석 사 학 위 논 문

Master's Thesis

3세대 및 3.5세대 무선 네트워크의 성능측정 연구

Performance Measurement over 3G and 3.5G Wireless Networks

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Korea Advanced Institute of Science and Technology

2008

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A thesis submitted to the faculty of the Korea Advanced Institute of Science and Technology in partial fulfillment of the requirements for the degree of Master of Engineering in the Department of Electrical Engineering and Computer Science, Division of Computer Science .

Daejeon, Korea
2007. 12. 4.
Approved by

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위 논문은 한국과학기술원 석사학위논문으로 학위논문심사 위원회에서 심사 통과하였음.

2007년 12월 4일

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MCS 20063609

한 몽 남. Han, Mongnam. Performance Measurement over 3G and 3.5G Wireless Networks. 3세대 및 3.5세대 무선 네트워크의 성능측정 연구. Department of Electrical Engineering and Computer Science, Division of Computer Science . 2008. ??p. Advisors Prof. Moon, Sue Bok, Prof. Yoon, Hyunsoo.

Abstract

In this work, we have conducted extensive end-to-end measurements to capture the characteristics of variability of wireless link, to analyze its impact on TCP behavior, and to evaluate QoS of VoIP applications over operational 3G and 3.5G wireless network such as CDMA-1xEVDO, HSDPA, and WiBro. Our measurement include UDP/TCP traffic traces and VoIP traffic simulated by D–ITG, as well as several physical layer information of a base-station to which a mobile device is connected. In order to capture the baseline performance of the 3G and 3.5G wireless network we measure and analyze the characteristics of throughput, delay, and loss under stationary and mobile scenarios. In addition, we quantify the degree of variability and demonstrate it graphically using second-order difference plot. Then to investigate TCP behavior on variable wireless link state which becomes more significant in mobile environment, we further analyze TCP performance metrics such as ACK compression and serveral types of retransmission. Finally, we evaluate QoS of VoIP applications using the E–Model of ITU–T G.107. Our measurements show that the achievable maximum throughput and variability in uplink and downlink of each wireless network. We note that even in stationary environment, there is an inefficiency of the interplay between TCP congestion algorithm and wireless link mechanisms, which results in performance degradation. VoIP quality is better than or at least as good as toll quality despite user mobility exceeding the protected limit of each wireless network mobility support. Using RAS and sector identification information, we show that the handoff is correlated with throughput and quality degradation.

목 차

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제 1 장 Introduction

Recent emerging wireless networks such as 3G cellular and wireless LAN (WLAN) allow users choices in accessing the Internet based on one's need and cost. WLAN with a high data rate (up to 54Mbps) supports low mobility and limited coverage. Cellular networks support high mobility with low bandwidth.

CDMA Evolution Data-only (EV-DO) is standarized by the third Generation Partnership Project 2 (3GPP2) [?] and allows a mobile user access the Internet with speeds up to 2.4Mbps for downlink. HSDPA (High-Speed Downlink Pcaket Access) is a competing technology of WCDMA and currently supports 3.6Mbps in downlink [?]. The broadband wireless access (BWA) systems address the market between WLAN and cellular networks. Their goal is to support higher bandwidth than 3G cellular networks, but less mobility for mobile end-user devices. The IEEE 802.16 family of standards specifies the air interface of fixed and mobile BWA systems. WiMax is a subset of the 802.16 standards whose main goal is product compatibility and interoperability of broadband wireless products, just as WiFi is to the 802.11 standards. WiBro has been developed as a mobile BWA solution in Korea, and is generally considered a precursor to WiMax. It is a subset of consolidated version of IEEE Standard 802.16-2004 (fixed wireless specifications), P802.16e (enhancements to support mobility), and P802.16-2004/Cor1 (corrections to IEEE Standard 802.16-2004). The profiles and test specifications of WiBro will be harmonized with WiMAX Forum's mobile WiMAX profiles and test specification, drawing a convergence of the two standards.

Several characteristics including the latency for FEC and interleaving, dynamic data rate due to inteference and mobility, the link asymmetry, a delay spike, high bit error rates and handoffs can significantly affect the performance of transport protocols [?, ?]. The representatives of negative effects are burst loss, spurious timeout, data/ACK compression and occational short or long disconnections, which result in degradation of the performance. Today's Internet users not only write emails and surf the web, but also make Voice over IP (VoIP) calls, play online games, and watch streaming media. These real-time applications have stringent Quality of Service (QoS) requirements on delay and loss. WiMax and WiBro standards have defined multiple service types in order to guarantee different levels of QoS. However, at the initial phase of deployment, often only the best-effort service is made available, while users do not limit themselves to emails and web surfing over emerging wireless technology networks.

In this work, we conduct experiments to evaluate the performance of UDP and TCP as well as QoS of VoIP applications over the 3G and 3.5G network. In order to capture the baseline performance of the 3G and 3.5G network, we measure and analyze delay, loss, and throughput of constant bit rate streams in both stationary and mobile scenarios. We have measured maximum throuhgputs of downlink and uplink. Packet loss and throughput exhibit more variability in the mobile scenario than stationary. Then we evaluate QoS of VoIP applications using the E–Model of ITU–T G.107 also in both stationary and mobile scenarios. VoIP quality is better than or at least as good as toll quality even in the mobile scenario. By combining the packet traces with physical layer information, we show that the handoff is correlated with throughput and quality degradation on VoIP quality. We note that the deployed WiBro network is lightly loaded.

The rest of this paper is organized as follows. In Chapter 2, we describe the background and related work. In Chapter 3, we describe our measurement methodology in the 3G and 3.5G wireless network and present the VoIP quality evaluation methodology. We present our analysis results in Chapter 4 and wrap up the paper with a summary in Chapter 5.

제 2 장 Background and Related Work

2.1 CDMA 2000 1x EVDO

CDMA 2000 1x Evolution-Data Only (or just EVDO), 3G mobile communication system, is optimized system for packet data while using previous CDMA system. It has been commercialized in Korea from 2000 and a representative wireless Internet connectivity service at present. Packet data does not use radio link continously but just use when necessary. EVDO makes the best use of dormant mode which uses this characteristic of burst transmission and provides a peak rate of 2.4Mbps for downlink. Data rate is allocated dynamically per a user depending on a request of user based on a signal to noise ratio(SNR) and varies from 38.6Kbps to 2.4Mbps and from 9.6Kbps to 153.6Kbps on downlink and uplink, respectively. And accrording to mobility it is specified for a stationary user to provide a average rate of 2Mbps and for a mobile user to provide a average rate of 600Kbps and a peak rate of 1.25Kbps in CDG(CDMA Development Group). To achieve higher peak data rate on downlink, EVDO uses TD-CDMA(Time Division-Code Division Multiple Access) that serveral channels are devided in logically and radio signals are spreaded in physically. A base station allocates the channel for a user based on optimized channel scheduling algorithm. In addition, it makes the best use of hybrid ARQ and a diversity to improve link efficiency. In aspect of system, EVDO uses wireless IP architecture which is specified in IS-835 and processes packet data in distinction from a voice signal.

2.2 HSDPA

HSDPA (High Speed Downlink Packet Access) which is an evolution of the WCDMA provides higher data transfer speeds up to 14.4 Mbps per cell in the downlink, that is faster over 3∼7 times than WCDMA. In Korea, SKT and KTF launched HSDPA from June, 2006, and they drive forward evolution to HSUPA(High Speed Uplink Packet Access) which improves uplink speed and to HSOPA which adopt next generation communication technology such as OFDM, MIMO, and smart antenna. In abroad, around Japan and Europe, HSDPA will be serviced as a next key mobile communication, especially in Japan it is called for Super 3G and is driven for evolution to 4G.

HSDPA uses a high speed downlink shared channel(HS-DSCH), and get the increase of downlink speeds. It also uses fast packet scheduling and fast retransmission known as HARQ(Hybrid Automatic Repeat Request) algorithm. In addition to improved data rates, it reduces latency and round trip time for real-time applications.

2.3 WiBro

Fixed WiMax was first used to assist in the relief effort for the 2004 tsunami in Aceh, Indonesia, and now has more than 350 service providers around the world [?]. WiBro, a mobile BWA service, had its public demonstration in December 2005, and has been in service since June 2006 in Korea. The network architecture of WiBro in the phase I standardization [?] is shown in Figure ??. The WiBro network consists of Access Control Routers (ACR), Radio Access Stations (RAS), Personal Subscriber Stations (PSS), and the network service provider's IP network. An RAS is the interface between PSSs and the core network at the physical layer and it also controls the radio resource at the data link layer in conjunction with an ACR. One of the distinguishing features of WiBro from cellular networks is that Internet Protocol (IP) is used beween an ACR and RASs and also between ACRs. WiBro uses Time or Frequency Division Duplexing (TDD or FDD) for duplexing and Orthogonal Frequency Division Multiple Access (OFDAS) for robustness against fast fading and narrow-band co-channel interference. So far, five service types have been proposed and incorporated into 802.16e: unsolicited grant service (UGS), real-time polling service (rtPS), extended real-time polling service (ertPS), non-realtime

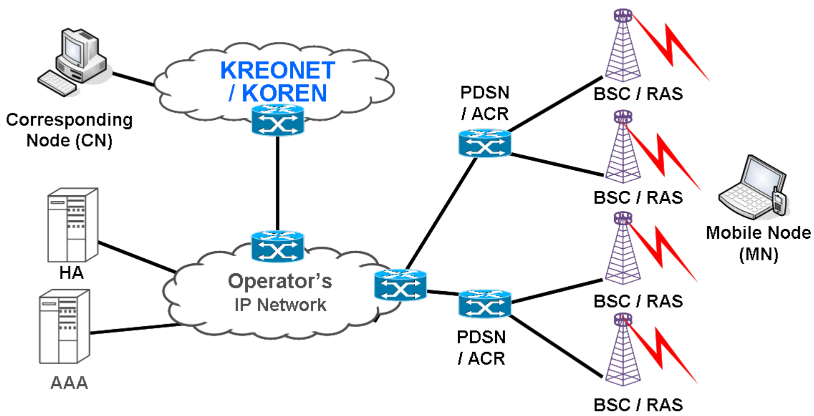


Figure 2.1: Experimental environment over WiBro

Service	CDMA-EVDO	HSDPA	WiBro
Generation	3G	3.5G	3.5G
Based technology	CDMA	WCDMA	OFDMA
Max. throughput	2.4Mbps	14.4Mbps	3Mbps
Mobility	250Km/H	250Km/H	100Km/H
Cell coverage	10Km	10Km	1Km
Target services	Data	Voice&Data	Data

Table 2.1: The comparison among three services

polling service (nrtPS), and best effort service (BE). However, only BE is used in current deployment in Korea.

There are a number of work for modeling the characteristics in wireless networks as well as for proposing solutions to deal with better [?, ?]. Most of these researches are focused on improving the performance of TCP over wireless network, since TCP is the most common transport protocols on the Internet but has many problems due to congestion control relying on the assumption that a packet loss and timeout indicate the congestion. The solutions for improving TCP can be classfied into three categories. First, the link layer protocols such as RLP/RLC which is already implemented in 3G networks to provide reliable link and hide error losses from transport layer by correcting corruption and using local retransmission. However this approach has a side effect of incresing delay. Second, TCP at the end-host is modified to distinguish wireless loss from congestoin and determine sending rate acurately by delay-based calcualtion [?, ?, ?, ?]. Several new timeout and retransmission mechanisms are also proposed to avoid spurious retransmission and to deal with burst loss [?, ?]. Finally, split connection approach tries to separate a slow and lossy wireless link from wired network at the base station. The base station has additional function and buffers to enhance and to keep the concurrent connections. It is reported that several cellular providers implemented split-TCP or TCP-proxy to gain the performace improvement [?, ?].

Ghosh *et al.* use a link-level simulation to analyze the 802.16 fixed WiMax system [?]. Cicconetti *et al.* analyze the effectiveness of the 802.16 MAC protocol for supporting QoS by simulation and evaluate various scheduling algorithms [?]. In [?], the authors evaluate UGS, rtPS, and ertPS scheduling algorithm in IEEE 802.16e system in OPNET simulation. Most of prior work focuses on investigating the performance at the physical layer and MAC layer largely through either simulation or experiment with limited mobility. In this work, we focus on the end-to-end performance at the application layer considering mobility in real life. Our work is unique in that we focus on empirical measurements from a real deployed network, world's first commercial deployment of the mobile WiMax technology.

제 3 장 Experiment Setup and Evaluation Methodology

We begin this chapter with a description of our measurement experiment setup. Then in Section 3.2, we describe the ITU-T E-model for VoIP quality evaluation.

3.1 Performance Measurement

A key factor in this measurement is a mobility. Thus, we have made following four scenarios to conduct UDP, TCP and VoIP experiments considering a mobility and an usage pattern in real-life. Also, the experiments are taken not only by the stationary host but also by the mobile host:

1. Stationary
2. Mobile in a car
   - Around city at the speed of about 100Km/H
   - Campus wide at the speed of about 30Km/H
3. Mobile on a KTX train
   - Across the nation at the maximum speed of about 300Km/H
4. Mobile on a subway
   - Along the Seoul subway line number 6

For all scenarios, we have placed a mobile node (a laptop with an exclusive modem for accessing each network) in each wireless network and installed a stationary node (a desktop PC) connected to the Internet over a fixed line so that we could focus on the 3G and 3.5G wireless network performance. We refer to the laptop as the Mobile Node (MN) and the PC as the Corresponding Node (CN), and mark them as such in Figure **??**. In order to place the CN as close to the wireless network as possible, we use a PC directly connected to a router on KOrea advanced REsearch Network (KOREN) for UDP and TCP experiment and on Korea Research Environment Open Network (KREONET) for

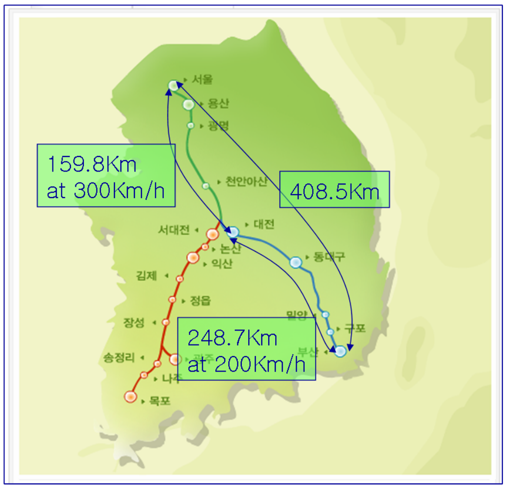
VoIP experiment. Those are research networks that interconnects super computing centers in Korea and also is used as a testbed for new networking technologies. It peers with each operator's IP backbone network at one of Korea Internet eXchange (KIX) points.

The scenario 2 is for a experiment by a MN in a car running at about 100Km/H on a highway around the city of Daejeon. The route is shown in Figure **??** (a). To maintain as the same mobility as possible, the car is drived without stop during a experiment. The measurements over CDMA-EVDO and HSDPA are taken by this scenario. In case of WiBro, it is yet fully deployed across the nation except Seoul in Korea. KT Corporation (also known as Korea Telecom) began the WiBro service in June, 2006, and at the same time built a few trial network around the country. It installed two RASs on the KAIST campus, and turned them on in late 2006. It uses these trial network to evaluate new scheduling algorithms, policies, and equipments before production-line deployment. So we use this trial network for a scenario of mobile car over the WiBro network and the mobility is less than 30Km/H due to restricted speed in a campus.

For a high mobility, scenario 3 is taken on a KTX train, which is running with the maximum speed of 350Km/H across Korea. However, only a measurement over the CDMA-EVDO network is taken since the other networks are yet deployed across the nation. The route is from Seoul to Busan as shown in the Figure **??** (b). Total distance is 1136.6Km and duration is 7 hours. Finally, we have made scenario 4, which is conducted on a subway for VoIP experiment considering usage pattern in real-life. The mobile route is shown in Figure **??** (c). In Korea, KT launched WiBro coverage for nine subway lines in Seoul on April, 2007. The Seoul subway system moves millions of people a day through an extensive network that reaches almost all corners within the city and major satellite cities outside. The maximum speed of Seoul subway trains is 90 km/h, and it takes about 1∼2 minutes between two stations. We have considered measurement experiments in vehicles moving at or under 60 km/h, the upper limit of WiBro, but chosen the subway, as it presents a more popular scenario with users. Commuters in subway are more likely to use mobile devices than those in moving vehicles, as the measurement experiment on a subway train is easier for us. We have conducted our measurement experiments on subway line number 6. It has 38 stations over a total distance of 35.1 km and six RASs.

For our measurement experiments, we generate three types of traffic: UDP constant bit rate (CBR), TCP long-lived bulk traffic, and VoIP. The difference between CBR and VoIP traffic lies in the packet sending rate and follow-up analysis. For all types of traffic, we take measurements when the MN is stationary and mobile. We use *iperf* for CBR and TCP traffic generation [**?**], and *D-ITG* [**?**] for VoIP traffic generation. We configure D-

(a) In a car around city



(b) On a KTX train across nation

(c) On a subway along the Seoul subway line 6

Figure 3.1: Mobile scenarios and routes

ITG to measure round-trip time (RTT) instead of one-way delay, as the MN and CN do not have GPS-quality clock synchronization.

Multiples types of handoff are possible in the 3G and 3.5G network. An inter-PDSN/ACR handoff takes longer than inter-BS/RAS or inter-sector handoff. An inter-sector handoff is between two sectors within a BS/RAS. A BS/RAS typically has three sectors. Using a custom tool developed to monitor inter-sector and inter-BS/RAS handoffs, we collect BS/RAS identifiers and corresponding sector identifiers. By aligning the changes in BS/RAS and sector identifiers with the measurement data, we can pinpoint the moments of handoffs in our data.

3.2 VoIP Quality Evaluation

The classic way to evaluate speech quality is Mean Opinion Score (MOS) [?]. However, it is time consuming, costly, and not repeatable, as human experts are involved in the evaluation. Perceptual Speech Quality Measure (PSQM) [?] and Perceptual Evaluation of Speech Quality (PESQ) [?] are the most common objective measurement methods for voice quality. Both still require a reference signal to compare a degraded speech signal against and predict a MOS value. They are called psychoacoustic models. The ITU–T E-model does not depend on a reference signal, but uses a computational model to predict voice quality directly from network measurements. The output of the E-model is a single value, called an "R–factor", derived from delays and equipment impairment factors. The ITU-T G.107 defines the relationship between the R factor and MOS as below

$$MOS = \begin{cases} 1, & \text{For } R \leq 0 \\ 1 + 0.035R + R(R-60)\cdot(100-R)\cdot 7\cdot 10^{-6} & \text{For } 0 < R < 100 \\ 4.5, & \text{For } R \geq 100 \end{cases} \tag{3.1}$$

The E-model is based on a mathematical algorithm. Its individual transmission parameters are transformed into different individual "impairment factors" that are assumed to be additive on a psychological scale. The algorithm of the E-model also takes into account the combination effects for those impairments in the connection which occur simultaneously, as well as some masking effects.

The R-factor calculated by the E-model ranges from 0 (poor) to 100 (excellent) and can be obtained by the following expression,

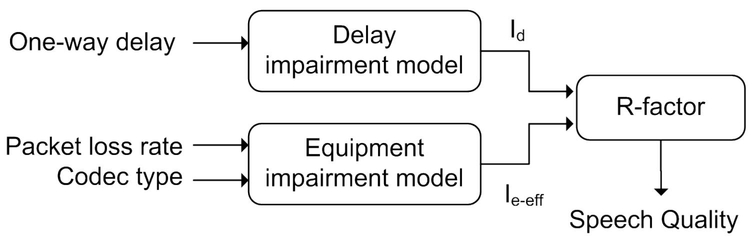


Figure 3.2: Analysis model for the QoS of VoIP

$$R = R_o - I_s - I_d - I_{e-eff} + A, \tag{3.2}$$

where

R_o :	Basic signal-to-noise ratio
I_s :	All impairments that occur more or less simultaneously with the voice signal
I_d :	Delay impairment factor
I_{e-eff} :	Effective equipment impairment factor caused by low bit-rate codec and by packet loss on the network path
A :	Advantage factor

Cole et al. has reduced (**??**) to (**??**) after taking default values for those parameters other than delay and loss [**?**].

$$R = 94.2 - I_d - I_{e-eff} \tag{3.3}$$

In this paper, we use (**??**) in our WiBro VoIP quality and apply Equations (5) and (10) of [?] to translate one-way delay and loss rate to I_d and I_{e-eff}.

제 4 장 Analysis

We have collected a number of data sets of UDP, TCP, and VoIP traffic from Aug 2006. For stationary experiments, we placed the MN on KAIST campus. For mobile experiments, we rode a car along the highway around city of Daejeon, the KTX train moving across nation, and the Seoul subway line 6. For traffic logging, we used *windump* at both the MN and CN. For the VoIP experiments, the MN and CN also dump log files including sequence numbers, packet departure times, acknowledgement arrival times, and calculated

Table 4.1: The summary of UDP, TCP and VoIP experiments (upload/download)

(a) Data sets of UDP experiments

Service	The number of tests			Option
	Stationary	Car	KTX	UDP CBR (Kbps)
SKT-EVDO	9/9	16/16	17/17	120/500
KTF-EVDO	10/10	4/4	14/14	120/500
SKT-HSDPA	10/10	15/15	-	350/1300
KTF-HSDPA	15/15	13/13	-	120/1200
KT-WiBro	14/14	5/5	-	1200/1200

(b) Data sets of TCP experiments

Service	Stationary	Car	KTX	TCP buffer length (Kbyte)
SKT-EVDO	20/20	16/16	15/15	20/64
KTF-EVDO	21/21	9/9	28/28	20/64
SKT-HSDPA	12/12	15/15	-	45/200
KTF-HSDPA	10/10	12/12	-	32/512
KT-WiBro	11/11	10/10	-	128/128

(c) Data sets of VoIP experiments

Service	Stationary	Subway	Codec Type
KT-WiBro	10/10	10/10	G.711 (64Kbps)

round trip time. The complete set of UDP, TCP and VoIP experiments are listed in Table ??.

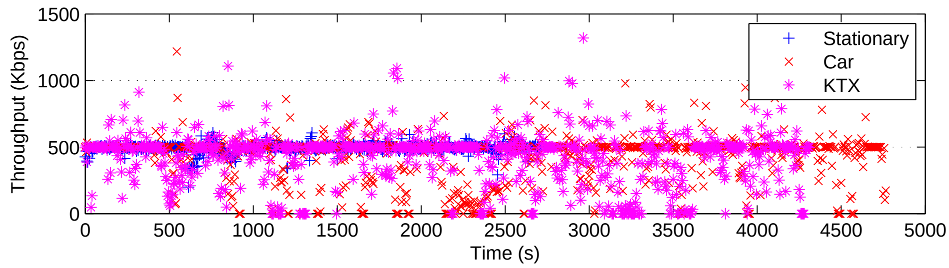
4.1 CBR Traffic Analysis

In order to capture the baseline performance of each wireless network, we first measure the maximum achievable throughput of UDP, and found the capped bandwidth of each network.

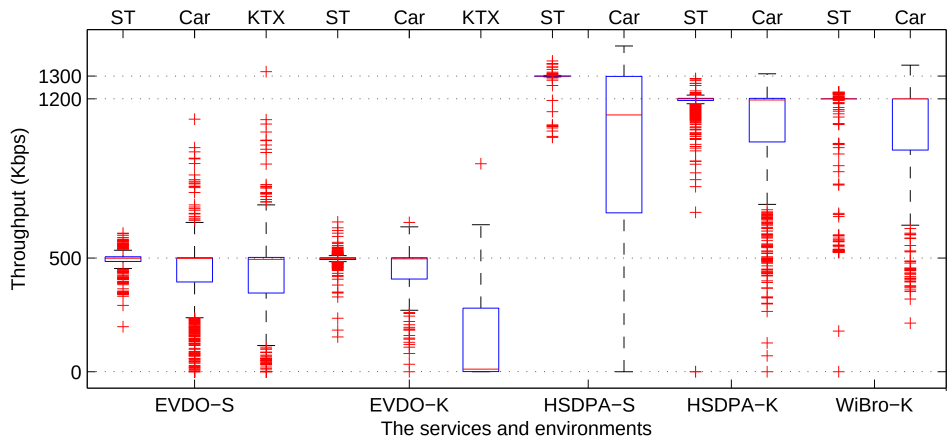
Then we set the transmission rate of our CBR traffic at a rate as listed in Table ?? (a) for uplink and downlink with the packet size of 1460 bytes and saturated the link. We conducted a number of sets of 300-second-long uploads and downloads. Due to limited space, we present only the downlink performance. We first compute the average throughput of CBR traffic over time and plot a result from a SKT-EVDO data set in Figure ?? as a represtative instance due to limited space.

From 9, 17, and 19 about 300-second-long sets for stationary, mobile in a car, mobile on a KTX train case, we get a time span of about 4500 seconds, which is the range on the x-axis. We use a 5-second interval to compute the average throughput. In Figure ??(a), we see that the throughput remains almost constant when the MN is stationary. When the MN is mobile, the throughput fluctuates much more. It is interesting that the flucatuation is similiar to each other in both mobile cases. That is, irrespective of degree of mobility, its impact is consistent. We plot the inter-quartile dispersion of throughput of both the stationary and mobile experiments in Figure ??(b). In the stationary experiment, the inter-quartile range is so small that most 5-second throughput values converge to a rate we set. In the mobile experiment, the inter-quartile range spans widely, and has noticeably more points of outliers. To view the dispersion of throughput in a more visually intuitive way, we plot the variability in Figure ??(c) using the second-order difference plot. The difference between two consecutive throughputs are plotted against that between next two consecutive values. Again in this figure, we find that the mobile station experiences much more throughput fluctuation than the stationary case.

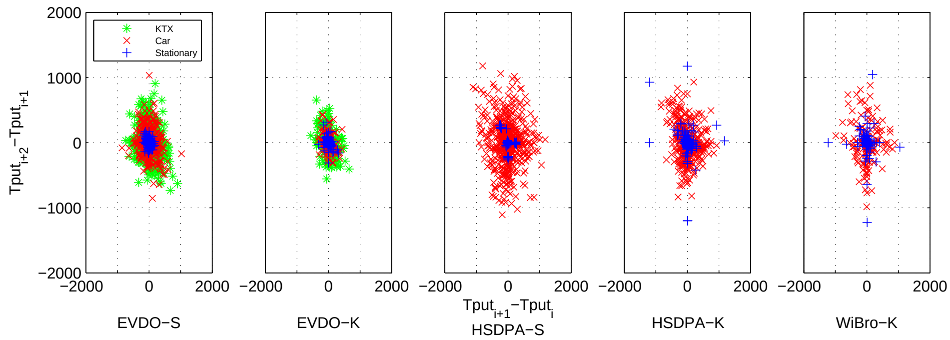
Next, we analyze the jitter and loss rates of CBR traffic. The jitter is defined as the difference between sending intervals and arrival intervals. Figure ??(a) depicts a cumulative distribution function (CDF) of CBR traffic jitter. Although a mobile host experience slightly higher jitter than stationary host, in both environments, more than 90% of jitters are below 100 milliseconds. Given that our traffic by itself saturated the link, this result is encouraging for real-time applications. Now we look at the loss rate of our CBR traffic. In Figure ??(b) the loss rate in the mobile environment is much higher than in stationary. In 3G and 3.5G networks, a MAC layer retransmission mechanism called Hybrid Auto Repeat reQuest (HARQ) is adopted to reduce loss rate at the cost of increased



(a) Fluctuation

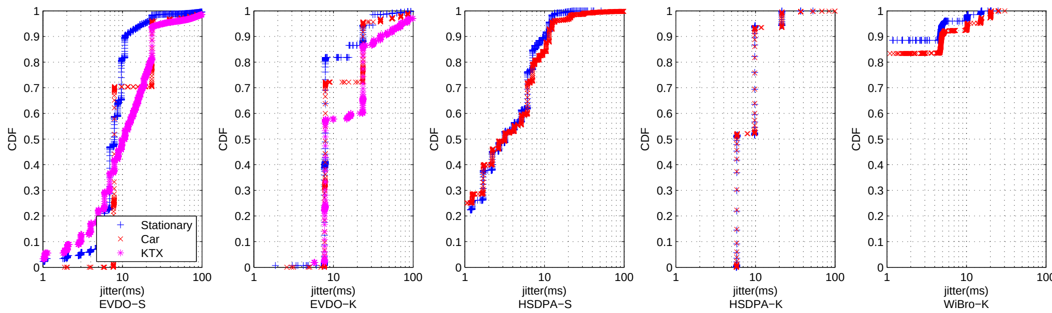


(b) Dispersion

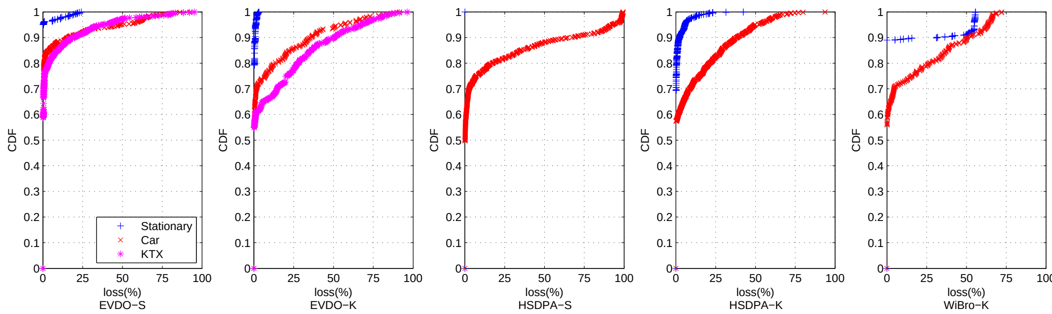


(c) Variability

Figure 4.1: Analysis of CBR traffic throughput over WiBro



(a) CDF of CBR traffic jitter



(b) CDF of CBR traffic loss rate

Figure 4.2: Analysis of CBR jitter and loss over WiBro

delay. As our CBR traffic used UDP as an underlying transport protocol and saturated the link, we expect the loss rates to decrease once we lower the sending rate. We revisit the discussion of the loss rate in the next chapter.

4.2 The TCP behavior

We note the fact that the variability of wireless link state is significantly increased by mobility from the analysis of UDP. The performance of TCP over wireless link suffer from serveral problems such as spurious time-out and inaccurate sending rate estimation caused by those variable rate and delay largely due to mobility. So we can conjecture that TCP with mobility can acheive much less performance than TCP without mobility. However it is not true and the performance of TCP in both environments are similar to each other unlike UDP. Figure **??** shows overall statistics of TCP performance results for both stationary and mobile hosts in box-whisker plot. It is interesting that the although meadian value for a mobile host is slightly lower than that for a stationary host, but 50 percentile

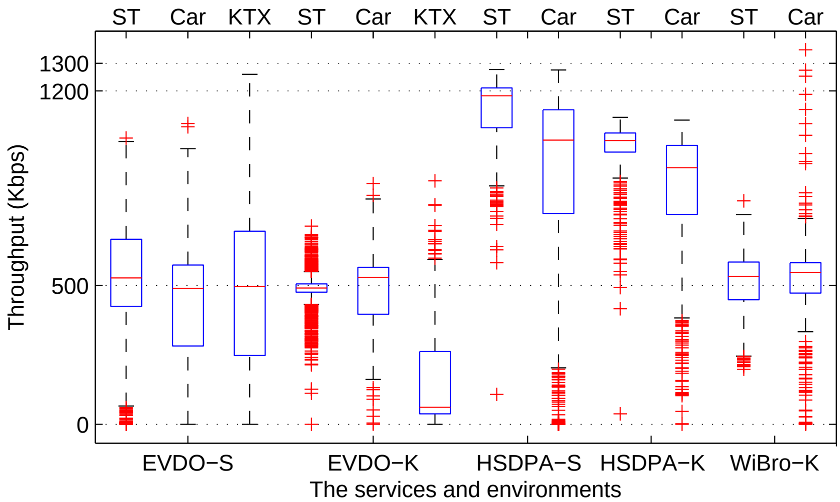


Figure 4.3: The TCP download throughput

range and overall range are almost a similar to each other over most of 3G and 3.5G wireless networks.

To investigate the cause of TCP performance degradation in stationary environment, we further analyze the TCP behavior in detail. First, we analyze ack compression resulted from the effect of variable rate and delay. When a packet is held up at some point due to poor link state, the later packets are accumulated and transferred all together in short period when the link state becomes good. This phenomenon is called as compression and if Ack packets are compressed, this results in overeatimating sending rate at the source. There are several causes but it is largely due to variable rate and delay [?]. We calculated the sending and receiving intervals of Ack packets from the traces collected at both side, which is denoded as $\Delta T_s, \Delta T_r$, respectively If $\Delta T_r/\Delta T_s < 0.75$ in at least three consecutive packets, we consider this a Ack compression event [?]. Figure **??** shows the cumulative distribution function(CDF) of $\Delta T_r/\Delta T_s$. For each ack packet, about 20% and 30% of ack from a stationary and a mobile host are compressed, respectively. However, if we consider only more than three consecutive acks compressed, the proportion of ack compressions are 10.781% and 22.201% in stationary and mobile environment, respectively. Many of ack compressions are occured by duplicated acks. Even for stationary host, the ack compression events are significant and results in misbehavior of TCP.

The TCP interpretes duplicate ack and timeout event as a loss indication and recovers through retransmitting corresponding packets. Sudden-increased delays due to temporal poor link state can lead to an unnecessary timeout and retransmission, so that degrade

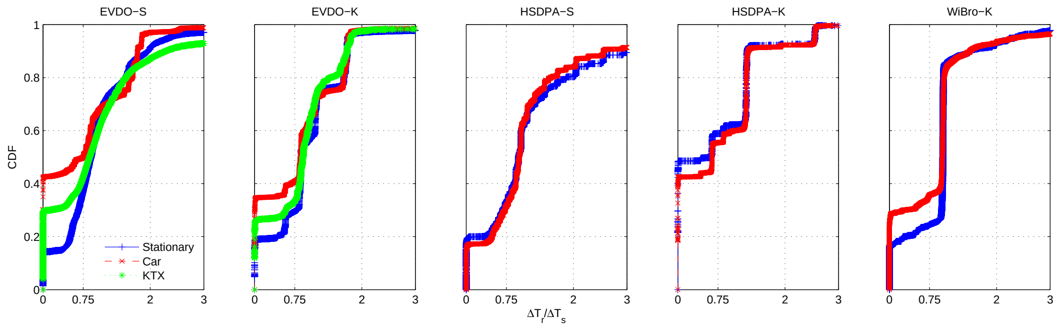


Figure 4.4: CDF of $\mathit{\Delta} T_r/\mathit{\Delta} T_s$

Table 4.2: Propotion of each retransmission type

Rate(%)	% Rxt.	N-Rxt.	S-Rxt.	U-Rxt.	M-Rxt.
Avg. for stationary	0.32	26.13	73.40	0.47	0.991
EVDO-S	0.29	83.31	16.69	0.00	4.074
EVDO-K	0.42	12.09	86.32	1.59	0.469
HSDPA-S	0.54	12.52	87.48	0.00	0.062
HSDPA-K	0.02	23.81	76.19	0.00	0.000
WiBro-K	0.25	6.52	93.17	0.31	0.311
Avg. for mobile	0.62	27.67	71.05	1.28	9.206
EVDO-S	1.42	41.08	57.27	1.66	10.396
EVDO-K	0.98	15.85	82.45	1.70	3.585
HSDPA-S	0.47	4.95	94.97	0.08	7.889
HSDPA-K	0.17	7.75	92.25	0.00	6.295
WiBro-K	0.69	44.01	53.55	2.44	12.860

the performance because TCP reduces congestion window size and enters into slow start phase. We analyze all retransmitted packets in TCP traces at both sender and receiver side and classify into three type of retransmission. Normal retransmission (N-Rxt) is due to loss of data packet, spurious retransmission (S-Rxt) is when there is neither loss of ack nor loss of data packet, and unavoidable retransmission (U-Rxt) is due to loss of ack packet. In addition, we analyze whether there are serial (multiple) retransmissions (M-Rxt) which means that a packet is retransmitted serveral times and suffers from exponential backoff. Table **??** summarize the results of three types of retransmission and serial retransmission.

As shown in Table **??**, the overall retransmission rate for stationary host is lower than for mobile host, but spurious retransmission rate is slightly higher. On the other hand, mobile host signaficantly suffer from serial timeout. That is, there are few significant delays in stationary environment, but TCP for a stationary host unnecessarily expires retransmission timer so that results in spurious retransmissions and degradation of performance.

4.3 VoIP Traffic Analysis

For VoIP experiments, we have generated voice traffic that has the same characteristics of the G.711 voice codec without Packet Loss Concealment (PLC) [**?**]. The payload size is set to 160 bytes and the sending interval to 20 ms in G.711 codec without PLC. The resulting throughput of VoIP traffic is 64 Kbps. We collected 10 300-second-long data sets after transmitting voice packets between the MN and the CN. Because the clocks on the MN and CN were not synchronized, we could not measure the one-way delay accurately. Instead, we took round-trip measurements of VoIP traffic, and halved the delay. In the rest of this section, all the delays we use are calculated as described.

Figure **??** plots the delay, loss, and R-factor of mobile VoIP traffic measurements. We

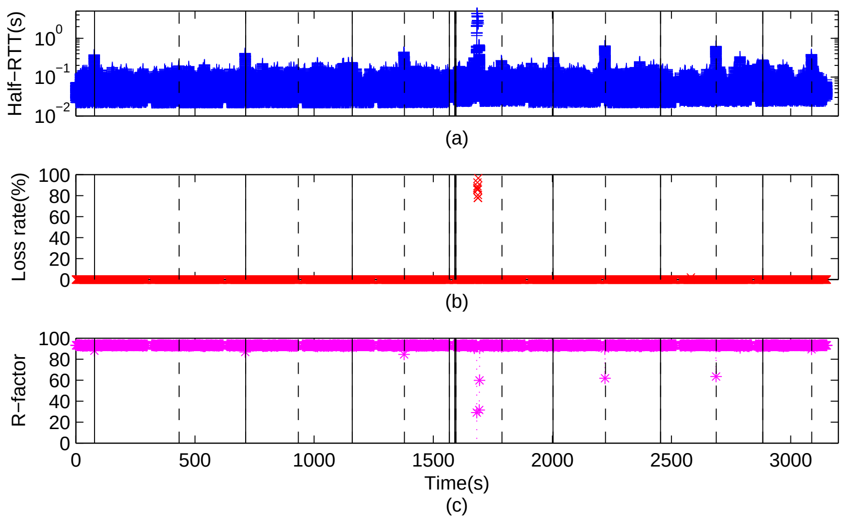


Figure 4.5: The time-series plots of (a) half-RTT, (b) loss rate, and (c) R-factor. Each vertical line indicates the time when either inter-RAS(solid line) or inter-sector(dashed line) handoff occurs

plot loss rates and R-factors calculated over 5 seconds as before. During the mobile experiment, handoffs occurred 17 times and most of delay spikes and burst losses occur near handoffs. We mark the points of the inter-RAS handoffs in solid lines and inter-sector handoffs in dashed lines in Figure **??**. Almost all packets experiences delay below 200 ms, but during handoffs some packets experiences delay over 400 ms. We note a delay spike of 5 sec about half way through the measurement experiment between 1600 and 1700 seconds on the x-axis. Delay spikes and burst losses have been reported in both wired and wireless networks, and there are many possible causes, such as cell or sector reselection, link-level error recovery, wireless bandwidth fluctuation and blocking by higher priority traffic [**?**, **?**]. However, this delay spike does not coincide with a handoff, and needs further investigation.

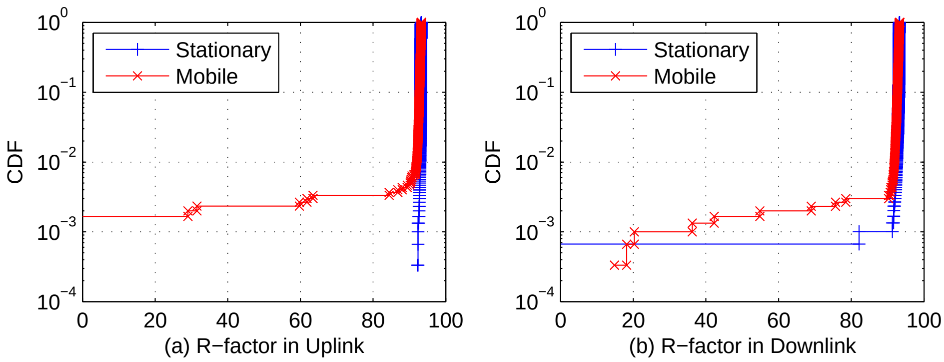


Figure 4.6: CDF of R-factor in uplink and downlink

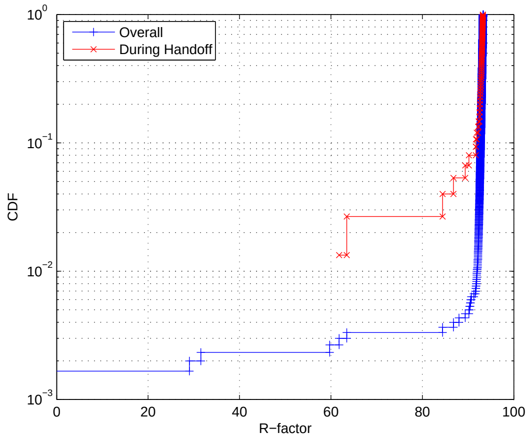


Figure 4.7: CDF of R-factor in 5-sec-intervals vs. during handoffs

Figure **??** shows the R-factor of VoIP quality of uplink (MN to CN) and downlink (CN to MN) in the stationary and mobile cases. The R-factor is calculated every 5 seconds. From the figure we have found that more than 99% of R-factors are above 90 in both the stationary and the mobile case. The R-factor of 70 or above is considered toll-quality, and thus mobile devices attached to the WiBro network are likely to experience toll-quality using VoIP applications.

To quantify the impact of handoffs on QoS of VoIP, we have computed R-factors using average delay and loss rate for the interval of 5 seconds before and after the handoff and compared with the overall cumulative distribution function (CDF) of R-factors in Figure **??**. Here again we observe that the about 99% of R-factors during handoffs are more than 85, which translates to good quality for voice communication.

제 5 장 Summary and Future Work

In this paper, we study the impact of wireless link state and mobility on the performance of transport protocols with end-to-end measurements of UDP and TCP traffic over operational 3G and 3.5G wireless network called as CDMA-EVDO, HSDPA and WiBro. In addition, we have conducted experiments to evaluate QoS of VoIP applications over the WiBro network. We first analyzed the dynamics of wireless link through the UDP experiment and graphically represent the variability with second-order difference plot. In our result of UDP, we observed that the mobility make the wireless link significantly variable in terms of rate and delay as well as more errornous, on the other hand the stationary host acheived stable rate and delay with little loss in the end-to-end long lived bulk traffic. One can conjecture that TCP can adopt to wireless link state derived from UDP results without mobility. However we found out that TCP neither could get enough performance gain compared with UDP for the stationary host nor with TCP for the mobile host. To investigate the cause of perfrmance degradation, we further analyzed two metrics affecting TCP behavior, one is an ack compression and the other is a spurious retransmission. Our analysis explains that TCP for the stationary host suffer from high rate of ack compression and spurious retransmission similar to TCP for the mobile host due to inefficient interplay between TCP congestion control algorithm and wireless link. Although a numerous previous works proposed solutions in several approach, e.g. link level improvement, TCP modification, split sonnection, etc., these have a lack of measurement-based analysis with mobility over operational 3G and 3.5G wirelss network. Finally, we evaluate QoS of VoIP applications using the E–Model of ITU–T G.107. VoIP quality is better than or at least as good as toll quality despite user mobility exceeding the projected limit of WiBro mobility support. Using RAS and sector identification information, we show that the handoff is correlated with throughput and quality degradation. In future work, we will evaluate some promising proposals to improve the performanc over wireless network such as M-TCP, Ack regulator, and F-RTO. Since most of those solutions are based on either a simulation or a experiment with limited mobility, evaluation results can be different in a scenario considering mobility of real-life. We also plan to conduct more experiments with cross-traffic injection and TCP traffic.

이 논문에서는 현재 상용화되어 있는 3세대 및 3.5세대 무선 네트워크, 즉 CDMA-1xEVDO, HSDPA, WiBro, 상에서 광범위한 종단간 측정연구를 통해 무선 링크의 가변적

특성을 알아보고, 그것이 TCP에 미치는 영향을 분석하며, VoIP 애플리케이션의 서비스 품질을 평가하고 있다. 측정된 데이터는 UDP 및 TCP 트래픽과 D–ITG에 의해 시뮬레이션된 VoIP 트래픽 뿐만 아니라 이동기기가 접속한 기지국에 대한 몇가지 물리계층 정보를 포함하고 있다. 3세대 및 3.5세대 무선 네트워크의 기본적인 성능을 확인하기 위해 단말이 고정상태일때와 이동상태일때 모두를 고려한 환경에서 Throughput, Delay, Loss 특성을 측정하고 분석한다. 추가적으로, 가변성의 정도를 정량화하고 이를 Second-order difference plot을 이용하여 시각적으로 제시한다. 그런다음, 특히 단말이 이동하는 환경에서 더 크게 가변적인 무선 링크 상태에서 TCP의 동작 특성을 연구하기 위해, ACK compression 현상, Retransmission의 형태등과 같은 TCP 성능 요소에 대해 추가적인 분석을 수행한다. 마지막으로, ITU–T G.107에 정의된 E-Model을 이용하여 VoIP 애플리케이션의 서비스 품질을 평가한다. 측정결과에서는 각 네트워크의 상향과 하향 링크 최대 Throughtput과 가변성의 정도를 보여준다. 여기서 주목할 점은 심지어 단말이 고정 상태일 때에도 TCP의 Congestion 알고리즘과 무선 링크 관리 Mechanism 사이의 비효율적인 상호작용으로 인해 성능 감소 현상이 발생한다는 것이다. VoIP의 서비스품질은 각 네트워크가 지원하는 최대 이동속도를 초과하는등 사용자의 이동성과 무관하게 최소한 전화의 통화품질과 비슷 하거나 그보다 나은 품질을 보인다. 하지만, 기지국 및 기지국의 Sector 확인 정보를 이용하여 Handoff와 Throughput 및 서비스 품질의 감소 현상이 서로 관련되어 있음을 보인다.

감 사 의 글

가방끈이 너무 짧고 또 낡아서 더 늦기 전에 하나라도 더 배워야 겠다는 결심으로 들어선 길이었기에, 어렵고 힘들었지만 더욱 값지고 보람있는 시간이었습니다. 지난 2년간 애정어린 관심과 격려로 저를 보아 주신 모든 분들 덕분에 제가 이곳에서 석사과정을 무사히 마치고 더 의미있는 삶을 살아갈 수 있게 되었습니다.

2005년 한 통의 이메일로 인연을 시작해, 2년동안 스승으로써 인생의 선배로써 배움을 주시고 이끌어 주신 문수복 교수님께 진정으로 감사의 말씀을 드립니다. 알고 있는것이 부족하고 할 수 있는것이 적어 항상 기대에 미치지 못하고 많은 실망을 드렸지만, 때로는 따뜻한 격로로 때로는 날카로운 비평으로 문제의식과 열정을 그리고 과학도로써의 길을 가르쳐 주셨기에 석사과정을 무사히 마칠 수 있었습니다. 과제를 함께 수행하면서 마찬가지로 많은 조언과 가르침을 주신 충남대학교의 이영석 교수님께도 감사의 말씀을 드립니다. 석사논문의 주제를 결정하도록 아이디어를 주시고 깊은 관심으로 피드백을 주셔서 논문작성에 많은 도움이 되었습니다. 과제를 만들어 연구를 할 수 있도록 기회를 주신 KT의 이재화 부장님과, 조수현 박사님, 곽동진 박사님께도 감사의 말씀을 드립니다. 그리고 석사학위논문 심사를 흔쾌히 허락해 주시고 많은 조언을 주신 윤현수 교수님과 염익준 교수님께도 감사의 말씀을 드립니다.

연구실에 처음 왔을때가 어렴풋이 기억이 납니다. 군에서만 있다가 넘치는 열정과 창의적 사고를 가지고 자유롭게 연구하는 모습을 보고 선뜻 적응 할 수 있을까 하는 의문을 가졌었습니다. 하지만, 반갑게 일원으로써 맞아준 연구실 모두가 너무나 고맙습니다. 항상 모범생으로써 그리고 국제 활동의 본보기로 바쁘면서도, 작은 일에도 신경써주고 논문작성때에도 흔쾌히 리뷰를 해주는 미영씨, 연구실의 든든한 박사 고참으로 무슨 일에든 합리적인 문제해결을 시도하고, 후배 석사들까지도 챙겨준 동기, 공부도 잘하고, 연구도 잘하고, 논문도 잘쓰고 거기다가 유머 감각까지 남다른 해운이, 과제하면서 실험을 위해 운전을 대신해 줄뿐만 아니라 프로그램 코딩까지 도와줘 논문연구에 많은 도움을 준 장건, 나름의 연구를 성실히 완수하고 무사히 석사를 같이 졸업하게된 그리고 너무나 밝은 모습으로 주변의 모든 사람을 밝게 해주는 현우, 이제 한해정도 밖에 인연을 가지지 못하고 헤어져야 하는 아쉬움이 남는 윤찬이와, 창현이, 모두들 2년동안 따뜻하게 대해주고 공부하는데, 연구하는데 많은 도움을 주어 고맙습니다. 지난해 졸업한 그리고 지금은 파리에서 유학중인 종건이, 그리고 세계에서 인정받는 대기업에서 열심히 일하고 있을 태희, 두영이, 토안에게도 고맙고, 덕분에 무사히 졸업할수 있게 되었다고 감사의 뜻을 전하고 싶습니다. 연구실 행정을 맡아 항상 음으로 양으로 도와주신 은진 누님에게도 가족분들 그리고 자녀들 모두 건강하게 항상 잘 지내시길 기원하며 감사의 말씀을 드립니다.

좋은일이나 나쁜일이 있을때 항상 우리 연구실과 함께 기뻐하고 걱정해준 2층연구실의 준복이, 유성이, 영재, 그리고 지금은 졸업한 상호에게도 이 인연을 계속 이어나가길 희망하면서 고맙다는 말을 전하고 싶습니다. 복도를 사이에 두고 서로에게 깊은 관심을 보여준 차성덕 교수님 연구실의 상록이, 준섭이, 동건이 등에게도 감사의 뜻을 전합니다.

처음 학교에 왔을때 익숙하지 않은 환경에 적응하고 군인으로써의 소속감을 잊지않게 도와주셨던 졸업하신 고영일 중령님, 최경식 소령님, 그리고 수업하면서 항상 챙겨주셨던 김찬오 소령님, 권태형 소령님, 졸업하고 애기를 잘 키우면서 군복무를 열심히 하고있을 구다라, 그리고 선배보다 훌륭한 후배들 석중이와 철기, 언제나 든든한 동반자가 되어준 태훈이 에게 졸업후 군에서 다시 만나뵙길 바라면서 감사의 말씀을 드립니다. 큰 힘이 되고, 도움을 주셨던 박성제 선배님, 우리 동기 좌진이, 사랑하는 후배 영구, 졸업한 정훈이, 그리고 이제 새롭게 시작하는 명성이에게도 감사의 말씀을 드립니다.

어머니 보살피면서, 동생마저 큰 도움을 주지못하고 짐을 안겨줘서 항상 죄스럽게 생각하는 형님, 형수님, 동생을 용서하시고 그리고 너무나 고맙습니다. 대전에 있으면서 자주 찾아뵙지도 못하고 도움도 주지 못했지만, 언제나 동생 걱정에 처남 걱정에 신경써주신 큰누나, 매형, 그리고 부산에 계신 작은누나와 매형에게도 감사의 말씀을 전합니다. 대전에서 만나 결혼하고 상준이를 낳아준 우리 이쁜 아내, 언제나 모자라고 부족하지만, 믿고 따라주고 그리고 공부하는 동안 고생하며 참아준 사랑하는 나영이에게, 그리고 힘들고 지친 아빠에게 너무나 벅찬 감동과 희망을 안겨준 우리 복덩이 상준이에게, 수십년을 자식들 뒷바라지 하면서, 이제 그 자식들 덕을 보아야 하는데 몸이 불편해 더욱 가슴이 아픈 어머님께 제 2년의 시간과 노력 그리고 그 마지막 결실인 이 논문을 바칩니다.

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