Chapter 1 Introduction

The wireless networks are under rapid growth and continuous evolution in order to accommodate an increasingly large number of applications with various service requirements. In order to provide a better service for the wireless network systems, there are many things to be done. Among them, providing Quality of Service (QoS) is a must in many situations. QoS is a set of parameters to measure the networks' performance. The parameters include packet delay bound, delay jitter, bandwidth, packet lose rate, etc. In this thesis, we will study the topics of QoS support for wireless network systems.

The history of modern wireless communications [1,12] started in 1896. The researcher, named Marconi, demonstrated wireless telegraphy by sending and receiving Morse code based on long-wave, whose wavelength is greater than 1 km using high-power transmitters. In 1907, the first commercial trans-Atlantic wireless service was initiated using huge ground stations. Wireless voice transmission between New York and San Francisco was achieved in 1915. In 1920, Marconi discovered short-wave, less than 100m wavelength, transmission. Such waves undergo reflections, refractions absorption, and bouncing off the ionosphere, which make short-waves more efficient. And in the same year, the first commercial radio broadcast took place in Pittsburgh, Pennsylvania. In 1921, police cars in Detroit, Michigan, were equipped with wireless dispatch radios. In 1935, the first telephone call around the world was made.

Ubiquitous access to information, anywhere, anyplace, and anytime, will characterize a whole new kind of information system in the 21st Century. Over the last two decades, development of wireless communication services was very phenomenal.

Without the restrictions of the communication wires, the wireless network system is so convenient for communicating with others. Demand for mobile communication service grows quickly. Nowadays, there are two variations of mobile wireless networks. The first is known as cellular personal communication systems (PCS), i.e., those networks with fixed and wired gateways. The bridges between the wireless networks and wired networks are known as base stations. A mobile node within these networks connects to, and communicates with, the nearest base station that is within its communications radius. The second type of mobile wireless network has no wired infrastructure. It is commonly known as an ad hoc network. The ad hoc networks have no fixed routers; all nodes are capable of movement and can be connected dynamically in an arbitrary manner. Mobile nodes of these networks function as routers, which discover and maintain routes to other nodes in the network.

In this thesis, we present a Quality of Service (QoS) supporting handoff procedure in the multimedia infrastructured wireless networks. A strategy called Signal Strength for Multimedia Communications (SSMC) is proposed to handle the handoff request in infrastructured wireless networks. SSMC calculates the handoff request priority by three values: the static priority value, the degradation rate of the received signal strength and the level of received signal strength. A handoff request with highest handoff priority will be first served. On the next part of the thesis, we propose a QoS supported routing algorithm in the ad hoc networks. Based on the AODV [44] routing protocol, we propose an AODV-RSS (AODV with Received Signal Strength) routing protocol to support QoS guarantees in ad hoc networks. The link available time (*LAT*) between two mobile nodes is forecasted by the received signal strength and the received signal strength changing rate. In accordance with the connection's QoS request, our method will discover a minimum hop count routing path that satisfies the QoS requirement if there is one.

The rest of this thesis is organized as follows. Chapter Two depicts background knowledge and surveys related work of our research. In Chapter Three, we will provide a QoS supported algorithm for the handoff procedure in the multimedia personal communication system. A QoS supported routing algorithm for ad hoc networks is presented in Chapter Four. Finally, conclusions and future work will be given in Chapter Five.

Chapter 2 Background Knowledge and Related Work

2.1 Personal Communication System

In this section, we briefly present the evolution and the concept of cellular communication system used in the United States and Europe. The mobile host registration procedure and a call establishment processes are presented.

The first public mobile telephone service started in 25 major United States of America cities in 1946 by AT&T. These mobile telephone systems were based on Frequency Modulation (FM) transmission. The FM mobile telephone channels used 120 kilohertz (kHz) of bandwidth to transmit voice with an effective bandwidth of only about 3 kHz in half-duplex mode. Then, the Federal Communications Commission (FCC) doubling the number of mobile channels and improved technology cut the bandwidth to 60 kHz in 1950. In 1960, the transmission bandwidth was further cut down to 30 kHz. In the mid-1960s the Bell System introduced the Improved Mobile Telephone Service (IMTS) with enhanced features, including automatic trunking, direct dialing, and full-duplex service. In the late 1960s and the early 1970s, work began on the cellular telephone system. In 1968, AT&T presented the cellular concept to the FCC. By 1976, 543 customers (12 channels) could be accommodated in the NY Bell mobile system. Then, in 1983, the FCC allocated 666 duplex channels for the Advance Mobile Phone System (AMPS). In 1985, the FCC released the unlicensed ISM (industrial, scientific, and medical) bands. And the FCC granted an additional 166 channels (10 MHz worth) to AMPS in 1989. In the late 1980s interest emerged in the digital cellular system, where both the voice and the control signal were digital. In 1991, US digital cellular (USDC), or IS-54, which supports three users in each 30 kHz channel, was released. This was later improved to accommodate six users per channel. In 1993, 1.8 GHz was released for the digital Personal Communications System (PCS), followed in 1994 by the introduction of IS-95 code-division multiple access (CDMA) [2]. During that year, approximately 16 million cellular phones were in use. The evolution of wireless technologies in the United States is shown in Figure 2-1.

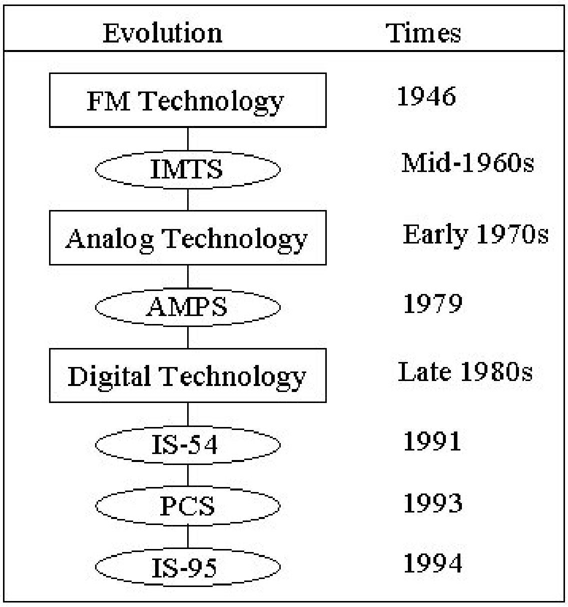

Figure 2-1. Evolution of wireless technologies in the United States

In 1982, the European Global System for Mobile Communication (GSM) [4] was established. However, it was incompatible with systems in other European nations. In 1984, the Cordless Telephone-1 (CT-1) 900 MHz analog system was implemented or proposed in 13 European and Scandinavian countries. In 1987, the digital Cordless Telephone-2 (CT-2) is proposed. And then, in 1989, the GSM 900 MHz is started with full digital 120 channels. In 1991, the DCS-1800 had been developed by the European Telecommunications Standards Institute (ETSI). It is a derivative of the GSM 900 MHz cellular standard. The evolution of wireless communication in Europe is shown in Figure

2-2.

In Asia, Japan built its own cellular system by Japan's Nippon Telephone & Telegraph (NTT). Now, current digital cellular systems use one of three standards:

1. North American Electronic Industry Association (EIA) Standards IS-54
2. Western Europe—Global System for Mobile Communications (GSM)
3. Japanese Digital Cellular (JDC)

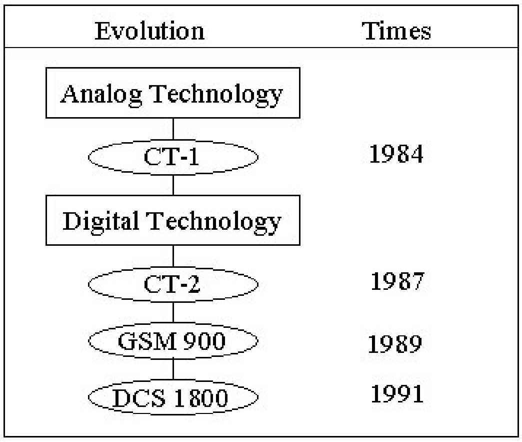

Figure 2-2. Evolution of Wireless Technologies in Europe

In the PCS network, the system area is divided into many small regions (cells) each served by a transmitter antenna. The frequency of the adjacent cell is not the same to avoid interference. The size of the cell is dependent on the power of the transmitter antenna. A higher power antenna covers a larger cell. All these cells cover the whole system. By this reason, the PCS is also called a cellular telephone system.

The architecture of PCS network is shown in Figure 2-3. The system is made up by cells, which are served by base stations (BS). The mobile host (MH) moves around the whole PCS network. Through the air interface, the mobile host communicates with the base station directly. Base station controller (BSC), an interface between the base station and mobile switching center (MSC), controls one or more base stations. It manages the frequency used by each base station, frequency hopping and mobile host handoff. One or more BSCs are managed by the MSC which is the pivot of the base station controller and the wired network system. Generally, the wired network is public switched telephone network (PSTN) or integrated services digital network (ISDN). MSC has the responsibility to transmit the message from BSC to the wired network. On the other side, we can see the MSC also connects with other MSCs. We call this network, connected by MSCs, as the signaling network. The architecture of signaling network is based on the advanced intelligent network (AIN) and signaling system number 7 (SS7) [3,4] adopted and published by the International Consultative Committee for Telephone and Telegraph (CCITT).

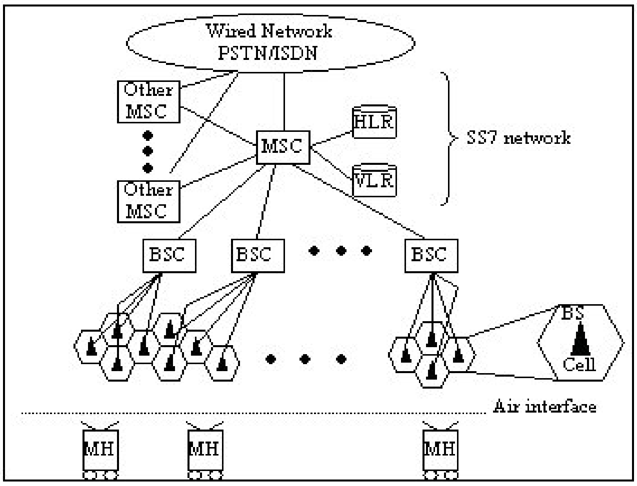

Figure 2-3. The architecture of PCS network

Because the network is divided into cells and the mobile user moves in the system time by time, how to keep the location information about the users is an important mission in PCS. In GSM, HLR/VLR (Home Location Register/Visitor Location Register) location management scheme is used. All the users' location information is kept by HLR and VLR. HLR is a database that stores all the system users' profile and contains a pointer to the VLR to assist in routing incoming calls. In generally, a PCS network has a HLR and one or more VLRs. HLR can be implemented as a distributed database system. To save the cost of signal transmitting, each MSC has its own VLR. The HLR/VLR location registration scheme will be explained in the following paragraphs.

The HLR/VLR location registration is used in IS-41 [3] and GSM [4]. Although there are some differences between two systems in control signaling, the registration spirit is the same. In the following paragraph, the location registration based on IS-41 protocol will be described. The registration processes are shown in Figure 2-4.

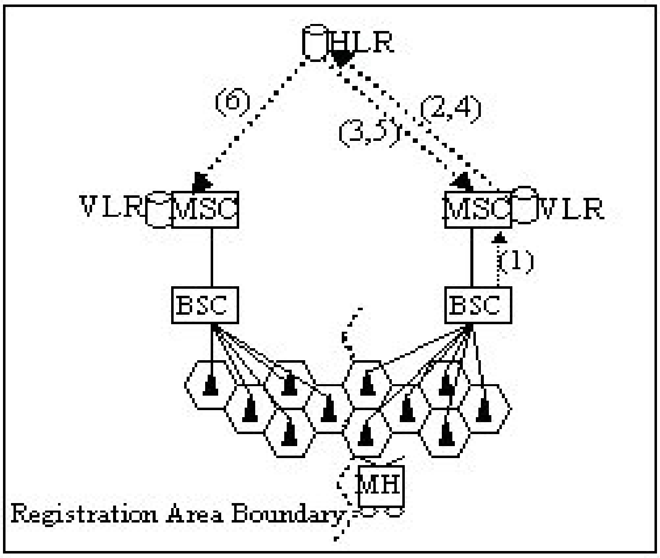

Figure 2-4. Signal flow diagram for registration

Step 1: The mobile host (MH) moves into a new registration area (crosses the registration area boundary, the dot line in Fig. 2-4) and sends a registration request to the MSC of this area by the serving base station of the new cell.

Step 2: The MSC sends the AUTH_REQ (authentication request) message to its VLR to authenticate this MH. Because the MH hasn't any registration record in this VLR, VLR sends AUTH_REQ to its HLR.

Step 3: The HLR sends its response in the AUTH_REQ message.

Step 4: If the MH is authenticated, the MSC sends a REG_NOTI (registration notification) message to its VLR. And then, VLR in turn sends a REG_NOTI message to the HLR serving the MH. HLR updates the registration record of this MH to point the new serving MSC/VLR.

Step 5: The HLR sends a response back to the VLR, which may contain relevant parts of the MH's service profile. The VLR stores the service profile in its database and also responds to the serving MSC.

Step 6: If the MH has an old registration record in previous registration area's VLR, the HLR sends a REG_CANC (registration cancellation) message to the previous VLR. On receiving this message, the previous VLR erases all entries for this MH from the record and sends REG_CANC message to the previous MSC, which then erases all entries for this MH from its memory.

The goal of mobile host's registration is to track the location where the MH is visiting now. When a call arrives for a specific MH, the MSC has to establish the routing path for this request. The signal flow diagram of call establishment is shown in Figure 2-5.

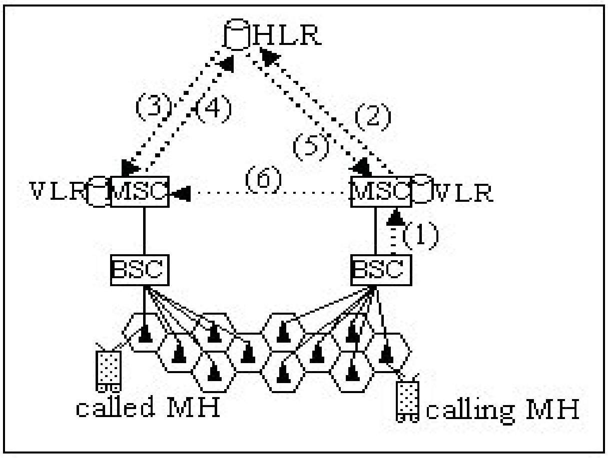

Figure 2-5. Signal flow diagram for call delivery

Step 1: Dialing up the called MH's mobile identification number (MIN), the calling MH initiates a connection request to its MSC.

Step 2: The MSC determines the associated HLR serving the called MH and sends a LOC_REQ (location request) message to the serving HLR.

Step 3: Through the registration process, the serving HLR determines the serving VLR of the called MH and sends a ROUTE_REQ (routing address request) to this serving VLR. And then, the VLR forwards this message to the serving MSC.

Step 4: The MSC allocates a temporary identifier, called a temporary local directory number (TLDN), to the called MH and returns a response to the HLR containing this information.

Step 5: The HLR forwards this information to the originating MSC in response to its LOC_REQ message.

Step 6: The originating MSC requests a call setup to the called MH's serving MSC by SS7 signaling network using the usual call setup protocols.

To enhance PCS, many standardizations have been released. In 1997, the ETSI proposed the standardization of general packet radio service (GPRS) [5,6] for GSM system.

GPRS is a step between GSM and 3G cellular networks. GPRS offers faster data transmission via a GSM network. The data transmission rate is improved from 9.6Kbits to 115Kbits. This new technology makes it possible for users to make telephone calls and transmit data at the same time. In 1998, the short message service (SMS) point-to-point (PP) [7,8,9] was released by ETSI. With SMS, mobile users are able to exchange alphanumeric messages (up to 160 characters) with other mobile users over digital cellular networks, almost anywhere in the world. Although SMS is a good service for delivering short text messages from one mobile user to another, there is clearly a need for a more advanced messaging service allowing mobile users to send and receive longer messages with richer contents. This new messaging service called multimedia messaging service (MMS) [10,11] is being standardized by the 3GPP (3rd Generation Partnership Project) since 1999. Besides the alphanumeric messages, this service enables the mobile users to send multimedia messages to one another.

The applications in PCS are numerous. However, the PCS is based on the wireless networks. When a mobile user moves to the border of a cell, the mobile user must handover to the target base station to continue the connection. However, sometimes the handover request will be rejected due to the insufficient resource in the target cell. How to ensure the resource is enough to handle the handover request and how to line up the handover request in the base station has been widely discussed [33,34].

In non-prioritized call handling schemes, the handoff requests are treated in the same manner as the new call requests so that the probability of handoff failure equals the probability of call blocking. In order to decrease the probability of handoff call failure, there are many methods. We classify the proposed schemes into the following types:

1) Call Admission Control Schemes

2) Guard Channel Schemes

3) Channel Reservation Schemes

4) Handoff Queuing Schemes

The Call Admission Control Schemes restrict the number of new calls accepted to decrease the probability of handoff call failure. The paper by Naghshineh and Schwartz [20] proposed a distributed call admission control scheme that takes the number of calls in adjacent cells into consideration when making a call admission decision. An admission control scheme based on the population of neighboring cells is proposed in [21]. The *shadow cluster* concept has been proposed to estimate future resource requirements and to perform call admission decisions in wireless networks in [22]. However, no real method is proposed to determine where the shadow cluster will be. Aljadhai and Znati [23] proposed a *Most Likely Shadow Cluster* framework to support predictive timed-QoS guarantees in wireless networks. Yu and Leung [24] proposed using Ziv-Lempel data compression algorithm to predictive the user mobility, which is then used in the call admission control.

The Guard Channel Schemes reserve a fixed or dynamically adjustable number of channels exclusively in every cell for handoff requests. Yoon and Un [25] presented and compared three call handling schemes, one with guard channels, for a base station to handle new calls and handoff calls. The results showed that the schemes without guard channels have a good call blocking probability without a severe penalty on handoff calls. A novel dynamic guard channel scheme has been proposed in [26], which adapts the number of guard channels in each cell according to the current estimate of the handoff call arrival rate derived from the current number of ongoing calls in neighboring cells and the mobility pattern. Sivalingam et al. [27] investigated static and dynamic resource allocation schemes. The dynamic scheme probabilistically estimates the potential customers that will be handed off from neighboring cells. The result, as to be expected, indicates that dynamic schemes are better than static schemes.

The Channel Reservation Schemes reserve bandwidth only in these cells where the mobile users are expected to visit in the near future. The real problem lies in how to determine the cell that a user will visit in the future. Oliveira et al. [28] proposed to reserve bandwidth in all neighboring cells of the currently residing cell. Oliver and Borras [29] studied a variable reservation scheme. Each neighboring cell has different amount of reservation. The amount will depend on the proximity of the mobile station to the cell border, its moving speed, and its mobility pattern. Chang et al. proposed several schemes to predict the moving trajectory of a mobile host. The methods include linear prediction and group prediction [69], statistical prediction [62], and prediction based on historical moving trajectories [63]. Awduche and Agu [61] also used the historical data for estimating the trajectory. But they do not specifically point out how to do it. Chiu and Bassiouni [65] use the Global Positioning System (GPS) to predictive the user's next position and make bandwidth reserve in advance. To accurately reserve bandwidth in the right cell, some papers [30,31] try to characterize the trajectory and the mobility pattern to predict the location of mobile users in the near future.

Hsu et al. [67] distinguished the cells into four types: active, candidate, neighbor, and remaining. Each type of cell has different probability of being migrated. Depending on the probability of success that a user wants, resources in a sufficient number of cells are reserved. Mahadevan and Sivalingam [71,72] proposed that we do not pre-reserve all the way to the sender in the neighboring cells. Instead, we only reserve the resources between the current base station and the neighboring base stations.

If resources are reserved in advance, the handoff-dropping rate will decrease. However, other performance measures such as new call blocking rate will increase. To remedy this, most papers classify the reserved resources into two types. One is called active [62,68,70,73], or committed [61], or conventional [64], or in-use [66], where the resources are currently used by the mobile host. The other is called passive, or quiescent, or predictive, or reserved, respectively, where the resources are reserved but not used. To increase the resource utilization, most papers [61,62,63,64,66,70,73] allow the reserved but unused resources to be used by other hosts temporarily. However, they will be preempted when the mobile host needs the resources.

Handoff Queuing Schemes discusses how to line up the handoff requests. Tekinay and Jabbari [32] presented a Measurement Based Prioritization Scheme, MBPS, to employ a dynamic priority queuing discipline instead of FIFO. Ebersman and Tonguz [33] proposed a Signal Prediction Priority Queuing (SPPQ) scheme to improve MBPS algorithm by using both RSS and the change in RSS (ΔRSS) to determine the priority ordering in the handoff queue. Bang et al. [34] studied a system where there are n types of traffic and each type is further divided into handoff traffic and new traffic. There is a single buffer shared by all traffic types. Each has its threshold for buffer space. Once the buffer level exceeds a type's threshold, new arrival for this type is discarded. In another papers [35,36], the same authors presented a Selective-Delay Push-In-Buffering mechanism for multimedia wireless ATM system providing QoS guarantees for handoffs. In this scheme, once a buffer threshold is reached, delay sensitive traffic can replace delay tolerant traffic already in the buffer.

In this thesis we will propose an approach to guarantee the QoS of multimedia transmission in the wireless personal communication system. Our method is based on the received signal strength (*RSS*), received signal strength degradation rate (*ΔRSS*) and the handoff priority of the traffic to line up the handover request. The handoff request with the highest handover priority will be first served by the base station. This topic will be discussed in Chapter Three.

2.2 Ad hoc Networks

The IEEE 802.11 subcommittee defines two types of networks: ad hoc network and client/server network. The ad hoc network is a network where communications are established directly between mobile nodes, without any access point or server. The client/server network uses the access point that controls and allocates the wireless resource for all mobile nodes. The mobile users are allowed to roam from cell to cell in this network. And the access point is also used to interface the mobile node to the wired or wireless backbone. In this section, we will concentrate our attention on the ad hoc networks.

An ad hoc [12,13] network is a collection of communications devices (mobile hosts) that wish to communicate, but have no fixed infrastructure available, and have no pre-determined organization of wired link (such as PSTN in PCS network). This capacity of mobile host will be very valuable in setting up a routing path. However routing protocols are more difficult to implement in this case as each mobile host must act as a router. An ad hoc network is shown in Figure 2-6. Depending on the radio power, every mobile host has it own transmission range. We assume the distance between mobile host *2* to mobile host *3* is too far to communicate with each other in Figure 2-6. If the mobile host *2* wants to communicate with mobile host *3*, it must send the packet to other mobile hosts, which can forward this packet to mobile host *3*. For example, mobile host *4* forwards this packet to mobile host *1*, and then mobile host *1* forwards this packet to mobile host *3*. So, mobile host *1* and mobile host *4* work as routers in this communication.

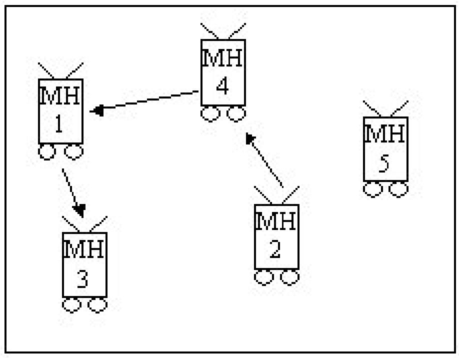

Figure 2-6. An example of ad hoc network

The history of ad hoc networks can be traced back to 1972 when the DoD sponsored the Packet Radio Network (PRNET), which evolved into the Survivable Adaptive Radio Networks (SURAN) program in the early 1980s. The PRNET used a combination of ALOHA and CSMA approaches for medium access, and a kind of distance-vector routing. In 1987, the IEEE 802 standards committee formed the 802.11 Wireless Local Area Networks Standards Working Group [16]. In 1994, the idea of an infrastructureless collection of mobile hosts was proposed in two conference papers [14,15], and IEEE 802.11 adopted the term “ad hoc networks”. In 1997, a global standard for radio equipment and networks operating in the 2.4 GHz unlicensed frequency band, with data rates 1 and 2 Mbps was obtained. In 1999, IEEE 802.11b with data rates up to 11 Mbps was adopted. Currently, IEEE 802.11a is looking into even faster rates (54 Mbps) in the 5 GHz band. Other IEEE 802.11 activities are shown in Table 2-1.

Table 2-1. Wireless related IEEE 802.11 working group

Group	**Task**
802.11	Wireless LAN (WLAN) working group
802.11 MAC	MAC for WLAN
802.11 PHY	Three PHY's: IR, 2.4 GHz FHSS and 2.4 GHz DSSS
802.11a	54 Mbps PHY
802.11b	Higher rate (11 Mbps) PHY
802.11c	Collaborate with 802.1 group
802.11d	Physical layer in new markets
802.11e	MAC enhancements
802.11f	Access points interoperability
802.11g	Higher 802.11b speeds
802.11h	Enhance 802.11 MAC and 802.11a PHY
802.11i	Enhance 802.11 MAC security
802.11 SG	Placement in Standards
802.11 5GSG	Globalization of 5GHz
802.11 PC	Publicity
802.11 R-REG	Regulatory issues
802.15	Wireless Personal Area Network (WPAN) working group

The subcommittee 802.15 is working on the Wireless Personal Area Network

(WPAN), which brings a new standardization called Bluetooth. Bluetooth (named after the Viking king who unified Denmark and Norway in the 10^{th} century) is an open standard for short-rang ad hoc wireless voice and data networks, operating in the unlicensed 2.4 GHz frequency band. Bluetooth was originally conceived by Ericsson in 1994. In 1998, Ericsson, Nokia, IBM, Intel and Toshiba formed a special interest group (SIG) to expand the concept and to develop a standard under IEEE 802.15. Currently, over 2000 companies are participating in the Bluetooth SIG, and many are developing Bluetooth products.

The physical layer of ad hoc networks uses high-frequency electromagnetic waves, either infrared (IR) or radio frequency (RF), to transmit information from one point to another. It is generally agreed that RF will be more practical than IR in home and office networking, since it can propagate through solid obstacles. The media access control (MAC) layer is based on carrier-sense multiple access with collision avoidance (CSMA/CA) [17]. The CSMA/CA is similar to the 802.3 Ethernet wired-line standard, called CSMA/CD. This protocol prevents collisions instead of detecting them.

IEEE 802.11 carrier sensing shall be performed both through physical and virtual mechanisms. Physical carrier sensing detects the medium to see whether the medium is busy or not. The virtual carrier sense mechanism is achieved by distributing and announcing reservation information before the data transmission. The exchange of Request-To-Send (RTS) and Clear-To-Send (CTS) frames prior to the actual data frame advertises this medium reservation information.

The physical carrier sensing media access mechanism is depicted in Figure 2-7. If a station wants to transmit a frame, it senses the channel to see whether medium is busy or not. If it is busy, it waits until medium becomes idle for DIFS period and then computes a random back-off time. If the timer reaches zero, the station transmits its packet. If channel becomes busy before timer reaches zero, then the timer is suspended.

The time interval between frames is called the inter-frame space (IFS). Four IFS intervals are specified in the standard. From the shortest to the longest, they are Short-IFS (SIFS), Point Coordination Function-IFS (PIFS), Distributed Coordination Function-IFS (DIFS), and Extended-IFS (EIFS). A station that only requires waiting for an SIFS has higher access priority than those stations required to wait a PIFS or DIFS or EIFS before transmission.

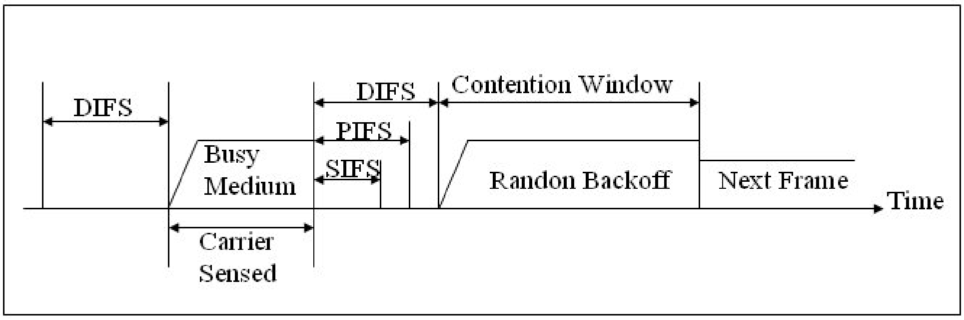

Figure 2-7. Media access timing of 802.11

The virtual carrier sense mechanism can avoid the hidden node problem, shown in Figure 2-8. Because mobile nodes *A* and *C* can not hear from each other, it will occur that the node *C* starts transmitting to node *B* while node *A* is also transmitting to node *B*. The collision occurs in node *B*. The CSMA/CA uses RTS/CTS frames to avoid this problem. The 4-way handshake [18] mechanism is shown in Figure 2-9. Using the RTS/CTS packet the CSMA/CA avoids the collision in advance. Before sending the data packet, the sender sends the RTS packet and waits to receive the CTS packet from receiver.

If the CTS packet is received in the time period, the data packet is sent and then waiting for hearing its ACK message from the receiver.

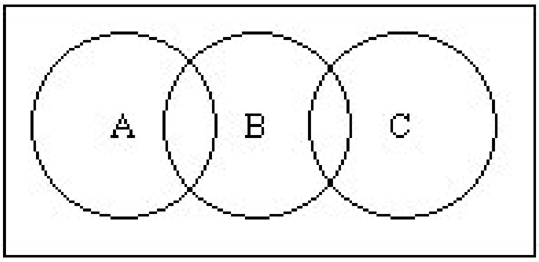

Figure 2-8. Hidden node problem

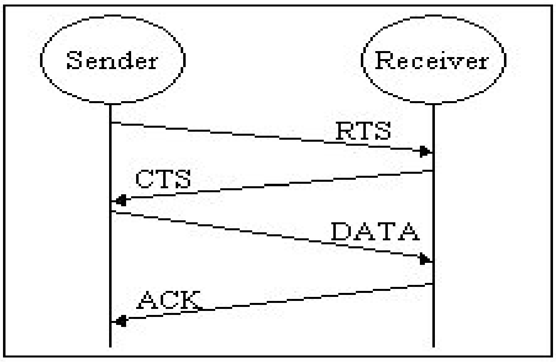

Figure 2-9. 4-way handshake

The network layer needs to determine and distribute information used to calculate routing path. The routing protocols for ad hoc networks have been widely discussed, such as DSDV [41], CGRS [42], WRP [43], AODV [44], DSR [45], TORA [46], ABR [47], SSR [48], ZRP [49], FSR [50], etc. In an ad hoc network environment, the routing protocol must keep up with the changing topology. Because of the mobility and infrastructureless characteristics, ensuring the routing path effectiveness and cutting the maintenance cost are the great challenges for ad hoc networks.

According to the routing message maintenance types, these routing protocols may generally be categorized as table-driven and on-demand driven. Besides the pure routing protocol, we also discuss the related QoS supported routing protocols. The categorization of these routing protocols is shown in Table 2-2. The following paragraphs briefly describe the protocols.

Table 2-2. Categorization of ad hoc routing protocols

Categorization	**Protocols**
Table Driven	1.Destination-Sequenced Distance-Vector Routing (DSDV) 2. Clusterhead Gateway Switch Routing (CGSR) 3. Wireless Routing Protocol (WRP)
On-Demand Driven	1. Ad-hoc On-demand Distance Vector Routing (AODV) 2. Dynamic Source Routing (DSR) 3. Temporally-Ordered Routing Algorithm (TORA) 4. Associativity-Based Routing (ABR) 5. Signal Stability Routing (SSR)
Hybrid	1. Zone Routing Protocol (ZRP) 2. Fisheye State Routing (FSR)
QoS Supporting	1. Core-Extraction Distributed Ad Hoc Routing Algorithm (CEDAR) 2. Flow Oriented Routing Protocol (FORP) 3. AODV Reliable Route Selection (AODV-RRS) 4. Ad hoc network with Mobile Backbones (MBN) 5. Adaptive QoS routing scheme based on local performance 6. Genetic Algorithm based routing for Ad hoc Networks (GAMAN) 7. Backup Routing in Ad hoc Networks (AODV-BR)

2.2.1 Table Driven Routing Protocols

The table driven routing protocols (also known as proactive protocols) attempt to maintain consistent, up-to-date routing information from each node to every other node in the network. These protocols require every node to maintain one or more routing tables and have to propagate its routing table contents throughout the network in order to have a consistent network view for every node. When a new connection request arrives, the mobile node can find out a route path from its routing table directly. Some of the table driven routing protocols are described below.

Destination-Sequenced Distance-Vector Routing (DSDV) [41] is a table driven routing protocol based on the Bellman-Ford routing mechanism [56]. Every mobile node in the network maintains a routing table in which all the possible destinations in the network and the number of hops to each destination are recorded. The contents of routing table are exchanged periodically.

Clusterhead Gateway Switch Routing (CGSR) [42] is a cluster based DSDV routing protocol. A cluster head controls a set of mobile nodes. In a cluster, the DSDV routing protocol is used to maintain the routing path. A gateway is the bridge of two or more cluster heads. A packet sent by a node is first routed to the cluster head by the DSDV routing protocol, and then the packet is routed from the cluster head through the gateway to another cluster head, and so on until the cluster head of the destination node is reached. This concept is shown in Figure 2-10.

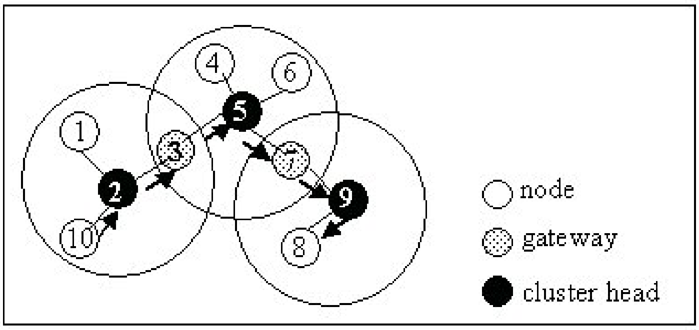

Figure 2-10. CGSR routes from node 10 to node 8

In the Wireless Routing Protocol (WRP) [43], each node in the network is responsible for maintaining four tables: (a) distance table, (b) routing table, (c) link-cost table, and (d) message retransmission list (MRL) table. Mobile nodes send update messages after processing updates from neighbors or detecting a change in a link to a neighbor. The neighbors then update their distance table entries and check for new possible paths through other nodes. Any new paths are relayed back to the original nodes, so they can update their tables accordingly.

2.2.2 On-demand Driven Routing Protocols

The on-demand routing protocols (also known as reactive protocols) create routes only when desired by the source node. When a node requires a route to a destination, it initiates a route discovery process within the network. Once a route has been established, it is maintained by some form of route maintenance procedure until the destination is unreachable or the route is no longer desired.

Ad-hoc On-demand Distance Vector Routing (AODV) [44] protocol builds on the DSDV algorithm. It is a pure on-demand protocol, as nodes that are not on a selected path do not maintain routing information or perform routing table exchanges. When a source node wants to send a message to some destination node and does not have a valid route to that destination, it broadcasts route request (RREQ) packets to its neighbors. The intermediate nodes re-broadcast the RREQ packet until the destination node receives the RREQ. Upon receiving the RREQ packet, the destination node sends the route reply (RREP) packet back to the neighbor from which it first received the RREQ. Figure 2-11 shows the concept of the routing path establishment process in AODV protocol. Another protocols, the Dynamic Source Routing (DSR) [45], is very similar to AODV. The request packets of DSR contain a list of mobile nodes that have seen the packet.

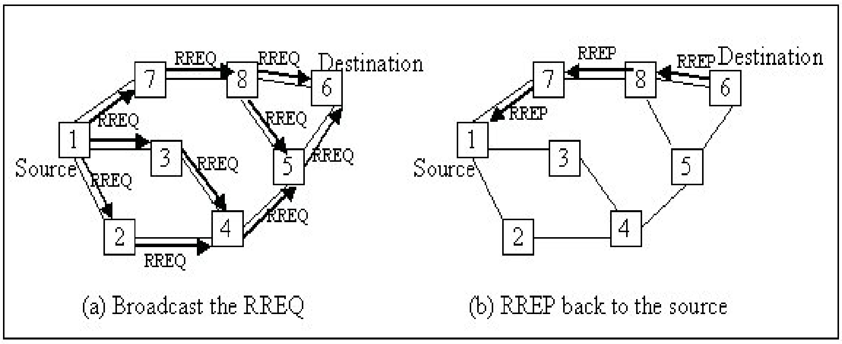

Figure 2-11. AODV path discovery procedure

Temporally-Ordered Routing Algorithm (TORA) [46] is based on the concept of link reversal. During the route establishing and maintenance phases, each mobile node in the system uses a "height" metric to establish a directed acyclic graph (DAG) rooted at the destination. Using the links of DAG, the routing path from the source to the destination is found.

In order to discover a more stable path, the degree of association stability opinion is used in Associativity-Based Routing (ABR) protocol [47]. The essence of ABR lies on the fact that a mobile host's association with its neighbor changes as it is migrating. Its transiting period can be identified by the associativity "ticks". There are stable and unstable ticks. The best route must be computed based on the degree of association stability.

The Signal Stability Routing (SSR) protocol [48], which selects routes based on the signal strength between nodes and on a node's location stability, contains two cooperative protocols: the Dynamic Routing Protocol (DRP) and the Static Routing Protocol (SRP). This method maintains a routing tables and the signal stability table. The signal strength recorded in the signal stability table is characterized as strong or weak by the average received signal strength in the past few beacons.

The Zone Routing Protocol (ZRP) [49] is for the re-configurable, large scale, and highly mobile ad hoc networking environment. Through the use of the zone radius parameter, the scheme exhibits adjustable hybrid behavior of proactive and reactive routing schemes. Each node is required to know the topology of the network within its routing zone only and nodes are updated about topological changes only within their routing zone. The routing procedure of inter-zone is like the on-demand protocols and the intra-zone routing procedure is like the table-driven protocols. A similar method, the Fisheye State Routing (FSR) algorithm [50], introduces the notion of multi-level "scope" to reduce routing update overhead in large networks.

2.2.3 QoS Supported Routing Protocols

Generally speaking, the topics of Quality of Service (QoS) may be related to bandwidth, throughput, packet delay, delay jitter, packet lose rate, etc. Having no fixed network infrastructure, QoS support in ad hoc network reveals more difficulties in many manners [51]. The related papers discussing the QoS in ad hoc routing protocol are depicted as follows.

A Core-Extraction Distributed Ad Hoc Routing Algorithm (CEDAR) [52] uses three key components: a) the establishment and maintenance of a self-organizing routing infrastructure; b) the propagation of the link state of high bandwidth and stable links; and c) a QoS-route computation algorithm to support the QoS routing in ad hoc networks. The Flow Oriented Routing Protocol (FORP) [53] uses the mobile's moving direction, speed, and transmission range to predict the link expiration time, and then find out a route path according to this link expiration time. Depending on the link expiration time concept, QoS support by bandwidth reservation is proposed in [54]. If the available bandwidth on the path can't fit the bandwidth requirement, the connection request will be rejected. Using the Global Positioning System (GSM), AODV Reliable Route Selection (AODV-RRS) [65] introduces the concept of *stable zone* and *caution zone* to discover a more stable routing path in ad hoc networks.

Xiao et al. proposed a QoS routing protocol for ad hoc network with Mobile Backbones (MBN) [74]. This routing algorithm is modified from the k-shortest path algorithm and the limited path Dijkstra's algorithm. Barolli et al. [75] proposed a Genetic Algorithm (GA) based routing method for Mobile Ad hoc Networks (GAMAN). Based on Fuzzy Logic, Neural Networks, the GAMAN is a source-based routing algorithm which can find out a more robust routing path. An adaptive QoS routing scheme based on the prediction of the local performance in ad hoc networks is proposed by Sun and Hughes [76]. It is implemented by link performance prediction procedure. Integrated QoS performance in each local area is estimated based on translating the effects of the lower layer parameters into the link state information.

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Backup Routing in Ad hoc Networks (AODV-BR) by Lee and Gerla in [77] was proposed for alleviating packet delay problem while rediscovering a new routing path by intermediate backup nodes. Those backup nodes are arranged when route discovery phase and would forward packets automatically if they detect the original radio link failure. For improving network stability and total throughput, multiple paths routing protocols are proposed [78,79]. The total bandwidth of those paths cannot just be sum up because of "interference". The paper [80] discussed the "available bandwith" network capacity and "interference" according to different Media Access Control (MAC) protocols. The Race-Free Bandwidth Reservation Protocol [81] is proposed for parallel bandwidth reservation in ad hoc networks. This algorithm relies on the maintenance of three-state slot status information (*free/allocated/reserved*) at each node, synchronous and asynchronous slot status updates to 1-hop and 2-hop neighbor nodes, wait-before-reject strategy, TTL timers for allocated slots, maximum QREQ node wait time, and max QREQ/QREP total wait time.

In order to fit the multimedia communication requirement, QoS supporting in ad hoc routing protocol is taken into consideration in recent researches. The QoS supported routing protocol is a complex and difficult issue because of the dynamic nature of the network topology and imprecise network state information. In this thesis, we propose a QoS supported routing protocol for ad hoc networks, called AODV-RSS (Ad hoc On-demand Distance Vector routing protocol with Received Signal Strength). This algorithm is described in Chapter Four.

Chapter 3 Handoff Ordering Using Signal Strength for Multimedia Communications in Wireless Networks

3.1 Introduction

Advances in microelectronic technologies have make our electronic gadgets become smaller and smaller. As such, it is easier to carry computer, phones, personal digital assistants, and many other devices around. Therefore, to connect all these devices together all the time, wireless networks become a necessity. A typical infrastructure for wireless networks is organized into geographical regions called cells [19]. The mobile users in a cell are served by a base station. Future wireless networks, however, will have to provide support for multimedia services (video, voice, and data). As such, it is important that the network provides quality-of-service (QoS) guarantees. However, satisfying the QoS guarantees is hard due to user mobility. When a mobile user moves from a cell to another, if the new cell does not have enough resource to accommodate the handoff user, his/her service will be disrupted.

Therefore, to maintain a consistent service of a user, either a sufficient resource must be reserved in each cell or we must handle the handoff users selectively such that the high priority user can get a better service. In this chapter, we will focus on the handoff procedure of the wireless networks. In Figure 3-1, there are two cells served by base station *A* and base station *B*. These two cells have overlapping area *C* denoted as the shaded region. Assume a mobile user *M* move from base station *A* toward base station *B*. The RSS (Received Signal Strength) from this user viewed by base station *A* is decreasing (curve *A* in Fig. 3-1). On the other side, the RSS viewed by base station *B* is increasing (curve *B* in Fig. 3-1). When the RSS in base station *A* is below the handoff-threshold level (point *H* in Fig. 3-1), a handoff request is sent to base station *B*. Base station *B* will handle the handoff request from time t_1 to t_2. If base station B cannot find enough available bandwidth for allocating to the mobile user M in this period and the RSS in base station A degrades to the receive-threshold level (point R in Fig. 3-1), the on going call of mobile user M will be force-terminated.

How to ensure that base station B (in Fig. 3-1) has enough bandwidth to handle the handoff call request has been widely discussed. We will review these methods in next section. In this chapter, we will propose a handoff request queuing ordering scheme for multimedia wireless networks. In our method, every base station has a handoff request queue to line up the handoff requests. We use a priority scheme to order the mobile user's handoff request. The handoff priority for every multimedia service is calculated using three values: the static priority value, the degradation rate of the received signal strength (ΔRSS) and the RSS level itself. We then analyze its performance in a 25-cell network. Finally, simulation results indicate it can effectively reduce the handoff call dropping probability compared to other methods.

The remainder of this chapter is organized as follows. Our method is described in Section 3.2. The simulation model and analysis are shown in Section 3.3. In Section 3.4, we demonstrate the simulation results. Finally, Section 3.5 concludes this chapter.

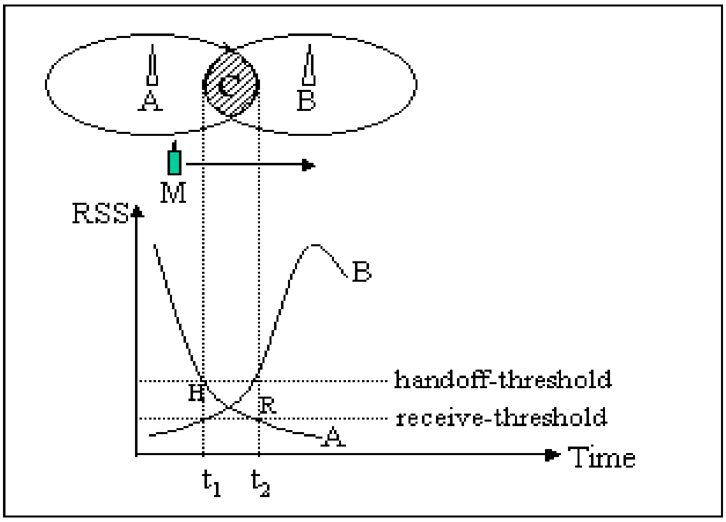

Figure 3-1. Handoff and RSS

3.2 Signal Strength for Multimedia Communication Scheme

In this chapter, we will extend the SPPQ algorithm to handle the multimedia traffic. Our method uses the change in received signal strength to offer a handoff priority value for different service classes. The following paragraphs will discuss this method in detail.

If the target base station has no available bandwidth to serve a new request, queuing the handoff request is possible because the mobile users spend some period of time in the overlapping area between two cells (as previously stated). In this time period, the target base station must deal with many things such as new call requests, handoff call requests, call terminations and so on. Each event changes the conditions of a base station and the RSS of every handoff request is changing too. When an on going call is finished, the occupied bandwidth will be released. Our goal is to decide who has the highest priority to use the released bandwidth and how to line up these handoff requests in the handoff queue of a base station.

In this chapter, we propose a Signal Strength for Multimedia Communications Scheme, abbreviated as SSMC. SSMC offers a line up method for handoff requests in the multimedia wireless system. In SSMC, if all channels of a base station are occupied, the new call requests within that cell are simply blocked and the handoff requests to that cell are queued. Requests are queued according to a priority value. To calculate the priority, first assume that there are *k* types of different service classes in the wireless system. The priority for the service class *i* is p_i, where $1 \le p_1 < p_2 < \cdots < p_i < \cdots < p_k$ for $1 \le i \le k$. In the time period *t1* to *t2* in Fig. 3-1, the $\Delta RSS(j)$ for a handoff request *j* is measured by the following formula:

$$\Delta RSS(j) = \left| \frac{RSS_{t2}(j) - RSS_{t1}(j)}{t2 - t1} \right|$$

Then the handoff priority for handoff request *j*, with service class priority p_j, is calculated as $P(j) = p_j * \Delta RSS(j) * \frac{1}{RSS(j)}$.

The value of ΔRSS is the absolute value of slope of the decreasing part of RSS curve. If it is large, it means that the received signal strength changes rapidly. A possible reason may be that the mobile is moving at a very fast speed. Therefore, the current cell will lose track of it and it needs to be handed off quickly. Similarly, if a mobile is leaving a base station, $RSS(j)$ will get smaller. Then $1/RSS(j)$ will get larger to promote its priority. Therefore, the handoff request with a higher static service class priority or a larger ΔRSS value or a smaller *RSS* value will get a higher priority than other handoff requests.

In SSMC, when the base station has an available bandwidth, the handoff request with the highest handoff priority, $P(j)$, will be served first. The SSMC method is a non-preemptive (once a request is served, it will not be disrupted until finished), dynamic priority discipline. ΔRSS and RSS levels are monitored continuously, and the priority of a handoff request dynamically changes depending on $\Delta RSS(j)$ and RSS, while waiting in the queue. The handoff requests waiting for the bandwidth in the handoff queue are sorted continuously according to their priorities. It can be seen that our approach is simple and does not require much overhead and computation in the part of base stations.

Figure 3-2 shows the flowchart of SSMC. If there has no enough bandwidth in a cell, a new call request will be simply dropped. After a new call is accepted, the RSS level of this call is monitored continuously. When the RSS level is lower than the handoff-threshold level, a handoff call request is proposed to the target cell where the mobile user is heading. If there is no enough bandwidth, the handoff request will be put into the target cell's queue and sorted by the priority, $P(j)$. In this waiting period, the value $P(j)$ of all requests queued is calculated and sorted continuously. If there has free bandwidth, the highest priority handoff request gets the channel.

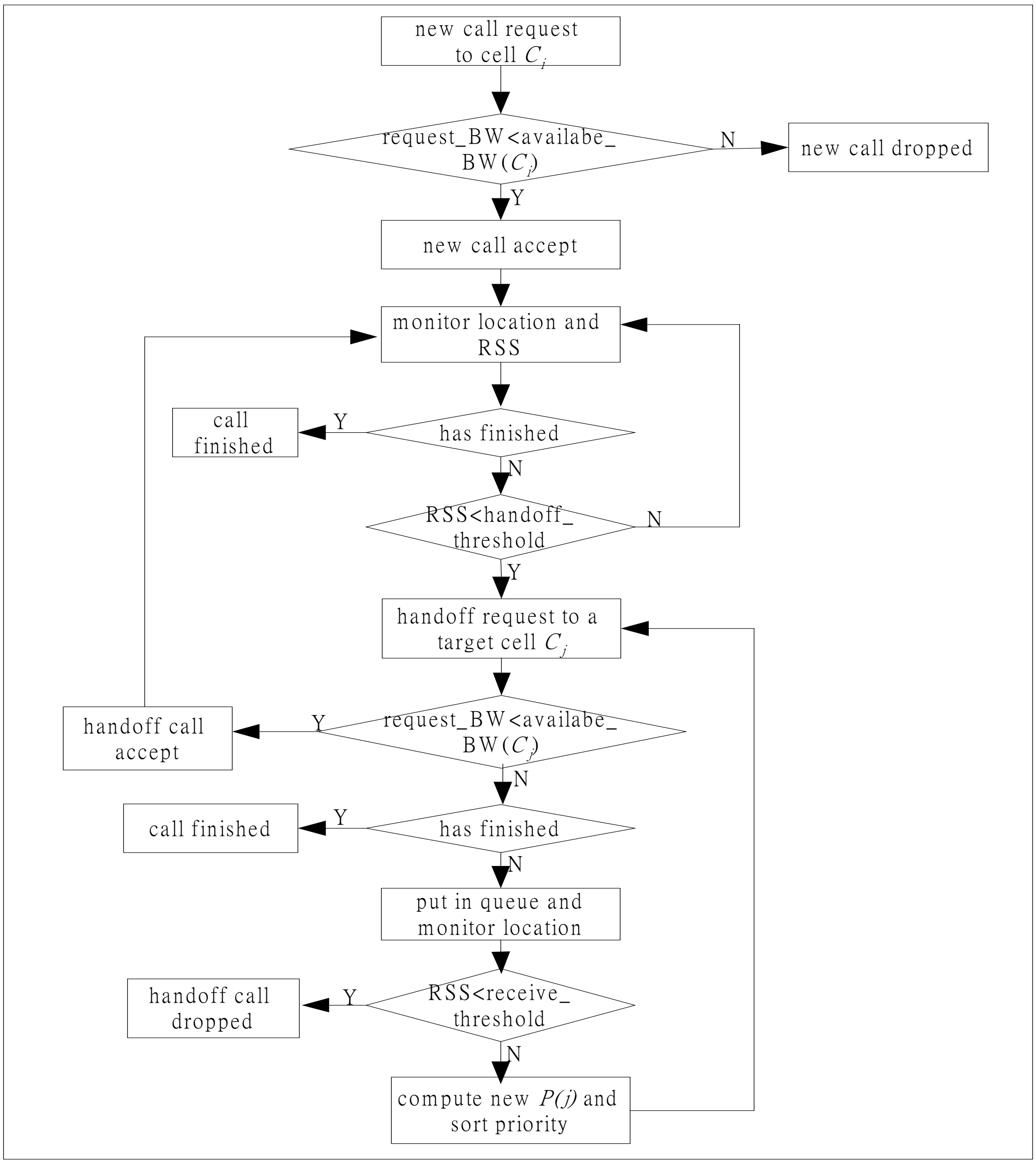

Figure 3-2. The flowchart of SSMC

In the following section, we will first analyze the handoff situation in a 25-cell network. We then verify its performance by simulations. As the simulation results shown, our method will reduce the handoff call dropped probability effectively.

3.3 Simulation Model and Analysis

Unlike the simulation in [38,39] that used only one cell, our network system includes twenty-five cells as shown in Fig. 3-3. The area of one cell is 4*4 km^2. We assume the top cells (cells 21, 22, 23, 24 and 25) and the bottom cells (cells 1, 2, 3, 4 and 5) are connected. That is, if a user comes out of cells 21 from top, he will come into cell 1. Analogously, we assume the left cells (cells 1, 6, 11, 16 and 21) and right cells (cells 5, 10, 15, 20 and 25) are connected too.

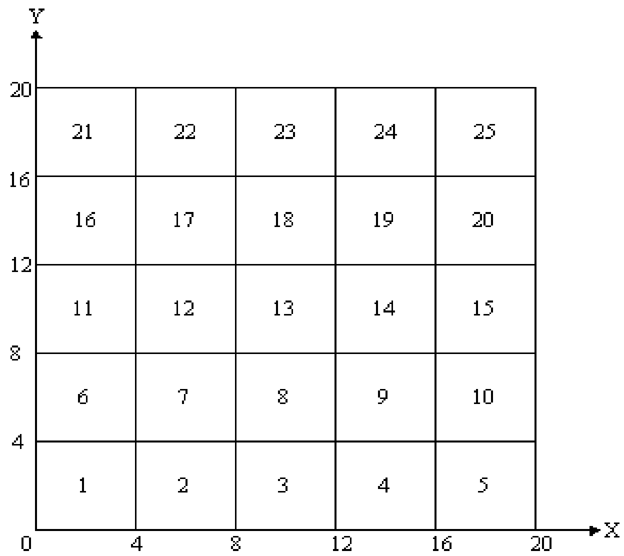

Figure 3-3. The simulated wireless network

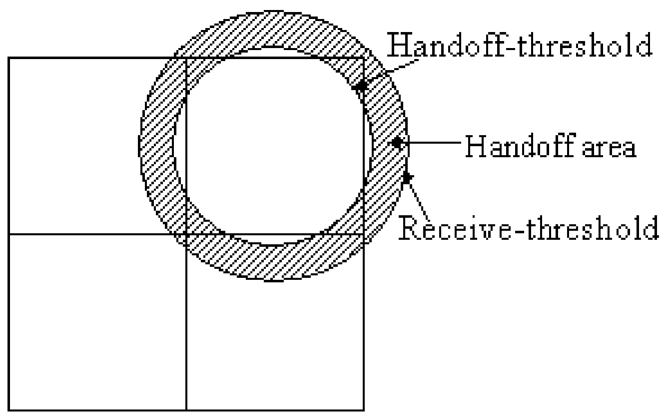

Figure 3-4. Handoff threshold and receive threshold

Figure 3-4 shows the concept of handoff-threshold and receive-threshold setting. Assume the base station of each cell is at the center of the square, we set the receive-threshold to 2.9 km (about half of the diagonal length) in order to cover all the cell area. The handoff-threshold can be set at any distance between cell-center to receive-threshold. The area between handoff-threshold and receive-threshold is called handoff area (the shaded area of Fig. 3-4). We set the handoff-threshold to 2.7 km. If a mobile user moves at a speed of 60 km/hr, he/her will have 12 seconds for handling handoff before moving out of this handoff area [39].

The user mobility pattern is described as follows. When a new call request is accepted, the original location of the mobile user is a random variable of the whole network system, and the moving direction is set by a random angle between 0 degree and 360 degree. The moving speed is uniformly distributed between 30 km and 90 km. This assumption suits for a mobile user who is driving a car on the freeway or highway. In our simulation, we monitor the user's location and *RSS* at every second.

Our simulations in next section include three different handoff call handling strategies. The first strategy, labeled *No priority*, is a non-prioritized call handling scheme in which a handoff request will be simply dropped if the target base station has no available channel. The second strategy, labeled *FIFO*, is prioritized call handling scheme in which a handoff request will be saved in the queue with first-in first-out strategy if the target base station has no free channel. The third strategy is the proposed *SSMC* strategy.

We will first compare the results of the new call blocking probabilities between those computed by the queuing model [37] and those obtained by the simulations. For non-prioritized call handling scheme, the probability of handoff call dropping (P_d) equals the probability of new call blocking (P_b), which is given by the well-known Erlang B formula for the *M/M/c/c* queue.

$$P_b = P_d = \frac{\dfrac{(c\rho)^c}{c!}}{\displaystyle\sum_{i=1}^{c} \frac{(c\rho)^i}{i!}}, \qquad \rho = \lambda / c\mu$$

If we limit the connection duration time to 40 seconds, the distance that a connection call moves is at most 1 km (at the highest speed 90 km/hour). This concept is illustrated by figure 3-5. The new call arrival rate for cell 13 is $\lambda / 25$ when the whole system arrival rate is λ. Besides the new call arrival rate, cell 13 will handle the handoff requests from cells 7, 8, 9, 12, 14, 17, 18 and 19. So the total call arrival rate of cell 13 will include the new call arrival rate $\lambda / 25$ and the handoff calls from the neighboring cells. Since the call duration is 40 seconds, we extend the size of cell 13 by 1 km in each direction, which is shown in Fig. 3-5 by the shaded square area.

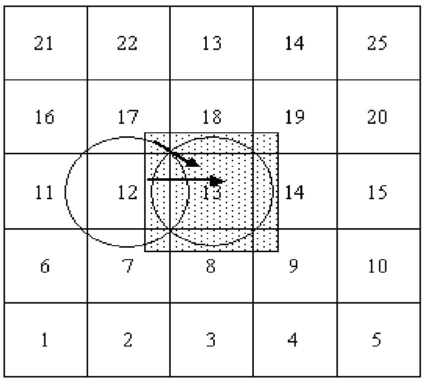

Figure 3-5. The call arrival rate analytical model

Under the 40 seconds connection duration constraint, a connection call in cell 12 that handoffs to cell 13 must start at the 1 km from right side cell boundary (the shaded area in cell 12). The probability that a call will start at this area is 1/4 for a 4*4 km^2 cell. Since handoff-threshold is 2.7 km from the cell center, the distance from the call starting point to the handoff-threshold boundary is 1.7 km. The probability that a connection call will request handoff is 1/1.7 at most. Because the moving direction of a mobile is uniformly distributed between 0 degree to 360 degree, the probability that a connection call will handoff to cell 13 from cell 12 is 1/4. By the above analysis, the probability that a connection call starting at cell 12 will handoff to cell 13 is $1/4*1/1.7*1/4 \approx 1/32$. The same analysis can be applied to cells 8, 14 and 18.

By a similar reasoning, the probability that a connection call starting at cell 17 (or 7, 9, 19) will handoff to cell 13 is $1/16*2/3*1/16$. Adding all the probabilities, the handoff request arrival rate is $13/96*\lambda/25$. Figure 3-6 shows the call blocking probabilities of non-prioritized scheme by simulation and the M/M/C/C results. In this simulation, there is only class 1 traffic and every cell has 50 channels.

In Fig. 3-6, M/M/C/C(1) is the analytic value of $13/96*\lambda/25+\lambda/25$ call arrival rate, and M/M/C/C(2) is for $\lambda/25$ arrival rate. Why are the simulation results close to the result of $\lambda/25$ arrival rate? This is because there are only a few number of handoff call requests in the wireless system under the simulation condition. Even if a call needs handoff, the remaining service time in the new cell is short.

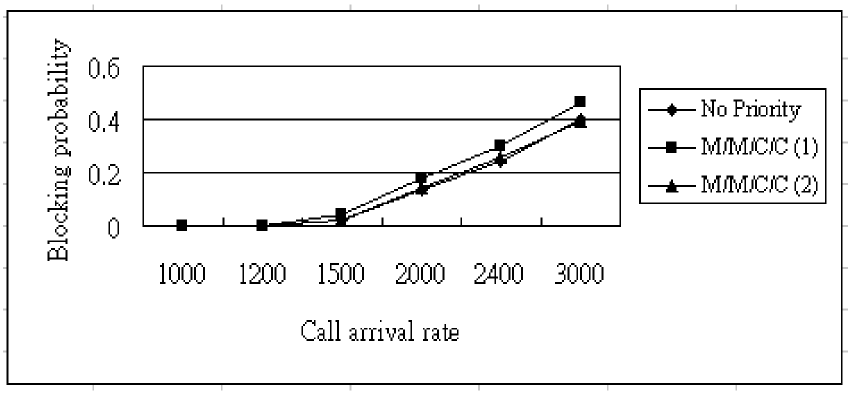

Figure 3-6. Call blocking probability of non-priority scheme vs. offered load (comparison between analysis and simulations)

The M/M/c queue is used to approximate the FIFO and SSMC queuing schemes. Until all channels are occupied, the arrival rate is the sum of new calls and handoff calls. Once all the channels are busy, only handoff calls are queued. The new call blocking probability of originating calls is simply given by the probability of the number of users in the system being equal to or more than the number of channels, c, i.e.,

$$P_b = \sum_{n=c}^{\infty} p_n$$
$$= \frac{1}{c!}\left(\frac{\lambda_c + \lambda_h}{u}\right)^c \left(\frac{cu}{c\mu - \lambda_h}\right) p_0$$

where λ_c and λ_h stand for the arrival rates of new calls and handoffs, respectively,

and $p_o = \left[\sum_{n=0}^{c-1} \frac{1}{n!}\left(\frac{\lambda_c + \lambda_h}{u}\right)^n + \frac{1}{c!}\left(\frac{\lambda_c + \lambda_h}{u}\right)^c \left(\frac{cu}{c\mu - \lambda_h}\right)\right]^{-1}$.

Fig. 3-7 demonstrates the accuracy of FIFO and SSMC queuing methods by comparing the analytical and simulation results. In this figure, M/M/C(1) is the numerical results computed by $\lambda_c = \lambda/25$ and $\lambda_h = 13/96 * \lambda/25$ call arrival rates, and M/M/C(2) is computed by $\lambda_c = \lambda/25$ and $\lambda_h = 0$ call arrival rates. The result shows our simulations for FIFO and SSMC are really close to the numerical results computed by M/M/C(2).

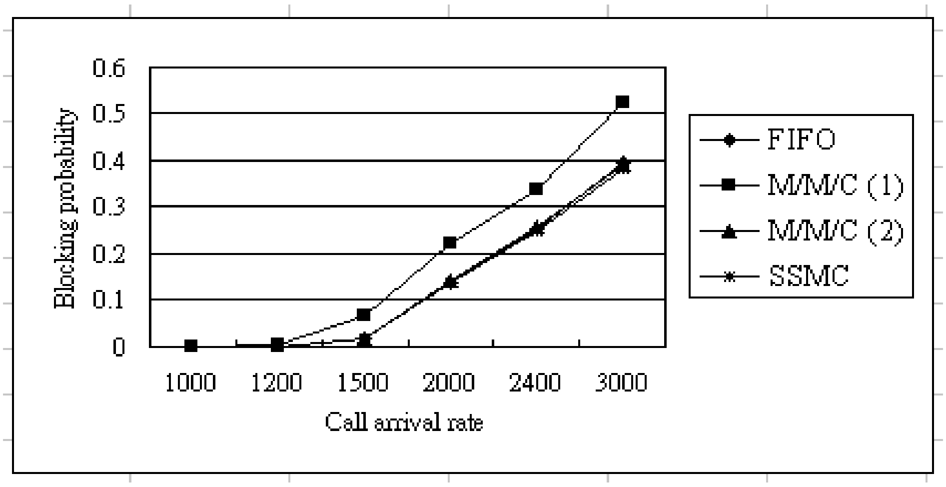

Figure 3-7. Call blocking probability of FIFO and SSMC vs. offered load (comparison between analysis and simulations)

3.4 Simulation Comparisons between SSMC, FIFO, and Non-Priority

In this section, we compare the performance of SSMC strategy with the other two schemes. The inputs to the simulator are a model of the wireless network and the characteristics of multimedia traffic in this network. The outputs of the simulator include the dropping probabilities for handoff requests and blocking probabilities for new connection requests.

There are three service classes in our simulation, called class 1, class 2 and class 3. The required bandwidth for each service classes is 64 Kbps which is suitable for data transmission [32], 64*2 Kbps and 64*4 Kbps respectively. The connection mean duration time for each class is 60 seconds, 60*5 seconds and 60*15 seconds respectively. The handoff priority for each class is 1, 4 and 8 respectively. And the channel capacity of a cell is 50*64 Kbps.

All the simulations are done with arrival rate ratio, 40:10:1, for service classes 1, 2 and 3. If the new call arrival rate is 51 calls per second, there will be 40 calls of class 1, 10 calls of class 2 and 1 call of class 3. All the results are average of 10 runs. Figures 3-8 and 3-9 show the handoff call dropping probability and new call blocking probability when the arrival rate is 510 calls per second.

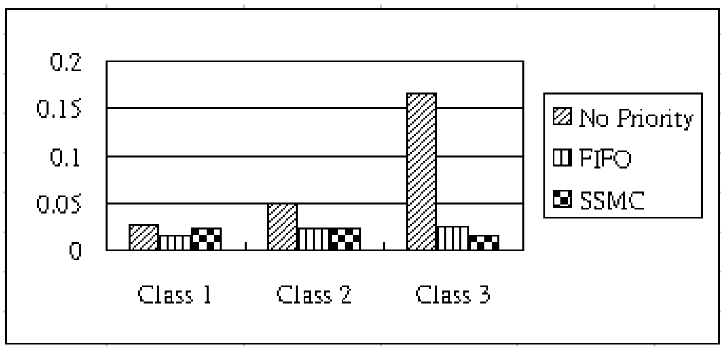

Figure 3-8. Handoff call dropping probability

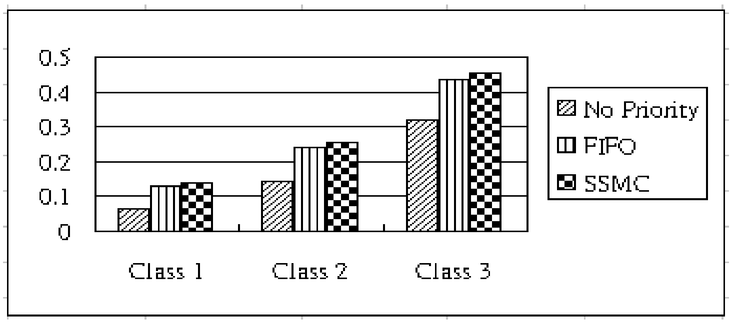

Figure 3-9. New call blocking probability

According to the above results, SSMC can reduce about 15% handoff call dropping probability than non-priority method for class 3. Also SSMC gives preferences to high priority traffic such that it has the lowest handoff dropping probability for class 3. Of course the price paid is the increase in new call blocking probability.

Figures 3-10, 3-11 and 3-12 show the handoff call dropping probability of three service classes under different offered load. It shows that the SSMC is effective in reducing class 3's handoff call dropping probability. The differences of handoff call dropping probability between FIFO and SSMC (FIFO minus SSMC) for class 1 and class 3 are presented in Fig. 3-13. When the new call arrival rate is higher than 510 calls/per second, the SSMC has outstanding efficiency in decreasing class 3's handoff dropping probability.

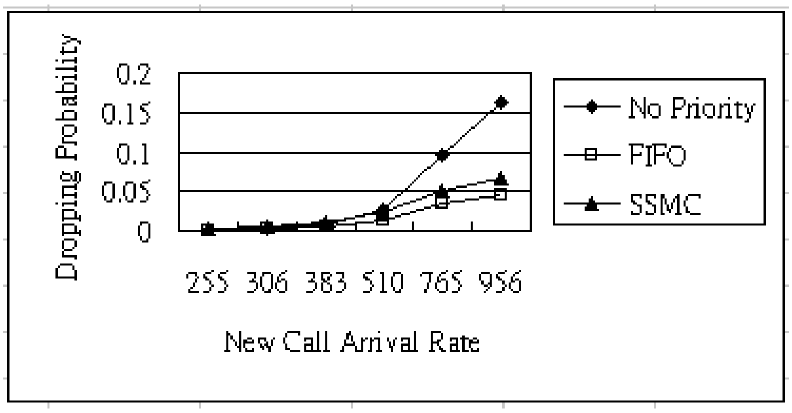

Figure 3-10. Handoff call dropping probability of class 1

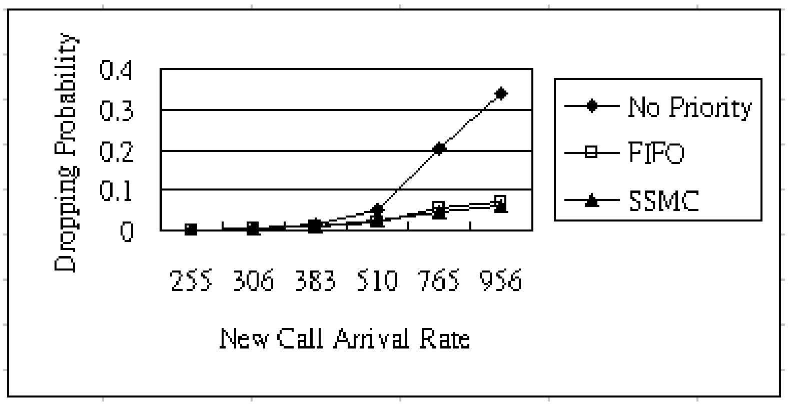

Figure 3-11. Handoff call dropping probability of class 2

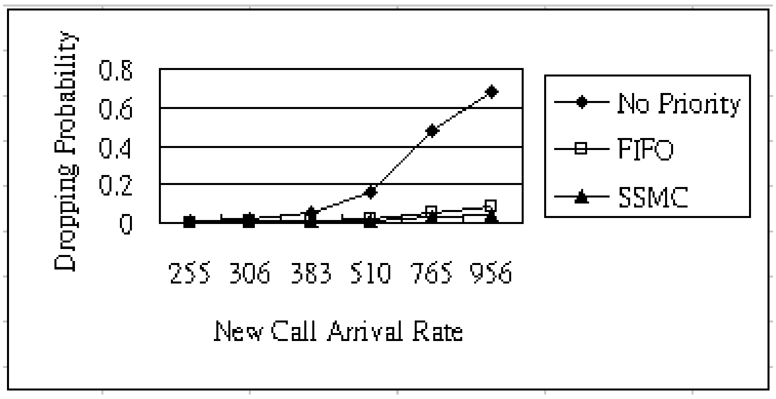

Figure 3-12. Handoff call dropping probability of class 3

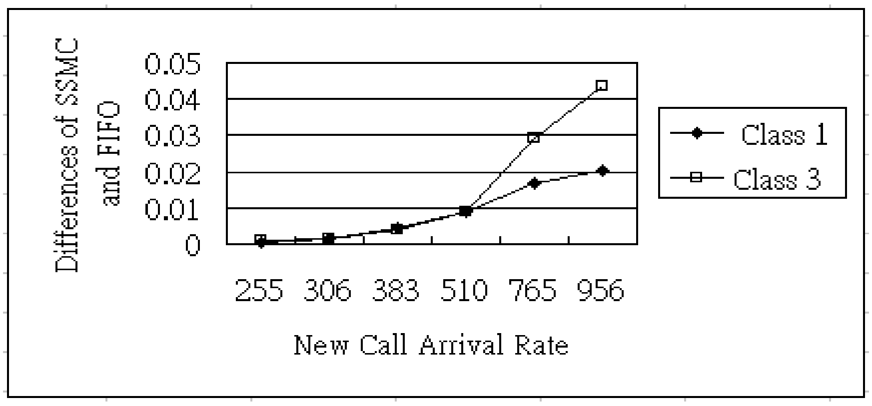

Figure 3-13. The differences of handoff call dropping probability (FIFO minus SSMC)

Figures 3-14, 3-15 and 3-16 represent the new call blocking probability under different offered load. According to these results the blocking probability of SSMC is almost the same as that of FIFO. Note that class 3's blocking probability is not affected as much as other classes.

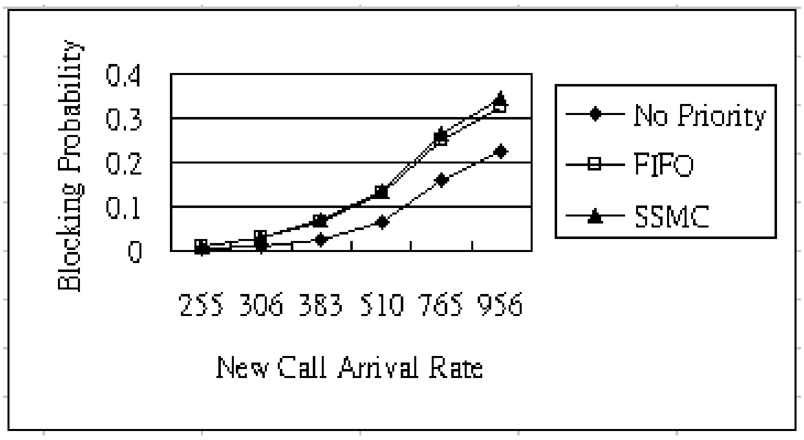

Figure 3-14. New call blocking probability of class 1

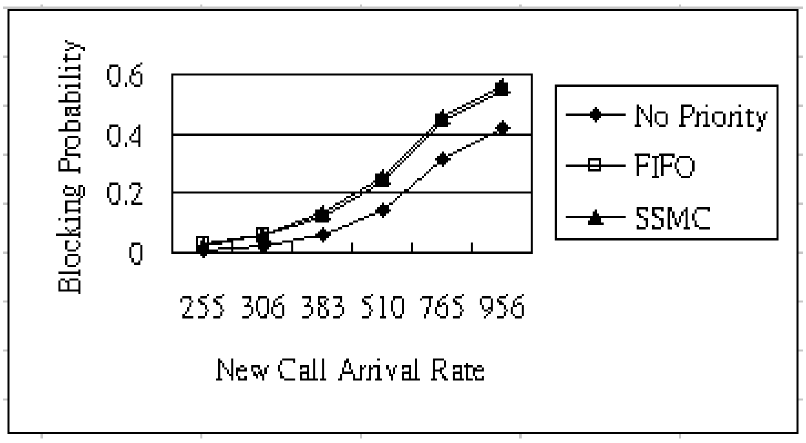

Figure 3-15. New call blocking probability of class 2

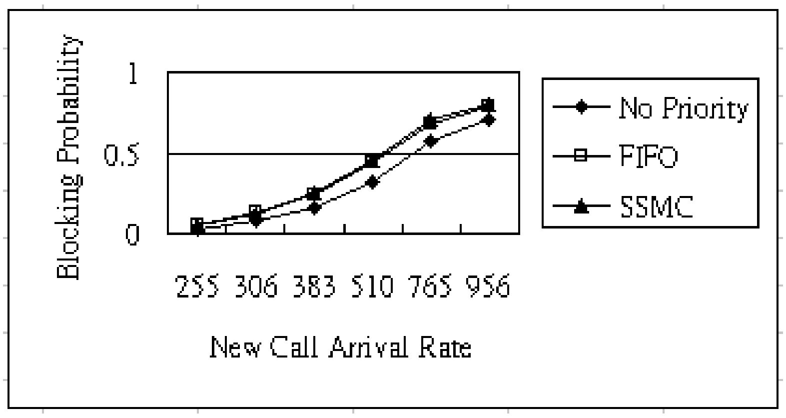

Figure 3-16. New call blocking probability of class 3

The average queuing times of FIFO and SSMC are shown in Fig. 3-17. The results show that SSMC has shorter handoff processing delay than FIFO.

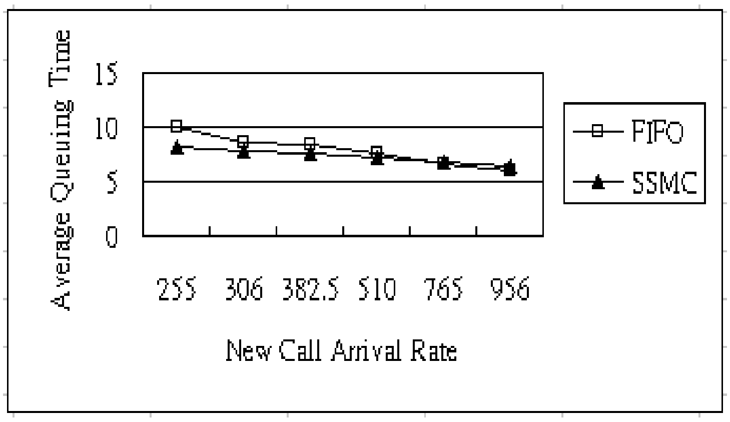

Figure 3-17. Average queuing time versus offered load

3.5 Remarks

In wireless networks, handoff between cells is unavoidable. The probability of forced termination is the most important quality of service (QoS) parameter, which requires special method to manage the handoff requests when the micro-cells are used. There are many methods proposed to handle this problem such as Call Admission Control Schemes, Guard Channel Schemes, Channel Reservation Schemes and Handoff Queuing Schemes. In this chapter, we propose a Handoff Queuing Scheme, which focuses on the problem of handoffs in a multimedia communication wireless networks.

A simple algorithm called SSMC is proposed, which can reduce the handoff call dropping probability in multimedia wireless networks. Different from the previous one cell simulation model, we propose a user mobility model and do a simulation in a 25-cells wireless network for three service classes. The simulation results show our method can reduce handoff call dropping probability for every service class. It is excellent in decreasing the dropping probability of high priority service class. There is a tradeoff between the handoff call dropping probability and new call blocking probability. SSMC increases about 15% in new call blocking probability, almost the same as the FIFO queuing method, to have a low handoff dropping probability. Therefore, it is an important research topic that to reduce the handoff dropping probability, we hope that the new call blocking probability is not unduly increased.

Chapter 4 QoS Routing With Received Signal Strength for Ad Hoc Networks

4.1 Introduction

Wireless networks provide mobile users with ubiquitous communication and information access capabilities regardless of their locations. There are currently two variations of mobile wireless networks. The first is known as infrastructured networks, i.e., networks with fixed and wired gateways [38]. The base station is the communication bridge between mobile users and wired networks. The mobile user communicates with the nearest base station within its communication radius. As the mobile user travels out of range of one base station and into the range of another, a "handoff" occurs from the old base station to the new. It allows the mobile user to communicate seamlessly. A typical network is the cellular personal wireless communication system.

The second type of mobile wireless network is the infrastructureless mobile networks, commonly known as a Mobile Ad Hoc NETwork, or MANET [39]. The infrastructureless network has no fixed routers. Each mobile node operates not only as a host but also as a router, which discovers and maintains routes to other nodes in this network. Example applications of ad hoc networks are emergency search-and-rescue operations and data acquisition operations in inhospitable terrains.

The mobility and limited computing capability of mobile hosts [39] make the design of routing protocols challenging. Many protocols have been proposed, such as DSDV [41], CGRS [42], WRP [43], AODV [44], DSR [45], TORA [46], ABR [47], SSR [48], ZRP [49], FSR [50], etc. Among them, AODV (Ad hoc On-demand Distance Vector) routing protocol is one of the most frequently mentioned. It chooses a minimal hop-count path to be the routing path. However, a path having the minimal hop-count does not always mean that it is the optimal routing path in various respects. Even the smallest hop-count path may be longer than other paths in real distance. It is highly probable that the spatial distance between intermediate nodes on the route is larger than another path. The actually longer distance between neighboring nodes may give rise to path maintenance cost, reduce the quality of service and suffer more path broken frequency. The whole system performance will be affected by the inferior routing path.

In this chapter we propose a quality of service supported routing scheme, called AODV-RSS (AODV with Received Signal Strength), for mobile ad hoc networks. Our method is based on the received signal strength, RSS, and the received signal strength changing rate, *ΔRSS*. We use these parameters to forecast the link available time (*LAT*) between two mobile nodes. In accordance with the connection's QoS request, such as minimum link available time, our method will find out a minimum hop count routing path that satisfies the QoS requirement. It is very useful in the multimedia communication environment where there are real time flows and best effort flows. A real time flow needs QoS guarantees, such as packet delay bound, delay jitter, bandwidth, etc. Using the value of *LAT*, our routing protocol will support a more stable routing path for the real time flows.

The remainder of this chapter is organized as follows. Section 4.2 describes our Quality of Service supported routing protocol. In Section 4.3, the simulation environment and the results of comparing the performance of our method and related work are presented. Section 4.4 concludes this chapter.

4.2 Received Signal Strength Routing Algorithm

Unlike the table-driven schemes, the on-demand schemes can save the node's resources usage in distributing the network control message distribution and routing table maintenance. Our proposed scheme is inspired by on-demand schemes. Based on the Ad hoc On-demand Distance Vector (AODV) routing protocol, our scheme will add the function of quality of service. This new routing protocol is called AODV-RSS (Ad hoc On-demand Distance Vector with Received Signal Strength) routing protocol.

The received signal strength will be larger when the distance of two mobile nodes is closer. Our QoS supported routing algorithm uses the Received Signal Strength, *RSS*, and Received Signal Strength changing rate, *ΔRSS* to predict the link available time between two mobile nodes. Let $RSS_{i,j}(t)$ denote the received signal strength seen by node *j* with respect to node *i* at time *t*. Assume a symmetric wireless network. Then $RSS_{i,j}(t)=RSS_{j,i}(t)$. Define $\Delta RSS_{i,j}(t_1,t_2)=\frac{RSS_{i,j}(t_2)-RSS_{i,j}(t_1)}{t_2-t_1}$, $t_2>t_1$. $\Delta RSS_{i,j}(t_1,t_2)$ can be seen as the *RSS* changing rate from time t_1 to t_2. Since $RSS_{i,j}(t)=RSS_{j,i}(t)$, we have $\Delta RSS_{i,j}(t_1,t_2)=\Delta RSS_{j,i}(t_1,t_2)$. If $\Delta RSS_{i,j}(t_1,t_2)>0$, it means that node *i* and node *j* are closer at time t_2 than they were at time t_1. On the contrary, a negative $\Delta RSS_{i,j}(t_1,t_2)$ means that node *i* and node *j* are leaving away from each other. Usually, $RSS_{i,j}(t)$ depends on the distance between node *i* and node *j* at time *t*. $\Delta RSS_{i,j}(t_1,t_2)$ depends on how fast node *i* and node *j* are moving relatively. Therefore, we can use $\left|\Delta RSS_{i,j}(t_1,t_2)\right|$ as an indication of the relative speed between node *i* and node *j*. A large $\left|\Delta RSS_{i,j}(t_1,t_2)\right|$ means node *i* and node *j* are moving toward or leaving away from each other quickly. Its concept is illustrated in Figure 4-1. In Figure 4-1, the distance between node *B* and node *C* is shorter than the distance between node *B* and node *A*. So the received signal at time t_1, $RSS_{B,C}(t_1)$, is greater than $RSS_{A,B}(t_1)$ at first. But node *B* is moving toward node *A*, and node C is leaving from node *B*. The received signal strength between *B* and *C* is decreasing over the time. On the contrary, the received signal strength between *A* and *B* is increasing. After a few minutes, the received signal strength of $RSS_{A,B}(t_2)$ is larger than $RSS_{B,C}(t_2)$. In our research, we are interest in using the received signal strength, *RSS*, and the received signal strength changing rate, ΔRSS, to calculate the *LAT* (Link Available Time) between two mobile nodes.

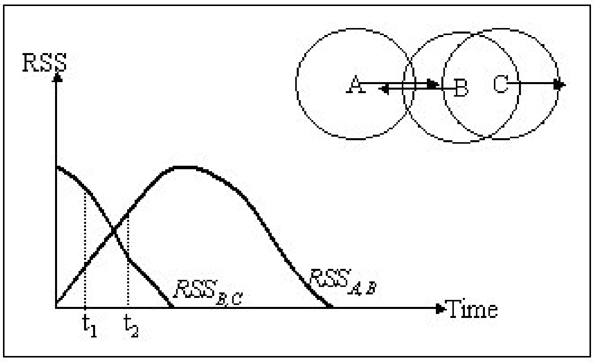

Figure 4-1. Moving direction and *RSS*

Let t_1 be the time when node *i* and node *j* first detect the presence of each other. With this understanding, we can simplify $\Delta RSS_{i,j}(t_1,t_2)$ to $\Delta RSS_{i,j}(t_2)$. Therefore, without loss of generality, we use $\Delta RSS_{i,j}(t)$ to denote the *RSS* changing rate between time *t* and the time node *i* and node *j* first met.

To calculate the link available time, let $D_{i,j}(t)$ denote the distance between node *i* and node *j* at time t and $S_{i,j}(t)$ denote the relative speed. Assume *TR* is the radio transmission range of a mobile node. If $\Delta RSS_{i,j}(t)$ is positive, it means node *i* and node *j* are moving toward each other, as shown in Figure 4-2(a). The time needed before they are at their closest point (Figure 4-2(b)) can be approximated by $\frac{D_{i,j}(t)}{|S_{i,j}(t)|}$. After the closest point, they are leaving away from each other. Their connection will be broken when each is at the border of the transmission range of the other (Figure 4-2(c)). The time for them to move from the position in Figure 4-5(b) to that of Figure 4-5(c) can be approximated by $\frac{TR}{|S_{i,j}(t)|}$. Therefore, the total time for the link between node *i* and node *j* to be effective is $\frac{D_{i,j}(t)+TR}{|S_{i,j}(t)|}$. Similarly, if $\Delta RSS_{i,j}(t)$ is negative, node *i* and node *j* are leaving away from each other. The time needed for them to lose contact can be approximated by $\frac{TR-D_{i,j}(t)}{|S_{i,j}(t)|}$. Combining the above results, the link available time between node *i* and node *j*, $LAT_{i,j}(t)$, can be defined as:

(1) if $\Delta RSS_{i,j}(t)$ is positive, $LAT_{i,j}(t)=\frac{D_{i,j}(t)+TR}{|S_{i,j}(t)|}$; else

(2) if $\Delta RSS_{i,j}(t)$ is negative, $LAT_{i,j}(t)=\frac{TR-D_{i,j}(t)}{|S_{i,j}(t)|}$.

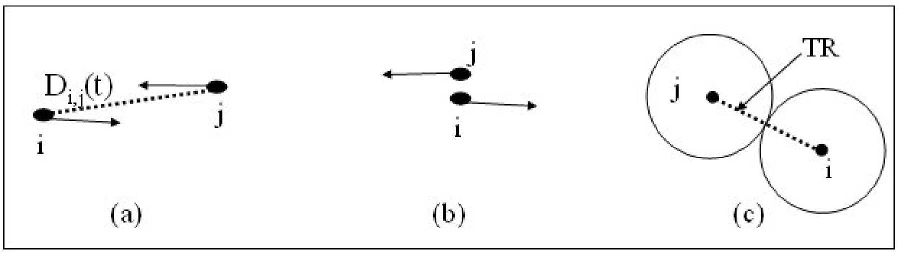

Figure 4-2. The relative position of two approaching nodes

However, from the point of view of node *j* (or node *i*), it does not have the values of $D_{i,j}(t)$ and $S_{i,j}(t)$. From [59], assume $RSS_{i,j}(t)$ is proportional to $\left[D_{i,j}(t)\right]^{-2}$. Then $\Delta RSS \approx \frac{dRSS}{dt} \propto \frac{dD^{-2}}{dt} = \frac{-2D^{-3}dD}{dt} = -2D^{-3}S(t)$. Therefore, $S(t) \propto D^{3}\Delta RSS \propto \left(\frac{1}{\sqrt{RSS}}\right)^{3}\Delta RSS$. Using the above reasoning, $LAT_{i,j}(t)$ can be redefined as:

(1) if $\Delta RSS_{i,j}(t)$ is positive, $LAT_{i,j}(t) = \frac{\left(RSS_{i,j}(t)\right)^{-\frac{1}{2}} + TR}{\left|\left(RSS_{i,j}(t)\right)^{-\frac{3}{2}} * \Delta RSS_{i,j}(t)\right|}$; else

(2) if $\Delta RSS_{i,j}(t)$ is negative, $LAT_{i,j}(t) = \frac{TR - \left(RSS_{i,j}(t)\right)^{-\frac{1}{2}}}{\left|\left(RSS_{i,j}(t)\right)^{-\frac{3}{2}} * \Delta RSS_{i,j}(t)\right|}$.

Since $RSS_{i,j}(t)$ and $\Delta RSS_{i,j}(t)$ can be estimated in each node without extra information, the link available time can thus be computed. The calculated *LAT* represents a measurement of how long two nodes can keep connected. Assume a connection wants to transmit *M* bytes and a link's data transmission rate is *N* bits per second, the whole data transmission time for this connection will be *M*8/N* seconds. The AODV-RSS algorithm can use the data transmission time in the routing path discovery procedure as an *LAT* constraint. Intuitively, if we set the *LAT* constraint to the data transmission time, it means that AODV-RSS algorithm can find out a routing path that can transmit the entire data without suffering any broken routing path. That is, a connection using data transmission time as the *LAT constraint* will have good QoS guarantee. However, a higher *LAT* constraint will suffer from a higher connection rejection probability. This is because a higher *LAT* constraint needs a longer link available time between any two mobiles. The number of available paths for a higher *LAT* constraint will be less than others. To decrease the route discovery time and increase the route discovery probability, less rigid *LAT* constraints may be used.

Using the above *LAT* constraint, our QoS supported routing algorithm will find out a minimum hop-count routing path that satisfies the *LAT* constraint. This concept is shown in Figure 4-3. In this figure, the number above the link between two mobiles means the link available time. Assume, the *LAT* constraint is 14. Because the *LAT* of *B-C* does not satisfy the *LAT* constraint, the RREQ packet will not be sent to node *C* in AODV-RSS algorithm. So, our algorithm will select *A-B-D-E-F* as the routing path. Every link of this routing path satisfies the *LAT* constraint. On the other hand, the routing path discovered by AODV will be *A-B-C-E-F* or *A-B-D-E-F*, depending on which path's RREQ packet is received first by *F*. If the routing path discovered by AODV is *A-B-C-E-F*, it will suffer from more frequent path broken than the path found by AODV-RSS.

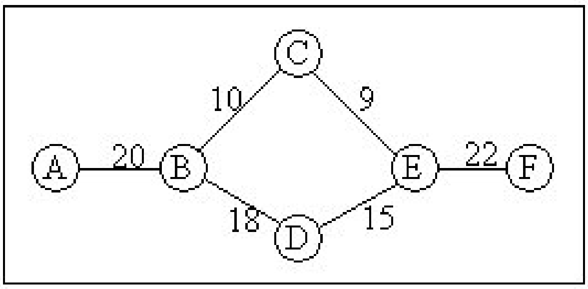

Figure 4-3. Routing path discovery example

The routing path establishment and maintenance procedure of AODV-RSS is similar to AODV. In the routing path discovery procedure, AODV-RSS algorithm just sends the RREQ packet to the nodes, whose link available time satisfies the *LAT* constraints. If a

RREP packet is not received within the path discovery time [58], this connection request is rejected. On the contrary, the packet is sent through the routing path until the path is broken due to the mobile node's moving. If the source node receives a RERR packet or an ACK message is not heard from the receiver within a time period, the source node will re-send the RREQ packet to re-establish a new path.

In order to assess the performance of our algorithm, simulations are done to compare with AODV algorithm. The simulation environments and results are described in the following section.

4.3 Performance Evaluations

4.3.1 Simulation Environments

We evaluate the proposed AODV-RSS protocol by simulation and compare the performance with that of AODV protocol. The link breakage is detected by the feedback of MAC layer in both protocols. No additional network layer mechanism is used. And the bandwidth of each mobile node is not taken into consideration to simplify the simulation model.

In our simulation model, we generate 100 mobile nodes in a 1000*1000 square meters area. The sides of the square are wrapped around, so the simulation area represents a sphere. The moving direction of each mobile node is a random variable of 0 degree to 360 degree. The mobile node's transmission range is 240 meters [55, 57]. The link capacity is 2 Mbps. The data transmission time of each connection is an exponential distribution with mean 50 seconds. Generally, an ad hoc network is not applicable to the extremely high-speed environment. We run our simulations with movement patterns generated by 12 different speeds (1, 2, 3, 4, ... 9, 10, 11, 12 meters per second) [55, 57]. The related simulation parameters are shown in Table 4-1. Net diameter is defined as the maximum possible number of hops between two nodes in the network [58]. So, net diameter of our simulation is 100. The node traversal time is a conservative estimate of the time spent on a node for packets and should include queuing delays, interrupt processing times and transfer times. The node traversal time is set to 40 milliseconds in our simulations [58].

Table 4-1. Simulation parameters

Parameters Name	***Value***
Net Diameter	*100*
Node Traversal Time	*40 milliseconds*
Net Traversal Time	*2*(Node Traversal Time)*(Net Diameter)*
Path Discovery Time	*2* (Net Traversal Time)*
Active Route Timeout	*2* (Net Traversal Time)*
Moving Speed	*1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11, 12 meters/sec*
Moving Direction	*Random number between 0 to 360 degree*
Transmission Range	*240 meters*
Data transmission Time	*Exponential with mean 50 seconds*
LAT Constraints	*0.2, 0.4 and 0.6 of data transmission time*

The following characteristics are taken into consideration to measure the AODV and AODV-RSS routing protocols' performance. These characteristics are depicted as follows.

- **Average Route Discovery Time**: The time period from RREQ packet is sent from source node to the RREP packet is received by source. In our simulations, if the source node does not receive the RREP packet within the path discovery time period, the connect request is given up.
- **Path Discovery Failure Probability**: The path discovery failure probability is the probability that the routing path for a new connection is not established in the first connection request.
- **Average Route Connected Time**: The time period from the source node sends a packet to the time the route is broken.
- **Average Route Discovery Frequency**: The moving of mobile node will lead to a broken routing path. So a routing path will be re-established. We consider the average of route discovery frequency for a each connection.
- **Connection Broken Probability**: When the routing path is broken, we find a new routing path for this connection. The connection broken probability is the probability where the new path is not established in the first re-routing discovery time.

Each simulation is executed for 60*60 seconds. In our simulation study, the simulation results are the average of 10 runs. The simulation results are shown in the following section.

4.3.2 Simulation Results

Figure 4-4 shows the average route discovery time per connection as a function of mobility (speed) for the proposed AODV-RSS, with 0.2, 0.4 and 0.6 *LAT* constraint, and AODV. The average route discovery time is about 0.36 second. In all values of the speed, the results show our protocol will not increase the route discovery time in three different *LAT* constraints. The average frequency of route discoveries for each connection is shown in Figure 4-5. According to simulation results, our AODV-RSS routing algorithm performs excellently in reducing the route re-discovery frequency. This is because AODV-RSS does not choose paths blindly. It selects paths that satisfy the *LAT* constraints.

Due to the mobility, an established routing path will be broken. Figure 4-6 shows the probability that a new routing path will not be re-established by the new RREQ sent after path broken. In the 6 meters per second mobility mode, the AODV-RSS with 0.6 *LAT* constraint can reduce the broken probability about 15% than the AODV routing protocol. Figure 4-7 shows the average route continuing connected time for AODV-RSS with different *LAT* constraint versus AODV routing protocol. Based on the simulation results, we can see the routing connected time is shorter when the moving speed is higher. The AODV-RSS algorithm is doing a good job in keeping the continuous connection. In the moving speed of 6 meter per second, AODV-RSS with 0.6 *LAT* constraint will increase route connected time about 13 seconds. Figure 4-8 shows the route path discovery failure probability for AODV and AODV-RSS. The results show AODV-RSS will increase about %3, 6% and 9% path discovery failure probability than AODV in different mobility patterns for 0.2, 0.4 and 0.6 LAT constraints, respectively. It is easy to understand that AODV-RSS needs to choose a more stable routing path, so the path discovery failure probability will increase.

The AODV-RSS algorithm is good in improving the routing path's duration and reducing the route broken probability. So we are interested in the *LAT* constraints study. The following paragraph will show the AODV-RSS in the performance of broken probability, the frequency of route path discovery, and route continuous connection time with 0.1, 0.2, 0.3, 0.4, 0.5, 0.6, 0.7, and 0.8 *LAT* constraints in 5 meters per second moving speed.

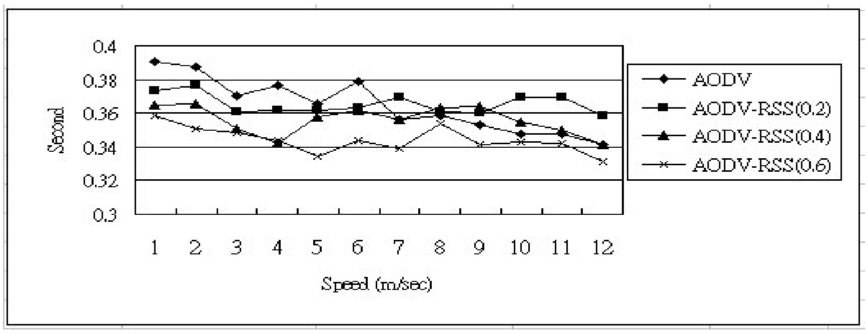

Figure 4-4. Average route discovery time vs. mobility

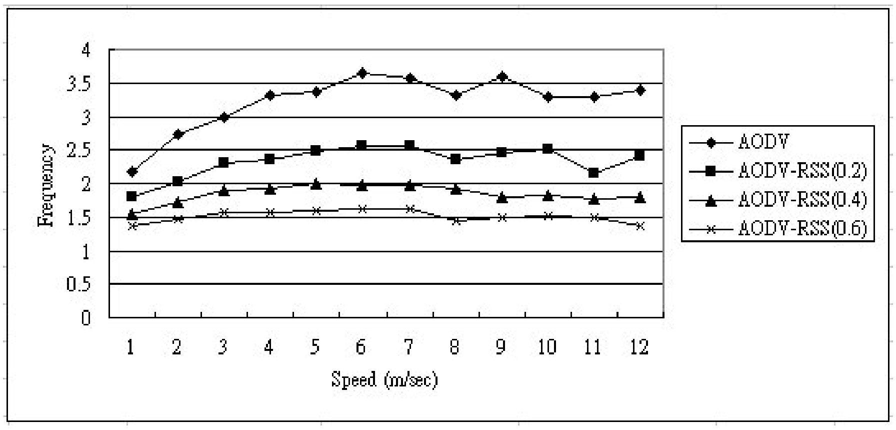

Figure 4-5. Average route discovery frequency vs. mobility

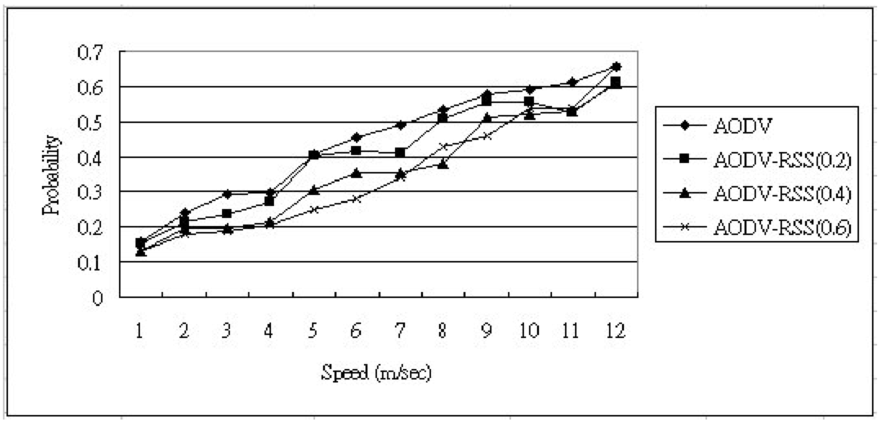

Figure 4-6. Connection broken probability vs. mobility

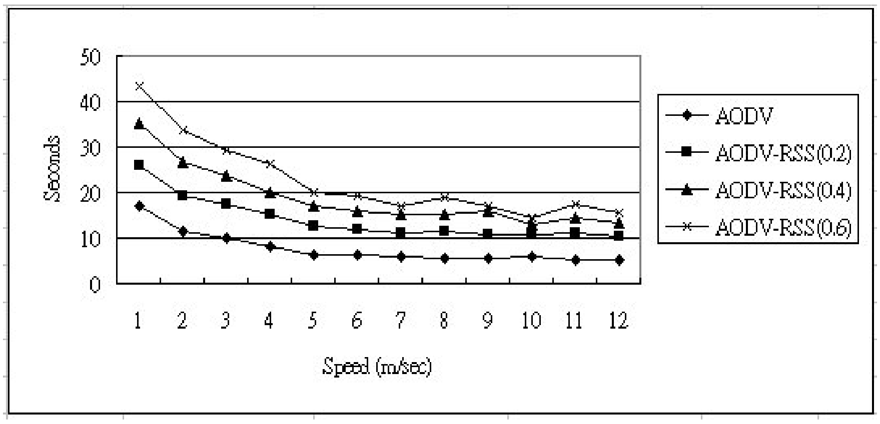

Figure 4-7. Average route connected time vs. mobility

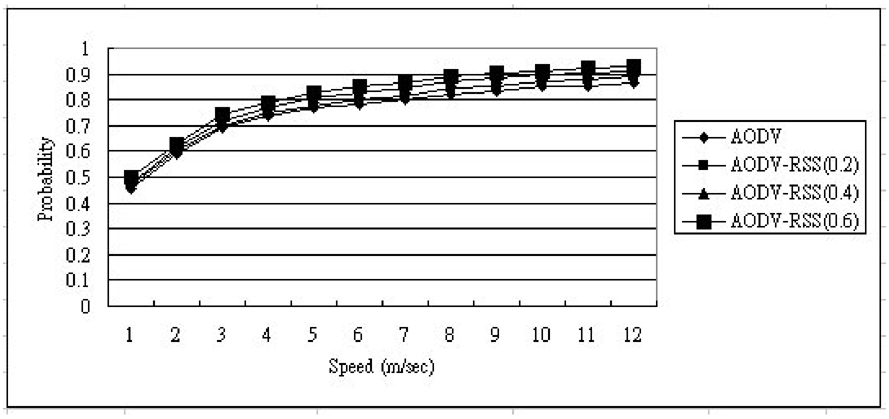

Figure 4-8. Path discovery failure probability vs. mobility

Figures 4-9, 4-10 and 4-11 show the route broken probability, route discovery frequency and route continuous connected time of the proposed AODV-RSS routing algorithm with different *LAT* constraints, respectively. All the results are done in 5 meters per second moving speed. In order to compare with AODV, the AODV simulation results with 5 meters per second are shown in these figures too. According to these results, we know the higher *LAT* constraint is the better QoS guarantee will be in low route broken probability, small route discovery frequency, and long continuous route connection time. These results are suitable for the characteristics of multimedia communications.

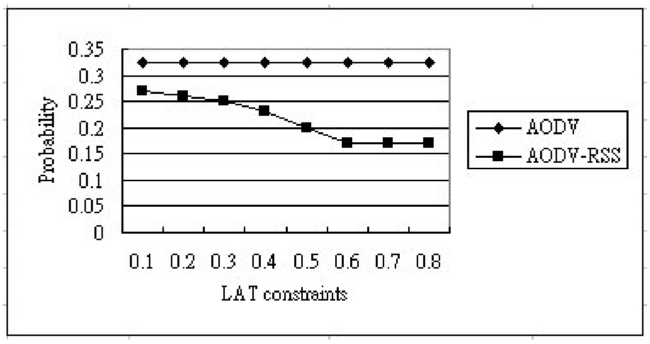

Figure 4-9. AODV-RSS route broken probability vs. *LAT* constraints

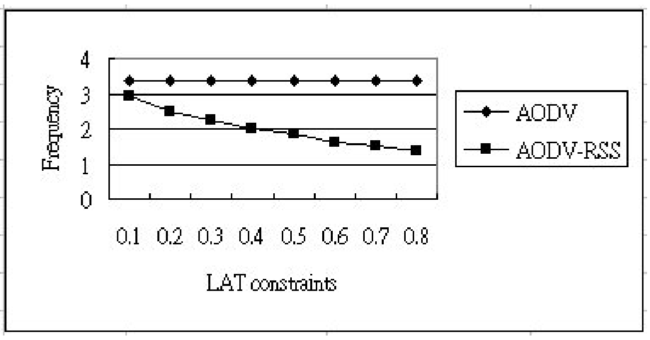

Figure 4-10. AODV-RSS route discovery frequency vs. *LAT* constraints

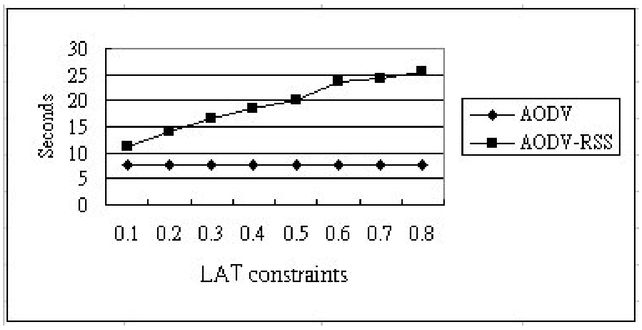

Figure 4-11. AODV-RSS route continues connected time vs. *LAT* constraints

4.4 Remarks

The ad hoc network's topology changes drastically and unpredictably due to the mobile node's mobility. There are many routing protocols proposed. In this chapter we review these routing protocols briefly firstly. Then, we propose an AODV based routing algorithm. In order to improve the QoS of routing path, we use the received signal strength to predict the link available time, *LAT*. This algorithm is called AODV-RSS.

The simulation results show that AODV-RSS can excellently improve the route QoS guarantee in route broken probability, route continuous connection time, and route re-establishment frequency. In 0.6 LAT constraint and 6 meters per seconds mobility pattern, our method will reduce half the route re-discovery probability, and increase 14 seconds for the route continuingly connected time, and reduce 18% for connection broken probability, compared with the AODV protocol. The higher *LAT* constraint is, the better QoS guarantee is. These characteristics are suitable for the multimedia communications. In multimedia communications environment, we can set different *LAT* constraints for different QoS requirement.

In our AODV-RSS algorithm, the *LAT* constraint is simply defined by the percentage of data transmission time. In the future research, other parameters can be added into the *LAT* constraint calculation to make it more precise. On the other hand, we ignore the effect of link bandwidth. However, bandwidth is an important and limited resource in wireless networks. How to ensure a connection request has enough bandwidth to transmit data is an important research in wireless networks. In the future research, we can add the bandwidth usage mechanism in our routing protocol.

Chapter 5 Conclusions and Future Work

Over the last two decades, development of wireless communication services was very phenomenal. The wireless communication systems provide the ubiquitous access to information, anywhere, anyplace, and anytime. According to the infrastructure of the wireless communication system, the wireless communication networks can be divided into two types: personal communication systems and ad hoc networks. The topics of this thesis are proposing a QoS supported handoff procedure for the multimedia wireless communication systems, and presenting a QoS supported routing algorithm for the ad hoc networks.

In wireless cellular communication systems, handoff between cells is unavoidable. In order to guarantee the base station has enough resources (usually bandwidth) to handle the handoff call request, many methods have been proposed. We briefly divided those methods into four types. Those are: Call Admission Control Schemes, Guard Channel Schemes, Channel Reservation Schemes, and Handoff Queuing Schemes. We proposed a Handoff Queuing Scheme, which focuses on the problem of handoffs in a multimedia communication wireless networks. Using the received signal strength, the received signal degradation rate and the service priority, our method can reduce the handoff call dropping probability in multimedia wireless networks.

Nowadays, there are various services by the wireless network, such as GPRS, SMS, MMS, WAP, etc. Multimedia messages transmitted by the mobile hosts are common. The real time and non-real time traffic must be dealt with differently manners to fit their different transmission requirements. Besides the handoff request queuing method, the other QoS supporting mechanisms, such as bandwidth reservation, call admission control, etc., may be taken into consideration to improve the QoS guarantee in the future research.

The mobility and infrastructurelessness increase the complexity of routing algorithm in the ad hoc networks. The routing protocols for ad hoc networks have been widely discussed in the recent researches. In order to support the multimedia communications, the QoS assisting routing algorithm is an important study in the ad hoc networks. In Chapter Four, we propose a QoS supported routing algorithm. Based on the received signal strength and the received signal strength changing rate, our routing algorithm can improve the AODV routing algorithm's performance in route broken probability, route continuous connection time, and reducing the route re-establishment frequency. The simulation results show that our QoS supported routing algorithm is suitable for the multimedia communications by setting different QoS constraints for different QoS requirements. Besides considering the received signal strength, other parameters can be taken into consideration to improve our method in the future research, such as the mobility pattern, bandwidth requirement, message priority, etc.

In conclusion, the development of the wireless communication is the important milestone in the human history. In the near future, they will also play a major role for human communications. How to implement a QoS guaranteed communication environment over the wireless networks is a fundamental issue for researches.

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作 者 簡 介

姓　　名：呂幸娟 (Shing-Jiaun Leu)

籍　　貫：臺灣省桃園縣

出生日期：民國 59 年 10 月 11 日

學　　歷：

1997.9 ~ 迄今　　國立台灣科技大學資訊管理系博士班研究生

1993.9 ~ 1995.6　　國立台灣科技大學資訊管理碩士

(原名台灣工業技術學院)

1991.9 ~ 1993.6　　國立台灣科技大學資訊管理系

經　　歷：

1995.8 ~ 迄今　私立東南技術學院工管系講師　(原名東南工專)

研究領域：資源預留、轉手、繞徑、通訊服務品質保證、無線網路、無線網際網路

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