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A New Adaptive EDCA Approach to QoS of Wireless Communications

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ABSTRACT

Under the background of intelligent transportation application, QoS for various services is different in wireless communication. Based on the MAC layer protocol, this paper analyzes the QoS in IEEE 802.11 MAC protocol framework, and proposes a new design of a Differentiation Enhanced Adaptive EDCA (enhanced distribution channel access) approach. The proposed approach adjusts the window zooming dynamically according to the collision rate in sending data frames, makes random offset, and further distinguishes the competition parameters of the data frames that have the same priority, so as to reduce the conflict among the data frames, and improve the channel utilization. Experiments with different service cases were conducted. The simulation results show that: comparing with the conventional EDCA method, the proposed approach can ensure that high priority services are sent with priority, and the overall QoS is highly improved.

Keywords: Intelligent Transportation Systems, Wireless Communication, Mac Layer Protocol, Enhanced Distribution Channel Access

1. Introduction

In the intelligent transportation systems (ITS), there are two wireless communication modes: car to car and car to road. They are the foundation of providing information exchange and service. Wireless communication improves the service active safety technology further. In early days, the main research topics of the intelligent transportation wireless communication technology are vehicles location and distance measurement between vehicles etc [1,2] in collaborative driving application. In recent years, more and more researches focus on transmission of service information and mass data flows such as audio and video and so on. In researches and applications, the modern wireless communication technologies are adopted. For example, in Fleetnet-road network project (see http://www.et2.tu-harburg.de/fleetnet, Fleetnet) and CarTalk 2000 project (see http://www.cartalk2000.net, Cartalk 2000 web site), wireless LAN and cellular networks are employed.

In different intelligent transportation application scenarios, each service has stringent requirements for QoS of wireless communications. For example, in the active safety application scenarios, the delay of sending the security message requires to reach ms level. While in emergency voice communication application, packet loss rate of voice data stream can’t exceed 1%. QoS can be described in different aspects in the communication, such as the network throughput and delays. At present, the method of improving QoS generally uses separate-layer or cross-layer. Cross-layer program has the advantage of improving the efficiency of the protocol stack. But its network protocol is complex and is difficult to maintain. Therefore, in the application background, the main issue is separate-layer QoS program in recent days.

The MAC layer protocol controls node accessing to wireless channel and message transmission occupying wireless medium, so as to provide reliable data transfer for upper layer, and guarantee the overall performance of the network. Therefore, MAC layer is the focus of communication protocol in intelligent transportation application. For example, in the EDCA (enhanced distribution channel access) mechanism of the IEEE 802.11p [3] standard, the service data accesses to channel by competitive way. And further distinction of priority-based QoS through distributing competitive parameters is achieved. CarTalk 2000 Project proposed the mechanism of Ad Hoc MAC [4]. It adopts dynamic TDMA mechanism, and each service accesses to the channel by making an appointment. The article [5] proposed the D-MAC mechanism, which reduces transmission conflict by using the directional
antenna. Kaichi F. et al. proposed the VRCP mechanism [6], which can work in Central control or ad hoc network. And it can dynamically switch according to whether the node is in the coverage area of the central controller.

However, the existing programs for intelligent transportation cannot fully meet the requirements in real-world applications. For network throughput and delay, this paper lays special stress on analyzing the MAC layer, and emphatically studies the performance of QoS of EDCA mechanism in intelligent transportation and designs a new mechanism to improve it. Then simulate the algorithm and analyze the experiment results. This paper is structured as follows: Section 2 introduces the principle of EDCA mechanism and the support to QoS. Section 3 designs and describes a differentiation enhanced adaptive EDCA mechanism. Section 4 simulates on the OPNET platform and analyzes the results. Section 5 is the summary.

2. Related Work

The MAC layer provides three mechanisms and protocols in the intelligent transportation with wireless communication, which are described as follows:

1) CSMA/CD-based Protocol, such as the 802.11p. CSMA/CD uses the mechanism of listen-before-talk (LBT). Each node in the system senses the channel before sending the packet. If the channel is idle, send data. If the channel is busy, continue to sense until it becomes idle, and then use binary backoff mechanism to reduce the probability of collision.

2) TDMA-based protocol, such as RR-ALOHA. Protocol, a channel is divided into several time slots (called BASIC CHANNEL, BC). Besides the payload of each channel, FI (Frame Information) is also transmitted. FI includes the idle state of each slot. Any terminal can ask for the establishment of BC in idle time slot. And the channel is remained by the terminal that occupies it before in busy time slot.

3) Directional antenna-based mechanism, such as D-MAC protocol. It is developed from IEEE 802.11. The protocol requires each node to know the location of themselves and their neighbors. Source node first launches a RTS/CTS/ACK handshake mechanism before sending data packets. RTS or Omn is sent directionally according to the situation of the transmission around. The directional antenna will be blocked when receiving the RTS or CTS, and the source nodes delay sending packets according to the received information. The conflict of transmission can be reduced by using directional antennas and channel reusing rate can be increased. But the directional antenna systems are expensive, and it’s hard to maintain in the actual implementation.

TDMA-based MAC protocols require strict time synchronization and its slot is fixed. When a node sends no data, its slots cannot be occupied by other nodes that need to send data. So it can’t use channel resources effectively. For example, in Japan ALOHA is mainly used in MAC layer protocol of Dedicated Short Range Communication (DSRC) [7]. It is designed for the electronic toll collection and other applications, and it can’t meet the demands in many service application scenarios and services.

CSMA-based protocol does not require time synchronization. It uses carrier sense method to judge whether the channel is idle before sending data. If the channel is busy, it adopts backoff mechanism to make the node return to sense, and tries again. So the conflict probability is small and the channel utilization is effective. CSMA-based protocol is more suitable for the rapid movement of vehicles. Therefore, this paper focuses on the MAC protocol based on CSMA/CD mechanism.

3. Conventional EDCA Mechanisms and Support of QoS

QoS refers to a series of performance indexes that need to be satisfied when the network transmits some data flows. QoS guarantee reflects the capabilities that network devices or protocols ensure data transmission and meet specific service. It is described by the throughput, delay, delay jitter and packet loss rate and other parameters. In order to support the application of the Intelligent Transportation System, IEEE 802.11 Working Group established wireless access standard for the vehicle environment, named IEEE 802.11p. In this standard, the MAC layer adopted the EDCA mechanism based on CSMA/CD.

The working principle of EDCA mechanism is shown in Figure 1. EDCA provides the priority-based QoS function. This mechanism defines the access category (AC). Reference to 802.1D standard, it casts 8 priorities into 4 ACs, each of which has a separate backoff parameter. Hence, there are 4 separate competition entities in each node. The distinction between priorities is obtained by setting the values of different parameters such as Arbitration Inter Frame Space (AIFS), minimum contention window (CWmin) and maximum contention window (CWmax) are used for different ACs. For each AC[i] (i = 1,2,3), the backoff mechanisms are the same as DCF. To achieve the distinction between priorities, for 0 ≤ i < j ≤ 3, demand $CW_{\text{min[i]}} \geq CW_{\text{min[j]}}$, $CW_{\text{max[i]}} \geq CW_{\text{max[j]}}$ and $AIFS[i] \geq AIFS[j]$. As can be seen, higher-priorities have smaller CWmin, CWmax, AIFS parameters. They can enter the backoff process and finish it firstly in the competition. Thus they have a top priority to access to channel. If several entities finish backoff at the same time, the higher-priority entity will be sent as a priority.

Furthermore, the EDCA mechanism introduces the concept of transmission opportunity (TXOP). After AC competing for channels and having sent a frame successfully,
if there are still data frames need to be sent, the channel can be used all the time in the TXOP time limit. When we reach the TXOP time limit or there are no data frames to send, TXOP is over and starts a new competition.

The EDCA mechanism ensures the higher-priority service to be sent as a priority by the means of distinguishing priorities. However, in high load, for middle and low-priority service flows, the performance will become rather poor. For EDCA mechanism, parameters configuration is the key of effectively using channel resources. And the reasonable parameter adjustment algorithm plays an important role in expressing the system performance.

4. New Design of Differentiation Enhanced Adaptive EDCA

On one hand, the EDCA mechanism supports to send the high priority service firstly, but backoff parameters cannot be adjusted dynamically with the change of network situation. Thus the network performance can’t be utilized effectively. On the other hand, EDCA can distinguish competition parameters from different ACs that belong to the same node. It leads to data frames from the same AC of multiple sites choosing the same competition parameters, which tries to send at the same time, especially for the high priority AC with the same AIFS, its scope of the CW is small, thus the probability of choosing the same parameters is very large. This phenomenon is even more evident with the nodes increasing. For the above two aspects, this paper will design a Differentiation Enhanced Adaptive EDCA (DE-AEDCA). The mechanism can adapt to different network congestions by calculating the collision rate of the sending-data frames, dynamically adjusting the window zooming, maintaining an appropriate contention window range. While the random offset achieves the further distinction of data frames competition parameters between the same priority, and reduces their conflict probability. Consequently, it cuts down the numbers of idle time slots caused by conflict, and improves the channel utilization.

4.1 Selection of Determining the Network Status Parameters

In the successive, same interval of time period \( T_n \), we record the collision rate of different ACs periodic. We assume the number of data frame conflicts to be \( \text{Num}_{\text{collision}}[i] \) in one \( T_n \), using parameter \( P_n[i] \) to show the conflicts in the current time period \( T_n \). The range of the \( P_n[i] \) value is 0 to 1. The formula is shown in (1).

\[
P_n[i] = \frac{\text{Num}_{\text{collision}}[i]}{\text{Num}_{\text{total}}[i]} \quad (n = 1, 2, 3, \cdots)
\] (1)

Collision rate changes in each time period \( T_n \) dynamically, which \( T_n \) is an integer multiple of slot duration. If \( T_n \) is too large, it cannot reflect real-time network status. On the contrary, if \( T_n \) is too small, the computation will increase. When OFDM technology is adopted in
the physical layer, one time slot length is 9μs, data interval is 0.01s in this article. Then we take the time data sources to produce 9 data frames as a statistical cycle, which means \( T_n = 90000\mu s \). In order to suppress judging errors of network state due to the burst of data frame, we deal with the current \( P_s[i] \) with weighted average method of multiple time periods collision rate. Weights after treatment is \( P_{\text{weight}}^w[i] \), which is computed as (2):

\[
\begin{align*}
P_{\text{weight}}^w[i] &= k \times P_s[i] + (1 - k) \times P_{\text{weight}}^w[i] \quad (n \geq 2) \\
&= P_s[i] \quad (n = 1)
\end{align*}
\]

where \( k(0 \leq k \leq 1) \) is the smoothness factor. It is changed by the emergent state of network, for instance, estimated by comparing \( \text{Num}_{\text{weight}}[i] \) with \( \text{Num}_{\text{total}}[i] \). If the nodes generate stable value of data frames in each time period, \( k \) can take a higher value. In this paper, it is more stable to produce data of source node. According to experimental results, \( k \) takes 0.8 in the simulation of this paper.

4.2 Conflict Process of Data Frame

The algorithm for conflict process of data frame is given as:

\[
\begin{align*}
CW_{\text{new}}[i] &= \min\{|CW_{\text{max}}[i], CW_{\text{old}}[i] \times (1 + 2^\varphi)|
\varphi = P_{\text{weight}}^w[i], \\
(i = 0,1,2,3)
\end{align*}
\]

where \( \varphi \) is the current weighted value of collision rate, \( \varphi \in [0,1] \). The scope of \( 1+2^\varphi \) is \([2,3]\), thus the competition window of \( \text{AC}[i] \) can progressively increase in different multiples depending on collision rate of its data frame. The larger the value of \( P_{\text{weight}}^w[i] \) is, the greater the progressively increasing multiples is, and vice versa, thus competition window can adaptively adjust depending on network status indication.

In this algorithm, each access category \( P_{\text{weight}}^w[i] \) is statistical respectively, because the high-priority data has higher right to access to channel than that of the low-priority data, so \( P_{\text{weight}}^w[i] \leq P_{\text{weight}}^w[j] \) (0 \( \leq i \leq j \leq 3 \)), it means that the higher the priority is, the smaller collision rate is. Thus the progressively increasing multiples of their competition window are smaller. Therefore, this algorithm still strictly ensure the priority relations among all categories of data, it is critical for the application of intelligent transportation system, because it must ensure that high-priority data (such as emergency short message data) to send first.

4.3 Judgement of Sending the Data Frames Successfully

The equations are an exception to the prescribed specifications of this template. You will need to determine whether or not your equation should be typed using either the Times New Roman or the Symbol font (please no other font). To create multileveled equations, it may be necessary to treat the equation as a graphic and insert it into the text after your paper is styled.

First, we define a variable \( \gamma_i \), whose value shows the current consecutively sending number of data frame of node \( \text{AC}[i] \). Its value adds 1 when it successfully sends a data frame, and the value is cleared when a conflict occurred.

Secondly, we subtract the value \( CW_{\text{min}}[i] \times (1 - P_{\text{weight}}^w[i]) \) of the contention window when \( \gamma_i = 1 \). Subtract \( 2 \times CW_{\text{min}}[i] \times (1 - P_{\text{weight}}^w[i]) \) of the contention window when \( \gamma_i = 2 \), and so on. That means the decreasing value increases when \( \gamma_i \) increases every time. Every time, the increasing value is \( \Delta \):

\[
\Delta = CW_{\text{min}}[i] \times (1 - P_{\text{weight}}^w[i])
\]

Finally, when the value of contention window reduced to half of the initial value, it achieves the congestion avoidance phase, in order to avoid excessive competition which results from contention window decreasing too fast. Hereafter, when \( \gamma_i \) keeps on increasing, it resumes taking \( CW_{\text{min}}[i] \times (1 - P_{\text{weight}}^w[i]) \) as original decreasing value and beginning to accelerate declining as before-mentioned steps until declining to \( CW_{\text{min}}[i] \).

The basic idea of this algorithm is similar to slow-start decreasing algorithm SSDS (Slow-start Decrease Scheme. It increases the adaptive capacity of the decreasing scheme, and it adds congestion control to avoid further competition which results from contention window decreasing too fast. The range of \( (1 - P_{\text{weight}}^w[i]) \) is \([0,1]\) in the algorithm, and the range of \( \Delta \) is \([0,CW_{\text{min}}[i]]\). The larger \( P_{\text{weight}}^w[i] \) is, the more severe the current network congestion is, and the smaller \( \Delta \) is, the slower the declining pace of contention window is, and vice versa. The algorithm can judge the current network state and adjust the contention window size dynamically, and maintain a good state of congestion avoidance.

The algorithm can be described as (7):

\[
\begin{align*}
CW_{\gamma_1}[i] &= \max\{CW_{\text{min}}[i], CW_{\gamma_1}[i] - \Delta \times \gamma_1\}, \quad (\gamma_1 \leq m) \\
CW_{\gamma_m}[i] &= \max\{CW_{\text{min}}[i], CW_{\gamma_m}[i] - \Delta \times (\gamma_m - m)\}, \quad (\gamma_m > m)
\end{align*}
\]

where \( \gamma_m \) is the number of sending data frame successfully after the congestion avoidance phase, and \( CW_{\gamma_1}[i] \) is the value of the contention window which is calculated by the algorithm. For example, \( CW_{\gamma_1}[i] \) shows the initial
the value of network congestion condition is, the larger changes with the state of the whole network. The larger competition parameter value by designing an added random offset to ease the probability of the same AC choosing different avoidance. So the basic idea of this algorithm is to increment another leads to the drastic decline of the whole network.

Another improvement in this paper is the differentiation enhancement. It is designed to resolve the problem that the same AC data frames of different nodes compete with each other, which leads to the decline of the whole network’s performance when EDCA is under the high load.

Usually, the waiting time before trying to transfer each AC data frame equals to the sum of the avoidance time and the delay time. The avoidance time is random generated between 0 and \( CW[i] \). The delay time which is decided by \( AIFS[i] \) is the time necessary to wait before avoidance. So the basic idea of this algorithm is to increase the probability of the same AC choosing different competition parameter value by designing an added random offset to \( AIFS[i] \) and \( CW[i] \), on this condition, it’s easier to distinguishing the competition parameters of the same AC data frames of different nodes. The concrete processes of the algorithm are as follows:

- The CW-based offset

The biggest offset \( CW_{\text{max}}[i] \) is computed by

\[
CW_{\text{max}}[i] = \frac{1}{2} CW[i],
\]

\[
m \in \gamma_i
\]

\[
\text{if } CW_{\text{new}}[i] < \frac{1}{2} CW[i],
\]

4.4 Discrimination of Same Access Categories

Another improvement in this paper is the differentiation enhancement. It is designed to resolve the problem that the same AC data frames of different nodes compete with each other, which leads to the drastic decline of the whole network.

Usually, the waiting time before trying to transfer each AC data frame equals to the sum of the avoidance time and the delay time. The avoidance time is random generated between 0 and \( CW[i] \). The delay time which is decided by \( AIFS[i] \) is the time necessary to wait before avoidance. So the basic idea of this algorithm is to increase the probability of the same AC choosing different competition parameter value by designing an added random offset to \( AIFS[i] \) and \( CW[i] \), on this condition, it’s easier to distinguishing the competition parameters of the same AC data frames of different nodes. The concrete processes of the algorithm are as follows:

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\]

\[
m \in \gamma_i
\]

\[
\text{if } CW_{\text{new}}[i] < \frac{1}{2} CW[i],
\]

5. Simulation Studies and Analysis

The simulation experiments are conducted on OPNET platform. The OPNET uses process domain model, node domain model and network domain model to react the relevant features of the networks comprehensively. Finite-state machines, OPNET core function, and standard C and C++ can make up the process model to describe bottom layer algorithm and the pipe stage model. OPNET Modeler uses discrete-time driven simulation mechanism that can get very detailed simulation results. It greatly improves the efficiency of the simulation.

5.1 Design of Simulation Cases

The simulation in this paper only considers the impact of MAC layer protocol on QoS requirements of multiple intelligent transportation service. Meanwhile, to cast the impact of routing layer on the wireless communication performance, this paper only considers one-hop communication without considering the impact on wireless communication of barrier and relative speed. The results in [11] shows that in one-hop range, the impact on throughput and delay of distance and speed of the cars is not obvious, but it has much relation with the number of the vehicles communicating at the same time. This paper mainly simulates different cases by setting different numbers of vehicles. In the simulation, the service scenario can be divided into high density case, middle density case and low density case by the number of vehicles communicating at the same time.

5.2 Parameter Settings

The services in Intelligent transportation systems can be divided into four categories: active safety, such as warning of dangerous road, collision warning; public service, such as emergency vehicles (fire engine, ambulance) which can pass prior on congested road, emergency speech communications; driving assistance service, such as roads navigation, video monitoring; business and entertainment.
Figure 2. The flow chart of the differentiation enhanced adaptive EDCA

service, such as Internet access, electronic map download.

In view of the above service categories, application-layer sets four services in the simulation: emergency short message, speech, video, background, and the priority are from high to low. The service parameters are in Table 1. This paper only considers the MAC layer’s effects for QoS, neglect routing layer and the above ones. The high layer is simulated by sending source modules.

Physical layer uses OFDM modulation technique, its bandwidth is 20 MHz, transmission rate is 24 Mbits/s, SIFS is 16 us, DIFS is 34 us, and a SlotTime is 9 us. It distinguishes the different cases by setting different numbers of vehicle nodes. 32 nodes are set in high-density vehicle case; 20 nodes are set in middle-density vehicle case; and 12 nodes are set in low-density vehicle case. In each case every vehicle node carries only one service, and in different cases the numbers of vehicle nodes which carry different services are same. The parameters of

<table>
<thead>
<tr>
<th>service</th>
<th>Inter-packet Gap(s)</th>
<th>Packet size</th>
<th>Transmission rate (Kbits/s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>emergence</td>
<td>0.01</td>
<td>12</td>
<td>9.6</td>
</tr>
<tr>
<td>speech</td>
<td>0.01</td>
<td>80</td>
<td>64</td>
</tr>
<tr>
<td>video</td>
<td>0.01</td>
<td>1250</td>
<td>1000</td>
</tr>
<tr>
<td>background</td>
<td>0.01</td>
<td>1500</td>
<td>1200</td>
</tr>
</tbody>
</table>

Table 1. Parameter list
The MAC layer parameters are shown in Table 2. AC[3], AC[2], AC[1], AC[0] are emergency short message, speech, video, background services.

The AIFSN offset range of the mechanism proposed in this paper is in Table 3.

### 5.3 Simulation Results

In the simulation, the performance of network throughput and delay are analyzed. Network throughput refers to the summation of data packets received and successfully transmitted to the application layer by MAC layer in all sites. Delay means different packets start when sending node in the application layer generates and end when destination node receives those packets, including media access delay and transmission delay (ms).

Figures 3, 4 and 5 are the simulation results with high density, medium density, and low density, respectively. The corresponding values of the four data streams in EDCA mechanism are described as the curves with round, left triangle, upper triangle and square symbols, according to their descending priority. The corresponding values of the four data streams in DE-AEDCA mechanism are described as the curves with the plus sign, lower triangle, right triangle and diamond symbols, by their priority from high to low. In figures, the x-axis represents the simulation time (sec.), the y-axis represents the delay time (sec.) and throughput (bit/s) separately. Table 4 is the statistic results for the three types of cases in simulation.

The simulation results with high-density of vehicles, show that, in comparison to conventional EDCA, the delays of various services in the proposed DE-AEDCA mechanism are decreasing. Thus, it can meet the demands of security-related emergency short messages and QoS requirements of speech service. Meanwhile the throughput of video service increases from 6100 kbits/s up to 8000 kbits/s, and the packet loss rate drops to 6.25%. In addition, the throughput of background service rises from 1850 kbits/s to 3250 kbits/s. Thus throughput of the video service and the background service respectively increases by 31.1% and 75.7%. Therefore the total throughput improves 38.6%. To sum up, the overall performances are improved.

For the middle-density case, the delay of various services has a certain decrease; the bandwidth needs of background service are met; no packet loss occurs; and the throughput performance is improved. In comparison to the EDCA mechanism, its overall performance is improved to some extent.

For the low-density vehicles, the delay is smaller than that in EDCA mechanism. Network packet loss does not occur any more. So the DE-AEDCA mechanism can meet the needs of various service applications properly.

As can be seen from the simulation results in the three situations of high-density and high network load, middle-density and high network load, as well as low-density and low network load, DE-AEDCA mechanism has both a higher network throughput and a smaller delay compared with EDCA mechanism. The reason is DE-AEDCA

<table>
<thead>
<tr>
<th>Table 2. Mac layer parameter</th>
</tr>
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<tbody>
<tr>
<td>CWmin</td>
</tr>
<tr>
<td>AC[3]</td>
</tr>
<tr>
<td>AC[2]</td>
</tr>
<tr>
<td>AC[1]</td>
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<tr>
<td>AC[0]</td>
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<table>
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<th>Table 3. Aifsn offset range of different Ac</th>
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<tr>
<td>Ni</td>
</tr>
<tr>
<td>AC[3]</td>
</tr>
<tr>
<td>AC[2]</td>
</tr>
<tr>
<td>AC[1]</td>
</tr>
<tr>
<td>AC[0]</td>
</tr>
</tbody>
</table>

Figure 3. The simulate results at high density. (a) medium access delay; (b) network throughput
mechanism considers the adjustment of competition parameters in conditions of a low network load and a high network load, so whenever in high-density and middle-density, or in the low-density vehicle cases, DE-AEDCA mechanism can have better QoS performance than EDCA mechanism.

In addition, among the data flows of same priority, DE-AEDCA mechanism increases the range of the parameters and reduces the possibility of choosing the same parameter by choosing a random offset, thus it can reduce conflict of the same priority data in network and improve the channel utilization. Experiments show that the DE-AEDCA mechanism always has better performance than EDCA mechanism in all kinds of intelligent

Table 4. The statistical results of simulations

<table>
<thead>
<tr>
<th></th>
<th>EDCA</th>
<th>DE-AEDCA</th>
</tr>
</thead>
<tbody>
<tr>
<td>Delay/average (ms)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>high</td>
<td>6</td>
<td>34</td>
</tr>
<tr>
<td>middle</td>
<td>2.1</td>
<td>3.7</td>
</tr>
<tr>
<td>low</td>
<td>1.1</td>
<td>1.7</td>
</tr>
<tr>
<td>Throughput/average (Kbits/s)</td>
<td></td>
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</tr>
<tr>
<td>high</td>
<td>76.8</td>
<td>512</td>
</tr>
<tr>
<td>middle</td>
<td>48</td>
<td>320</td>
</tr>
<tr>
<td>low</td>
<td>28.8</td>
<td>192</td>
</tr>
</tbody>
</table>
service case.

6. Conclusions

This paper considers QoS demands of various services in intelligence transportation, and analyzes the QoS based on the EDCA mechanism in IEEE 802.11p standard framework. A Differentiation Enhanced Adaptive EDCA is designed, which improves the QOS of EDCA by dynamically adjusting the size of competition window and further distinguishing the competition parameters of the data frames that have the same priority. The simulation results show that, in different situations of network congestion, the mechanism can discriminate the service priority, and make the high-priority service with low delay and high throughput. Meanwhile the overall performance of the network is improved significantly. Its performance is better than EDCA.

7. Acknowledgements

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The Performance Gain of Cognitive Radio in Adaptive Modulation Scheme

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ABSTRACT

Cognitive radio is considered as one of the main enablers for provisioning dynamic and flexible spectrum/channel allocation in wireless communications. The reliable data transmission over cognitive radio should employ modulation, coding etc. and thus the performance of such a new communication system should be realized. In this paper, we provide the performance analysis of adaptive modulation over a cognitive radio system in order to study the potential gain of cognitive radios in terms of spectral efficiency. The results obtained show that the performance gain of cognitive radio in adaptive modulation is remarkable.

Keywords: Cognitive Radio, Adaptive Modulation, Performance Gain

1. Introduction

The Cognitive Radio (CR) concept brought the idea to exploit the spectrum holes which result from the under-utilization of the electromagnetic spectrum in wireless communications. This fact is corroborated by the Spectrum Policy Task Force of the Federal Communications Commission (FCC) which ascertains that the legacy regulation on spectrum availability begets snags in potential spectrum access by users. More precisely, it was identified that spectrum bands seem to be unoccupied most of the time and some of them seem to be occupied partially by the primary (or licensed) users [1]. There are two types of cognitive radio systems namely opportunistic spectrum access or spectrum pooling and spectrum sharing or spectrum underlay [2]. Due to the fact that an OSA model presents a significant potential for studying the design aspects of spectrum utilization in CRNs with minimal deployment requirements, we have chosen it as the cognitive radio system in this work. In the following text, we provide the details of an OSA-based CRN model and its performance in terms of capacity achieved over Rayleigh fading channels. To this end, we assume a spectrum pooling system with cognition capabilities. Such a system is capable to reliably sense the spectrum range. Moreover, considering that the frequency carrier is divided into subbands then the mobile users are being served in a multi-band context. In such a context, the task of detecting holes in spectrum bands could be performed for instance by a ‘listen-before-talk’ strategy [3].

By assuming such a cognitive radio system we should be able to assess the performance of reliable transmission that us taking place at the physical layer. To this direction, we have chosen to study the performance analysis of adaptive modulation over such a cognitive radio system [4]. More specific, since the channel allocation in spectrum pooling systems employs optimum power control policy, we study the variable rate variable power case of adaptive modulation which also follows such a transmission policy [5]. In order to make an objective performance analysis and evaluation, we present first the performance of cognitive radio system over a particular fading propagation environment. Thus, we calculate the capacity of such a system over a flat fading channel with Rayleigh coefficients [6]. We conclude this paper with the numerical results of this performance analysis and evaluation of adaptive modulation over cognitive radio.

2. Background

2.1 Spectral Efficiency of Cognitive Radio in Rayleigh Fading Channels

In spectrum pooling systems, the spectrum or channel allocation technique gives priority to primary users. Subsequently, the secondary users are assigned the detected spectrum holes. These holes are assumed voids in sub-band range. Thus, secondary users fill these voids as long as they achieve the desirable transmit power level. More specific, the spectrum is divided into \( N \) sub-bands and

\[
N = \frac{N_{\text{total}}}{N_{\text{spectrum holes}}}
\]
each user $l$ is trying to transmit by using an optimum power control policy. The system’s operation is considered to be under a wide-band context. Hence, the $N$ sub-bands of the cognitive radio channel extend to infinite ($N \rightarrow \infty$). The channel is assumed with fading components that are varying slowly in time, i.e. the receiver is able to sense and track the channel fluctuations. The channel fluctuations represent the channel gains $h_l$ for each user $l$ and are assumed over a block fading length in order to be able to retain their values constant during the processed block. Under these assumptions, the average capacity of user $l$ in bits/sec/Hz obtained by the following equation:

$$C_{l,\infty} = \int_0^\infty \left( \frac{1}{\gamma_0} - \frac{P_l(t)t}{N_0} \right) e^{-t} dt$$

for $l \in [1, L]$ that is the number of users being served by the cognitive system with $N$ sub-bands. However, the channel allocation for each user is performed in order to maximize the transmission rate when the optimal power is achieved. Hence, the power of $P_l$ is subject to the average constraint:

$$\int_0^\infty \left( \frac{1}{\gamma_0} - \frac{N_0}{t} \right) e^{-t} dt = 1$$

This technique for optimal power and rate adaptation is relied on water-filling algorithm and the corresponding SNR cut-off levels. In our case, the cut-off SNR levels are equal to $\gamma_0 N_0$ that denote the values of the channel gain above which a user can transmit on a sub-band. The term $N_0$ denotes the additive Gaussian noise at the receiver, while the $\gamma_0$ is calculated by keeping the aforementioned transmit power constraint expressed by equation (2). However, in order to obtain the performance of adaptive modulation over cognitive radio, we keep below the formulation of Rayleigh channel fading model [6]. More specific, the capacity per unit bandwidth is expressed as follows when optimal power and rate adaptation is considered:

$$\left\langle C_{1,\infty} \right\rangle_{Rayleigh} = \log_2 \left( e^{\gamma_0/\gamma} \right) \left( \frac{\gamma_0}{\gamma_0 + \gamma} \right)$$

In this case, the cut-off level is related to average SNR $\gamma$ values.

In previous analysis, we keep on purpose the notation of the first user as $\left\langle C_{1,\infty} \right\rangle_{Rayleigh}$, since the spectral efficiency over cognitive radio is related to the average capacity of the primary user and the band factor gain of the cognitive radio itself. The band factor gain $\Delta_{\infty}$ of a cognitive radio system is defined as the spectrum band, which is being sensed as void from user $l$ to user $l + 1$ over the total bandwidth $W$. This gain is obtained as follows:

$$\Delta_{\infty} = 1 - \exp(-SNR_{off})$$

where $SNR_{off}$ denotes the cut-off level in the corresponding SNR values. Taking into account the aforementioned formulas for the Rayleigh fading channel, we can note that the SNR cut-off value for the Rayleigh fading channel is equal to $SNR_{off} = \gamma_0 / \gamma$. In case of the whole radio system, the sum spectral efficiency is considered which is obtained as follows [3]

$$\left\langle S_{sum} \right\rangle_{Rayleigh} = \frac{1 - \Delta_{\infty}}{1 - \Delta_{\infty}} \left\langle C_{1,\infty} \right\rangle_{Rayleigh}$$

Figure 1 shows the spectral efficiency achieved in cognitive radio and the one achieved in conventional radio systems. The numerical results derived from the aforementioned analysis are dedicated to Rayleigh channels. More specific, the line with cross marker type illustrates the spectral efficiency of optimal power and rate adaptation using the closed-form (3). The line with square marker type depicts the sum spectral efficiency of cognitive radio using the combination of closed-forms (3-6). It is evident that the propagation over cognitive radio yields an increase in capacity that has a considerable value in low average SNR regions that is ranged between 0.2-0.3 bits/sec/Hz. For comparison purposes, we also depict the sum spectral efficiency of cognitive radio when only the primary user is considered that is in fact the capacity of user 1 over Rayleigh fading channel model.

### 2.2 Spectral Efficiency of Adaptive Modulation

Although, adaptive modulation was introduced in [5] studying both variable rate and variable power (VRVP)
modulations, it was followed with the analysis of variable-rate and constant-power (VRCP) case presented in [4]. The VRVP instead of VRCP uses power adaptation that is relied on optimal power control policy that we mention above. On the other hand, the achievable performance of cognitive radio depends on the band factor gain which relies on the corresponding channel allocation technique. As specified in [3], this capability is related to the frequency variation that is expressed by the channel fading distribution. Since power control policy is employed, the frequency variation depends on the cut-off fade level. The obvious inference is that we use the VRVP case for studying the performance of adaptive modulation over cognitive radio, since the cognitive radio system employs such an optimal power control policy for allocating channels to each user. To this end, we should rely on the performance of VRVP case of adaptive modulation and its spectral efficiency in particular [5].

The VRVP scheme deploys MQAM constellations denoted as the set \( \{ M_j : j = 0,1,\ldots,N \} \) that can be chosen according to the instantaneous signal-to-noise ratio \( \gamma \) during the symbol period. Throughout this set, the \( M_0 \) choice means no data transmission. Particularly, each constellation is associated with a fading region that is emerged by the division of the range \( \gamma \) into \( N+1 \) regions known as fading regions. Thus, when the fading level is set in the \( j \)th region then the constellation \( M_j \) is chosen. Consequently, the current data rate of the adaptable system is \( \log_2 M_j \). However, since the transmit power \( S(\gamma) \) should also be adapted in order to retain the average power constraint \( \bar{S} \) and therefore the received SNR is equal to \( \gamma \cdot S(\gamma) / \bar{S} \). Thus, the adaptive scheme should be able to decide by which rate should transmit in the next period and which will be the transmit power either. Both rate and power are controlled under the required BER value (target BER) which should be retained at the physical layer.

Afterwards, the aim of such a system with power control policy is to maximize the spectral efficiency subject to the power constraint

\[
\frac{S(\gamma)}{\bar{S}} = \begin{cases} 
\frac{1}{\gamma_0} - \frac{1}{\gamma \cdot K}, & \gamma \geq \gamma_0 / K \\
0, & \gamma < \gamma_0 / K
\end{cases}
\]  

(6)

It is obvious that the cut-off level is equal to \( \gamma_0 / K \) where \( K \) is in relation with the target BER \( (K = -1.5 / \ln(5BER)) \). Denoted the \( \gamma_0 / K \) as \( \gamma_\epsilon \), the data rates are fallen in the fading region with optimum power allocation expressed as \( \gamma / \gamma_\epsilon \). Afterwards, the spectral efficiency of VRVP is the sum of data rates multiplied by the probability that the fading level falls in that region

\[
R = \frac{B}{B} \sum_{j=1}^{N} \log_2 (M_j) p(\gamma_j \leq \gamma / \gamma_\epsilon < M_{j+1}) 
\]

(7)

3. Adaptive Modulation over Cognitive Radio

As described above, the achievable performance of cognitive radio depends on the band factor gain. Therefore, the band factor of the VRVP adaptive modulation, is expressed as follows

\[
\langle \Delta_\epsilon \rangle_{VRVP} = 1 - \exp(-\gamma_\epsilon / \gamma) 
\]

(8)

Taking into account the sum spectral efficiency of cognitive radio system, we derive the sum spectral efficiency of VRVP adaptive modulation over cognitive radio which obtained as follows

\[
\langle S_{\epsilon_{sum}} \rangle_{VRVP} = \frac{1 - \langle \Delta_\epsilon \rangle_{VRVP}^L}{1 - \langle \Delta_\epsilon \rangle_{VRVP}^U} \left( \frac{R}{B} \right) 
\]

(9)

3.1 Numerical Results

1) Sum Spectral efficiency: Figure 2 shows the numerical results of adaptive modulation scheme over cognitive radio.

We keep as reference the numerical results derived in [5]. The regions denote the number of constellation that the system employs i.e. the number of deployed M-QAM at the physical layer. It is obvious the performance gain of cognitive radio at the lower average SNR regions while the gain is negligible at the higher average SNR regions. More specific, when 5 regions are considered then the performance gain in terms of bits per second per symbol period is approximately 0.2 in the average SNR region below 10 dB. In case of 4 regions, this gain seems to be smaller than 0.2 bps/Hz while in 3 regions is follows the same scaling procedure i.e. decrease. However, the decrease is more evident in average SNR regions higher that 10 dB.
2) Performance Gain: Figure 3 shows the performance gain in adaptive modulation with 5 regions. We depict the results using $L$ users. It is obvious from the results that the increase in number of users leads to increase in the performance gain. However, the increase is getting negligible when the number of users is getting increased either. For instance, above 10 users the performance gain is approximately equal to that achieved with 10 users precisely. Moreover, this figure corroborates the performance gain of cognitive radio in low average SNR regions. We should remind that we have resulted in the same conclusion for the capacity of cognitive radio over fading channel illustrated in Figure 1, where the results are related to instantaneous received SNR. Afterwards, it can be noted that the performance gain of cognitive radio at the physical layer in general, is remarkable at low fading regions and is retained at the same level for more than a small number of users.

REFERENCES


A Perspective on Traffic Measurement Tools in Wireless Networks

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ABSTRACT

To understand the characteristics of the wireless networks, the network usage data from wireless measurement tools are essential. The data collection is a process of collecting the network time-varying information in standardized format and from standard interfaces. The characteristics include signal propagation, received signal quality, network traffic, active applications and mobility of the mobile terminal (MT). The purpose of the measurement is to collect vital data of the wireless network. There are several tools available for this purpose. The most widely used network measurement tools are client side measurement tool, Syslog, Simple Network Management protocol (SNMP), network sniffing, wireless sniffing. This paper discusses the different wireless measurement tools and their benefits and limitation these tools.

Keywords: SNMP, Wireless Sniffing, Syslog, Network Sniffing, Wireless Networks

1. Introduction

The data collection is a process of collecting the network time-varying information in a standardized formats and from a standard interfaces. This needs a Portable tool for data collection. The collected data need to be processed effectively without losing the “tail” of the data and identifying holes and cleaning data. In the pre-processing mechanism, the time-varying network parameters are arranged in an order. These time series may have few missing entries, due to the minor flaws in the measurement tools, which are estimated and filled using time series techniques.

There are many implicit differences in wired and wireless medium. Wired medium will have clear points of connection but wireless medium is physically dispersed. The mobility in wireless networks and novel devices used inspires new usage patterns. In this prevailing scenario, the measurement of wireless network information is essential. This strengthens our understanding of user and network behaviours. The better understanding leads to better network models. The improved network models are momentous to improvement in terms of network protocols, distributed algorithms, applications and improved deployment strategy.

The NGWN provides users with a wide range of services across HWNs coexisting with diverse throughput and coverage with a single MT. The existing cellular networks will provide communication services over a wide geographical area but has limited bandwidth to support emerging data services. But the future 3G cellular and 4G systems, such as UMTS, Wi-MAX (802.16), have lesser coverage and higher bandwidth when compared to cellular networks. The WLAN (IEEE 802.11 a/b/g/n) is able to provide higher data rate but with lesser coverage compared to cellular and 4G systems. Therefore an integration of cellular networks, Wireless Local Area networks (WLAN) and Wi-MAX would result in higher bandwidth, more network coverage and will also help in enhanced user mobility and with choice of new services and enhanced QoS [1]. Figure 1 illustrates the Speed v/s Mobility comparison of wireless networks. The characteristics of the different wireless networks are depicted in Table 1.

The process of network switching will involve the following three phases – network discovery, switching decision and execution [2]. The decision phase will play an important role in balancing network utilization, fulfilling the user requirements and QoS requirements of network
A Perspective on Traffic Measurement Tools in Wireless Networks

Figure 1. Speed vs. mobility comparisons of different wireless networks

Table 1. Attribute comparisons of different wireless networks

<table>
<thead>
<tr>
<th>Wireless Network</th>
<th>BW (Mbps)</th>
<th>Modulation Technique</th>
<th>Freq (GHz)</th>
<th>Coverage</th>
</tr>
</thead>
<tbody>
<tr>
<td>IEEE802.11a</td>
<td>20</td>
<td>OFDM</td>
<td>5</td>
<td>Indoor: 35 meters, Outdoor: 120 meters</td>
</tr>
<tr>
<td>IEEE802.11b</td>
<td>11</td>
<td>DSSS</td>
<td>2.4</td>
<td>Indoor: 38 meters, Outdoor: 140 meters</td>
</tr>
<tr>
<td>IEEE802.11g</td>
<td>54</td>
<td>OFDM/ DSSS</td>
<td>2.4</td>
<td>Indoor: 38 meters, Outdoor: 140 meters</td>
</tr>
<tr>
<td>IEEE802.11n</td>
<td>600</td>
<td>OFDM</td>
<td>5</td>
<td>Indoor: 70 meters, Outdoor: 250 meters</td>
</tr>
<tr>
<td>HiperLAN2</td>
<td>54</td>
<td>OFDM</td>
<td>5</td>
<td>Indoor: 50 meters, Outdoor: 50 meters</td>
</tr>
<tr>
<td>802.16e</td>
<td>Up to 125</td>
<td>OFDMA</td>
<td>2-6</td>
<td>Up to 35000 meters (35Kms)</td>
</tr>
<tr>
<td>802.16m</td>
<td>Up to 300</td>
<td>OFDM</td>
<td>Up to 6</td>
<td>Up to 50000 meters (50 Kms)</td>
</tr>
<tr>
<td>EDGE Evolution</td>
<td>9.6-384</td>
<td>TDMA/FDD</td>
<td>900/1800 /1900 MHz</td>
<td>Up to 40000 meters (40Kms)</td>
</tr>
<tr>
<td>UMTS/W-CDMA</td>
<td>2</td>
<td>FDD, TDD</td>
<td>Up to 20000 meters (20Kms)</td>
<td></td>
</tr>
</tbody>
</table>

The signal fading in a wireless system is a common phenomenon of the radio channel. They are classified into two types, *Flat fading* and *Frequency selective fading*. In a narrowband wireless channel, the consistency bandwidth of the channel is larger than the bandwidth of the signal. In such channels all frequency components of the signal will experience the same amount of fading. Such a fading is called as ‘*Flat fading*’. On the other hand, in a wideband wireless channel the coherence bandwidth of the channel is smaller than the bandwidth of the signal. This result in different frequency components of the signal, experiencing the different amount of fading called as ‘*frequency selective fading*’. Apart from these two types of fading, when the MT is moving at a high speed, the signal strength varies severely and undergoes deep fading within the small time frame. This type of fading is named as ‘*Fast fading*’ [9].

The next generation wireless systems typically have higher bandwidth and support optimal mobility, need to challenge with the frequency selective fading and fast fading. The next generation wireless systems make use of low complexity techniques such as Orthogonal Frequency Division Multiplexing (OFDM) in the physical layer and Orthogonal Frequency Division Multiple Access (OFDMA) mechanisms in the link layer to prevail over the effect of frequency selective fading [10].

2. Wireless Network Measurements

To understand the characteristics of the wireless networks, the network usage data from wireless measurement tools are essential. The characteristics include signal propagation, received signal quality, network traffic, active applications and mobility of the MT. The purpose of the measurement is to collect vital data of the wireless network. There are several tools available for this purpose. The most widely used network measurement tools are client side measurement tool, Syslog, Simple Network Management protocol(SNMP), network sniffing, wireless sniffing.

2.1 Client Side Network Management Tools

The wireless measurement tools mentioned above *i.e.* Syslog, SNMP, network sniffing and wireless sniffing tools are intended to monitor the network from the viewpoint of the network. In client side methods the measurement tools are installed in client to measure the activities at the client side. This client side measurement has many advantages.

A client side tool can accurately determine what exactly a client is doing. While Syslog will provide information about set of clients which are associated to the particular AP/BS, a client side tool can list all the APs/BSs that a client can handle, which are useful for mobility tracing. A client side tool can list all the applications that are running on it, rather than just those applications that...
generate network traffic. Client side tools are extensively used in WMAN and WWAN measurements [11,12].

Writing a generic client side program, such as tcpdump, Wireshark formerly called Ethereal and kismet will be a challenging task, because it has to run on variety of operating systems and different device drivers.

2.2 Syslog

Syslog records detail steps of association, and have been used effectively for studying user activity patterns [13, 14]. To all intents and purposes Syslog is a standard for sending and receiving of log messages [15]. The wireless APs and BSs can be configured to log appropriate events in the network. The Syslog messages are used to understand the state of an MT in the wireless network. The AP or BS can generate a time stamped message whenever an MT authenticates, de-authenticates, associates, dis-associates or roams to that AP or BS. By collecting these messages it is possible to determine the state of the MTs on the network. The Syslog messages are stored and analyzed locally in the BS or transmitted across the network for storage and analysis by a dedicated computer.

There is no standard format for Syslog messages. The messages that APs or BSs send can vary in format and amount of information contained. In most of the cases APs and BSs manufactured from same manufacturer will have different Syslog message formats. In certain cases the message formats differ for each version of the same product. In a heterogeneous wireless environment, multiple type of APs and BSs with varieties of Syslog message formats. It is necessary to translate these messages into an intermediate format prior to the data analysis. In some of the measurement studies [16,17], the multiple Syslog message formats are translated to general, intermediate parsed format for the purpose of analysis. Figure 2 indicates the parsed Syslog trace data format.

2.3 SNMP

The SNMP is a generic tool in measuring and managing a network device, called ‘network object’ in the network management terminology [18]. The SNMP provides information on both traffic volume and the number of active users. This makes the SNMP the most suitable technique used for both traffic studies [14,19,20] and user mobility studies [21].

A network administrator runs a tool known as ‘manager’, which communicates with SNMP ‘agents’. Agents run on network objects and provide interface between the object and manager. A network object can contain several objects, such as statistics or configuration items, arranged in a database known as Management Information Base (MIB). The network statistics are stored in the MIB variables and these variables are represented in a standard format known as Abstract Syntax Notation (ASN). The manager queries the agent for the purpose of measurement and agent replies by extracting information from the MIB variables. Both request and reply will be in the standard SNMP message format [22]. In the recent version of SNMP few MIB variables, like MAC address, IP address, Signal strength, Power saving mode, Network session length and Traffic of the MT associated with AP or BS, are specific to the wireless network [23]. The SNMP messages are shown in Figure 3.

Some of the advantages of the SNMP are

- SNMP messages provide more detailed information about the status of the network than Syslog messages.
- SNMP provides information on both traffic volume and the number of active users. Hence it is suitable to be used for both traffic studies and user mobility studies.
- SNMP messages are generally device independent and are usually available in a standard format.

The drawbacks of SNMP are

- SNMP-based approaches is that they require an interval between SNMP polls (typically every 1-5 minutes), and it has been shown that long poll intervals may miss wireless clients that associate with APs for less than this poll interval [24].

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Figure 2. Parsed syslog format

Figure 3. Set of SNMP messages
The SNMP-based approaches may be able to retrieve such detailed wireless MAC/PHY information through the use of a properly defined MIB, the most existing SNMP MIBs for APs (MIB-I (RFC 1066), MIB-II (RFC 1213), and 802.11 MIB (IEEE Std 802.11-1999)) provide very limited visibility into MAC-level behaviour.

2.4 Network Sniffing

The network or packet sniffing refers to the process of capturing of the network traffic at the network interface. For the purpose of sniffing, the network interface should be in a promiscuous mode. In this mode the interface will ignore its assigned address and captures all the frames/packets present in the network. There are programs, such as tcpdump, Ethereal and kismet, which will capture and analyze the frame/packet [25-27].

Kismet is an 802.11 layer2 wireless network detector, sniffer, and intrusion detection system. Kismet will work with any wireless card which supports raw monitoring (rfmon) mode, and (with appropriate hardware) can sniff 802.11b, 802.11a, 802.11g, and 802.11n traffic. The Kismet is good for WLAN surveillance. It is capable to sense the details of all wireless access points (WAPs) and WLAN nodes, showing channels, use of encryption and signal strength.

Ethereal is a network packet analyzer. A network packet analyzer will try to capture network packets and tries to display that packet data as detailed as possible. You could think of a network packet analyzer as a measuring device used to examine what’s going on inside a network cable. The Ethereal is not an intrusion detection system. It will not warn when someone does strange things on the network that the user isn’t allowed to do. However, if strange things happen, Ethereal might help you figure out what is really going on. Ethereal will not manipulate things on the network, it will only “measure” things from it. Ethereal doesn’t send packets on the network or do other active things (except for name resolutions, but even that can be disabled). The trace of an ethereal is shown in Figure 4.

The important concern with network sniffing is that the volume of data generated from the sniffing process is much larger than Syslog and SNMP. A typical sniffing of 802.11b wireless network operating at 11 Mbps speed can generate several gigabits of data within few minutes. It is vital to ensure that sufficient disk space is available to store the captured frames/packets in the hard disk. Another major concern in the network sniffing is the privacy of captured information. The frame/packet that is captured through sniffing may contain sensitive data especially when the data within the frame/packet is not encrypted. The issue of privacy may be alleviated by only capturing the header data, which may be sufficient for a network measurement. Even with this, the privacy problem is not completely overcome as some vital information, such as packet size, MAC/IP address, higher layer protocol and inter-arrival time, stand exposed. The result of such a sniffing is referred to as a trace.

2.5 Wireless Sniffing

The wireless sniffing is a WLAN measurement tool [28]. Syslog, SNMP and network sniffing are the generic measurement tools which will be used in measuring all types of wireless as well as wired networks. The wireless sniffing is a measurement tool useful only for a wireless network. It will operate at AP/BS or at a switch that connects wireless network to the wired backbone. The disadvantage of wire side measurement is that not all wireless data observable from the wired network, such as management frames, beacons, retransmissions and collisions, send traffic via wired network. The wireless sniffer is widely used to collect the MAC level frame information in a wireless network. Even though wireless sniffer can be installed on a host under measurement, in majority of cases, it is installed on an autonomous device. This independent device could be a laptop or any MT or a PDA system. This makes the wireless sniffer to monitor the wireless network in promiscuous mode without interfering with the stations under study/monitoring. Wireless sniffers capture both the data frames as well as management frames. The management frames captured by wireless sniffer includes beacon frames, request to send (RTS) frames, clear to send (CTS) frames and acknowledgement (ACK) frames. Nevertheless, there is need of special hardware and software in form of drivers is essential for effective working of a wireless sniffer. Ethereal and Kismet are the most admired wireless sniffer and analyzer software. There are good amount of re-
search works reported on wireless performance using Wireless sniffers. The measurement of streaming media over wireless link using independent sniffers [29,30], measurement of congestion in wireless LAN [31], in the network monitor research in [32], a complete wireless sniffer system is implemented and used to characterize a typical computer science department WLAN traffic.

Wireless measurement can be applied to the mobile host. This is accomplished by placing wireless network interface card in a monitor mode. In this mode, the wireless card captures all types of frames/packets. These frames/packets may be analyzed similar to those of network sniffing. Since this mode is not a promiscuous mode it limits the wireless sniffer in the mobile host as a work sniffing. Since this mode is not a promiscuous mode it limits the wireless sniffer in the mobile host as a simple network monitoring tool. Figure 5 shows an example of wireless sniffing trace.

The advantages and disadvantages of wireless sniffing are as listed below.

Advantages of wireless sniffing are:
- Wireless Sniffing done be an independent sniffer in a promiscuous mode will not cause any interference with the hosts under test in wireless experiment. Therefore, sniffing can be used to measure these devices, such as the wireless game consoles, which do not provide general accesses for measurement purpose.
- Wireless sniffing can provide frame level information and wireless network conditions, such as the RSSI and sending capacity.
- Wireless sniffers can be used as wireless network diagnostic tools as they are capable to capture wireless management frames, such as RTS, CTS, Authentication / De-authentication frames and Association / Disassociation frames.

Disadvantages of Wireless sniffers are:
- Wireless sniffers cannot record all the frames that are transmitted over the network [31,33] since the sniffer is only capturing the frames at its own location this results in non capturing of the packets lost due to a hidden terminal and packets lost due bit errors.

- The Received Signal Strength Indicator (RSSI) is measured relative to the wireless sniffer installation location. This measurement of received signal strength may not be same as the AP or the clients that are remote from the wireless sniffer installation location.

<table>
<thead>
<tr>
<th>No.</th>
<th>Time</th>
<th>Source</th>
<th>Destination</th>
<th>Protocol Info</th>
</tr>
</thead>
<tbody>
<tr>
<td>2458</td>
<td>55.951347</td>
<td>XXX_1a:97:ab (RA)</td>
<td>IEEE 802.11</td>
<td>Clear-to-send</td>
</tr>
<tr>
<td>2459</td>
<td>55.951553</td>
<td>XXX_1a:97:ab YYY_11:13:30:a8</td>
<td>IEEE 802.11</td>
<td>Data</td>
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<td>IEEE 802.11</td>
<td>Clear-to-send</td>
</tr>
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<td>XXX_1a:97:ab YYY_11:13:30:a8</td>
<td>IEEE 802.11</td>
<td>Data</td>
</tr>
<tr>
<td>2462</td>
<td>55.952847</td>
<td>XXX_1a:97:ab (RA)</td>
<td>IEEE 802.11</td>
<td>Clear-to-send</td>
</tr>
<tr>
<td>2463</td>
<td>55.953895</td>
<td>XXX_1a:97:ab YYY_11:13:30:a8</td>
<td>IEEE 802.11</td>
<td>Data</td>
</tr>
<tr>
<td>2464</td>
<td>55.954070</td>
<td>XXX_1a:97:ab(RA)</td>
<td>IEEE 802.11</td>
<td>Acknowledgement</td>
</tr>
</tbody>
</table>

Figure 5. Wireless sniffing trace in WLAN

- The location of the sniffer plays an important role in the wireless sniffing. For example, a location very close to an AP is helpful when studying the AP behaviour, but may miss some traffic sent from a distant client due to signal attenuation and on the other hand the similar effect is experienced when the sniffer is near to the client and away from the AP. This results in ‘Generic losses.
- The wireless sniffing suffers from ‘AP losses due to the firmware incompatibility between AP and monitoring device. These losses can be minimized by using redundant sniffers or sniffers with interface cards having different chipset and using antennas of different gains and positioning the sniffers at strategic places [34].

3. Conclusions

The wireless Measurement is an important phase of any study on wireless networks. The data collection phase acts as the building stone of the study of wireless measurements. The various wireless measurements tools used to measure the characteristics will have their own strength and weaknesses. The wireless sniffing is one of the measurement techniques that could be used for effective measurement of wireless network time varying characteristics. The data collection of wireless networks can be supported by standardization of interfaces and formats of information which is common to all network vendors. The archival of the network data will help in better understanding and methodical study of wireless networks. Our future work includes the building up the effective measurement framework and step ahead for predicting the missing values in measurements by applying intelligent hybrid technique like Fuzzy neural approach.

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WiMAX System Simulation and Performance Analysis under the Influence of Jamming

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ABSTRACT

This paper presents simulation of WiMAX based system under jamming. The performance of the system was found out to greatly differ with the use of different jamming signals, allowing central areas to be identified, where system development should be focused on. In addition, from the basic theory point of view, rather surprising results were also found. This work should give a clear picture of how the studied WiMAX system performs under jamming as well as without jamming. The results show that some forms of interference degrade the performance of the system rapidly, thus the form of incoming jamming should be known and considered before deploying the system. Single carrier jamming and multi–carrier jamming are discussed here. The issues related to jamming and jamming reduction techniques are also covered. Jamming can destroy communication in the targeted area. Multi–carrier jamming is challenge in WiMAX because WiMAX is having OFDM based physical layer. Simulation is the main approach in this paper. OPNET MODELER is the software used for the simulation purpose.

Keywords: Jamming, Multicarrier, WiMAX, OFDM

1. Introduction

IEEE 802.16 is the standard for WiMAX. WiMAX is also known as wireless broadband. IEEE 802.16d – 2004 is known as fixed WiMAX and IEEE 802.16e – 2005 is known as mobile WiMAX [1]. In wired networks physical layer threats are not important but in wireless air is used as medium so physical layer threats comes into picture [2]. In wireless jamming and scrambling are considered as physical layer threats. Mac layer threats are different than physical layer threats. Here simulation approach is used to see the performance of the IEEE 802.16e – 2005 fixed NLOS (Non line of sight) system in jamming environment. Jamming is achieved by introducing a source of noise strong enough to significantly reduce the capacity of the WiMAX channel. The information and equipment required to perform jamming are not difficult to acquire. Resilience to jamming can be augmented by increasing the power of signals or increasing the bandwidth of signals via spreading techniques such as frequency hopping or direct sequence spread spectrum. The practical options include a more powerful WiMAX transmitter, a high gain WiMAX transmission antenna, or a high gain WiMAX receiving antenna [3]. It is easy to detect jamming in WiMAX Communications as it can be heard by the receiving equipment. Law enforcement can also be involved to stop jammers. Since jamming is fairly easy to detect and address, so it does not pose a significant impact on both the WIMAX users and systems. Single carrier jamming and multi–carrier jamming are considered here for simulation approach. Single carrier jamming is used to jam the particular band of frequencies. In single carrier jamming carrier frequency and bandwidth of the targeted system should be known. In multi–carrier jamming the frequencies of carriers of targeted system should be known. Simulation approach is easy compare to practical approach. The issues related to practical approach will be described in the later part.

2. The Investigated Physical Layer

The primary operation bands of WiMAX include frequencies 10–66 GHz, 2–11 GHz and license–exempt frequencies below 11GHz (primarily 5–6 GHz). According to these operation bands, WiMAX PHY defines five specifications for different operation scenarios. Among them, Wireless MAN–OFDM PHY is based on orthogonal frequency–division multiplexing (OFDM) technology and designed for NLOS operation in the frequency bands below 1 GHz [4]. It is selected to be the air interface of...
the system under investigation in this paper. Denial of service is very similar to jamming but it is MAC layer threat [5].

At the transmitter side, the information data first undergoes channel coding composed of randomization, forward error correction (FEC), and interleaving. Randomizer uses a Linear Feedback Shift Register (LFSR) to scatter long data strings of zeros or ones. FEC concatenates an outer Reed–Solomon encoder with an inner rate compatible convolutional encoder. FEC helps to correct the errors in subcarriers to a certain limit. The interleaver takes two permutations to rearrange the subcarriers so that the burst errors are distributed more uniformly at the demodulation input. After channel coding, data bits are mapped and modulated onto the allocated subcarriers by QPSK, 16–QAM and 64–QAM modulation. Subsequently, data are transmitted by OFDM method. In the receiver side, all the procedures carried out in the transmitter side are implemented again but in a reverse direction. One OFDM symbol can be divided into two parts in time domain: the cyclic prefix (CP) time and the useful symbol time [6,7]. The CP locates in the beginning of the symbol and is a duplication of the tail of the useful symbol, which is introduced to mitigate the effect of multi-path. In frequency domain, an OFDM symbol is composed of a series of subcarriers. In Wireless MAN–OFDM PHY, the number of subcarriers is 256. As per Figure 1, three types of subcarriers can be categorized: 192 data subcarriers carrying payload, 8 pilot subcarriers mainly for channel estimation, and 56 null subcarriers for guarding purpose. The pilot subcarriers distribute evenly among the data subcarriers. This is standard symbol in frequency domain.

Channel estimation is mandatory for the OFDM systems employing coherent detection. Comb type pilot channel estimation is capable of collecting instant information of the channel and therefore used in this research. The channel estimation for the payload subcarriers is achieved by interpolation, using the channel information obtained at the 8 pilot subcarriers. In this paper IEEE 802.16e–2005 is simulated under jamming. In Mobile WiMAX, the FFT size is scalable from 128 to 2.048. Here, when the available bandwidth increases, the FFT size is also increased such that the subcarrier spacing is always 10.94 kHz. This keeps the OFDM symbol duration, which is the basic resource unit, fixed and therefore makes scaling have minimal impact on higher layers. A scalable design also keeps the costs low. The subcarrier spacing of 10.94 kHz was chosen as a good balance between satisfying the delay spread and Doppler spread requirements for operating in fixed and mobile environments. This subcarrier spacing can support delay–spread values up to 20 micro seconds and vehicular mobility up to 125 km per hour when operating in 3.5 GHz. A subcarrier spacing of 10.94 kHz implies that 128, 512, 1,024, and 2,048 FFT are used when the channel bandwidth is 1.25 MHz, 5 MHz, 10 MHz, and 20 MHz, respectively. 2–11 GHz is used for Fixed NLOS and 2–6 GHz is used for mobile NLOS. It should, however, be noted that mobile WiMAX may also include additional bandwidth profiles. For example, a profile compatible with WiBro will use an 8.75 MHz channel bandwidth and 1,024 FFT. This obviously will require different subcarrier spacing and hence will not have the same scalability properties. The number of subcarriers may be 512, 1024 and 2048. Data subcarriers, Null subcarriers and Pilot subcarriers are also given in next section for 512, 1024 and 2048 subcarriers.

3. Jamming in Detail

There are two types of jamming: single carrier jamming and multicarrier jamming.

3.1 Single Carrier Jamming

The goal of single carrier jamming is to insert an interference signal into the enemy communication system so that the wanted signal is completely submerged by the interference. This form of jamming is also known as denial of service attack or obscuration jamming. The optimal jamming waveform is intuitively white Gaussian noise (WGN), since from the information theory point of view, it has maximum entropy. This conclusion can also be drawn from the fact that the receiver can not distinguish between jammer injected noise and its own [8-9]. Based on the relationship between jammer bandwidth and that of the equipment, single carrier jamming can be categorized into narrow– (spot) and wideband (barrage) jamming. The relationship is conveniently expressed as

$$\frac{B_J}{B_{VS}} = \frac{\text{Jammer_bandwidth}}{\text{Victim_system_bandwidth}}$$  \hspace{1cm} (1)

Typically, if the ratio BJ/BVS is less than 0.2 jamming is considered to be spot jamming and if greater than 1, barrage jamming. The main advantage of single carrier jamming is that, very little information about the enemy’s equipment is required. However, there are great many factors, which make the performance of a noise jammer to drop below its theoretical capability. The fact that a noise jammer has to function on victim systems using arbitrary polarizations, generally leads to usage of...
either 45 degrees slant polarized or circularly polarized jammer radiations. This causes a rather modest effective radiated power (ERP) drop of typically 3 dB, but more serious losses in the order of tens of dB occur as a result of bad noise quality and e.g. orthogonal polarization between jammer and victim antennas. The easiest way of creating an effective noise jammer is to pass band–limited noise through an RF–amplifier and to the transmitting antenna. This method is also known as direct noise amplification (DINA).

3.2 Multicarrier Jamming

Multicarrier jamming differs from single carrier jamming by being suitable only for jamming the system it is designed for. The general idea is to determine the most critical vulnerability of the victim system in terms of the carriers used and then inject a very narrowband signal, e.g. zero bandwidth sine signal, onto the those carriers. If data subcarriers are destroyed by jamming then information is lost so throughput is reduced so bit error rate (BER) is increased. If pilot subcarriers are destroyed by jamming then channel estimation is very difficult. More information about the enemy’s equipment is required in multicarrier jamming, because only some subcarriers are targeted. In simulation only fixed NLOS is considered. In OFDMA there are data subcarriers, pilot subcarriers, guard subcarriers. The simulation parameters required for multicarrier jamming are described in next section.

4. Simulation Setup

4.1 Simulation Parameters for Single Carrier Jamming

In this paper fixed NLOS WiMAX system is simulated under jamming. Single carrier jamming scenario is shown in Figure 2.

The node model of jammer is given in Figure 3.

Source parameters for the jammer are packet inter–arrival time, packet size, jammer start time, stop time and transmitter power. According to target system these parameter are chosen.

Radio parameters are data rate, packet formats, frequency and bandwidth. Frequency is chosen according to carrier frequency of target system.

Antenna can be isotropic, WiMAX Omni and WiMAX sector antenna. Polarization is also much important in practical case of jamming. If polarization of jamming system antenna is not proper then it results in wastage of power.

4.2 Simulation Parameters for Multicarrier Jamming

Multi carrier jamming is difficult to simulate. In this paper multi carrier jamming effect is modeled. Symbol duration is calculated by the following equation. In scalable OFDMA the symbol duration and subcarrier spacing is

\[ T_s = T_b + T_g \]  \hspace{1cm} (2)

\[ T_b = \frac{1}{\text{delta}_f} \]  \hspace{1cm} (3)

\[ \text{delta}_f = \frac{\text{bandwidth}}{\text{subcarriers}} \times n \]  \hspace{1cm} (4)

\[ T_g = \frac{T_s}{8} \]  \hspace{1cm} (5)

where, \( n = \text{sampling factor} \)

\( \text{delta}_f = \text{tone spacing} \)

\( T_s = \text{symbol duration} \)

\( T_g = \text{guard time} \)

\( T_b = \text{useful symbol duration} \)

fixed. Number of sub–carriers and bandwidth is changed to keep symbol duration and subcarrier spacing fixed. Equation (4) shows number of subcarriers is proportional to bandwidth of channel for scalable OFDMA. It is multicarrier scheme. Multicarrier jamming scenario is shown in Figure 4.
WiMAX System Simulation and Performance Analysis under the Influence of Jamming

Figure 4. Scenario used to simulate multicarrier jamming

In this scenario fixed NLOS is considered. There are two subscriber stations. Application and profile node is used to create an application. WiMAX configuration node is used to configure WiMAX properties. IEEE 802.16e uses fixed and mobile NLOS. There are two subcarrier permutation modes FUSC (full usage subcarrier) and PUSC (partial usage subcarrier). Uplink and downlink both can use different permutation modes in single application [10,11].

In uplink PUSC subcarrier permutation is given in Table 1 [11]. Same way in Downlink PUSC subcarrier permutation is given in Table 2. In Simulation depending on number of subcarriers and permutation scheme used parameters are chosen from the given tables.

<table>
<thead>
<tr>
<th>Table 1. PUSC permutation scheme for uplink [11]</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of sub channels</td>
</tr>
<tr>
<td>Data subcarriers used</td>
</tr>
<tr>
<td>Pilot subcarrier</td>
</tr>
<tr>
<td>Left–guard subcarriers</td>
</tr>
<tr>
<td>Right–guard subcarriers</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Table 2. PUSC permutation scheme for downlink [1]</th>
</tr>
</thead>
<tbody>
<tr>
<td>128 1024 2048</td>
</tr>
<tr>
<td>Number of sub channels</td>
</tr>
<tr>
<td>Data subcarriers used</td>
</tr>
<tr>
<td>Pilot subcarrier</td>
</tr>
<tr>
<td>Left–guard subcarriers</td>
</tr>
<tr>
<td>Right–guard subcarriers</td>
</tr>
</tbody>
</table>

Downlink PUSC permutation scheme is also used in simulation. Numbers of data, pilot and guard subcarriers required for scalable OFDMA are shown in the table.

Steps involved in multicarrier jamming simulation are described below.

1) Take the scenario as shown in Figure 4. Set the application which is to be examined under jamming effect. Select parameters for the same.

2) Now choose number of subscribers and base stations. Set their attributes according to application.

3) Set WiMAX attributes of all subscribers and base station. Set the symbol duration according to standards.

4) Set permutation mode as per the simulation criteria.

5) Now simulate the scenario and see the throughput and delay.

6) Take new scenario as shown in Figure 4 and repeat step number 1 to 4. Reduce number of data or pilot or guard subcarriers in uplink or downlink as per simulation criteria but keep symbol duration, subcarrier spacing and bandwidth same as the first scenario.

7) Now simulate this scenario and compare the throughput and delay result with previous result.

By following above steps multicarrier jamming effect can be simulated and results can be noted down.

5. Results

In first scenario two subscriber stations are there and jammer is moving towards subscriber stations and then moving away as show in Figure 2. This scenario is build to simulate single carrier jamming. The simulation time is taken by considering movement of jammer. The result shown in Figure 5 is BER performance under jamming for different modulation schemes.

Second result shown in Figure 6 is taken using the same scenario but the antennas used in subscriber stations are changed to see the impact antennas under jamming.

Figure 6 shows the BER vs. simulation time for different antenna used at receiver side under jamming effect. Jammer has isotropic antenna. Now new scenario is taken as shown in Figure 4 to simulate the effect of scalability property of OFDMA. In this scenario two subscriber stations and one base station are there. Any application
can be used to simulate multi carrier jamming. In this scenario video conferencing application is chosen. Number of subcarriers chosen for this application is 512 and bandwidth is 5 MHz for this application. The symbol duration is 102.86 micro seconds and subcarrier spacing is 10.94 KHz. Uplink and downlink both is using PUSC. Take new scenario as shown in Figure 4 and keep all the parameters same except number of subcarriers and bandwidth. Now choose 1024 subcarriers and 10 MHz bandwidth. The results for scalability property are shown in Figure 7 and Figure 8. Simulation time is set to 300 seconds. Throughput and delay of whole system is considered in the results. The system throughput is the sum of all data rates that are delivered to all terminals in a network. Delay is time taken by a packet to reach its destination starting from the time it leaves the source.

Now new scenario is taken as shown in Figure 4 to simulate multicarrier jamming effect. Sampling factor n is 28/25 and number of subcarriers are 1024. This scenario is run for 300 seconds. Bandwidth is 10 MHz, subcarrier spacing is 10.94 KHz same as second scenario. Data subcarriers in uplink PUSC and downlink PUSC are 560 and 720 respectively. Number of sub channels in uplink PUSC and downlink PUSC are 35 and 30 respectively.

Now follow the steps that are already mentioned in simulation section to simulate multi carrier jamming. Procedure to simulate multicarrier jamming is already mentioned. In uplink the effect of data subcarrier reduction is more compare to that in downlink. Results for uplink multicarrier jamming are shown in Figure 9 and Figure 10. Graph of throughput and delay vs. simulation time are shown.

Downlink is less sensitive to data subcarrier reduction. Here Figure 7 and Figure 8 shows the effect of scalability property of OFDMA on throughput and delay. The symbol duration and subcarrier spacing keep same for both 512 and 1024 subcarriers but the number of subcarriers and bandwidth is changed. Figure 9 and Figure 10 show the effect of number of data subcarrier reduction on throughput and delay in uplink. Throughput is decreased as numbers of data subcarriers are destroyed simulation result is shown in Figure 11. Delay is increased as the number of data subcarrier destroyed shown in Figure 12. In this paper only data subcarriers are destroyed and effect is noted down. If pilot subcarriers are destroyed then channel estimation becomes very difficult Multicarrier jamming effect on throughput and delay in downlink is shown below.
6. Conclusions

The performance of BPSK and QPSK is better than other modulation scheme under single carrier jamming. WiMAX sector antenna gives better performance than isotropic and WiMAX Omni antennas. Throughput is decreased when data subcarriers are destroyed in uplink and downlink. Throughput is reduced means BER is increased. Delay is increased when data subcarriers are destroyed in uplink and downlink. Uplink is more sensitive to data subcarrier reduction compare to downlink. Single carrier jamming and multicarrier jamming both are destructive in nature. Single carrier jamming can be detected easily compare to multicarrier jamming. The performance of the downlink PUSC is better than uplink PUSC under the influence of the multicarrier jamming. Multicarrier jamming effects are modeled in this paper by following particular algorithm which is already mentioned earlier. In this paper only simulated results are shown. In practical multicarrier jamming effect on throughput and delay remains same as shown by the simulated results. In practical case of multicarrier jamming antenna polarization and its pattern have to be considered. The paper shows results for fixed NLOS only. Mobile NLOS can be considered with some additional simulation parameters related to mobility under jamming effect.

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Design of an Acoustic Communication System Based on FHMA for Multiple Underwater Vehicles

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ABSTRACT

This paper designs an underwater acoustic communication system based on tamed spread spectrum and Frequency Hopping Multiple Access (FHMA) for multiple underwater vehicles. In this system, multiple underwater vehicles can communicate with the console on the water surface simultaneously successfully. The communication system is composed of tamed spread spectrum modulation and demodulation, frequency hopping modulation and demodulation with synchronization function, 4FSK modulation and demodulation and Rake Receiver. In order to make the system more feasible, underwater channel and the effect of user number on Bit Error Ratio (BER) are also taken into account. The simulation results indicate that it is feasible to use this system to accomplish underwater communication reliably for multiple users due to the low BER.

Keywords: Underwater Acoustic Communication, Tamed Spread Spectrum, FHMA, Multiple Underwater Vehicles

1. Introduction

As more and more attention is paid to the ocean, there are growing interests in exploring and exploiting the ocean these days. In order to realize some tasks such as collection of the information, monitoring of the environment, exploration of energy sources, forecast of natural disaster, assistant navigation in the ocean, underwater communication technology is necessary, which has become a very important scientific research field.

Because acoustic wave is adopted to actualize communication in the water instead of electromagnetic wave, it is called underwater acoustic communication. However, the complex environment in the ocean makes it rather difficult to communicate normally, such as the problems of high noise, multipath effect and large delay. So far, many researches have been made to ensure the communication reliability underwater, including the simulation and measure of the channel [1]; research on the use and algorithm of signal processor in the receiver [2]; diversity reception technique [3] and coding technique (compression coding and error correction coding) [4]. As the tasks that we require underwater vehicles to undertake grow more and more complicated, precise and various, only an underwater vehicle can hardly satisfy all the requirements. Multiple underwater vehicles system becomes one of the most important developmental directions. Our goal is to set up a communication network to ensure the vehicles to work under control normally. At present, multiple access based on spread spectrum technique is necessary to realize communication between multiuser. However, the limitation of frequency bandwidth in underwater acoustic channel results in low rate of original data [5-9]. Some researches focusing on high rate of underwater acoustic communication have been made [10,11], but how to put them to use in the situation of multiuser has not been reported yet. Reference [12] proposed a novel multichannel detection technique based on adaptive multichannel receiver to realize high data rate in multiuser underwater acoustic communication. But as a whole system, the communication quality was unknown (such as bit error rate) and the near-far effect wasn’t taken into account. In order to solve these problems, this paper proposes an underwater acoustic communication system for multiple underwater vehicles. Tamed spread spectrum and FHMA techniques are adopted to deal with the contradiction between the data rate and the limitation of frequency bandwidth. While the data rate is raised, the communication quality still keeps a high level.

This paper is organized as follows: Section 2 analyzes underwater acoustic channel, which is a very important influencing factor in communication system. Section 3 expounds the communication approaches, including the working principle, the components and the structure of
the system. The simulation results are shown in Section 4. Finally, the conclusions are made and the future research in this field is presented.

2. Underwater Acoustic Channel

2.1 Characteristics of Underwater Acoustic Channel

Underwater acoustic channel is the most complex communication channel that we have known about. It brings many difficulties to underwater communication. The characteristics of underwater acoustic channel are limitation of frequency bandwidth, varying multipath, fast fading and high noise.

The absorption and the diffusion of acoustic energy relate to transmission distance and frequency. That is to say transmission loss increases as transmission distance and frequency increase. This characteristic results in great attenuation of high frequency signal in long distance transmission. Reference [13] points out that the bandwidth can be larger than 100 kHz in near distance (< 1 km); in middle distance (1-20 km), the bandwidth is limited in the magnitude of ten kilohertz; the bandwidth is only about several kilohertz in long distance (20-2000 km).

There are many noise sources in the ocean. Some typical noise sources are listed as the frequency rises [14]. The effect of hydrostatic pressure caused by tide and wave; disturbance of earthquake; onflow; sailing ship; surface wave; thermal noise. For the frequency in the magnitude of ten kilohertz, a main noise source is surface wave. The high noise will cause the original signal difficult to recover.

Because of the reflection of the surface and the floor of ocean as well as the existent of reflectors and scatterers caused by the organisms, acoustic wave will reach the receiving part along several different paths after it is sent. This phenomenon is called multipath transmission. It’s a most important factor that affects the performance of underwater acoustic communication. Multipath transmission results in signal distortion (fast fading) and selective fading. The amplitude and the phase of the signal will change along with time and frequency which brings on errors in the receiving part. In order to solve the problem, equalization technique, diversity technique, spread spectrum technique and array technique can be adopted.

2.2 The Model of Underwater Acoustic Channel

Because underwater acoustic channel is very complex, it can’t be represented by a precise simulation model. Generally speaking, underwater acoustic channel is a kind of slow time-varying coherent multipath channel. In the length of coherent time, it can be simplified as a coherent multipath channel, which only has multipath effect. In this paper, a typical model of acoustic ray is adopted to implement the simulation. The model is expressed as

\[
\sum_{k=1}^{L} a_k(t) s(t - \tau_k(t)) + n(t) \tag{1}
\]

where \( L \) is the number of multipath; \( a_k(t) \) is the time-varying attenuation factor in the \( k \)th transmission path; \( \tau_k(t) \) is the delay; \( n(t) \) is additive white noise.

3. Communication Approaches

Because the underwater environment is rather complex, it is very important to decide the structure and the method of communication to ensure the reliability. In this section, the working principle and the structure of the system will be expounded in detail.

3.1 Tamed Spread Spectrum Communication

Developing from Direct Sequence Spread Spectrum (DSSS), tamed spread spectrum technique is widely used in communication field in recent years with the characteristics of resisting interference, great security and large capacity. Different with DSSS, tamed spread spectrum accomplishes spread spectrum by way of encoding (\( N, k \)). A binary data of \( k \) bits is expressed by a sequence of \( N \) bits Pseudo Noise (PN) code. The spreading gain of \( G = N/k \) is smaller than that of DSSS, and may not be integer, which is very suitable to the limitation of the frequency bandwidth in underwater acoustic communication. There are \( 2^k \) states in a binary data of \( k \) bits; accordingly, \( 2^k \) sequences of PN code are necessary. This means that these sequences need to have good characteristics of autocorrelation and cross correlation. In other words, \( 2^k \) sequences of PN code should be orthogonal.

Figure 1 is the model of tamed spread spectrum communication system. The signal after encoded can be expressed as

\[
\{a_k\} \rightarrow \text{K bits Shift Register} \rightarrow \text{Encoder} \rightarrow \{b_N\}
\]

Figure 2. The model of tamed spread spectrum communication system
\[
b(t) = \sum_{i=0}^{\infty} c_i(t - iT)
\]

where \(c(t)\) is the PN code, and \(j\) is decided by the weight of \(a_k(t): j = \sum_{i=0}^{\infty} a_i 2^i\), \(a_i(t) = \sum_{m=0}^{N-1} a_m g_m(t - mT)\), \(g_m(t)\) is gate function, \(T = kT_c\) is the period of the PN code.

**Figure 3** is the framework of the receiver. There are \(2^k\) paths to execute correlation processing to the \(2^k\) sequences. Because there is only one correlator that is correlated with the received signal in a period of the PN code, the \(k\) bits data in this path can be recovered if the output of the correlator is higher than the threshold, and the demodulation is accomplished.

After correlation processing in the \(l_{th}\) path, the output of the correlator is

\[
y_i = \int_0^T r(t)c_i(t)\cos\omega dt\]

where \(r(t)\) is the input of the correlator, \(c_i(t)\) is the reference code of the \(l_{th}\) path. \(y_i\) is composed of signal component and noise component, and the signal component is

\[
y_i = \frac{1}{2T} \int_0^T c_i(t)c_j(t)dt
\]

It is the correlation function of the PN code. If the PN code is orthogonal, the maximum of the autocorrelation value is \(T(m = f)\), and the cross correlation value is 0 \((m \neq f)\).

### 3.2 FHMA System

Controlled by the PN code, the carrier frequency of Frequency Hopping (FH) system hops continually and randomly. Compared with DSSS, it has the characteristics of high utilization of the frequency band and solves the problem of near-far effect. They are very important in underwater acoustic communication system, especially in the condition of multiple users.

**Figure 4** is the framework of frequency hopping communication system. In the transmitting part, the original signal modulates the carrier produced by the frequency synthesizer, which is controlled by the PN code. After one frequency hopping to another, it is very difficult to keep phase coherence. Therefore, FSK and ASK that can realize noncoherent detection are adopted as modulation method.

In the receiving part, the signal after processed by the amplifier will be sent into the mixer. In order to achieve de-spread spectrum of FH, the outputs of the frequency synthesizer should be identical with that in the transmitting part. That is to say, the synchronization of the PN codes in the two parts is necessary. Through mixing, the frequency of the signal will be fixed, which can be demodulated, and the signal is recovered finally. For the undesired signal, it doesn’t know the hopping regularity, so the frequency is not correlated with the outputs of the frequency synthesizer in the receiving part. Thus, it can hardly bring on interference to the FH system.

Based on FH, we adopt FHMA to realize underwater acoustic communication simultaneously between multiple users. In the FHMA system, the bandwidth is divided into several channels. The carrier frequency will continually hop along with the time instead of fixed to one channel. The hopping regularity is decided by the PN code of each user. The random hopping of the carrier frequency of each user results in the possibility of multiple accesses in a large frequency extent. In the receiving part, the same as FH, the PN code of each user should keep synchronous with that in the transmitting part. The key point in FHMA system is that the PN code of each user should be mutually orthogonal so as not to affect each other.

### 3.3 Receiving Technique of Multipath Signals

Diversity is a technique which is to reduce the influence of signal fading caused by multipath effect. The basic idea is to divide the received multipath signals into uncorrelated signals, and then to combine the signal energy according to some rule to obtain the energy of the useful signals as much as possible. In this way, the harmful multipath signals are turned into useful signals. The method can maximize the energy of the received useful signals in order to improve Signal to Noise Ratio (SNR).

**Figure 4. The framework of frequency hopping communication system**
Because of the large multipath delay and the dispersedness of energy in underwater acoustic channel, tap automatic adjusted Rake Receiver is a better choice. Different from tap fixed Rake Receiver, the delay of each correlator in tap automatic adjusted Rake Receiver is adjustable and it is dependent on the estimation made by the multipath searching module. The working principle of Tap automatic adjusted Rake Receiver is shown in Figure 5.

In the figure, the output \( \hat{z}(t) \) is

\[
\hat{z}(t) = \sum_{i=1}^{L} c_i(t) z_i(t)
\]

where the weighted coefficient \( c_i(t) \) is expressed as

\[
c_i(t) = \frac{z_i^2(t)}{\sum_{n=1}^{L} z_n^2(t)} \quad i = 1, 2, \ldots, L
\]

Maximum ratio combination is selected as combination rule. This method combines the signals based on the SNR of each path. Larger weight is assigned to the path with larger SNR. The SNR and the processing gain \( G_m \) of the combined signals are as follows:

\[
\text{SNR} = \sum_{i=1}^{L} c_i \cdot \text{SNR}_i
\]

\[
G_m = L
\]

3.4 Synchronization

Synchronization is a very important part in a communication system. Only when the receiving part keeps synchronous with the transmitting part, the system can work normally. Many kinds of synchronization need to be considered in a spread spectrum communication system, such as carrier synchronization, bit synchronization and frame synchronization. In this paper, hopping frequency synchronization is mainly discussed.

The requirements of FH synchronization in underwater acoustic communication include quick establishment, high precision, high reliability and easy realization. According to the requirements and taking the limitation of frequency bandwidth into account, we adopted an improved waiting-type self-synchronization method based on matched filter. The working principle is shown in Figure 6.

In this method, one matched filter is set to work at the frequency \( f \) firstly, waiting for the corresponding frequency \( f' \) of the FH signal form the transmitting part. When \( f' \) is detected, the frequency controller will begin to control the matched filter based on the FH pattern. At the same time, synchronization decision is executed. If the received FH sequence is identical with that in the receiving part (decision value is larger than the threshold), that means the synchronization acquisition is successful. The group of matched filters will continue working and the synchronization tracking begins. Otherwise (decision value is smaller than the threshold), the synchronization acquisition is finished by mistake. In this situation, the frequency controller will stop controlling the matched filters and shift \( f \) by a chip to the previous frequency point.

The probability density of the synchronization for the first time is

\[
\rho_1(t) = \frac{1}{N T_e - T_i}
\]

where \( T_i \) is the waiting time for the first time, \( N \) is the length of the FH sequence, \( T_e \) is the detention time at each FH point.

If the synchronization fails for the \((n - 1)th\) time, the probability density of the synchronization becomes

\[
\rho_n(t) = \frac{1}{N T_e - T_n} , \quad T_n = (n-1)T_e + \sum_{m=1}^{n-1} T_m
\]

Figure 6. Working principle of the improved waiting-type self-synchronization method
This equation indicates that the more the times of failure are, the more the probability density of the synchronization is.

For synchronization tracking, a representative tracking technique of FHSS signal is feasible, which is shown in Figure 7 [15].

### 3.5 Structure and Model of the Communication System

The framework of the communication system is shown in Figure 8. It is composed of several underwater vehicles and a console on the surface of water. All the underwater vehicles can communicate with the console simultaneously.

Figure 9 is the principle model of the communication system. In the transmitting part, the original data (digital binary sequence) is tamed spread firstly and then it is 4FSK modulated. Next, the signal is mixed by a series of other frequencies produced by a frequency synthesizer which is controlled by the PN code. Finally, the digital signal is turned into acoustic signal and emitted to the channel through the underwater acoustic transducer. Accordingly, the receiving part is composed of four parts: 1) Rake Receiver; 2) FH de-spread (with synchronization function); 3) 4FSK demodulate; 4) tamed de-spread. Based on the characteristics of underwater acoustic channel combined with FH system, 4FSK noncoherent detection is adopted as demodulation method in this paper.

This section will give the simulation results. The original data and the received data, the frequency spectrogram and the BER curve are included.
4. Simulation Results

This section will give the simulation results. The original data and the received data, the frequency spectrogram and the BER curve are included.

4.1 Design of the Parameters

In the simulation, the bit rate of original digital sequence is \( f_o = 650 \text{ bit/s} \). Taking the capability and the complexity of the system into account, we use Reed-Solomon (RS) [16] code to realize time spread spectrum and the parameter of RS code is \((15, 5)\). In this situation, the spread factor is 3 and the bit rate after time spread spectrum is \( f_s = 1.95 \text{ kbit/s} \). In the 4FSK modulation part, the modulator inputs one of the four frequencies every \( T_f = 1.026 \text{ ms} \), which means the modulator will change the frequency after transmitting every two bits. In order to avoid interference between neighboring channels in FHMA system, the minimum frequency interval should be

\[
\Delta f = n f_d \quad (n = 1, 2, 3, \ldots)
\]

(11)

where \( f_d \) is the bandwidth of the signal. In this paper, the interval between the four frequencies of 4FSK is \( \Delta f = 2/T_f = 1.95 \text{ kHz} \) and the interval between the frequencies from the frequency synthesizer in the FH part is \( \Delta f' = 4\Delta f = 7.8 \text{ kHz} \). Considering the frequency bandwidth in underwater acoustic communication, we use four frequencies produced by the frequency synthesizer to combine with the four frequencies of 4FSK to form 16 hopping frequencies which is listed in Table 1. The spread factor of FH part is 16, and the total gain of the system is \( \log (3 \times 16) = 16.8 \text{ dB} \).

In the FH part, Gold code is selected as the PN code. Gold code is a kind of combined codes of \( m \) sequence, which is obtained by modular two additions of optimum pairs of two \( m \) sequences with the same length, the same rate and the different codes. Its characteristics of autocorrelation and crosscorrelation are very good and the number of available sequences is much more than that of \( m \) sequence. The FH pattern of the system is shown in Figure 10.

The transmission distance is 1 km from the users to the console on condition that the sound velocity is 1460 m/s. Also we suppose that there are three transmission paths in the communication channel. One is direct path and the other two are reflection paths. The transmission delay of the two reflection paths are 0.096 s and 0.168 s respectively. The transmission losses of the three paths are 9.4 dB, 10.7 dB and 11.7 dB respectively.

For the convenient of simulation, we suppose there are two users in the system. In fact, because of the good autocorrelation and crosscorrelation of Gold code, the number of users has little influence on the communication effect.

4.2 Graph of the Signal

Figure 11 and Figure 12 show the original and the received signal of the communication system. From them we notice that when the SNR is \(-10 \text{ dB}\), the original signal can be recovered correctly in the receiving part and the two users can communicate simultaneously and successfully.

![Figure 10. The FH pattern of the communication system](image)

![Figure 11. Binary sequence of User 1](image)
Figure 13, Figure 14, Figure 15, Figure 16 show the transforming process of the signal frequency spectrogram (User 1 and User 2) in the transmitting part and the channel. From Figure 13 and Figure 14 we can see that after tamed spread, the frequency spectrogram is spread. Sixteen FH points are shown clearly in Figure 15. Figure 16 indicates that the original signal is submerged in the noise when it is transmitting in the channel. However, due to the adoption of techniques such as spread spectrum, channel coding and Rake receiver, the energy of the noise is reduced in the receiving part and the original signal is recovered.

4.3 Discussion of BER

BER is a most important index to estimate the performance of a communication system. In fact, because of the existence of the channel and the noise caused by hardware itself, there are always errors in the communication. What we could do is to reduce the errors to a certain degree according to the requirement so that they can’t affect the basic communication. Generally speaking, in the instruction communication system, the requirement of BER is very strict ($< 10^{-6}$). Correspondingly, in the voice and image communication system, the requirement of BER is lower ($10^{-3}$-$10^{-5}$). The relationship between BER and SNR of the communication system designed in this paper is shown in Figure 17. From the figure we notice that the BER is less than $10^{-4}$ when the SNR is $-14$dB. The original bit rate is 650 b/s, it can carry out some data and simple voice communication and the BER level is acceptable in practice.

Figure 18 shows the effect of user number on BER. It indicates that BER grows as user number grows on average because of the interference between each other. But on the whole, the interference is very low due to the good correlation of PN code.
5. Conclusions

This paper designs a kind of underwater acoustic communication system based on tamed spread spectrum and FHMA for multiple underwater vehicles. Aiming at the characteristics of limitation of frequency bandwidth and high noise in underwater acoustic channel, two key techniques of tamed spread spectrum and FHMA are adopted in order to ensure the communication reliability. Besides, Rake Receiver is also an important part in the system to reduce the influence caused by multipath effect. Meanwhile, the synchronization of FH part and underwater channel are also discussed. The simulation results indicate that this system could enable multiple vehicles to communicate simultaneously successfully. The system could realize some data and simple voice communication, which is acceptable in practice. The theory and the results in this paper have referenced value in the research of underwater acoustic communication.

Next work is to set up duplex communication links between multiple users and the console in order to ensure the equipments can transmit and receive simultaneously at any time, which is a very important problem in the underwater acoustic networks.

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Design of an Acoustic Communication System Based on FHMA for Multiple Underwater Vehicles


Cognitive Radio Sensing Using Hilbert Huang Transform

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ABSTRACT
Vast segments of the frequency spectrum are reserved for primary (licensed) users. These legacy users often under-utilize their reserved spectrum thus causing bandwidth waste. The unlicensed (secondary) users can take advantage of this fact and exploit the spectral holes (vacant spectrum segments). Since spectrum occupancy is transient in nature it is imperative that the spectral holes are identified as fast as possible. To accomplish this, we propose a novel adaptive spectrum sensing procedure. This procedure scans a wideband spectrum using Hilbert Huang Transform and detects the spectral holes present in the spectrum.

Keywords: Empirical Mode Decomposition (EMD), Ensemble Empirical Mode Decomposition (EEMD), Intrinsic Mode Function (IMF), Noise Assisted Data Analysis (NADA), Cognitive Radio, Guard Band, Spectrum Sensing

1. Introduction
Wireless networks, till date, are regulated by fixed spectrum allocation policies to operate in a particular time frame, certain frequency bands and are also constrained to geographical regions. Recent trends show that certain radio bands are overcrowded while some are moderately or sparsely used. In order to utilize the spectrum optimally and efficiently, cognitive radio technology has been proposed as a potential communication paradigm [1]. Cognitive radios are defined by the Federal Communications Commission (FCC) [2] as radio systems that continuously perform spectrum sensing, dynamically identify unused spectrum, and then operate in those spectral holes where the licensed (primary) radio systems are idle. In CR networks secondary users are allowed to utilize unoccupied bandwidths provided they do not cause interference to primary users. Many have worked on the sensing of a wideband spectrum. The main functions of cognitive radios are [3]:

Spectrum Sensing: refers to detect the unused spectrum and sharing it without any interference with other users, by sensing spectrum holes. Spectrum sensing techniques can be classified into three categories, namely 1) Transmitter detection by inspecting the spectrum generated, 2) Cooperative detection, which pools information from multiple cognitive radio users and, 3) Interference based detection, by using an interference temperature model.

Spectrum Management: Having known the spectral holes, it has to decide on the best spectrum band to meet the Quality of Service requirements over all available spectrum bands

Spectrum Mobility: As the objective is to use the spectrum in a dynamic manner they should enable transition to better spectrum by secondary users.

Spectrum Sharing: Involves spectrum scheduling for efficient spectrum usage.

In this paper transmitter detection method is used. Some authors used wideband spectrum joint detection for the presence of a primary user [4]. Some have used spatial diversity to find the presence of a primary user by sharing the information over a network [5]. Cognitive Radio is essential locally reuse unused spectrum by primary users (spectral holes) in order to increase the total channel capacity. Efficient spectrum utilization can be achieved by making a secondary user to access a spectrum hole created by the primary user at any location and time. The greatest task in sensing spectrum is in developing techniques which are able to detect even very weak primary user signals at the same time reasonably fast and low computational cost. This can be encountered by utilizing a novel technique called Hilbert Huang Transform [6].

2. System Model
2.1 Hilbert Huang Transform
Traditional data analysis methods are based on assump-
tions that the data is linear and stationary. In recent years new methods in the field of data analysis have been introduced. For example, wavelet analysis [7] and Wagner-Ville distribution [8,9] were designed for linear but non-stationary data. Additionally, various nonlinear time series analysis methods [10-12] were designed for nonlinear but stationary and deterministic systems. Unfortunately, in most real systems, natural or man-made, the data is most likely nonlinear and non-stationary. Analyzing the data of such a system is very difficult because the most sophisticated basis function cannot be relied as a basis function. Thus an adaptive basis function should be data defined. The Hilbert Huang Transform [6,13-15] seems to meet few of the above faced challenges.

The Hilbert Huang Transform is an empirically based data analysis method. Its basis of expansion is adaptive so that physical meaning can be derived from the nonlinear and non-stationary processes. The Hilbert Huang Transform consists of two parts: empirical mode decomposition (EMD) and Hilbert spectral analysis (HSA). We have incorporated a variant of the EMD called Empirical Ensemble Mode Decomposition (EEMD) to deal with input signals that are contaminated with noise.

2.2 Empirical Mode Decomposition

The empirical mode decomposition method is vital to deal with data from non-stationary and nonlinear process. The decomposition is based on the assumption that any data consists of different simple intrinsic modes of oscillations. Each intrinsic mode, linear or nonlinear, represents a simple oscillation, which will have the same number of extrema and zero-crossings. Furthermore, the oscillation will also be symmetric with respect to the “local mean.” At any given time, the data may have many different coexisting modes of oscillation, one superimposing on the others. The result is the final complicated data. Each of these oscillatory modes is represented by an intrinsic mode function (IMF) with the following definition:

1) In the whole dataset, the number of extrema and the number of zero-crossings must either equal or differ at most by one, and
2) At any point, the mean value of the envelope defined by the local maxima and the envelope defined by the local minima is zero.

An IMF represents a simple oscillatory mode as a counterpart to the simple harmonic function, but it is much more general: instead of constant amplitude and frequency, as in a simple harmonic component, the IMF can have a variable amplitude and frequency as functions of time.

IMFs can be generated by following the steps:
1) Identify all the local maxima and join these points using a cubic spline to give an upper envelope
2) Repeat the above procedure for local minima’s to give lower envelope. (Figure 1)
3) The mean of the envelopes is designated as m1.
4) The difference between the data and mean is the first component h1.
5) h1 is treated as a proto-IMF. In the next step h1 is treated as the input data and this procedure continues until the new component satisfies the IMF definition. The input data is then sifted and decomposed and the procedure continues until we obtain a monotonic residue.

2.3 Ensemble Empirical Mode Decomposition

To overcome some of the shortcomings of EMD, noise assisted data analysis method (NADA) EEMD was incorporated [6]. EMD cannot deal with signals contaminated with white noise efficiently. Due to the presence of noise a consequence of signal intermittency occurs which is termed as Mode mixing. This consequence causes serious aliasing in time-frequency distribution and also makes the physical meaning of IMF unclear. The critical concept advanced here is based on the following observations:

1) A collection of white noise cancels each other out in a time-space ensemble mean; therefore, only the signal can survive and persist in the final noise-added signal ensemble mean.
2) Finite, not infinitesimal, amplitude white noise is necessary to force the ensemble to exhaust all possible solutions; the finite magnitude noise makes the different scale signals reside in the corresponding IMF, dictated by the dyadic filter banks, and render the resulting ensemble mean more meaningful.

2.4 Hilbert Spectral Analysis

After obtaining the IMF (Figure 2) components, Hilbert Transform is applied to each IMF component. This transform gives the amplitude and the frequency of the IMF components as functions of time. The frequency-time distribution of the amplitude is termed as “Hilbert amplitude spectrum” or simply “Hilbert spectrum” (Figure 3). Squaring the amplitude gives the energy spectrum (Figure 4) which is quite useful in determining the dominant signal among a series of signals.

![Figure 1. The data (blue) and the envelopes (green) defined by the local maxima and minima respectively. The mean (red) of then upper and lower envelope](Image)
The combination of the ensemble empirical mode decomposition and the Hilbert spectral analysis is also known as the “Hilbert–Huang transform” (HHT) for short. Empirically, all tests indicate that Hilbert Huang transform is a superior tool for time-frequency analysis of nonlinear and non-stationary data. It is based on an adaptive basis, and the frequency is defined through the Hilbert transform. Consequently, there is no need for the spurious harmonics to represent nonlinear waveform deformations as in any of the a priori basis methods, and there is no uncertainty principle limitation on time or frequency resolution from the convolution pairs based also on a priori basis.

Table 1 [6] shows that the Hilbert Huang transform is indeed a powerful method for analyzing data from nonlinear and non-stationary processes: it is based on an adaptive basis; the frequency is derived by differentiation rather than convolution; therefore, it is not limited by the uncertainty principle; it is applicable to nonlinear and non-stationary data and presents the results in time-frequency-energy space for feature extraction.

![Figure 1](image1.png)

**Figure 1. The signal to be analyzed**

<table>
<thead>
<tr>
<th>Basis</th>
<th>Fourier</th>
<th>Wavelet</th>
<th>Hilbert Huang</th>
</tr>
</thead>
<tbody>
<tr>
<td>Frequency Convolution</td>
<td>Global, uncertainty</td>
<td>Regional, uncertainty</td>
<td>Differentiation: Local, certainty</td>
</tr>
<tr>
<td>Presentation Energy-frequency</td>
<td>Energy-time-frequency</td>
<td>Energy-time-frequency</td>
<td></td>
</tr>
<tr>
<td>Nonlinear</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>Non-stationary</td>
<td>No</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Feature Extraction</td>
<td>No</td>
<td>Discrete: no Continuous: yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Theoretical Base</td>
<td>Theory complete</td>
<td>Theory complete</td>
<td>Empirical</td>
</tr>
</tbody>
</table>

**Table 1. Comparison of the various transforms**

2.5 Adopting HHT in Wideband Sensing

Hilbert Huang Transform has proven from time to time its efficiency in dealing with nonlinear and non-stationary input data signals. As mentioned earlier, the day to day signals that are encountered are signals that are nonlinear and non-stationary in nature. To deal with such signals Hilbert Huang Transform is a very good tool. In this paper we take a wideband spectrum of 1.5 GHz ranging from 250 MHz to 1.75 GHz. The input data signal is shown in Figure 1. In this paper we utilize this tool (HHT) to sense a wideband spectrum for the presence of any legacy (primary) signal. We apply the Hilbert Huang Transform to the wideband spectrum.

From the Figure 5 it can be seen that the frequency ranges of the input data signal is found.

2.6 Guard Band

Every primary user has an interference limit, after identifying the frequency bands of the primary user a guard band is allocated to each of the primary user. This guard band is a varying guard band based on the priority each

![Figure 2](image2.png)

**Figure 2. The IMF of the input data**

![Figure 3](image3.png)

**Figure 3. A frequency plot**
user gets. \( r \) is a variable that signifies the priority of a user. The guard band is formulated as: \( \text{Guard Band} = r \times (\text{fixed Guard band}) \). In this scenario we have taken the guard band as 30 MHz.

Each primary user is given a priority ranging from 0.1 to 1. Thus the final guard band a primary user is assigned lies between 3 MHz to 30 MHz. A high priority primary user gets a 30 MHz guard band whereas on the other hand a low priority user might get about 5 MHz of guard band.

In Figure 6 the red band represents the guard band. From the figure it can be seen that the thickness of the guard band varies for different primary users. This is governed by the priority variable \( r \).

When the frequency bands that are used have been found and the guard band attached to each primary user calculated, the frequency bands that are unused have to be found. (Shown in Figure 7)

![Figure 5. Shows the frequency the signal is present (1 for presence and 0 for absence of signal)](image)

![Figure 6. The guard (red) band and the primary user bands (blue)](image)

3. Conclusions

In this paper we have introduced Hilbert Huang Transform for wideband spectrum sensing in Cognitive Radio Networks. The basic strategy is to take into account the detection of primary users jointly rather than detecting narrow bands individually. We have presented a novel technique HHT, explained its various components and implemented it in the cognitive radio sensing module. The computational cost was found to have reduced considerably when compared to a traditional wideband spectrum sensing module. The authors utilized a cognitive radio network to deal with weak primary users. Due to the high accuracy of HHT this network can be dropped as HHT successfully detects weak primary users. We utilized SVMTool [17] to find the frequency bands used by the legacy users. The results from SVMTool and HHT were compared and it was found that there was 98.7% accuracy with an ambiguity of 0.0674 per token. The above results clearly show that the procedure successfully found the spectral holes and also decreases the computational cost in a cognitive radio.

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Sensor Placement to Improve the Positioning Performance Based on Angle of Arrival (AOA)

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ABSTRACT

In this paper, a method for sensor placement to improve the placement quality based on angle of arrival of signal in a specific area is proposed. The installation place of sensors may be constrained with specified boundaries. In this method, the criterion of maximum quality of placement is the Cramer-Rao bound. The generalized pattern search as an effective method is used to maximize error bound of the placement problem by angle of arrival. Better results are obtained in comparison with results of genetic algorithm. The derived results are compared from two aspects of run time and result quality.

Keywords: Sensor Placement, Positioning, Cramer-Rao Bound, Numerical Optimization

1. Introduction

Positioning based on receiving and processing of radio signals has military, communication, navigation and even biologic application in the today world. Thus, study of positioning quality and methods to improve it has also a considerable importance. In a first glance, many methods are came to mind to improve the quality of positioning, for instance, improving the efficiency of sensors, improving the efficiency of positioning algorithms and proper design for installation location of sensors are some examples of these methods. In this paper, a method for determination of sensor position to improve the positioning quality in a specific area is considered. It is notable that in this paper the sensors are also constrained in the sense of the installation location and their position is not eligible to be in any desired location in the plane.

Yang and Scheuring are analytically and in some special cases prove that to decrease the positioning error in regular ranges such as circle and sphere, the existing sensors must be located on the perimeter of circle or surface of sphere symmetrically [1,2]. The case of study in these papers, is a very special case of sensor placement, because in some cases, all margins of positioning area is not available. In real applications, it is not possible to freely position the sensors and some places are not permitted to be positioned by a sensor; thus, in sensor placement, the location constraints of sensors must be considered.

From another aspect, there are several criterions to define and measure the positioning quality. These criterions might be related to average, maximum or cumulative density function (CDF) of error in the considered area. After defining the quality criterion, a method to calculate it must be proposed. For this aim, a complete simulation of positioning system with a Mont-Carlo method or an approximation of positioning error with existing bounds might be used. In this paper, the criterion of maximum positioning error in the considered area is chosen which is approximated with Cramer-Rao bound and this criterion is minimized with changing place of sensor locations.

After defining the quality criterion, the optimization method for sensor placement must be studied. For this purpose, we arrange all sensor places in an array and employ a cost function to calculate the quality criterion for this array. Then, optimizing the obtained function (minimization of the error or maximization of quality) the problem of sensor placement is solved.

This paper is organized as follows: in Section 2 the positioning problem using angle of arrival is investigated and the Cramer-Rao bound is mentioned for it. In Section 3, the effect of sensor placement on positioning is studied. The proposed method for sensor placement is presented in Section 4. In Section 5, a method based on genetic algorithm as a reference method for comparison is introduced. In Section 6 the results are compared and finally the paper is summarized and concluded in Section 7.
2. The Placement Problem

In this paper, for investigation of proposed sensor placement method, the positioning problem with angle of signal is considered. This type of positioning is efficient in applications in which the sent signal of desired source is a pulse signal.

AOA equations for a source in place of \((x, y)\) and \(N\) sensors at locations of \((x_i, y_i)\) \(i = 1, \ldots, N\) are as follows

\[
a_i = \theta_i + n_i = \arctan\left(\frac{x_i - x}{y_i - y}\right) + n_i, \quad i = 2, \ldots, N
\]

where, \((x_i, y_i)\) denotes place of \(i\)th sensor and \(n_i\)'s are uncorrelated Gaussian noise with covariance matrix of

\[
\Sigma_n = \sigma^2 \begin{bmatrix}
1 & 0 & \cdots & 0 \\
0 & 1 & \cdots & 0 \\
\vdots & \vdots & \ddots & \vdots \\
0 & 0 & \cdots & 1
\end{bmatrix}_{N \times N}
\]

Cramer-Rao lower bound is obtained for positioning using AOA as follows [3]

\[
\text{CLRB}(x) = \sqrt{\text{trace}\left(J^T(x) \Sigma_n J^{-1}(x)\right)}
\]

where, operations of \(\sqrt{()}\), \(\text{trace}()\), \((\cdot)^T\) and \((\cdot)^{-1}\) are square root, summation of main diagonal of matrix, transpose of matrix and matrix inverse operation. Furthermore \(J(x)\) is Jacobian matrix of noiseless angle with respect to target position, which is defined as

\[
J(x) = \left[\frac{\partial}{\partial x} a_1(x), \ldots, \frac{\partial}{\partial y} a_N(x)\right]^T = \begin{bmatrix}
\frac{y - y_1}{r_1^2} & \frac{x - x_1}{r_1^2} \\
\frac{y - y_2}{r_2^2} & \frac{x - x_2}{r_2^2} \\
\vdots & \vdots \\
\frac{y - y_N}{r_N^2} & \frac{x - x_N}{r_N^2}
\end{bmatrix}
\]

where, \(i = 1, \ldots, N\) and \(r_i\) is distance between target to \(i\)th sensor. Moreover, operations of \(\frac{\partial}{\partial x}\), \(\frac{\partial}{\partial y}\) and \(\otimes\) shows derivation respect \(x, y\) and Kronecker product operation.

The Cramer-Rao lower bound is achievable when position of target respect to position of sensor is inappropriate. For example, the Gauss-Newton [4] method if it is converged, the error reaches to Cramer-Rao lower bound. Positioning methods such as factor graph [5] or Levenberg–Marquardt [6] have better convergence and reach to Cramer-Rao lower bound. So, this bound is appropriate approximation for showing positioning error in practical systems.

3. Effects of Sensor Placement

To investigate the effect and importance of sensor placement in positioning, a supposed scenario is considered. Figure 1 shows the lower bound of Cramer-Rao error for the specified area in upper half-plane with symmetrical sensor placement in lower half-plane. Standard deviation of error in angle of arrival recognition of signal in each sensor is supposed to be 50 meter. In Figure 1, the right bar, shows the amount of error in each point proportional to the color of that point on the screen.

For optimization in this case, the maximum of positioning error in each area which is the positioning quality index, will be calculated. In Figure 2 this index is minimized and the sensors are replaced to minimize the maximum of positioning error in the considered area. In this case, also it is assumed that the sensors are constrained to place in lower half-plane.
For the first case, in which the best symmetrical arrangement is chosen, the maximum error is 815.3 meters, while with optimal placement of sensors this error decreases to 766.1 meters which is an improvement of 6.05%. In practical applications, the requirements of federal communication commission considers a standard for the error which says that in the 65% of considered area the positioning error must be lower than 100 meters or in 95% of mentioned area the error must be lower than 300 meters. Considering maximum error index in an area means that in 100% of that area, the positioning error is lower than determined limit. Note that this design and calculation of maximum positioning error index using the approximation of Cramer-Rao bound is a theoretical calculation of maximum positioning error index using the lower than determined limit. This design and calculation of maximum positioning error index using the approximation of Cramer-Rao bound is a theoretical process, thus, if in an area, the positioning error in 100% of the points is lower than a determined amount, this will guarantee a certainty margin of 5% to satisfy the standard index of 95%.

The numerical method to calculate the cost function or the maximum positioning error in an area is using a grid of points inside the supposed area. In this method, for all points of this grid, the positioning error bound is calculated, and then these values in all points are compared and their maximums are chosen as the numerical amount of criterion. The numerical method of optimization, which will be considered in Section 4, minimizes this index by replacing the position of sensors. In this paper, to accelerate the calculation of cost function a grid of 40 × 40 points is employed, but the final results are reported using a grid of 200 × 200 points.

4. Propose Method for Sensor Placement

The proposed method to minimize the cost function is using generalized pattern search which will be described in brief. Respect to the comparison of next section, this method is less complex from two aspects of realization and efficiency compared with genetic algorithm; thus this method is a proper method for sensor placement which this application of the method has not been reported in literature, since now.

Generalized pattern search is method for numerical minimization of functions without the need for gradient or other derivatives of functions. In this method, with a start point for the position of sensor and using a series of specified patterns to replace the sensors and with a search radius, the enquiring points of initial one are investigated and in the case of finding a better point, the position of sensor is transferred to that point and the search is continued with applying an expansion factor and with a larger search radius. The number of used patterns in each stage is four times of the number of sensors means that each action of increase or decrease in latitude or longitude coordinates of each sensor is considered as a pattern. In this method, if a better point is not found, the search radius is decreased applying a condensation factor and the search stages are repeated. It is not easy to prove that there is no local minimum for a numerical problem such as our positioning problem, but respect to the simulation results, with a high certainty, there is no local minimum in this problem.

Although it is not necessary, to certify that the initial point is not too improper and the convergence happens with acceptable rate, we can use the best member of a random small population as the initial point. Table 1 shows the parameters used in the generalized pattern search. Respect to Table 1, the condensation factor of search radius is 1/4, while the expansion factor of search radius is 2. The default values are generally 1/2 and 2, but in this problem, for an accelerated convergence and converges to a precise solution in lower number of iterations.

5. Method Based on Genetic Algorithm

To optimize the defined cost function, genetic algorithm, as a strong numerical optimization method, can be employed. However, using genetic algorithm considerably increases the possibility to escape the local minimums [7, 8]. But for this certainty, a heavier calculation load is imposed compared with the proposed method. Note that, in genetic algorithm the selected values for parameters of algorithm must be proper. In this paper, due to the lack of references for the use of genetic algorithm in sensor placement, these parameters are selected via try and error method in some experiments.

Several genetic algorithm parameters which are used in this paper are provided in Table 2.

For the fraction of next generation which is born by crossover, the typical value is 0.8 and the values between 0.4 and 0.8 are acceptable. In this problem the value of 0.5 is selected with an aim to have more new members in each generation and thus a larger search area for the algorithm. Based on the performed simulations for parent selection function, competition method [7] has less final error respect to roulette wheel [8] and other methods and therefore is considered in simulations. Adaptive mutation and efficiency compared with genetic algorithm; thus this method is considered in simulations. Adaptive mutation...
Table 2. Different parameters of genetic algorithm

<table>
<thead>
<tr>
<th>Selected value</th>
<th>Parameter</th>
</tr>
</thead>
<tbody>
<tr>
<td>uniform</td>
<td>Function of primary population generation</td>
</tr>
<tr>
<td>Adaptive</td>
<td>Mutation function</td>
</tr>
<tr>
<td>competition</td>
<td>Selection function</td>
</tr>
<tr>
<td>4</td>
<td>Size of competition</td>
</tr>
<tr>
<td>25</td>
<td>Number of generation</td>
</tr>
<tr>
<td>10 times of number</td>
<td>Size of population in each generation</td>
</tr>
<tr>
<td>sensors</td>
<td>Number of elites transferred to next generation directly</td>
</tr>
<tr>
<td>5</td>
<td>Percent of each generation generated from crossover</td>
</tr>
<tr>
<td>Scattered</td>
<td>Crossover function</td>
</tr>
<tr>
<td>Rank</td>
<td>Fitness Scaling</td>
</tr>
<tr>
<td>Forward</td>
<td>Migration Direction</td>
</tr>
<tr>
<td>20</td>
<td>Percent of Migration</td>
</tr>
<tr>
<td>20</td>
<td>Interval of Migration</td>
</tr>
<tr>
<td>200 times of number</td>
<td>Maximum calculation number of cost function</td>
</tr>
<tr>
<td>sensors</td>
<td></td>
</tr>
</tbody>
</table>

Table 3. Simulation results of proposed method and genetic algorithm for maximum positioning error criterion (optimal value is accessible for maximum error of 766.1)

<table>
<thead>
<tr>
<th>Maximum error criterion in region</th>
<th>Genetic algorithm</th>
<th>Proposed method</th>
</tr>
</thead>
<tbody>
<tr>
<td>mean</td>
<td>907.2</td>
<td>785.5</td>
</tr>
<tr>
<td>Standard division</td>
<td>50.3</td>
<td>6.7</td>
</tr>
<tr>
<td>Division percentage from optimum point</td>
<td>18.42%</td>
<td>2.47%</td>
</tr>
</tbody>
</table>

Table 4. Results for mean error on the region for 100 times run of methods which are being compared (optimal value is accessible for mean error of 383.1)

<table>
<thead>
<tr>
<th>Mean error criterion in region</th>
<th>Genetic algorithm</th>
<th>Proposed method</th>
</tr>
</thead>
<tbody>
<tr>
<td>mean</td>
<td>401.2</td>
<td>385.1</td>
</tr>
<tr>
<td>Standard division</td>
<td>17.4</td>
<td>3.8</td>
</tr>
<tr>
<td>Division Percentage from Optimum point</td>
<td>4.72%</td>
<td>0.52%</td>
</tr>
</tbody>
</table>

In Table 4 mean of error for 100 times of run are reported for proposed methods in the mentioned region. Optimal value is obtained through minimizing mean of error (rather than maximum of error) on the proposed region but results of methods under comparison are calculated and reported with maximum error criterion.

Table 3 and Table 4 show that the proposed method performs better than genetic algorithm in the region from both view points of maximum and mean of positioning error criterion.

Comparing with the optimal results, maximum error of proposed method which optimization is performed on has 2.5% difference with optimal value while genetic algorithm has error rate of 18.42. Consequently proposed method has better performance from the view point of functionality. Moreover in 100 runs of both methods, standard deviation for results of proposed method is just 7.2; while this value equals 50.3 for genetic algorithm. This shows that the results from genetic algorithm have more variations than those of proposed method.

It can be concluded from Table 4 that if two methods are compared from the view point of mean error in the whole region, it will be concluded that proposed method has 0.84% error respect to optimal value which has a good superiority respect to 4.02 percent for genetic algorithm. Similar to Table 3, standard deviation resulted from 100 repetitions of both methods signifies suggested algorithm’s better (less) standard deviation respect to genetic algorithm.

Generally, it can be seen that the proposed method has better results respect to genetic algorithm from the view function is used due to constraint features of the problem in order to prevent problems related to authenticity of mutated offspring in each generation. For constrained optimization problems, it is possible to use uniform or Gaussian distributed mutation but the run time will increase due to infeasibility of some offspring.

6. Comparison Results

The results of these two methods are compared with results of genetic algorithm to evaluate proposed method. From computational complexity view point, proposed method has the same number of function evaluation as genetic algorithm but realization of proposed method sounds simpler than genetic algorithm. To investigate from the view point of positioning quality, the same scenario of Section 3 shown in Figure 1 is used with the difference that in this case 4 sensors are deployed in the lower half-plane. The results of Table 3 and Table 4 are obtained by 100 runs of two methods. It is necessary to note that different scenarios are considered in different simulations in order to compare two methods which have resulted fairly similar results to proposed ones.

In all simulations, the constraint of maximum function evaluation number is assumed 200 times of the number of sensors which leads to reasonable simulation time. In optimizations, cost functions are calculated with a grid of 40 × 40 while final result is calculated and reported with accuracy of 200 × 200 points in the region.

Both methods are evaluated and compared with the criterion of maximum positioning error. Mean and standard deviation and deviation percentage from optimum value are shown in the Table 3 for 100 times of test repetition.
point of error percentage, stability of results from different runs for two cases of maximum error and mean error.

7. Conclusions

In this paper, a method is proposed based on mutual utilization of random search and generalized pattern search for placement of sensors in positioning applications. This method has higher speed in runtime in addition to less realization complexity and has completely better results respect to genetic algorithm in a way that is very near to optimum. Although this method is investigated for a specific positioning problem, it is possible to generalize it for other positioning applications only with some changes such as in cost function.

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