THE UNIVERSITY OF CALGARY

Implementation of a Peer-to-Peer Wireless System

by

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ABSTRACT

For most telephone companies, installing and maintaining switching and transmission facilities as well as supporting infrastructure is a big burden, especially in the areas with small population density and minimal subscribers growth. New mobile satellite systems offer a potential alternative of serving rural areas, but the local calling within a community will require two satellite channels for communication link. A simple peer-to-peer wireless system can be used to provide local calling within a small community. This reduces the use of the mobile satellite channels for local calls and, at the same time, telephone companies can reduce their cost for installing and maintaining local switching and transmission facilities. In this thesis, a single hop peer-to-peer wireless system is implemented with spread spectrum technology, and its performance is examined to ensure all the requirements are met.

ACKNOWLEDGMENTS

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Chapter 1

INTRODUCTION

1.1 Overview

Following the pioneering work of Hertz, the experiments of Marconi at the end of the 19th century demonstrated the feasibility that radio communications could take place between transceivers that were mobile and far apart. Henceforth telegraphic and voice communications were not inherently limited to the user's equipment tethered by wires [1]. As a result, North American telephony did not have to be defined by the physical challenge of wiring the continent any more. In fact, the installation and maintenance of copper wires to provide telephone service represents a major investment of telephone companies, and the cost of this investment is higher with lower population density [2]; it is expensive to provide telephone service to rural areas.

A common method to provide telephone service in the rural areas of northern Canada is to install a local fixed satellite earth station with a telephone switch. This centralized configuration is required for common access to the telephone network and ease of maintenance for the telephone company. In the 1970's telephone companies began to serve fixed-location wireless access systems such as subscriber radios to users in Canada [3]. Such a system is composed of a central station and multiple subscriber stations. With the addition of a wireless network, the telephone service provider can increase profits over the exclusive use of wired network. Both of these wired and wireless networks require centralized configuration with a local switch or central station.

The emergence of the new mobile satellite system, either geostationary (GEO) or low earth orbit (LEO), provides a wireless access alternative to the telephone network. Each subscriber only needs a terminal with an antenna pointing to the satellite to have telephone service. These systems can be used to provide telephone service in the rural areas. But using this kind of systems is still not very satisfactory, because all calls have to be routed through the mobile satellite system, whether they are local or long distance. Picture this. Two neighbors live in the same rural area, their houses only few hundred meters apart. When one of them calls the other, the call first goes to the satellite, and then comes down to the called person, and the air-time charges for making such a local call has to be paid. To alleviate this problem, a peer-to-peer wireless system consisting of a group of identical terminals can be used in the rural areas for local calling. This system with a mobile satellite system provides you a much better picture of providing telephone services in rural areas.

Current peer-to-peer wireless systems are designed for military use and for mobile users. The jamming threat concern and the terminal mobility requirement increase the complexity of the system. They are not applicable for the rural area local communications [6].

1.2 Unlicensed ISM bands

In 1985 the United States Federal Communications Commission (FCC) released its Part 15 rules which allowed unlicensed use of spread-spectrum radios in the industrial scientific and medical (ISM) bands (902 - 928 Mhz ; 2.400 - 2.4835 GHz; 5.725 - 5.850 GHz). The maximum allowed transmitted power is 1 Watt in these bands when using Spread Spectrum.

1.3 Spread Spectrum and TDMA

Spread spectrum technologies consist of direct sequence (DS) spread spectrum and frequency hopping (FH) spread spectrum. They are explained further in Section 3.2. TDMA is the acronym for time division multiple access.

1.4 Scope of Thesis

The peer-to-peer wireless system implemented in this thesis is a single hop system, which consists of a signaling channel and a message channel. This peer-to-peer wireless system is in the 902 - 928 Mhz band. TDMA and frequency hopping spread spectrum are employed in this peer-to-peer wireless system. In this system, the code for handling the signaling channel protocol is implemented from scratch, and code for handling the message channel is partially modified from the original FH code in the wireless modem -Hopper, and partially developed from scratch.

An overview of the existing configurations to provide telephone services in the rural areas of northern Canada, and the problem of local calling within a community using mobile satellites, as well as the motivation for implementing this peer-to-peer wireless system will be discussed in chapter 2. The operating requirements for this peer-to-peer wireless system and accompanying technologies will be introduced in chapter 3. A detailed implementation of this peer-to-peer wireless system based on the wireless modem Hopper^{TM}, made by Wi-LAN inc., is covered in chapter 4. The system performance is verified in chapter 5 and in chapter 6 the conclusions are presented.

Chapter 2

TELEPHONE SERVICE IN RURAL AREAS OF NORTHERN CANADA

2.1 Present Provided Telephone Service Configuration

A typical centralized architecture is used for providing telephone service in both urban and rural areas. In urban areas it is often combined with other transmission facilities based on optical fiber or digital radios. Here we especially concentrate on the configuration of providing telephone service in the Yukon Territory and the Northwest Territories of Canada. A simplified telephone network configuration in the rural areas of Northern Canada is shown in Figure 2.1. With this telephone network configuration, the telephone company has to provide local switching and satellite transmission facilities, as well as supporting infrastructure such as building and power. Such an architecture is expensive to install and maintain, especially in areas with low population density and minimal subscriber growth areas. The cost per line for this type of network increases dramatically when the population density of served area decreases. In the rural areas of northern Canada, usually there are less than 200 subscriber lines in each small community. As a result, serving the rural areas may cost five to ten times more per line as compared to urban areas [4].

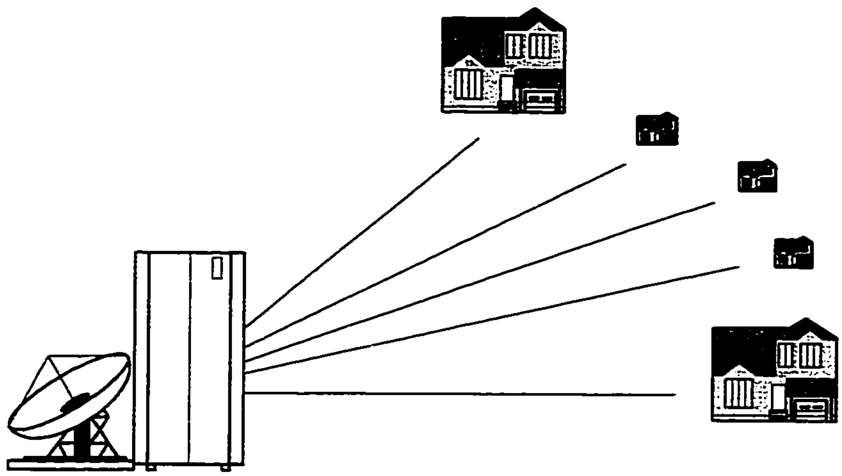


FIGURE 2.1 A TYPICAL (SIMPLIFIED) TELEPHONE NETWORK CONFIGURATION IN THE RURAL AREAS OF NORTHERN CANADA

2.2 Mobile Satellite Systems

Since the first communication satellite launch in 1962, satellite communication systems have been employed as an integral part of most major wide-area telecommunication networks throughout the world. In 1992, the World Administrative Radio Conference allocated new frequencies for mobile satellite communication in the 2 GHz band [5]. Several proposed nongeostationary mobile satellite systems will provide voice grade telephone service including MSAT from Telesat Mobile Inc., Iridium from Motorola, and Globalstar from Qualcomm. By using these mobile satellite systems, a user only needs a terminal to access the public telephone network through a satellite gateway station as shown in Figure 2.2. Mobile satellite systems provide an alternative for rural area subscribers to access the public telephone network. The local switching and transmission facilities are no longer necessary with the application of mobile satellite systems. The only equipment needed here is the mobile satellite terminal. This distributed architecture frees the telephone company from the maintenance of their own facilities.

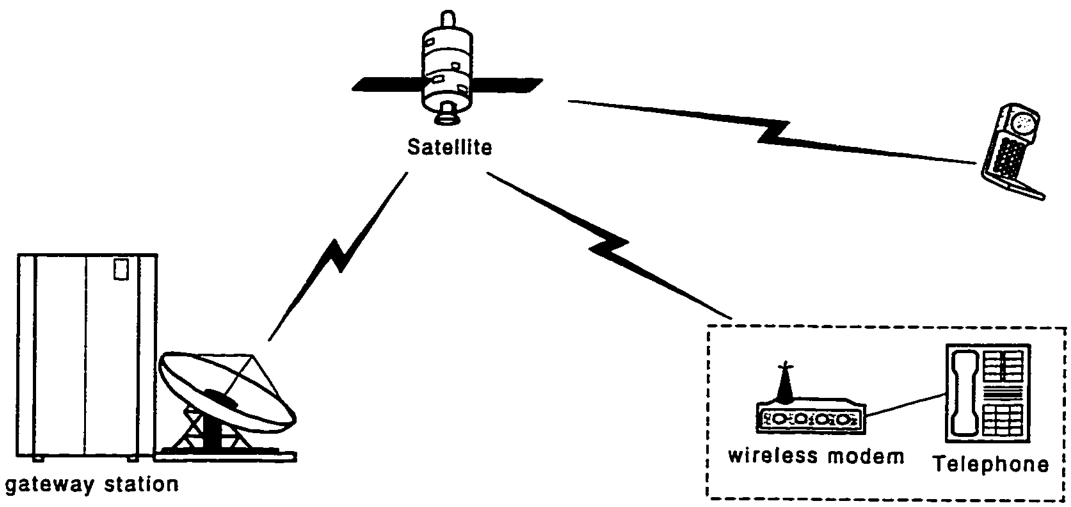


FIGURE 2.2 USING THE MOBILE SATELLITE SYSTEM TO ACCESS THE TELEPHONE NETWORK

2.3 Using Mobile Satellite System in rural areas

Although a mobile satellite system can provide telephone service in the rural areas, all calls, whether local or long distance, still have to be routed to the satellite and sometimes to the gateway station. This is not very satisfactory, especially for the local calls. As indicated in Figure 2.3, a local call within a community needs two satellite channels, which is definitely not economical for the telephone service providers due to the expensive air time fee.

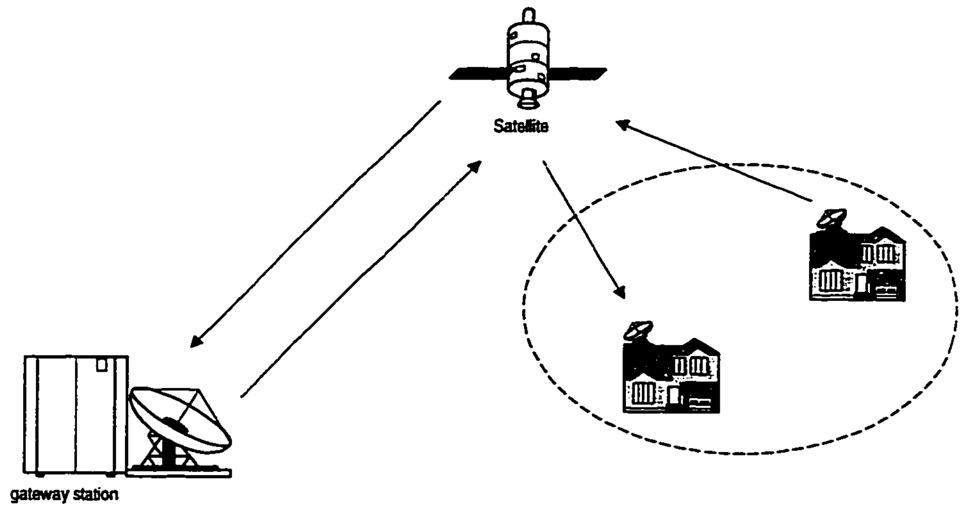


FIGURE 2.3 LOCAL CALLING WITHIN A COMMUNITY USING THE MOBILE SATELLITE SYSTEM

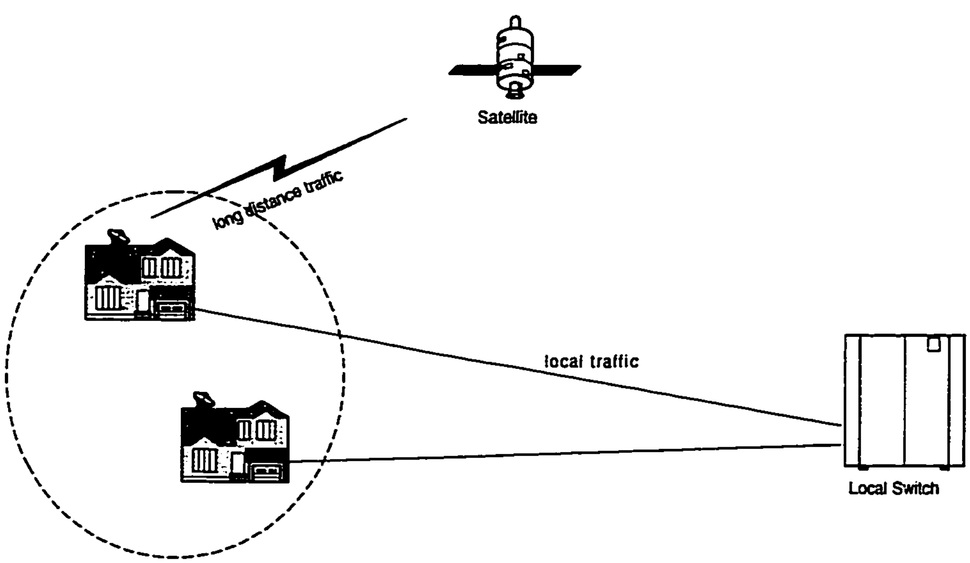


FIGURE 2.4 USING A PRIVATE BRANCH EXCHANGE ON SITE FOR THE LOCAL CALLS

One solution for this problem is to install a private branch exchange on site to route the local calls as shown on Figure 2.4. But this solution leads back to the former problem, the maintenance of the installed local facilities. We need a more satisfactory approach.

2.4 A better solution - peer-to-peer wireless system

The use of a peer-to-peer wireless system for handling local calls and routing only the long-distance calls to the satellite can solve the problem mentioned above. As shown in Figure 2.5.

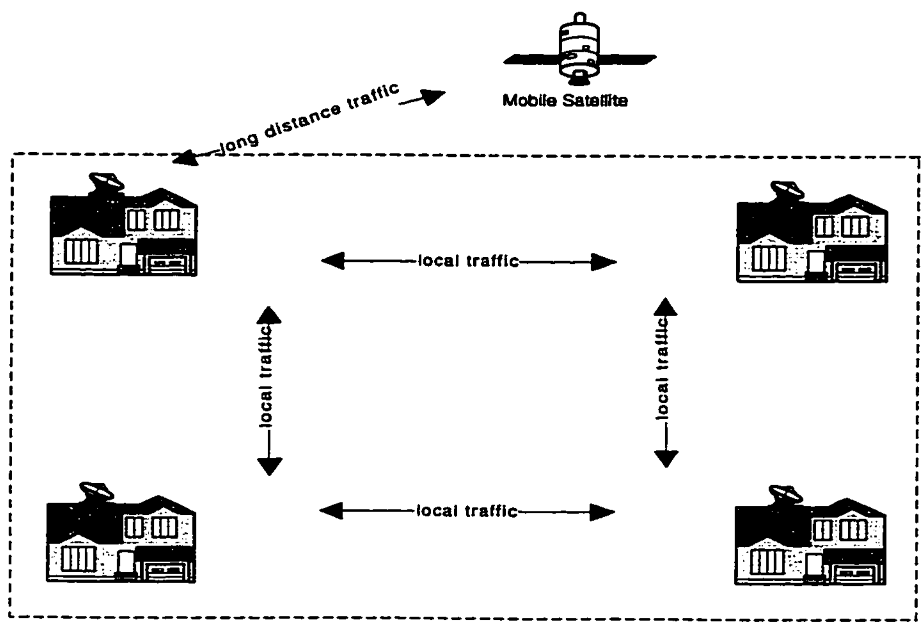


FIGURE 2.5 USING A PEER-TO-PEER WIRELESS SYSTEM TO HANDLE LOCAL CALLS IN THE RURAL AREAS OF NORTHERN CANADA

This architecture offers several advantages:

1) The use of a peer-to-peer wireless system eliminates the use of copper wire in rural areas, and all switching hardware related infrastructure, such as buildings and power. This system can be set up for local communication in a short time. This is very attractive in areas where no telecommunication infrastructure exists, but is needed temporarily, such as mining and exploration camps, or in search and rescue operations.

2) No installation infers no maintenance. The maintenance of the local switching and transmission facilities is a big burden to every telephone company. This burden can be simply eliminated by using a peer-to-peer wireless system. As a result, the cost of service in rural areas can be dramatically decreased.

3) Using a peer-to-peer wireless system, no local calls need to be routed to the satellite. This reduces air-time usage which results in savings for the telephone service providers, and free satellite channels for other uses.

Therefore, the use of a peer-to-peer wireless system in conjunction with a mobile satellite system is a very good option for serving rural areas.

Besides providing the local communications in the rural areas, a peer-to-peer wireless system can also have other applications; for instance, wireless local loop and cordless extension of cellular systems [6].

In the next chapter, the operating requirements of this peer-to-peer wireless system as well as some background knowledge are introduced.

Chapter 3

OPERATING REQUIREMENTS OF THE PEER TO PEER WIRELESS SYSTEM

3.1 Operating Frequency Band

Since the system is designed to serve as local communication in the rural areas of northern Canada, the radio standards set by the Industry and Science Canada (ISC) (or equivalently the Federal Communications Commission (FCC) in the U.S.) must be met. Also, the system should be simple and cost effective.

According to the above, the unlicensed or ISM (industrial, scientific, and medical) bands under the Industry and Science Canada RSS-210 [7] or FCC Part 15 in the United States are suitable operating frequency bands for this peer-to-peer wireless system. This is because the use of such bands does not require any licenses. The unlicensed frequency bands include the 902 - 928 Mhz, the 2400 - 2483.5 Mhz, and the 5725 - 5850 Mhz bands. All these unlicensed bands require the use of spread spectrum. Among these three bands, the 902 - 928 Mhz band is chosen for the following reasons:

1) 902 - 928 Mhz band is the lowest frequency band among these three frequency bands, so under the same transmission power, this band allows the maximum transmission range between terminals. The relation between free-space loss on a line-of-sight path and the frequency is given by equation 3-1.

$$L = (4\pi l / \lambda)^2 = (4\pi f l / c)^2 \qquad (3\text{-}1)$$

where L is the free space loss on a line-of-sight path;

f is the signal frequency;

λ is the signal wavelength;

l is the distance between terminals; and

c is the speed of light.

From equation 3-1-1, we can see that for a certain amount of path loss, the lower the signal frequency the longer the transmission distance.

2) The existence of a working wireless modem on the 902 - 928 Mhz band: Hopper, a wireless substitution for the RS-232 interface made by Wi-LAN Inc., is in this band, and is designed for the use of spread spectrum, which made the Hopper an ideal terminal for this peer-to-peer wireless system.

3.2 Spread Spectrum Techniques

As stated before, in the 902 - 928 Mhz unlicensed band only the application of spread spectrum techniques is allowed. Spread spectrum techniques can be grouped as direct sequence (DS) and frequency hopping (FH).

3.2.1 Direct Sequence Spread Spectrum

In direct sequence spread spectrum, a pseudo-random binary sequence is used to modulate the transmitted signal, and the relative rate between the pseudo-random sequence and the user's data is called the processing gain. Typically, the processing gain is between 10 and 100 for commercial systems [8]. After demodulation of the received signal, the baseband signal is correlated with the same pseudo-random sequence to recover the end user data. An example of direct sequence spread spectrum techniques with processing gain 12 is illustrated in Figure 3.1.

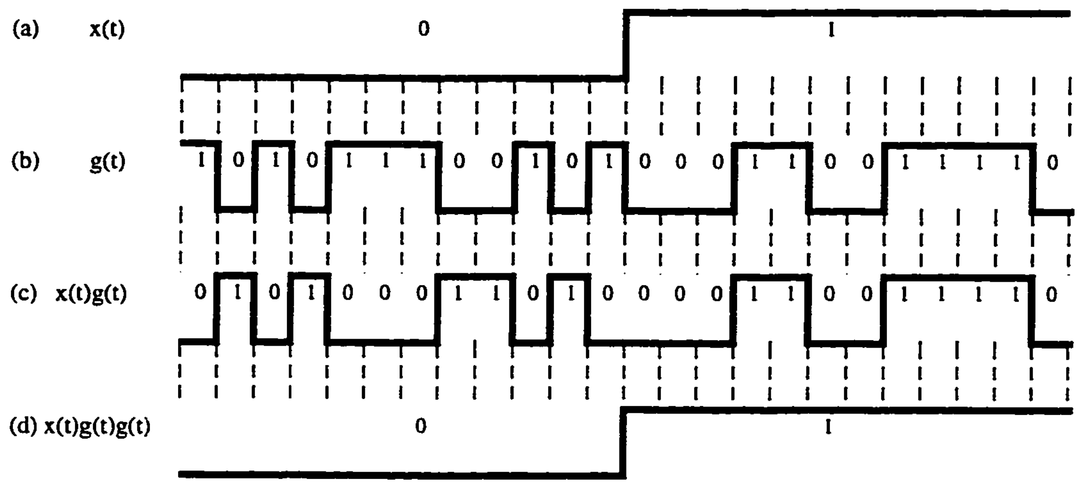


FIGURE 3.1 SPREAD-SPECTRUM EXAMPLE USING DIRECT SEQUENCE

(a) Binary data wave-form to be transmitted
(b) Code sequence (c) Transmitted sequence
(d) Demodulated data wave-form

3.2.2 Frequency Hopping Spread Spectrum

Unlike direct sequence, with frequency hopping technology the frequency of the carrier in both the transmitter and the receiver hop from one frequency to the next synchronously in a pre-determined hopping pattern. Frequency hopping system can be further classified either as slow frequency hopping which means that there are several modulation symbols per hop, or as fast frequency hopping which means that there are several frequency hops per modulation symbol [9]. A system with fast hopping technology is usually more costly to build, so slow hopping is more often implemented. An example of frequency hopping spread spectrum is shown in Figure 3.2.

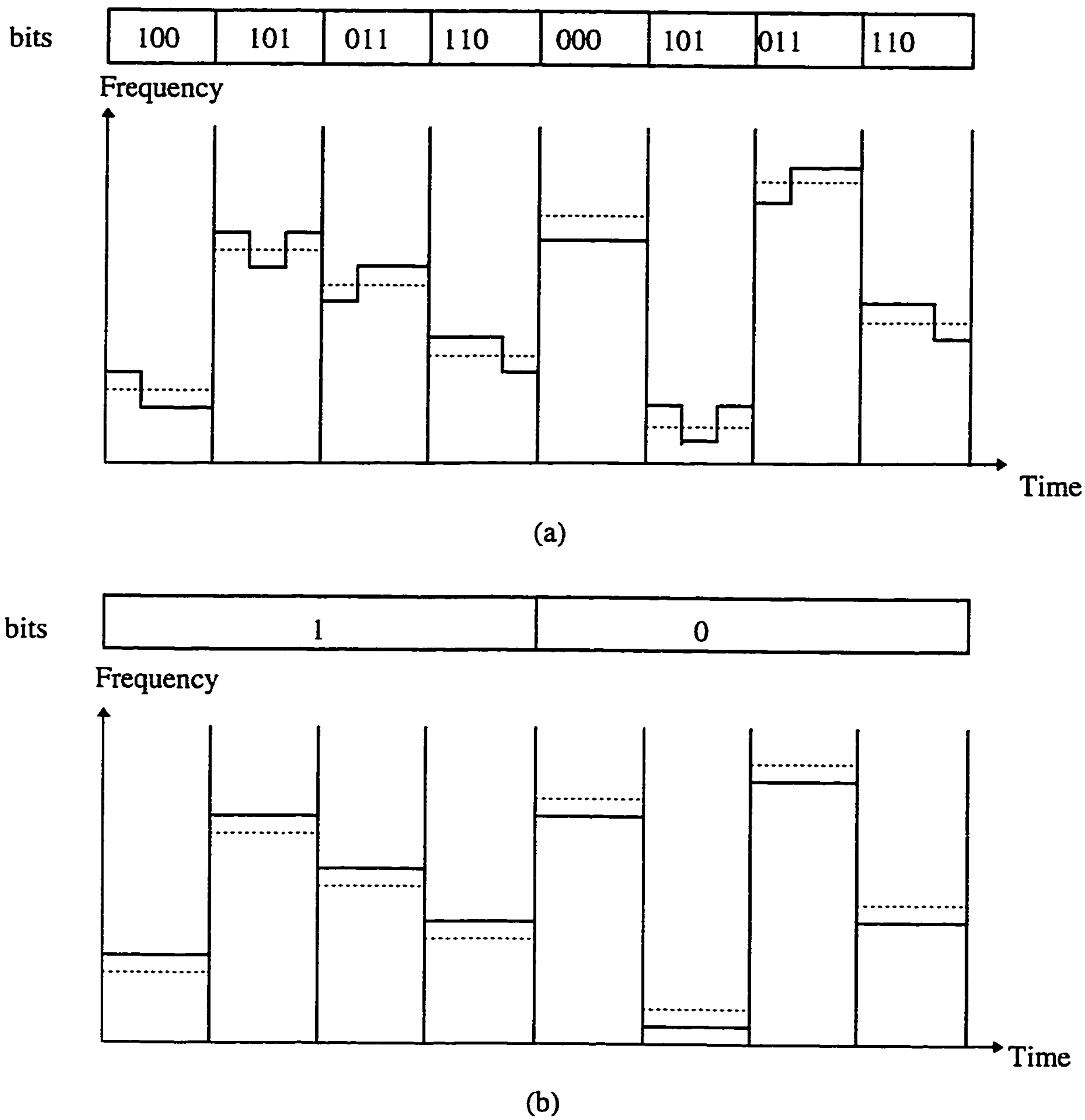


FIGURE 3.2 EXAMPLE OF SLOW HOPPING AND FAST HOPPING IN A BINARY SYSTEM

(a) Slow hopping (b) Fast hopping

Both frequency hopping spread spectrum and direct sequence spread spectrum systems have anti-jamming immunity to narrow-band interference. More specifically, direct sequence spread spectrum is best used in a multi-path environment, such as in an urban environment, while frequency hopping spread spectrum is best used to overcome narrow band jammers [10]. In addition to the advantages mentioned above, spread spectrum systems also have other features, such as:

1) Spread spectrum signals can be overlaid onto bands where other systems are already operating, with minimal performance impact to or from other systems.

2) The anti-interference characteristics of spread spectrum are important in some applications, such as networks operating on manufacturing floors, where signal interference environment can be harsh [10].

3) Cellular systems designed with code division multiple access (CDMA) spread spectrum technology can offer greater operational flexibility and possibly a greater overall system capacity than do systems built on frequency division multiple access (FDMA) and TDMA access methods [10].

The peer-to-peer wireless system implemented in this thesis employs frequency hopping technology. The reasons for selecting frequency hopping over direct sequence are as follows:

1) Direct sequence systems are highly sensitive to differences in the received signal power levels due to the non-zero cross correlation of the pseudo noise (PN) sequences assigned to individual users. This is known as the near-far problem [11]. The only solution to this problem is power control. Power control is difficult to implement in direct sequence spread spectrum. In frequency hopping systems, power control is not necessary.

2) Synchronization of direct sequence systems is more difficult to achieve than in frequency hopping systems, because of the former's high sample rate, corresponding to the chip rate.

For the same system performance, the implementation of a peer-to-peer wireless system based on frequency hopping technology is easier and simpler than one based on direct sequence technology.

3.3 Frequency Hopping System Requirements in 902 - 928 Mhz band

According to Industry and Science Canada RSS-210 or FCC Part 15, the requirements for a frequency hopping system operating in the 902 - 928 Mhz band [7] are listed below:

1) The minimum number of hopping frequencies is 50.

2) In a 20 second period, the average time occupancy on any hopping frequency cannot be over 0.4 second.

3) The peak output power cannot be over 1 W.

4) The maximum 20 dB bandwidth of the hopping channel is 500 Khz.

5) The Minimum hopping frequency separation is 25 Khz or the 20 dB bandwidth of the hopping channel, whichever is greater.

Knowing the kind of Spread Spectrum technique to be utilized in this peer-to-peer wireless system and the system operating frequency, the channel access scheme is introduced in the following section.

3.4 Channel Access Scheme of peer-to-peer Wireless system

In order to serve multiple users in this peer-to-peer wireless system, a channel access scheme is necessary. It is composed of a signaling channel and a message channel, where the signaling channel is for call set up and system synchronization, and the message channel is for exchanging data.

3.4.1 TDMA on signaling channel and FH on message channel

The signaling channel is a broadcast channel, occupying ideally only one frequency band. Sometimes more bands are used in order to avoid an access failure when the signaling channel is jammed. The message channel is assigned the remaining frequency bands. When this peer-to-peer wireless system operates in the 902 - 928 Mhz band, the total number of frequency bands is at least 50. In order to support multiple terminals while complying with RSS-210 or FCC Part 15, the signaling channel has been divided into many time slots. The number of time slots is determined by the total number of frequency bands assigned to the signaling channel and the message channel. By doing this, we can make sure that no single terminal will transmit on the frequency band assigned to the signaling channel more than 0.4 second in a 20 second period. On the other hand, all idle transceivers constantly monitor the signaling channel until a call setup is initiated. Thus, one can view the channel access scheme over the signaling channel as time division multiple access while over the message channel to be frequency hopping code division multiple access.

3.4.2 Switch between signaling channel and message channel

Given that the access scheme for the signaling channel is TDMA, while for the message channel is FH CDMA, the question becomes: how to put them together? The scheme proposed here answers this question.

Before talking about the channel access scheme, some relationships between the time slots on the signaling channel and the frequency band occupancy time on the message channel should be understood first:

1) The time duration of each time slot in the signaling channel should be equal to the frequency band occupancy time on the message channel. In other words, the time length of the time slot on the signaling channel is determined by the frequency hopping rate on the message channel.

2) The frequency hopping scheme used in this system is slow hopping, and the data is sent out and received by packet. For a fixed baud rate, the frequency hopping rate is determined by the packet length. That is, the time occupancy on each frequency band by the hopping terminals is determined by the packet length.

In order to describe the channel access scheme in an easier way, an example is shown in Figure 3.3. In this example, there is only one frequency band used for the signaling channel, and three frequency bands used for the message channel, so there are in total four time slots in the signaling channel.

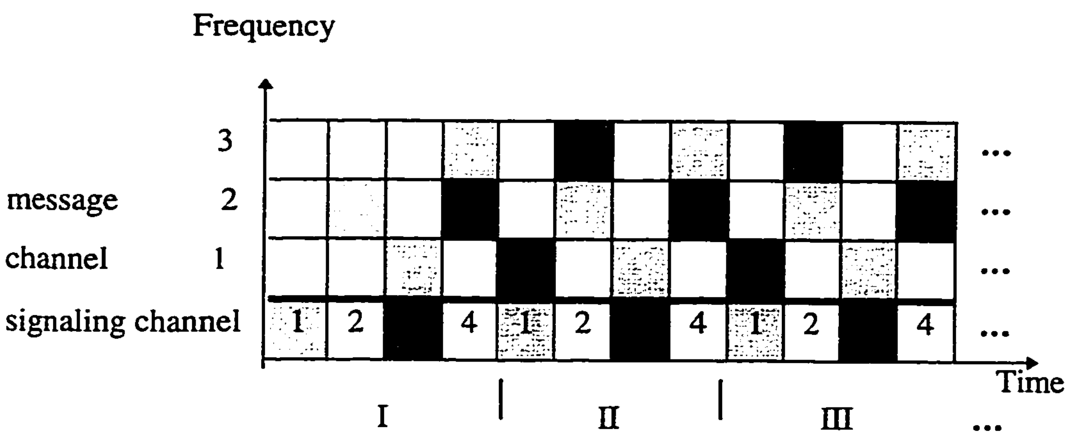


FIGURE 3.3 EXAMPLE OF THE CHANNEL ACCESS SCHEME

Narrow band frequency used by user pair A

Narrow band frequency used by user pair B

I, II, III, ...are cycle numbers

In Figure 3.3, three cycles are shown, and the frequency hopping pattern used in this example is 2 - 1 - 3 followed by a time slot at the signaling channel. This pattern is repeated in each cycle. For those who want to initiate a call, first they have to find an empty time slot on the signaling channel and use it to set up the call. After they finish setting up the call, both the one who initiated the call and the one who is called switch to message channel in order to communicate and hop according to the preset hopping pattern bands used for the message channel. When the hopping cycle is complete, it is time for the calling party and the called party to come back to the time slot they have chosen on the signaling channel to re-synchronize. The message and signaling cycle repeats until the call is finished. In Figure 3.3 the user pair A has chosen time slot 1 on the signaling channel, and user pair B has chosen time slot 3. According to the channel access scheme, no frequency bands are used by any two or more user pair at the same time. Theoretically co-channel interference cannot happen by using this channel access scheme.

3.4.3 The Selection of the Packet Length

As stated before, the time slot length in the signaling channel is determined by the frequency hopping rate in the message channel. Because this peer-to-peer wireless system is a slow hopping system, the hopping rate is determined by the transmitted packet length at a certain baud rate. As a result, the packet length is a critical value. There are several issues concerning the selection of the packet length:

1) **Time Delay**

In this peer-to-peer wireless system, time division duplex (TDD) [6] is used, as shown in Figure 3.4. In a TDD scheme, the information transmission is not continuous.

The delay between successive packet must be small enough to enable voice transmission. This must be a consideration while choosing the packet length. As illustrated in Figure 3.4, the largest time delay between two successive Tx or Rx happens when the calling party and the called party return to the signaling channel. That delay time is as long as 1.5 times the length of the time slot.

Sync/ Tx, Rx	**Tx**	**Rx**	**Tx**	**Rx**	**...**	**Sync/ Tx, Rx**	**Tx**	**Rx**	**Tx**	**Rx**	**...**
signaling channel	message channel					signaling channel	message channel				

FIGURE 3.4 TIME DIVISION DUPLEX

2) **Throughput**

Considering the time delay, the shorter the time slot the better. But under certain transmission rates (19.2 kbps for this system), if the time slot is too short, the throughput would be very low. In every packet, in addition to the data word, there are still three word packet header. Thus if the time slot is short, the packet length will be short. As a result the percentage of data information in every packet will be low and the throughput will be low. The usage efficiency of a time slot is calculated as shown below:

Time Slot = Time for programming the radio synthesizer * 2 (one for Tx, one for Rx)
+ Time for transmitting and receiving the packet header
+ Time for transmitting and receiving the data

$$\text{Efficiency} = \frac{\text{Time for transmitting and receiving data}}{\text{Time Slot}}$$

The throughput at varying bit error rates for the Hopper when working in FH mode is given by equation 3-2 [10] .

$$D = \frac{l\,(1 - 2P_{be}\,(l + l'))}{2*1 + 2*(48/50000) + (2*10/(8*50000))*(l + l')} \qquad (3\text{-}2)$$

Where: D is system throughput;

l' is the length of packet header in bits. In this system $l' = 48$ bits;

l is the length of data in bits;

P_{be} is the probability of a bit error;

$2 * 1\ ms$

is the time to settle the synthesizer;

$2 * (48 / 50000)$

is the time for bit synchronization at 50 kbps;

$(2 * 10 / (8 * 50000)) * (l + l')$

is the time for transceiving a packet including header and data.

The equation with $P_{be} = 0.00001$ and 0.0001 is plotted in Figure 3.5.

From Figure 3.5 we can see that when the packet length varies between 0 - 200 bits, the throughput changes dramatically, but after 200 bits, the throughput does not change very much. We can choose therefore a packet length near 200 bits per packet, so that the packet is not too big, the delay is acceptable and the throughput is reasonable.

The packet is chosen as 15 words per packet. There are 16 bits per word. The packet length is 240 bits, and include a 48 bit packet header and 192 data bits. For this packet length, the throughput is about 12 kbps and if transmitted at 19.2 kpbs. It takes less than 16 ms to transmit one packet. According to this, the time slot length can be set at 16 ms. In this case, the largest delay is 24 ms, which is shorter than the 40 ms used on IS-136.1-A TDMA cellular system [12].

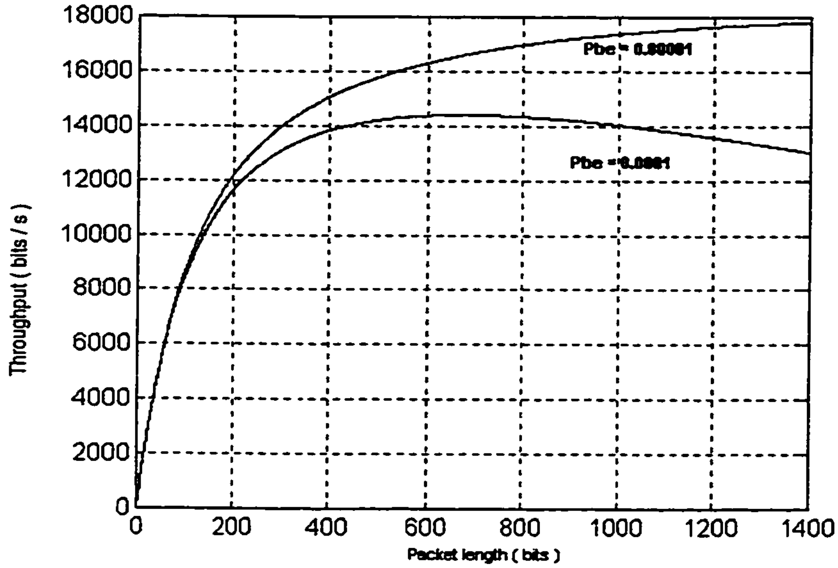


FIGURE 3.5 THROUGHPUT AT TWO BIT ERROR RATE

Chapter 4

IMPLEMENTATION OF THE PEER TO PEER WIRELESS SYSTEM

4.1 Introduction

This chapter covers the specifics of the algorithm used to implement the peer-to-peer wireless system. Detailed operation of the algorithm and implementation are described, including the signaling channel and the message channel as well as some design limitations and requirements.

4.2 Implementation Limitations and Requirements

Since this peer-to-peer wireless system is implemented using the Hopper wireless modems made by Wi-LAN Inc., familiarity with the Hopper would be helpful.

The central processing unit in the Hopper is an Analog Devices 2101 Digital Signal Processor (DSP). The details of the hardware and processor are described in Appendix A. From the architectural design we can see that the lack of some important features increased the difficulty of this implementation. For instance, this processor can only be programmed using assembly, which increased the programming complexity, and reduced the maintainability. The fact that only 2048 24 bit words of program memory and 1024 16 bit words of data memory are available also caused problems. In order to run the software of the peer-to-peer wireless system, the program had to be sectioned into "pages". When required, the DSP would be rebooted to load the appropriate program page. This caused some difficulties in the system synchronization. However, this processor does have some advantageous features: it has a " secondary " set of registers, and can be switched in one processor cycle. It is useful for interrupts as the context switch time is minimal.

Another factor to care about is the original code in the Hopper. I used only parts of the original code with modification due to the requirements for this peer-to-peer wireless system, and then included my code on top of it. So it was necessary to grasp the key settings in the original code. Table 4.1 shows the time interval between bits while the Hopper working in FH operation [10].

Baud Rate (bps)	Time between Interrupts	
	μs	**Processor Cycles**
frequency hopping Operation		
24992	13.4	213
52805	6.3	100

Table 4. 1 Time interval between bits

In the original FH code each bit must be dealt with three times for synchronization purposes [10]. From Table 4.1 we can see that if the Hopper is running at a baud rate of 52805, there are only 100 processor cycles between interrupts. This is tight, because some flags must be checked and some logical selections must be processed between interrupts, otherwise the code would be working in a different way from the way it should be.

4.3 Code Structure

The peer-to-peer wireless system implemented in this thesis is a single-hop system, system initialization is not necessary. Assuming the network connectivity exists with each terminal being able to communicate with all other terminals in the system, the only thing that needs to be done is to assign every terminal in the system a unique address. The advantage of this system is that a new terminal can be added at any time without disturbing the system.

Since the Hopper wireless modem was used in this implementation as the peer to peer wireless terminal, its protocol structure must be explained. The main high level functions, which compose the program, are shown below [10]:

1. UART
2. Service transmit request
3. Service transmit end request
4. Service receive request
5. Service receive end request
6. Timer handler
7. Flow control
8. Rx int
9. Tx int

The protocol operation of the Hopper is controlled by the Service transmit request, Service transmit end request, Service receive request, Service receive end request, and the Uart. The Flow control controls the Hopper's UART. The Rx int and Tx int are two interrupt driven routines used for talking to the RF physical layer.

The protocol of the peer-to-peer wireless system is composed of two main parts:

1) signaling channel ; and

2) message channel.

The signaling channel protocol is responsible for handling call set up. All procedures for setting up a call are processed in this function including the system synchronization scheme. In the signaling channel, three subroutines, Uart, Timer handler and Flow control are called. The message channel protocol is responsible for exchanging data after the call is set up. In the message channel, all subroutines existing in the Hopper are called for transmitting and receiving.

As mentioned before, this peer-to-peer wireless system uses spread spectrum technology. In the signaling channel, it uses TDMA. In the message channel, it uses FH. Figure 4.1 shows the relation between the signaling channel and the message channel.

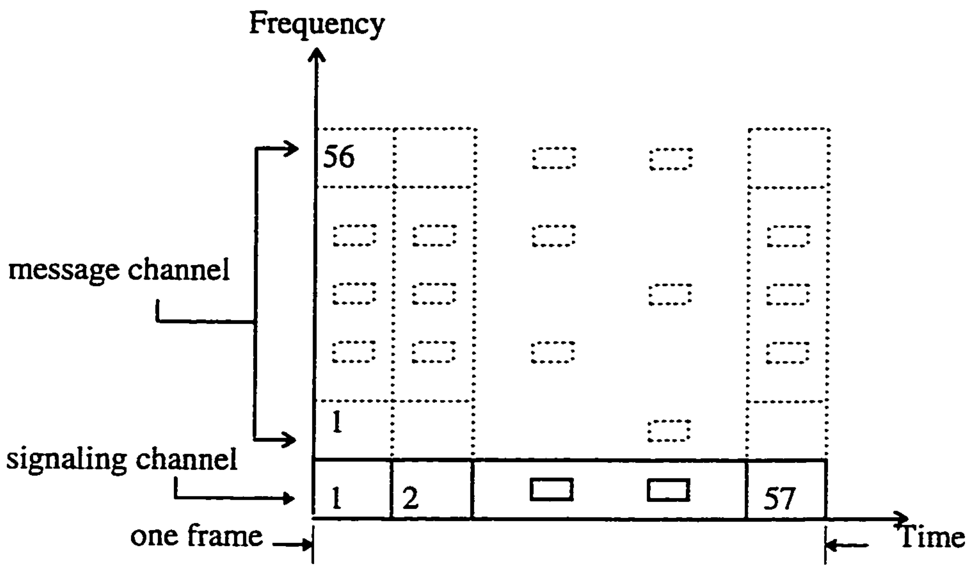


FIGURE 4.1 SIGNALING CHANNEL & MESSAGE CHANNEL

4.4 Signaling Channel

The signaling Channel in Figure 4.1 uses TDMA with 57 slots per frame. Each time slot lasts 16ms at 19.2kbps. Every slot of the signaling channel corresponds to a message channel. In each message channel, there are 56 different frequencies, the use of each frequency will be chosen by a PN sequence. The reason why 57 is chosen for the number of slots per frame is that according to the channel access scheme, the number of slots in the signaling channel should be equal to the number of frequency bands used in the message channel plus the number of frequency bands used in the signaling channel. Since the message channel is assigned 56 different frequency bands for frequency hopping, and only one more frequency band is used for the signaling channel, there are a total of 57 frequency bands used in this peer-to-peer wireless system. The signaling channel time slots are shown in Figure 4.2.

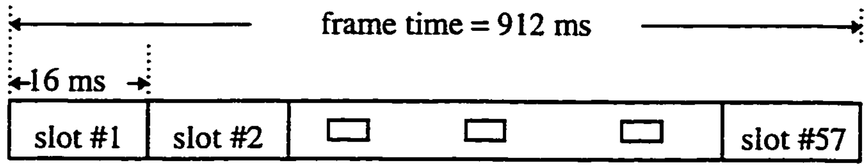


FIGURE 4.2 SIGNALING CHANNEL TIME SLOTS

This channel access scheme meets two of the ISM band requirements:

1. Since the frame time of the signaling channel is chosen to be 912 ms, each of the 57 time slots has a duration of 16 ms. Each message channel contains 56 distinct frequencies. During a period of 912 ms, a user pair hops on 56 distinct frequencies on the message channel and on the signaling channel once. This brings the total number of hopping frequencies to 57 which is greater than the minimum requirement of the ISM band.
2. The dwell time of each frequency of the message channel is 16 ms within a period of 912 ms. Therefore, within a 20 second period, a user pair occupies each frequency for a total of 350.88 ms. This meets the ISM band requirement.

4.4.1 Signaling channel protocol

The signaling channel is used for setting up a call and system synchronization.

The basic structure of the signaling channel is shown in Figure 4.3.

In order to fulfill this function, several packets are used in the signaling channel.

4.4.1.1 Signaling channel packet types

Each time slot of the signaling channel could be occupied by one of the following five packet types:

1. Idle Packet (an empty slot)
2. Booking Packet
3. Call request Packet
4. Call Busy Packet
5. Call ACK Packet

All the packet types are shown in Figure 4.4.

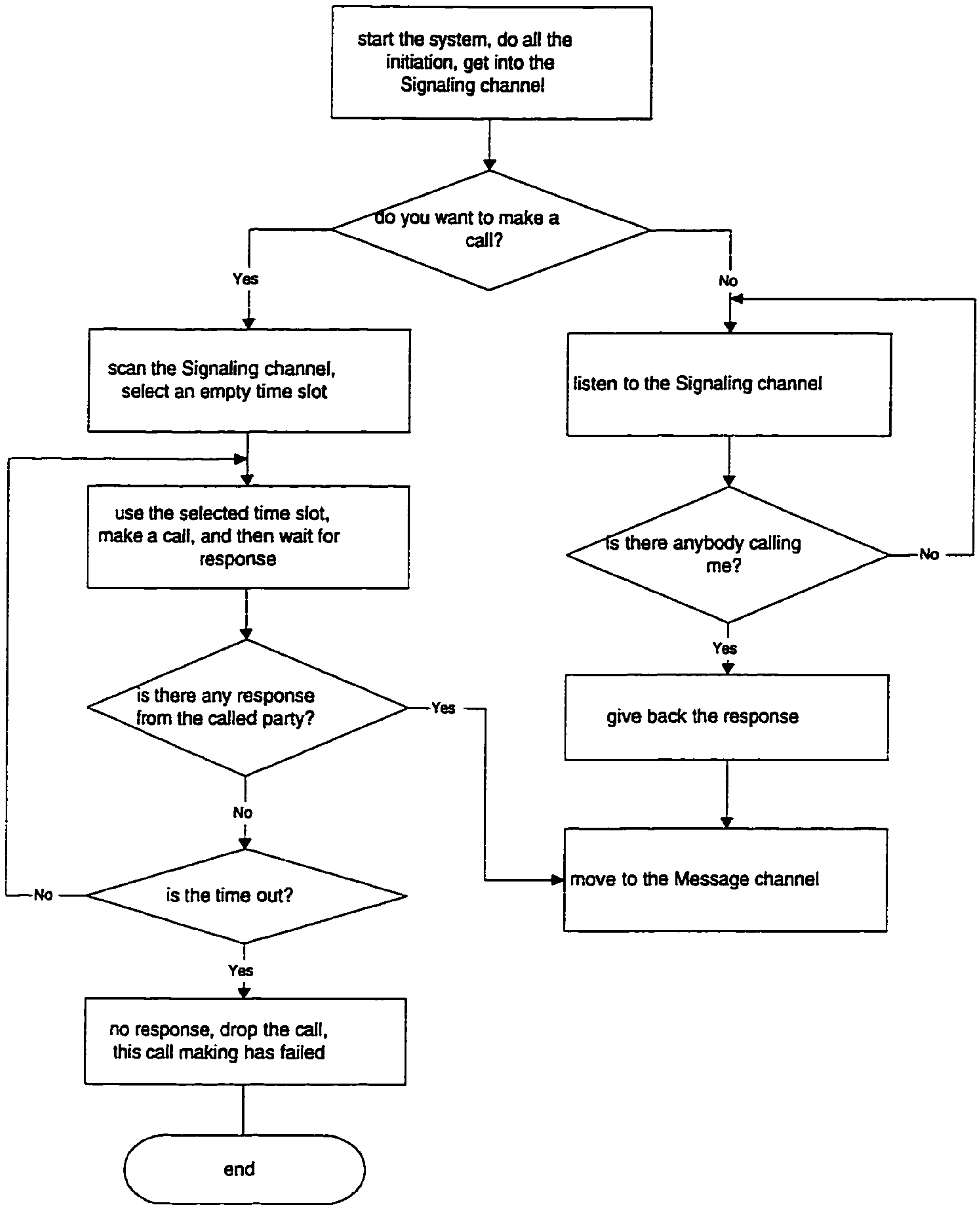


FIGURE 4.3 LOGIC FLOW CHART OF SIGNALING CHANNEL

Empty Packet (Empty Slot)

(a)

<table>
<tr><td></td><td>upper byte</td><td colspan="2">lower byte</td></tr>
<tr><td>first word</td><td colspan="2">Reserved (13 bits)</td><td>Type (3 bits)</td></tr>
<tr><td>second word</td><td>Calling Address (8 bits)</td><td colspan="2">Calling Address (8 bits)</td></tr>
</table>

(b)

<table>
<tr><td></td><td>upper byte</td><td colspan="2">lower byte</td></tr>
<tr><td>first word</td><td>Reserved (7 bits)</td><td>Slot ID (6 bits)</td><td>Type (3 bits)</td></tr>
<tr><td>second word</td><td>Called Address (8 bits)</td><td colspan="2">Calling Address (8 bits)</td></tr>
<tr><td>third word</td><td colspan="3">Synchronization Information (16 bits)</td></tr>
</table>

(c)

<table>
<tr><td></td><td>upper byte</td><td colspan="2">lower byte</td></tr>
<tr><td>first word</td><td>Reserved (7 bits)</td><td>Slot ID (6 bits)</td><td>Type (3 bits)</td></tr>
<tr><td>second word</td><td>Called Address (8 bits)</td><td colspan="2">Calling Address (8 bits)</td></tr>
<tr><td>third word</td><td colspan="3">Synchronization Information (16 bits)</td></tr>
</table>

(d)

<table>
<tr><td></td><td>upper byte</td><td colspan="2">lower byte</td></tr>
<tr><td>first word</td><td>Reserved (7 bits)</td><td>Slot ID (6 bits)</td><td>Type (3 bits)</td></tr>
<tr><td>second word</td><td>Called Address (8 bits)</td><td colspan="2">Calling Address (8 bits)</td></tr>
</table>

(e)

FIGURE 4.4 PACKET TYPES

(a) Idle Packet
(b) Booking Packet
(c) Call Request Packet
(d) Call Busy Packet
(e) Call Acknowledgment Packet

An empty slot (Idle Packet) indicates that this slot is available for use.

A slot with booking packet indicates that this slot is booked by someone for future use, and is not available for others.

A slot with Call Request Packet indicates that this slot is used for setting up a call. When the calling party wants to make a call, after he found an empty time slot, he has to send this Call Request Packet to inform the called party. When the called party receives this Call Request Packet, he knows that somebody is calling him.

A slot with Call Busy Packet indicates that this slot is used for processing a call, and its corresponding hopping pattern is used. It is also used for synchronization of a user pair. The slot ID of either the Call Request Packet or Call Busy Packet links to a frequency hopping pattern (message channel). Therefore, any user who seizes a slot will automatically acquire a message channel.

A slot with Call Acknowledgment Packet indicates that a call is set up in this time slot, and this time slot is not available for others. A Call Acknowledgment Packet is used by the Called party to respond to the calling party and accept the call.

4.4.2 Media Access Control Protocol

A Media Access Control (MAC) Protocol is basically a set of rules applying to any user (terminal) who initiates a call to access an idle time slot on the signaling channel.

The MAC Protocol used in this implementation is :

After the user goes off-hook, the terminal must:

1. Scan the signaling channel for 912 ms(1 period);

2. Determine which slots are free for use, and choose one free slot.

After the Calling party finishes scanning the signaling channel, three situations can be met:

1. All the signaling channels are empty, nobody is using the channel.

2. Part of the signaling channel is empty, somebody is using the channel, and the channel is not full.

3. The signaling channel is full; all slots are in use.

According to these three situations, the terminal will give out different responses.

In situation 1, the first time slot is selected by the terminal for processing its call set up. As shown in Figure 4.5.

In situation 2, the first unused time slot after the first used time slot is selected. As illustrated in Figure 4.6.

In situation 3, since no time slot is available, the user cannot initiate a call. The terminal cannot do anything, except for being given the "channel full" information. It has to try it again later, if it still wants to initiate a call.

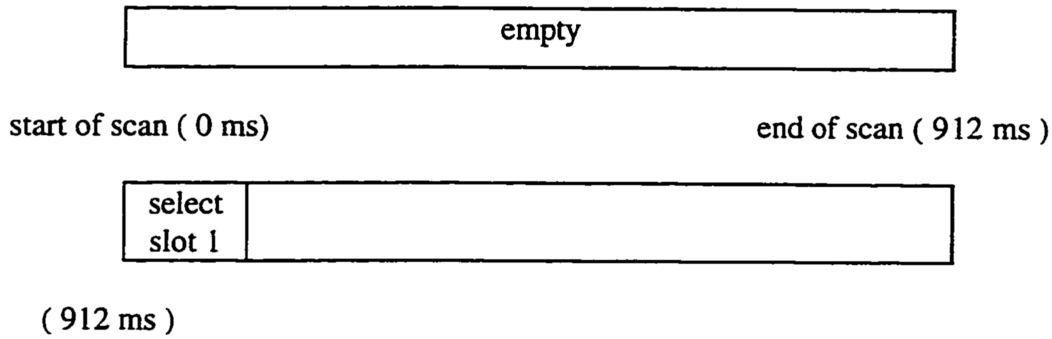


FIGURE 4.5 SELECTED SLOT FOR EMPTY SIGNALING CHANNEL

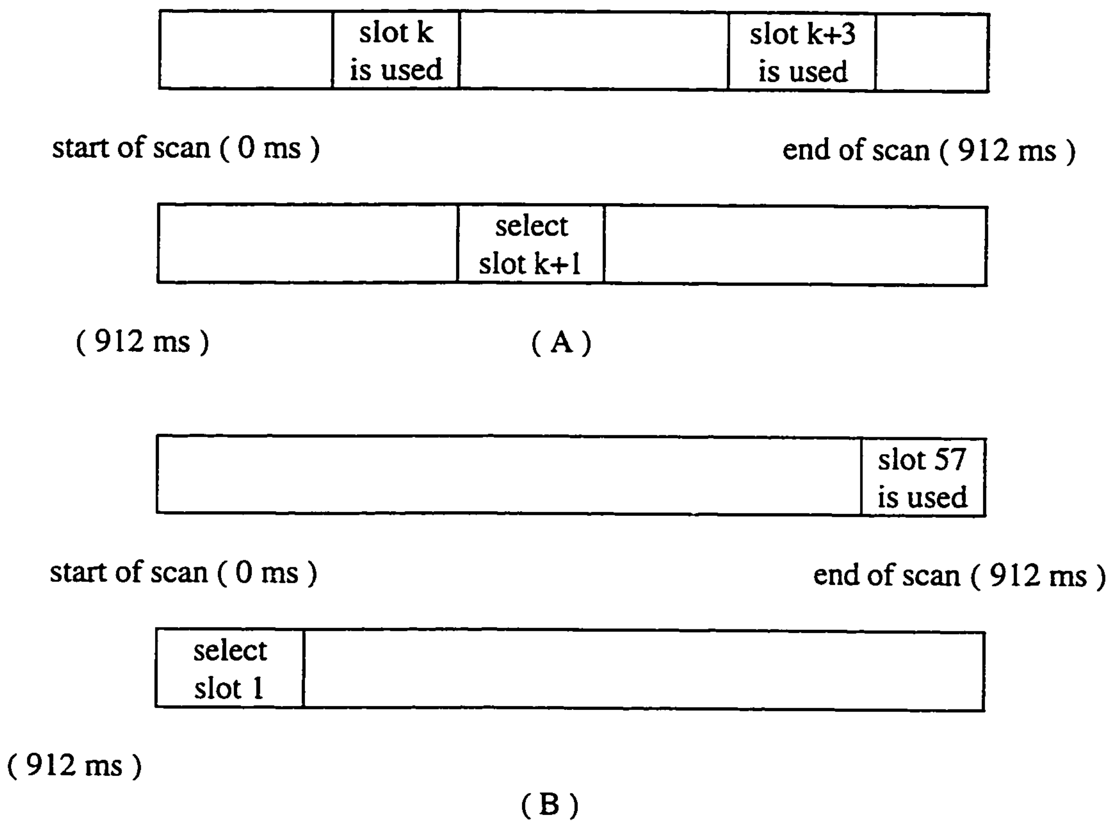


FIGURE 4.6 SELECTED SLOT FOR NON-EMPTY SIGNALING CHANNEL

4.4.3 System Synchronization

Since this peer-to-peer wireless system is a single hop system, no system initialization is required, but it does need synchronization among all terminals. With respect to the person who intends to initiate a call (the calling party), two kinds of synchronization could happen in the MAC protocol:

1. Corresponding to the empty signaling channel, the calling party does not need to synchronize with anyone else, because no one is using the signaling channel. In this case, the calling party will name the time slots according to his own timing.

2. Corresponding to the non-empty non-full signaling channel, the calling party will synchronize himself with the one who is using the time slot ahead of him. In this case, the calling party has to name the time slot according to the timing of the user he is synchronized with.

With respect to those who do not want to initiate a call, no synchronization will happen until one of them receives a call request packet, and then he will synchronize with the one calling him. Generally, all the terminals, which are in call processing, including call set up in the signaling channel and the exchange data in the message channel, will be synchronized, and those, which are not in call processing, will stay idle.

The logic flow chart of the implementation of this MAC protocol is shown in Figure 4.7. This MAC protocol applies to call initiators. From the flow chart, one can notice that the one who doesn't want to initiate a call, will keep listening to the signaling channel waiting for someone to call him. He can catch everything happening on the signaling channel, but only respond to Call Request Packet directed to him. After he receives the Call Request Packet, several things need to be processed, as shown in Figure 4.8.

After sending out the Call Request Packet, the call initiator switches to receiving mode, not only to receive the Acknowledgment from the called party, but also to check the availability of the called party. Three possible results can be obtained:

1. The expected Acknowledgment is received.
2. From the received packet, it is figured out that the called party is not available, because he is engaged in another "conversation".
3. Nothing related to the called party is received in the time slot which the calling party used.

Result 1 and result 2 are normal, but 3 is not. Two situations can cause result 3 to happen. One situation is that the called party didn't receive the call request packet from the calling party. This may be caused by bit errors or the off-line of the called party, so no ACK will be received by the calling party. After receiving a packet, the receiving party will check a number of things to determine if it is the expected packet or not. These include calling and called address, slot_ID, and packet type. If there are bit errors in the received packet, the packet will be overlooked. Because of this, the code cannot stop here, the calling party has to try a number of times. If the Acknowledge still cannot be received, the calling attempt fails.

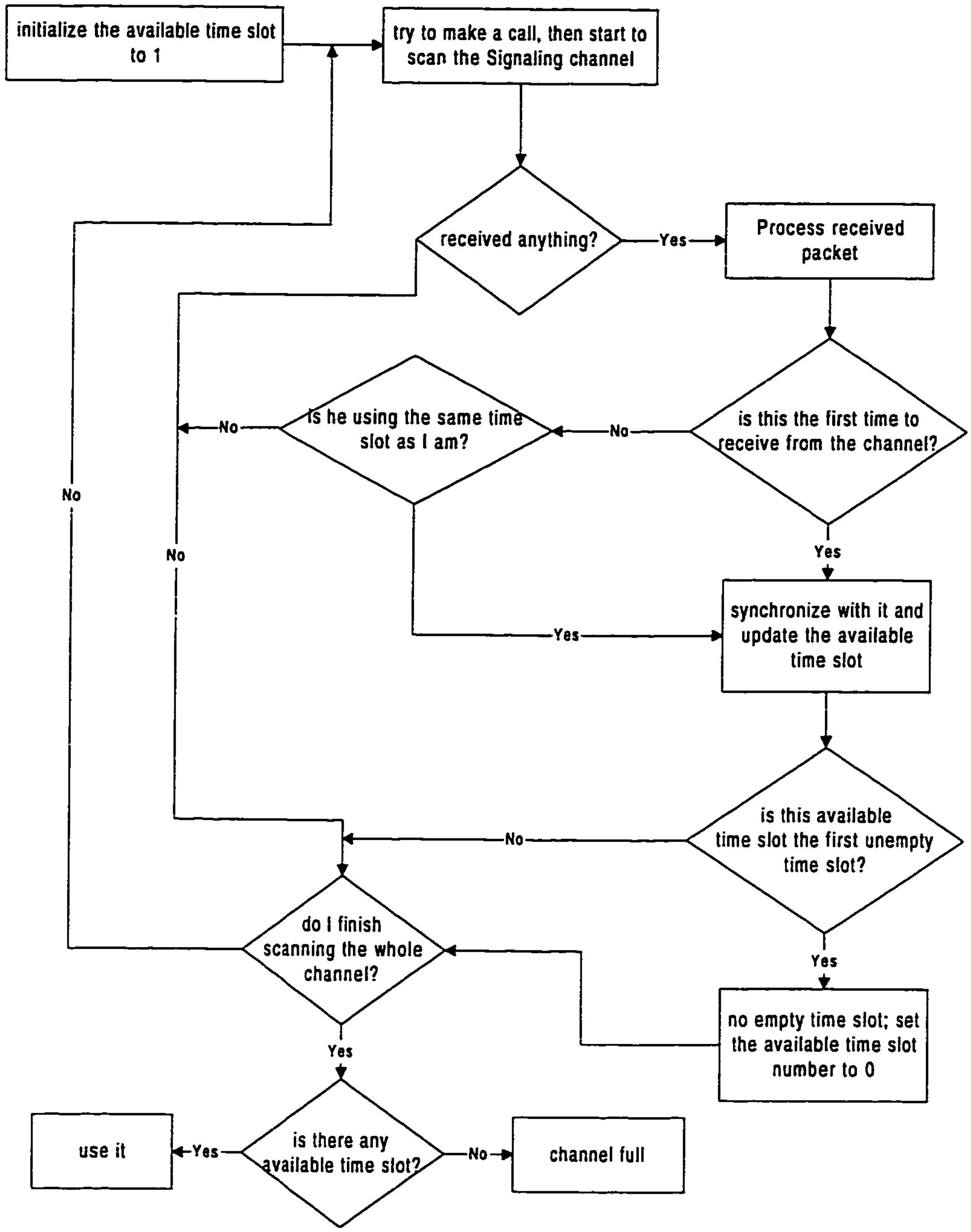


FIGURE 4.7 LOGIC FLOW CHART OF MAC PROTOCOL

It can happen that the called party receives the call request packet, and acknowledges, but the calling party does not receive it. In this case, the called party moves to the message channel by himself. This will also cause failed communication. Under this situation, the only thing that can be done is manually reset the modem. This is just like when we hear the phone ring, pick up the phone, say several " Hello". If there is no response from the other side, we just hang up the phone. The logic flow chart is shown on Figure 4.9.

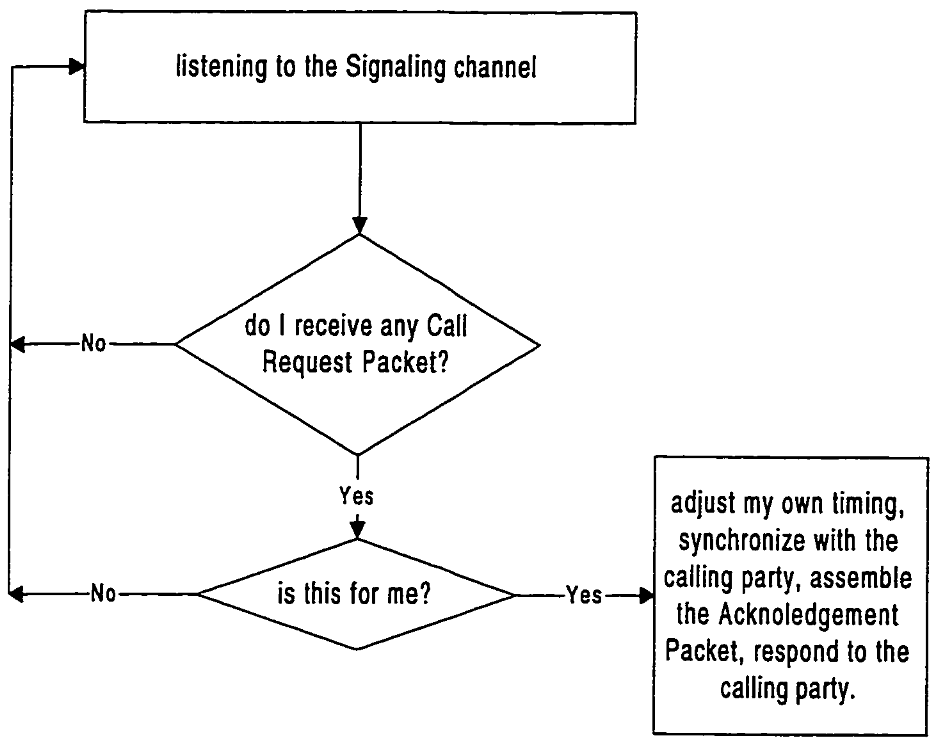


FIGURE 4.8 FLOW CHART FOR ACQUIRING THE CALL REQUEST PACKET

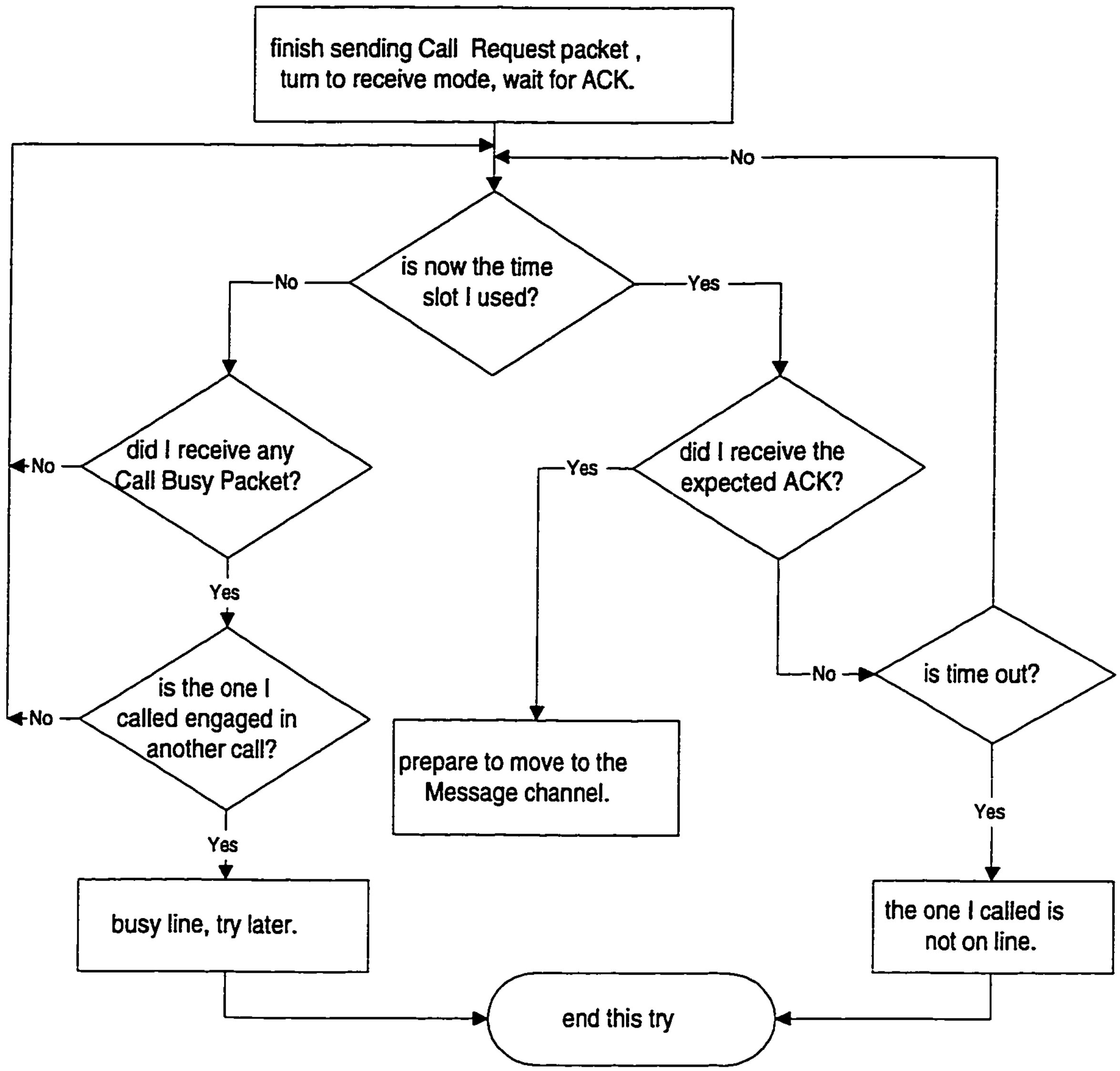


FIGURE 4.9 FLOW CHART OF RECEIVING ACKNOWLEDGMENT

4.5 Message channel protocol

The message channel is used for exchanging information after one user pair has been linked using the signaling channel. As previously mentioned, frequency hopping technology is employed in the message channel.

There are in total 57 different frequency bands used by the peer-to-peer wireless system. Among them, one frequency band is assigned to the signaling channel, the other 56 frequency bands are assigned to the message channel.

Switching from the message channel to the signaling channel and back from the signaling channel to the message channel happens once every cycle. The flow chart of the implementation is shown in Figure 4.10.

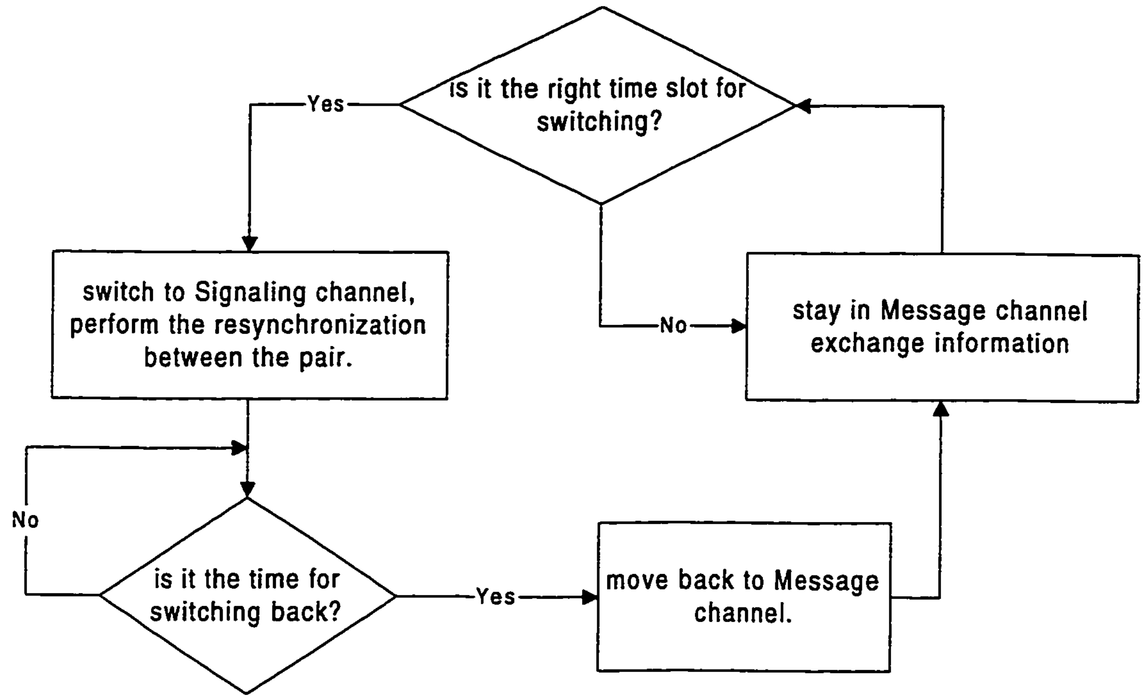


FIGURE 4.10 FLOW CHART FOR SWITCHING BETWEEN THE MESSAGE CHANNEL AND THE SIGNALING CHANNEL

In the message channel, every use of one of the 56 different frequency bands is controlled by a PN sequence. The frequency assigned to the signaling channel is not used by the message channel through the control of the PN sequence. By separating the frequency band used by the signaling channel from those frequency bands controlled by the PN sequence, the control logic for managing the signaling channel and the message channel becomes simple. Whenever we want to switch to the signaling channel from the message channel, we need not worry about disturbing the PN sequence. We can simply stop the PN sequence, which stops the Frequency hopping, switch to the signaling channel, perform all tasks required by the signaling channel, switch back to the message channel, and let the PN sequence restart where it was stopped. The channel switch technique is shown in Figure 4.11.

Recall that all the transmitted and received packets in the signaling channel are three word long, but all the transmitted and received packets in the message channel are 15 word long. These 15 words are composed of 3 header words and 12 data words. From Figure 4.11 we can also tell the difference between the packet length of the signaling channel packet and of the message channel packet.

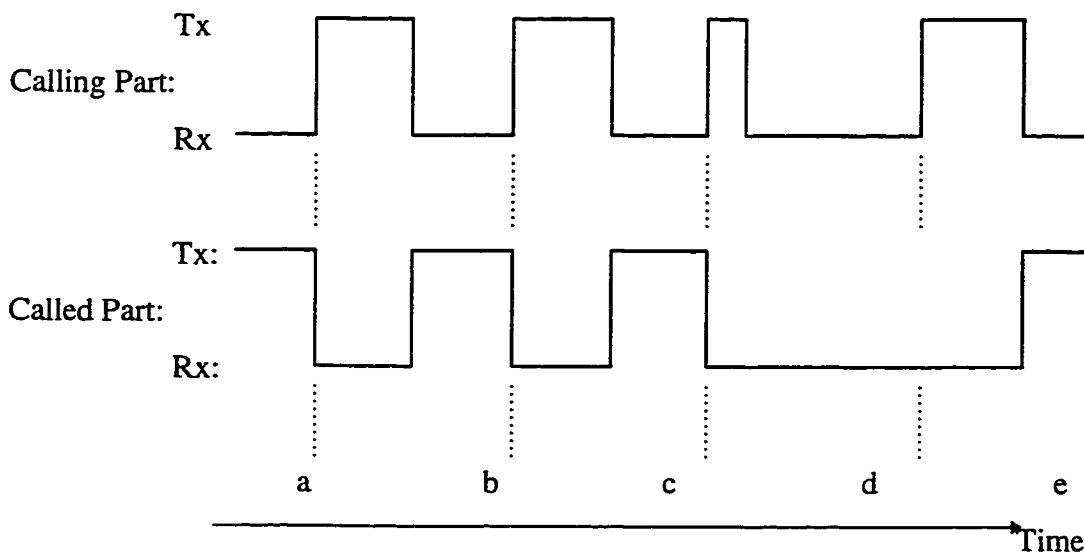


FIGURE 4.11 SWITCH FROM THE MESSAGE CHANNEL TO THE SIGNALING CHANNEL

In Figure 4.11, all labeled points 'a', 'b', 'c', 'd' and 'e' are where the frequency is changed. ab, bc, cd and de are four different time slots, where ab, bc and de are in the message channel, and cd is in the signaling channel. As previously mentioned, in the time slots ab, bc and de the frequency chosen is controlled by the PN sequence. One should notice that when switching back to the signaling channel, the behavior of the calling party and that of the called party is not the same. On the signaling channel the calling party transmits a three word long Busy Packet. This three word Busy Packet has two functions: (1) to inform others that this time slot is being occupied; (2) to synchronize the called party. The called party does nothing on the signaling channel but listens to it, and tries to acquire the Busy Packet sent out by the calling party. After acquiring the Busy Packet, the called party will use the information carried in the busy-packet to finish the re-synchronization within this user pair.

There is no acknowledgment sent back to the calling party by the called party, after the called party receives the Busy Packet. If by chance the called party does not receive the Busy Packet he is supposed to receive, by the end of the time slot of the signaling channel, he will leave the signaling channel, switch back to the message channel, and keep transmitting and receiving. When the time comes for switching back to the signaling channel, the called party tries again to receive the Busy Packet. If the called party does not receive the busy-packet for many cycles, he might loose his synchronization with the calling party, because of the time drift between internal clocks.

In the case of losing synchronization, there is a special treatment for this. When synchronization is lost between the pair, the called party fails to receive anything from the calling party. When the called party fails to receive anything from the calling party, he knows he lost his synchronization with the calling party. In this case, he jumps ahead 4 frequency bands, and stays there and waits until he receives the calling party again. This scheme can be used since the calling party transmits in every frequency band. Even if there is no data to transmit, he will transmit some dummy characters. On the other hand, the called party only transmits if he has data to send out, otherwise, he only receives. The synchronization recovery mechanism is shown in Figure 4.12.

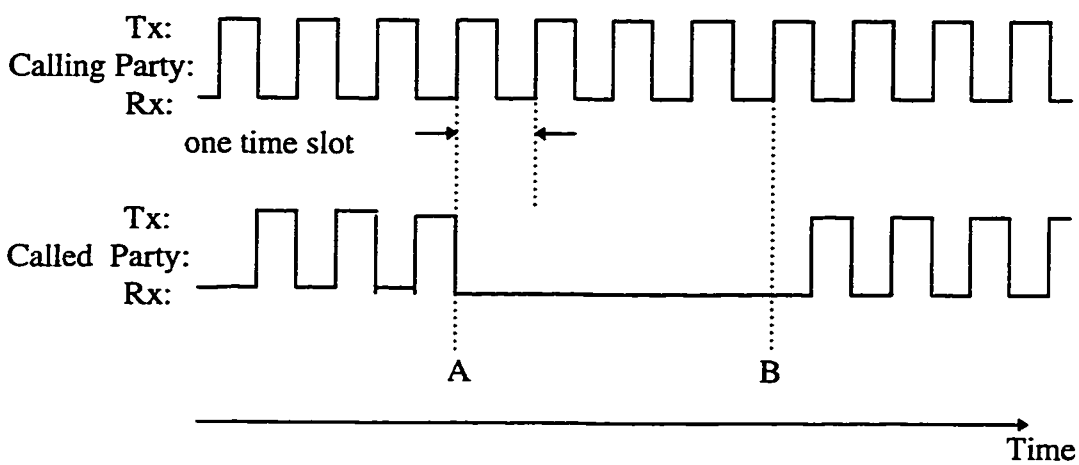


FIGURE 4.12 LOST SYNCHRONIZATION RECOVERY

A: Lose Synchronization B: Recover Synchronization

From Figure 4.12 we can also notice that every time slot is separated into two parts: one part for transmitting, and one part for receiving. In the real implementation the transmitting part does not occupy the same time length as the receiving part in every time slot, since the receiving part is 1 ms longer than the transmitting part. The reason for this is to make sure that the receiving party can receive what the transmitting party transmits. For the same reason, after the calling party switches back to the signaling channel, he does not transmit the Busy Packet right away. Instead he waits 1 ms, and then transmits the Busy Packet. See Figure 4.13.

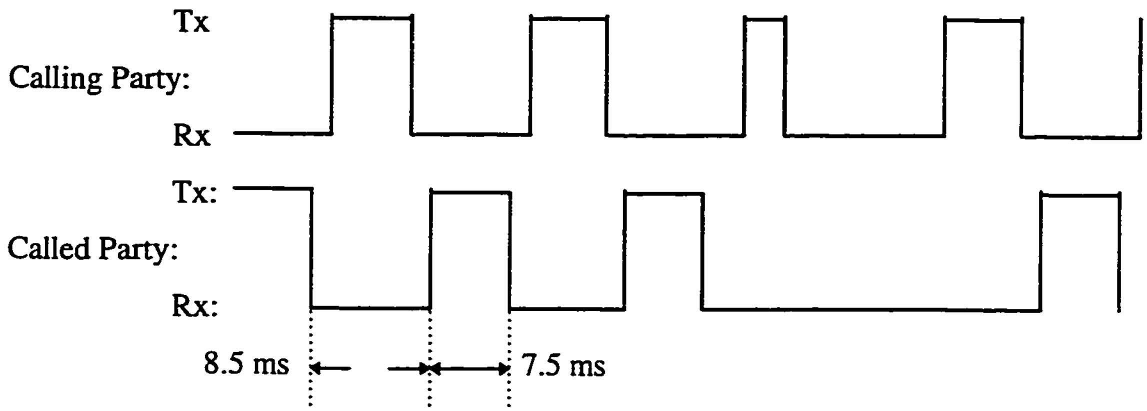


FIGURE 4.13 ACTUAL TRANSMIT AND RECEIVE CYCLE

4.6 Second signaling channel

One of the advantages of frequency hopping over other access schemes is anti-jamming. However, since our peer-to-peer wireless system only employs FH over the message channel when the signaling channel is jammed, the whole system eventually fails, because no new calls can be set up, and no re-synchronization can be performed either.

Because of such a critical reason, this peer-to-peer system needs a second signaling channel, so that if the first signaling channel is jammed, the second one can be used. One requirement for the second signaling channel is that it has to be a certain frequency range away from the first signaling channel, otherwise it is possible that the two signaling channels be jammed by the same jamming signal simultateously.

Before adding the second signaling channel, one can recall that one frequency band is used for the signaling channel, while 56 frequency bands are used for the message channel. After adding the second signaling channel, there will be 58 frequency bands in total. Since the number of time slots is always equal to the number of the frequency bands, if one more frequency band is added, one more time slot should be added too. That means there should be 58 time slots in one cycle. But in fact, adding one more time slot is not necessary, there are two points we can take advantage of in order to avoid adding one more time slot. First, the frequency bands used for the signaling channel are set separately from those used for the message channel, that is the selection of the frequency is not controlled by the PN sequence. Second, one time slot lasts 16 ms long, while transmitting one packet over the signaling channel lasts only 4 ms, so we can use the remaining 12 ms to move to the second signaling channel and to transmit there too. Thus, two frequency bands for the signaling channel and 56 frequency bands for the message channel can work fine in 57 time slots. Now the time slot used for the signaling channel is like the one shown in Figure 4.14.

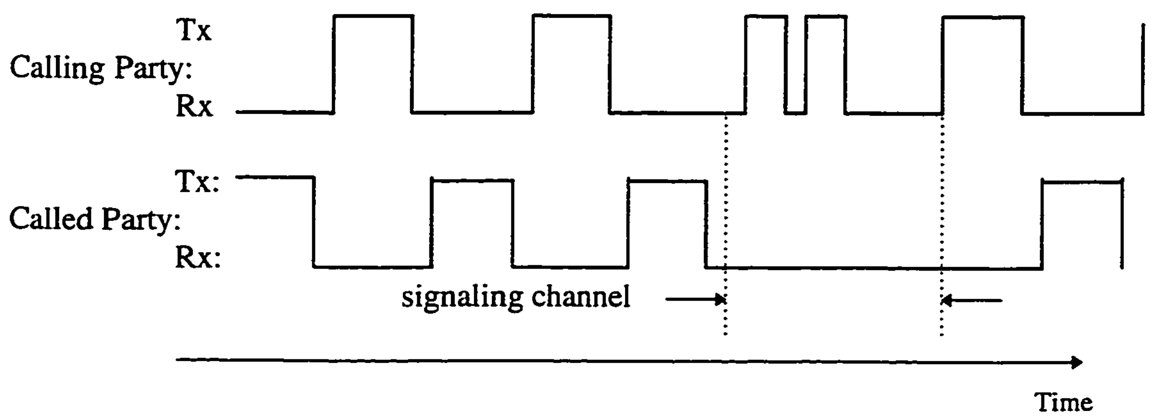


FIGURE 4.14 TWO SIGNALING CHANNEL IN ONE TIME SLOT

4.6.1 Modifications of the code

4.6.1.1 Initiate a call

After adding the second signaling channel, some modifications on the code are necessary. First, for those who want to initiate a call, compared with before, the second signaling channel does not affect them very much. Before, the calling party used to send the call request packet repeatedly over the signaling channel until the expected acknowledgment was received. Now, the calling party sends the call request packet alternately over the two signaling channels until the expected acknowledgment is received on either one. The logic flow chart is shown in Figure 4.15.

In Figure 4.15 there is a decision made on " did I find the same empty time slot on both signaling channels". Why do I need this decision? Usually, the same empty time slot is found in both signaling channels, but when one signaling channel is jammed, there is a chance that the calling party does not find the same empty time slot in both signaling channels. The reason for this is directly related to the way the Hopper is used for packet synchronization. The Hopper uses a packet synchronization header, which is composed of 48 bits, for packet synchronization. Before every packet is transmitted, this packet synchronization header is sent first.

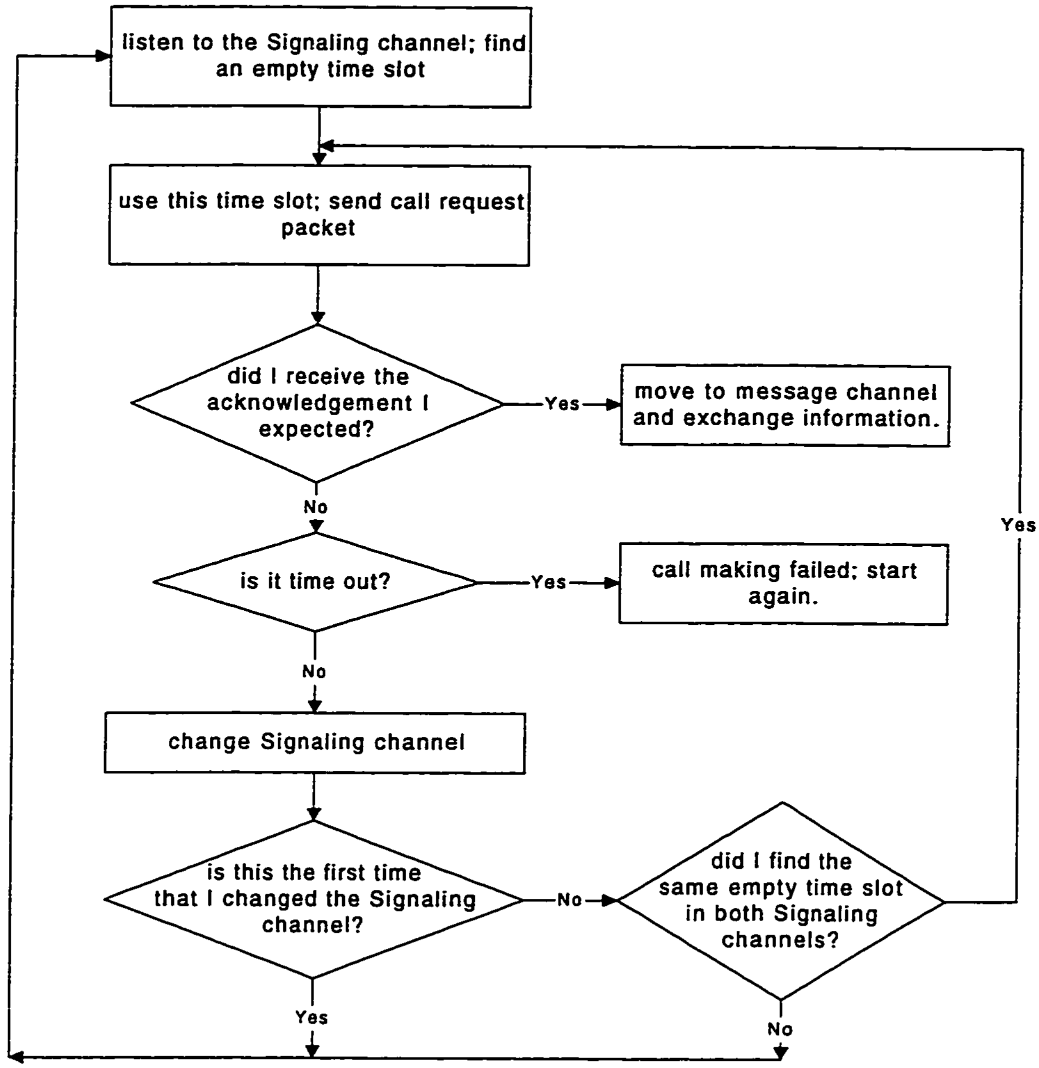


FIGURE 4.15 FLOW CHART OF MODIFIED INITIATING A CALL

When the channel is jammed, no packet synchronization header can be received, so to the one who is scanning the channel, the channel is empty, and consequently the time slot can be picked up.

4.6.1.2 Listening to the channel

Now let us turn to those who do not want to initiate a call. Before the second signaling channel was added, the only thing they needed to do was to stay in the signaling channel and listen to it. Now they have to do the same thing as before but listen to both signaling channels instead of only one. The logic for their switching between two signaling channels is very simple, they just stay in every signaling channel for five cycles and then move to the other signaling channel until they receive a call request packet. This is shown in Figure 4.16.

After the call is set up, both the calling party and the called party move to the message channel. In each cycle, the calling party uses the booked time slot to send out a busy packet, and the called party uses the booked time slot to acquire the busy packet to fulfill his re-synchronization to the calling party. The calling party, as mentioned before, can send out the Busy Packet on both signaling channels within one booked time slot, but the called party cannot listen to both signaling channels within one booked time slot. This leads to the following question: which signaling channel should the called party listen to? The answer is: both, but not simultaneously. This means that the re-synchronization cannot depend only on ONE signaling channel. The method for selecting the signaling channel for the called party in this system is:

1 First listen to the signaling channel on which the call was set up;

2 If the called party does not acquire a busy packet in two consecutive cycle,

3 move to the other signaling channel.

If the called party does not receive a Busy Packet for many cycles, thereby losing his synchronization with the calling party, the same scheme is applied as the one for the single signaling channel system.

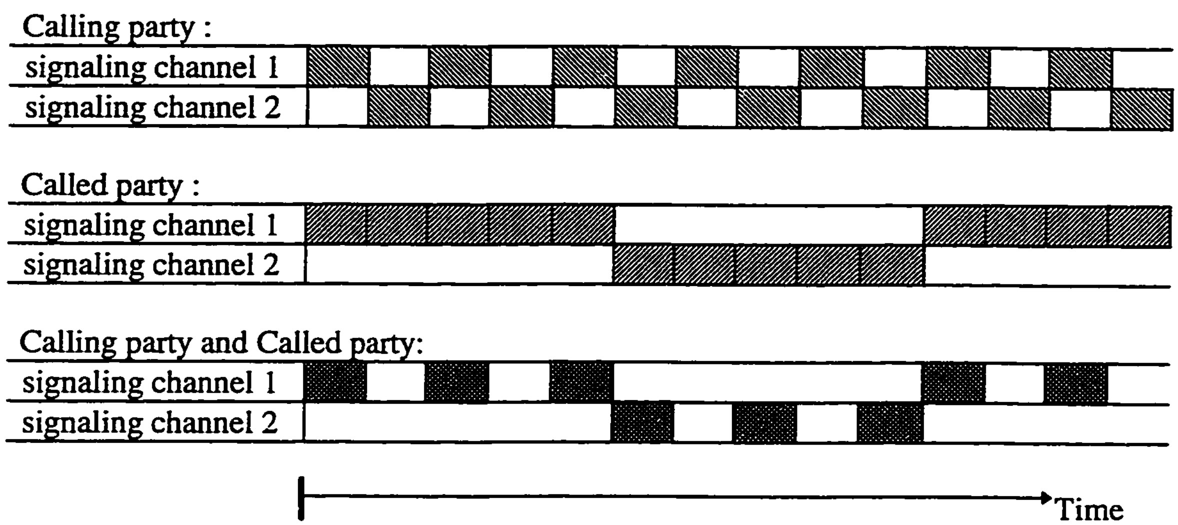


FIGURE 4.16 CHANNEL OCCUPATION OF THE CALLING PARTY AND THE CALLED PARTY

Signaling channel employed by calling party in one cycle

Signaling channel employed by called party in one cycle

Signaling channel employed by calling party and called party in one cycle

4.7 Reboot a new page

Before moving to the next chapter, there is one more thing to mentione. For the ADSP-2101 chip, there are only 2048 24 bit words of instruction memory, this amount of programmable memory is very limited. In another word, 2048 24 bit words are not enough to program both signaling channel and message channel. So rebooting a new page enough to program both signaling channel and message channel. So rebooting a new page is necessary in the middle of the code. Here the software-forced rebooting is employed. The software-forced rebooting can be accomplished by setting the BFORCE bit in the processor's system control register.

There is a very critical effect to the code caused by rebooting a new page. During the booting and rebooting, all interrupts including serial port interrupts are masked. This means that no interrupt could happen during this period of time. The wireless modem Hopper is an interrupt driven modem, and all the peer-to-peer system synchronization is based on interrupts. So to the pair of users, who encounter the rebooting, this period of time for rebooting is "missing", just like a time gap. This time gap has to be erased. Otherwise, after rebooting the new page, a synchronized system will lose the synchronization between those who encounter rebooting a new page and those who did not encounter rebooting a new page. The simplest way to solve this problem is that before rebooting a new page, let the timer inside the terminals jump ahead of the amount of time which is needed for rebooting a new page. The rebooting time will vary from modem to modem, but can be tested. Example is shown in Figure 4.17.

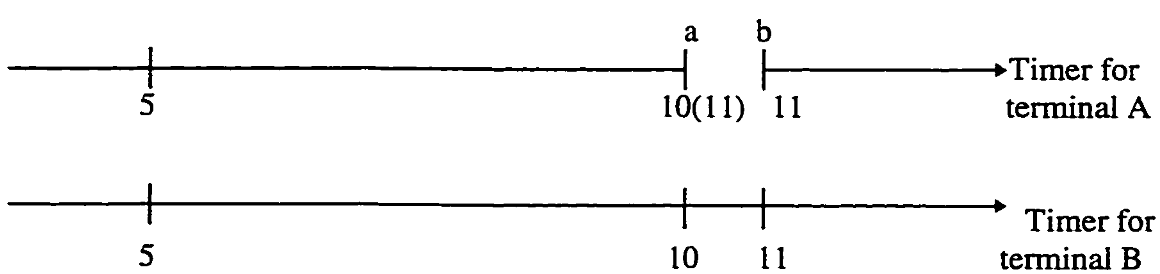


FIGURE 4.17 EXAMPLE OF REBOOTING THE PAGE

From timer for terminal A we can see that at point 'a' the timer is 10, and at exactly that time a new page needs to be rebooted. We also know that the time needed to reboot a new page is 1 unit of time, so we just let the timer jump ahead 1 unit before it starts to reboot the new page. When rebooting the new page is completed at point 'b', we can see that terminal A is still synchronized with terminal B which did not encounter rebooting for a new page.

Chapter 5

PEER TO PEER WIRELESS SYSTEM PERFORMANCE

In this chapter, the peer-to-peer wireless system performance is examined. The examination is divided into two groups. In the first group, the path loss, jamming and latency tests are performed. In the second group, the system synchronization is examined.

5.1 Throughput Vs Path Loss

For a wireless system, one of the important specifications is the range between transceivers. In general, this is hard to obtain because of the varieties of environments the system can be used in. So usually the throughput versus path loss test is taken as an alternative to evaluate an actual range.

5.1.1 Equipment Set-up

To perform the path loss test, the system shown in Figure 5.1 is used. The signal is transmitted via co-axial cable to provide a consistent channel. The two Hoppers used in this test are placed in two different rooms, about 15 meters apart. This set up reduces the possibility that the Hoppers could receive leaked signals from one another. Also to minimize the interference, this test is carried during the evening, when all the RF equipment nearby is turned off.

The attenuation of all the components used in the test is overlooked, so this test is a worst case scenario.

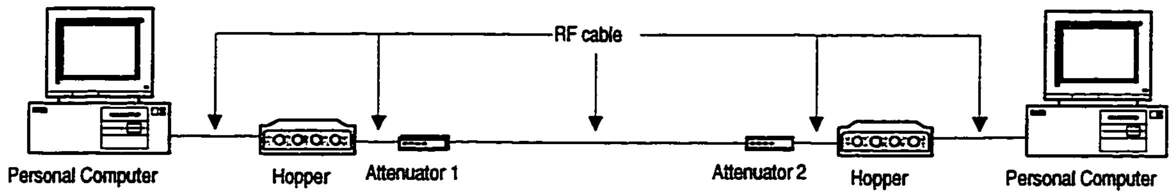


FIGURE 5.1 EXPERIMENTAL SETUP FOR PATH LOSS TEST

5.1.2 Fulfill the Throughput Vs Pass Loss Test

The RF cable used in this test can be thought of as an Additive White Gaussian Noise (AWGN) channel. Even though some of the real world characteristics are not included in the test, such as multi-path, the test can used to obtain a reasonable estimate of the range between units. This is due to the fact that path loss and range are directly related.

To gather data, two files are transmitted: one file is 716548 byte long, the other is 58880 byte long. The reason for using two different length files to accomplish the test is that when the path-loss reaches a certain level, there are many termination in transmitting a file, so transmitting a long file is very time consuming. All files are transmitted using the Zmodem protocol which is chosen mainly because it was used for testing the original FH code performance. So a fair comparison can be made. The Zmodem implementation in the *DOS*™ communications program *PcPlus*™, which is used for the experiments, reveals the throughput in characters per second. The test is performed with 15 word long packet, that is 30 characters per packet, which is equal to 240 bits per packet. The throughput Vs path loss result is shown in Figure 5.2. The data is collected five times, and then averaged up. The statistics are shown in Table 5.1.

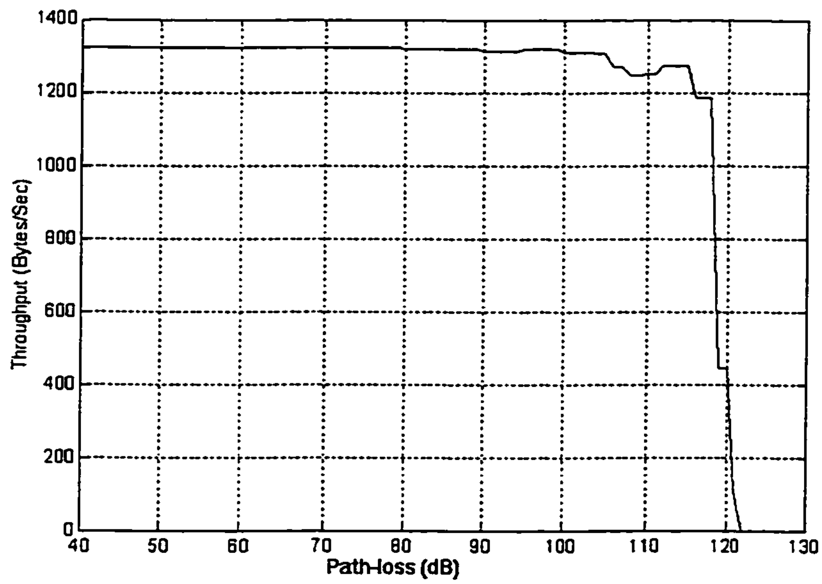


FIGURE 5.2 PATH LOSS MEASUREMENT FOR 240 BIT PACKET

The number of packets is calculated in the following way:

$$\text{Number of packets} = \frac{\text{Tx file size in byte}}{\text{24 information byte per packet}},$$

and the percentage of the bad CRC is obtained in this way:

$$\text{Percentage of bad CRC} = \frac{\text{num. of bad CRC}}{\text{num. of packets}} * 100\ \%$$

Path-loss in dB	Tx file size (byte)	Throughput (byte/s)	bad CRC	Num. of Packets	% of bad CRC
40	716,548	1,325	15	29,856	0.05024%
50	716,548	1,325	12	29,856	0.04019%
60	716,548	1,323	7	29,856	0.02344%
70	716,548	1,325	18	29,856	0.06028%
80	716,548	1,319	7	29,856	0.02344%
90	716,548	1,315	14	29,856	0.04688%
95	716,548	1,318	12	29,856	0.04019%
97	716,548	1,320	15	29,856	0.05024%
99	716,548	1,312	12	29,856	0.04019%
103	716,548	1,306	23	29,856	0.07074%
105	716,548	1,271	18	29,856	0.06028%
107	716,548	1,248	19	29,856	0.06363%
109	716,548	1,253	24	29,856	0.08038%
111	716,548	1,276	19	29,856	0.06363%
115	716,548	1,186	27	29,856	0.09043%
118	716,548	1,199	26	29,856	0.08708%
120	716,548	446	559	29,856	1.872%
121	58,880	111	235	2,453	9.58%
122+		Not		Available	

Table 5. 1 Statistics result of Figure 5.2

From Figure 5.2 we can see that the throughput does not change very much when the path loss is under 118 dB. After 118 dB the throughput drops dramatically when increasing the path loss. More specifically, in four dB the system goes from working well to complete failure. This result is expected due to the FM cut-off effect [10]. One thing to be mentioned is the bad-CRC item. This CRC check is built inside the Zmodem of the *DOS*™ program *PcPlus*™ instead of inside the peer-to-peer system code. There was a 16 bit CRC check function in the peer-to-peer system code, and it was disabled for two reasons :

1) The CRC check is not mandatory for this peer-to-peer wireless system.

2) The programmable memory is too tight to put in this 16 bit CRC check routine. There is only 4 bytes in the programmable memory left without the 16 bit CRC check routine.

From Figure 5.2, we also can notice that the throughput is around 1300 bytes/s when the path-loss is above 111 dB. Comparing this value with the theoretical calculation which is shown in chapter 3 Figure 3.5 to be around 1500 bytes/s, there are 200 bytes/s difference between the two. The reasons being as follows. The theoretical calculation is for a frequency hopping code without any interrupts, that is the system is hopping all the time, and transmitting data all the time. But for this peer-to-peer wireless system, both units have to go back to the signaling channel to fulfill its re-synchronization once every 56 hops, so in this peer-to-peer system data is not transmitted all the time.

Now let's compare this throughput with the throughput of the original FH code, which exists in the Hopper modem. The throughput of the original FH code is shown in Table 5.2 [10]. In the original FH code, with a 1024 bit packet length, the throughput is about 1900 bytes/s, while for a packet length of 512 bits, the throughput is about 1300 bytes/s, and for a packet length of 256 bits, the throughput is about 500 bytes/s. At first, it seems that the throughput of the original FH code is better, but in fact, it is not. The following calculation will prove it.

Packet length (bits)	Throughput (bytes/s)	Cut-off path loss (dB)
1024	1900	108
512	1300	110
256	500	110

Table 5. 2 Path loss measurements for frequency hopping from Jason's result

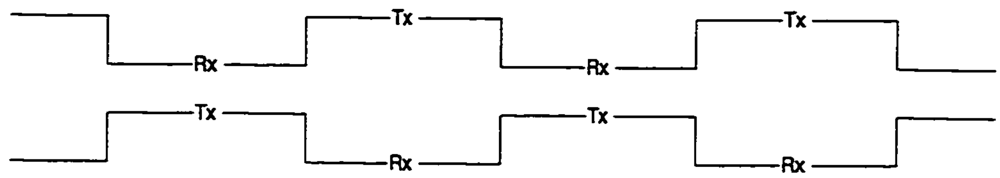
The theoretical throughput can be calculated using equation 3-2. For a 1024 bit packet, the ideal throughput is 2168 bytes/s. So the ratio of practical result to the ideal result is about 87.6%. For a 240 bit packet, as employed in this peer-to-peer wireless system, the ideal throughput is 1500 bytes/s. So the ratio of practical result to the ideal result is about 88.3%. One should realize that this comparison is made between a 240 bit packet system and a 1024 bit packet system, in which the latter system should have a better theoretical throughput. If we compare the throughput of a 240 bit packet used in this peer-to-peer wireless system with the throughput of a 256 bit packet used in the original FH code, the throughput of this peer-to-peer wireless system is much better. In fact, the absolute throughput, to transmit a 240 bit packet using this peer-to-peer system code is comparable to the throughput to transmit a 512 bit packet using the old FH code.

From the above comparison, this peer-to-peer system code has a high throughput while transmitting 240 bit packets compared to the original FH code with 256 bit packets.

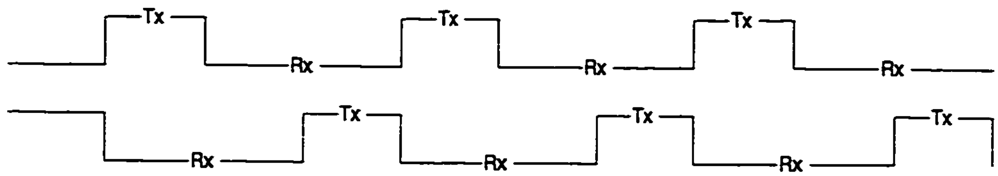
One possible explanation is that the original FH code was designed mainly for transmitting 1024 bit packets. In the original FH code, the transmitting time depended on the packet length, the receiving time depended on the receiving timer, and all the timer parameters for receiving data were set up based on a 1024 bit packet. Figure 5.3 can give us a better understanding of the above reason.

Case A in Figure 5.3 shows that both modems never stay in the same state when transmitting 1024 bit packets with 1024 bit packet receiving timer, in full duplex mode. In case B, with 1024 bit packet receiving timer, instead of transmitting 1024 bit packets, 512 bit packets were transmitted. From the Figure one can tell that in one third of the total active time, both modems stay in the same state, which is the receiving state. In case C, both modems stay in the same receiving state for 60 percent of the total active time. If we set the system efficiency for case A to be 100 percent, case B would have only a 66.7 percent efficiency, and case C only a 40 percent efficiency. The actual throughputs for the three cases is generally a 100 : 66.7 : 40 ratio.

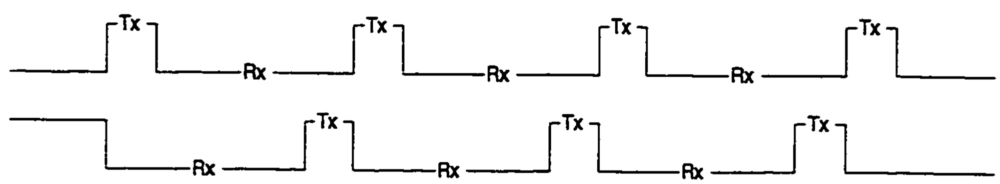
The receiving timer in this peer-to-peer wireless system is set up according to transmitting 240 bit packets, which is the same situation as in case A in Figure 5.3.



case A



case B



case C

FIGURE 5.3 TRANSMIT DIFFERENT LENGTH PACKET WITH SAME RECEIVING TIMER

case A)	Transmit 1024 bit packet with 1024 bit packet receiving timer
case B)	Transmit 512 bit packet with 1024 bit packet receiving timer
case C)	Transmit 256 bit packet with 1024 bit packet receiving timer

Comparing Figure 5.2 with Table 5.2, one can observe that the cut-off path loss for this peer-to-peer wireless system is about 122 dB, but the cut-off path loss for the old Hopper modem is about 110 dB. This 12 dB gain comes from hardware improvement of the Hopper modem, in fact the improvement is higher than 12 dB, if the path loss of the cables is taken into account.

After comparing the two systems with respect to throughput and cut-off path loss, let us take a look at the percentage of bad CRC. For this peer-to-peer wireless system the percentage of bad CRC is shown in Figure 5.4. This figure shows that under the cut-off path loss, the percentage of bad CRC is fairly low and generally growing with the increasing path loss. When the path loss reaches 120 dB, which is only 2 dB away from the cut-off path loss, the percentage of bad CRC increases sharply. The main reason is that under this amount of path loss, the receiving signal for both modems is very weak, so the signal is easily polluted by the channel noise, eventually increasing the percentage of bad CRC.

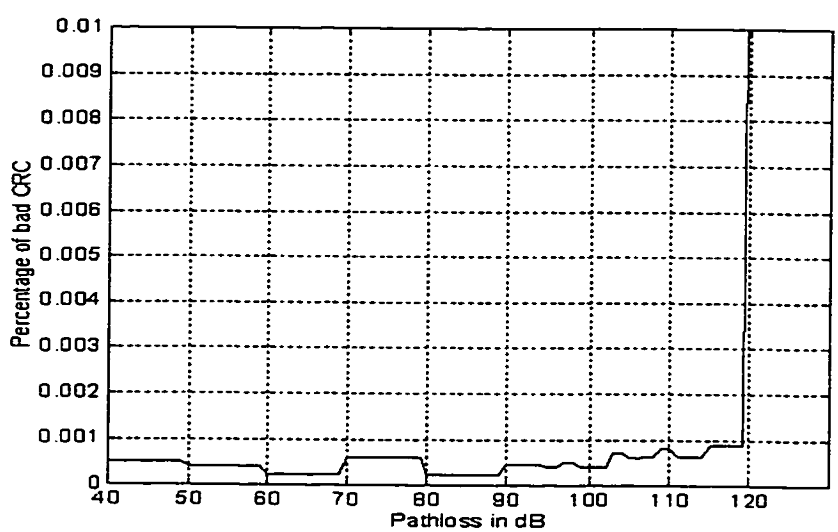


FIGURE 5.4 PERCENTAGE OF BAD CRC

From the above experiment one can conclude that the way of coding a system can affect the system throughput dramatically. However, the actual system practical range is mainly determined by the sensitivity of the modem receiver. Although the coding algorithm can improve the system throughput, it can hardly improve the transmission range.

Another question to ask is: is it good enough to transmit voice using a 10kbps bit rate? The answer is YES! Since the speech coder VSELP, which is used in IS-54, has a bit rate of 8 kbps before forward error control (FEC) coding. Furthermore speech coder CELP, which is used in IS-95, has a bit rate of 9.6 kbps before FEC coding [13]. Both vocoders require a bit rate smaller than this peer-to-peer system throughput. And for this peer-to-peer system, FEC is not necessary, since its actual bandwidth is over 20 Mhz. This is much wider than the bandwidth for IS-54 or IS-95.

5.2 Throughput Vs Narrow Band Jamming

As mentioned previously in Chapter 3, ideally, a frequency hopping system can overcome a narrow band jammer. So next, the peer-to-peer wireless system is evaluated at the presence of a narrow band jammer.

5.2.1 System setup

The system setup for testing throughput Vs narrow band jamming is shown in Figure 5.5. Two Hopper modems are put in two rooms about 15 meters apart, connected by an RF cable. The jammer is a single tone jammer, which is generated by a signal generator. The jammer is added to the signal via a splitter.

The test proceeded in the following manner: The signal strength where the splitter is connected in is measured. The variable attenuators are then adjusted in order to balance the signal strength since the RF cables are not of the same length. After adjustment, the attenuation between the two Hopper modems is found to be about 80 dB. During the test, the strength of the jammer is the only variable.

5.2.2 Test Result Examination

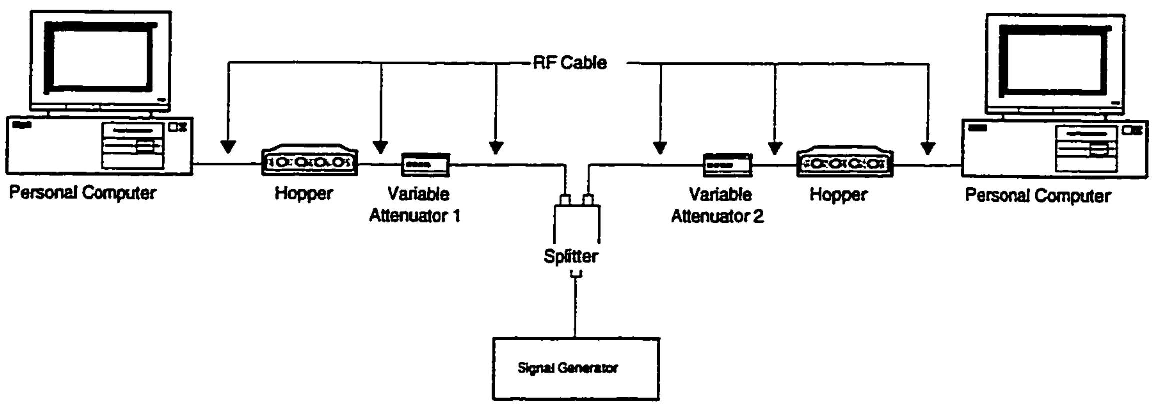


FIGURE 5.5 EXPERIMENTAL SETUP FOR NARROW BAND JAMMING TEST

The data collection is identical to the test for throughput Vs path-loss. The Z Modem of *PcPlus*^{TM} is used for collecting data. The result is shown in Figure 5.6. The data is shown in Table 5.3.

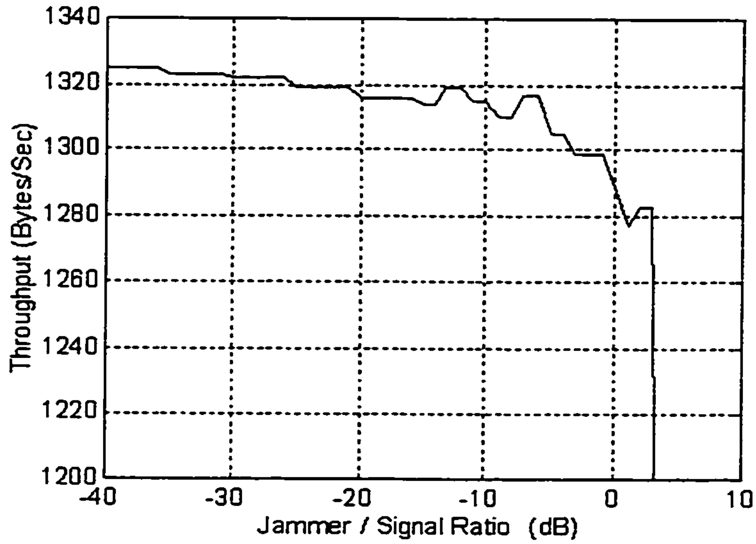


FIGURE 5.6 THROUGHPUT VARIATION AT PRESENT OF A NARROW BAND JAMMER

Jamming/Signal Ratio (dB)	File Size (Byte)	Throughput (Byte/s)
-40	716548	1325
-35	716548	1323
-30	716548	1322
-25	716548	1319
-20	716548	1316
-15	716548	1314
-13	716548	1319
-11	716548	1315
-9	716548	1310
-7	716548	1317
-5	716548	1305
-3	716548	1299
0	716548	1289
1	716548	1277
3	716548	1283
5	716548	1061

Table 5. 3 The data of throughput Vs narrow band jammer test

From Figure 5.6 one can note that when the jammer/signal ratio is less than -5 dB, the jammer does not affect the signal, but when the jammer/signal ratio becomes greater than -5 dB, the signal is affected. The average throughput from -40 dB to -5 dB is 1318 bytes/s, and the average throughput from -5 dB to 3 dB is 1290 bytes/s. So because of the influence of the single tone jammer, the throughput drops 28 bytes/s, that is:

$$\frac{(1318 - 1290)}{1318} = \frac{29}{1318} * 100\ \% = 2.2\ \%$$

Compared with the theoretical loss in throughput, which is (1 / 56) * 100 % = 1.78 %, these two results are quite similar. But when the jammer/signal ratio is higher than 5 dB, the throughput loss increases sharply, the reason is that the band pass filter used in Hopper modem is not very narrow, so when the strength of the jammer is much stronger than the signal, it does not only affect one frequency band, it affects the adjacent frequency bands as well.

The above result is identical to the result obtained from the original FH code. This is expected since the filter used in the Hopper modem has not been improved since the last test of the original FH code. It also proves that, the modification made on the original FH code based on the peer-to-peer wireless system requirements, does not change the feature of the original FH code, which is complies with the FCC part 15 rules.

5.3 Latency Test

System latency is another essential index for a wireless system, in addition to the throughput. Latency is the length of time required to send data from wireless terminal A to wireless terminal B. Latency is especially important for this peer-to-peer wireless system, because the system is designed for transmitting voice. Unlike data systems, voice systems usually cannot tolerate a large latency.

5.3.1 System Setup and the Result for Latency Test

To fulfill the latency test, two Hoppers and one oscilloscope were used. The system setup is shown in Figure 5.7. Because the latency is the time required to send data from wireless terminal A to wireless terminal B, the system latency can simply be tested by sending one packet, and measuring the time difference between sending the packet in terminal A and receiving the packet in terminal B. The test was processed in this manner: First let Hopper A connect with computer A, and Hopper B connect with computer B. Then use one channel of the oscilloscope to trace the Tx port of the RS-232 standard cable which connects Hopper A and computer A. Use another channel of the oscilloscope to trace the Rx port of another RS-232 standard cable which connects Hopper B and computer B. The oscilloscope is set at single trigger mode. 300 packets are sent one by one. The result is shown in Table 5.4. This latency is tolerable for voice transmission.

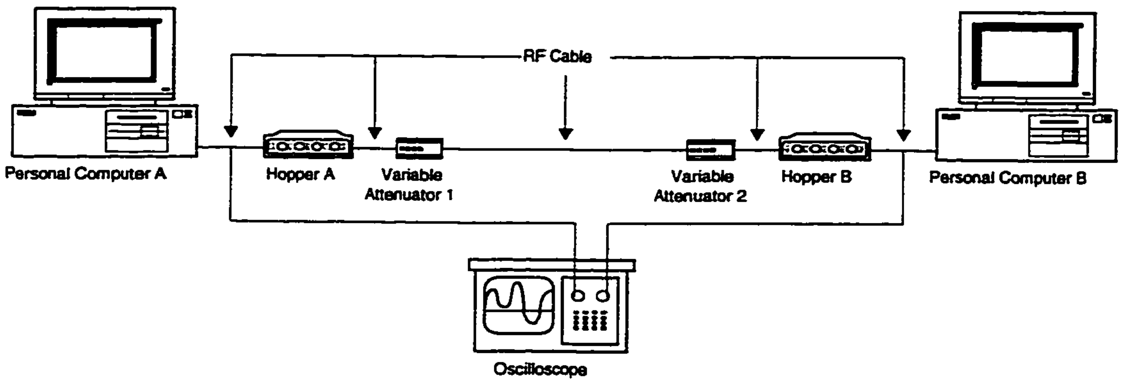


FIGURE 5.7 SYSTEM SET UP FOR LATENCY TEST

Longest latency (ms)	Shortest latency (ms)
30	8.5

Table 5. 4 Result of the latency test

5.4 Signaling Channel Performance Test

All the above tests were carried out in order to measure the performance of the message channel, which corresponds to data communications. But for a peer-to-peer wireless system, data communications between wireless modems is only one of two main aspects of the system. The other is system management, which is fulfilled using the signaling channel. To evaluate the performance of the signaling channel, four tests are performed: a call making test, an empty time slot pick up test, an inside calling pair re-synchronization test, and a time shift test between adjacent user pair.

5.4.1 Call Making Test

The basic idea for the call making test is to evaluate the robustness of the call initiation between user pairs with increasing path loss. The system setup for this test is the same as the system setup for testing the throughput Vs path loss.

The test was processed this way, first, modem A initiates a call, calling modem B, then modem B initiates a call, calling modem A. For each path loss ten call making attempts on each direction are made, and the statistics are shown in Table 5.5. One thing to notice from Table 5.5 is that after a 122 dB path loss, the connection between modems is totally broken, this result is consistent with the results obtained from the throughput Vs path loss test. One can also notice that, under the 122 dB path loss, all the call making attempts are 100% fulfilled. Compared with the result obtained from the throughput Vs path loss test, where the throughput under the 118 dB path loss is much different from the throughput above the 118 dB path loss, the result for call making test seems a little bit more reliable. This is due to the fact that the packet for call making is only 3 word long and timing for call making is not critical.

Path loss (dB)	Tried times A →B	Tried times B →A	Failed A →B	Failed B →A	Percentage of failed tries A →B	Percentage of failed tries B →A
40	10	10	0	0	0%	0%
50	10	10	0	0	0%	0%
60	10	10	0	0	0%	0%
70	10	10	0	0	0%	0%
80	10	10	0	0	0%	0%
90	10	10	0	0	0%	0%
100	10	10	0	0	0%	0%
103	10	10	0	0	0%	0%
105	10	10	0	0	0%	0%
107	10	10	0	0	0%	0%
109	10	10	0	0	0%	0%
111	10	10	0	0	0%	0%
115	10	10	0	0	0%	0%
117	10	10	0	0	0%	0%
118	10	10	0	0	0%	0%
119	10	10	0	0	0%	0%
120	10	10	0	0	0%	0%
121	10	10	0	0	0%	0%
122	10	10	10	10	100%	100%

Table 5. 5 Result of call making test

5.4.2 Empty Time Slot Pick-up Test

The purpose of this test is to evaluate the channel access scheme. The system setup for this test is shown in Figure 5.8. Four Hopper modems are used in this test, and all signals are transmitted through the air. Four Hopper modems are separated in four corners of a square, so every Hopper modem has the same influence from the other three Hopper modems, and the signal transmitted from each Hopper is 40 dB attenuated. The test is accomplished in this way. First a connection is built between modem A and modem B. Then a connection is built between modem C and Modem D. After modem C and modem D start transmitting and receiving over the message channel, modem A and modem B are disconnected, and then reconnected again. After modem A and modem B switch to the message channel and start transmitting and receiving, modem C and modem D are disconnected. The above procedure is repeated, until all time slots have been selected. Before this test, the code is modified to print out the selected time slot number on the computer screen, which is connected to the corresponding Hopper modem.

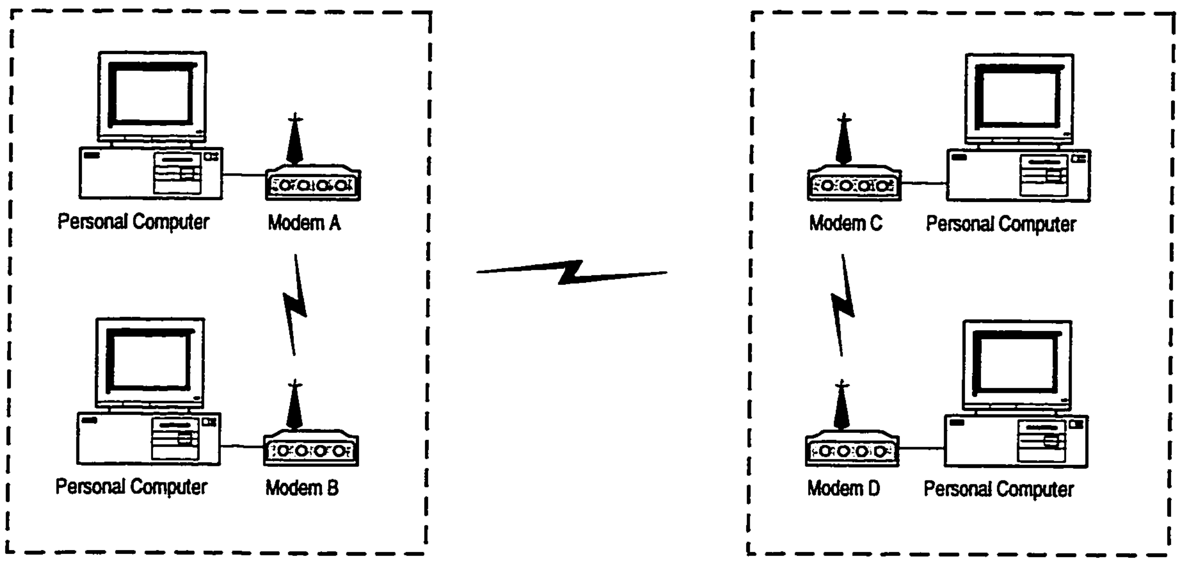


FIGURE 5.8 SYSTEM SET UP FOR EMPTY SLOT PICK-UP TEST

According to the channel access scheme, when the first connection is built up between modem A and modem B, there is no time slots employed in the signaling channel, so time slot #1 should be selected, and when the connection between modem C and modem D is built up, time slot #2 should be selected. Following the same rule, when the connection between modem A and modem B is broken, then re-built, time slot #3 should be selected for their connection. The result is shown in Table 5.6.

No	Time slot modem A-B selected	Time slot modem A-B occupied	Time slot modem C-D selected	Time slot modem C-D occupied
1		1	2	
2	3			2
3		3	4	
4	5			4
...	...	...	...	...
54	55			54
55		55	56	
56	57			56
57		57	1	

Table 5. 6 Statistic result for Time-slot pick up test

Table 5.6 shows that the result obtained from the time slot pick up test is consistent with the original design purpose.

This time slot pick up test is repeated in the case that one of the signaling channels is jammed, and the same result is obtained.

5.4.3 Inside calling pair re-synchronization test

The peer-to-peer wireless system implemented in this thesis is a digital system, and for a digital wireless system, synchronization between communicating modems is critical, because loss of synchronization between communicating modems means loss of communication. In other words, if the communication between communicating modems is preserved, they are in synchronization. So the evaluation of the synchronization between communicating modems can be completed by evaluating the throughput Vs time. The system setup for this test is shown in Figure 5.9, and all signals are transmitted through the air.

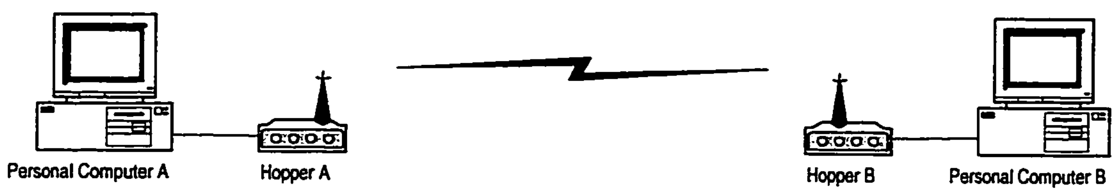


FIGURE 5.9 SYSTEM SETUP FOR INSIDE PAIR SYNCHRONIZATION TEST

In the test, one text file is transmitted 6 times, taking about 30 minutes every time, so the total transmitting time is around 3 hours. The test proceeded in the way described below:

First the connection between Hopper A and Hopper B is build. Then the text file is sent from Hopper A to Hopper B. When the transmission is over, Hopper B sends back the file to Hopper A. This procedure is repeated three times. The Zmodem implementation in the *DOS*™ communications program *PcPlus*™ is used for data collection, which is the real-time throughput. The statistic result is shown in Table 5.7 and the graphic result is shown in Figure 5.10.

Time (minute)	Throughput (bytes/second)	Time (minute)	Throughput (bytes/second)	Time (minute)	Throughput (bytes/second)
5	1331	60	1316	115	1322
10	1325	65	1320	120	1321
15	1322	70	1325	125	1325
20	1311	75	1323	130	1329
25	1324	80	1320	135	1327
30	1321	85	1319	140	1324
35	1318	90	1315	145	1325
40	1319	95	1323	150	1323
45	1326	100	1322	155	1325
50	1320	105	1325	160	1327
55	1322	110	1327	165	1327

Table 5. 7 Statistic Result of Throughput Vs Time Test

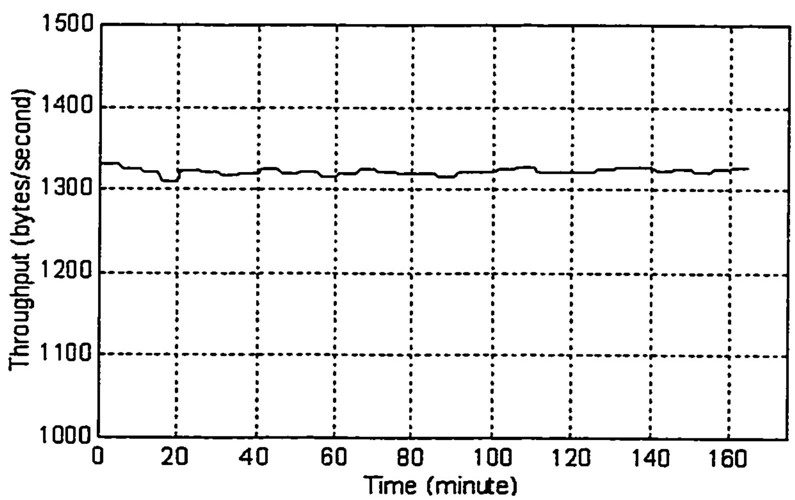


FIGURE 5.10 GRAPHIC RESULT OF THROUGHPUT VS TIME TEST

From the result shown in Table 5.7 and Figure 5.10 one can see that in a 165 minute period, the throughput stayed in the range of 1331 bytes/s to 1315 bytes/s, with no sudden drop or sudden gain in the throughput, which means there is no loss of synchronization during the testing period.

5.4.4 Throughput Vs Time between Transmitting Pair of Modem

For this peer-to-peer wireless system, all synchronization is accomplished over the signaling channel, including inside communication pair synchronization and synchronization between communicating pairs. The inside communication pair synchronization is updated every time the pair uses the signaling channel, which is every 912 ms. So the effect of time drift between the two modems' inner clock can be overlooked. But for the synchronization between two communicating pairs, the effect of time drift between the modems' inner clock has to be considered, because the synchronization between the two pairs has no chance to be updated since they have both switched to the message channel. In this case, the use of a more accurate inner clock modem can give a better system performance. This test is to evaluate the system performance with the use of the Hoppers.

5.4.4.1 System Setup

System setup for this test is shown in Figure 5.11. Four Hoppers were involved in this test. All signals were transmitted through the air. Hopper A and Hopper B was one communicating pair, while Hopper C and Hopper D was another communicating pair. In the test, Hopper A and Hopper B were put in one room, while Hopper C and Hopper D were put in another room. These two rooms are about 10 meters apart. By doing this the signal inside each communicating pair is stronger than the signal between the other communicating pair. The main reason for this is due to the fact that the band-pass filter used in the Hopper is not ideal, so there is a chance that one communicating pair affects the other pair if they are using adjacent frequency bands. But with a stronger inside communicating pair signal, the effect of the other communicating pair can be minimized.

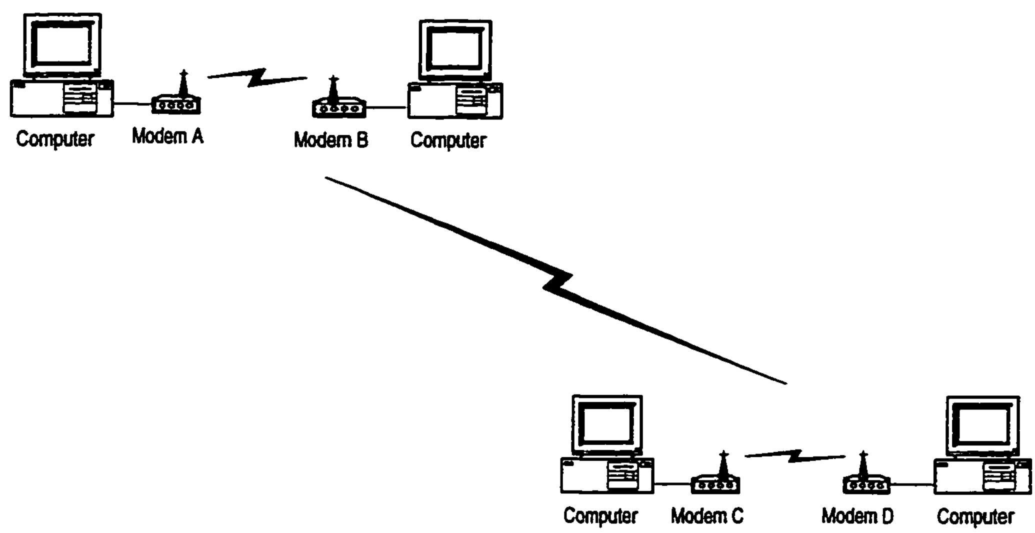


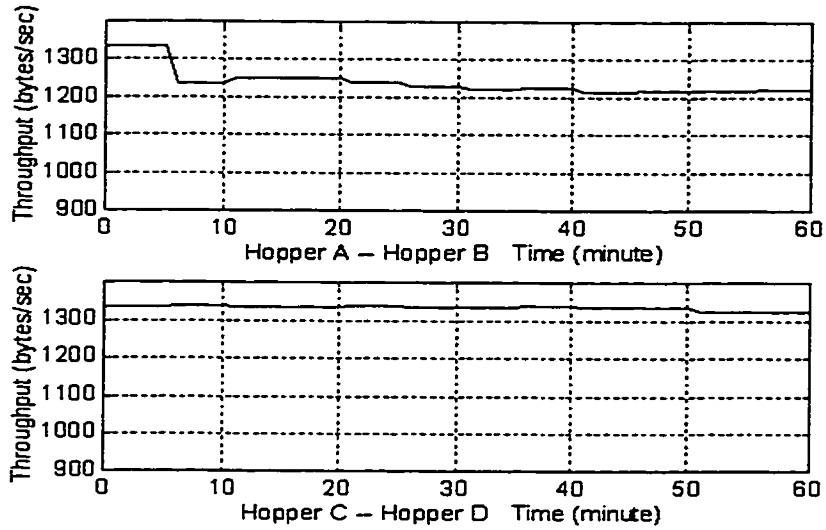
FIGURE 5.11 SYSTEM SET UP FOR BETWEEN PAIR SYNCHRONIZATION TEST

5.4.4.2 Test Result and Evaluation

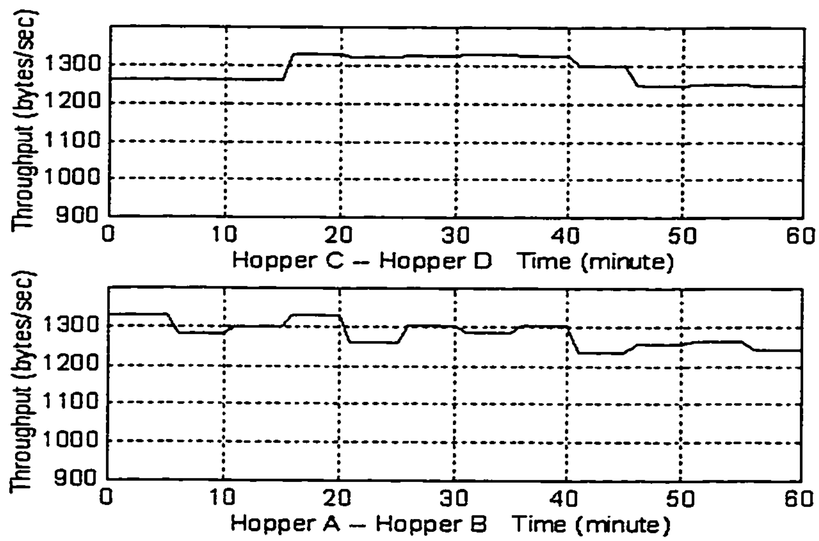
The test was done in two steps. In the first step, Hopper A and Hopper B started first, then Hopper C and Hopper D started. In the second step, Hopper C and Hopper D started first. So in this way, no matter wether the time shift is forward or backward, the effect on the system can be detected. The data was collected for about an hour using Z modem implemented in *DOS*™ communications program *PcPlus*™, and the result is shown in Table 5.8 and Figure 5.12.

Time (minute)	Throughput (bytes/sec) between Hopper A and Hopper B, start first, using time slot 1.	Throughput (bytes/sec) between Hopper C and Hopper D, start second, using time slot 2.
5	1334	1333
10	1235	1337
15	1250	1336
20	1250	1336
25	1238	1338
30	1227	1336
35	1222	1336
40	1225	1338
45	1214	1336
50	1217	1336
55	1220	1321
60	1221	1321
Time (minute)	Throughput (bytes/sec) between Hopper C and Hopper D, start first, using time slot 1.	Throughput (bytes/sec) between Hopper A and Hopper B, start second, using time slot 2.
5	1265	1331
10	1262	1283
15	1263	1302
20	1331	1331
25	1321	1260
30	1327	1304
35	1332	1284
40	1328	1304
45	1300	1234
50	1249	1255
55	1252	1264
60	1248	1243

Table 5. 8 Statistic result for running two pair of Hoppers in the same time



(A)



(B)

FIGURE 5.12 GRAPHIC RESULT FOR RUNNING TWO PAIR OF HOPPERS IN THE SAME TIME

(A) Hopper A and B using time slot 1, Hopper C and D using time slot 2.

(B) Hopper C and D using time slot 1, Hopper A and B using time slot 2.

Based on the statistic result shown in Table 5.8 and Figure 5.12, generally one can conclude that the time drift between Hopper pairs is not fatal to this peer-to-peer wireless system. We can notice that during the whole test period, all four Hoppers' throughput is above 1200 bytes/sec, and also there is no sudden drop in throughput for both pair of Hoppers, no matter which pair started first. All of this indicates that during the testing period, which is one hour, the two pairs of Hoppers are in synchronization, i.e. one can conclude that there was no significant time drift between the two pairs of Hoppers.

Chapter 6

CONCLUSIONS

6.1 Concluding Remarks

The conventional method to provide telephone service in rural areas is expensive. This is because the telephone company must install and maintain the local switching and transmission facilities. The use of a peer-to-peer wireless system in conjunction with a mobile satellite system presents a low cost alternative for serving rural areas. A simple single hop peer-to-peer wireless system is implemented and examined in this thesis. In this implementation, the wireless modem Hopper made by Wi-LAN Inc. is used as the wireless terminal.

The implementation of a peer-to-peer wireless system proposed in this thesis is mainly focused on software development. The software can be grouped into two main parts: the signaling channel and the message channel. The message channel employed some of the original frequency hopping software which handled the basic frequency hopping functionality. In addition, new software has been added to fulfill the message channel operation and to manage the connection to the signaling channel. In the signaling channel, the channel access scheme is accomplished, also the system synchronization is performed. Moreover, the use of two frequency bands for the signaling channel made the peer-to-peer wireless system more reliable in the presence of a jammer.

In order to evaluate the software, 7 tests were carried out. From the results, it was clear that all the system design goals were met: the throughput was high enough for voice transmission, the synchronization was steady enough for telephone service, and the system performance complied with the RSS-210 or FCC part 15 limitations for unlicensed ISM bands.

6.2 Future Work

There are a number of potential improvements to the system design which could increase the system performance and its reliability. As discussed before, this peer-to-peer wireless system uses spread spectrum technology: TDMA in the signaling channel, FH in the message channel. According to the system design, once a user terminal hooks up with its communicating party and moves to the message channel, at any time, one frequency band can only be occupied by one pair of communicating user. In other words, once a user pair starts its communication, it has no information exchange with other user pairs. So the potential problem here is, after a long communication time, the synchronization between user pairs, which takes place in the beginning of a call over the signaling channel, could be lost. This is caused by clock drifting inside each modem.

The above problem can be solved by changing the design. First, a CRC check routine is necessary. This CRC check routine is not for re-transmitting the missed packets, but for detecting the quality of the channel. If the bad CRC packets are received at a high rate, it means that the channel quality is not very good. Since the peer-to-peer wireless system is designed for operating in rural areas, if suddenly the channel becomes contaminated, there is a greater possibility that the reason is co-channel interference generated by losing synchronization between user pairs. Therefore, the rate of bad CRC can be an indicator to determine whether the synchronization between user pairs is lost or not. The basic logic for realizing this idea is shown in Figure 6.1.

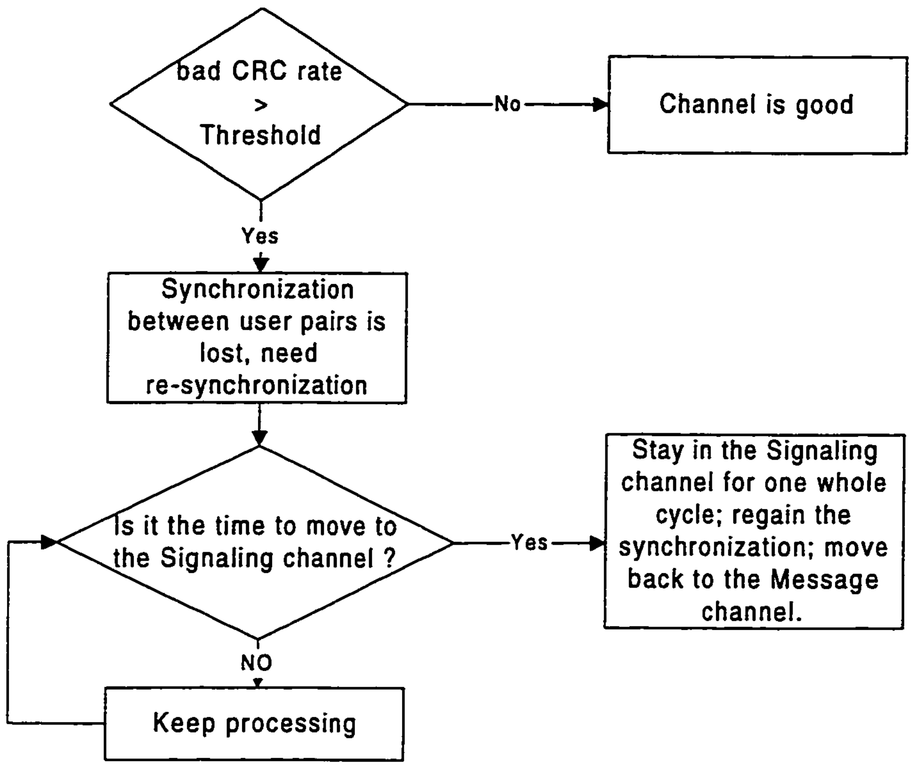


FIGURE 6.1 BASIC LOGIC FOR REGAINING THE SYNCHRONIZATION BETWEEN USER PAIR

There is a sacrifice have to make in order to fulfill this modification. One can notice from Figure 6.1 that the modem has to stay in the signaling channel for a whole cycle in order to finish regaining synchronization. But with a faster processor, this sacrifice is not very critical.

Currently, Wi-LAN Inc. is building a prototype for a new Hopper with the ADSP 2181 processor, which has a 16K programmable memory and a 32 Mhz clock. This new

Hopper would be an ideal modem to use for further investigating the peer-to-peer wireless system.

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APPENDIX A HARDWARE - WIRELESS MODEMS

8.1 Introduction

The original software support provided a "dumb" link between two nodes. Then the functionality of the full-duplex support was added by Jason Conrad Sokolosky. This thesis was to develop software for the microprocessor controller to support the peer-to-peer functionality. The modem hardware is described below.

8.2 Unit Description

The Hopper modem was designed at Wi-LAN Inc.. The design uses two printed circuit boards which are interconnected. One board is for the digital section and the other is for the RF (radio frequency) section. A block diagram showing the flow of data between each section is shown in Figure A.2.1.

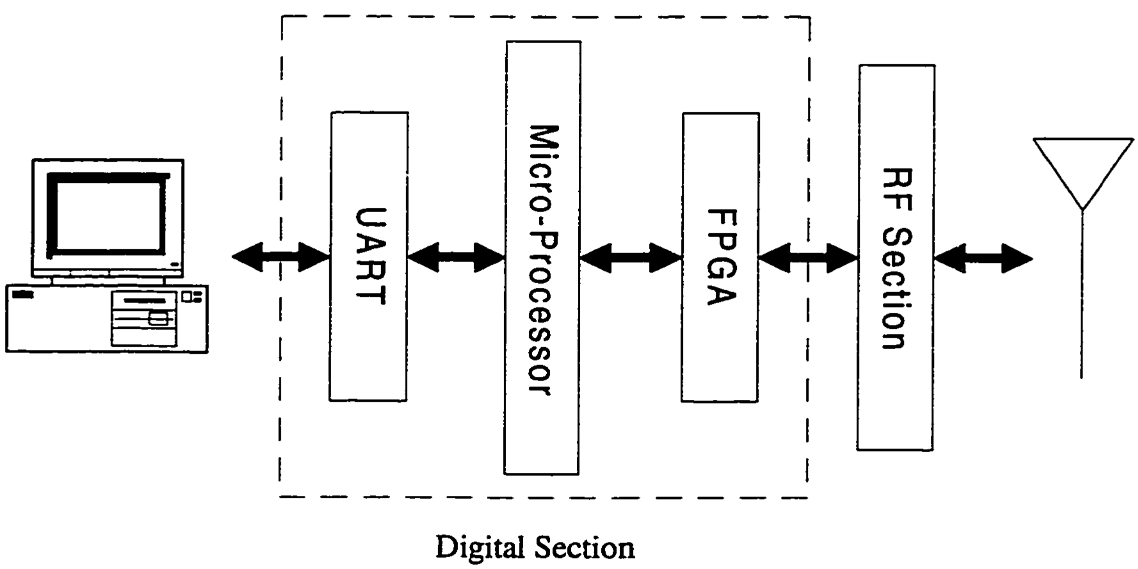


FIGURE A.2.1 BLOCK DIAGRAM OF HOPPER MODEM

8.2.1 Digital Section

The Digital section receives and sends data to the Data Terminal Emulator (DTE) and is also responsible for controlling the modem's operation. The purpose of each section is described below:

1. UART

The UART (universal asynchronous receiver/transmitter) converts serial data from the DTE to parallel data that is used by the microprocessor and vice versa. The UART is a one character double-buffered device. All aspects of the serial bit synchronization are performed by the UART.

2. Microprocessor

The microprocessor controls the operation of the modem. It receives data from the UART and passes it along to the Field Programmable Gate Array (FPGA). It controls the frequency synchronization for the frequency hopping operation. As well as the user interface.

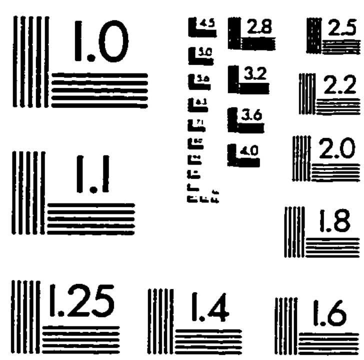
3. FPGA

The FPGA (field programmable gate array) performs baseband modulation PN code generation, bit post detection, and digital integration.

8.2.2 RF section

The RF (radio frequency) section performs down and up conversion from the base-band to the final carrier frequency. It also contains a power amplifier for the amplification of the signal to a transmit power of 1 Watt. The RF section also performs the FM demodulation for the received signal.

IMAGE EVALUATION TEST TARGET (QA-3)



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