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# A study of Skype over IEEE 802.16 networks: voice quality and bandwidth usage

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**A study of Skype over IEEE 802.16 networks:  
voice quality and bandwidth usage**

by

Kuan-Yu Chen

A thesis submitted to the graduate faculty  
in partial fulfillment of the requirements for the degree of  
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Major: Computer Engineering

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## ABSTRACT

Skype, one of the popular VoIP applications, has its own redundancy mechanism to mitigate the impact of packet loss at the expense of additional bandwidth usage. However, the benefit of voice quality improvement through redundancy is reduced greatly in the case of consecutive loss of Skype packets. When Skype is running over wireless networks, the high likelihood of having consecutive packet loss in a noisy wireless environment could lead to an interesting scenario: using more bandwidth for a lower voice quality.

In this thesis, we study the impact of the HARQ retransmission mechanism of WiMAX to the voice quality of Skype. Two key parameters for the HARQ mechanism of WiMAX (i.e. Maximum numbers of retransmission and the delay of sending ACK) and their interplay with voice quality are investigated in this research. While HARQ retransmission may reduce the redundancy usage in Skype, it introduces additional bandwidth usage during the retransmission. The overhead of WiMAX retransmission with different HARQ parameters and overall bandwidth usage of Skype are also discussed in this thesis. Numerical analysis and simulation of voice quality and bandwidth usage of Skype over WIMAX are presented. Results show that HARQ retransmission can reduce the impact of bursty noise and improve the voice quality of skype up to 22%. The overall bandwidth consumption can be reduced by 76% when HARQ is engaged. Combining our studies on voice quality and bandwidth usage, we attempt to obtain the optimal HARQ parameters that can lead to the highest voice quality over bandwidth usage ratio for Skype over WiMAX.

## CHAPTER 1. INTRODUCTION

VoIP is a technology which converts analog voice signal to digital signal and transmit it over Internet Protocol (IP)(2). It costs less because the digital signal travels on the computer network and the cost of computer network becomes lower and lower. Therefore, the number of user of VoIP telephony grows very fast in the recent years and it is a big portion of all telephony on the world. However, the quality of service cannot be guaranteed on VoIP telephony due to the computer network which is not designed to support real time service such as VoIP or video stream. TCP is a Internet Protocol which can guarantee the packets transmit to the destination successfully. It retransmits the lost packet until the destination receives the packet correctly. Although TCP can guarantee the packet transmission, it causes much delay to the packets.

TCP source retransmits the packet if it does not receive the corresponding acknowledgment from the destination. It wastes time on waiting ACK and packet retransmission because the delay is increased while the packet needs to be retransmitted. It is not acceptable for VoIP applications because the quality of voice will be degraded significantly when the delay is more than hundreds milliseconds which is happened in the TCP retransmission. UDP is another Internet Protocol which does not produce additional delay because it does not retransmit the packets. Usually, the VoIP applications use UDP to avoid additional delay.

Skype, one of the popular VoIP applications, has to deal with the problem on delay and packet loss. Recently, many papers studied the Skype protocol. But it is hard to analyze the protocol because the Skype traffic is encrypted. Researches only can observe the behavior of Skype in the different network environment. The authors in (7) found that the UDP is applied when there is no firewall between the caller and callee and the TCP is applied when the firewall is existed. The reaction of Skype on packet loss and delay also have been studied recently (11).



The authors observed the size of Skype packet is doubled when the packet loss rate is getting higher. They concluded that the Skype's packet piggybacks a previous packet as redundancy to recover the lost packet. This redundancy mechanism plays a key role to improve the voice quality at the expense of additional bandwidth usage. However, the benefit of redundancy is reduced greatly in the case of consecutive loss of Skype packets.

The wireless physical link is vulnerable due to the noisy wireless environment. The packet loss rate in wireless networks tends to be much higher than it in a wired network. When Skype is running over wireless networks, Skype's redundancy mechanism is invoked as the result of high packet loss rate. This redundancy mechanism consumes twice the bandwidth to carry the additional redundant packets. The extra bandwidth usage is a much more serious cost in wireless networks than it in a wired network. Moreover, the high likelihood of having consecutive packet loss in a noisy wireless environment could reduce the benefit of the redundancy mechanism. This leads to an interesting scenario: Skype over wireless networks might use more bandwidth for a lower voice quality. Obviously, the desirable situation is to have higher voice quality under a lower bandwidth usage. This motivates us to study how to achieve the highest voice quality over bandwidth usage ratio for Skype over wireless.

Retransmission of lost packet is commonly used in wireless network to cope with the noisy channel condition. These retransmission mechanisms, different from the one at the TCP layer, are available at Media Access Control (MAC) and Physical (PHY) layers. The MAC/PHY layer of wireless networks (e.g. WLAN and WiMAX) tends to limit the number of retransmission (of the same packet) to avoid monopolizing the medium. Thus, these retransmission mechanisms cannot guarantee to recover every lost packet. Apparently, all the retransmissions lead to higher bandwidth usage and longer delay. However, the packet delay can be controlled through the number of retransmission and the delay of sending ACK from the receiver. It is worth noting that the number of retransmission also determines the effective packet loss rate. In our recent study, we found that, with just a few retransmissions, the occurrence probability of consecutive loss for the same packet can be reduced greatly. In this thesis, we study the impacts of MAC/PHY layer retransmission which complements the redundancy mechanism of Skype. The objective of this study is to obtain the optimal parameters of retransmission mechanism

that can lead to the best voice quality over bandwidth usage ratio in a wireless network.

Worldwide Interoperability for Microwave Access (WiMAX), a wireless technology based on IEEE 802.16 standard, is deployed over more than 150 countries (6) as 4G networks. VoIP is one of the popular mobile applications. To cope with the high packet loss rate caused by the noisy channel in the wireless network, the Skype will increase the redundancy ratio of packet transmission to improve voice quality at the expense of extra bandwidth usage. In WiMAX Networks, bandwidth is a precious resource to the operators since it determines how many mobile users can be served by a base station. This research investigates the trade-offs between voice quality and bandwidth usage when the Skype is running over WiMAX.

WiMAX employs two retransmission mechanisms: Automatic Repeat Request (ARQ) and Hybrid ARQ (HARQ). The ARQ is implemented in MAC layer and the HARQ is available in MAC/PHY layer. The retransmissions are performed between Mobile Station (MS) and Base Station (BS). The HARQ's retransmission tends to have shorter response time. Therefore, the HARQ is considered as a better candidate for VoIP applications. In this research, we focus on the HARQ mechanism.

In this thesis, we study the impact of HARQ on the voice quality of Skype. The retransmission can reduce the packet loss rate but increase delay. However, both of packet loss rate and delay are the important factors to voice quality. By limiting the maximum number of retransmission and retransmission interval, delay caused by HARQ retransmission can be limited to a tolerable range, so the voice quality can be maintained. Therefore, the voice quality of Skype over WiMAX could be improved by HARQ retransmission. The comparison of voice quality of Skype with and without HARQ is studied in this research. Voice quality we measured in this research is in terms of Mean Opinion Score (MOS).

The benefit of Skype's redundancy mechanism could be degraded by the consecutive packet loss. How to reduce the probability of potential consecutive packet loss is one of the important study in this thesis. The impact of busy noise on the voice quality of Skype can be reduced by retransmission. Maximum numbers of retransmission and the delay of sending ACK (called HARQ\_ACK\_DELAY in WiMAX; it also determines the retransmission interval) are the two key parameters for the HARQ mechanism of WiMAX. In this research, we perform a comprehensive

analysis of the relationship between voice quality of Skype and these HARQ parameters.

By retransmitting packets in MAC/PHY layer, packet loss rate is decreased. Thus, the Skype's redundancy mechanism shall reduce the redundancy ratio after the HARQ is enabled. This reduces the additional bandwidth usage introduced by the redundancy of Skype packets which are at IP layer. However, the HARQ retransmission itself also consumes extra bandwidth, and it may reduce the above mentioned benefit of HARQ. The overhead of WiMAX retransmission with different HARQ parameters and the overall bandwidth usage are also discussed in this thesis. Combining our studies on voice quality and bandwidth usage, we attempt to obtain the optimal HARQ parameters that can lead to the highest voice quality over bandwidth usage ratio for Skype over WiMAX.

We provide numerical analysis for the voice quality and bandwidth usage of Skype under different level of packet loss and noise burstiness. The interplay of quality and bandwidth with HARQ parameters is also considered. The result of numerical analysis is computed by Matlab. The dummynet (1) is used in this research to set the packet loss rate. Thus, we can get the Skype redundancy information under different packet loss rate. We use the information of Skype redundancy in our simulation and simulate a WiMAX wireless network with bursty noise to verify our numerical analysis.

The contributions of this thesis are organized as follows:

- To the best of our knowledge, we are the first to introduce the voice quality over bandwidth usage ratio in the study of Skype over wireless networks.
- We provide a detail analysis of Skype over WiMAX. Our model is used to evaluate the delay, packet loss rate and voice quality of Skype under different probability of packet loss and burstiness packet loss.
- We show the significant improvement of voice quality of Skype when the WiMAX HARQ is enabled.
- Based on our model and simulation, we show the bandwidth usage of Skype's redundancy mechanism could be reduced by enabling HARQ.

- With given probability of packet loss and burstiness of packet loss, we attempt to obtain the optimal HARQ parameters that can lead to the best voice quality over bandwidth usage ratio.

The rest of this thesis is organized as follows. In Chapter 2, we introduce some related works on Skype and retransmission scheme. In Chapter 3, we describe the background of HARQ and the quality evaluating method of VoIP. In Chapter 4, we provide our analytical model of Skype's piggyback algorithm and HARQ retransmission. We present our numerical and simulation result in the Chapter 5. The conclusion is given in Chapter 6.

## CHAPTER 2. RELATED WORK

The voice quality of VoIP is an important issue and the capacity of WiMAX is another. Because all the users share a limited bandwidth in a wireless network, the capacity of wireless network is not unlimited. Once each user requests more bandwidth or the signaling message is increased, the user capacity of network is decreased. Either decreases the bandwidth usage of each user or reduces the size of signaling message can increase the user capacity. For VoIP users, the lower the bit rate could downsize the packet but it also lower the quality of voice. Reducing the size of signaling message by ignoring some message causes extra delay for VoIP users and it also degrade the voice quality. The challenge is, therefore, how to increase the user capacity and keep the quality of voice at the same level.

Some researchers found that the signaling messages in WiMAX may waste the bandwidth in certain conditions. For example, MAP message could be a large portion of downlink subframe. Because the MAP message is used to inform all the mobile stations of the allocation in the frame, the message should be robust enough to ensure that all the stations can receive the correct MAP message, or the users do not know its allocation in the frame. Therefore, the low data rate modulation is used to transmit MAP message. It consumes resource especially that the large amount of mobile stations are accommodated in the same frame.

The persistent scheduling proposed in (4) is designed to reduce the overhead of MAP message for the periodic traffic which has fixed payload size such as VoIP traffic. When the jitter happens in the transmission through the backbone network, the packets will not arrive the base station periodically. Moreover, the size of VoIP packets are different in the silence and talk period. The resource will be wasted because the fixed allocation size. (16), (15), (17) studied this issue and proposed new schemes to improve the performance of persistent allocation on VoIP.

The interference in the wireless network is also a big challenge and therefore there are many methods to ensure that the packet could be transmitted to the receiver in the wireless network on MAC or PHY layers. Lowering the coding rate of modulation, retransmission and additional FEC bits also can guarantee the the receiver can receive correct packet.

Some papers study the legacy retransmission mechanism over wireless network such as TCP (10). (12) found that if both ARQ and HARQ are enabled in the WiMAX, a single TCP data could be acknowledged up to five times. (13) suggested that the base station buffers the TCP segment and locally retransmits the lost segment can improve the throughput significantly. (8) evaluate the TCP performance under 3G wireless network. Although they proposed a solution called Ack Regulator to mitigate the effect of rate and delay variability, the RTT is still too high to maintain the quality of voice when TCP performs the retransmission.

Moreover, increasing the FEC bits leads the length of payload decreasing, so the trade off between packet correction and goodput should be considered. Fortunately, in the real time service, the loss is tolerable for the quality of voice application. In the other words, the users still satisfy the quality of service when the packets are delay or lost in the transmission. Therefore, the large FEC size and many retransmissions is not necessary.

In (14), the authors studied the effect of FEC and ARQ retransmission on the quality of VoIP under different network environments. They proposed an algorithm for MAC protocol data unit (MPDU) size optimization based on the varying feedbacks from the receiver. There scheme aggregates the multiple MSDUs to a MPDU in the when the channel condition is good to reduce the overhead due to header and increase the goodput. In the bad channel condition, the scheme decreases the payload size to reduce the quality degradation due to packet loss and increase the FEC size to increase the probability of successful packet decoding. Their scheme gets higher packet restore probability and higher capacity compared with the scheme without ARQ. They found the aggregate can reduce the delay impairment, but there is not improvement of loss impairment. The authors also observed the quality of VoIP decreased dramatically when the ARQ retransmission has more than two retransmission due to the delay produce by the retransmissions.

To eliminate the impact of packet delay and loss, not only the MAC layer's retransmission

can mitigate the influence of VoIP, but the VoIP applications themselves also have their own schemes to reduce those effects. The authors in (11), observed the Skype's redundancy mechanism in a controlled network. They evaluated the quality of voice in the network with certain packet loss rate. The authors found the Skype's algorithm can maintain the quality of voice when the noise burst ratio is equal to one. Once the burst ratio is higher than 1, the Skype's redundancy scheme loses performance dramatically. They proposed a solution which tunes the redundancy ratio with different noise bustiness to improve Skype's algorithm. The model they proposed, although, can maintain the quality of voice in different noise burstiness, the voice quality with packet loss rate higher than 5% could not be improved when the burst ratio higher than 1.5.

In this thesis, we do not change the redundancy algorithm to improve the quality of voice. We studied the impact of HARQ on Skype's redundancy mechanism and also analyzed the overall bandwidth usage with and without HARQ retransmission. Moreover, the quality of voice between before and after enabled HARQ in the system and the effect of different HARQ parameters are studied.

## CHAPTER 3. BACKGROUND

In this section, we provide the background information of ARQ and HARQ in WiMAX and describe the function of mean opinion score (MOS).

### 3.1 ARQ

Automatic Repeat Request (ARQ), a mechanism works in the MAC layer of WiMAX, has two major functions, one is fragmentation and another is retransmission. When ARQ is enabled for a connection, a Service Data Unit (SDU) from upper layer may be partitioned into a set of blocks. The transmitter can pack the blocks into a fragment even the blocks are from different SDUs and encapsulate a fragment into a Protocol Data Unit (PDU).

Retransmission, the another function of ARQ, works like the retransmission mechanism on IEEE 802.11. ARQ blocks are retransmitted depending on the ARQ feedback. There are two types of ACK in the ARQ feedback; selective ACK and sequence ACK. The ARQ block or the sequence of blocks should be retransmitted if the corresponding bit is set to zero in the selective ACK MAP or in the sequence ACK MAP.

### 3.2 HARQ

Hybrid ARQ (HARQ) combines error correction and retransmission mechanism and works across MAC layer and PHY layer. HARQ may pack MAC PDUs together and map onto PHY burst. HARQ scheme is a simple stop-and-wait protocol at the FEC block level. The received retransmission packets are combined together to correct the error. The HARQ packet in a HARQ channel should wait for its corresponding ACK, otherwise the HARQ packet could not be removed from the buffer. If the receiver cannot decode the HARQ packet, it should send a



NACK to the sender. If the sender does not receive a ACK or receive a NACK, the sender will start the retransmission.

There are two types of coding for HARQ, one is Chase Combining and another is Incremental Redundancy (IR). The PHY encodes a HARQ packet into several subpackets. For Chase Combining, the HARQ subpackets are encoded in one version, so the all retransmitted packets are the same. The subpackets for IR are encoded in different versions, so all the retransmitted subpackets are totally different and each subpacket has a unique Subpacket ID (SPID). Instead of combining the received packets to correct the error, any error packet will be dropped in our simulation.

### 3.3 Measurement of Quality of VoIP

There are many methods can quantify the quality of VoIP. The ITU-T Recommendation G.107 (5) defines a E-model which based on the equipment impairment to determine the voice transmission quality from mouth to ear. The output of this model is R-factor which is defined by

$$R = 100 - I_s - I_d - I_e + A. \quad (3.1)$$

$I_s$  is the signal-to-noise impairment associated with typical switched circuit network path.  $I_d$  is the impairment caused by mouth-to-ear delay.  $I_e$  represents the impairment factor contributed by data lost in the transmission.  $A$  is an expectation factor which depends on the communication system of users.

Because VoIP works on packet network which is different from the typical switched circuit networks, the R-factor can be expressed as

$$R = 94.2 - I_d - I_e, \quad (3.2)$$

where only  $I_d$  and  $I_e$  are required in the equation (3.2).

$I_d$  expresses the effect of delay from mouth to ear and it is composed by three factors: codec delay ( $d_{codec}$ ), network delay ( $d_{network}$ ), and playout delay ( $d_{playout}$ ). The codec delay is the delay due to the codec which needs time to encode or decode the voice. It is varies from codec to codec because different codec has different algorithm to sample and quantify

the voice. A network delay is produced when the digital voice data which is transmitting through the network, the routing and the propagation needs certain amount of time depends on the transmission distance. A playout delay is happened when the receiver side buffered the data after receiving the digital data from network for decoding. Therefore the delay could be expressed as

$$d = d_{codec} + d_{network} + d_{playout}. \quad (3.3)$$

Moreover, there is a delay value which will affect the quality of voice very critically. When the total delay of VoIP excess 177.3ms, it will affect the quality of VoIP seriously. Therefore the impact of delay is defined as follows,

$$I_d = 0.024d + 0.11(d - 177.3)H(d - 177.3). \quad (3.4)$$

The  $H(x)$  is a Heavyside function where

$$H(x) = \begin{cases} 0, & \text{if } x < 0 \\ 1, & \text{else} \end{cases}$$

The packet lost rate of VoIP between the caller and callee also affect the quality of voice. The packet loss rate  $e$  includes not only the packet loss rate but the codec itself. The network loss rate ( $e_{network}$ ) and receiver's playout loss rate ( $e_{playout}$ ) both involve the total packet loss rate, so the  $e$  could be writhen as

$$e = e_{network} + (1 - e_{network})e_{playout}. \quad (3.5)$$

Besides the packet loss rate, the codec itself also produces the losses. The effect of packet loss rate is presented as

$$I_e = \gamma_1 + \gamma_2 \ln(1 + \gamma_3 e) \quad (3.6)$$

where  $\gamma_i$  is the parameter which determines the impairment to the R-factor for different codecs. For G.729,  $\gamma_1 = 11$ ,  $\gamma_2 = 40$ , and  $\gamma_3 = 10$ .

The R-factor could be rewritten as

$$R = 94.2 - (\gamma_1 + \gamma_2 \ln(1 + \gamma_3 e)) - (0.024d + 0.11(d - 177.3)H(d - 177.3)) \quad (3.7)$$

To estimate the MOS from R-factor, (5) provides a transfer function as

$$MOS = 1 + 0.035R + 7 \times 10^{-6}R(R - 60)(100 - R) \quad (3.8)$$

for  $0 < R < 100$ . We will use these functions to measure the quality of VoIP in the our experiments.

## CHAPTER 4. ANALYSIS

In this section, we present the observation of Skype's redundancy mechanism and the analysis of its algorithm. We obtained the result under different packet loss rate and different busrty noise. The Skype's redundancy algorithm will be integrated into our WiMAX HARQ simulation.

### 4.1 The observation on Skype's redundancy mechanism

In the papers (7), (11), the authors observed that the Skype's packet size is increased when the network loss rate is increased. To figure out the Skype's packet redundancy ratio in different packet loss rate, we use dummynet (1) to control the packet loss probability and observe the reaction of Skype. In our experiment, we setup two computers with Windows operation system and connect the machines to a wired Local Area Network. Skype is installed in the two computers with version 3.1. The Wireshark (3) is used in this experiment to monitor the Skype's data flow and collect the packets on IP layer packet. We only consider the packet transmitted under UDP since UDP is commonly used in Skype application. The packet loss rate is controlled from 0% to 10% in 1% incremental every 60 seconds to observe the reaction of Skype's redundancy mechanism.

To determine whether the packet has redundancy, the first step is to define the size of packet without redundancy. Because we only test on Skype-to-Skype traffic, the iSAC codec is used by Skype (11), (7), which support variable bit rate. We have to find out the threshold of payload size of Skype's packet without redundancy. To avoid the Skype generates packet with redundancy, we set the probability of packet loss to 0%. Therefore, we can determine the maximum payload size of Skype's packet without redundancy. Conversely, the packet size over

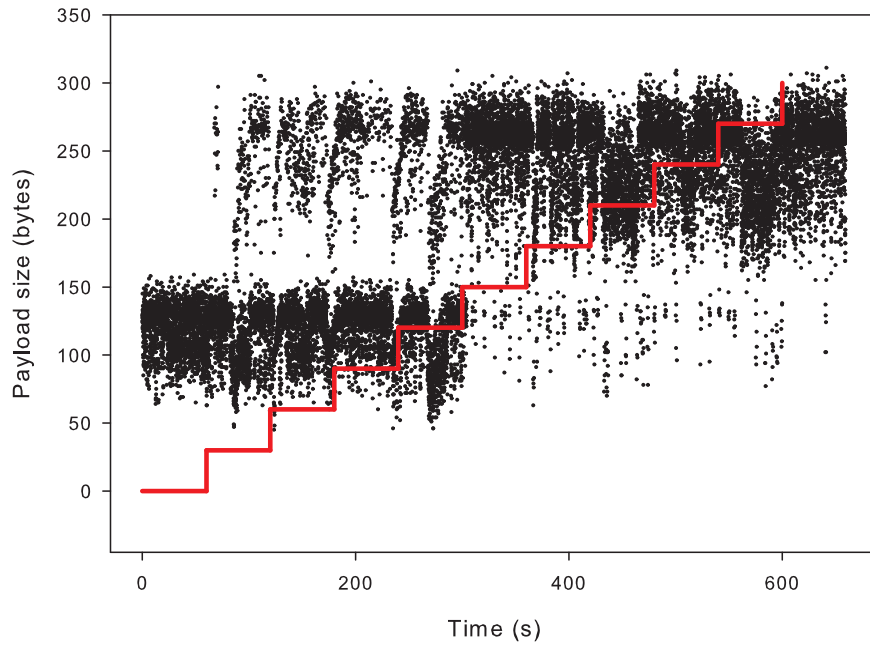


Figure 4.1 The payload size of Skype packet

the threshold is considered as packet with redundancy. The payload size under different packet loss rate is shown in Fig. 4.1.

The maximum payload size in the interval of 0% packet loss rate we could find is 159 Kbytes, so any packet which has payload size larger than 159 Kbytes is considered as the packet with redundancy. The redundancy ratio we define here is that the percentage of the packets has redundancy. The relationship between redundancy ratio and packet loss rate is shown in Fig. 4.2. In Fig. 4.2, we observe that the redundancy ratio grows gradually from 0 to 0.3 when the packet loss rate increases from 0% to 3% and it raises to 0.9 tremendously when the packet loss rate is over 4%. It means that almost every packet piggybacks a previous packet when the packet loss rate exceeds 4% and bandwidth usage is almost doubled by comparing with the bandwidth usage at 0% packet loss rate.

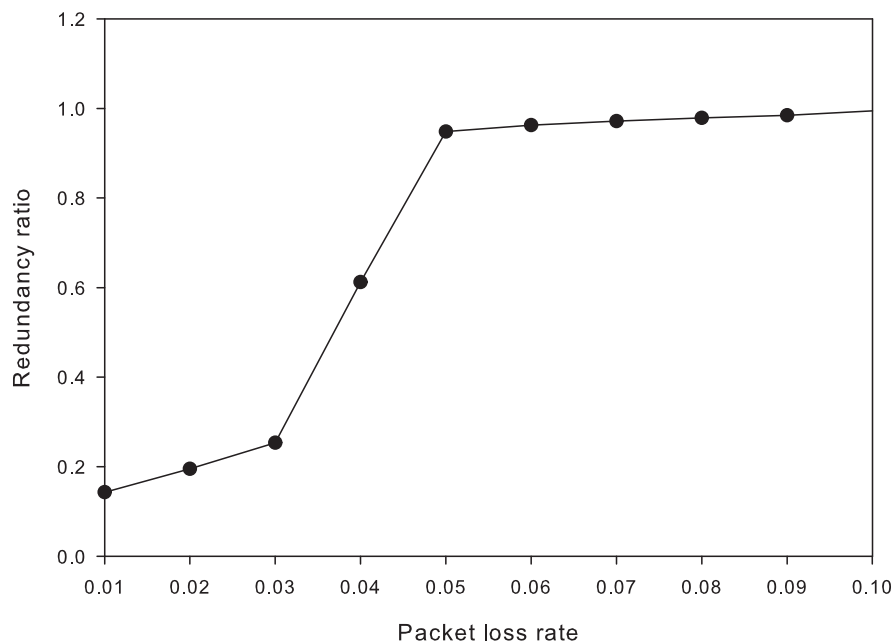


Figure 4.2 Skype's redundancy ratio under different system loss rate

## 4.2 The impact of Bursty Packet Loss

The limitation of piggyback mechanism is that the packet is hard to be recovered under continuous packet loss. Here we study the relationship between voice quality of Skype and bursty packet loss. The quality of voice is degraded tremendously when noise burst ratio ( $R_{burst}$ ) is increased (11). In (5), the burst ratio of loss is defined as follow.

$$R_{burst} = \frac{L_c}{L_r},$$

where  $L_c$  is the average length of observed consecutive losses and  $L_r$  is the average length of consecutive losses under random losses. To perform the burstiness of packet loss, we apply Gilbert model (9) to our analysis and simulation. The Gilbert model uses a two states Markov chain to generate bursts. One state is called *good state* and another is called *bad state*. The packet is received by the receiver in the good state and the packet is lost when the state is bad. Therefore, we also call good state as received state and call bad state as loss state. The probability the state transits from good to bad is  $p$  and the  $q$  is the probability that the state

Table 4.1 Notation and Description

Notation	Description
$T_f$	MAC frame duration
$P_l$	Probability of packet loss in transmission
$D$	Receiver should send ACK after a fixed delay which is in number of MAC frames (HARQ_ACK_Delay)
$R_x$	The maximum number of retransmission
$T_r$	Retransmission interval
$p$	transition probability from loss to received
$q$	transition probability from received to loss
$R_{burst}$	Burst ratio
$P_{LR}(n)$	The transition probability from loss to received, number of $n$ frames are in between
$P_{LL}(n)$	The transition probability from loss to loss, number of $n$ frames are in between
$P_{RR}(n)$	The transition probability from received to received, number of $n$ frames are in between
$P_{RL}(n)$	The transition probability from received to loss, number of $n$ frames are in between
$T_{PIGGYBACK}$	The delay produces by Skype's redundancy mechanism without enabling HARQ
$T_{HARQ}$	The delay of a packet without duplicate packet when WiMAX HARQ is enabled
$T'_{PIGGYBACK}$	The delay produces by Skype's redundancy mechanism with enabling HARQ

goes from bad to good. Moreover, the probability of the state stays at good is  $1 - p$  and the state stays at bad with probability  $1 - q$ .

Table 4.1 summarizes notations we used in the analysis. We assume that the packet which is generated in  $n$ th frame will be sent in frame  $n + 1$ , and the state status is changed frame by frame depends on the transition probability.

The transition probability  $p$  and  $q$  can be determined by target burst ratio ( $R_{burst}$ ) and loss rate ( $P_l$ ),

$$q = (1 - P_l)/R_{burst} \quad (4.1)$$

$$p = P_l/R_{burst} \quad (4.2)$$

There are two scenarios that the packet cannot arrive the receiver. First, a packet without a duplicate packet will be considered lost when the packet is transmitted in the time frame with bad state and the loss probability is  $P_l$ . The ratio of the packet without a duplicate packet is  $1-R_r$  where  $R_r$  is the Skype's redundancy ratio. Second, the packet which has a duplicate packet will be considered lost when the original packet is transmitted in the time frame with bad state and the duplicate packet is transmitted in the next time frame with bad state as well and therefore the original and duplicate packet are both lost. The loss rate in the second situation can be expressed by  $P_l(1-q)$  and the ratio of these packets is  $R_r$ . The overall packet loss rate is expressed as

$$P_{PIGGYBACK} = P_l \cdot (1 - R_r) + P_l(1 - q) \cdot R_r, \quad (4.3)$$

where  $P_l \cdot (1 - R_r)$  is the probability of a random packet which is lost without a duplicate packet, and  $P_l(1 - q) \cdot R_r$  is the probability of a random packet which is lost with a duplicate packet.

Similar to the loss rate, the delay are discussed in two parts. Due to the packet is generated in frame  $N$  and transmitted in frame  $N + 1$ , every packet has at least one MAC frame delay ( $T_f$ ). The delay of original packet is one  $T_f$  and the probability of receiving the original packet is  $1 - P_l$ . The probability of the receiver without receiving the original packet but receives the duplicate packet is  $P_l \cdot q$  and the delay is  $2T_f$ . Because the delay of lost packet is ignored, we only take the delay of received packet into account. Thus, we normalize the delay equation by the factor of the percentage of received packet. The expected delay is expressed as

$$T_{PIGGYBACK} = T_f \frac{((1 - P_l) + 2R_r P_l \cdot q)}{(1 - P_{PIGGYBACK})}. \quad (4.4)$$

### 4.3 Impact of WiMAX HARQ on the voice quality of Skype

#### 4.3.1 Delay

In this section, we model the impact of WiMAX HARQ on the quality of Skype. The WiMAX HARQ retransmission mechanism is briefly summarized as follow. The sender retransmits the HARQ packet at  $n$ th frame and wait corresponding ACK at  $n + D$ th frame,



where  $D$  represents the HARQ\_ACK\_Delay. If the sender does not receive the corresponding ACK at frame  $n + D$ , the sender should retransmit the packet at frame  $n + D + 1$ . Thus, the retransmission interval  $T_r$  is  $D + 1$ .  $P_{LR}$  is the transition probability from bad state to good state. There are four probabilities,  $P_{LR}$ ,  $P_{RR}$ ,  $P_{LL}$ , and  $P_{RL}$  in Table 4.1, represent four probabilities of state transition from  $N$ th frame to  $N + n$ th frame are expressed as

$$P_{LR}(n) = P_{LR}(n-1)(1-p) + P_{LL}(n-1)q \quad (4.5)$$

$$P_{RR}(n) = P_{RR}(n-1)(1-p) + P_{RL}(n-1)q \quad (4.6)$$

$$P_{LL}(n) = P_{LL}(n-1)(1-q) + P_{LR}(n-1)p \quad (4.7)$$

$$P_{RL}(n) = P_{RL}(n-1)(1-q) + P_{RR}(n-1)p, \quad (4.8)$$

where  $P_{LR}(0) = q$ ,  $P_{RR}(0) = 1 - p$ ,  $P_{RL}(0) = p$ , and  $P_{LL}(0) = (1 - q)$ .

To model the packet delay and loss probability, we distinguish the Skype packets in two cases: one case is the packet which has a duplicate packet and the other is the packet which has no duplicate packet. When a packet has no duplicate packet, the delay of a packet received at  $n$ th retransmission is  $T_f \cdot (1 + nT_r)$ , where  $T_r$  is the retransmission interval (in number of frames) and  $T_f$  is MAC frame duration. Without a duplicate packet, the probability of a packet received at  $n$ th retransmission is

$$a'_n = P_l \cdot P_{LL}(T_r)^{n-1} \cdot P_{LR}(T_r)$$

The expected delay of a packet is received at  $n$ th retransmission is

$$\begin{aligned} a_n &= T_f \cdot (1 + nT_r) \cdot a'_n \\ &= T_f \cdot (1 + nT_r) \cdot P_l \cdot P_{LL}(T_r)^{n-1} \cdot P_{LR}(T_r), \end{aligned}$$

where  $n$  is the number of retransmission. The expected delay of a packet received at first transmission is  $T_f \cdot (1 - P_l)$ .

Without a duplicate packet, the expected delay of a packet with maximum number of retransmission  $R_x$  is

$$\begin{aligned}
T_{HARQ} &= (T_f(1 - P_l) + \sum_{n=1}^{R_x} a_n) \cdot \frac{1}{1 - P_l P_{LL}(T_r)^{R_x}} \\
&= T_f[(1 - P_l) + \\
&\quad P_l \sum_{n=1}^{R_x} (1 + nT_r) P_{LL}(T_r)^{n-1} P_{LR}(T_r)] \cdot \\
&\quad \frac{1}{1 - P_l P_{LL}(T_r)^{R_x}},
\end{aligned} \tag{4.9}$$

where  $1 - P_l P_{LL}(T_r)^{R_x}$  is the percentage of received packet. Because we only take the delay of received packet into account, (4.9) is normalized by the percentage of received packet.

With the Skype's redundancy mechanism, the duplicate packet is sent in  $N + 1$ th frame when the original packet is transmitted in  $N$ th frame. Either the original or duplicate packet is received by the receiver, the packet is considered received. Both original and duplicate packet are transmitted and retransmitted individually by HARQ retransmission mechanism. The probability that only the original packet is received at  $n$ th retransmission is

$$\begin{aligned}
b'_n &= P_l \cdot (1 - q) \cdot P_{LL}(T_r - 1)^{n-1} \cdot (1 - q)^{n-1} \cdot P_{LR}(T_r - 1) \\
&= P_l \cdot (1 - q)^n \cdot P_{LL}(T_r - 1)^{n-1} \cdot P_{LR}(T_r - 1).
\end{aligned}$$

The expected delay that an original packet is received at  $n$ th retransmission is

$$\begin{aligned}
b_n &= T_f \cdot (1 + nT_r) \cdot b'_n \\
&= T_f \cdot (1 + nT_r) \cdot P_l \cdot (1 - q)^n \cdot P_{LL}(T_r - 1)^{n-1} \cdot P_{LR}(T_r - 1)
\end{aligned}$$

and expected delay that an original packet is received at first transmission is  $T_f \cdot (1 - P_l)$ .

The probability for only the duplicate packet received at  $n$ th retransmission is

$$c'_n = P_l \cdot ((1 - q) \cdot P_{LL}(T_r - 1))^n \cdot q.$$

The expected delay of receiving the duplicate packet at  $n$ th retransmission is

$$\begin{aligned}
c_n &= T_f \cdot (2 + nT_r) \cdot c'_n \\
&= T_f \cdot (2 + nT_r) \cdot P_l \cdot ((1 - q) \cdot P_{LL}(T_r - 1))^n \cdot q
\end{aligned}$$

and the expected delay of receiving a duplicate packet at first transmission is  $2T_f \cdot P_l \cdot q$ . Therefore, the packet delay with Skype's redundancy mechanism and HARQ retransmission is expressed as

$$\begin{aligned}
& T'_{PIGGYBACK} \\
& = \{T_f \cdot (1 - P_l) + 2T_f \cdot P_l \cdot q + \sum_{n=1}^{R_x} (b_n + c_n)\} \cdot \\
& \quad \frac{1}{1 - P_l(1 - q)^{R_x+1} \cdot P_{LL}(T_r)^{R_x}} \\
& = T_f \cdot \{(1 - P_l) + \\
& \quad P_l \sum_{n=1}^{R_x} (1 + nT_r)(1 - q)^n P_{LL}(T_r - 1)^{n-1} P_{LR}(T_r - 1) + \\
& \quad P_l \cdot q [2 + \sum_{n=1}^{R_x} (2 + nT_r)((1 - q) \cdot P_{LL}(T_r - 1))^n]\} \cdot \\
& \quad \frac{1}{1 - P_l(1 - q)^{R_x+1} \cdot P_{LL}(T_r)^{R_x}}.
\end{aligned} \tag{4.10}$$

The (4.10) is normalized by the factor of percentage of received packet because the delay of dropped packet is not counted.

From (4.9) and (4.10), the overall expected delay under redundancy ratio  $R_r$  is derived as

$$T_D = T'_{PIGGYBACK} \cdot R_r + T_{HARQ} \cdot (1 - R_r). \tag{4.11}$$

### 4.3.2 Packet loss rate

Similar to delay, we also model the packet loss rate in two parts: the model with and without Skype's redundancy mechanism. A packet without Skype's redundancy mechanism is considered loss when all the transmission of the certain packet are all transmitted in the frames with bad state. The loss probability for the first attempt is  $P_l$ . For the same packet, the probability of failing in retransmission is  $P_{LL}$ . The sender will not retransmit any more when the attempt of retransmission reaches  $R_x$  times. Therefore the probability of packet loss without Skype's redundancy mechanism is expressed as

$$P_{HARQ} = P_l \cdot P_{LL}(T_r)^{R_x}. \tag{4.12}$$

By applying the Skype's redundancy mechanism, a packet is considered lost if the original and the duplicate packet are both lost. The probability of original packet at the first transmission in frame  $N$  with bad state is  $P_l$ . Because the duplicate packet is piggybacked by the next packet and transmitted in the frame  $N + 1$ , the probability for frame  $N + 1$  with bad state is  $P_l \cdot (1 - q)$  which also represent the loss probability of the duplicate packet. Since the first retransmission of original packet is transmitted after  $D$  frames of the first transmission of duplicate packet, the packet is retransmitted in frame  $N + 1 + D$ . The probability of the frame  $N$ ,  $N + 1$ , and  $N + 1 + D$  that are all in the bad state is  $P_l \cdot (1 - q) \cdot P_{LL}(D)$ . Therefore, the probability of the frame  $N + D + 2$  with bad state is  $P_l \cdot (1 - q) \cdot P_{LL}(D) \cdot (1 - q)$  which is the frame of the second attempt for the duplicate packet. Therefore, the loss probability of a packet with Skype redundancy mechanism with enabling WiMAX HARQ can be represented as

$$P_{PIGGYBACK} = P_l \cdot [(1 - q) \cdot P_{LL}(T_r - 1)]^{R_x} \cdot (1 - q). \quad (4.13)$$

From (4.12) and (4.13), the overall probability of Skype's packet loss under redundancy ratio  $R_r$  with WiMAX HARQ retransmission can be derived as

$$P_L = P_{PIGGYBACK} \cdot R_r + P_{HARQ} \cdot (1 - R_r). \quad (4.14)$$

## CHAPTER 5. NUMERICAL AND SIMULATION RESULT

### 5.1 Simulation Setup

In this section, we validate the analytical model with our simulation and show the results of our simulation. The WiMAX parameters used in simulation are shown in Table.5.1. We develop our own simulator which is written in C language. Each subscriber station only has one connection and the each simulation is last for one hundred thousand frames which is equal to two thousand seconds. The call is made from MS to landline phone and the connection between BS and landline phone is wired without any network error. All the results are the average value from all the subscriber stations.

### 5.2 Numerical result of the impact of HARQ on voice quality

The Fig. 5.1(a), 5.1(b), 5.1(c) shows the numerical result of voice quality with and without enabling HARQ, and it also shows the voice quality under different bustiness noise. In our analysis and simulation, channel error rate is the probability that a wireless connection is in the bad state. The packet loss rate is the end-to-end loss rate. A packet loss is considered as end-to-end loss when the packet and its own duplicate packets are all lost.

Table 5.1 WiMAX parameters

Number of BS	1
Number of SS	100
MAC Frame Duration	20 ms
Modulation	64 QAM

### 5.2.1 Comparison without HARQ under different noise burst ratio

Without HARQ, we observe that the the Skype's redundancy mechanism cannot maintain the quality of voice when the channel error rate is lower than 3% because the drop of MOS is almost 10% by comparing the MOS with channel loss rate of 0% and 3%. Moreover, the Skype's redundancy mechanism does not maintain the same voice quality when the burst ratio is increased. When the channel loss rate is 10%, the MOS with burst ratio 2 is 20% less than the MOS with burst ratio 1. In the conclusion, the Skype's redundancy has limitation under burstiness noise, which corrupt series of packets.

In Fig.5.1(a) and 5.1(b), with higher burst ratio, the delay is lower and the effective loss rate is higher. Because higher burst ratio represents more consecutive frames in bad state, a packet and its duplicate packet, which is the packet piggybacked by the next packet, are both have higher loss probability. It limits the effect of Skype's redundancy mechanism and causes the higher packet loss rate. The delay is lower due to the higher packet loss rate and the delay of lost packet is ignored.

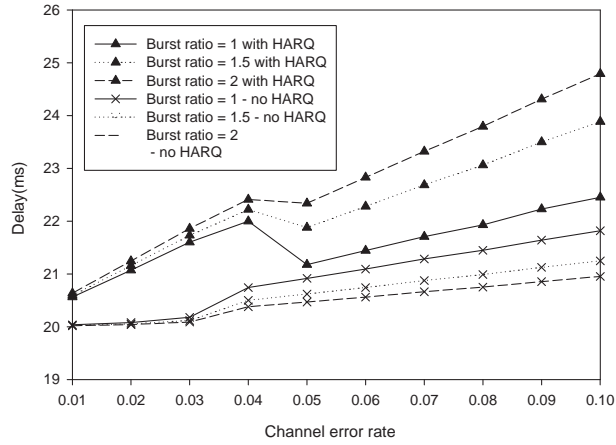
We also show the comparison of numerical and simulation result is Fig. 5.2. We show the percentage of difference by the equation:

$$\frac{\text{Simulation result} - \text{Numerical result}}{\text{Numerical result}} \quad (5.1)$$

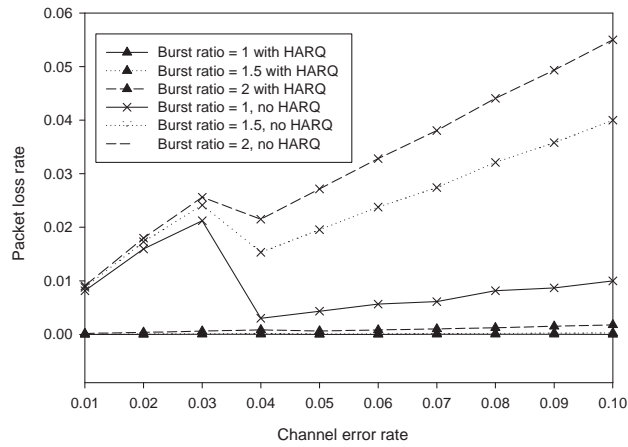
It shows the numerical result of our analytical model is very close to the simulation.

### 5.2.2 Comparison between enabled HARQ and disabled HARQ

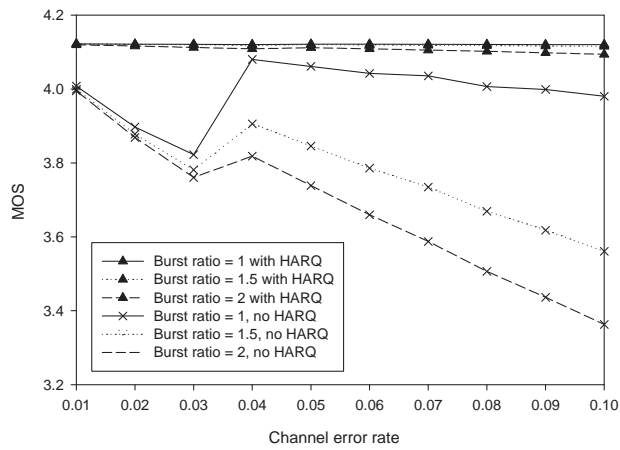
The Fig.5.1 shows the impact of HARQ to the voice quality under burst ratio 1, 1.5, and 2; the Maximum retransmission and HARQ\_ACK\_DELAY are both set to two in this evaluation. We observe the delay in Fig. 5.1(a) with HARQ is higher than Skype's redundancy mechanism alone. It is because the delay produced by Skype's redundancy mechanism is only one additional frame duration. The amount of delay produced by HARQ is depended on the two parameters,  $R_x$  and  $D$ . Each retransmission produces  $1 + D$  frame duration delay and  $R_x$  limits the maximum delay of HARQ packet.



(a) Delay versus channel error rate



(b) Packet loss rate versus channel error rate



(c) MOS versus channel error rate

Figure 5.1 Comparison of delay, packet loss rate, and MOS with and without HARQ

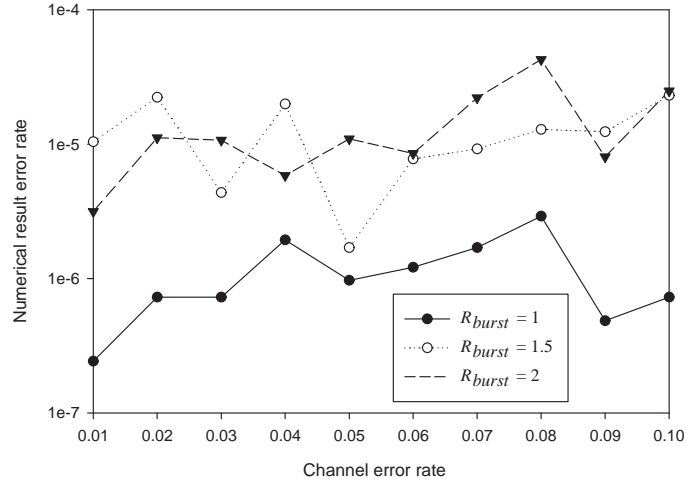


Figure 5.2 Difference between Simulation and numerical result.

As shown in Fig.5.1(b), enabling HARQ lower the effective loss rate. It is because the Skype's redundancy mechanism only transmits a duplicate packet to recover the potential loss. HARQ provides extra retransmissions for both original packet and its duplicate packet to increase the probability of packet recovering. For example, without HARQ, one packet can be transmitted twice; with HARQ, a packet will be transmitted at most  $2 \cdot (1 + R_x)$  times. Although the HARQ retransmission produces extra delay from 10% to 20%, the MOS increases more than 20% due to the packet loss rate which is reduced more than 90%. The comparison of MOS is shown in Fig.5.1(c).

### 5.3 Voice quality with different value of two HARQ parameters

In this section, we show the impact of HARQ to the quality of Skype under different WiMAX HARQ parameters,  $D$  and  $R_x$  in Table 4.1. The value of each parameter should not produce large delay which degrades the voice quality tremendously. To control the impact of this two parameters on voice quality, we set the maximum delay to 177.3ms to prevent the dramatic quality decline(14). The maximum delay causes by using HARQ is expressed as

$$T_f \cdot (1 + R_x \cdot D) \leq 177.3ms. \quad (5.2)$$



Table 5.2 HARQ parameters

Maximum retransmission( $R_x$ )	HARQ_ACK_DELAY( $D$ )
1	1,2,3,4,5,6,7
2	1,2,3
3	1,2

Table. 5.2 shows the values of the parameters of WiMAX HARQ we use in our analysis and simulation.

In this research, we found WiMAX HARQ can mitigate the loss that the Skype's redundancy mechanism cannot recover under bursty noise. Because in the consecutive bad frames, WiMAX HARQ's retransmission has lower probability to retransmit the packet in the frame with bad status, which could avoid the consecutive packet loss.

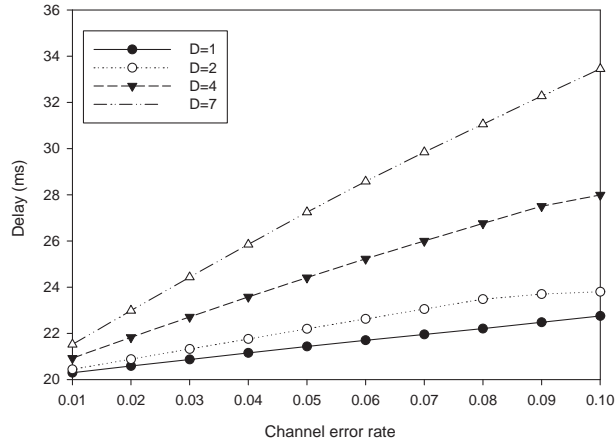
### 5.3.1 Fixed value of $R_x$

In the same value of  $R_x$ , the longer  $D$  has better ability to resist the bursty noise. It is because the longer  $D$ , the probability of losing two consecutive retransmission packet is lower.

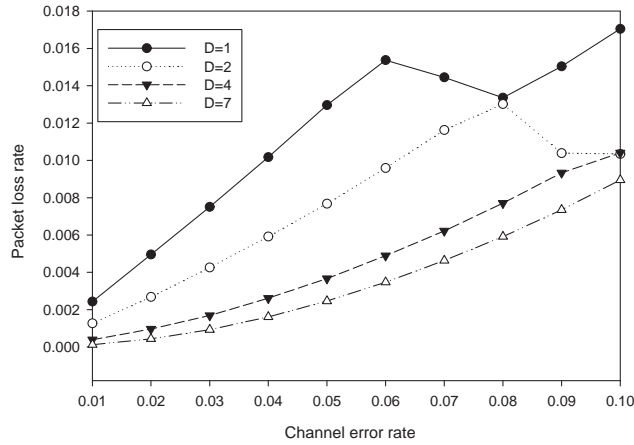
Fig. 5.3(a) shows the delay is higher when  $D$  is higher. The reason is that the larger  $D$ , the interval between retransmission is longer, which can be derived from our model 4.9, 4.10, and 4.11.

Fig. 5.3(b) shows the higher  $D$  has lower loss rate. We observe that the packet loss rate in certain channel error rate is decreased with higher channel error, including  $D = 1$  and  $D = 2$ . The reason is that the shorter  $D$ , the error rate of two consecutive transmissions (the first attempt and the second attempt) is high enough to trigger the Skype's redundancy mechanism which raises the redundancy ratio to higher level.

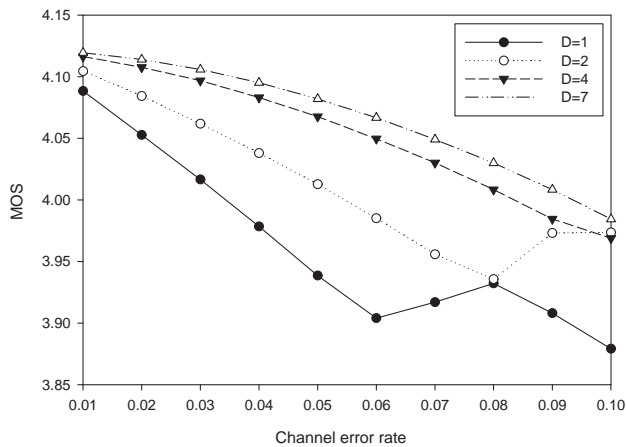
In Fig. 5.3(c), MOS is higher while  $D$  is higher. Although the delay is higher with larger  $D$ , the effect of packet loss on MOS is higher than the effect of delay, which makes MOS still higher with higher  $D$ .



(a) Delay versus channel rate

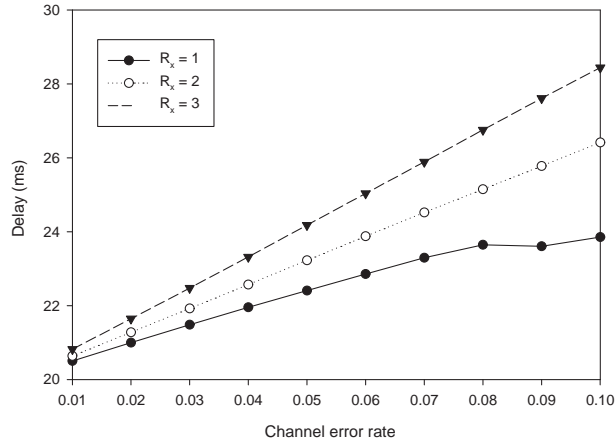


(b) Packet loss rate versus channel error rate

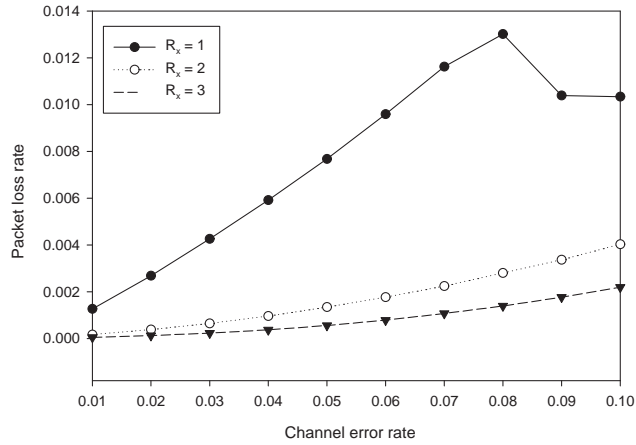


(c) MOS versus channel error rate

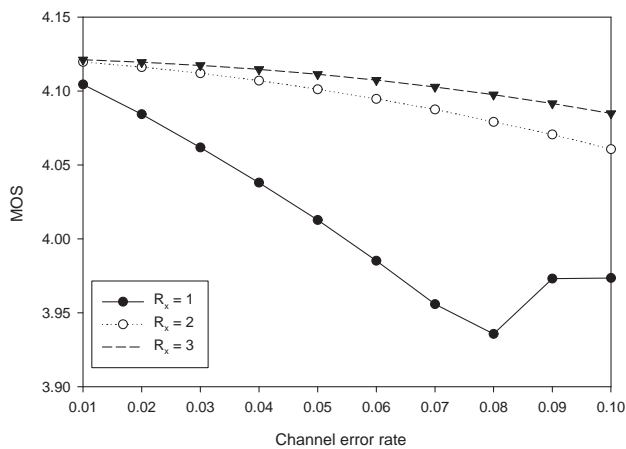
Figure 5.3 Comparison of effective loss rate and redundancy ratio when maximum retransmission is 1 and burst ratio is 2 with the value of HARQ\_Ack\_Delay 1, 2, 4, and 7



(a) Delay versus channel error rate



(b) Packet loss rate versus channel error rate



(c) MOS versus channel error rate

Figure 5.4 Comparison of delay, packet loss rate and MOS with HARQ\_ACK\_DELAY is 2 and maximum retransmission from 1 to 3

### 5.3.2 Fixed value of $D$

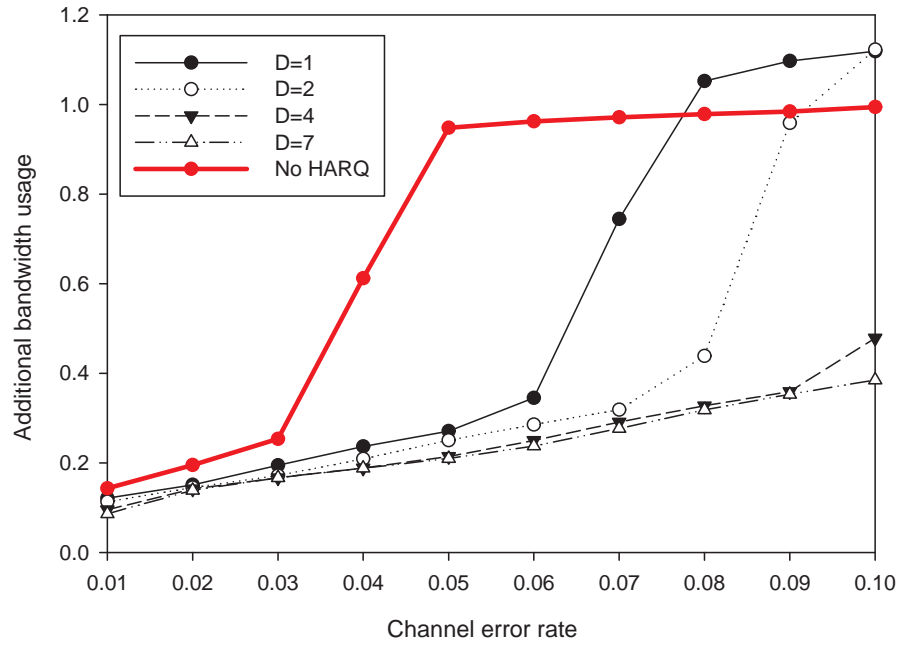
Here we consider the case of fixed  $D$  with different value of  $R_x$ . The simulation result is shown in Fig.5.4. The higher  $R_x$  allows more attempts for a HARQ packet retransmission. It results in the higher  $R_x$  yielding the lower packet loss rate and the higher packet delay. We also observe that the MOS is greater with the increase in the system loss rate. Despite the delay is increased with higher value of  $R_x$ , the MOS is still increased. The reason is same as the reason discuss in the section of fixed  $D$ .

## 5.4 Bandwidth usage of HARQ

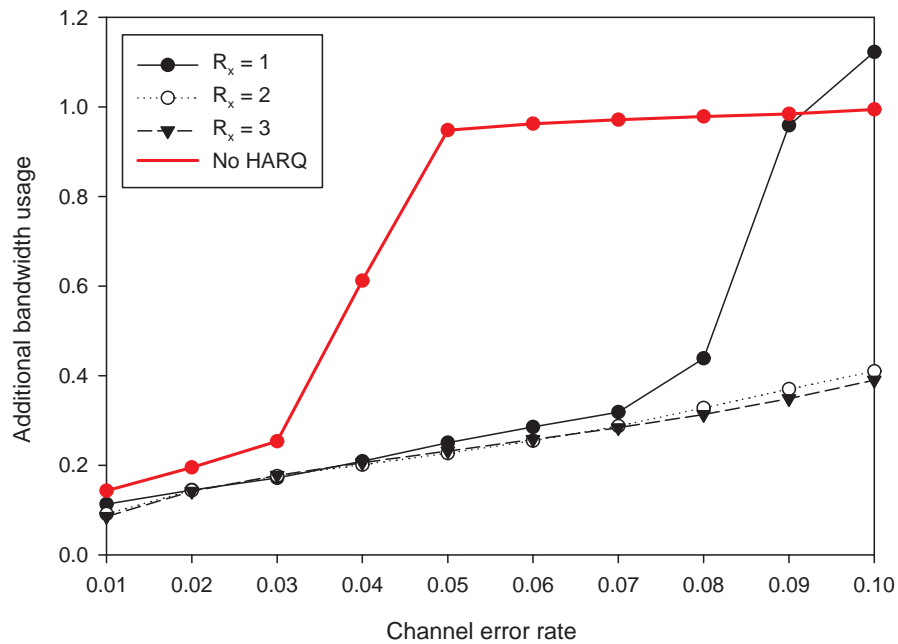
Although HARQ mitigates the loss impairment, it consumes more bandwidth on transmissions. The redundancy produced by HARQ retransmission and Skype's redundancy mechanism is shown in Fig.5.5 under noise burst ratio 2. The redundancy in the graph represents the ratio of total duplicate packet per original packet. The original packet indicates the packet generated by Skype without applying any redundancy mechanism, and the duplicate packet indicates the piggybacked packet generated by Skype's redundancy mechanism and the packet generated by the retransmission of WiMAX HARQ. We observe that except the  $D$  is 1 and 2 in Fig.5.5(a) and  $R_x$  is 1 in the Fig.5.5(b), the overall redundancy ratio is lower than the applying Skype's redundancy mechanism without HARQ. It is because the HARQ's retransmission reduces the impact of bursty noise which decreases the packet loss rate and leads the Skype lower its redundancy ratio. Thus, it is notable that under the environment with bursty noise, the HARQ not only improve the quality of voice but also decreases the bandwidth utility of Skype.

## 5.5 Optimal HARQ parameters

There is a trade-off between voice quality and bandwidth usage. We study the voice quality and bandwidth usage under burst ratio 2 and 10% channel error rate and the result is shown in the Table 5.3. The value of HARQ parameters, Maximum retransmission and HARQ\_ACK\_DELAY, we tested are listed in the first two columns from the right in Table 5.3. B/Q is the bandwidth usage over voice quality (MOS) ratio.



(a) Maximum retransmission = 1



(b) HARQ\_ACK\_DELAY = 2

Figure 5.5 Additional bandwidth usage produced by HARQ and Skype's redundancy mechanism versus packet loss rate

Table 5.3 MOS and Bandwidth usage comparison

$D$	$R_x$	MOS	Bandwidth Usage	B/Q
1	1	3.879217	1.118893	3.467013
1	1	4.042374	1.122765	3.600374
1	3	4.075768	0.433489	9.402240
1	4	4.103960	0.400904	10.236764
2	1	3.973558	1.122601	3.539599
2	2	4.060694	0.410016	9.903745
2	3	4.084810	0.390113	10.470837
3	1	3.995783	0.934826	4.274360
3	2	4.085024	0.397171	10.285302
4	1	3.968849	0.478620	8.292275
5	1	3.976321	0.403855	9.845912
6	1	3.981675	0.388049	10.260753
7	1	3.984368	0.385025	10.348335

Here we found the B/Q is the highest when  $D$  is 2 and  $R_x$  is 3. Which means that by choosing this set of parameters, the voice quality is better and the bandwidth usage is lower than most of other sets of parameters.

## CHAPTER 6. CONCLUSION

The voice quality is crucial to the success of VoIP service. Skype, one of the popular VoIP applications, is also widely used in wireless networks. Skype employs a redundancy mechanism to improve the voice quality through sending redundant packets. This consumes additional bandwidth which is precious resource in a wireless network. In this thesis, we study the voice quality and bandwidth usage of Skype over WiMAX network which is considered as one of the 4G wireless network technologies. We perform a detailed study on the voice quality and the HARQ retransmission mechanism used in WiMAX. The key parameters of HARQ mechanism are investigated. We argue that the HARQ retransmission of WiMAX and the redundancy mechanism of Skype should work together to achieve the best voice quality over bandwidth usage ratio.

In this thesis, the Skype's redundancy mechanism and the WiMAX's HARQ retransmission have been fully analyzed. The Skype's redundancy mechanism does not work well under the environment with bursty noise, which is happened frequently in wireless network. First, we show the functionality of Skype's redundancy mechanism is limited when the channel suffers bursty noise, which degrades voice quality more than 20% . Then we analyze the voice quality of the Skype with HARQ enabled. We found that the voice quality is improved significantly by enabling HARQ. This is because the retransmission of HARQ can reduce the probability of consecutive loss in wireless network where the packet loss rate tends to be high.

We present analytical model to compute the delay, loss rate, and voice quality (i.e. MOS). The models are validated through simulations, and there is only 1% difference in most of our simulations. Our research shows that by comparing to the environment with disabled HARQ, enabled HARQ on WiWAX could improve the voice quality of Skype up to 22%. The bandwidth usage could be reduced by 76% because the packet loss rate has been decreased.

It also attributes the lower bandwidth usage to the lower redundancy ratio used by Skype when the packet loss rate is improved through the HARQ retransmission. From our simulation results, we can obtain the optimal HARQ parameters which can lead to the best voice quality over bandwidth usage ratio when Skype is running over WiMAX. Therefore, our research not only shows the bursty noise problem can be eased by enabling HARQ, but also finds the optimal HARQ parameters that can increase the system throughput of WiMAX and maintain the high voice quality for Skype application.



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