RTCP XR), IETF RFC 3611, Nov. (2003)

[6] Jonathan Rosenberg, et. al.: SIP: Session Initiation Protocol, IETF RFC 3261, Jun. (2002)

[7] Athina P. Markopoulou, Fouad A. Tobagi, and Mansour J. Karam: Assessing the Quality of Voice Communications Over Internet Backbones, IEEE/ACM Transaction on Networking, Vol. 11, No. 5, Oct. (2003)

[8] Shengquan Wang, Zhibin Mai, Dong Xuan, and Wei Zhao: Design and Implementation of QoS-Provisioning System for Voice over IP, IEEE TRANSACTIONS ON PARALLEL AND DISTRIBUTED SYSTEMS, VOL. 17, NO. 3, MAR. (2006) 276—288

[9] Shengquan Wang, Zhibin Mai, Walt Magnussen, Dong Xuan, and Wei Zhao: Implementation of

QoS-Provisioning System for Voice over IP, IEEE Real-Time and Embedded Technology and Applications Symposium (RTAS'02), (2002)

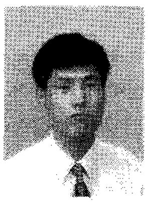
[10] 박석천, "FMIPv6에서 Qos를 고려한 트래픽 제어 메커니즘의 설계 및 평가", 한국컴퓨터정보학회논문지, 제 10권, 제 6호, pp. 261-267, 2005.

[11] 박대우, "VOIP 보안 게이트웨이에 관한 연구", 한국컴퓨터정보학회논문지, 제 10권, 제 5호, pp. 237-243, 2005.

저 자 소 개

강 현 중

1980년 2월 성균관대대학교
수학교육학과 졸업
1986년 2월 연세대학교 대학원
전자계산학과 석사
1996년 2월 성균관대학교 대학원
정보공학과 박사
1979년 11월~1982년 2월
한국과학기술연구소(KIST)연구원
1982년 3월~1989년 2월
한화종합금융(주) 전산팀장
1989년 3월~현재
서일대학 인터넷정보전공 교수
〈관심분야〉 데이터통신,
프로그래밍언어



남 홍 우

2002년 2월 경희대학교
컴퓨터공학과 졸업
2005년 2월 고려대학교 대학원
컴퓨터학과 석사
2005년 9월~현재
고려대학교 대학원
전자.컴퓨터학과 박사과정
2006년 9월~현재
서일대학 정보통신과
시간강사
〈관심분야〉 센서네트워크, RFID