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Performance of Voice over IP in Mobile Ad Hoc Networks

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Performance of Voice over IP in Mobile Ad Hoc Networks

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Abstract

The objectives of this thesis are to study the performance of voice over IP in ad-hoc networks, and to study the relationship among routing protocols, coding schemes and the quality of voice. The contribution is an adaptive rate control scheme which uses Mean Opinion Score (MOS) as criteria to improve the performance of the mobile ad hoc network. We first discuss and analyze the characteristics of wireless ad hoc network and Adaptive Multirate-Wideband (AMW) speech coding, and then provide the operation of the adaptive source-network rate control scheme consisting of the voice coding and transmission control, adaptive network rate control, the source rate control, and the frame combination. We implement the scheme in a test-bed and test the performance of speech in different scenarios using Multi-path Source Routing (MSR). Finally, we verify our testbed measurements with OPNET simulation.

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Table of Acronyms and Abbreviations

		Section of 1st Appearance
3GPP	The third Generation Partnership Project	2.5
ACK	Acknowledgement code	1.1
ACELP	Algebraic-Code-Excited Linear Prediction	2.5
ACU	Aironet Client Utility	A.1
AMR-WB	Adaptive Multi-Rate WideBand	1.2
AODV	Ad hoc On command Distance Vector Routing	1.1
ARQ	Automatic Repeat-request	1.2
ASNC	Adaptive Source-Network rate Control scheme	1.6
CBR	Constant Bit Rate	5.1
CCK	Complementary Code Keying	2.1
CELP	Code-Excited Linear Prediction	2.5
CRC	Cyclic Redundancy Check	3.0
CSMA/CA	Carrier Sense Multiple Access/Collision Avoidance	1.1
DNS	Domain Name System	1.1
DCF	Distributed Coordination Function	2.1
DIFS	Distributed Inter-frame Space	2.1
DN	Directory Number	1.1
D-ITG	Distributed Internet Traffic Generator	3.1
DSDV	Destination Sequenced Distance Vector	1.2
DSR	Dynamic Source Routing	1.1
DSSS	Direct Sequence Spectrum Spread	1.1
DTX	Discontinuous Transmission	3.1
ENUM	Telephone Number Mapping	1.1
FEC	Forward Error Correction	1.2
FFT	Fast Fourier Transform	2.4
FHSS	Frequency Hop Spectrum Spread	1.1
GSM	Global System for Mobile	2.1
GUI	Graph User Interface	4.3
HDLC	High Data Link Control	2.1
IAI	Initial Address Information	1.1
ISM	Industry Scientific Medical Frequency	4.7
ISP	Immittance Spectrum Pairs	2.5
ITU	International Telecommunication Union	2.5
LAN	Local Area Network	1.1
LLC	Logic Link Control	1.1
LP	Linear Prediction	2.5
LPC	Linear Predictive Coding	2.5
LTP	Long Term Prediction	2.5
MA	Moving Average Prediction	2.5
MAC	Media Access Control	1.1

MANET	Mobile Ad hoc Networking	7.2
MDC	Multiple Description Coding	2.4
MGCP	Media Gateway Control Protocol	2.3
MIMO	Multiple Input Multiple Output	1.2
MOS	Mean Opinion Score	1.3
MPDU	MAC Protocol Data Unit	3.2
MSB	Most Significant Bit	2.5
MSR	Multiple path Source Routing	1.1
MTU	Maximum Transmission Unit	3.2
OSI	Open System Interconnection	1.1
PCF	Point Coordination Function	2.1
PCM	Pulse-Code Modulation	2.4
PDA	Personal Digital Assistant	1.1
PESQ	Performance Evaluation of Speech Quality	1.5
PLCP	Physical Layer Convergence Procedure	1.1
PMD	Physical Medium Dependent	1.1
PPDU	PLCP Protocol Data Unit	3.2
PSDU	Physical layer Service Data Unit	3.2
PSTN	Public Service Telephone Networks	2.3
RTP	Real Time Transportation Protocol	1.1
RTS/CTS	Request to Send / Clear To Send	2.1
RTT	Round Trip Time	4.5
SC	Scalable Speech Coding	2.4
SCR	Source Controlled Rate Operation	3.1
SID	Silence Insertion Descriptor	2.5
SIP	Session Initiation Protocols	2.3
SSID	Service Set Identifier	4.3
UDP	User Datagram Protocol	1.1
VAD	Voice Activity Detection	2.5
VoIP	Voice over IP	1.1
WCDMA	Wideband Code Division Multiple Access	2.4
Wi-Fi	Wireless Fidelity	1.1
WLAN	Wireless Local Area Network	1.1

Table of Notations and Symbols

Notation	Explaination	Section of 1^{st}Appearance
$A(z)$	The inverse filter with unquantized coefficients	2.5
$\hat{A}(z)$	The inverse filter with quantized coefficients	2.5
$a\hat{}i$	The quantified linear prediction parameters	2.5
d	The distance between transmitter and receiver	4.7
γ_1	The perceptual weighting factor	2.5
g_p	The adaptive codebook gain	2.5
G_t	Transmitting gain respectively	4.7
h_r	The antenna height of receiver. *d* denotes the distance	4.7
h_t	The antenna height of transmitter	4.7
m	The order of the LP model	2.5
n	The number of nodes	3.2
n_i	The number of ith node neighbors	3.2
$MOS(m, pl)$	The lookup MOS value	3.3
P_t	The transmitting power	4.7
R_e	The effective transmission rate	3.2
$RS(n,k)$	Reed-Solomon Coding	2.7
T	The integer pitch lag nearest to the closed-loop fractional pitch lag of the subframe	2.5
$W(z)$	The perceptual weighting filter (un-quantized coefficients)	2.5
Wi	The transmission rate	3.2

Chapter One
Introduction

1.1 Overview

The acme of personal communications is to implement the "anywhere, anytime, and any way" paradigm. A general objective is to use one handset to receive and send high quality voice and smooth video, and download data fast. Wireless LAN (Local Area Network) technology provides us an opportunity to realize this objective.

Wireless LAN technology is a promising technology in the near future because it provides not only voice service but also data service and it can satisfy next generation wireless networks with high speed and multi-service. The transition from a circuit switch to a packet switch becomes an irreversible trend. It is possible that the latter will substitute the former completely. WLAN (Wireless Local Area Network) also provides a challenge to cellular phones, just like VoIP (Voice over IP) competing with the traditional telephone, because they can provide good quality (wideband) at a low cost.

Wireless LAN uses IEEE 802.11b/g ratified in 1999/2003 [IEEE07] provides 11/54M data bit rate at work frequency of 2.4 GHz. WLAN stations can use access points to access the Internet. They can also connect each other to become a network without infrastructure. Called an ad hoc network, network stations work in peer-to-peer modes. The stations can work as servers, not just as clients used for sharing content files and real-time streams. The stations are usually laptop or PDA (Personal Digital Assistant) with a wireless adaptor. In an ad hoc network, a path should be found so that the packet is forward by intermediate nodes. Since wired routing protocols can not respond rapidly to change topology or link quality, the on-demand protocols are used to satisfy the requirement. The routing protocols include DSR [JoMa96], MSR [WaZh01] and AODV [PeRo99].

Compared with wired systems, however, wireless systems are prone to errors. They are contributed by factors in the physical layer, from interference and due to limited bandwidth. There is fast path-loss, shadow fading and multi-path fading in wireless channels. Therefore, we should use suitable source coding and channel coding, particularly a kind of control scheme against bit errors and congestion so as to make voice transportation more robust to the bit error or packet loss.

In order to implement the VoIP [Burg02] over wireless networks, we need both signaling and media protocols. The signaling protocol allows us to obtain the routing information (e.g. terminal IP address or Directory Number (DN)). The media protocol is to transport the voice packets by the given routing information. We also use ENUM (tElephone NUmber Mapping) that provides mapping from telephone number to Internet address to implement translation. For example, all devices (PC, IP phone, Wi-Fi phone, cell phone, or telephone) are first registered its presence status (available IP or DN) manually or automatically in the DNS (Domain Name System) database, which is similar to looking up the domain name. When we dial +1-613-562-5862, the IP address or directory number is searched by 5862.562.613.1 in the database. Then we get available device IP address or telephone number. Finally, we use IP address and/or IAI (Initial Address Information) by Gateway to get the destination.

Packetized voice packets are transported using the OSI model [ITUT02e]. Physical layers of WiFi consist of Physical Layer Convergence Procedure (PLCP) and Physical Medium Dependent (PMD) by direct sequence spectrum spread (DSSS) or frequency hop spectrum spread (FHSS) modulation. MAC (Media Access Control) layer uses CSMA/CA (carrier sense multiple access/collision avoidance) with ACK (Acknowledgement code), it also uses LLC (Logic Link Control) protocol to ensure reliable transmission. By encapsulating them with proper headers, voice packets are delivered by best effort when transported in IP by Network Layer and UDP (User Datagram Protocol) in Transport Layer. Application Layer uses RTP (Real Time Transportation Protocol) protocol to implement the reliable transportation. Thus, VoIP is implemented in wireless LAN.

1.2 Literature Review

In order to improve the performance of voice over IP in ad hoc networks, we shall review related papers of recent years. There are many reasons to cause the quality of voice to deteriorate such as packet loss, error, delay and jitters. First, there are many types of interference such as co-channel, adjacent channel and inter-symbol interference in wireless networks. In addition, the bandwidth of wireless network is limited. These factors lead to packet drops because of bit error and congestion. Packet loss and delay also is caused by collisions in the media access layer and congestion in the network layer. There are many methods proposed to solve those issues. Among these methods, congestion control schemes and speech/channel coding are two basic technologies.

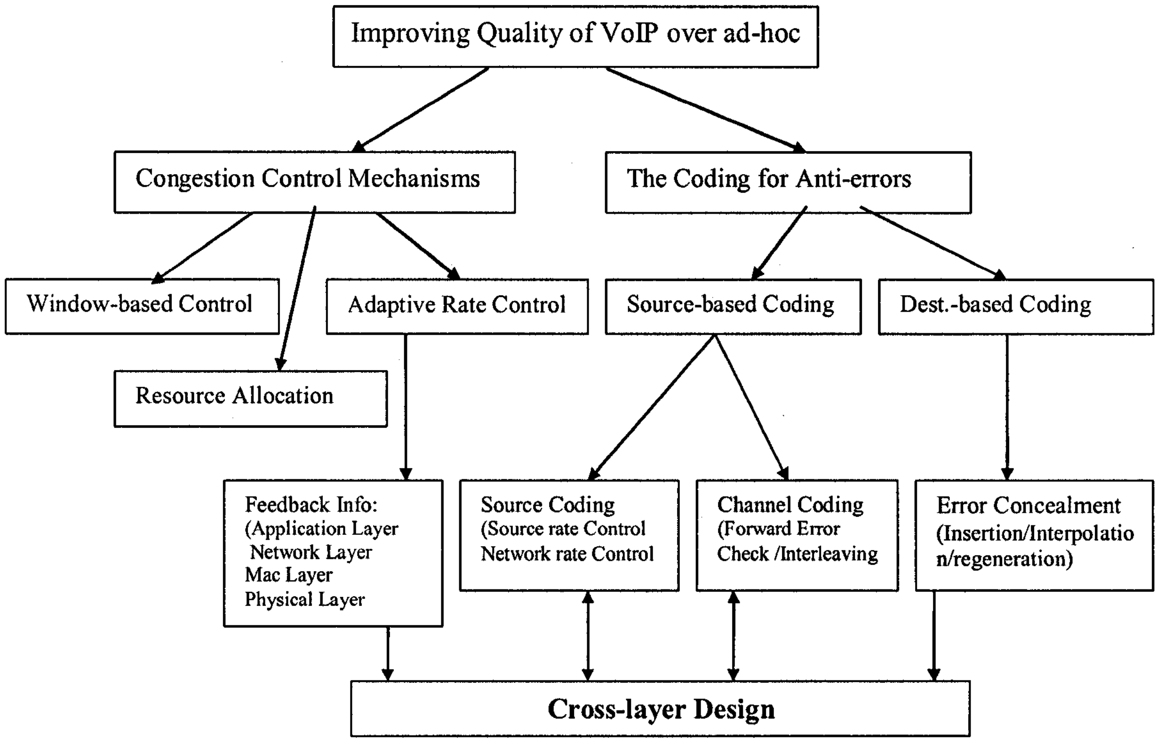


Figure 1.1: Literature Classification for Performance of VoIP over Ad Hoc

An adaptive rate control scheme can be used to adjust the bit rate according to channel capacity [ZhLo06]. We can also use source coding and channel coding to adapt coding bit rate and enhance the code robustness. For the destination, we can use the error concealment method to be resilient to errors. Many recent papers, e.g. [KoHo06] [ZhTa06] [KsNa06], have focused on cross layer control mechanism in applications of wireless ad hoc network, which combines adaptive control mechanisms and source and channel coding together. Their objective is to obtain a high quality of service of the ad hoc network. Figure 1.1 shows the classification of papers about improving the performance of voice over wireless ad hoc networks.

There are also different testbeds developed for the study of WLAN and ad hoc networks. Hardware-fitted Modeling [FeKa00] is used to analyze the behavior of the wireless VoIP by taking hardware measurements for a small number of hardware sources, then duplicated and extended in the OPNET simulation. A large-scale Testbed [LuLu02] is designed to assess several different routing protocols with up to 37 physical nodes. It introduces a virtual mobility by the measured signal quality instead of the geometric location variation so as to compare the performance of ad hoc routing protocols. MSR testbed [ZhYa05] is implemented to evaluate and compare its performance.

1.2.1 Speech Coding and VoIP

The speech coding standard Adaptive Multi-Rate Wideband (AMR-WB) has recently been discussed widely because it can provide high speech quality with low coding bit rates but robust to bit loss. AMR-WB makes communication more natural, reduces listening effort and eases speaker recognition. Therefore, AMR-WB is a good candidate for cross-layer architecture. There are many papers to discuss the AMR-WB speech coding. For example, its good quality over narrowband speech [OjLa06], effect of codec characteristics on the performance of wideband speech [VaDe06], Wideband coding architecture, applications and the distinctive operational features [AhJe06] [ShGr06].

There are also many papers discussing how to improve the performance of VoIP. For example, a voice multiplex-multicast (M-M) scheme for overcoming the large overhead effect of VoIP over WLAN is used to reduce the overhead of the voice packet [WaLi05]. E-model is used to study the relationship between resource utilization in the wireless LAN and the quality of VoIP calls transmitted over the wireless medium [NaDa06]. The capacity of an WLAN network carrying voice calls is evaluated in a wide range of scenarios including delay constraints, channel conditions and voice quality requirements [HoTo04]. However, there is no proposal to improve voice quality in ad hoc networks.

1.2.2 Packet Error Recovery

There are also many error resilient techniques which have received significant attention in recent years and many error mitigation techniques which have been investigated. Among these techniques, forward error correction and automatic repeat-request (ARQ) are two basic error control mechanisms. ARQ is a closed loop error control technique by retransmitting corrupted data frame. Since this also introduces additional delay, it is not acceptable for real-time conventional services. Forward Error Check (FEC) is generally used instead to combat packet bit error [QuPe05]. It is traditionally used for real time multimedia traffic since it requires no feedback and the delay can be bounded. Forward error correction codes are usually employed to reduce packet errors at the expense of the associated increase in bit payload. Error concealment techniques [PeHo98] have been used in decoders to extrapolate information from received bits in order to reconstruct the lost information. For burst errors, frame interleaving techniques are used to reduce bit errors at the cost of confusing frame order, leading to longer delay. One can also use channel coding (e.g. Reed-Solomon coding) to protect channel errors [PaMo99]. All these papers focus on the characteristics and performance of AMR-WB coding, and the error recovery methods.

1.2.3 Cross-Layer Design

Cross layer design is an attractive method used recently to improve the performance of wireless networks. It is an emerging research field which involves several areas, such as adaptive coding, channel and traffic modeling. A cross-layer design has to conduct the functionalities in several layers such as the physical layer, the data link layer, the network layer and application layer. There are many papers to discuss this issue recently. For example, a cross-layer architecture optimizing the cross-layer feedback signal to enhance the performance of existing protocol stacks [KoHo06], a physical model dealing with MIMO diversity and adaptive modulation coding [ZhTa06], a cross-layer architecture for efficiently transmitting H.264 video [HaTa06][KsNa06], application-driven cross-layer design and optimization [KhPe06] and delay and distortion models [QuPe05] to choose FEC and paths, a cross-layer control mechanism [ZhLo06] by collecting routing information and frame loss information. All these papers use a synthesis method to compensate wireless network weakness and then improve the quality of wireless networks.

1.2.4 Congestion Control Mechanism

There are many works that explore the technology against packet loss of wireless networks. Among these technologies, resource allocation [Mase04], adaptive rate control [KrMo04] [FrFr85], window-based control are three basic congestion control mechanisms that are widely used to combat transmission loss caused by congestion shown in Fig 1.1.

Resource allocation is routed-based control and uses scheduling the use of bandwidth resources or uses virtual circuits to guarantee bandwidth. This technique can eliminate network congestion by blocking traffic that is in excess of the network capacity such as admission control [Mase04].

Window-based control sets a source a maximum number of credits (i.e., the window size) to transmit its packets. The source has to withhold its transmission until acknowledgments are returned [Stal03] when the credits are exhausted. Window-based control methods can be divided into source-based and router-based.

Source-based control attempts to control the packet transmission rate in each source based on the current network congestion situation. Using an end-to-end window-based protocol based on the original TCP/IP, the window size is halved whenever a packet is dropped in an overflowed router buffer. This scheme is called the AIMD (Additive-Increase Multiplicative-Decrease) [Jaco88] which is very effective in the conventional IP networks where the propagation delays are short compared to the source transmission rates. However it is unreasonable to exclusively rely on sources to perform end-to-end congestion control.

The AQM (Active Queue Management) which is router-based control uses congestion avoidance to participate in the network traffic control. Random-drop rather than the drop-tail method is proposed for managing the queue length in an outbound buffer. Packets are dropped whenever congestion is anticipated rather than waiting until the queue is full. The router adjusts packet drop probability directly to regulate the source transmission rate indirectly. ERD (Early Random Drop) [Hash89] is a congestion recovery mechanism proposed. When the queue length exceeds a threshold, packets are dropped with a fixed probability.

Rate-based control (also called adaptive rate control) allows an end system to adjust its sending rate for best-effort service traffic so as to make the maximum use of network resources. Thus, it is the most effective and desirable solution under a network environment with long round trip delays and dynamical changes in available bandwidth to achieve a high QoS (Quality of Service) for streaming media transmission in the Internet [HuXu03]. This scheme reduces the bit rate of the transmitted stream signal when the channel is anticipated to be bad. A scalable and adaptive source-channel rate control scheme [KrMo04] are used to optimize the level of adaptation to reduce the bandwidth requirement while guaranteeing delay and loss bounds.

Rate-based control can also be classified as either router-based or source-based. The router-based control also uses AQM to implement network traffic control by the router participation. Basically, by calculating the advertised source transmission rate directly, the router converts it into the source window size to regulate the source transmission rate "directly" [GeLo02, HoYa05b]. A rate-based router control scheme was presented for transparently augmenting the end-to-end TCP performance by controlling the sending rate of a source in [KaKa00].

For source-based rate-based control, some methods have been proposed recently for AQM in IP router supporting best effort service traffic (e.g., streaming media such as audio and video over IP). For example, TFRC (TCP-Friendly Rate Control) proposal was presented for equation-based congestion control for unicast traffic in [FlHa00]. RAP (Rate Adaptation Protocol) employed an improved AIMD source control scheme for real-time stream transmission [ReHa99].

The utility-based control [KeMa98, HoYa05a] is a variation of the rate-based source control method. The source adjusts its transmission rate based on the feedback network congestion information and its own utility function, which is called utility-based control. Utility function is applied in the controller design in order to make the optimal use of the network resources [KeMa98]. A framework model is used to analyze the stability and fairness of the rate control algorithm, and a primal algorithm is provided to adjust the users' sending rates to converge at an equilibrium state. An optimal solution to the resource allocation problem is then obtained under the assumption that there are no communication delays between the end users.

In addition, a proportional rate control for the high-speed data networks [ZhYW05] is developed to stabilize the network, overcome the oscillation problem and achieve the fairness for varying round-trip delays and packet loss. A proportional-integrative (PI) congestion controller for best-effort traffic in the Internet [HoYa03] is designed to regulate the source rate to make the steady value of queue length exactly equal to the specified threshold value.

1.2.5 Ad Hoc Routing Protocol

There are many papers on the suitable routing mechanisms of voice stream over ad hoc networks. These mechanisms include Destination Sequenced Distance Vector (DSDV), Ad hoc On-demand Distance Vector (AODV), Dynamic Source Routing (DSR), and Multipath

Source Routing (MSR). Basically, they can be classified into two categories: Table-driven and on-demand.

DSDV is a table driven routing protocol [PeBh94]. DSR models the mobile computers as routers cooperating to forward packets to each other as needed. Each entry in the routing table is marked with a sequence number assigned by the destination. Routing table updates are transmitted throughout the entire network periodically. The router has to issue broadcasts to announce every change in the overall connectivity of the ad hoc network. Therefore, local movements have global effects.

AODV is on-demand routing protocol [PeRo99] that is improved from DSDV. With AODV, a router localizes to effects caused by local movements. AODV distinguishes itself from DSDV for having no broadcast. In AODV, all routing information is added in packet header so intermediate nodes do not need a routing table. In order to find a route, the source node floods the whole network with a route request broadcast packet. Only the destination nodes can reply by sending back a route reply to the source along a reverse path, thus the path establishing.

DSR (Dynamic Source Routing) [JoMa96] is also a popular on-demand routing protocol. DSR is better than AODV in light traffic and low mobility. DSR is a source routing protocol and inserts the routing information in the packet header. DSR can be introduced as a single and efficient ad hoc routing protocol to allow the network to be completely self-organizing and self-configuring without the need of careful design and administration.

MSR (Multi-path Source Routing) [WaZh01] is extended from DSR to distribute the load into the multiple paths which can alleviate the congestion problem by load balance. Multiple path routing also has high network stability than single path since the other links will continue to work even though one of links breaks.

In summary, on-demand routing protocols are better than table-driven routing protocols because they do not need to maintain the routing table containing all the route information to the other nodes frequently. When a node requires a route to a destination, it initiates the routing discovery process to find routes to the destination. Thus, on-demand routing protocol will be used in ad hoc networks because it utilize network resources more efficient therefore better performance than table driven ones.

1.3 Motivations

As reviewed above, although many different methods have been offered to improve the quality of stream transmissions, there has not been an efficient control scheme to combat packet loss or error in ad hoc networks. So we would like to study the wireless channel and speech coding to propose an adaptive rate control scheme in which MOS (Mean Opinion Score) is used as evaluation criteria. As we understand, these methods have not been used in voice transport over ad hoc networks. Therefore, we would like to use these methods of improving voice transmission over ad hoc networks.

Secondly, as summarized in Section 1.1 earlier, VoIP over ad hoc networks is a promising technology and there are many issues needed to be researched and solved. For example, approaches to improve the quality of voice over ad hoc networks are a good research topic. Furthermore, the ad hoc technology is a good complementary of next generation communication technology. Ad hoc's peer-to-peer is a good conception and has already been implemented in a wire-line Internet environment. Voice over ad hoc networks will become an important ubiquitous (anywhere and anytime) application and will promote the development of ad hoc technology. Therefore, one would like to study the voice performance of the ad hoc network in outdoor and indoor settings so as to obtain whole understanding.

Thirdly, previous graduates use simple applications such as FTP and PING over ad hoc networks [ZhYW05][Wang04]. They evaluated the performance of DSR, MSR and BSR routing software provided by Tianjin University. They also reported and debugged the software bugs because the software is an immature version. However, they did not deal with the speech application and improvement of speech. In this thesis, we would like to make use of the routing software to implement voice over IP applications over ad hoc networks and also propose an adaptive control scheme to improve the quality of voice over ad hoc networks.

1.4 Objectives

Our general objective is to develop a control model for voice performance analysis of ad hoc networks in different traffic interference. Specifically, we are interested in the following:

1. To propose an adaptive source-network rate control scheme that can improve the quality

of voice.

2. To implement this adaptive source-network rate control scheme on a testbed.
3. To investigate the performance of VoIP under different traffic and scenarios in wireless ad-hoc networks.
4. To implement an OPNET simulator for the adaptive source-network rate control scheme.
5. To use the testbed measurement results to validate the simulation.

1.5 Methodology

In order to improve the quality of voice over an ad hoc network, we will propose an adaptive source-network rate control scheme for mobile ad hoc networks and measure the voice quality under different traffic parameters.

We first thoroughly study and analyze the characteristics of wireless channel and speech coding. We create an ad hoc queuing network model to analyze the throughput of the wireless ad hoc network. We also create a speech coding model for voice over ad hoc networks. Under these models, we implement the adaptive rate control schemes to improve the performance of speech transmission over ad hoc networks.

We have taken several measures to implement the adaptive control scheme by means of adjusting the packet size and the packet transmission rate to make maximum use of network resources and also to avoid congestion. We shall use network status information (including packet loss rate and routing information) to control speech coding rate. We plan to utilize Voice Activity Detection (VAD) to decrease the coding rate further. Furthermore, the frame combination method can be used to save overhead by merging two frames together to reduce overhead of packet header.

To verify all schemes above, we setup an ad hoc testbed consisting of several laptops, and their wireless cards. We shall implement/code the VoIP and the control scheme on a Linux platform. Speech codec AMR-WB (Adaptive Multi-Rate WideBand) will be used because it can provide a variable coding rate to suit the adaptive rate control scheme with a high quality and lower bit rate. We shall add PESQ (Performance Evaluation of Speech Quality) code to evaluate the quality of speech. We have borrowed the code from Tianjin University to implement the routing protocols DSR and MSR. We shall conduct our experiment for each specifically-chosen topology, both indoor and outdoor as well as both mobile and static environments. All these different environments would provide us with different propagation and interference conditions for our testing.

We evaluate the performance of networks in our testbed using throughput, packet loss rate, delay, jitter and MOS. We use VoIP program to generate voice stream and use packet generator D-ITG (Distributed Internet Traffic Generator) [AGEP04] to generate traffic packets. Delay, jitter and throughput are measured by PING and FTP. The MOS is measured by the PESQ embedded in VoIP program. We evaluate the performance of CBR (constant bit rate) and then evaluate and compare the performance of ASNC with that of CBR to verify the effectiveness of our adaptive rate control scheme ASNC.

Besides testbed measurements, we also conduct simulation to study the VoIP with ASNC further. Simulation is more time-saving and energy-saving than practical measurements. It can be used to simulate the complexity and uncertainty desired so that performance evaluation can be easily done via analysis of simulation data. Simulation also can test complex topology without the physical limitations of testbed. Out of the many simulation software (such as NS2, GloMoSim, Quarnet and OPNET), we choose OPNET as our simulation tool because it has many ad hoc models in its library and has a friendly graphical user interface environment for network modeling and simulation. Also its module structure and visual interface allow us to design scenarios and models more easily.

Although testbed measurements of VoIP performance are more time-consuming than simulation, they can get more useful results. The reason is that testbed can reflect real physical layer of radio communication and interlayer communication of protocols since physical layer laws are too complex for investigation by simulation. Furthermore, testbed can also be used to verify the correctness of software simulation.

1.6 Contributions

The contributions of this thesis are:

1) The ASNC (Adaptive Source-Network rate Control) scheme to transport VoIP over wireless ad hoc network: This scheme combines network controlled coding rate, source controlled rate operation, and frame combination and Cyclic Redundancy Check together. The coding rate is optimized to adapt the network bandwidth while guaranteeing delay and loss bounds so as to ensure the quality of speech.
2) Testbed implementation and testing of the ASNC scheme: Using laptop computers, we implement socket programming, AWR-WB speech coding, packet loss/error detection

and concealment, sound card programming, and performance evaluation of speech quality (PESQ). We performed testing, debugging and measurements in different outdoor and indoor scenarios.

3) An OPNET Simulator: The ASNC has been implemented in OPNET and verified by testbed measurements. This would support any extensive network scenarios in future.

1.7 Thesis Organization

The remainder of the thesis is organized as follows. Chapter 2 provides the preliminaries involved in the thesis. Chapter 3 proposes the adaptive source-network rate control scheme for mobile ad hoc networks. Chapter 4 describes the testbed setup including hardware and software development details. Chapters 5 and 6 evaluate respectively the performance of ad hoc networks on the static and mobile cases. Chapter 7 provides the network models, operations and definitions. Chapter 8 simulates the performance of ad hoc networks on static and mobile cases using OPNET. Chapter 9 concludes this thesis and proposes some future work.

Chapter Two
Backgrounds

In this chapter, we will provide some background required for this research. In order to explore the improvement of voice over IP over wireless ad hoc networks, there are three aspects of knowledge related. The first one is the wireless technology and the ad hoc routing protocols to provide a platform. The second aspect is the principle of voice over IP and digital speech coding AMR-WB. The third aspect is packet loss recovery and the quality evaluation by using PESQ.

2.1 The Wireless LAN (Wi-Fi) Operation

Wi-Fi (Wireless Fidelity), a modern wireless access technology, can provide convenient, high speed access to the networks that does not need cable connection. The service areas may range from a single room to an entire campus. There are two Wi-Fi standards used widely now [IEEE07]. IEEE 802.11b operates at 2.4 GHz with a data rate of 11Mbps and throughput of 6.5 Mbps. It uses Complementary Code Keying (CCK) modulation technique that is an extension to single carrier DSSS radio technology. IEEE 802.11g uses multi-carrier OFDM (Orthogonal Frequency Division Multiplexing) radio technology to provide higher bit rate of 55 Mbps. It also operates at 2.4 GHz with a data rate of 54 Mbps and throughput of 20 Mbps that is compatible with 802.11b Wi-Fi. They both use Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) as the medium access method. We discuss the Wi-Fi operation in detail as following.

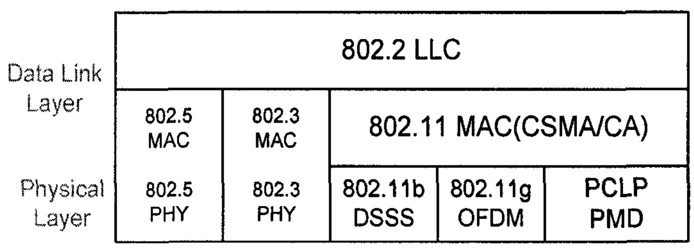


Figure 2.1: IEEE 802 Relationship to the OSI Model

2.1.1 IEEE 802.11b Layers

Wi-Fi protocol covers both MAC and the physical layer shown in Fig. 2.1. Physical layer consists of Frequency Hopping Spread Spectrum (FHSS) and Direct Sequence Spread Spectrum (DSSS). They both use the Code Division Modulation Access technology in which one bit of data is transferred to N bit spread code. In FHSS, the spread code is used to control a frequency agile local oscillator. In DSSS, the spread code is transmitted using different modulation schemes such as BPSK and QPSK. The physical layer is split into two components: the Physical Layer Convergence Procedure (PLCP) and a Physical Medium Dependent (PMD) system to map the MAC frames to medium. The coding uses Complementary Code Keying (CCK) instead of a Barker Code for high bit rate transmission.

The MAC layer provides two access methods: the Distributed Coordination Function (DCF) and Point Coordination Function (PCF) using scheduling. The 802.11 MAC works with a single first-in-first-out transmission queue. The CSMA/CA of DCF works as follows [WaLi05]: When a packet arrives at the head of a transmission queue and if the channel is busy, the MAC waits until the medium becomes idle, the deferring for an extra time interval, called the DCF Inter-frame Space (DIFS). If the channel stays idle during the DIFS deference, the MAC then start the back-off process by selecting a random back-off counter. For each idle slot time interval, the back-off counter is decremented. When the counter reaches zero, the packet is transmitted. The timing of DCF channel access is illustrated in Fig 2.2.

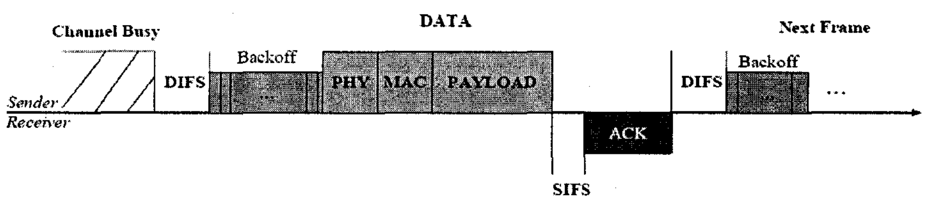


Figure 2.2: Inter-frame Space Relationship

Each station maintains a contention window, which is used to select the random back-off counter. The back-off counter is determined as a random integer drawn from a uniform distribution over the interval. If the channel becomes busy during a back-off process, the back-off is suspended. When the channel becomes idle again, it stays idle for an extra FIFs back-off counter value. For each successful reception of a packet, the receiving station immediately acknowledges the source by sending an acknowledgement packet. The ACK packet is transmitted after a Short IFS, which is shorter than the DIFS. If an ACK packet is not received after the data transmission, the packet is retransmitted after another random back-off.

The upper sublayer is the Logical Link Control Sublayer 802.2 which provides a means to exchange frames between different MAC protocols. It also provides some services of HDLC at data link layer. In this sublayer, there is an acknowledge signal. The relationship of the layer is shown in Fig. 2.1.

2.1.2 Management Operation

This section will introduce how the Wireless LAN operates. There are three types of WiFi Frames: Data Frames, Control Frames and Management Frames. Data Frame is used to encapsulate high layer packets for hauling data. Control Frame is used for medium access such as ACK, RTS/CTS and PowerSavePoll. Management Frame is used for exchanging management information to establish a connection with a network. The frames include Beacon to announce network existing, Probe Request/Response, Association Request/Response, Authentication, Re-association and Disassociation.

The procedure a station uses to access the WLAN networks is the following [WaLi05]: First, the station finds the network by scanning Beacon sent by the AP. It also can send Probe request to find the network. After that, the station joins the network by checking power levels, synchronization, physical channel, modulation type, etc. And then authentication is provided by physical access for security reasons. If the authentication is successful, the station will associate with an access point to gain full access to the network. Thus, the application data can be transported by the data frame. In the meantime, the station can use power management to save energy.

2.1.3 Ad Hoc Networks

Based on above Wi-Fi technology, Wi-Fi stations not only can be inter-connected via access points to form an infrastructure network, but also can communicate in peer-to-peer to form an ad hoc network. The access point is similar to a wire-line Ethernet hub to concentrate each station data stream to wire-line networks for the Internet access.

An ad hoc network enables devices to connect directly with each other without infrastructure base stations. The ad hoc network has no central infrastructure and also can form a wireless mesh network. Ad hoc assumption is that all nodes wishing to communicate with other nodes within the ad hoc network are willing to participate fully in the network protocols and each participating node should be willing to forward packets for other nodes. In an ad hoc network, each node functions as both a host and a router, and the routing of the network is distributed among the nodes. In the mean time, the network topology is dynamic because the connectivity among the nodes may vary with time due to nodes' movements. If a station can reach the range of another station, intermediate nodes may transfer its data as a relay. This connectivity mode is useful in consumer electronics and gaming applications. It can be used in a temporary network for example, in an academic meeting and field work group. It also extends the range of wireless networks.

Therefore, it is necessary to have efficient routing protocols to allow the nodes to communicate over multi-hop paths consisting of possibly several links. These protocols can use network resource efficiently.

2.2 Ad Hoc Routing

The routing is a critical issue of ad hoc networks. Wireless ad hoc networks consist of autonomous mobile nodes that spontaneously spawn a communication network between each other. The nodes are usually laptop or PDA with wireless interface and have enough computing power for ad hoc routing. Because of these nodes' mobility, the link status is never stable. Wired routing protocols can not fit to rapidly changing topology or link quality and thus cannot be used efficiently over such a network. Therefore, several kinds of ad hoc routing protocols are developed to fit the specific applications.

Routing protocols used in conventional wired networks, such as link state and distance vectors, are not suitable for the mobile environment due to the considerable overhead produced by periodic route update messages and the slow convergence to topological changes. Moreover, wireless links have limited bandwidth but most of the current Internet routing protocols rely on single path routing algorithms which often underutilize resources. However, on-demand routing protocols do not need to maintain the routing table containing all the route information to the other nodes frequently. When a node requires a route to a destination, it initiates the routing discovery process to find routes to the destination. Thus, on-demand routing protocol is more efficient to utilize network resources than table driven ones.

There are several kinds of on-demand routing protocols. Dynamic Source Routing (DSR) is a basic routing protocol [JoMa96], where the source finds a single path to transmit packets. It adds route information in the header of the packet so that the intermediate nodes do not need to maintain the routing table and just forward the packet according to the route information in the packet header; MSR is an extension of DSR. Multi-path Source Routing (MSR) is finding the multiple paths and load balance according to path delay constraints. When the one link breaks, the other path will continue the connectivity for source and destination. Fig. 2.3 shows the format of the on-demand packet.

IP Header	DSR/MSR/BSR Fixed-Portion Header	DSR/MSR/BSR Options	Transport Layer data

Figure 2.3: Ad Hoc Packet for On-demand

2.2.1 Dynamic Source Routing [JoMa96]

DSR implements source routing instead of hop-by-hop packet routing. Each data packet carries the complete path information from source to destination. The major advantage of DSR is source routing, where intermediate nodes are not required to keep route information because the path is explicitly specified in packet header. This allows loop-free packet routing and its on-demand features eliminate the need for periodic routing information updates and neighbor detection in the intermediate nodes through which packets are forwarded. It will decrease the computation and use of resources. DSR operates entirely on-demand, and routing packet overhead is minimized to react to changes in the current routes.

The DSR protocol consists of two mechanisms: Route Discovery and Route Maintenances. There are several kinds of DSR packets including path request packet, path reply packet, path error packet, acknowledgement packet, route probing packet, probing reply packet, and data packet to implement these mechanisms.

Route discovery is initiated by a source whenever a source has a data packet to send but does not have any routing information to the destination. To establish a route, the source floods the network with request packets carrying a unique request ID. Intermediate nodes which have no routing information to destination will rebroadcast path request packet appending their IP. When a request packet reaches the destination or a node that has route information to the destination, the nodes sends a route reply packet containing path information back to the source.

Route Maintenance is the mechanism by which a sender of a packet detects network topology changes that render useless its route to the destination if two nodes in the route have moved out of range of each other. When Route Maintenance indicates a source route is broken, source nodes are notified with a Route Error packet. The sender can then attempt to use any other route to destination that is already in its cache or can invoke Route Discovery again to find a new route [JoMa96]. These key factors ensure excellent performance in multi-hop wireless ad hoc networks.

2.2.2 Multiple Path Source Routing

DSR provides only single path routing, which does not use network resources fully and deals poorly with congestion and link breakage. Multi-path routing can overcome the above problems by providing load balancing and routing failure protection by distributing traffic among a set of diverse paths. Therefore, multi-path source routing (MSR) is a good candidate for mobile bandwidth limited ad hoc networks.

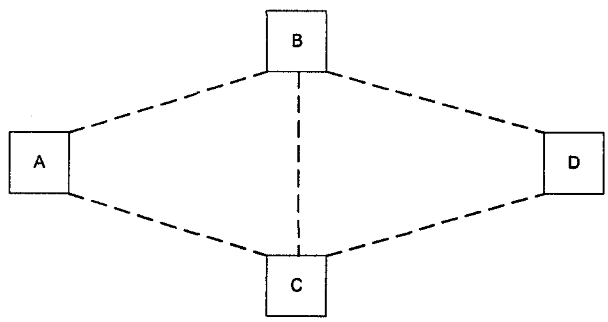


Figure 2.4: An Example of Path Finding

MSR [WaZh01] is based on DSR and has similar principles and benefits of DSR. Multi-path routing is a scheme to distribute loads among several paths according to the round trip time of every path. By utilizing source routing, MSR [ShYa02] can improve performance by applications distribution to multiple paths rather than the single path service. MSR first has to find the disjoint paths because they can provide more independent paths between two nodes. MSR uses the RREQ forwarding mechanism to find independent paths effectively. By this mechanism, each intermediate node only forwards the first received

RREQ while discarding all the others received later. An example is given in Figure 2.4 to show the procedure of finding independent path. Node A initials a route discovery and find four routes ABD, ACD, ABCD, and ACBD and only ABD and ACD are found as disjoint paths by that mechanism.

MSR also uses load balance at the source node that distributes the load and sends traffic to two more intermediate nodes. The delay is used for the metric to distribute the traffic over multiple routes and obtained by a periodic probing mechanism for each path. The destination node re-orders and combines several streams to obtain original data stream. Thus, MSR increases the network utility and balances the energy consumption of each node so as to improve the performance of ad hoc networks.

2.3 Voice over IP Principle

IP telephony uses packet switched network as the infrastructure of voice communications. It is a potential future replacement of PSTN (traditional public switched telephone network). It brings us mainly several benefits: First, it can generate significant revenue and provide competitive service because of great cost-reductions on long distance phone call. Second, network capacity and efficiency have been greatly improved. Third, it can make the future integrated infrastructure which can support any forms of communications possible. Finally, VoIP also makes the wideband telephone possible. Nowadays, it is taking over the traditional telecommunication business gradually. People have inevitably taken audio, video and teleconference as the most important feature of our next generation network. So, studying of VoIP is important, interesting, and helpful.

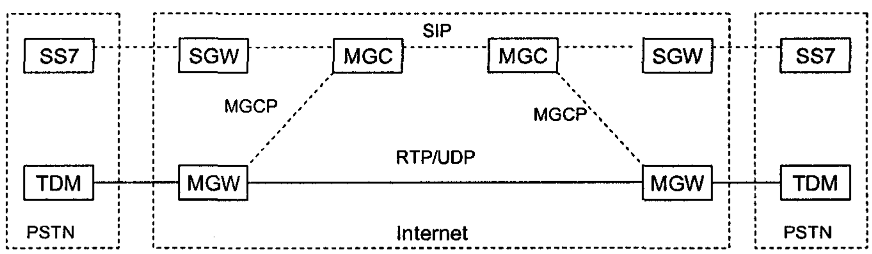


Figure 2.5: The Principle of VoIP

The principle of VoIP is shown in Figure 2.5. The VoIP network connects the Signaling System SS7 and Trunk TDM of telephone network PSTN via the Signaling

Gateway (SGW) and Media Gateway (MGW). There are two types of protocols required for implementing VoIP: signaling protocols, and media protocols [Burg02]. The signaling protocols include call signaling that is used for call setup and release and gateway signaling that is used for gateways' communication. Call signaling SIP (Session Initiation Protocols) are popularly used to create, modify, and terminate sessions with one or more participants. Gateway signaling MGCP (Media Gateway Control Protocol) is used for controlling telephony gateways from media gateway controllers (MGC). Media protocol RTP/RTCP is used to delivery voice packet to monitor the quality of voice.

2.3.1 Signaling Protocol SIP and MGCP [Burg02]

In order to obtain the destination IP address or directory number to establish a call, we need the signaling protocols. Current implementation of VoIP signaling has two main types of protocols: SIP and MGCP.

SIP is primarily used in setting up and tearing down voice or video calls. It also can be used in any applications where session initiation is a requirement. These include event subscription and notification, terminal mobility and so on. SIP is text-based protocol similar to the HTTP protocol so it is easy to decode messages. An objective for SIP was to provide a signaling for packet-based communications that can support call processing functions used presently in PSTN. These functions and features have been designed and built in Proxy Servers and User Agents that provide familiar telephony operations: dialing a number, causing a phone to ring, hearing ring back tones or a busy signal. SIP address format is similar to an Email address, for example, 8642345@skype.com.

MGCP (Media Gateway Control Protocol) is used to connection packet streams and circuit channels. A telephony gateway provides conversion between the audio signals carried on telephone circuits and data packets carried over IP networks. The gateway provides the call control such sending dial tone, collecting digits like a central switch office which provides these functions rather than the user terminal. Therefore, the MGCP is used to eliminate the need for complex, processor-intense IP telephony devices, thus simplifying and lowering the cost of these terminals.

2.3.2 Media Protocol RTP and RTCP [Burg02]

The Real-time Transport Protocol (RTP) defines a good standardized packet format for delivering audio and video over the Internet. RTP can carry any data with real-time characteristics, such as interactive audio and video. Call setup and tear-down is usually performed by the SIP/H.323 protocol. It carries the following information: payload-type identification that indicates what kind of content is being carried, PDU Sequence numbering, Time stamping that presents time of the content being carried in the PDU, and delivery monitoring. RTP itself does not provide mechanisms to ensure timely delivery and does not give any quality (QoS) guarantees either.

RTCP is used to provide feedback on the quality of service provided by RTP. The control packets is transmitted periodically to participants in a streaming multimedia session so as to obtain results of statistics on a media connection and information such as bytes sent, packets sent, lost packets, jitter, and round trip delay. Participants may use this information to increase the quality of service perhaps by limiting flow, or using a low bit rate codec instead of a high bit rate codec. There are several types of RTCP packets: sender report packet, receiver report packet, source description packet, goodbye packet and application specific packets.

2.4 Coding Scheme in Wireless Networks

Speech coding allows voice signal to be sampled, quantized, and encoded to digital format. We can use 8 kHz or16 kHz sampling rate with 8 bit or 16 bit representation. Since the bit rate of this uniform coding is too high to transport in the Internet, many encoding methods were explored to lower the bit rate. There are various kinds of voice coding schemes: PCM (Pulse Code Modulation) [ITUT93] is the simplest one which provides 64 kbps for telephone speech with A or μ law. Another coding is to utilize the connection between samples to get ADPCM (Adaptive Differential PCM) [CCIT90] can provide high quality code at 32 kbps. For the coding rate of 16 kbps and lower, linear predictive coding and code excited linear prediction coders technology will be adopted, for instance, G.729 coding. A speech coding scheme is evaluated by following attributes [Neme02]:

a) *Bit rate*: The coding bit rate ranges frame from 2.4 kbps (LPC-10) coders to 64 kbps (G.711). The rate of the coder determined the required channel bandwidth. The bit rate and the quality of voice have a tradeoff.

b) *Coding Delay*: The coding delay is a part of the overall end-to-end delay in VoIP call. The coding delay is caused by the framing as well as look-ahead delay. Small frame size and look-ahead have a low delay.

c) *Quality*: The quality of coder can be expressed in terms of how pleasant or comfortable speech sounds to the human ears. It is measured by subjective perceptual criteria, a mean opinion score (MOS) experiment.

d) *Complexity:* Speech coding algorithms generally computation intensive because they have to do a lot of digital signal processing operations such as convolutions, FFTs and digital filtering. The efficient algorithm can save the processing power and energy of digital signal processors.

In a wireless environment, the mobile radio channel is particularly dynamic due to multi-path propagation and Doppler spread which have a strong negative effect on the bit error rate of any modulation technique. So speech coding is a challenge in a wireless environment. Although equalization, diversity, and channel coding can be used to improve received signal quality and link performance over small-scale time and distance, the robust of speech coding itself is important. Recent studies focus on multiple description coding (MDC) and scalable coding of speech (SC) by utilizing the characteristics of multiple path of wireless environment [DoCh03]. More recently, AMR-WB were chosen by WCDMA and CMDA2000 as a speech coding standard of next generation communication because it can provide a wideband, lower bit rate, and high quality coding. It is introduced in following.

Table 2.1: Bit Allocation of AMR-WB Coding

Mode	Bit-rate (kbit/s)	MOS	VAD-flag	ISP	LTP-filtering	Pitch delay	Algebraic code	Codebook gain	HB-energy	Bit number frame
0	6.6	3.495	1	36		23	48	24		132
1	8.85	3.732	1	46		26	80	24		177
2	12.65	3.915	1	46	4	30	144	28		253
3	14.25	3.971	1	46	4	30	176	28		285
4	15.85	4.001	1	46	4	30	208	28		317
5	18.25	4.034	1	46	4	30	256	28		365
6	19.85	4.082	1	46	4	30	288	28		397
7	23.05	4.106	1	46	4	30	352	28		461
8	23.85	4.095	1	46	4	30	352	28	16	477
dtx	0.583		1(dither)	28				6		35

2.5 AMR-WB (G.722.2) [ITUT02]

AMR-WB standard is a developed speech coding standard as a result of ITU-T and 3GPP. The AMR-WB coding is expected to result in a compression ratio of 9 to 34 in bit rate by providing nine modes bit rate from 6.6 to 23.85 kbps with wideband characteristics (50-7000 Hz). It also provides source controlled rate operation (also called Discontinuous Transmission, DTX) by the voice activity detection which provides very low rate comfort noise abstracting from the speech. This can reduce about 50% of the bit rate compared to constant bit rate. In addition, the standard provides built-in error-resilience functions such as error concealment as well as flexible network adaptation to combat packet loss and error over error-prone wireless network. What is more, AMR is robust for packet loss and has better quality at a relatively low bit rate. These factors make AMR an attractive candidate for wireless voice communication.

The codec is based on the convention ACELP technology whose bit stream consists of typical ACELP encoder parameters shown in Table 2.1. The amount of encoder parameters determines the coding mode (bit rate) and coding quality. Different bit rates are caused mainly by different lengths of algebraic codebook. The AMR adaptive mode selection is typically based on the network capacity and radio channel conditions. Only a relatively simple source based rate adaptation is utilized using voice activity detection.

The principle of AMR-WB codec is shown as following. Analogue voice was first sampled at 16 kHz with14-bit PCM representation. Then this data can be compressed into code according to specify mode. As shown in Figure 2.6, the input signal is pre-emphasized first $H(z) = 1-\mu z^{-1}$ and then apply CELP model. A 16th order linear prediction, short-term synthesis filter is used which is given by:

$$H(z)=\frac{1}{\hat{A}(z)}=\frac{1}{1+\sum_{i=1}^{m}\hat{a}_i z^{-i}}, \qquad (1)$$

where $m = 16$ is the predictor order and $\hat{a}_i$, $i = 1,\ldots,16$ are the linear prediction parameters. The long-term, or pitch, synthesis filter is performed using the adaptive codebook approach and is given by:

$$\frac{1}{B(z)}=\frac{1}{1-g_p z^{-T}}, \qquad (2)$$

where g_p is the pitch gain and T is the pitch delay.

The CELP speech synthesis model is shown in Figure 2.6. In this model, the excitation signal at the input of the short-term Linear Predictive (LP) synthesis filter is produced by two excitation vectors from adaptive and fixed codebooks. The speech is synthesized by feeding the two properly chosen vectors from these codebooks through the short-term synthesis filter.

The optimum excitation sequence is chosen using an analysis-by-synthesis search procedure in which the error between the original and synthesized speech is minimized by a perceptually weighted distortion measure.

The perceptual weighting filter used is given by:

$$W(z) = A(z/\gamma_1)H_{\text{de-emph}}(z), \tag{3}$$

where $A(z)$ is the unquantized LP filter, $H_{\text{de-emph}} = \dfrac{1}{1-0.68z^{-1}}$, and $\gamma_1 = 0.92$ is the perceptual weighting factor. The encoder performs the analysis of the LPC, LTP and fixed codebook parameters at 12.8 kHz sampling rate. The coder operates on speech frames of 20 ms. At each frame, the speech signal is analyzed to extract the parameters of the CELP model including LP filter coefficients, adaptive and fixed codebooks' indices and gains. Besides these parameters, high-band gain indices are calculated in 23.85 kbps mode. These parameters are encoded and transmitted.

At the decoder, these parameters are decoded and speech is synthesized by filtering the reconstructed excitation signal through the LP synthesis filter.

2.5.1 The Principle of Encoder

The signal flow at the encoder is shown in Figure 2.6. The pre-process includes down sampling, high-pass and pre-emphasis filtering. LP analysis is implemented once per frame. The set of LP parameters is converted to immittance spectrum pairs (ISP) and vector quantized using split-multistage vector quantization. The speech frame of 20 ms is divided into 4 sub-frames each. The adaptive and fixed codebook parameters are encoded in every subframe. The quantized and unquantized LP parameters or their interpolated versions are used depending on the subframe. An open-loop pitch lag is estimated in every other subframe or once per frame [ITUT02].

The following operations are then performed for each subframe:

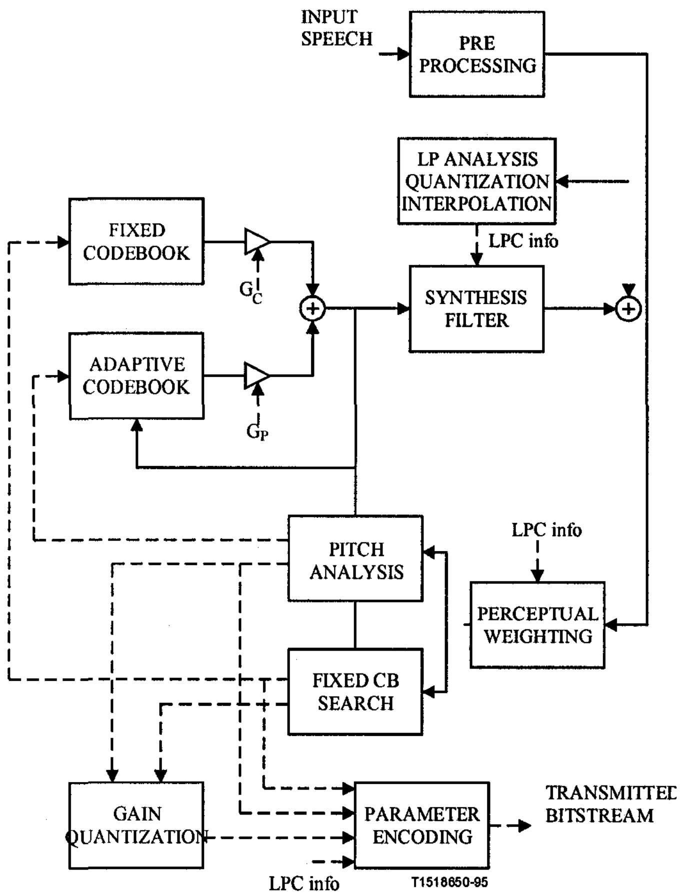


Figure 2.6 G.722.2 Block Diagram of the CELP Synthesis Model [ITUT02]

1) Filtering the LP residual through the weighted synthesis filter *W(z)H(z)* which is the error between LP residual and excitation to get the target signal *x(n)*.

2) The impulse response, *h(n)* of the weighted synthesis filter is computed.

3) Using the target *x(n)* and impulse response *h(n)*, by searching around the open-loop pitch lag to find the pitch lag and gain.

4) The target *x(n)* is removed the adaptive codebook contribution to get new target *x2(n)* used in the fixed algebraic codebook search.

5) The gains of the adaptive and fixed codebook are vector quantified with 6 or 7 bits with moving average (MA) prediction.

6) Finally, the filter memories are updated for finding the target signal in the next subframe.

The bit allocation and bit rate of the AMR-WB codec modes is shown in Table 2.1. Note that the most significant bits (MSB) are always sent first.

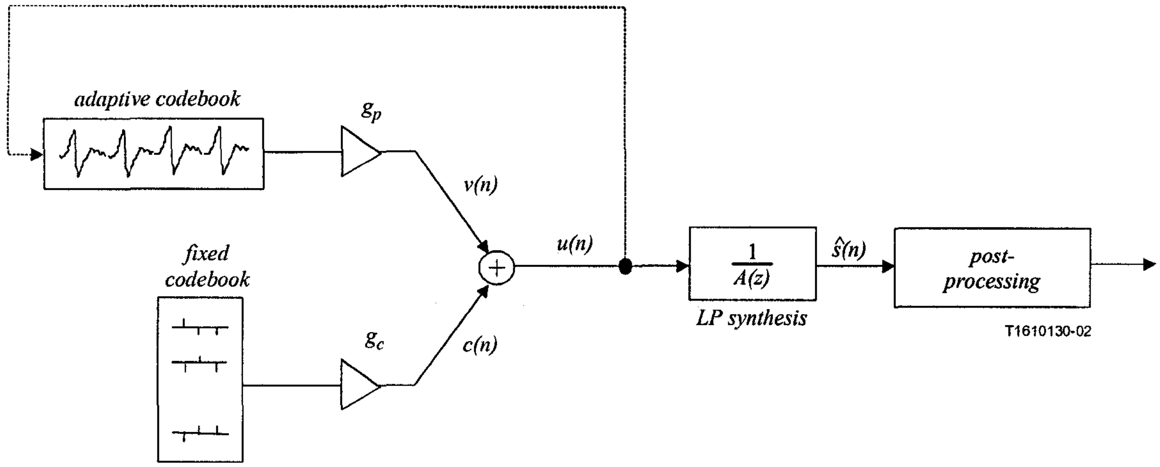


Figure 2.7: G.722.2 Simplified Block Diagram of the Decoder [ITUT02]

2.5.2 Principles of the Decoder

The signal flow at the decoder is shown in Figure 2.7. The coding indices are obtained from the received bit-stream and then are decoded to obtain the coder parameters at each frame [ITUT02]. These parameters include the ISP vector, the 4 fractional pitch lags, the four LTP filtering parameters, the four innovative code-vectors, and the four sets of vector quantized pitch and codebook gains show in Table 2.1. A high-band gain index is needed in 23.85 kbps mode. The ISP vector is converted to the LP filter coefficients and interpolated to obtain LP filters at each sub-frame. For each subframe:

1) Adding the adaptive and codebook scaled by their respective gains to construct the excitation.
2) Filtering the excitation through the LP synthesis filter to reconstruct speech
3) The reconstructed speech is de-emphasized.
4) The reconstructed speech is restored to 16 kHz sampling rate and high-band speech signal is added to the frequency band.

2.5.3 Error Concealment and Voice Activity Detection

AMR-WB is performed error detection for the most sensitive bits of AMR by the network with 35-bit data. Erroneous/lost speech frames should be substituted with either repletion or extrapolation of the previously good speech frames with a gradually decreasing output level [ITUT02i]. Lost SID also should be substituted by earlier received valid SID. The solution for substation and muting is based on a state machine with seven states.

AMR-WB performs voice activity detection (VAD) on the transmitting side by evaluating the background acoustic noise on the transmitting side in order to transmit characteristic parameters to the receiving side; on the receiving side, it generates a similar noise, called comfort noise during periods where the transmission is switch off every three frame with 35-bit coding. Handover needs seven inactive frame and send silence insertion descriptor (SID) frame by extracting ISF evaluation, frame energy calculation and stationary background energy.

2.6 Packet Loss Recovery

There are many causes to packet loss in Internet networks, such as congestion, router buffer overflow, linker failure and Ethernet problems. Packet loss causes the loss of all speech information of one frame or many frames. A lot of work has been done to combat packet loss of VoIP system. During the planning and implementing the VoIP system, people adopt call admission, route diversity, RSVP, and recovery etc. to decrease packet loss and improve system performance. There are also many packet-loss recovery techniques for VoIP. These techniques can be classified as sender-driven and receiver-based. They are introduced in the following.

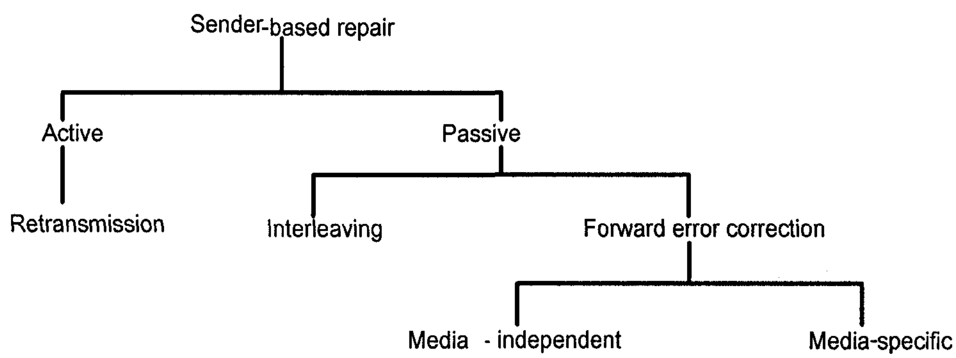


Figure 2.8: Hierarchy of Sender-Based Repair Techniques [PeHo98]

2.6.1 Sender Based Repair

Sender-Based repair techniques may be split into two major classes: active retransmission and passive channel coding. It is further possible to subdivide the set of channel coding techniques as interleaving technique and Forward Error Correction (FEC) shown in Figure 2.8. FEC can be categorized as either media independent, typically based upon exclusive-OR operations, or media-specific based on the properties of a speech signal [PeHo98].

Interactive speech applications have tight latency bounds, and end-to-end delays need to be less than 150 ms. Therefore retransmission is not typically employed for the recovery of lost packets. Many packets are interleaved together so as to overcome the effect of a burst loss. If the end-end delay can be tolerated, the interleaving is one useful method.

Forward Error Correction (FEC) is classified as media-independent and media-specific. FEC is an effective approach for error-prone channel for example wireless network for error detection and correction at cost of bandwidth. Reed-Solomon [PaMo99] is a widely used channel coding. Basically, this scheme operates by aligning k successive data packets vertically, each of which is subsequently partitioned in q bit symbols. An $RS(n, k)$ code is used to encode the vertically aligned q bit symbols to produce n-k parity packet [Wick95].

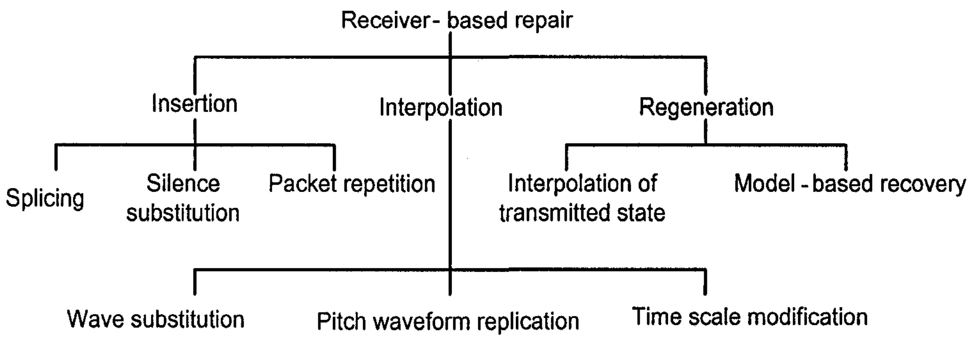


Figure 2.9: A Hierarchy of Receiver-based Repair Techniques

2.6.2 Receiver Based Repair

Figure 2.9 shows a hierarchy of receiver-based repair for packet loss in VoIP. Receiver-based repair called error concealment technology is initiated by the receiver and does not require assistance from the sender. It relies on producing a replacement for a lost packet which is similar to the original. That is possible since the audio or speech signals exhibit large amounts of short-term self-similarity. Thus, these techniques work for relatively small loss rates (<15%) and for small packets (4-40 ms). It will lose efficacy when the loss length exceeds length of a phoneme (50-100 ms), since whole phonemes may be missed by the listener. Error concealment forms a cheap and effective means of patching over the loss. It can be seen that these techniques split into three categories [Peho98]:

- Insertion-based schemes repair losses by inserting a fill-in packet. This fill-in is usually very simple: silence, noise or repetition of the previous packet with attenuation of gain.
- Interpolation-based schemes use some form of pattern matching and interpolation to derive a replacement packet which is expected to be similar to the lost packet by comparing previous packet and latter ones. These techniques are more difficult to implement and require more processing when compared with insertion-based schemes but have better performance.
- Regeneration-based schemes derive the decoder state from packets surrounding the loss and generate a replacement for the lost packet from that. This process is expensive to implement but can give good results. The erasure concealment in G.729 belongs to this category.

The receiver-based repair techniques almost do not incur extra delay for whole system. The combination of sender-based repair and receiver-based repair are expected to improve the reconstructed speech quality in the case of packet loss.

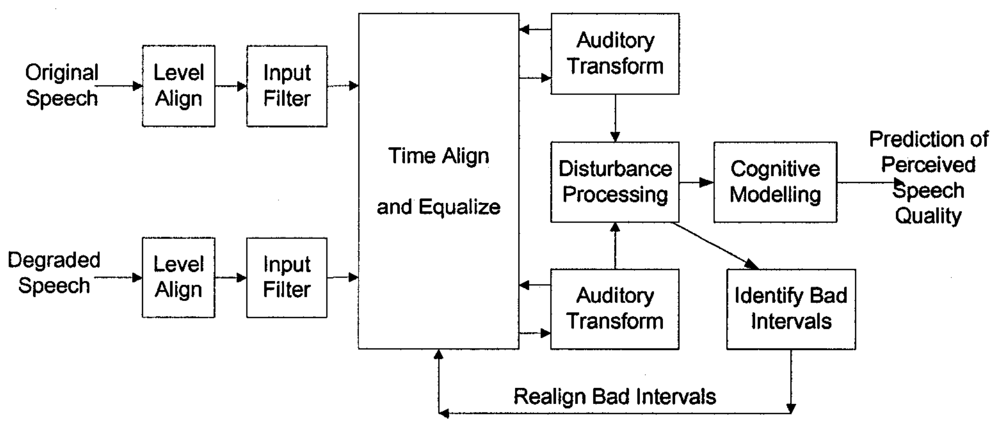


Figure 2.10: Principle of PESQ [ITUT01]

2.7 PESQ Measurement

The measurement quality of voice quality is an important indicator. In an analogue domain, the signal noise ratio is used for measurement. In digital domain, the criteria are no long valid because the perceptual model is used for speech coding. Therefore, subjective evaluation is used for quality evaluation. The most popular ranking system is mean opinion score (MOS) which evaluates the quality of voice with five-scores and each point associated with descriptions: 5 excellent, 4 good, 3 fair, 2 poor and 1 bad. However, these scores are obtained by the perception of a listener, and it is time-consuming and skillful. Therefore, ITU-T creates the standard P.862 PESQ (Perceptual Evaluation of Speech Quality), which utilizes PESQ model to evaluate the quality of speech by comparing the degraded signal and the original one [ITUT01].

The principle of PESQ is shown in Figure 2.10. The model first performs level aligning to both signals to a standard level. And then they are filtered (using an FFT) with an input filter to model a standard telephone handset. After that, the signals are implemented time aligning and then processed through an auditory transform which involves equalizing for linear filtering and for gain variation. Both the original and the degraded signals are transformed to an internal representation that is analogous to the psychophysical representation of audio signals. In the human auditory system, taking account of perceptual frequency and loudness are considered.

The evaluation is achieved in several stages: time alignment, level alignment to a calibrated listening level, time-frequency mapping, frequency warping, and compressive loudness scaling. Two distortion parameters are extracted from the disturbance (the difference between the transforms of the signals), and are aggregated in frequency and time and mapped to a prediction of subjective mean opinion score (MOS). PESQ measures the effects of one-way speech distortion and noise on the speech quality. The effects of loudness, loss, delay, side-tone, echo, and other impairments related to two-way interaction are not reflected in the PESQ scores.

2.8 Concluding Remark

In this chapter, we discuss the background knowledge, which is used in the thesis. We first introduced wireless network environment, and ad hoc routing protocols. We also discussed the principle of voice over IP and speech coding scheme AMR-WB. Finally, we investigated packet loss recovery technology and the perceptual evaluation of speech quality (PESQ).

Chapter Three
Adaptive Source Network Rate Control Scheme

In this chapter, we propose the Adaptive Source-Network rate Control scheme (ASNC) to transport voice over wireless ad hoc networks. We first discuss and analyze the characteristics of ad hoc networks and speech codec AMR-WB before presenting our proposal. Then we discuss our proposed ASNC scheme. Finally, we detail the implementation of the different components of the scheme including adaptive network based rate control, source-based control and frame combination.

3.1 The Model

In this section, we will discuss the model used in the proposed scheme. We will provide the models of AMR-WB speech coding and ad hoc channel used by different adaptive rate control schemes.

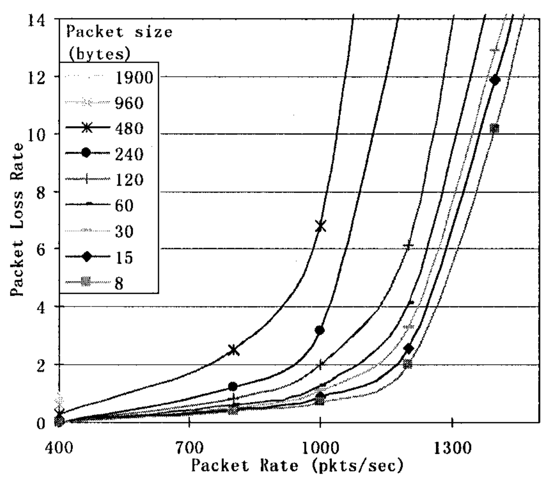
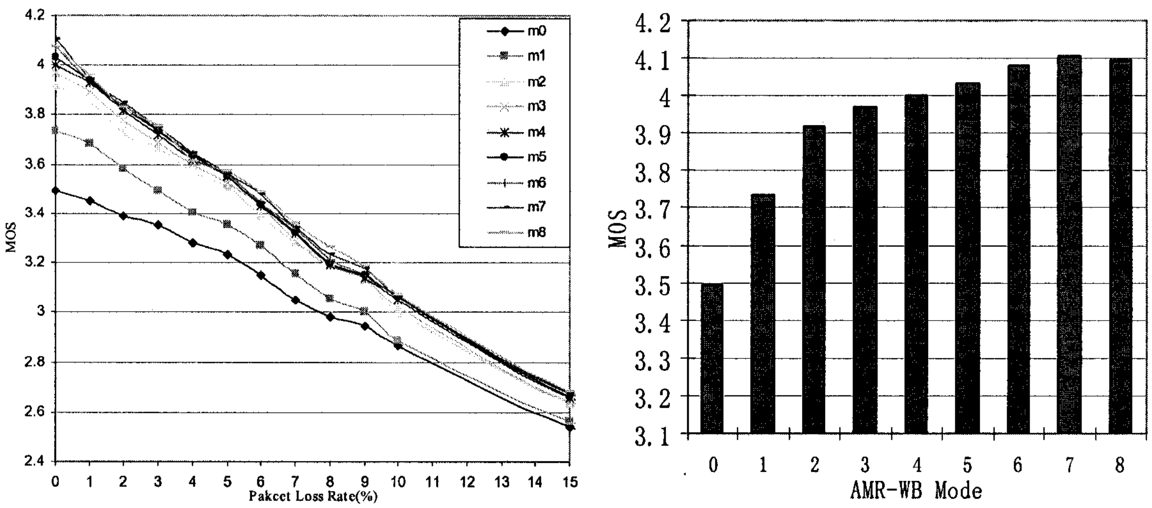


Figure 3.1: Packet Loss Rate vs. Packet Rate for Various Packet Size in a Single-Hop Ad Hoc Network.

3.1.1 Channel Characteristic of an Ad Hoc Network

We create the channel characteristics model of ad hoc networks to demonstrate the throughput of wireless channels versus traffic load. D-ITG (Distributed Internet Traffic Generator) is a packet generator which can produce traffic at packet level accurately replicating appropriate stochastic processes for both inter departure time and packet size. By using it to generate packets of a given rate and size, we can then test the throughput of wireless ad hoc channels, and evaluate the performance at the receiver. The measurement results shown in Figure 3.1 illustrate that the packet loss rate is an increasing function of packet rate and size in a single-hop ad hoc network. Consider the bottom curve (using packet size of 8 bytes), one observes that packet loss rate increases rapidly beyond a certain data rate, when congestion occurs. Similar observations made for different packet size. When packet size is very large, the packet loss performance is very bad because large size can cause fragmentation and serious congestion. For example, packet size of 960 bytes and 1900 bytes is measured at packet rate of 400 packets/sec. As the packet rate increases, the "take off" point of the packet rate is decreased before increasing once more.



(a) (b)

Figure 3.2: (a) Relationship between Packet Loss Rate and MOS in a Single-Hop Ad Hoc Network. (b) Relationship between Mode and MOS Value

3.1.2 Characteristics of the AMR-WB Speech Coding

There are 9 coding modes of AMR-WB speech coding that we can use in the adaptive source network rate control scheme. The relationship between the bit rates and the MOS values of AMR-WB modes is shown in Figure 3.2b. We also write programs to simulate different packet loss with which we can measure the MOS values at different packet loss rates. The standard speech clips provided by ITU-T are used in the tests and the results are shown in Figure 3.2a. The results show that the MOS value is a decreasing function of packet loss rate.

Consider the bottom curve (using coding Mode-0), one observes that MOS decreases linearly approximately as the packet loss rate increases. Similar observations can be made for different coding modes but at larger slopes. The higher the mode/bit rate, the larger the slope.

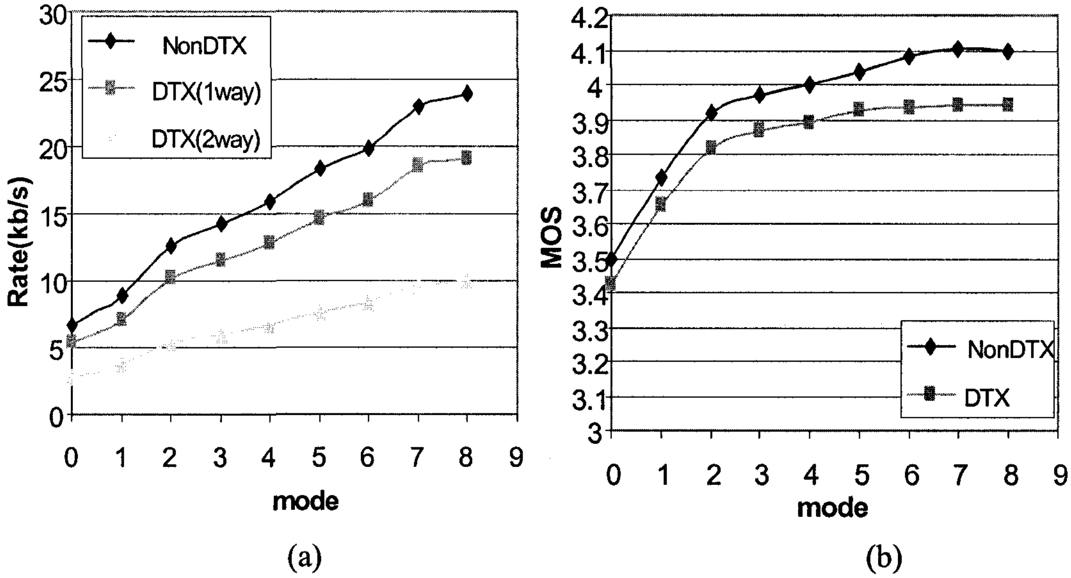


Figure 3.3: Relationship of Coding Mode and MOS for Discontinuous Transmission

3.1.3 Source Controlled Rate Operation

Source Controlled Rate operation (SCR) (called DTX (Discontinuous Transmission) sometimes) can be used to encode an input signal at a lower average rate by taking speech inactivity into account. It can reduce the load and overall interference and also save power in the source. In AMR-WB speech coding, the SCR uses a Voice Activity Detector to evaluate the background noise on the transmission side, in order to transmit characteristic parameters to the receiving side. The receiving side generates a similar noise, called "comfort noise" during periods when the transmission is switched off. The comfort noise is transmitted with a very low bit rate, and therefore it saves the bandwidth. Voice silence intervals have led to the strategy of multiplexing multiple connections to share the bandwidth of large transmission pipes.

The DTX can reduce the bit rate of AWR-WB coding as shown in Figure 3.3a. One-way DTX and two-way DTX have lower bit rate than non-DTX. Voice Activity Factor (VAF) is defined as percentage of active frame to total frames. The single way VAF is about

85% and two-way VAF is 42.5%. Both of them (shown as DTX in Fig 3.3b) suffer from the same loss of speech quality when compared with non-DTX.

Packet combination can be used when the payload of the speech packet is small (e.g. about 60 bytes) and header of packet is large (e.g. about 82 bytes). Let the actual bit rate be R the speech coding rate r, the packet header size H and the frame length T in time. Then the following relationship holds: $R=r+H/T$. The relationship shows that increasing T can reduce the actual bit rate. The frame length of AMR-WB is only 20 ms, so we can merge two frames into one 40 ms long packet so as to reduce packet rate and overhead. This gives the wireless channel a much better utilization and less congestion. Both schemes are very effective schemes to improve performance in wireless environment where the resource is limited.

3.1.4 Adaptive Control Schemes

Adaptive control involves modifying the control rule to cope with the parameters of the system that are slowly time-varying or uncertain. Adaptive control can be used in the physical layer, network and application layer of wireless LAN to improve the quality of transmission.

3.1.4.1 Adaptive Control in Physical Layer

The adaptive control in the Physical Layer of 802.11b employs a link adaptation method that will select one transmission mode out of several rate modes available at a particular frame time. It starts at 11 Mbps by default, but will decrease to 5.5, then 2, and finally to 1 Mbps depending on the frame-by-frame signal quality and strength. The lower data rates uses less complex modulation method but more redundant methods of data encoding so that they are less susceptible to corruption due to time-varying errors and signal attenuation, as well as interference from the contention with other nodes in a wireless channel. In general, one can expect to improve system throughput and the energy consumption performance [DoSC04].

3.1.4.2 Adaptive Control in Network and Application Layer

Adaptive control also is used in network and application layer to improve the network throughput. The performance of ad hoc networks is related to the number of hops increases and the number of nodes because all nodes of ad hoc networks have to share channel locally with other nodes. The topology and routing information of ad hoc networks can be used to determine the capacity of networks. The information of application layer such as the packet loss /error is also used to adjust transmission bit rate. Therefore, the adaptive rate control scheme to be discussed in Section 3.3 can improve the performance of transmission.

3.1.4.3 Adaptive Control in Source Coding

The codec determines the rate mode according to the source characteristics and network status information. The AMR adaptive mode selection is typically based on the network capacity and radio channel conditions. A source rate operation is utilized to reduce bit rate further by using voice activity detection. Thus, the network capacity is optimized during silence periods. It can be further optimized during active speech with source based adaptation and variable bit rate coding [MaOj02]. Our adaptive rate control scheme to be discussed in Section 3.3 determines the best codec mode to maintain high perceptual speech quality according source characteristics and network status.

3.2 Throughput Requirement Analysis

In order to find the method to improve performance of voice over ad hoc wireless networks, we need to analyze the throughput capacity of single hop and multiple hops ad hoc networks as introduced in Section 3.1. Since all n nodes share a common channel with bandwidth W in wireless networks, the transmission bit rate for each node is W/N. In an ad hoc network, this bandwidth is decreased because of multi-hop relay. Since the nodes have to communicate with each other through the shared channel, it is easy to produce collision and congestion that may lead to packet loss because of reduced bandwidth. What is more, transmission bandwidth is determined by the packet payload and header. The bigger the ratio of payload to overhead, the larger the transmission bandwidth is. Moreover, various kinds of interferences, attenuations and fading make the channel error-prone. This will lead to packet error and packet drops. We will discuss the channel throughput capacity of ad hoc network as following.

3.2.1 A Single Hop Wireless Network

We have reviewed the CSMA/CA operation of 802.11b in Section 2.1. Here, we shall first analyze the capacity of a single hop wireless network to give a reference to compare with multi-hop networks. We explore the relationship between the number of call sessions and packet payload, overhead and data rate.

<table>
<tr><td colspan="10">PPDU</td></tr>
<tr><td rowspan="5">PLCP Preamble 18 byte</td><td rowspan="5">PLCP Header 6 byte</td><td colspan="8">PSDU</td></tr>
<tr><td colspan="8">IEEE 802.11 Data Frame (MPDU)</td></tr>
<tr><td rowspan="3">MAC Header 30 byte</td><td colspan="6">Data (0-2312)</td><td rowspan="3">FCS 4 byte</td></tr>
<tr><td rowspan="2">LLC 3 byte</td><td rowspan="2">SNAP 5 byte</td><td colspan="4">UDP/IP Datagram (MTU size)</td></tr>
<tr><td>IP Header 20 byte</td><td>UDP Header 8 byte</td><td>RTP Header 12 byte</td><td>Data (MSS Size) 0-65495 byte</td></tr>
</table>

Figure 3.4: Packet Format of IEEE 802.11

The structure of the 802.11b packet is shown in Figure 3.4. The packet contains the header MAC, IP, UDP and RTP and payload. The time of sending a packet includes the transmission time of packet header H and payload P, sender's overhead time, and receiver's overhead time. We analyze them as following.

At MAC layer the overhead time at the sender includes DIFS, average back-off time CW_{arg} and physical layer header PHY, i.e.

$$T_{sd}=DIFS+CW_{arg}+PHY, \tag{1}$$

where $CW_{arg}=T_{slot}(CW_{min}-1)/2,$ and CW_{min} is minimal contention window and T_{slot} is minimal transmission detection time. The overhead time occurred at the receiver includes SIFS and the acknowledge time ACK, i.e.

$$T_{rcv}=SIFS+ACK \tag{2}$$

Therefore, the total sending time of a packet is

$$T=(H+P)/R+T_{sd}+T_{rcv} \tag{3}$$

$$H=H_{MAC}+H_{IP/UDP/RTP}$$

where R is data rate and the back-off and retransmission caused by collision is negligible compared with other overhead. Thus, the packet rate R_p, the transmission bandwidth W and the number of call sessions n are respectively

$$R_p=1/T= 1/((H+P)/R+T_{sd}+T_{rcv}) \tag{4}$$

$$W=P/T =RP/(P+H+(Tsd+Tsr)R) \tag{5}$$

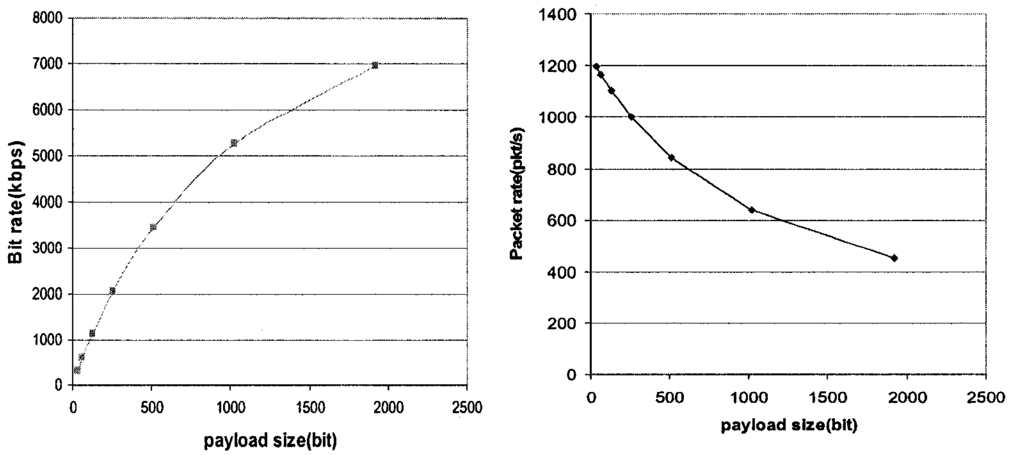
$$n=W/(2r) \tag{6}$$

where r is the speech bit rate.

The values of *DIFS, SIFS, PHY, ACK* for 802.11 are given in Table 3.1. Assuming the AMR-WB Mode 2 (12.8 kbps equivalent to a payload of 256 bits/frame and a frame rate of 50 frame/sec) is used, we can see WLAN provides a bandwidth of 305.9 kbps and the packet rate 1195 pkts/s for the speech coding. So, we can have about 11 sessions of two-way calls.

Table 3.1: Parameters of IEEE 802.11b DCF

Parameter	Value	Parameter	Value
DIFS	50 μsec	Data Rate	11Mb/s
SIFS	10 μsec	Basic Rate	2 Mb/s
T_{slot}	20 μsec	PHY header	192 μsec
CW_{min}	32	MAC header	34bytes
CW_{max}	1023	ACK	202.2 μsec
		Header(IP/UDP/RTP)	40bytes



(a) Bit Rate (b) Packet Rate

Figure 3.5: Bit Rate and Packet Rate vs. Payload Size in Wireless Networks

Figure 3.5 shows the bit rate and packet rate as a function of payload size. We can see that as the payload of voice packet increases, the bandwidth increases and the packet rate decreases due to the overhead of the wireless physical layer. We can use source coding to decrease packet rate and/or packet combination to merge two or more packets together to decrease overhead. Because the merged packets can share the same overhead, the packet overhead and packet rate will reduce greatly so that congestion will be alleviated. Thus, the efficiency of link is improved. The cost of this method is the increased delay budgets. For example, using 12.65 kbps Mode 2 for AMR-WB coding, we can get a capacity between 306 kbps and 595 kbps when the interval of transmission packet is doubled.

3.2.2 Multiple-Hop Ad Hoc Networks

In order to utilize the resource reasonably in ad hoc networks, it is necessary to choose the suitable transmission rate as the reference transmission rate. This rate is affected by the number of hops between source and destination and the number of interference neighbors of intermediate nodes [QuPe05]. The larger the number of neighboring nodes and hops, the smaller the transmission bandwidth is. For illustration, we can consider a wireless ad hoc network with homogenous nodes that have the same traffic pattern, and then the effective transmission rate R_e is given by

$$R_e = Min(\frac{W_i}{h(1+n_i)}) \tag{7}$$

where $n_i = n-1$ is the number of interference nodes of the ith node. The ith node has transmission rate Wi that can be calculated using Eq. (6). The h is the number of hops. Two parameters can be estimated by an ad hoc routing protocol, such as the DSR or the MSR. Therefore, we can use an adaptive rate control scheme to ensure node transmission rate (that is less than the channel capacity) to avoid congestion. For example, we can determine that five VoIP sessions can be supported for two-hop network with Mode-2 AMR-WB coding.

3.3 The Proposed Adaptive Source-Network Rate Control Scheme

In this section, we will propose the ASNC (Adaptive Source-Network Rate Control) scheme which combines source rate control scheme and network rate control scheme. The adaptive source-network rate control scheme utilizes the source and network status to control the AMR coding bit rate and its discontinuous transmission. The routing information such the number of hops is also used to control the initial coding rate.

3.3.1 Adaptive Network Rate Control Scheme

The adaptive network rate control scheme is described as follows. The receiving node measures a parameter indicative of the network performance and feeds back to the source. The source adjusts the coding rate according to the network status. Our objective is to use MOS value as the criteria to evaluate the overall end-to-end performance of a voice stream. Perceptual Evaluation of Speech Quality (PESQ) provides a numerical indication of the perceived quality of received speech after compression and transmission and it is easier to be implemented than other measurement tools. Therefore, in this thesis we choose to use PESQ to evaluate the quality of speech. However, the PESQ MOS can not be measured in real time because it needs the original voice as reference. Because the MOS value has a relationship with the packet loss rate for a certain mode, when we have packet loss, we can get the MOS value by mapping packet loss rate. We use this method to implement adaptive control schemes.

In order to use adaptive control schemes to get the optimal bit rate, it was found that the bit rate can be adjusted using formula (8). The Q= $MOS(m, pl)$ denotes obtaining MOS as shown in Figure 3.2. The m is a coding mode (bit rate), pl denotes packet loss rate that can be obtain from the feedback of the receiver. Thus, the adaptive control algorithm is shown by the following:

(1) $R_{old}=R_{ini}$; $Q_{old}=Q_{ini}$

(2) $R=R_{old}+(Q-Q_{old})*k$ (8)

(3) $R_{old}=R$; $Q_{old}=Q$; goto (2)

where k is multiplier k=14.38 kbps

First, we set the bit rate R as the average bit rate R_{ini} and set MOS Q as target quality Q_{ini}. Then we use (2) to calculate the new voice coding rate by comparing current MOS and previous MOS. Finally, we save the R and Q as old bit rate and quality and do next loop. Thus, we can get optimal coding mode.

3.3.2 Adaptive Source-Network Rate Control Scheme

Under the adaptive source network rate control scheme, the transmitting node first chooses average rate (Mode 4) as the initial coding rate. Then, the transmitting node adjusts the coding bit rate according to the feedback information such as packet loss rates from the receiving node. When the speech quality is lower than $Q1$, which means bandwidth not enough, the source uses a voice activity detector to implement discontinuous transmission control. When the quality of speech is lower than $Q2$, the packet combination scheme is used with a 40 ms packet. $Q1$ and $Q2$ are determined by the experiments (i.e. $Q1$=3.7 and $Q2$=3.5). In the mean time, the Cyclic Redundancy Check coding is used to detect the bit error. If packet loss and bit error occur, the receiving node will use error concealment to compensate it. Thus, our scheme can adapt to the changes in the network environments and source characteristics, that is, the speech coding system encodes a suitable bit rate according to the network and source status. MOS value is used as the criteria to evaluate the end-to-end performance. This scheme overcomes the performance loss due to the source coding rate insufficiency and excess. The algorithm implementation procedure is as follows:

- Choose an average rate as initial coding rate, and then adjust the coding rate according to network feedback information.
- When the quality of speech is lower than *Q1*, source rate control scheme is used.
- When the quality of speech is lower than *Q2*, the packet combination scheme is used.

When network traffic increases due to any reason, for example, more VoIP sessions start leading to the deterioration of the quality of transmission, the receiving terminal will feed back the information such as packet loss to the sending terminal. If the packet loss rate decreases, the sending node will increase rate to get higher quality. Therefore, the method will ensure the quality of voice optimization.

Besides, CRC (cyclic Redundancy checking) coding is employed to detect errors and use error concealment to relieve the degradation of quality caused by packet loss and packet error.

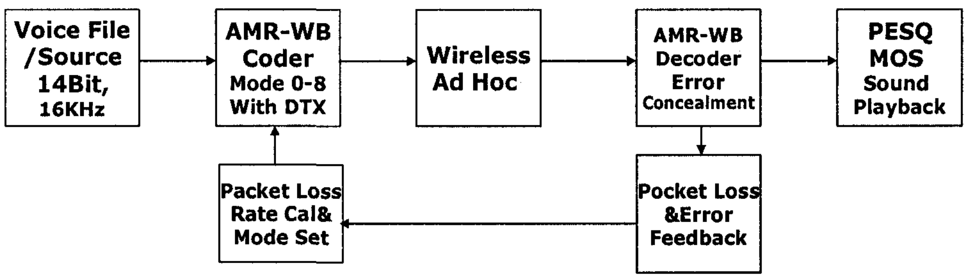


Figure 3.6: Block Diagram of Adaptive Rate Control using AMR-WB Coding

3.4 Implementation of ASNC Schemes

The implementation of the ASNC scheme is shown in the block diagram in Figure. 3.6. The AMR-WB coding is chosen as a candidate for the adaptive rate control scheme because it has many advantages. It provides wideband and high quality speech with a low variable bit rate. The study shows that it is still understandable under even 50% packet loss rate because it uses robust speech coding for packet loss. It also provides DTX (Discontinuous Transmission) that uses very low rate comfort noise abstracting from the speech at inactive period. In addition, it provides built-in error-resilience functions such as error concealment as well as flexible network adaptation to combat packet loss and error over error-prone wireless network. The main body of AMR-WB, comfort noise aspects and source controlled rate operation are implemented and are added into our VoIP program. Each function block of the program will be described in detail below.

3.4.1 Voice Sampling and Coding

The first and second blocks (on the left) perform digital-to-analogue conversion and AMR-WB coding respectively. The input speech is sampled by Windows Sound Recorder [Wrig02] with a 16 kHz sampling rate and converted to 14 bit uniform PCM. Then, the uniform PCM is converted to AMR-WB code, whose bit rate is decided by the Mode Set block in the bottom left. The AMR-WB coding has variable bit rates ranging from 6.6 to 23.85 kbps in 9 modes that can be used for adaptive rate control. The AMR-WB also has a source rate control function that provides lower bit rates. Finally, voice streams are transmitted over wireless ad hoc networks. Both periodic and aperiodic modes of sending packets are used.

The RTP/UDP/IP protocol stack is used to packetize and to transmit voice stream over ad hoc network. Rather than TCP/IP, we use a UDP/IP combination which provides a best effort packet delivery service because it can satisfy a stringent delay threshold for voice streams. The RTP (Real-time Transport Protocol) is also used to enable real-time multimedia applications over IP network.

The packet structure consists of class A data and class B data. Class B contains bits where increasing error rates gradually reduce the speech quality but usually possible without annoying artifacts. Class A is protected by the Cyclic Redundancy Check (CRC) since it contains the bits most sensitive to errors and can result in a corrupted speech frame. CRC parity bits are generated by the cyclic generator polynomial: $G(x)=x^8+x^6+x^5+x^4+1$, which is computed over all Class A bits of AMR-WB core frame for mode 0-8 and mode 9 for comfort noise bits.

3.4.2 Ad Hoc Routing

The third block (on the middle) is the transmission channel of wireless ad hoc networks. In order to transmit voice packet from a source to a destination, intermediate nodes are required to route and forward the packets. We use the routing protocols DSR, MSR for ad hoc networks developed by Tianjin University. DSR provides on-demand routing protocol in which each node maintains a routing table for each entry (destination) including intermediate nodes' information. MSR is extended from DSR to distribute the load to multiple paths which can alleviate the congestion problem. Thus, we implement the adaptive rate control scheme over ad hoc routing protocols so as to improve the performance of the network stability and the quality of speech.

3.4.3 Decoding and Evaluation

The fourth block (on the right) is the AMR-WB decoder which determines packet loss according to sequence number, timeout and packet error using CRC (Cyclic Redundancy Check). If packet loss or error occurs, the receiver feeds back to the sender and uses an error concealment mechanism to repair it by filling in suitable data. The decoder will output decoded wave data and drives the sound card to play sound so that we can check the quality of speech perceptually. When transmission finishes, PESQ (Perceptual Evaluation of Speech Quality) is used to evaluate the quality of speech by comparing the degraded file with original one to get speech MOS value.

The PESQ can measure the quality of active speech by comparing the original sample with the degraded speech at the output of a communications system [ITUT01]. It first implements time alignment by calculating the delay between the original and degraded speech. Then the original signal is compared with its aligned degraded voice by using a perceptual model to transform them to a psychophysical representation of audio signal. Finally, one can get the quality of speech from the difference according to the cognitive model. We integrated the source code of PESQ into our VoIP program.

3.4.4 Feedback and Mode Decision

This is the fifth block located in the bottom right corner. If the receiver detects packet loss or error by checking the sequence number, CRC code and reception timeout, the receiver sends a packet with loss/error indication to inform the sender. After collecting this information, the sender calculates the packet loss rate, and then decides a suitable bit rate (mode). There are three steps to be done for adaptive control. First, we adjust the coding bit rate according to network status. Then we use discontinuous transmission scheme when the quality is not enough. If they still do not satisfy the performance requirement, we can use packet combination to transmit double length packets to reduce the overhead of traffic at the cost of increasing delays.

Table 3.2: Characteristics of Speech Coding Schemes

Compression	KBits per second	Need fast CPU?	Sound Fidelity	Sound Fidelity	MOS
G.711	64.000	No	Best	Best	4.500
ADPCM	32.000	No	Good	Good	4.145
G.729	8.000	Yes	Good	Good	3.745
GSM	13.200	Yes	Good	Good	3.677
LPC	5.200	Yes	Fair	Fair	2.829
LPC-10	2.768	Extremely	Fair	Fair	2.709
AMR-WB	6.6-23.85	Yes	Best	Best	3.495-4.095

3.5 Testing Coding Schemes

In this section, we measure the performance of different speech coding schemes shown in Table 3.2. A good digital speech coding should generate good quality with low bit rate and more robust to the bit error or packet loss. The testing results show that LPC and LPC-10 have low bit rate but their quality is not good. G.711 and ADPCM have good quality but they have high bit rate. The present popular standard G.729 has low bit rate and can get good quality but it can not provide wideband and variable coding. AMR-WB can provide wideband speech coding, variable speech coding that has low bit rate and good quality. Therefore, AMR-WB is a good choice for our speech coding scheme.

3.6 Concluding Remarks

In this chapter, we first analyzed the characteristics of ad hoc networks, speech codec AMR-WB and adaptive control. We realize that packet loss rate is an increasing function of network traffic, the coding quality is a decreasing function of packet loss rate and throughput of ad hoc network is a decreasing function of the number of hops and the number of source nodes. Then we have proposed an adaptive source-network rate control scheme to improve the quality of voice transport over wireless ad hoc networks. The basic idea is to adjust the voice coding bit rate to adapt the available network bandwidth so as to maximize the quality of voice since the packet loss rate increases with increasing packet rate and size. We used MOS as a standard to control end-to-end performance of the output stream of a voice encoder. The effectiveness of the scheme will be validated by the testbed and simulation in next chapters.

Chapter Four
Testbed Experiments

In this chapter, we shall discuss the implementation of the ad hoc network testbed. The hardware setup and software development environment of Linux are described as well as the code development for voice over IP over ad hoc network. We also implement initial testing and verification, and discuss the performance measures and measurement scenarios.

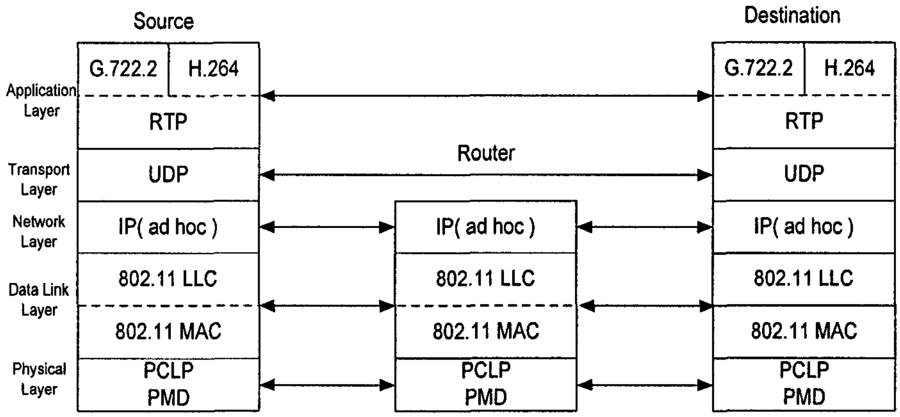


Figure 4.1: The Architecture of the Testbed

4.1 Testbed Architecture

Our testbed employs a five-layer architecture model shown in Figure 4.1. All management operations discussed in Section 2.1 are adopted. The functions of various layers are as follows.

4.1.1 Testbed Layer Model

Recall that in Section 2.1, the 802.11 physical layer is split into the PLCP (Physical Layer Convergence Procedure) and the PMD (Physical Medium Dependent) sublayer shown in Figure 4.1. The PMD takes care of the wireless DSSS encoding and the PLCP prepares 802.11 frames for transmission and directs the PMD to actually transmit signals, change radio channels, receive signals, and implement link adaptation which is employed to adjust the data transmission rate to ensure the data correct receipt. The data link layer within 802.11 consists of the LLC and the MAC sublayers. 802.11 LLC uses the same 802.2 LLC and 48-bit addressing as other 802 LANs. The 802.11 MAC is very similar in concept to 802.3 and uses CSMA/CA to avoid collisions by using explicit packet acknowledgment (ACK) [Geie03].

In the network layer, IP and ad hoc routing protocols are utilized to delivery data packets to destinations by choosing the suitable the path. The packets are routed according to the routing information that is encapsulated in the packet header between IP header and UDP header. In the application layer, the voice data are encapsulated by IP/UDP/RTP. RTP can provide packet sequence number and timestamp which are necessary for stream packets.

Consider transmitting a packet from source to destination in ad hoc networks shown Figure 4.1. Source encapsulates a voice packet with IP address of router and routing information and transmits the packet by the physical layer to router. The router extracts routing information from the packet in the network layer and encapsulates a new IP address (destination/next hop) and retransmits in the physical layer. Destination finally receives the voice packet from the physical layer and obtains the voice data in the application layer.

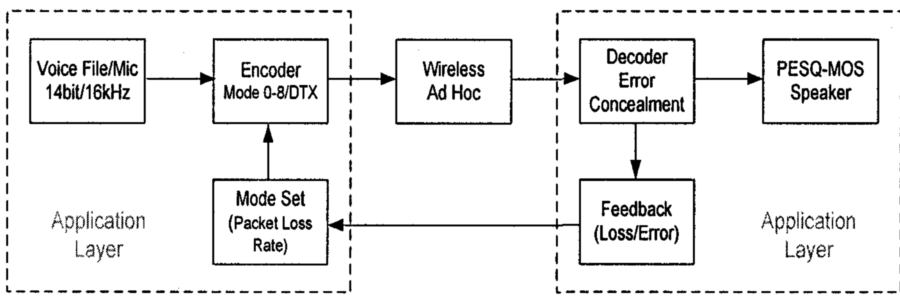


Figure 4.2: The Principle of the Testbed

4.1.2 Testbed Architecture

Figure 4.2 show the implementation of testbed for VoIP with ASNC. The encoding and mode setting are employed in the application layer. First, the input signal sampled with 16 kHz and 14 bit uniform PCM is encoded into AMR-WB coding with 9 modes (with bit rates from 6.6 to 23.85 kbps). After that, voice data are encapsulated by IP/UDP/RTP and ad hoc routing protocols in the network layer and then are transmitted over ad hoc networks. The intermediate nodes receive the packet and add a new IP address extracting from the packet and retransmit again in the network layer.

Finally, the destination receives the packet. These voice packets are decoded into PCM format by an AMR-WB decoder in the application layer. The decoder detects whether the packet loss occurs according to sequence number and timeout and packet errors occur according to CRC (cyclic redundancy check). If these happen, the error concealment mechanism is used to recovery them by inserting suitable data. The PCM voice is played to evaluate the quality of speech perceptively. PESQ is used to evaluate the quality of speech by comparing degrade files with original ones to get speech MOS value.

When packet loss or error occurs, the destination feeds back error information to the sender. The sender collects this information and calculates the packet loss rate to decide a suitable encoding mode (bit rate). The scheme also employs DTX (Discontinuous Transmission) operation according to the characteristics of sources. If the lowest rate still can not satisfy the performance, we can transmit more length packet for example 40 ms to reduce the overhead traffic at cost of increasing delay. Therefore, the method will ensure the quality of voice.

4.2 Hardware Configuration

The hardware is set up to provide a flexible implementation to test voice over IP and to verify the efficacy of the proposed adaptive coding control approach in our ad hoc voice transport testbed. Since ad hoc networks consist of many peep-to-peer computers with wireless card, we choose several commonly-used IBM laptops as network nodes. The IBM T40 and R40 laptops are used in our testbed because of their high performance with small size and light weight that make them good candidates for our testbed since we have to move the laptops to test the mobile cases. Each laptop is equipped with a Cisco PCMCIA wireless card set in ad hoc mode. All the wireless cards are set with the same network name and mask so that they can be interconnected in one ad hoc subnet without an infrastructure (e.g. using an access point). Since the internal wireless cards do not have driver software, Cisco Aironet 350 Series PCMCIA card is used to test because it can provide card drivers for Linux and its transmission power can be adjusted. It can provide six levels of adjustable transmission power: 1 mW, 5 mW, 20 mW, 30 mW, 50 mW and 100 mW. In our experiments, we will choose the lowest transmission power level so that we can test different scenarios in as small a range as possible.

4.3 Software Setup

The system software setup and tool used in our testbed system include the adaptive card configuration, UDP packet generation, MOS measurement and VoIP application programs.

4.3.1 Test-bed Software

We chose Red Hat Linux as the operation system because there is plenty of open-source software in the Linux platform. Aironet Client Utility is a GUI tool used to configure the wireless card. We first use it to load the suitable version firmware of wireless adaptive card. (version must be 4.25 rather than 5.6; otherwise, the wireless card can not work properly). In the profile manager menu, we configure the transmission power to 1 mW and configure the link speed to 11 Mbps using channel 1. Each laptop is designated the same network name called SSID. We also use the "iwconfig" to configure the card.

We write application software AMR-WB over IP with an adaptive control scheme. The AMR-WB coding algorithm uses ITU-U source code. We also implement an adaptive rate control scheme into it. All source code will be introduced in detail in the next section. Our routing protocol software is also based on the Linux operating system, which is developed by Tianjin University, China. It can provide DSR and MSR ad hoc routing algorithms. We also use the packet generator called D-ITG to generate background traffic with the VoIP stream. The commands ITGSend and ITGRecv are used to generate and to receive the UDP packets. In order to measure the quality of voice, we use ITU P.862 software to get the MOS value of voice stream. Our analysis tool is Ethereal to analyze protocol and packet content. Thus, we can generate, capture and measure the quality of VoIP stream.

4.3.2 VoIP Software

VoIP testbed software is the main software for the measurement and consists of several modules: sound recording/playback, AMR-WB coding, socket communication, and control schemes implementation. They are described in the following sections.

4.3.2.1 Sound Recording/Playback

In order to implement sound recording and playback, we first OPEN the sound card file and then configure the sound card using the ICNT command. Then we use either FREAD to read via a sound device to obtain recording data, or FWRITE to write sound date to drive the sound card play sound. Finally, we CLOSE the sound device. FREAD and FWRITE are blocking commands which require them to wait until the sound card finishes collecting data. They can be used to control the time of reading speech data from sound cards. Therefore they can be used for timing when we use a voice file as sending sources. The pseudo codes for opening, configuration, recording and playing sound are attached in following.

```
//open sound device
fd = open("/dev/dsp", O_RDWR);
//configure sound card parameter
arg = SIZE,CHANNELS,RATE;
ioctl(fd, SOUND_PCM_WRITE_RATE, &arg);
//recoding sound and playing
read(fd, signal, sizeof(signal));
// Playback the decoded stream
write(fd, synth, sizeof(synth));
// close the sound device*/
fclose(fd).
```

4.3.2.2. AMR-WB coding

AMR-WB coding provided by ITU has two kinds of modes: ITU and 3GPP mode. We use 3GGP mode because it can provide discontinuous transmission operation and bad frame indication. We compile coding and decoding codes as a static library so that other programs can call it up conveniently. This library is also used in our OPNET simulation program. In addition, we also call error concealment function when time-out for receiving occurs. The encoder, decoder functions and PESQ are declared in following.

```
//AMR CODER signal is compressed into stream
amrcoder(FILE *f_serial, short *signal,short mode, short *stream,int frame)
//AMR DECODER cpprsm is decompressed into synth
amrdecoder(FILE *f_serial, FILE *f_synth, short cpprms, short *synth, int
frame)
//PESQ measuring MOS comparing degraded file with reference file
pesqmain(char ref_file[], char deg_file[], long frequency);
```

4.3.2.3. Socket Communication

TCP/IP protocol is used to realize communication of nodes. Since the packet data structure of the layered model is too complex to implement, socket function provided by Linux is used to encapsulate data to TCP/UDP packets and realize the node communication. Figure 4.3 shows a typical diagram of the sequence of socket call for the UDP and TCP mode. At the client node, we first open a socket descriptor, and then CONNECT the server address and port. At last, we can use WRITER or READ to exchange UDP packet. At the server, we first

<u>Client</u> (UDP)		**<u>Server</u>**
socket ( )		socket ( )
	bind ( )	
sendto ( )	------------------→	recvfrom ( )
recvfrom ( )	←------------------	sendto ( )
close ( )		close ( )

<u>Client</u> (TCP)		**<u>Server</u>**
socket ( )		socket ( )
bind ( )		bind ( )
	listen ( )	
connect	←------------------→	accept ( )
	(connection negotiation)	
write ( )	------------------→	read ( )
read ( )	←------------------	write ( )
close( )		close( )

Figure 4.3: Communication by UDP and TCP between Two Computers

open a socket descriptor, and then BIND the local address and port. Finally, we use RECVFROM and SENDTO to exchange UDP packets to that port. For a TCP socket, first, a socket descriptor is opened, and then the server BINDs the well-known port. After that it LISTENs to the expected clients. Finally ACCEPT puts the server to sleep until the client uses CONNECT to establish TCP connection. READ and WRITE are used for data communication. We attach the UDP pseudo codes in following.

```
 //client configuration
 bzero(&servaddr, sizeof(servaddr));
 Set .sin_family = AF_INET;
     .sin_port = SERV_PORT;
 inet_pton(AF_INET, serv_IP, &servaddr.sin_addr)
 //open socket and attach destination.
 sockfd = socket(AF_INET, SOCK_DGRAM, 0);
 connect(sockfd, (struct sockaddr *)&servaddr, servlen);
 //send packets to server
  write(sockfd,streamcp, 2*frmno*(cplen+hdlen));

//server configuration
//init servaddr parameter defination
  bzero(&servaddr, sizeof(servaddr));
  Set .sin_family = AF_INET;
      .sin_addr.s_addr =INADDR_ANY;
      .sin_port = SERV_PORT;
  sockfd = socket(AF_INET, SOCK_DGRAM, 0);
//bind address and port to socket
  bind(sockfd, (struct sockaddr *)&servaddr, sizeof(servaddr));
//Waiting for receive data
  recvfrom(sockfd, cpprms,lin, 0, (struct sockaddr *)&cliaddr, &len);
```

4.4 Test Scenarios

Three test scenarios are used to validate the effectiveness of our proposed adaptive source-network control scheme when compared with the constant rate source coding. These test scenarios are the most basic types and can be conducted easily. The multiple-hop scenario is used to test the characteristics in different hops. The bottleneck scenario is used to test the capacity of single node transmission. Finally, the multiple-path scenario is used to test the ability of multiple path routing. All test scenarios are conducted in both static and mobile network as well as both indoor and outdoor. We first run our test without any control mechanism, and then repeat with the adaptive source-network rate control mechanism. MOS, packet loss rate and delay are measured in all scenarios. The routing protocols DSR and MSR are used to test single path and multiple paths. The rate control scheme updates the transmission rate every one second.

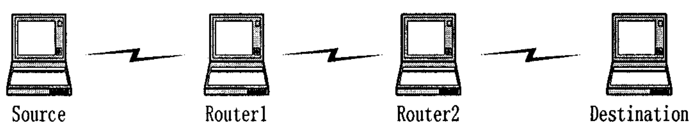


Figure 4.4: Multiple Hops Scenario

4.4.1 Scenario 1: Multiple Hops

Packets on ad hoc wireless networks need be relayed by intermediate nodes, which will save the sending node's power and reduce interference. This is at the expense of additional bandwidth, power and processing capacity of the intermediate nodes. In the multiple-hop scenario, we will test the relationship between throughput and the number of hops, and the maximum throughput for different hops. We will generate voice transport traffic and test the packet loss and packet delay. First, we test the performance of multiple-hops in static scenarios, and then fix the source and intermediate nodes and move destination node back and forth between the destination and source node positions shown in Figure 4.4. The node moving speed is 1.5 m/s. We will test the performance in difference the number of hops, and the performance of the ad hoc routing update.

4.4.2 Scenario 2: Bottleneck

The bottleneck scenario is to show the effect of a node with saturated packet forwarding capacity between source and destination. We will configure two sources, two destination and one intermediate node as shown in Fig. 4.5.

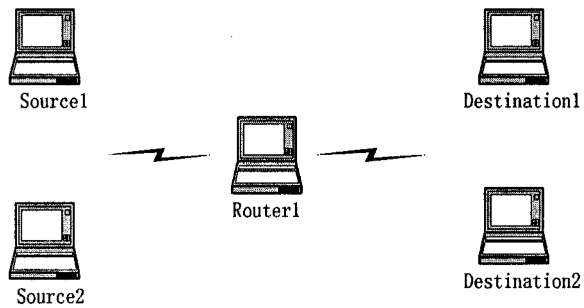


Figure 4.5: Bottleneck Scenario

Source 1 sends a voice stream to Destination 1 and Source 2 sends background traffic to destination 2 simultaneously. They all share wireless channel consisting of the intermediate node Router1 regarded as bottleneck node that moves around a circle at speed 1.5 m/s like Scenario 3.

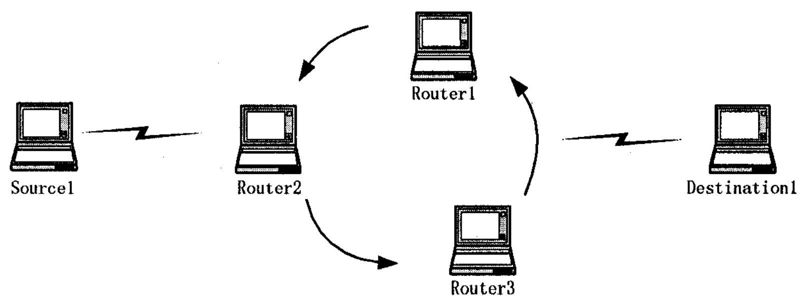


Figure 4.6: Multiple Path Scenarios

4.4.3 Scenario 3: Multiple Paths

Multiple path routing can be used to improve the stability although total throughput is sacrificed. In this scenario as shown in Figure 4.6, Source-1 communicates with Destination-1 via intermediate nodes Router1, Router2 and Router3 which are moving in a circle between the destination and source nodes. The experiment objective is to test the performance of the adaptive control scheme in multiple path routing and performance of route updating. We fix the source and the destination and move the intermediate nodes. The positions of the intermediate nodes change frequently which make the delay of the links between the intermediate nodes and the source or destination change frequently. Moving speed is set at 1.5 m/s.

4.5 General System Parameters

The performance of control schemes and routing protocols are analyzed in this section. We analyze the factors that affect the performance of wireless networks and speech coding in different environments: static and mobile, indoor and outdoor.

4.5.1 Static and Mobile

The propagation of radio waves has different characteristics in mobile and static environment. In the static communication, the system can take advantage of the very well-defined, time-invariant nature of the propagation channel between the transmitter and the receiver. In ad hoc networks, static systems do not need to do routing update frequently, so the wireless link is more stable. The signal strength of the static system does not change rapidly, that will benefit the reception of the receiver.

In the mobile environment, the distance between the transmitter and the receiver, the reflection, diffraction and obstruction of various kinds of objects will cause large-scale path loss and fading. These factors will cause the variation of signal strength with the position and time. They also cause the random frequency modulation due to the Doppler shift and time dispersion caused by the propagation delay. All these factors lead to the quality deterioration, while static model does not have the above issue

4.5.2 Indoor and Outdoor Environment

The propagation of indoor and outdoor radio waves has different characteristics. Indoor propagation conditions are more variable than outdoor, and affected mainly by blocking, reflection, diffraction and scattering. The range of wireless indoor is limited due to blocking of the walls and reflection causing the multiple-path fading. In addition, 802.11b uses the 2.4 GHz ISM (Industry Scientific Medical) frequency band, so there exists interference more than outdoor (such as microwave oven).

In an outdoor environment, the factor affecting wireless transmission that should be considered is terrain profile. Obstacles such as buildings and trees also must be taken into account. These factors cause reflection, diffraction and scattering. Open space outdoor has no blocking and so it has a large range. The signal strength is related to antenna height and the distance between antennas. The received power according to the ground reflection model [Rapp02] is

$$P_r = P_t\, G_t\, G_r\, (h_t\, h_r)^2 / d^4,$$

where P_t *and* G_t are the transmitting power and transmitting gain respectively, G_r is receiving gain. The parameters h_t and h_r are respectively antenna height of transmitter and receiver, and parameter d denotes the distance between the transmitter and the receiver. If h_t *and* h_r are equal, then $P_r = P_t\, G_t\, G_r\, (h/d)^4$, which suggests that the received power falls off with distance raised to the fourth power of distance, and is in direct proportion to the height of the antenna. Therefore, not only the distance but also the height is important for determining the quality of the wireless signal.

4.6 Performance Measures

In order to evaluate the performance of voice over ad hoc networks, we define several performance measures used in our evaluation.

1) Throughput: this is the total number of bytes of all packets sent successfully in one time unit. The FTP is used to measure this parameter by getting the file from the FTP server.
2) Packet loss rate: this is the ratio of the total number of the packets dropped before reaching the destination to the total number of packets sent from for a given measurement period. We will use the D-ITG packet generator software to measure this parameter.
3) End-to-end Delay: this is the time duration from the time instant when the first bit of a packet is sent out until the last bit is received at the receiver.
4) Round Trip Time (RTT) is the time duration from the instant a signal is sent to a remote destination and back a measuring the delay needs to be strictly synchronized.
4) Jitter: this is the variance in delay. It is potentially more disruptive for VoIP than the pure delays.
5) MOS (Mean Opinion Score): this is a five-point standard and is evaluated by the Perceptual Evaluation Speech Quality (PESQ) algorithm described in Section 2.7.

In testbed measurements, RTT and jitter are measured by PING. Synchronization is usually required in delay measurement.

We have integrated the PESQ in our VoIP program to measure the quality of voice MOS.

Two streams are used to test the quality of voice. One is background traffic, and the other is a voice stream. We send background traffic with a voice packet size of 60 bytes and increase the packet rate gradually. At the same time, the voice packets are transmitted along the same path and are used to test the quality of voice. In addition, PING is used to test delay because it is hard to synchronize a source and a destination and the Return Trip Time (RTT) can be used to represent delay approximately. FTP is used to test throughput. D-ITG packet generation program is used to generate UDP traffic and measure packet loss. This program can send UDP/TCP packet at certain rate and packet size and conduct statistic analysis of packet loss rate and throughput at the receiving end. In our test, we will fix the packet size and increase the packet rate and then test packet loss rate, throughput and delay.

4.7 Initial Testing and Verification

In this section, we implement initial testing and verification to verify the routing protocols and coding schemes that are main factors to determine the quality of voice.

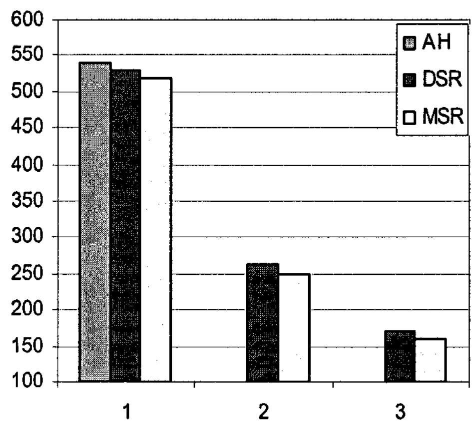
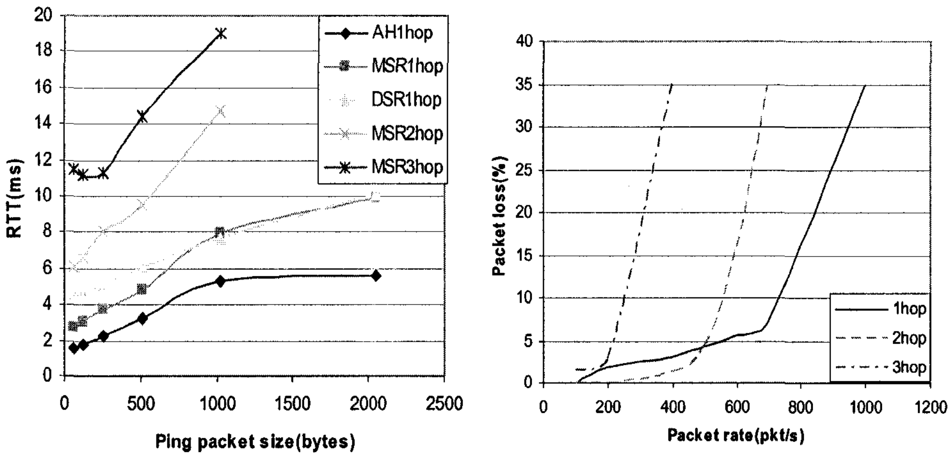


Figure 4.7: Throughput for Different Protocols

4.7.1 Routing Protocols Verification

First we measure different routing protocols in different hops. We measure throughput with different routing protocols in static indoor environment as shown in Figure 4.7. From the results of testing, we can see that throughput is a decreasing function of the number of hops. The scenario without Ad Hoc routing protocol (AH) has a larger throughput than others in one-hop case since it uses a simple scheme. DSR has better throughput than the MSR because MSR has to combine two path data. However, the MSR uses multiple links to avoid the packet drop caused by one link disconnection. In the experiment, we find that DSR is not stable because it freezes intermediate nodes. MSR is more stable in this respect.



(a) (b)

Figure 4.8: Performance using Different Number of Hops (a) RTT, b) packet loss

4.7.2 Performance Testing in Different Hops

We measure the RTT and packet loss rate for different hops in static indoor environments. RTT is measured by using a command PING with packet size from 64, 128, 256, 512, 1024 bytes without ping intervals. Figure 4.8a shows that the indoor performance of delay and jitter increases with the increment of PING packet size since the larger packet size can lead to more congestion and causes more delay and jitter. We also find that the RTT increases with the number of hops increasing since the large number of hops bring more delay. The figure also shows that the RTT without Ad Hoc protocol (AH) is the smallest while RTT of MSR is largest in one hop networks.

Figure 4.8b shows the performance of packet loss in different number of hops. The packet generator D-ITG is used to generate packets and measure the loss rate. The graph shows that the packet loss rate increases with the increment of the number of hops since the hops reduce the bandwidth of the wireless channel. We also can see the packet loss rate increases when the traffic packet rate increases since more traffic can lead to more congestion and packet loss in a wireless channel.

4.7.3 AMR-WB Testing Results.

In this section, we measure the performance of AMR-WB as shown in Figure 4.9a. We can see that the coding quality for both non-DTX(CNT) and DTX is a increasing function of the coding bit rate. The discontinuous transmission coding (DTX) reduces the bit rate and also produces the quality loss of MOS 0.1 to 0.15 comparing with non-DTX (CNT). However, the reduction of the bit rate outweighs the quality loss. Figure 4.9b shows that coding quality of speech is a decreasing function of the packet loss rate for different coding modes.

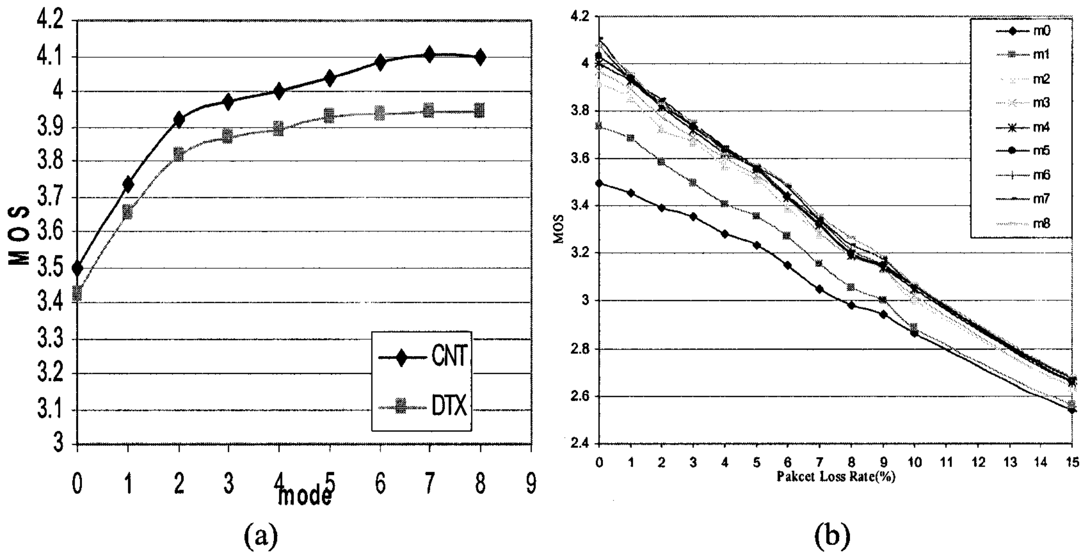


Figure 4.9: Relationship between Bit Rate, a) Packet Loss Rate and b) MOS

Consider the bottom curve (using coding Mode-0). One observes that MOS decreases linearly approximately as the packet loss rate increases. Similar observations can be made for different coding modes but at larger slopes. The higher the mode/bit rate, the deeper the slope is. Compared with other speech codecs in Table 3.2, AMR-WB has better quality and low bit rate as well as wideband feature.

4.8 Concluding Remarks

In this chapter, we discussed the architecture of the testbed as well as its hardware and software configuration. We designed three scenarios and defined performance measures to be used in our testbed measurements. We also tested and verified that the throughput of ad hoc network reduces when the number of hops increases for protocols DSR and MSR and speech coding schemes AMR has better quality than other coding schemes.

Chapter Five
QoS Measurement of VoIP on Static Ad-Hoc Networks

After implementing the testbed, we will now have an in-depth study of voice over ad hoc networks by making direct tests on the efficiency of our routing and coding schemes. In this chapter, we test the performance of static ad-hoc networks in both indoor and outdoor environment.

The performance of voice transmission using the proposed adaptive source-network rate control scheme is evaluated in terms of packet loss rate, delay, jitter and throughput. In our experiments, RTT and the variance of RTT are used to represent delay and jitter. We use these parameters and the Mean Opinion Score (MOS) to indicate the quality of transmission. Unless specified, every data point is the mean of 3 measurements.

5.1 Indoor Measurements

We shall present and discuss performance under various static indoor scenarios described in Section 4.5. The Constant Bit Rate (CBR) and the Adaptive Source-Network rate Control scheme (ASNC) are used. Each hop is separated by a blocking wall. We send a voice stream and a background stream traffic simultaneously at each source node and receive them at the destination node. A VoIP program is used to generate speech streams and a packet generator D-ITG is used to generate background traffic. We use FTP on different file sizes to measure throughput. RTT and jitter are measured by using PING command with a packet size of 8 bytes and an interval of 0.5 second. The packet loss rate and speech quality are measured using the VoIP program.

5.1.1 Throughput

Figure 5.1 presents the throughput performance of the static ad hoc network using both DSR and MSR. We use Scenarios 2 and 3 which are 2-hop networks and Scenario 1 which is a 3-hop network in order to compare with the one-hop networks. The results show that the throughput of the network decreases when the number of hops increases. The throughput of one-hop networks is the highest and that of Scenario 1 is the lowest. Because each node shares the same wireless broadcast channel and relay nodes also consume

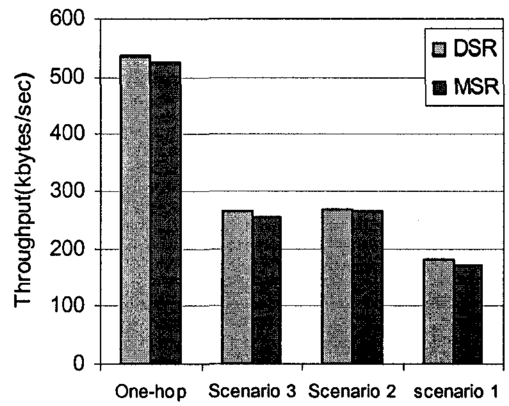
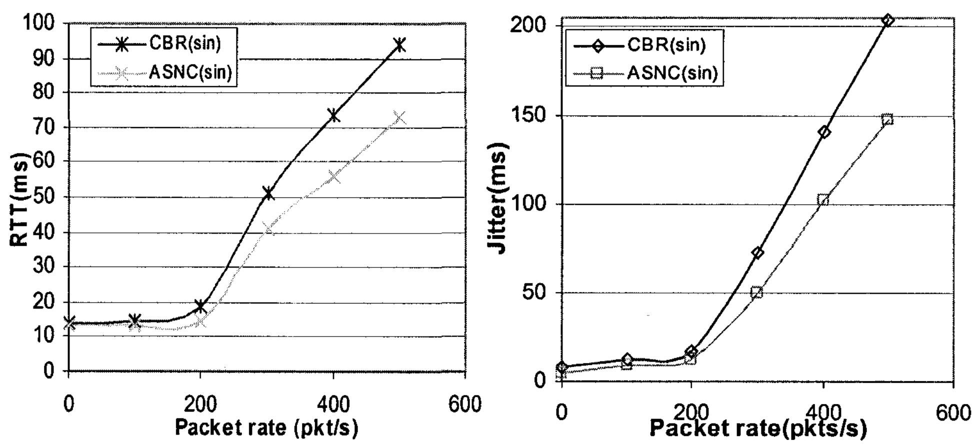


Figure 5.1: Throughput of Indoor Environments

bandwidth to forward packets, the more hops, the less the available bandwidth is. Although about the same, the DSR has a slightly better performance than MSR because MSR is a more complex algorithm. We will study MSR in the remaining experiments because we found that the intermediate nodes in DSR are unstable and can be frozen easily.

5.1.2 Delay and Jitter

We now measure the delay (RTT) and its jitter performance of speech transmission with a background traffic. MSR is used in an indoor static network. We have also compared the RTT among different routing protocols. There have been reported in Section 4.7.1 as an initial test and will not be repeated here.

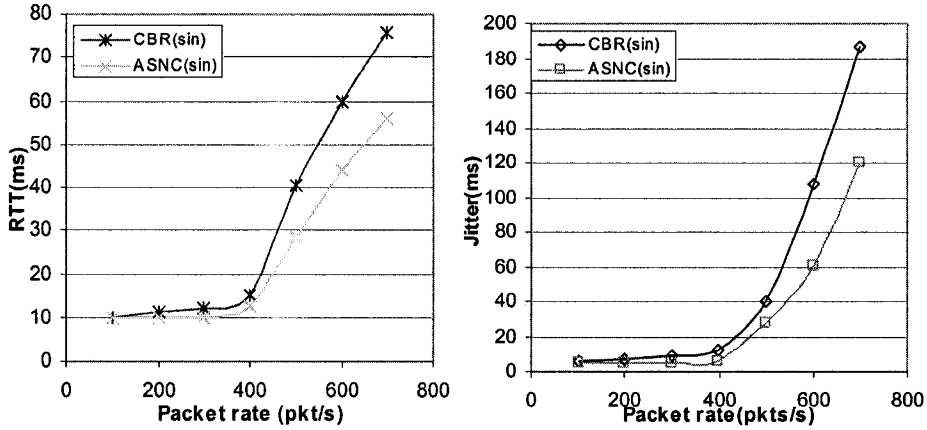


(a) RTT (b) Jitter

Figure 5.2: RTT and Jitter Performance of Scenario 1

5.1.2.1 Scenario 1

Figure 5.2 shows that RTT and jitter increase with increasing traffic. The traffic is increased by using the larger packets which leads to more congestion and increases delay and jitter. The figures also illustrate that the ASNC has a better performance (ASNC(sin)) in RTT and jitter than those (CBR(sin)) of CBR because ASNC is capable to reduce traffic through adaptation. This is especially the case when the packet rate is beyond 200 packets/sec. For example, at 400 packets/sec, ASNC has an RTT of 54 ms and a jitter of 102 ms, compared with 70 ms and 142 ms for CBR.

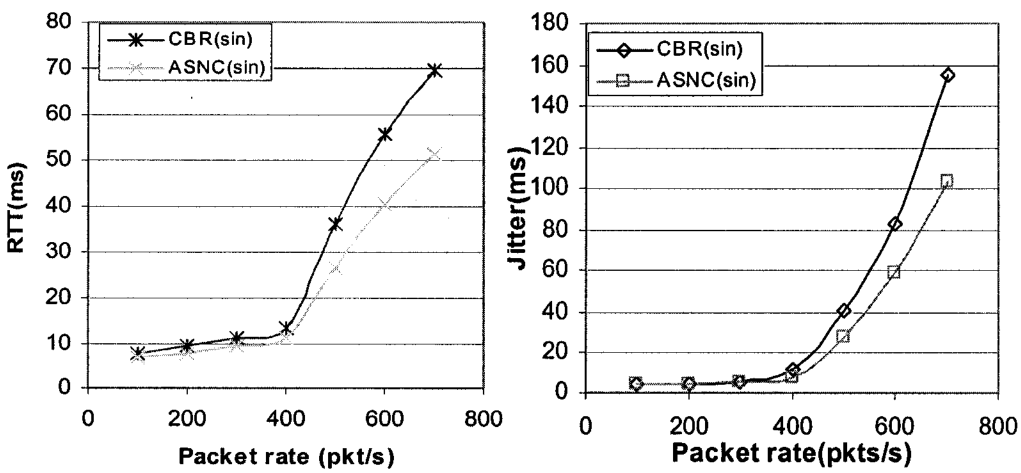


(a) RTT (b) Jitter

Figure 5.3: RTT and Jitter Performance of Scenario 2

5.1.2.2 Scenario 2

Figure 5.3 compares the delay and jitter performance of CBR and ASNC in Scenario 2. Observations similar to Figure 5.2 can be made. However, both RTT and jitter are much reduced (about half) because Scenario 2 has fewer hops than Scenario 1.



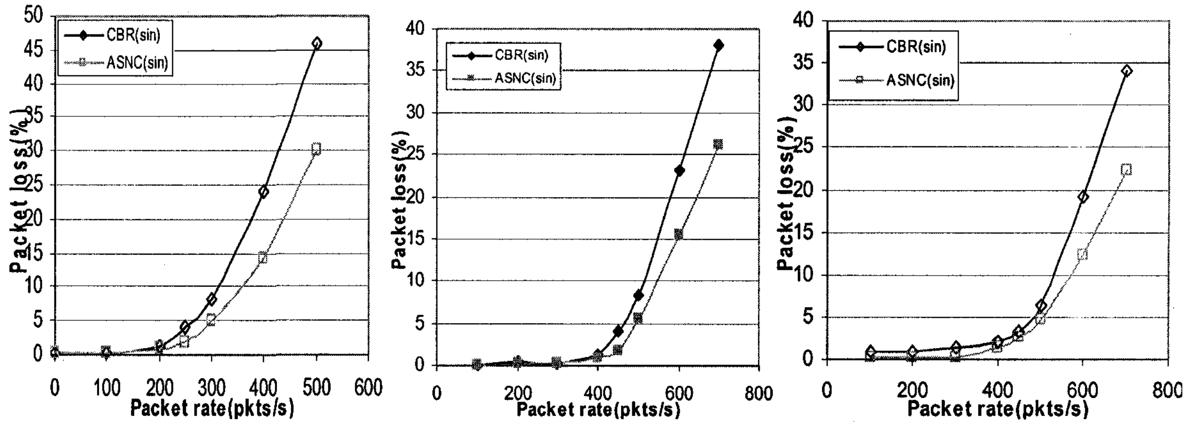
(a) RTT (b) Jitter

Figure 5.4: RTT and Jitter Performance of Scenario 3

5.1.2.3 Scenario 3

Figure 5.4 compares the delay and jitter performance of CBR and ASNC in Scenario 3. Observations similar to Figure 5.2 can be made. However, both RTT and jitter are also much reduced (about half) because Scenario 3 has fewer hops than Scenario 1.

By comparing the three scenarios, scenario 1 has the largest RTT and jitter because it has more hops (3 hops) than the others. Scenario 3 has the smallest RTT and jitter because Scenario 3 has fewer hops and utilizes multi-path.



(a) Scenario 1 (b) Scenario 2 (c) Scenario 3

Figure 5.5: Packet Loss Performance

5.1.3 Packet Loss Rate

Figure5.5 presents the packet loss rate performance under the three scenarios described in Section 4.5 using the MSR operation. CBR is compared to ASNC in each scenario.

As seen in Scenario 1 (Figure 5.5a), the packet loss rate under CBR (CBR(sin)) is increasing exponentially with regard to the packet arrival rate. The performance of ASNC is similar but much reduced. For example, the loss rate is 15% at 400 packets/sec as opposed to 24% for CBR. Similar performance of CBR and ASNC and their comparison are observed under Scenario 2 (Figure 5.5b) and Scenario 3 (Figure 5.5c), but the loss rate is decreasing. For example, at 500 packets/sce, ASNC loss rate is reduced from 30% in Scenario 1 to 6% in Scenario 2 and eventually to 5% in Scenario 3. This is because Scenario 1 has more hops than Scenario 2 and Scenario 3.

5.1.4 Speech Quality

Figure 5.6 presents the speech quality performance under the three scenarios described in

Section 4.5 using the MSR operation. CBR is compared to ASNC in each scenario.

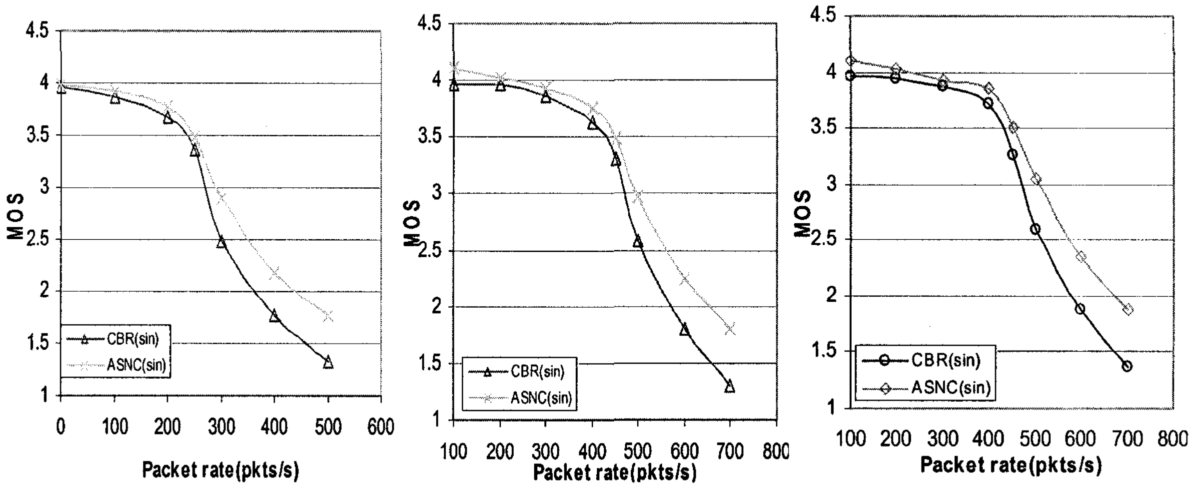


Figure 5.6: Speech Quality (a) Scenario 1 (b) Scenario 2 (c) Scenario 3

As seen in Scenario 1 (Figure 5.6a), the MOS under CBR (CBR(sin)) is decreasing with regard to traffic packet rate. The performance of ASNC is similar but more increased because ASNC is capable to reduce traffic. For example, the MOS is 2.9 at 300 packets/sec is opposed to 2.5 for CBR. Similar performance of CBR and ASNC and their comparison are observed under Scenario 2 (Figure 5.6b) and Scenario 3 (Figure 5.6c), but the MOS is increasing. For example, at 500 packets/sce, ASNC MOS is increased from 1.8 in Scenario 1 to 3.0 in Scenario 2 and eventually to 3.1 in Scenario 3. This is because Scenario 2 and 3 has fewer hops than Scenario 1.

5.2 Outdoor Measurements

We shall measure performance in static outdoor environment. We repeat the same experiments under the environment as discussed in Section 5.1 and compare the performance between indoor and outdoor environments in next section.

5.2.1 Throughput

Figure 5.7 presents the throughput performance of the static ad hoc network for both DSR and MSR. We use Scenarios 2 and 3 which are 2-hop networks and Scenario 1 which is a 3-hop network in order to compare with one-hop networks. The results show that the throughput of the network decreases when the number of hops increases. The throughput of

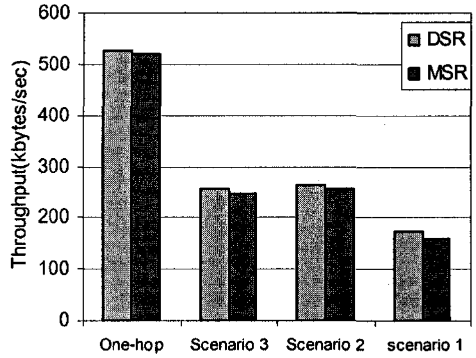
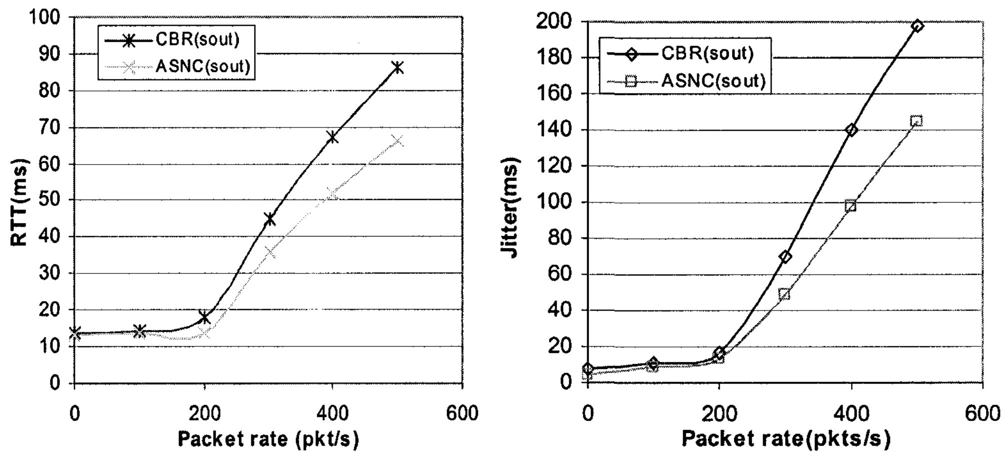


Figure 5.7: Throughput Performances in Outdoor Environments

one-hop networks is the highest and that of Scenario 1 is the lowest. Because each node shares the same wireless broadcast channel and relay nodes also consume bandwidth to forward packets, the more hops, the less the available bandwidth is. Although about the same, the DSR has a slightly better performance than MSR because MSR is a more complex algorithm. We will study MSR in the remaining experiments because we found that the intermediate nodes in DSR are unstable and can be frozen easily.



(a) RTT (b) Jitter

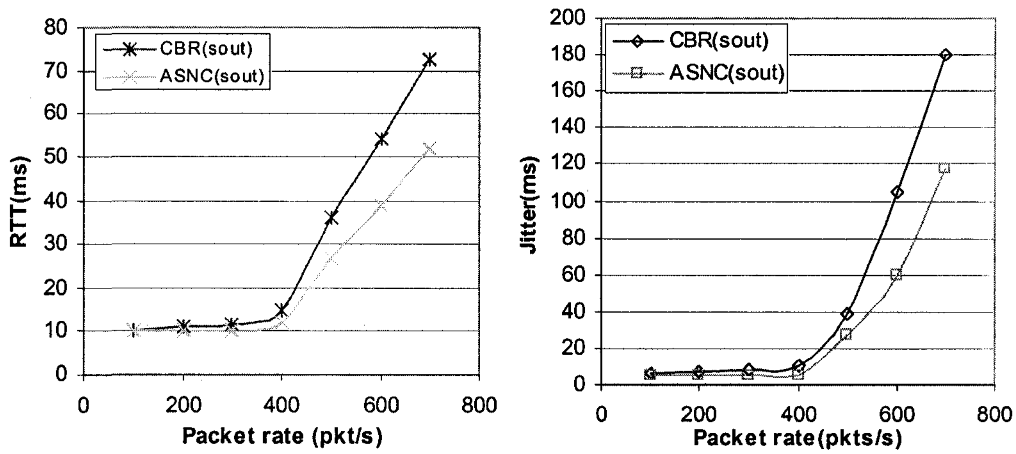
Figure 5.8: RTT and Jitter Performance of Scenario 1

5.2.2 Delay and Jitter

We now measure the delay (RTT) and its jitter performance of speech transmission with a background traffic. MSR is used in an outdoor static network.

5.2.2.1 Scenario 1

Figure 5.8 shows that RTT and jitter increase with the increasing traffic. The traffic is increased by using the larger packets which leads to more congestion and increases delay and jitter. The figures also illustrate that the ASNC has a better performance (ASNC(sout)) in RTT and jitter than those (CBR(sout)) of CBR because ASNC is capable to reduce traffic through adaptation. This is especially the case when packet rate is beyond 200 packets/sec. For example, at 400 packets/sec, ASNC has an RTT of 52 ms and a jitter of 99 ms, compared with 68 ms and 140 ms for CBR.

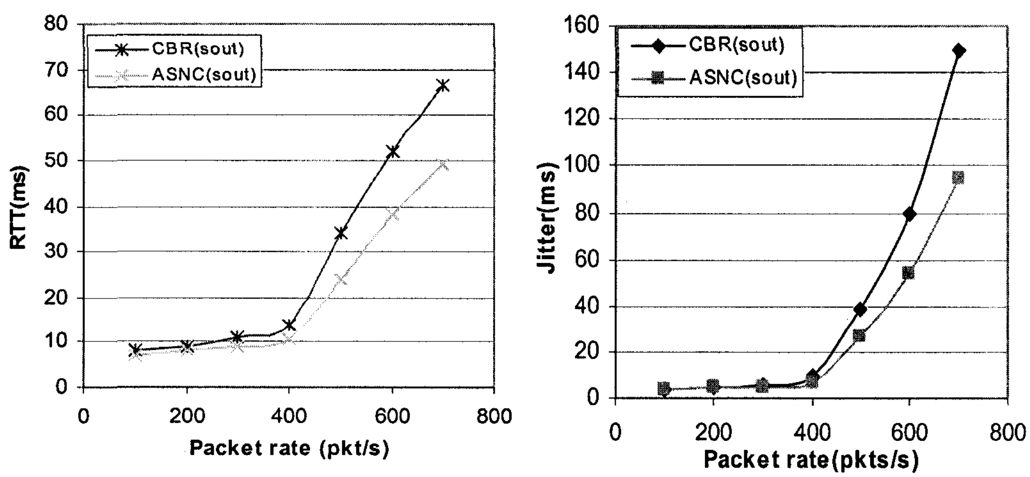


(a) RTT (b)Jitter

Figure 5.9: RTT and Jitter Performance of Scenario 2

5.2.2.2 Scenario 2

Figure 5.9 compares the delay and jitter performance of CBR and ASNC in Scenario 2. Observations similar to Figure 5.8 can be made. However, both RTT and jitter are much reduced because Scenario 2 has fewer hops than Scenario 1.



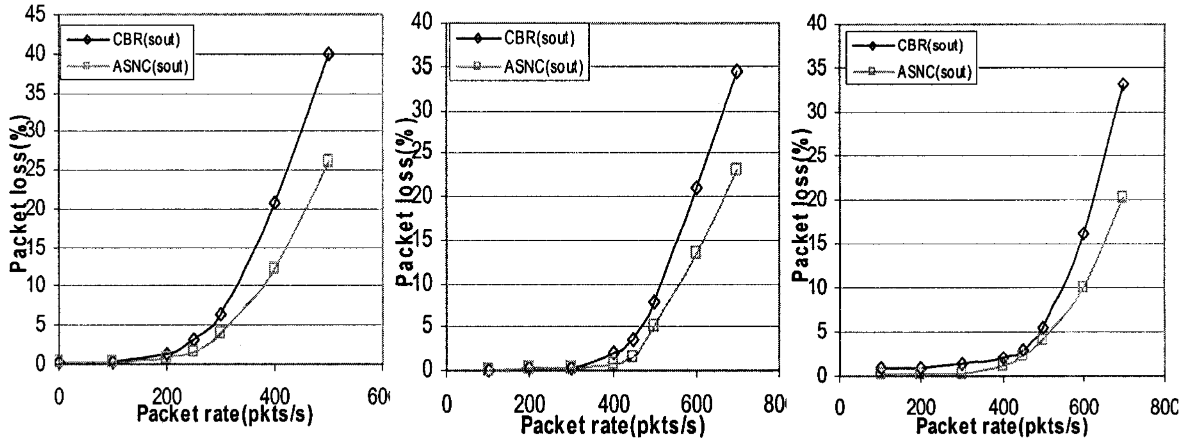
a) RTT (b) Jitter

Figure 5.10: RTT and Jitter Performance of Scenario 3

5.2.2.3 Scenario 3

Figure 5.10 compares the delay and jitter performance of CBR and ASNC in Scenario 3. Observations similar to Figure 5.8 can be made. However, both RTT and jitter are also much reduced because Scenario 3 has fewer hops than Scenario 1.

By comparing the three scenarios, scenario 1 has the largest RTT and jitter because it has more hops (3 hops) than the others. Scenario 3 has the smallest RTT and jitter because it has fewer hops and utilizes multi-path.



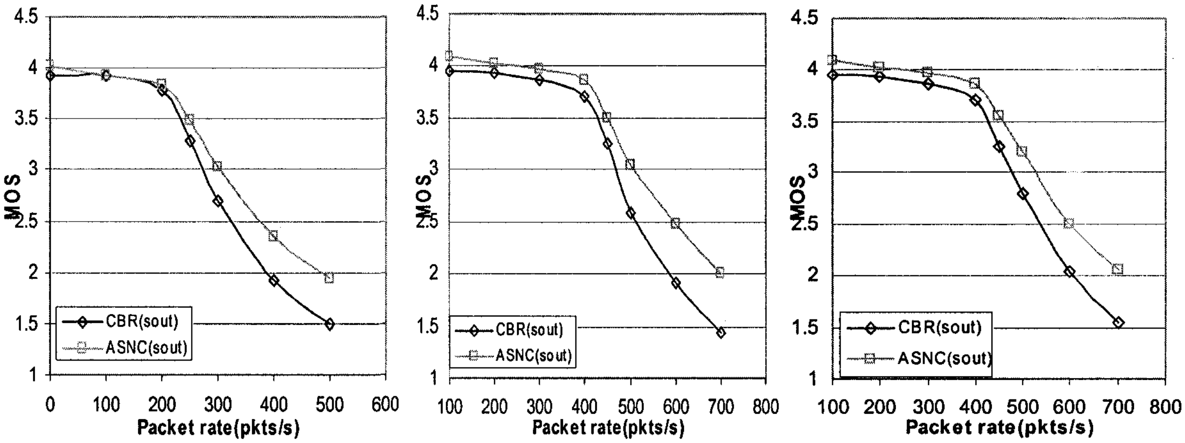
(a) Scenario 1 (b) Scenario 2 (c) Scenario 3

Figure 5.11: Packet Loss Performance

5.2.3 Packet Loss Rate

Figure 5.11 presents the packet loss rate performance under the three scenarios described in Section 4.5 using the MSR operation. CBR is compared to ASNC in each scenario.

As seen in Scenario 1 (Figure 5.11a), the packet loss rate under CBR (CBR(sout)) is increasing exponentially with regard to the packet arrival rate. The performance of ASNC is similar but much reduced. For example, the packet loss rate is 12% at 400 packets/sec as opposed to 21% for CBR. Similar performance of CBR and ASNC and their comparison are observed under Scenario 2 (Figure 5.11b) and Scenario 3 (Figure 5.11c), but the packet loss rate is decreasing. For example, at 500 packets/sec, ASNC loss rate is reduced from 26% in Scenario 1 to 5 % in Scenario 2 and eventually to 4% in Scenario 3. This is because Scenario 1 has more hops than Scenario 2 and Scenario 3.



(a) (b) (c)

Figure 5.12: Speech Quality (a) Scenario 1 (b) Scenario 2 (c) Scenario 3

5.2.4 Speech Quality

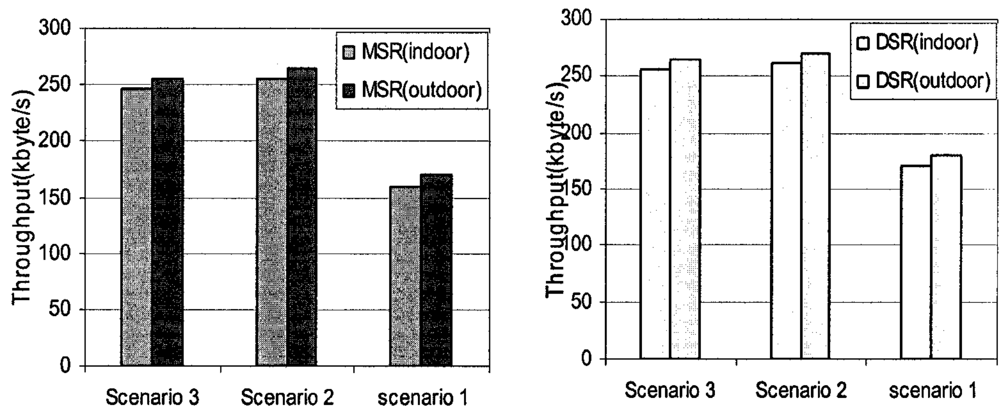
Figure 5.12 presents the speech quality performance under the three scenarios described in Section 4.5 using the MSR operation. CBR is compared to ASNC in each scenario.

As seen in Scenario 1 (Figure 5.12a), the MOS under CBR (CBR(sout)) is decreasing with regard to traffic packet rate. The performance of ASNC is similar but much increased because ASNC is capable to reduce traffic using adaptation. For example, the MOS is 3.0 at 300 packets/sec is opposed to 2.7 for CBR. Similar performance of CBR and ASNC and their comparison are observed under Scenario 2 (Figure 5.12b) and Scenario 3 (Figure 5.12c), but the MOS is much increased. For example, at 500 packets/sec, ASNC MOS is increased from 1.45 in Scenario 1 to 3.0 in Scenario 2 and eventually to 3.2 in Scenario 3. This is because Scenario 2 and 3 has fewer hops than Scenario 1.

5.3 Indoor and Outdoor Comparison

The propagation of radio waves has different characteristics indoor and outdoor. Indoor propagation conditions are more variable than outdoor, affected mainly by blocking, reflection, diffraction and scattering. The range of wireless indoor is limited due to blocking of the walls and reflection causing the multiple-path fading. In addition, 802.11b uses the 2.4 GHz ISM (Industry Scientific Medical) frequency band, so there is more interference such as microwave oven than outdoor.

In the outdoor environment, the issues above are not serious and the transmission of wireless signals is line-of-sight propagation in most time, so outdoor environment has longer transmission distance and better signal quality than indoor environments. For all these scenarios, we shall compare the indoor and outdoor performance after measuring them.



(a) MSR (b) DSR

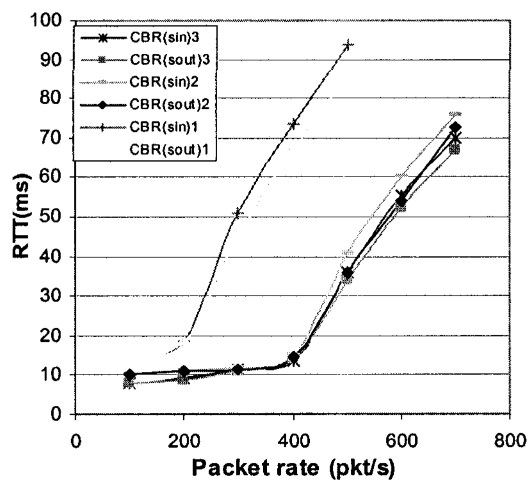
Figure 5.13: Throughput Comparisons between Indoor and Outdoor

5.3.1 Throughput

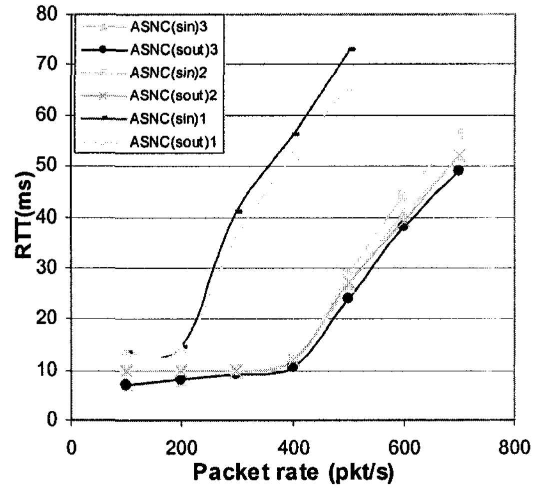
Figure 5.13 shows the indoor and outdoor comparison of throughput for MSR (Figure 5.13b) and DSR (Figure 5.13a). The graph shows that the outdoor results are better than indoor. For example, the outdoor throughput for MSR is about 10 KB/sec more than the indoor. The reason is that an outdoor environment has fewer interference signals than an indoor environment. The figure also shows that Scenario 1 has lower throughput than Scenario 2 and 3 since it has more hops than others. The throughput is increasing from 180 KB/sec to 262 KB/sec in Scenario 3 and eventually 270 KB/sec in Scenario 2.

5.3.2 RTT Comparison

Figure 5.14 shows the RTT comparison between indoor (sin) and outdoor (sout) in three scenarios. The graphs all show that the indoor RTT is larger than outdoor. For example, at 500 packets/sec for ASNC shown in Figure 5.14b, the indoor RTT is 73, 29 and 25 ms while the outdoor RTT is 65, 26 and 24 ms for Scenario 1, 2 and 3. The reason is that outdoor environment has better signal strength than indoor environments. For CBR, there are similar results. The figure also shows that RTT increase with the increasing traffic because the traffic is increased by using the larger packets which leads to more congestion and increases delay and jitter. This is especially the case beyond 200 packets/sec in Scenario 1 and 400 packets/sec in Scenario 2 and 3.

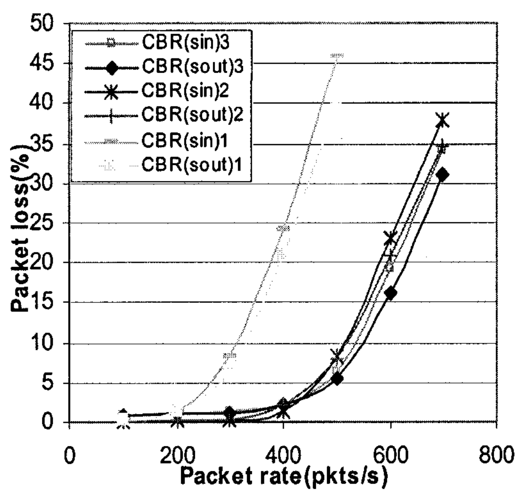


(a) CBR

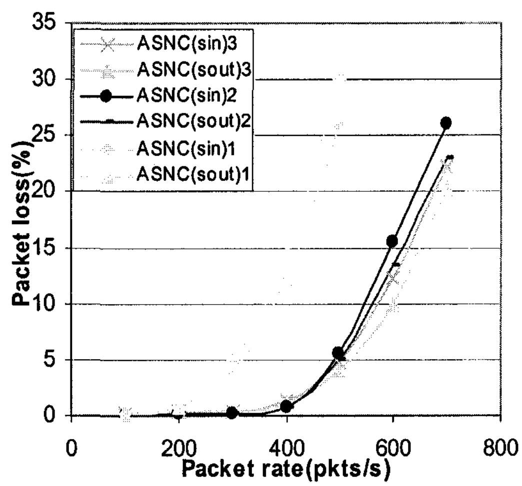


(b) ASNC

Figure 5.14: RTT Comparison between Indoor & Outdoor



(a) CBR



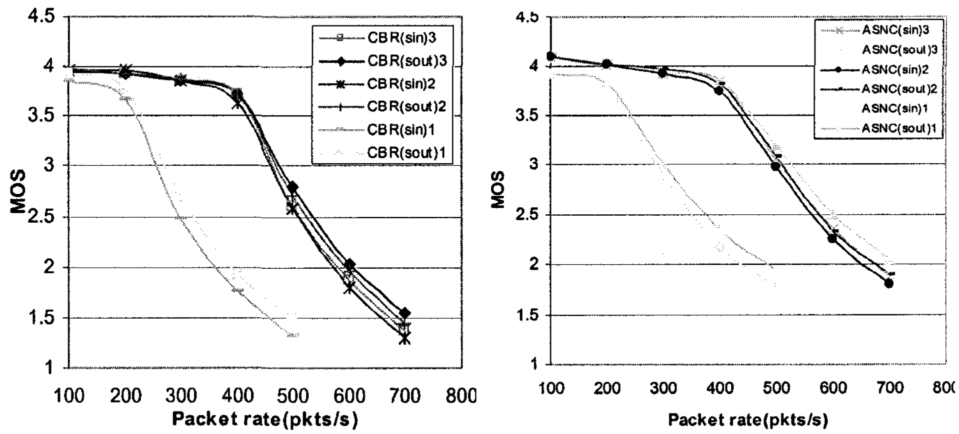
(b) ASNC

Figure 5.15: Packet Loss Comparisons between Indoor and Outdoor

5.3.3 Packet Loss Comparison

Figure 5.15 shows the packet loss rate comparison between indoor (sin) and outdoor (sout) in the three scenarios. The graphs show that the outdoor performance is better than indoor in both CBR (Figure 5.15a) and ASNC (Figure 5.15b) since the outdoor environment has better signal than the indoor environment. For example, at 500 packets/sec for ASNC shown in Figure 5.14b, indoor packet loss rate is 30%, 6% and 5% while outdoor packet loss rate is 26%, 5% and 4% ms for Scenario 1, 2 and 3. For CBR, there are similar results but the loss rate is much increased.

As seen in three scenarios, all packet loss rates under CBR and ASNC are increasing exponentially with regard to the packet arrival rate, but Scenario 1 has a more increasing packet loss rate than others. This is because Scenario 1 has more hops than Scenario 2 and Scenario 3. For example, at 500 packets/sec, ASNC loss rates reduce from 30% in Scenario 1 to 6% in Scenario 2 and eventually 5% in Scenario 3 for outdoor.



(a)CBR (b)ASNC

Figure 5.16: Speech Quality Comparison (a) CBR (b) ASNC

5.3.4 Speech Quality

Figure 5.16 shows the speech quality comparison between indoor (sin) and outdoor (sout) in both CBR and ASNC. The graphs show that the outdoor MOS is better than indoor because the outdoor environment has better signal than the indoor environment. For example, at 500 packets/sec for ASNC shown in Figure 5.16b, outdoor MOS is 1.9, 3.1 and 3.2 while indoor packet loss rate is 1.8, 3.0 and 3.1 ms for Scenario 1, 2 and 3. Similar performance of CBR is observed but MOS is worse because ASNC is capable to reduce traffic using adaptation.

We also can see the MOS under CBR and ASNC is decreasing with regard to the traffic packet rate in three scenarios, but Scenario 1 has a smaller MOS than Scenarios 1 and 2. This is because Scenario 1 has more hops than others. For example, at 500 packets/sec, ASNC MOS increases from 1.9 in Scenario 1 to 3.1 in Scenario 2 and eventually 3.2 in Scenario 3 for outdoor.

5.4 Concluding Remarks

In this chapter, we have conducted field experiments to test and compare the performance of voice over static ad hoc networks in different scenarios. Our comparison demonstrates the difference and effect of indoor and outdoor environments. The adaptive source-network rate control scheme is shown to have better performance than the CBR coding.

Chapter Six
QoS Performance Measurement of VoIP on Mobile Ad-Hoc Networks

In this chapter, we continue to study the characteristics of wireless ad hoc routing and verify the effectiveness of the proposed schemes. However, we shall test the adaptive rate control schemes in mobile ad hoc networks. As in the static environment, we measure the mobile performance measures such as throughput, delay and jitter, packet loss and MOS in both indoor and outdoor environment under three scenarios. Unless specified, every data point is also the average of 3 measurements.

6.1 Indoor Measurements

We shall discuss performance in mobile indoor environment. We follow the same setup and measurement procedure for the indoor measurement described in Section 5.1. The mobile nodes speed is 1.5 meters/sec.

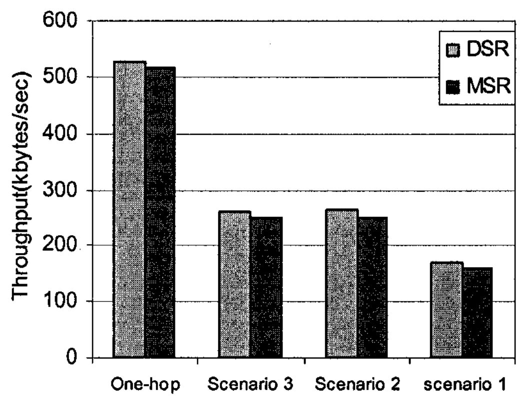


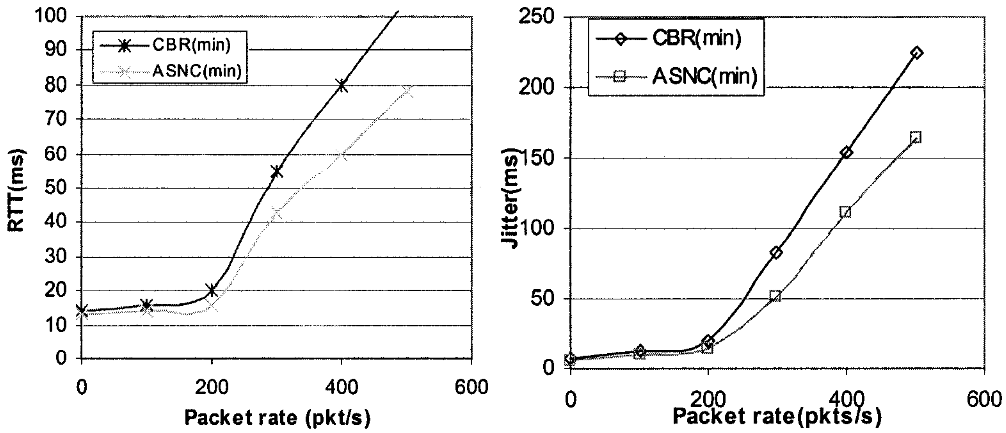
Figure 6.1: Throughput of Indoor Environments

6.1.1 Throughput

Figure 6.1 presents the throughput performance of the indoor mobile ad hoc network using both DSR and MSR. We use Scenarios 2 and 3 which are 2-hop networks and Scenario 1 which is a 3-hop network to compare with the one-hop networks. The results show that the throughput of the network decreases when the number of hops increases. The throughput of one-hop networks is the highest and that of Scenario 1 is the lowest. Because each node shares the same wireless broadcast channel and relay nodes consume bandwidth to forward packets, the more hops, the less the available bandwidth is. Although about the same, the DSR has a slightly better performance than MSR because MSR is a more complex algorithm. We will study MSR in the remaining experiments because we found that the intermediate nodes in DSR are unstable and can be frozen easily.

6.1.2 Delay and Jitter

We now measure the delay (RTT) and its jitter performance of speech transmission with a background traffic. MSR is used in an indoor mobile network.



(a) RTT (b) Jitter

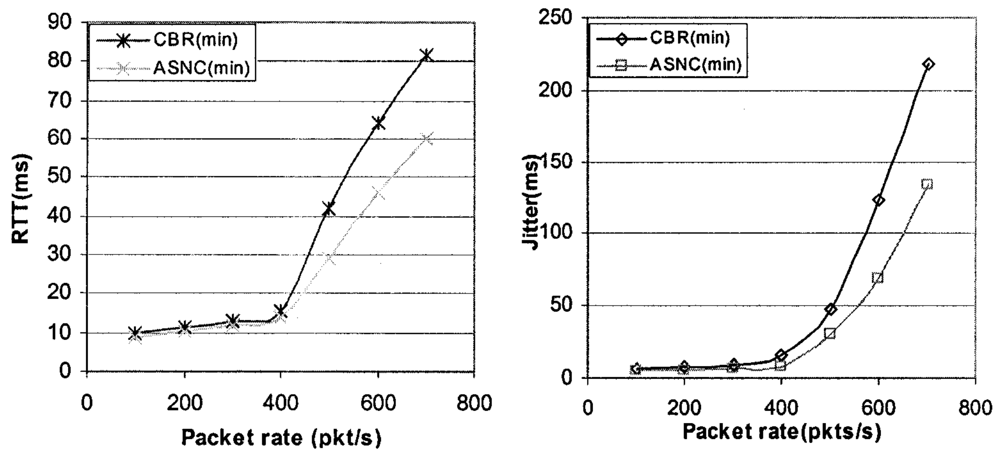
Figure 6.2: RTT & Jitter Performance of Scenario 1

6.1.2.1 Scenario 1

Figure 6.2 shows that RTT and jitter increase with the increasing traffic. The traffic is increased by using the larger packets which leads to more congestion and increases delay and jitter. The figures also illustrate that the ASNC has a better performance (ASNC(min)) in RTT and jitter than those (CBR(min)) of CBR because ASNC is capable to reduce traffic through adaptation. This is especially the case when packet rate is beyond 200 packets/sec. For example, at 400 packets/sec, ASNC has an RTT of 60 ms and a jitter of 110 ms, compared with 80 ms and 151 ms for CBR.

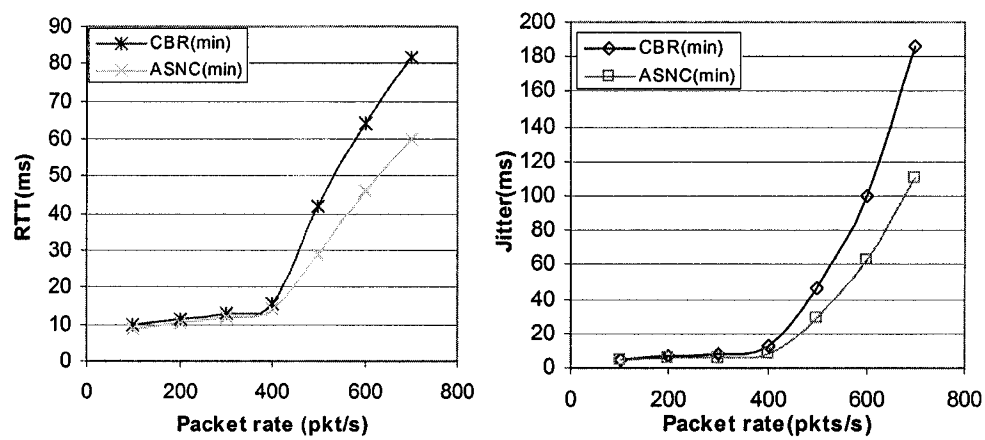
6.1.2.2 Scenario 2

Figure 6.3 compares the delay and jitter performance of CBR and ASNC in Scenario 2. Observations similar to Figure 6.2 can be made. However, both RTT and jitter are much reduced because Scenario 2 has fewer hops than Scenario 1.



(a) RTT (b) Jitter

Figure 6.3: RTT & Jitter Performance of Scenario 2



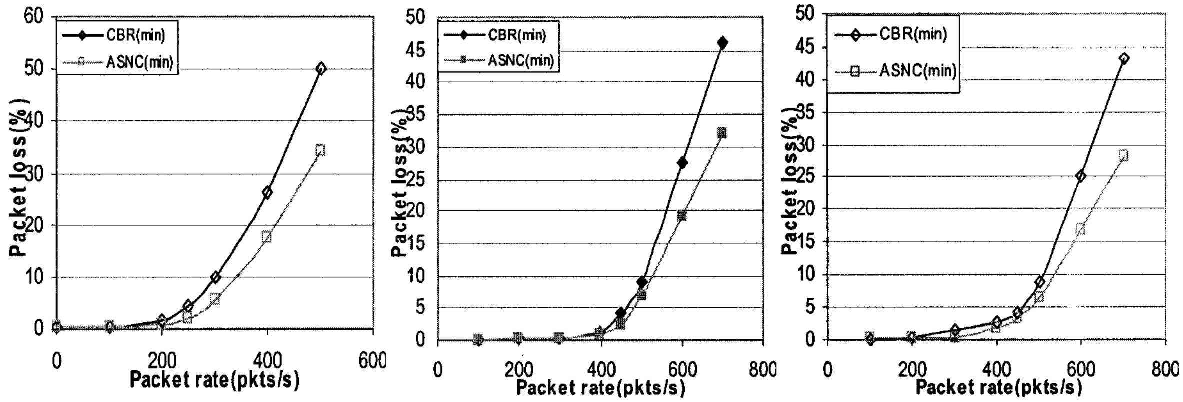
(a) RTT (b) Jitter

Figure 6.4: RTT & Jitter Performance of Scenario 3

6.1.2.3 Scenario 3

Figure 6.4 compares the delay and jitter performance of CBR and ASNC in Scenario 3. Observations similar to Figure 6.2 can be made. However, both RTT and jitter are also much reduced because Scenario 3 has fewer hops than Scenario 1.

By comparing the three scenarios, scenario 1 has the largest RTT and jitter because it has more hops (3 hops) than the others. Scenario 3 has the smallest RTT and jitter because Scenario 3 has fewer hops and utilizes multi-path.



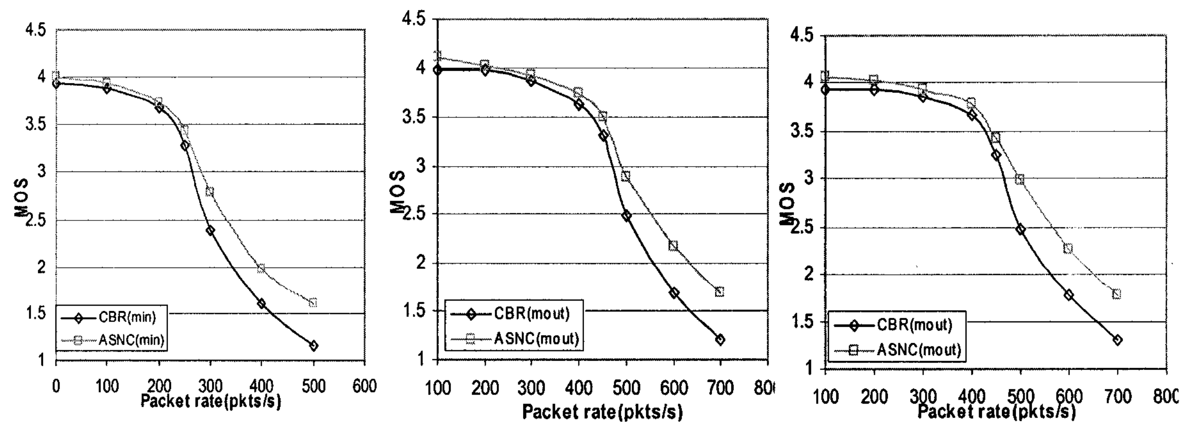
(a) Scenario 1 (b) Scenario 2 (c) Scenario 3

Figure 6.5: Packet Loss Performance

6.1.3 Packet Loss Rate

Figure 6.5 presents the packet loss rate performance under the three scenarios described in Section 4.5 using the MSR operation. CBR is compared to ASNC in each scenario.

As seen in Scenario 1 (Figure 6.5a), the packet loss rate under CBR (CBR(min)) is increasing exponentially with regard to the packet arrival rate. The performance of ASNC is similar but much reduced. For example, the loss rate is 18% at 400 packets/sec as opposed to 27% for CBR. Similar performance of CBR and ASNC and their comparison are observed under Scenario 2 (Figure 6.5b) and Scenario 3 (Figure 6.5c), but the loss rate is decreasing. For example, at 500 packets/sec, ASNC packet loss rate is reduced for 34% in Scenario 1 to 7% in Scenario 2 and eventually to 6% in Scenario 3. This is because Scenario 1 has more hops than Scenario 2 and Scenario 3.



(a) (b) (c)

Figure 6.6: Speech Quality (a) Scenario 1 (b) Scenario 2 (c) Scenario 3

6.1.4 Speech Quality

Figure 6.6 presents the speech quality performance under the three scenarios described in Section 4.5 using the MSR operation. CBR is compared to ASNC in each scenario.

As seen in Scenario 1 (Figure 6.6a), the MOS under CBR (CBR(min)) is decreasing with regard to traffic packet rate. The performance of ASNC is similar but much increased because ASNC is capable to reduce traffic. For example, the MOS is 2.8 at 300 packets/sec is opposed to 2.4 for CBR. Similar performance of CBR and ASNC and their comparison are observed under Scenario 2 (Figure 6.6b) and Scenario 3 (Figure 6.6c), but the MOS is increasing. For example, at 500 packets/sec, ASNC MOS is reduced from 1.6 in Scenario 1 to 2.9 in Scenario 2 and eventually to 3.0 in Scenario 3. This is because Scenarios 2 and 3 have fewer hops than Scenario 1.

By comparing the three scenarios, scenario 1 has the smallest MOS because it has more hops (3 hops) than the others. Scenario 3 has better MOS performance than Scenario 2 since Scenario 3 utilizes multiple paths to transmit data and increases the stability of links.

6.2 Outdoor Measurement

We shall discuss performance in mobile outdoor environment. We repeat the same experience under the same environment as described in Section 6.1. The difference is we measure in outdoor environments.

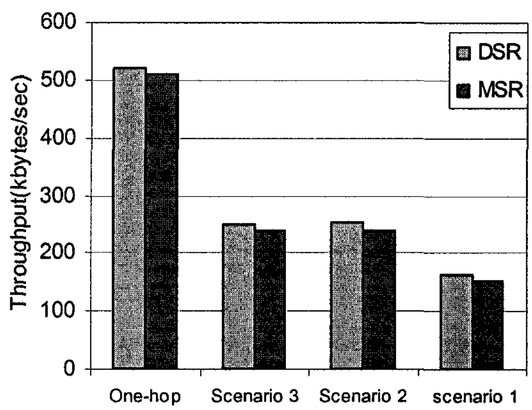


Figure 6.7: Throughput Performance in Outdoor Mobile Environment

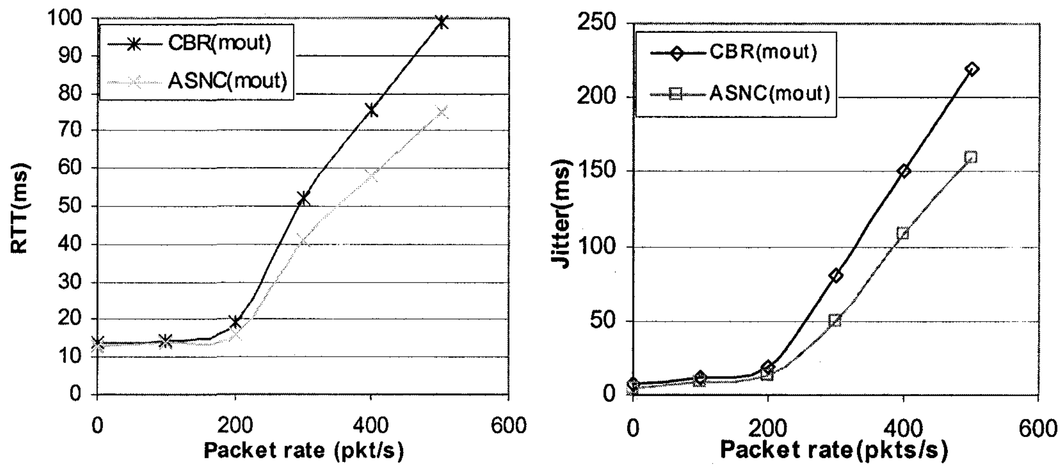
6.2.1 Throughput

Figure 6.7 presents the throughput performance of the static ad hoc network using DSR and

MSR. We use Scenarios 2 and 3 which are 2-hop networks and Scenario 1 which is a 3-hop network in order to compare with the one-hop network. The results show that the throughput of the network decreases when the number of hops increases. The throughput of one-hop networks is the highest and that of Scenario 1 is the lowest. Because each node shares the same wireless broadcast channel and relay nodes also consumes bandwidth to forward packets, the more hops, the less the available bandwidth is. Although about the same, the DSR has a slightly better performance than MSR because MSR is a more complex algorithm. We will study MSR in the remaining experiments because we found that the intermediate nodes in DSR are unstable and can be frozen easily.

6.2.2 Delay and Jitter

We now measure the delay (RTT) and its jitter performance of speech transmission with a background traffic. MSR is used in an outdoor mobile network.

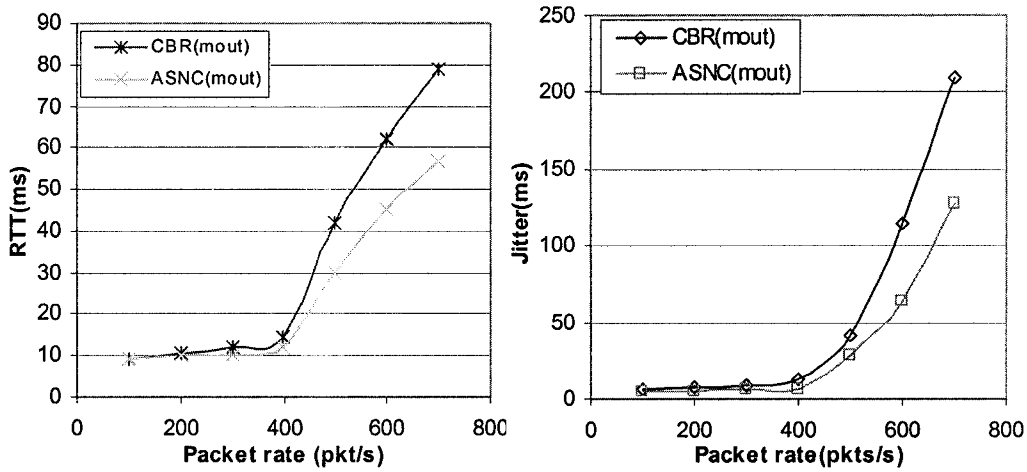


(a) RTT (b) Jitter

Figure 6.8: RTT & Jitter Performance of Scenario 1

6.2.2.1 Scenario 1

Figure 6.8 shows that RTT and jitter increase with the increasing traffic. The traffic is increased by using the larger packets which leads to more congestion and increases delay and jitter. The figures also illustrate that the ASNC has a better performance (ASNC(mout)) in RTT and jitter than those (CBR(mout)) of CBR because ASNC is capable to reduce traffic through adaptation. This is especially the case when packet rate is beyond 200 packets/sec. For example, at 400 packets/sec, ASNC has an RTT of 58 ms and a jitter of 110 ms, compared with 76 ms and 150 ms for CBR.

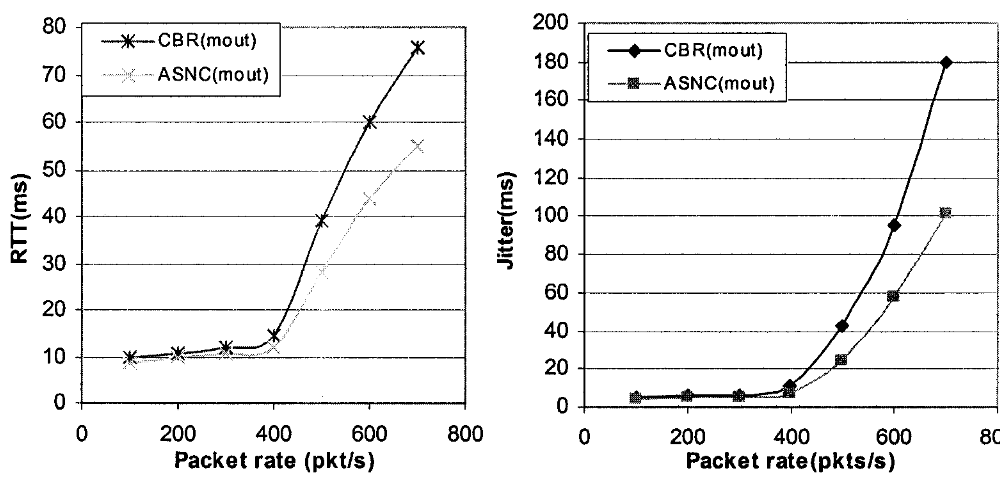


(a) RTT (b) Jitter

Figure 6.9: RTT & Jitter Performance of Scenario 2

6.2.2.2 Scenario 2

Figure 6.9 compares the delay and jitter performance of CBR and ASNC in Scenario 2. Observations similar to Figure 6.8 can be made. However, both RTT and jitter are much reduced because Scenario 2 has fewer hops than Scenario 1.



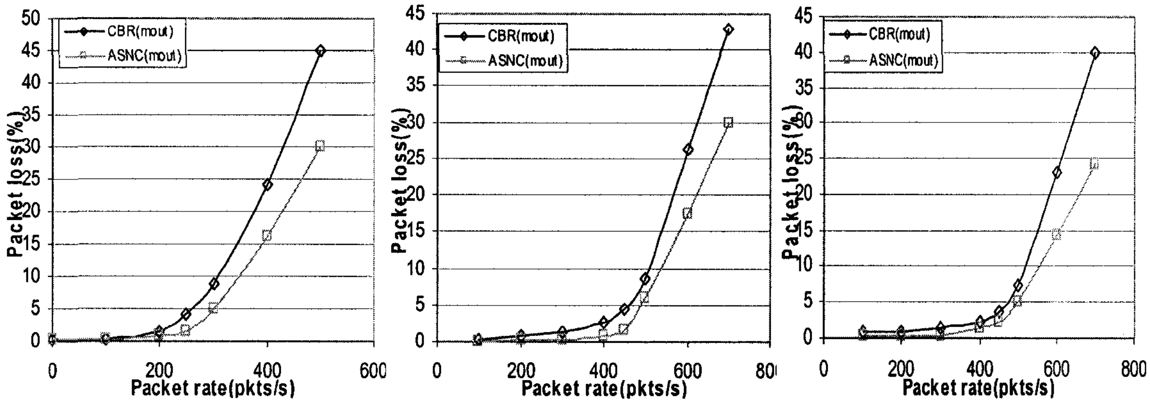
(a) RTT (b) Jitter

Figure 6.10: RTT & Jitter Performance of Scenario 3

6.2.2.3 Scenario 3

Figure 6.10 compares the delay and jitter performance of CBR and ASNC in Scenario 3. Observations similar to Figure 6.8 can be made. However, both RTT and jitter are also much reduced because Scenario 3 has fewer hops than Scenario 1.

By comparing the three scenarios, scenario 1 has the largest RTT and jitter because it has more hops (3 hops) than the others. Scenario 3 has the smallest RTT and jitter because Scenario 3 has fewer hops and utilizes multi-path.



(a)Scenario 1 (b) Scenario 2 (c) Scenario 3

Figure 6.11: Packet Loss Performance

6.2.3 Packet Loss Rate

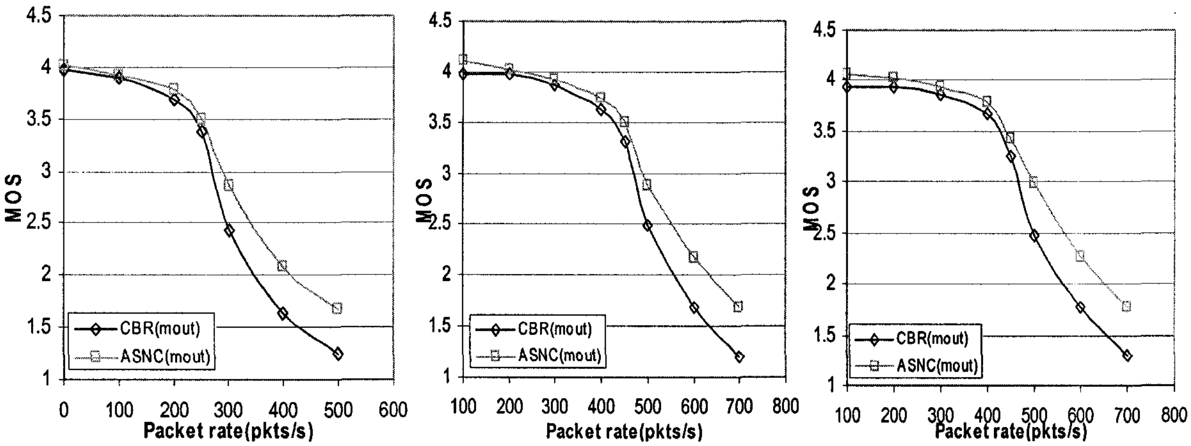
Figure 6.11 presents the packet loss rate performance under the three scenarios described in Section 4.5 using the MSR operation. CBR is compared to ASNC in each scenario.

As seen in Scenario 1(Figure 6.11a), the packet loss rate under CBR (CBR(mout)) is increasing exponentially with regard to the packet arrival rate. The performance of ASNC is similar but much reduced. For example, the loss rate is 16% at 400 packets/sec as opposed to 24% for CBR. Similar performance of CBR and ASNC and their comparison are observed under Scenario 2 (Figure 6.11b) and Scenario 3 (Figure 6.11c), but the loss rate is decreasing. For example, at 500 packets/sce, ASNC loss rate is reduced from 30% in Scenario 1 to 6% in Scenario 2 and eventually to 5% in Scenario 3. This is because Scenario 1 has more hops than Scenario 2 and Scenario 3.

6.2.4 Speech Quality

Figure 6.12 presents the speech quality performance under the three scenarios described in Section 4.5 using the MSR operation. CBR is compared to ASNC in each scenario.

As seen in Scenario 1(Figure 6.12a), the MOS under CBR (CBR(min)) is decreasing with regard to traffic packet rate. The performance of ASNC is similar but much increased because ASNC is capable to reduce traffic. For example, the MOS is 2.9 at 300 packets/sec is opposed to 2.4 for CBR. Similar performance of CBR and ASNC and their comparison



(a) (b) (c)

Figure 6.12: Speech Quality (a) Scenario 1 (b) Scenario 2 (c) Scenario 3

are observed under Scenario 2 (Figure 6.12b) and Scenario 3 (Figure 6.12c), but the MOS is better. For example, at 500 packets/sec, ASNC MOS is reduced from 1.7 in Scenario 1 to 2.9 in Scenario 2 and eventually to 3.0 in Scenario 3. This is because Scenarios 2 and 3 have fewer hops than Scenario 1.

By comparing the three scenarios, scenario 1 has the smallest MOS because it has more hops (3 hops) than the others. Scenario 3 has better MOS performance than Scenario 2 since Scenario 3 utilizes multiple paths to transmit data and increases the stability of links.

6.3 Static vs Mobile and Outdoor vs Indoor

The propagation of radio waves has different characteristics in mobile and static environment. In the mobile environment, the distance between the transmitter and the receiver, the reflection, diffraction and obstruction of various kinds of objects all cause large scale path loss and fading. These factors will cause the variation of signal strength with the position and time. They also cause the random frequency modulation due to the Doppler shift and time dispersion caused by the propagation delay. All these factors lead to the quality deterioration, while static model does not have the above issues, the static systems do not need to routing update frequently, so the wireless link is more stable.

As noted in the comparison in Section 5.3, the indoor propagation conditions are more variable than outdoor due to factors like blocking and reflection. Outdoor transmission is mostly line of sight, and therefore has a better performance. It would be interesting to see what other variations a mobile network will belong to. We will compare throughput, delay, packet loss and speech quality as following.

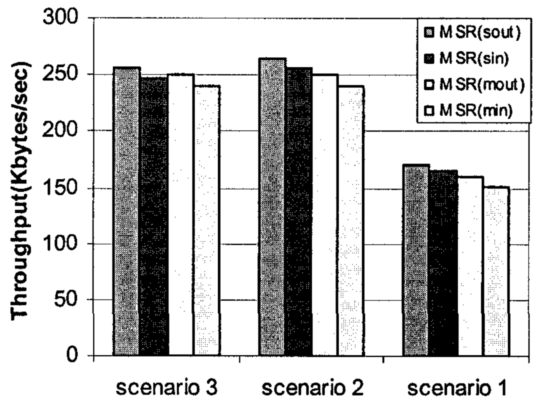


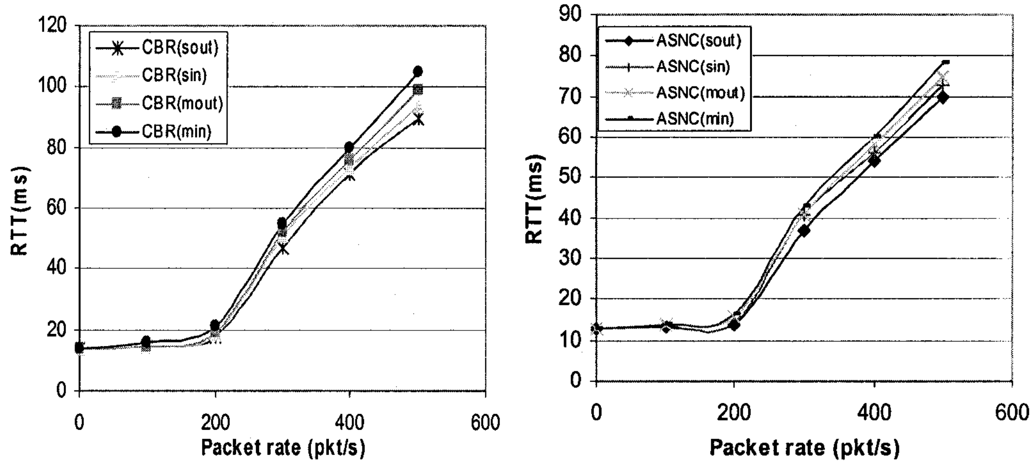
Figure 6.13: Throughput Comparison

6.3.1 Throughput Comparisons

Figure 6.13 present that the comparison of throughput using MSR in static indoor (sin) and outdoor (sout) environments, mobile indoor (min) and outdoor (mout) environments. The static outdoor scenario has achieved the best throughput while the mobile indoor scenario is the worst. The static indoor scenario has better throughput than the mobile outdoor. For example, throughput of Scenario 1 reduces from 170 KB/s in static outdoor to 165 KB/s in static indoor, 160 KB/s in mobile outdoor, and eventually 150KB/s in mobile indoor. The static indoor scenario has better throughput than the mobile outdoor one. The reason is that indoor propagation conditions are variable than outdoor and the signal of mobile environments is changeable and there is routing update. Similar performance comparison in Scenario 2 and Scenario 3, but the throughput is increasing because they have fewer hops.

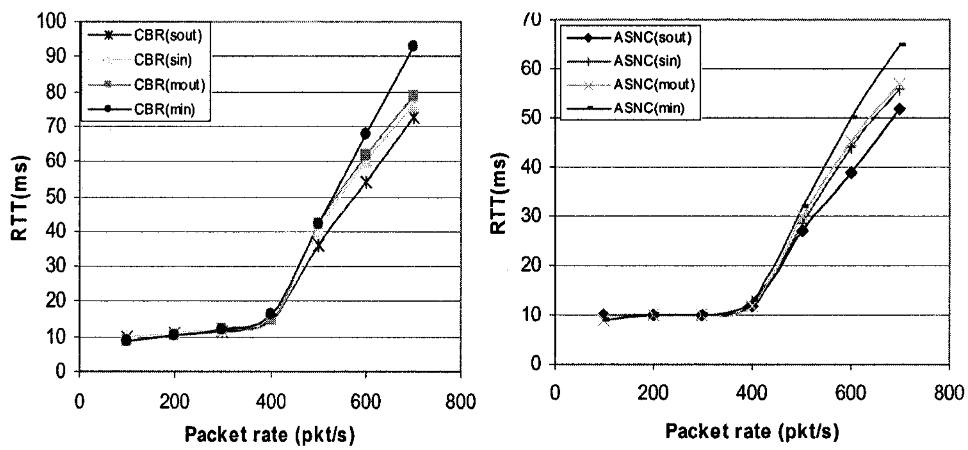
6.3.2 Delay Comparisons

Figures 6.14, 6.15 and 6.16 show the comparison of RTT in indoor and outdoor, mobile and static environments for the CBR and ASNC schemes in three scenarios. Figure 6.14a illustrates that the static outdoor of CBR (CBR(sout)) has the shortest RTT while mobile indoor (CBR(min)) has the longest RTT in Scenario 1. The static indoor (CBR(sin)) has shorter RTT than the mobile outdoor(CBR(mout)). The reason is that the outdoor propagation condition is better than indoor and the signal strength of mobile environments is



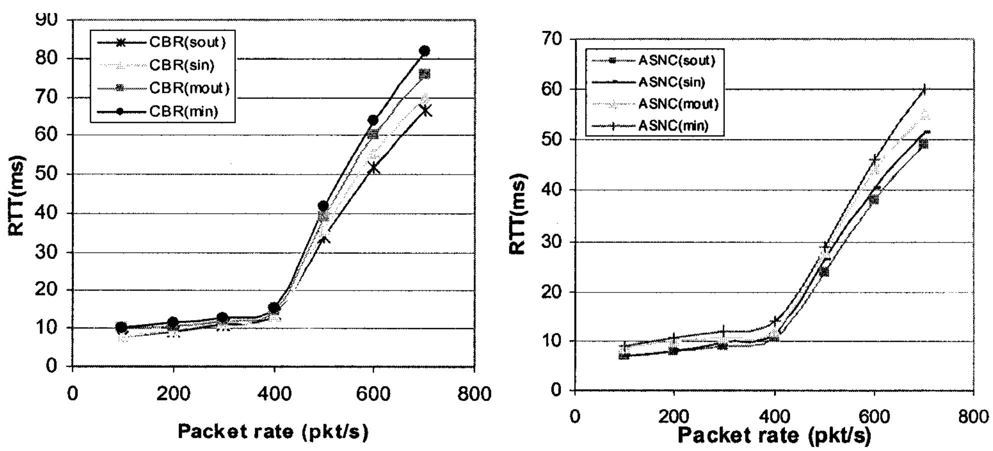
(a)CBR (b) ASNC

Figure 6.14: RTT Comparisons of Scenario 1



(a)CBR (b) ASNC

Figure 6.15: RTT Comparisons of Scenario 2

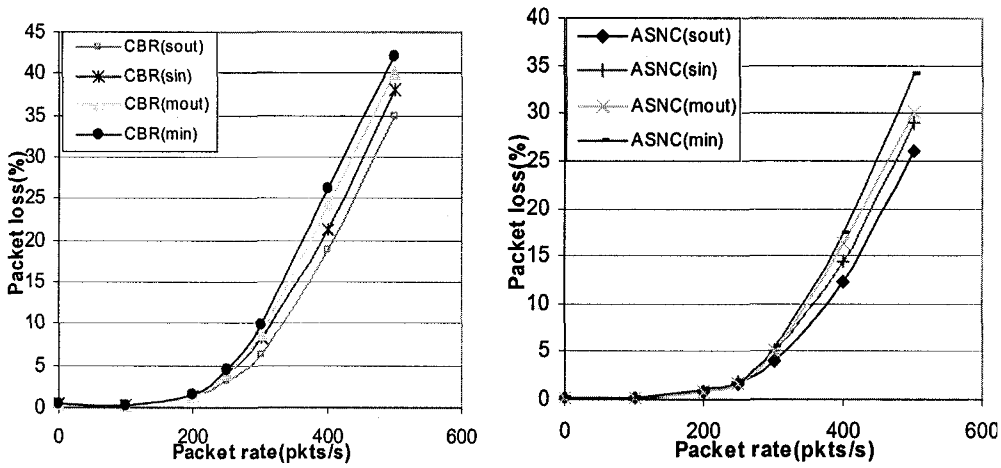


(a)CBR (b) ASNC

Figure 6.16: RTT Comparisons of Scenario 3

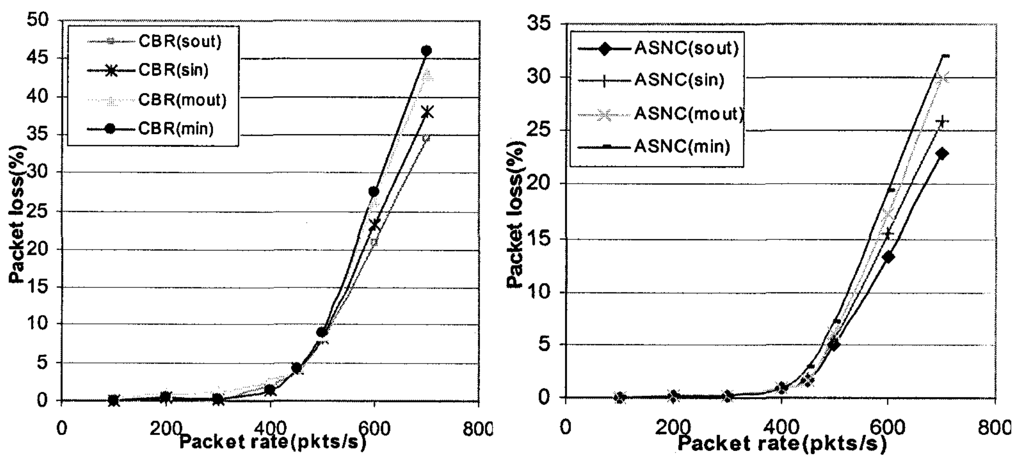
not stable due to routing update. For example, at 400 packets/sec, ASNC RTT is decreased from 80 ms of static mobile to 76% of static indoor, 73 ms of mobile outdoor and eventually to 70 ms of mobile indoor. Similar observations, comparisons and explanations can be made to ASNC in Figure 6.14b but it is much decreased because it uses the adaptive scheme to reduce traffic.

In fact, similar performance comparison of CBR and ASNC is observed under Scenario 2 (Figure 6.15) and Scenario 3 (Figure 6.16), but the RTT is much decreased since it has fewer hops. For example, at 500 packets/sec, ASNC loss rate is decreased from 70 ms in Scenario 1 to 28 ms in Scenario 2 and eventually to 24 ms in Scenario 3 for static outdoor environments. The explanations are similar to Scenario 1.



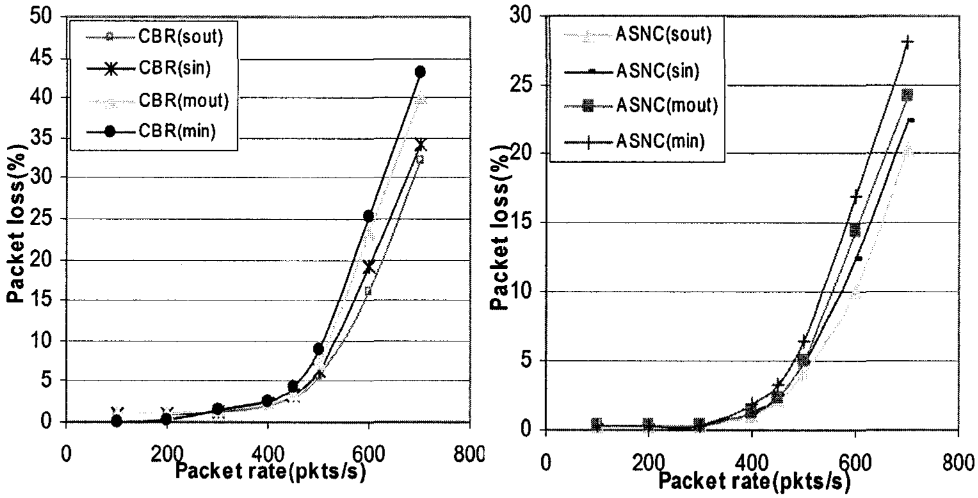
(a)CBR (b) ASNC

Figure 6.17: Packet Loss Comparisons of Scenario 1



(a)CBR (b) ASNC

Figure 6.18: Packet Loss Comparisons of Scenario 2



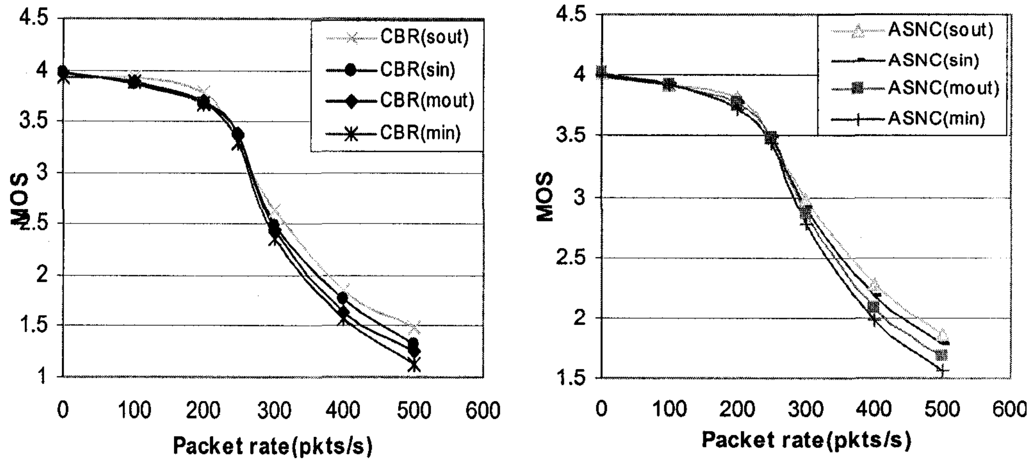
(a)CBR (b) ASNC

Figure 6.19: Packet Loss Comparisons of Scenario 3

6.3.3 Packet Loss Comparisons

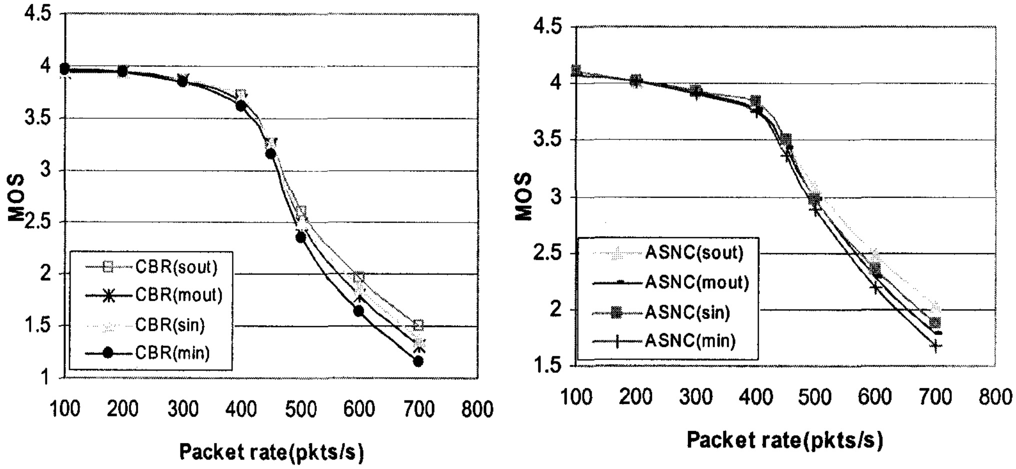
Figures 6.17, 6.18 and 6.19 shows that the comparison of packet loss rate in indoor and outdoor, mobile and static environments for the CBR and ASNC schemes in three scenarios Figure 6.17a illustrates that the static outdoor quality of CBR (CBR(sout)) is the best while mobile indoor quality (CBR(min)) is the worst in Scenario 1. The static indoor (CBR(sin)) also has lower packet loss rate than the mobile outdoor (CBR(mout)). The reason is that the outdoor propagation condition is better than indoor and the signal strength of mobile environments is not stable due to routing update. For example, For example, at 400 packets/sec, ASNC loss rate is decreased from 26% of static mobile to 24% of static indoor, 22% of mobile outdoor and eventually to 19% of mobile indoor. Similar observations, comparisons and explanations can be made to ASNC in Figure 6.17b, but it is much decreased because it uses adaptation to reduce traffic.

Similar performance comparison of CBR and ASNC is observed under Scenario 3 (Figure 6.18) and Scenario 3 (Figure 6.19), but the loss rate is much decreased since it has fewer hops. For example, at 500 packets/sec, ASNC loss rate is decreased from 26% in Scenario 1 to 23% in Scenario 2 and eventually to 20% in Scenario 3 for static outdoor environments. The explanations are similar to Scenario 1.



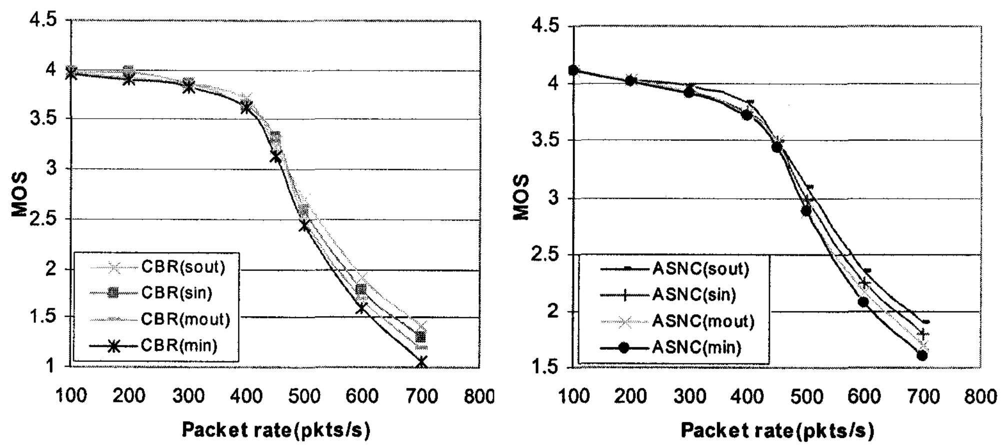
(a)CBR (b) ASNC

Figure 6.20: Speech Quality Comparisons of Scenario 1



(a)CBR (b) ASNC

Figure 6.21: Speech Quality Comparisons of Scenario 2



(a)CBR (b) ASNC

Figure 6.22: Speech Quality Comparisons of Scenario 3

6.3.4 Quality Comparisons

Figures 6.20, 6.21 and 6.22 show that the comparison of speech quality in indoor and outdoor, mobile and static environments for the CBR and ASNC schemes in three scenarios Figure 6.20a illustrates that the static outdoor quality (CBR(sout)) is the best while mobile indoor quality (CBR(sin))is the worst in Scenario 1 for CBR since the outdoor propagation condition is better than indoor and the signal strength of mobile environments is not stable and there is routing update. For example, For example, at 400 packets/sec, ASNC MOS is reduced from 1.9 of static mobile to 1.8 of static indoor, 1.7 of mobile outdoor and eventually to 1.6 of mobile indoor. This is because Scenario 2 and 3 has fewer hops than Scenario 1. Similar observations, comparisons and explanations can be made to ASNC in Figure 6.20b but it is much increasing because it uses the adaptive scheme to reduce traffic.

Similar performance comparison of CBR and ASNC is observed under Scenario 2 (Figure 6.21) and Scenario 3 (Figure 6.22), but the MOS is much increased since it has fewer hops. For example, at 500 packets/sec, ASNC MOS is increased from 1.9 in Scenario 1 to 3.1 in Scenario 2 and eventually to 3.2 in Scenario 3 for static outdoor environments. The explanations are similar to Scenario 1.

6.4 Concluding Remarks

In this chapter, we have tested the performance of voice over mobile ad hoc networks in mobile scenarios. We compared the performance of the ASNC scheme in mobile and static, indoor and outdoor and concluded that static and outdoor environment have better performance than mobile and indoor environment. We conclude that the ASNC scheme has better performance than the constant bit rate coding in various environments mentioned above.

Chapter Seven
Network Modeling, Simulation and Assumption

In this chapter, we shall discuss the models of our ad hoc networks, then operation and protocol layer. According to the operations in Section 3.2, we analyze the characteristics of the ad-hoc channel and create a mathematical ad hoc queuing model. We explain how system and interface of the network architecture are captured in the network model, node model process model and process model of our OPNET simulation.

7.1 Network Model

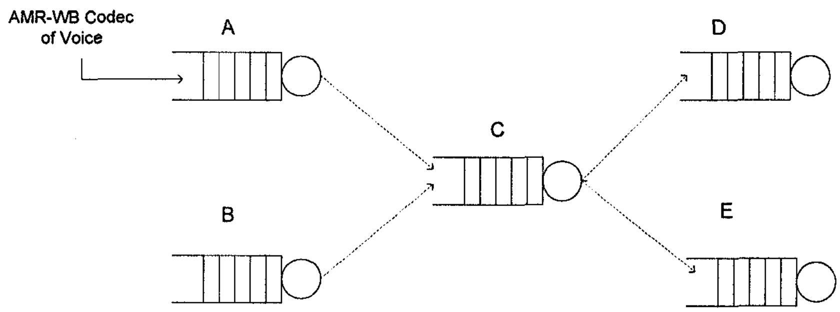


Figure 7.1: Ad Hoc Queuing Model

7.1.1 Ad Hoc Queuing Model

We consider an ad hoc network of N nodes with a maximum bit rate W. The number of links M is a random variable depending on the instant topology. For example, the topology at a given moment can resemble Scenario 2 in Figure 4.5 and therefore M=4 links as shown. This is no direct link between nodes A and D because signal of node A can not reach node D. The maximum average bit rate of each links is W/M.

The communication channel of ad hoc networks is shared by all nodes. So it is necessary to require an access mechanism. The distribution coordination function is used in ad hoc networks. In a wireless network only two nodes can communicate simultaneously since all nodes have to share channels locally with other nodes. Therefore, only one node can send information at one time and the others just wait and receive information. The multiple-path does not provide a larger bandwidth but only ensures stability of links. The multiple-hop reduces the link bit rate but provides energy saving. From the ad hoc queuing model shown in Figure 7.1, we can see if N source nodes of an n-node network with data rate W

have to send data, the bit rate of each node equals to W/N. For the ad hoc network, the N source nodes have to transfer and relay information in h hops. Therefore the bit rate of link nodes reduce to $W/(Nh)$.

Another characteristic of ad hoc networks is that the packets of wireless networks have a large overhead because of an acknowledgement and a complex physical layer used. Also, additional routing information is added in the packet header. For the voice stream, the packet size is very small (less than 60 bytes). A speech stream consumes a bandwidth R in order to support a payload rate r. Let L be the header overhead, P be the physical layer and wait timing overhead, T be the frame length of speech stream and W be the data rate of wireless networks. Then $R=r+L/T+P*W/T$. Therefore, we use the adaptive source-network rate control scheme described in Section 3.3 to reduce the bandwidth of speech stream so as to improve the quality of speech transmission.

7.1.2 Voice over Ad Hoc Model

For performance evaluation, we are interested in the end-to-end model described in Section 3.3 whose figure is repeated in Figure 7.2 below.

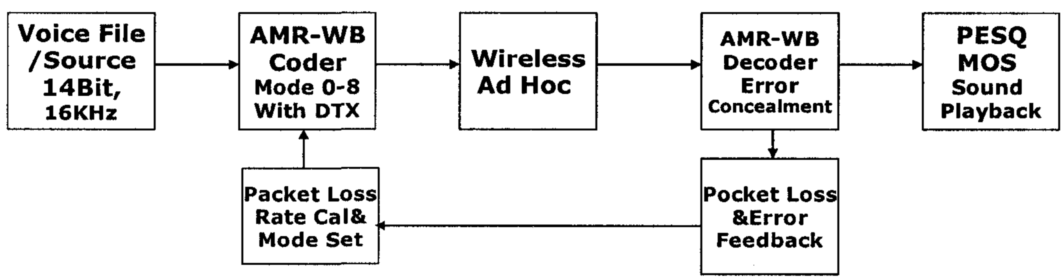


Figure 7.2: The Principle of Adaptive Rate Control Scheme

The input is the voice signal, which is recorded by "sound recorder", one of Windows Applications. The voice is converted to uniform PCM file with 16kHz sample rate and 14-bit representation. First the sending node will read the binary voice file and encode it to AMR-WB according to suitable mode and DTX indication. The coding mode and DTX are determined by the packet loss rate and previous mode. In the ad hoc system, a client and a server communicate with each other using the DSR and MSR protocol. Interference nodes produce more traffic to affect the voice transmission in ad hoc network and lead to the voice packet drop. At the server, we will compensate the drop using error concealment. The voice packets are decoded to wave format and played back. We also use the packet loss detection and CRC error check to utilize the error concealment to restore packet loss/error. We feed the error information back to the sending nodes so as to determine a suitable coding mode (mode and DTX) in the next transmission. Finally, we can evaluate the voice quality with PESQ software. In order to share the code written in Linux platform, the code is compiled into static library in Windows platform. Then the codes are called as static library in the OPNET.

7.2 Simulation Model

We choose OPNET Version 11.5 because it has provided a MANET (Mobile Ad hoc Networking) model along with several ad hoc protocols. OPNET 11.5 provides several routing protocols. We use Dynamic Source Routing (DSR) and Multipath Source Routing (MSR) which is implemented in our simulation model by modifying DSR. We use outdoor model for our simulations because there is no indoor model available from OPNET. The mobility configuration of a node is used to simulate mobile scenarios.

7.2.1 Model Implementation

Of the two kinds of MANET models provided by OPNET (Wireless LAN workstations and MANET stations), we shall use the workstation model because it has seven-layer model and can generate application traffic (The station model is only a 2-layer model but it can generate raw packets over IP over network.) [1]. When we create the network model, certain characteristics can be specified such as the geographic size of the model, the transmitter power, the routing protocols and node mobility (which includes random mobility, trajectories and direct manipulation of the node position.) Besides, nodes mobility configuration is used to generate mobile scenarios.

Each node has a packet generator for each voice stream based on a constant rate. It also has a packet receiver to receive voice and evaluate its quality. The voice transmission is

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[1] Three types of parameters are assigned to configure MANET nodes: The first type is ad hoc routing parameters. The second type is wireless LAN parameters which include transmitter power, Frequency hopping/direct sequence DSSS/FHSS, RTS/CTS, Fragmentation, Channel settings, etc. The third type is parameters used to generate raw MANET packets (packet size and interval).

affected by the channel capacity and other node interference. In addition, each node is assigned its own IP address in order to communicate with each other. We use C-programming language to implement the AWB-WB codec, VoIP with adaptive control scheme, PESQ evaluation and ad hoc routing. They are built into the OPNET platform. Finally, the wave file is compared with the original wave file to obtain transmitting quality MOS by an evaluation tool PESQ. Static libraries AMR-WB and PESQ are compiled in Windows Visual C++.

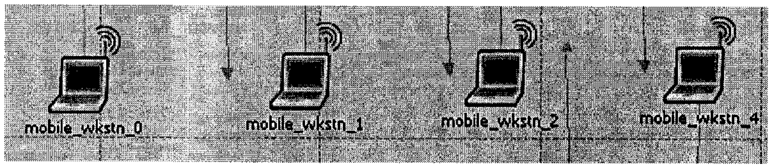
In order to test the performance over adaptive rate control and source rate control scheme, we embed the AMR-WB coding into above MANET model and use the core procedure such as op_pk_send and op_pk_get to realize the packet communication. We also add the control schemes and packet combination with discontinuous transmission into that model. Finally, we use PESQ to measure the MOS of the quality of speech.

Now, we create the network model, node model and module model and embed the code above into it. We create a model to transmit the speech packet in a pair of stations, for example the client and the server, other stations transmit packets as background traffic of their interference. OPNET Modeler provides a hierarchical modeling structure starting from Network Model on the top level, followed by further details repetitively under the Node Model, Process Model and Packet Model. They are summarized as follows.

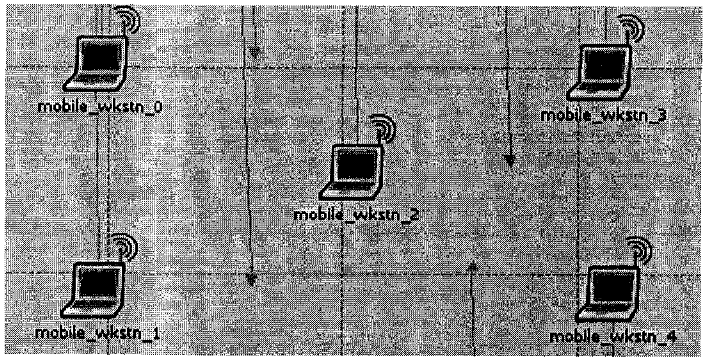
7.2.2 Network Models

The network model is shown in Figure 7.3. It is an example of the queue model described in Figure 7.1. As shown in Figure 7.3b for Scenario 2, Station 0 transmits constant rate voice stream, with some specified parameters and another Station 3 receives that voice traffic. Stations 1 and 2 just generate background inference traffic. The aim is that one station transmits voice stream and another receive it under the background traffic interference provided by the remainder stations. Station 2 is also a router to relay traffic.

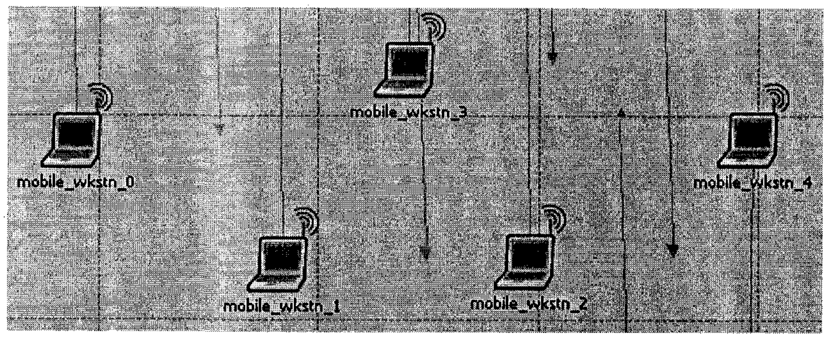
There are three network scenarios used in the simulation testing shown in Figure 7.3. Multiple hops scenario is used to test the characteristics in different hops as shown in Figure 7.3a. Bottle neck scenario is used to test the capacity of single node transmission shown in Figure 7.3b. Multiple paths scenario is used to test the ability of multiple path routing shown in Figure 7.3c. We will test the static and mobile, indoor and outdoor, and validate the effectiveness of our proposed adaptive control mechanism and fixed source coding rate and adaptive rate control scheme. The description of scenarios in detail can be seen in Section 4.5.



(a)



(b)



(c)

Figure 7.3: Network Model

7.2.3 Node Model

The node model shown in Figure 7.4 consists of a series of smaller biding blocks called modules and connections to support interaction between modules. It is particularly well suited to modeling arrangements of seven-layer communication protocols and their relationship. Here, we will simulate wireless ad hoc protocol 802.11b as shown in the following figure.

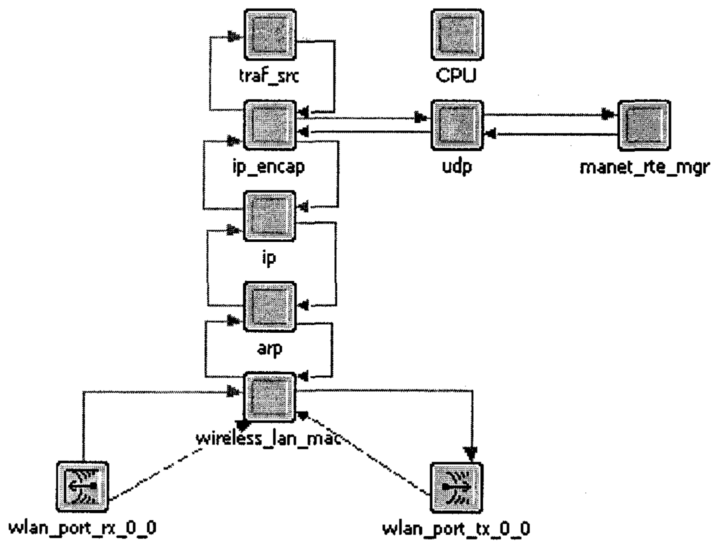


Figure 7.4: Node Model

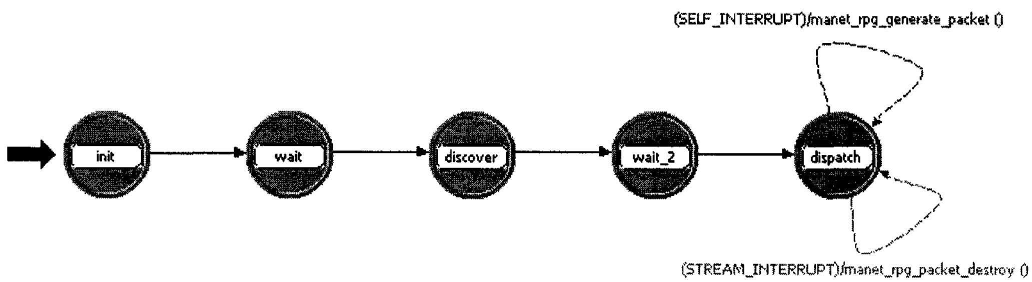


Figure 7.5: Process Model

7.2.4 Process Model

The module or process models shown in Figure 7.5 consist of a group of state transition diagram including states and transitions. The process model is implemented by a language called Proto-C which is compatible with C language. We modify the process code to embed the AMR-WB coding into above MANET model and use the core procedure such as op_pk_send and op_pk_get to realize the voice over IP with the adaptive rate control scheme.

We add PESQ module to measure the MOS of the quality of speech.

The Packet Format Model is also used to represent the packet of speech stream. The packet model includes head, mode, sequence number and cyclic redundancy check shown in Figure 7.6.

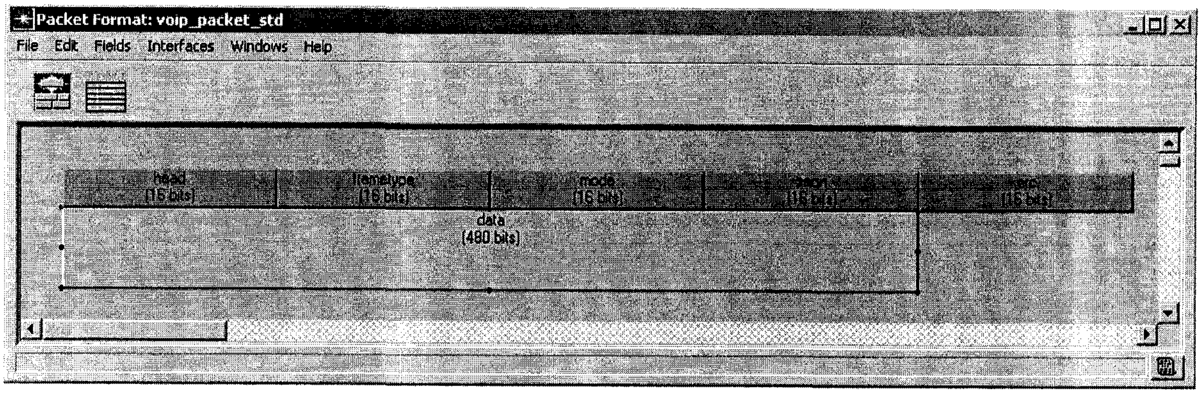


Figure 7.6: Packet Format

7.3 Assumptions

The following assumptions are used in the performance evaluation done in Chapter 8:

1. A wireless ad hoc network consists of homogeneous nodes.
2. All nodes have the same transmission power level when they communicate with each other.
3. The physical layer uses DSSS modulation.
4. Packet reception power threshold is configured to be -95 dBm.
5. The power of transmitter is 1 mW. This is the lowest power class for WLAN and the hop distance is 70 meters.
6. Data transmission rate is 11 MHz and Channel 1 is used with frequency 2412 ± 11 MHz.
7. The mobile node moving speed is 1.5 m/s as in the common walking speed.
8. Traffic packet size is 60 bytes which is the maximum coding payload of AMR-WB.

7.4 Concluding Remarks

In this chapter, we have provided the mathematical model and the simulation models for use of performance evaluation in later chapter. We discussed the implementation of the voice over ad hoc networks. We will simulate the VoIP using these models in the next chapter.

Chapter Eight
QoS Performance of VoIP on Simulated Ad-Hoc Networks

This chapter describes the simulation of our adaptive rate control mechanism in voice over static ad hoc networks in order to study the performance of voice over ad hoc network further. Performance measures of throughput, delay, packet loss and speech quality have been defined in Section 4.4. They are evaluated in both static and mobile environment so as to study the characteristics of ad hoc networks and verify the control schemes effectiveness. We also compare the performance of simulation with testbed measurements to verify the model of simulation. As implied in Section 7.2, all simulated results are measured in outdoor scenarios.

8.1 Simulated Static Networks

We shall simulate the performance in static outdoor environment because we only have outdoor model. The simulation scenarios are described in Section 7.2. The performance of the Constant Bit Rate (CBR) and the Adaptive Source-Network rate Control scheme (ASNC) are simulated respectively. We send a voice stream and a background traffic stream simultaneously at the source node and receive them at the destination node. The AMR-WB codec and PESQ evaluation program are embedded in the process models. The throughput is measured by sending the FTP traffic. RTT and jitter are measured by using probe model. The packet loss rate and speech quality are measured using a program embedded in OPNET.

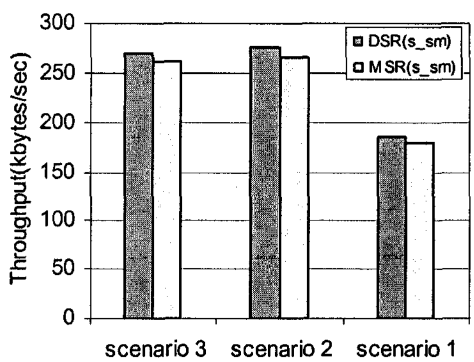


Figure 8.1: Throughput Performance

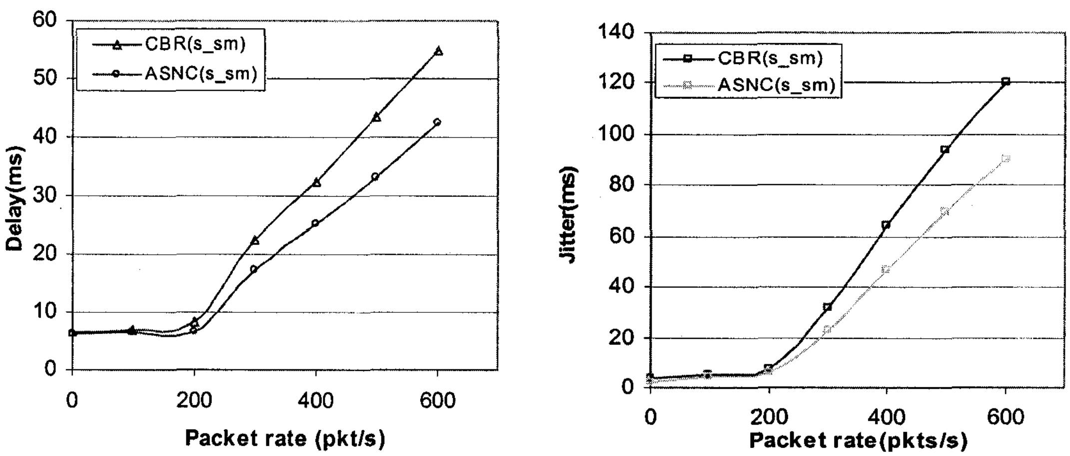
8.1.1 Throughput

The throughput is measured in the simulation by sending FTP traffic in static scenarios using the DSR and MSR routing protocol with background traffic.

Figure 8.1 shows that the throughput performance. We can see that Scenario 2 has achieved the best throughput while Scenario 1 is the worst since Scenario 1 has more hops than the other scenarios. Scenario 2 is better than Scenario 3 because the multiple paths used in Scenario 3 can cause the packets arriving at the destination out of order and the packets of retransmission. We can also see that the DSR has a slightly better performance than MSR because MSR is a more complex algorithm. For example, Scenario 1 has achieved 180 KB/s for DSR opposed to throughput of 170 KB/s for MSR.

8.1.2 Delay and Jitter

We obtain from simulation the end-to-end delay and its jitter performance of speech transmission with a background traffic, and compare the performance of CBR and ASNC using MSR.

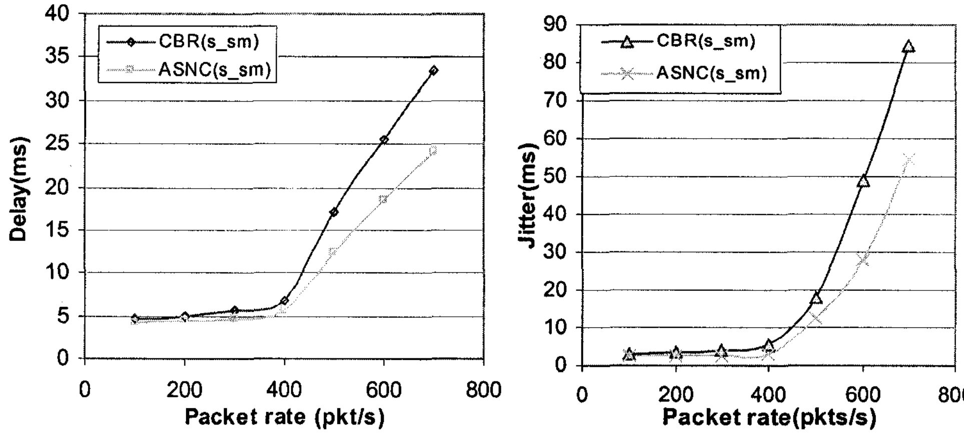


(a) Delay (b) Jitter

Figure 8.2: Delay Performance of Scenario 1

8.1.2.1 Scenario 1

Figure 8.2 shows that RTT and jitter increase with increasing traffic. The traffic is increased by using larger packets which leads to more congestion and increases delay and jitter. The figures also illustrate that the ASNC has a better performance (ASNC(s_sm)) in RTT and jitter than those (CBR(s_sm)) of CBR because ASNC is capable to reduce traffic through adaptation. This is especially the case when packer rate is beyond 200 packets/sec. For example, at 400 packets/sec, ASNC has a delay of 25 ms and a jitter of 48 ms, compared with 32 ms and 65 ms for CBR.

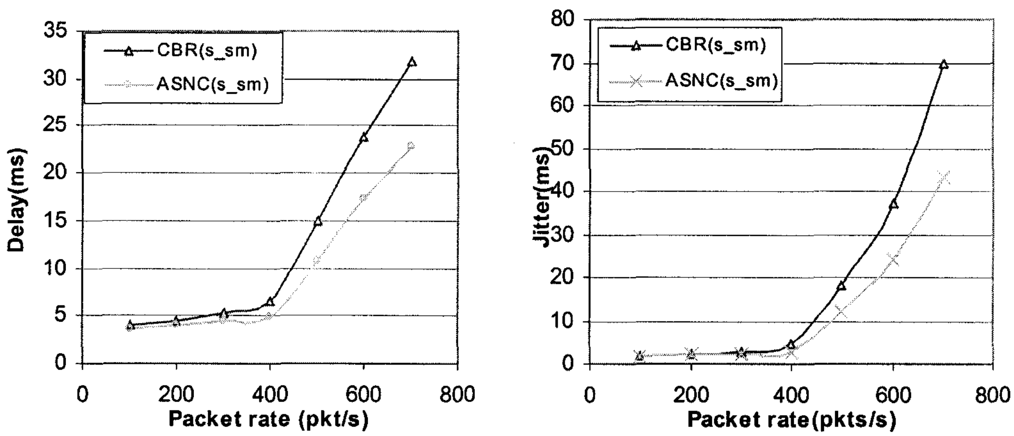


(a) Delay (b) Jitter

Figure 8.3: Delay & Jitter Performance of Scenario 2

8.1.2.2 Scenario 2

Figure 8.3 compares the delay and jitter performance of CBR and ASNC in Scenario 2. Observations similar to Figure 8.2 can be made. However, both RTT and jitter are much reduced because Scenario 2 has fewer hops than Scenario 1.



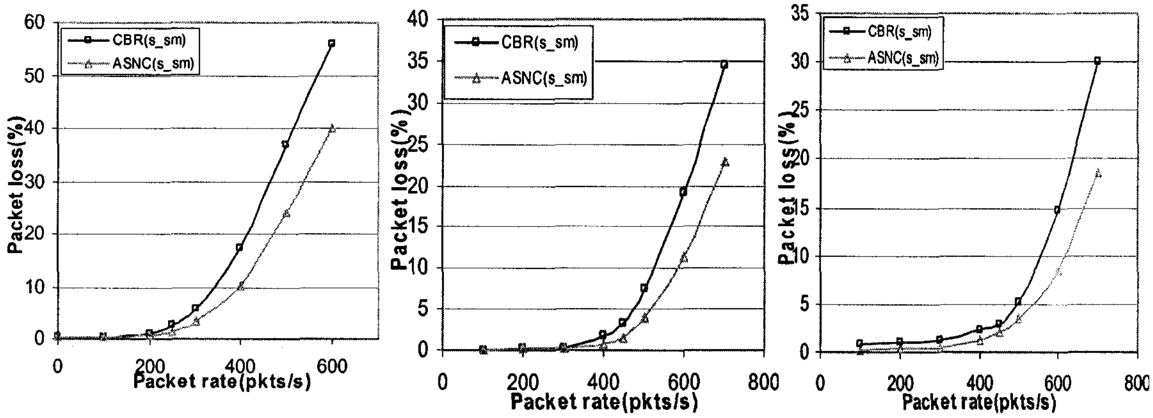
(a) Delay (b) Jitter

Figure 8.4: Delay & Jitter Performance of Scenario 3

8.1.2.3 Scenario 3

Figure 8.4 compares the delay and jitter performance of CBR and ASNC in Scenario 3. Observations similar to Figure 8.2 can be made. However, both RTT and jitter are also much reduced because Scenario 3 has fewer hops than Scenario 1.

By comparing the three scenarios, scenario 1 has the largest RTT and jitter because it has more hops (3 hops) than the others. Scenario 3 has the smallest RTT and jitter because Scenario 3 has fewer hops and utilizes multi-path.



(a) Scenario 1 (b) Scenario 2 (c) Scenario 3

Figure 8.5: Packet Loss Performance

8.1.3 Packet Loss Rate

Figure 8.5 presents the packet loss rate performance under the three scenarios described in Section 7.2 using the MSR operation. CBR is compared to ASNC in each scenario.

As seen in Scenario 1 (Figure 8.5a), the packet loss rate under CBR (CBR(s_sm)) is increasing exponentially with regard to the packet arrival rate. The performance of ASNC is similar but much reduced. For example, the loss rate is 11% at 400 packets/sec as opposed to 17% for CBR. Similar performance of CBR and ASNC and their comparison are observed under Scenario 2 (Figure 8.5b) and Scenario 3 (Figure 8.5c), but the loss rate is decreasing. For example, at 500 packets/sec, ASNC loss rate is reduced for 24% in Scenario 1 to 4% in Scenario 2 and eventually to 3% in Scenario 3. This is because Scenario 1 has more hops than Scenario 2 and Scenario 3.

8.1.4 Speech Quality

Figure 8.6 presents the speech quality performance under the three scenarios described in Section 7.2 using the MSR operation. CBR is compared to ASNC in each scenario.

As seen in Scenario 1(Figure 8.6a), the MOS under CBR (CBR(s_sm)) is decreasing with regard to traffic packet rate. The performance of ASNC is similar but much increased because ASNC is capable to reduce traffic through adaptation. For example, the MOS is 3.0 at 300 packets/sec is opposed to 2.7 for CBR. Similar performance of CBR and ASNC and their comparison are observed under Scenario 2 (Figure 8.6b) and Scenario 3 (Figure 8.6c), but the MOS is increasing. For example, at 500 packets/sce, ASNC MOS is increased for 1.9 in Scenario 1 to 3.0 in Scenario 2 and eventually to 3.2 in Scenario 3. This is because Scenarios 2 and 3 have fewer hops than Scenario 1.

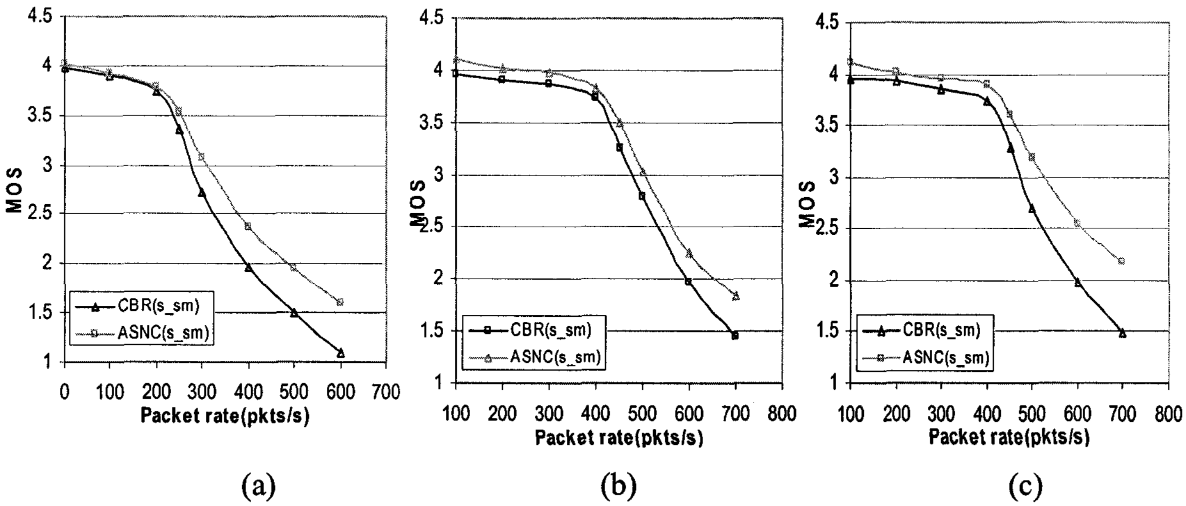


Figure 8.6: Speech Quality (a) Scenario 1 (b) Scenario 2 (c) Scenario 3

8.1.5 Comparison of Simulation and Testbed Measurements

In this section, we shall analyze and compare the OPNET simulation results with the measurement results in static outdoor environments obtained earlier.

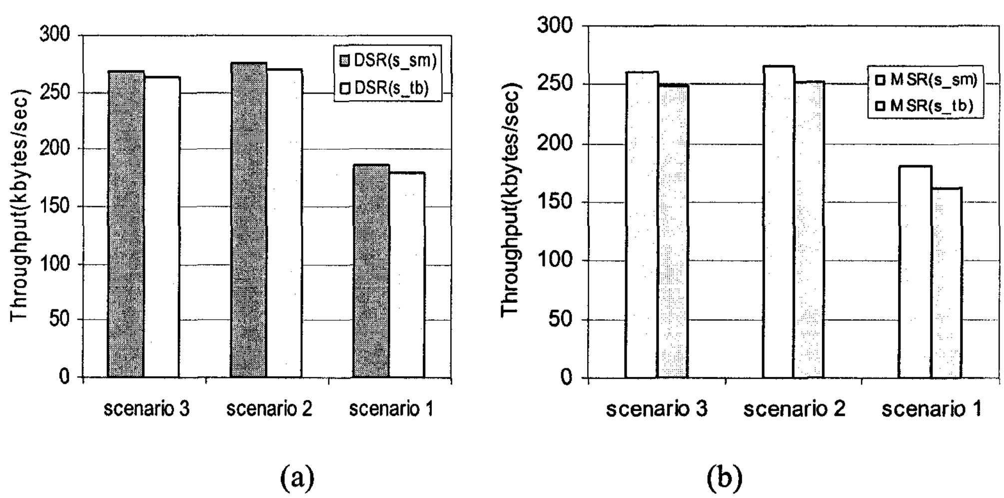
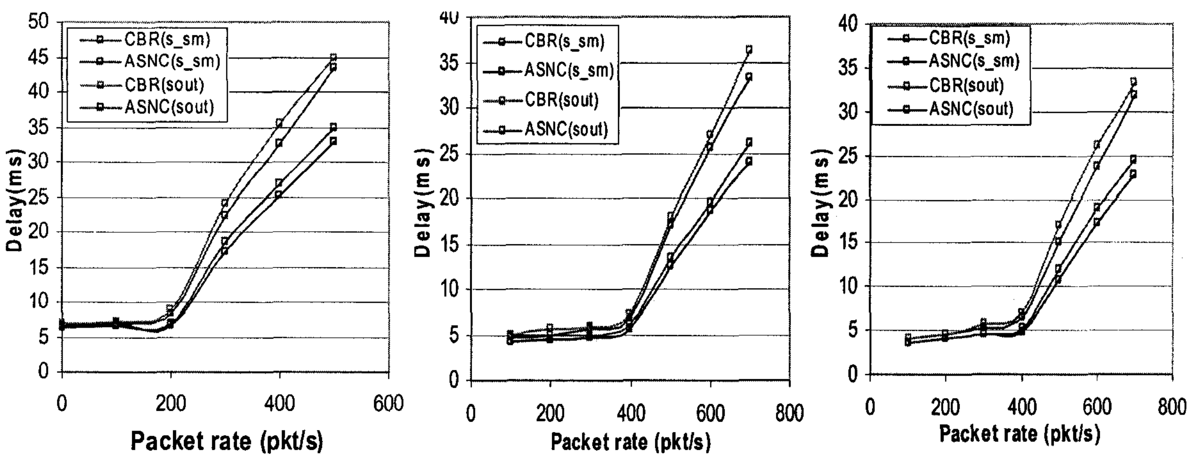


Figure 8.7: Throughput Comparisons of Simulation and Measurement

8.1.5.1 Throughput Comparisons

Figure 8.7 shows that throughput performance of the static scenario for measurement (s_tb) and simulation results (s_sm) using DSR. Figure 8.7a presents that the simulation for DSR has slightly better throughput than the testbed measurements in the three scenarios. The reason is that real physical layer is more complex than the simulation model. For example, Scenario 1 has achieved throughput of 180 KB/sec for testbed compared with 186 KB/sec for simulation. Similar performance comparison is observed for MSR shown in Figure 8.7b, but the throughput is decreasing because MSR has more complex algorithm.



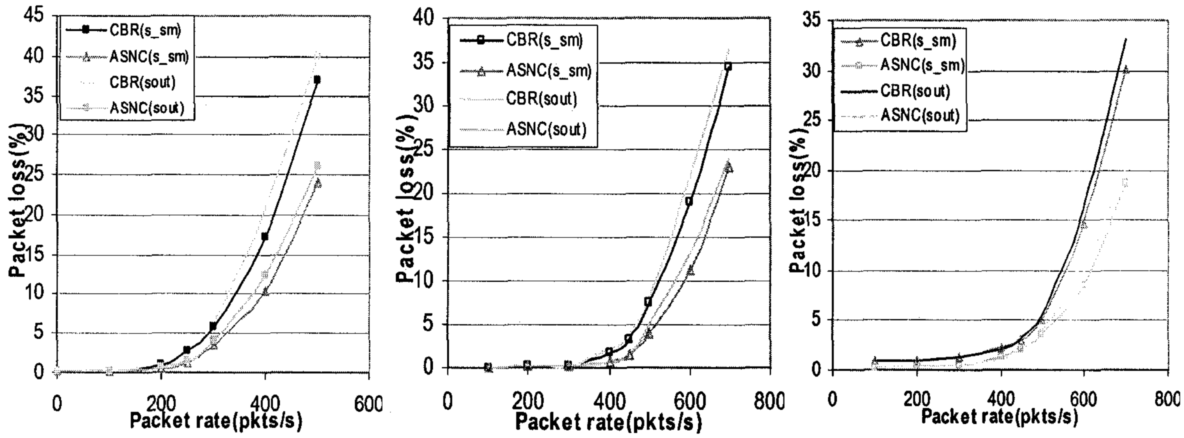
(a) Scenario 1 (b) Scenario 2 (c) Scenario 3

Figure 8.8: Delay Comparisons of Simulation and Measurement

8.1.5.2 Delay Comparisons

Figure 8.8 shows that the deley performance comparison between the static measurement (sout) and static simulation (s_sm) results in three scenarios described in Section 7.2. As seen in Scenario 1 (Figure 8.8a), the simulation for CBR has achieved shorter delay than the testbed measurement. The reason is that the simulation model has some limitations and assumptions while the testbed reflects real physical layer. For example, the delay of simulation is 25 ms at 400 packets/sec as opposed to 27 for testbed. The performance of ASNC is similar but much increased.

Similar performance comparisons are observed under Scenario 2 (Figure 8.8b) and Scenario 3 (Figure 8.8c) that the simulation has achieve shorter delay than the testbed measurement, but the delay is much decreased since they have fewer hops than Scenario 1.

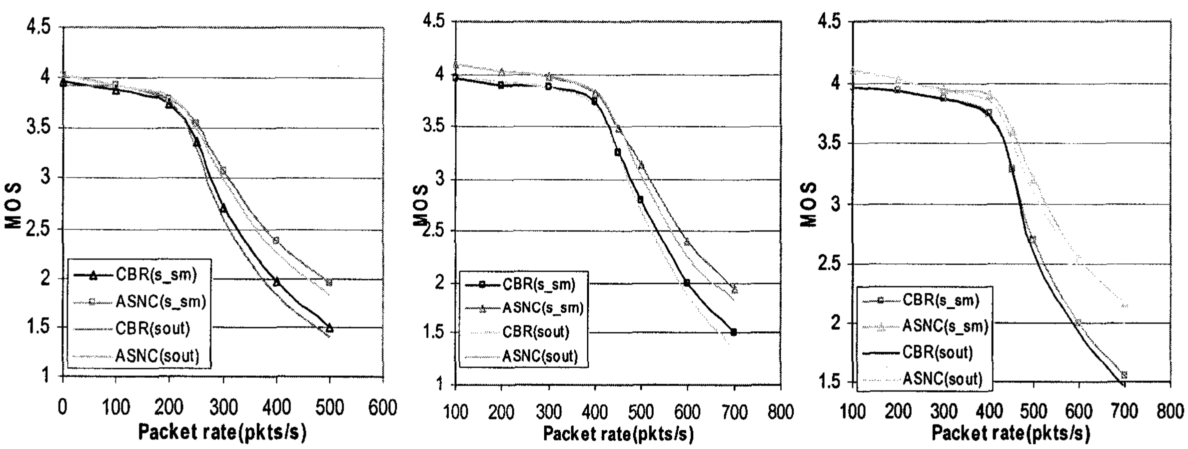


(a) Scenario 1 (b) Scenario 2 (c) Scenario 3

Figure 8.9: Packet Loss Rate Comparison of Simulation and Measurement

8.1.5.3 Packet Loss Rate Comparisons

Figure 8.9 show that MOS performance comparison between the static measurement (sout) and static simulation (s_sm) results in three scenarios described in Section 7.2. As seen in Scenario 1 (Figure 8.9a), the simulation for CBR has achieved smaller packet loss rate than the testbed measurement. The reason is that the simulation model has limitations due to simplification and assumptions while the testbed reflects real physical environment. For example, the loss rate of simulation is 10% at 400 packets/sec as opposed to 12% for testbed. The performance of ASNC is similar but much increased. Similar performance comparisons are observed under Scenario 2 (Figure 8.9b) and Scenario 3 (Figure 8.9c) that the simulation has achieved smaller loss rate than the testbed measurement, but the loss rate is much decreased since they have fewer hops than Scenario 1.



(a) Scenario 1 (b) Scenario 2 (c) Scenario 3

Figure 8.10: MOS Comparison of Simulation and Measurement

8.1.5.4 Speech Quality Comparisons

Figure 8.10 shows that MOS performance comparison between the static measurement (sout) and static simulation (s_sm) results in three scenarios described in Section 7.2. As seen in Scenario 1 (Fig 8.10a), the simulation for CBR has achieved better MOS than the testbed measurement. The reason is that the simulation model has some limitations and assumptions while the testbed reflects real physical layer. For example, the MOS of simulation is 2.0 at 400 packets/sec as opposed to 1.9 for testbed. The performance of ASNC is similar but much increased.

Similar performance comparisons are observed under Scenario 2 (Figure 8.10b) and Scenario 3 (Figure 8.10c) that the simulation has achieve better MOS than the testbed measurement, but the MOS is much increased since they have fewer hops than Scenario 1.

8.2 Simulated Mobile Networks

We now repeat the same experiments as described in Section 8.1 but for mobile outdoor networks.

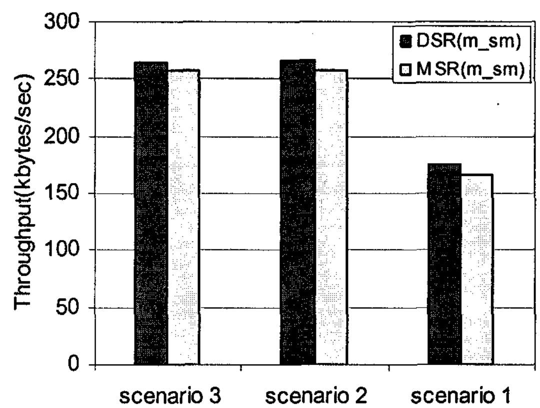


Figure 8.11: Throughput Performance of Scenario 1

8.2.1 Throughput

The throughput is measured in the simulation by sending FTP traffic in mobile scenarios using the MSR and DSR routing protocol with background traffic.

Figure 8.11 shows that the throughput performance. We can see that Scenario 2 has achieved the best throughput while Scenario 1 is the worst since Scenario 1 has more hops than the other scenarios. Scenario 2 is slightly better than Scenario 3 because the multiple paths used in Scenario 3 can cause the packets arriving at the destination out of order and the packets of retransmission. We can also see that the DSR has a slightly better performance than MSR because MSR is a more complex algorithm. For example, Scenario 1 has achieved 175 KB/s for DSR opposed to throughput of 166 KB/s for MSR.

8.2.2 End-to-end Delay and Jitter

We obtain from simulation the end-to-end delay and its jitter performance of speech transmission with a background traffic, and compare the performance of CBR and ASNC using MSR.

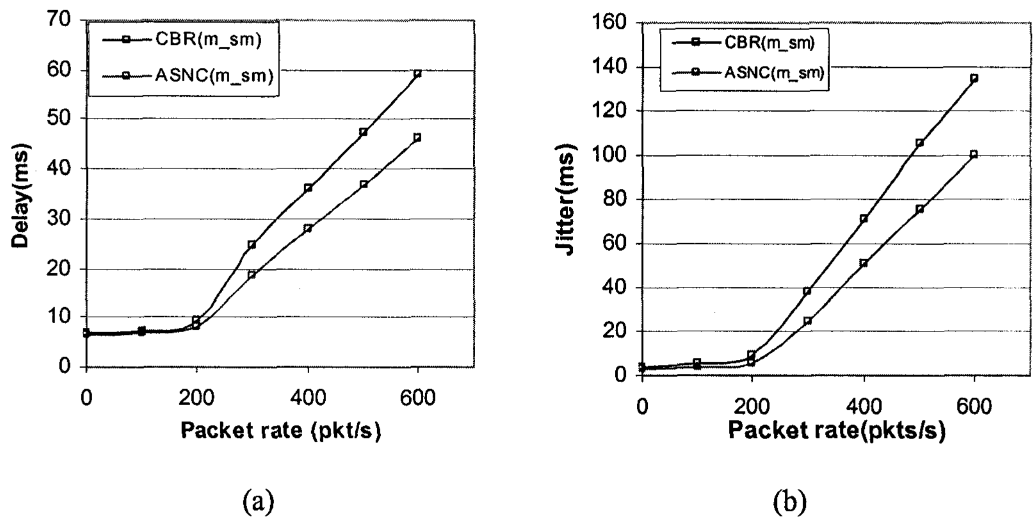


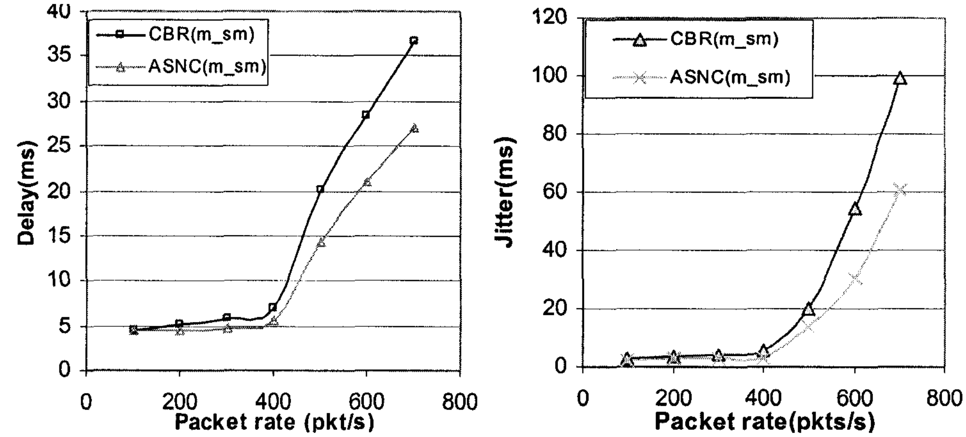
Figure 8.12: Delay & Jitter Performance of Scenario 1

8.2.2.1 Scenario 1

Figure 8.12 shows that RTT and jitter increase with the increasing traffic. The traffic is increased by using the larger packets which leads to more congestion and increases delay and jitter. The figures also illustrate that the ASNC has a better performance (ASNC(m_sm)) in RTT and jitter than those (CBR(m_sm)) of CBR because ASNC is capable to reduce traffic through adaptation. This is especially the case when packet rate is beyond 200 packets/sec. For example, at 400 packets/sec, ASNC has a delay of 28 ms and a jitter of 50 ms, compared with 35 ms and 68 ms for CBR.

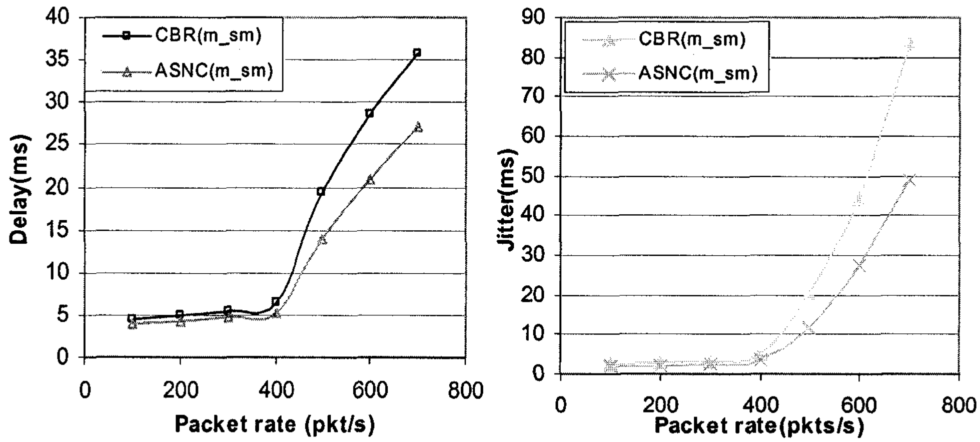
8.2.2.2 Scenario 2

Figure 8.13 compares the delay and jitter performance of CBR and ASNC in Scenario 2. Observations similar to Fig. 8.2 can be made. However, both RTT and jitter are much reduced because Scenario 2 has fewer hops than Scenario 1.



(a) (b)

Figure 8.13: Delay & Jitter Performance of Scenario 2



(a) (b)

Figure 8.14: Delay & Jitter Performance of Scenario 3

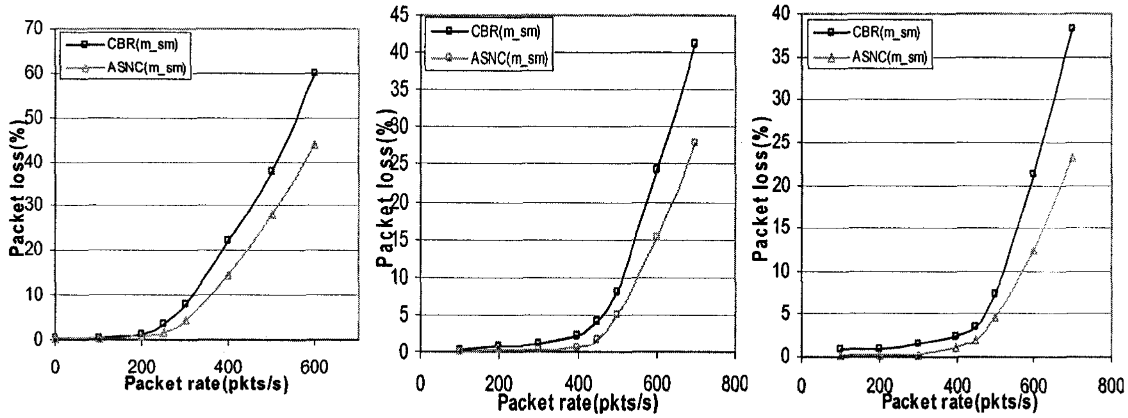
8.2.2.2 Scenario 3

Figure 8.14 compares the delay and jitter performance of CBR and ASNC in Scenario 3. Observations similar to Figure 8.2 can be made. However, both RTT and jitter are also much reduced because Scenario 3 has fewer hops than Scenario 1.

By comparing the three scenarios, scenario 1 has the largest RTT and jitter because it has more hops (3 hops) than the others. Scenario 3 has the smallest RTT and jitter because Scenario 3 has fewer hops and utilizes multi-path.

8.2.3 Packet Loss Rate

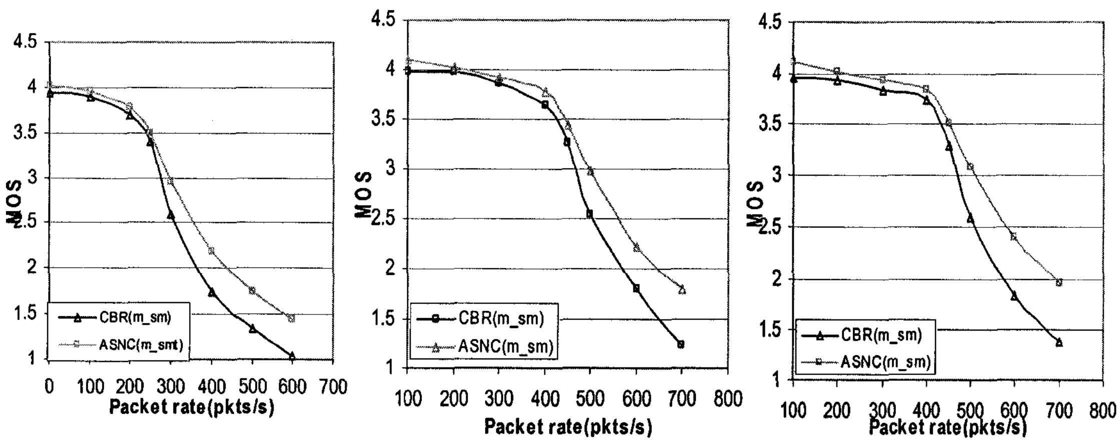
Figure 8.15 presents the packet loss rate performance under the three scenarios described in Section 7.2 using the MSR operation. CBR is compared to ASNC in each scenario.



(a) Scenario 1 (b) Scenario 2 (c) Scenario 3

Figure 8.15: Packet Loss Performance

As seen in Scenario 1 (Figure 8.15a), the packet loss rate under CBR (CBR(m_sm)) is increasing exponentially with regard to the packet arrival rate. The performance of ASNC (ASNC(m_sm)) is similar but much reduced. For example, the loss rate is 14% at 400 packets/sec as opposed to 22% for CBR. Similar performance of CBR and ASNC and their comparison are observed under Scenario 2 (Figure 8.15b) and Scenario 3 (Figure 8.15c), but the loss rate is much decreasing. For example, at 500 packets/sec, ASNC loss rate is reduced for 28% in Scenario 1 to 5% in Scenario 2 and eventually to 4% in Scenario 3. This is because Scenario 1 has more hops than Scenario 2 and Scenario 3.



(a) (b) (c)

Figure 8.16: Speech Quality (a) Scenario 1 (b) Scenario 2 (c) Scenario 3

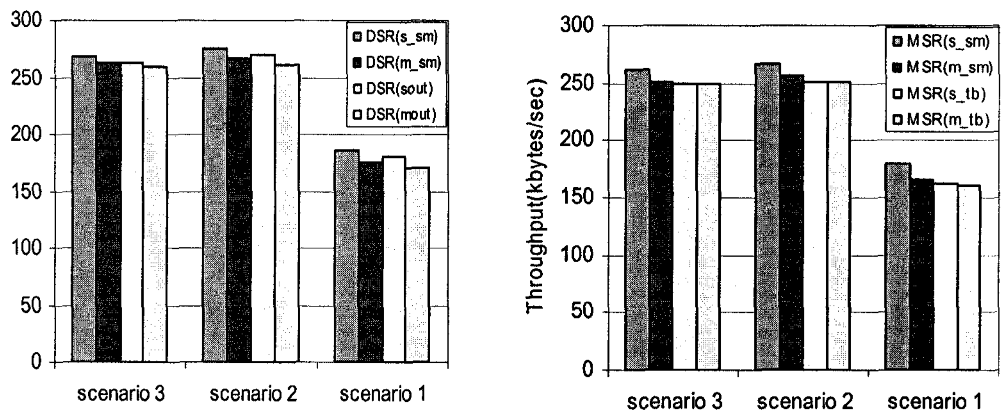
8.2.4 Speech Quality

Figure 8.16 presents the speech quality performance under the three scenarios described in Section 7.2 using the MSR operation. CBR is compared to ASNC in each scenario.

As seen in Scenario 1 (Figure 8.16a), the MOS under CBR (CBR(m_sm)) is decreasing with regard to traffic packet rate. The performance of ASNC(ASNC(m_sm)) is similar but much increased because ASNC is capable to reduce traffic through adaptation. For example, the MOS is 2.9 at 300 packets/sec is opposed to 2.5 for CBR. Similar performance of CBR and ASNC and their comparison are observed under Scenario 2 (Figure 8.16b) and Scenario 3 (Figure 8.16c), but the MOS is increasing. For example, at 500 packets/sce, ASNC MOS is increased for 1.8 in Scenario 1 to 3.0 in Scenario 2 and eventually to 3.1 in Scenario 3. This is because Scenarios 2 and 3 have fewer hops than Scenario 1.

8.2.5 Simulation and Testbed Comparison

In this section, we shall analyze and compare the OPNET simulation results with the outdoor measurement results in mobile environments obtained earlier. We also compare the simulation between static and mobile environments



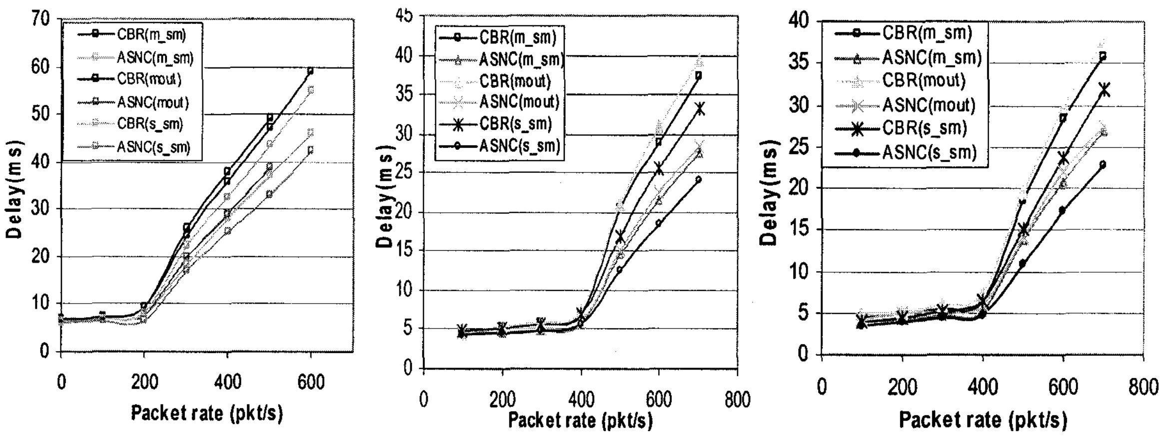
(a) (b)

Figure 8.17: Throughput Comparison of Simulation and Measurement

8.2.5.1 Throughput Comparisons

Figure 8.17 shows the throughput performance comparison between the mobile simulation (m_sm) and mobile measurements (mout). Figure 8.17a presents that the simulation for DSR has slightly better throughput than the testbed measurement in mobile environments. The reason is that real physical layer is more complex than the simulation model. For example, Scenario 1 has achieved throughput of 170 KB/sec for testbed compared with 175 KB/sec for simulation. We can also see than static simulation results (s_sm) are better than mobile simulation results (m_sm) since the mobility brings the signal and path unstable. For example, Scenario 1 has achieved throughput of 175 KB/sec for mobile simulation compared with 185 KB/sec for static simulation.

Similar performance comparison is observed under MSR (Figure 8.17b), but the throughput is decreasing because the packets of MSR easy out of order and have to be retransmitted. For example, Scenario 1 in mobile simulation for DSR has achieved the throughput of 175 KB/s compared with 166 KB/s for MSR.



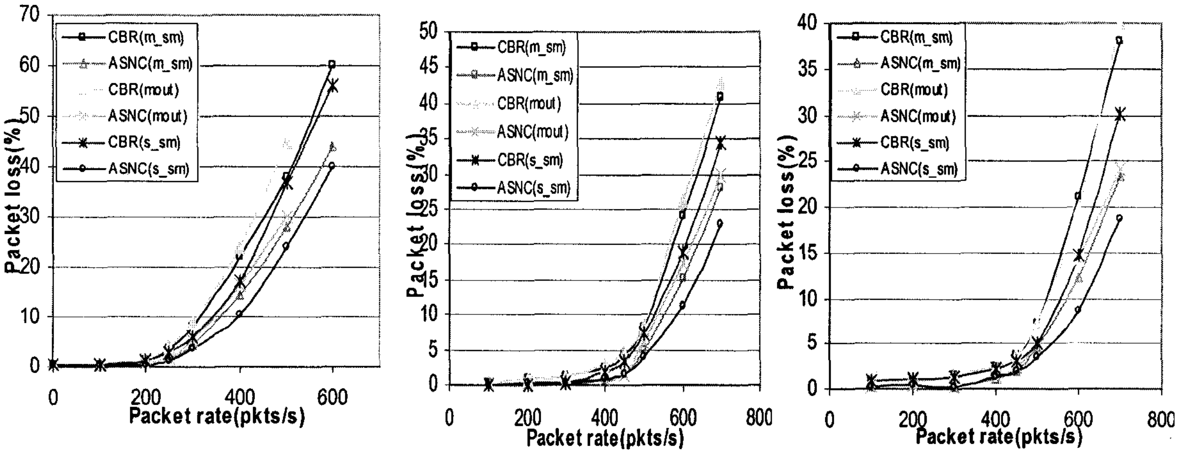
(a) Scenario 1 **(b) Scenario 2** **(c) Scenario 3**

Figure 8.18: Delay Comparisons of Simulation and Measurement

8.2.5.2 Delay Comparisons

Figure 8.18 shows that the delay performance comparison between the mobile measurement (mout) and mobile simulation (m_sm) results in three scenarios described in Section 7.2. As seen in Scenario 1 (Figure 8.18a), the simulation for both CBR and ASNC has achieved slight shorter delay than the testbed measurement. The reason is that the simulation model has some limitations and assumptions while the testbed reflects real physical layer. The figure also presents that static simulation (s_sm) results are better than the mobile simulation (m_sm) for both CBR and ASNC since the mobility brings the signal and path unstable. Similar performance comparisons are observed under Scenario 2 (Figure 8.18b) and

Scenario 3 (Figure 8.18c), but the delay is much decreased since they have fewer hops than Scenario 1.



(a) Scenario 1 (b) Scenario 2 (c) Scenario 3

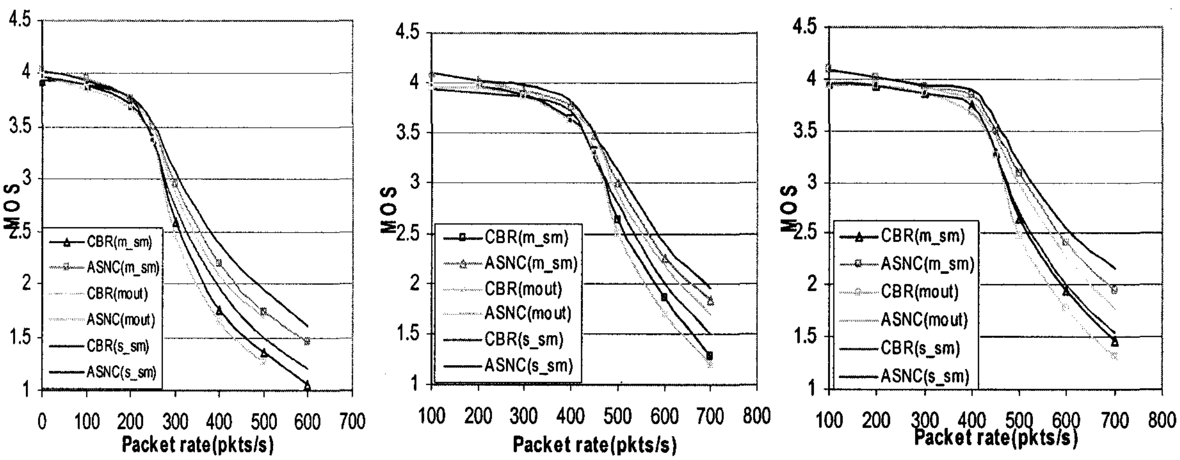
Figure 8.19: Packet Loss Comparisons of Simulation and Measurement

8.2.5.3 Packet Loss Rate Comparisons

Figure 8.19 presents that the packet loss rate performance comparison between the mobile measurement (mout) and mobile simulation (m_sm) results in three scenarios decribed in Section 7.2. As seen in Scenario 1 (Fig 8.19a), the simulation for both CBR and ASNC has achieved slight smaller MOS than the testbed measurement. The reason is that the simulation model has some limitations and assumptions while the testbed reflects real physical layer. The figure also shows that static simulation (s_sm) results are better than the mobile simulation (m_sm) for both CBR and ASNC since the mobility brings the signal and path unstable. Similar performance comparisons are observed under Scenario 2 (Fig 8.19b) and Scenario 3 (8.19c), but the packet loss rate is much decreasing since they have fewer hops than Scenario 1.

8.2.5.4 Speech Quality Comparisons

Figure 8.20 show that MOS performance comparison between the mobile measurement (mout) and mobile simulation (m_sm) results in three scenarios described in Section 7.2. As seen in Scenario 1 (Figure 8.20a), the simulation for both CBR and ASNC has achieved better MOS than the testbed measurement. The reason is that the simulation model has some limitations and assumptions while the testbed reflects real physical layer. The figure also shows that static simulation (s_sm) results are better than the mobile simulation (m_sm) for both CBR and ASNC since the mobility brings the signal and path unstable. Similar performance comparisons are observed under Scenario 2 (Figure 8.20b) and Scenario 3 (Figure 8.20c), but the MOS is much increased since they have fewer hops than Scenario 1.



(a) Scenario 1 (b) Scenario 2 (c) Scenario 3

Figure 8.20: Comparison of Simulation and Measurement

8.3 Concluding Remarks

In this chapter, we simulated voice over static and mobile ad hoc networks using OPNET tools. The results show the performance in different scenarios and verify the effectiveness of the adaptive network rate control scheme. In addition, the results show that OPNET simulation has slightly better results than measurement results since the simulation model is just a simplification of real world. Despite this, we can say that our simulation model can track the real system closely. Furthermore, unlike some measurements (e.g. at packet rate of 600 packets/sec) that can not be taken in real system before due to much higher packet loss rate, we can now make the measurements easily and with a good confidence. These data results can be found in Figures 8.2, 8.5a, 8.6a, 8.8a, 8.12, 15a and 8.16a.

Chapter Nine
Conclusions

In this thesis, we have proposed the adaptive rate control scheme to improve the performance of voice over wireless ad hoc networks. The basic idea is to adjust the voice coding bit rate according to the available bandwidth to make the maximum use of network resources. Based on our analysis of the characteristics of wireless ad hoc networks and the frame loss performance of speech codec AMR-WB, we proposed a source-network rate control scheme that yields the maximal speech quality by considering the network conditions and source characteristics at the source and choosing the best encoding rate. MOS is used as the standard to evaluate the quality of speech. The discontinuous transmission and packet combination are used to decrease the bit rate.

Both the testbed measurement and the OPNET simulation methods have been used to verify the effectiveness of the proposed adaptive rate control scheme. All test scenarios are conducted in both static and mobile network as well as both indoor and outdoor. Both measurement and simulation results demonstrated the effectiveness of our proposed scheme for speech communications over wireless ad hoc networks in each scenario. The results also show that OPNET simulation has approximate results (slightly better) with testbed results.

Based on our study, we would like to provide some valuable points on the design and measurement of ad hoc networks. We also describe the testing issues meeting in measurements.

9.1 Networks Configuration Guideline

In order to setup a wireless ad hoc network, we first need a device with wireless LAN interface. For example, a laptop with a PCMCIA wireless adaptor. In order to configure all mobile nodes in one ad hoc subnet without an infrastructure, each wireless card is set an ad hoc mode with the same SSID (Service Set Identification) and each device has to be set a static IP address with the same subnet mask. A suitable ad hoc routing protocol is also required for each device. For example, we use DSR or MSR ad hoc routing algorithms. Particularly, we wrote VoIP program using the adaptive control scheme to control the coding rate so as to make the maximum use of network resources. Because of bandwidth limitation, we should use adaptive control scheme to control the coding rate.

9.2 Environment Guideline

From the experiments we have seen that the outdoor has a better signal quality than the indoor because the signal outdoor propagation is mostly line of light. So the outdoor commutation distance is larger than the indoor. For example, when transmission power is only 1mW, the distance reaches about 120 meters. However, we found that terrain profile affects wireless transmission greatly because the height of ad hoc nodes is limited. The quality of reception signal is weak when the laptop's position is too low. So a higher position can ensure a better reception of wireless signal. Indoor propagation conditions are more variable than outdoor due the factors like blocking and reflection. The range of wireless indoor is limited. For example, the transmission distance is about 20 meters with the transmission power of 1mW. So we should be more power to enhance the coverage of wireless.

One also has to pay more attention to the signal of mobile environment which is variable due the distance variation between the transmitter and the receiver. The position variation will lead to path update frequently. It even causes the random frequency modulation and time dispersion caused by the propagation delay. These effects become more serious when the speed of nodes is higher. All these factors lead to the quality deterioration of signal. While static environment can take advantage of the very well-defined, time-invariant nature of the propagation channel between the transmitter and the receiver, the wireless link is more stable that will benefit the reception of the receiver. These factors should be considered in the experiments.

9.3 Combating Network Congestion

From the measurement results we can see that there is a congestion point for the wireless system. The performance decreases slowly before the congestion point while the performance decreases dramatically after that point when the traffic increases. The value of congestion point is inversely proportional to the number of hops. Admission control schemes and adaptive control schemes can be used to decrease the probability of congestion. The adaptive source network rate control proposed in this thesis is used to alleviate the network congestion and improve the quality of speech transmission in wireless ad hoc networks.

9.4 Future Work

Our adaptive rate control scheme is used in speech coding. It also has the potential to test other more complicated scenarios. For example, it can also be used in video stream using codec H.264 that is a new coding standard with a high quality and a lower bit rate. This will be more challenging because video needs more wideband than audio. In addition, we can also use channel coding for example Reed-Solomon coding to protect channel errors. We can study the performance in more complex environments. For example, there are interference sources such as microwave ovens; nodes move at high speed (on moving vehicles). We can make use of the technologies above to improve the performance in these environments.

In addition, we can conduct more simulations. For example, we can implement more complex scenarios to evaluate the performance of the ad hoc networks. We can use cross layer design to improve the performance. We can also compare the performance of different ad hoc routing schemes such as DSR, MSR and AODV. Furthermore, our simulation model can be improved so as to model ad hoc wireless networks more accurately.

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