ADAPTIVE CHANNEL CODING SCHEME USING FINITE STATE MACHINE FOR SDR

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ABSTRACT

This paper proposes and investigates a coding and decoding scheme to achieve adaptive channel coding using a Finite State Machine (FSM) for Software Defined Radio (SDR). Adaptive channel coding and decoding systems that can switch between different coding rates and error correcting capabilities in order to adapt to changing applications and environments are effective for SDR. However, in these systems, a receiver cannot always select the correct decoder which causes decoding errors, usually referred to as Decoder-Selection-Errors (DSE). We propose a trellis encoder estimation scheme that compensates for this problem. This scheme uses the circuit of FSM to limit the encoder transition and the Viterbi algorithm for maximum likelihood trellis encoder estimation. Computer simulations are applied for evaluating the DSE rate, the Bit Error Rate (BER) and Throughput of the proposed scheme in comparison with a conventional scheme.

1. INTRODUCTION

The wireless communication field has developed rapidly recently. Especially, Intelligence Transport System (ITS) is partially implemented and IMT2000 service has started, while fourth and fifth generation wireless communication are being researched. However, it is necessary to use the limited channel capacity more efficiently. Moreover, several terminals built for a large variety of systems are required. In order to address these problems it is required to pay attention to the SDR [1][2][3]. SDR calls for replacing the conventional circuitries with programmable digital circuitries. SDR is based on two major concepts: the universal radio and the adaptive radio. The universal radio can adapt to applications by replacing software, while the adaptive radio can adapt to changes in the environment via learning software. There are many applications for the SDR system. As is known in wireless communication, one such application is the multi-mode radio such as GSM, IS-95 and W-CDMA. Another is wireless local area networks such as 2.45 GHz band, ISM band and 20 GHz band services. In addition, there are radio wave monitoring systems, broadcasting systems, medical services and so on.

We focus on the adaptive radio which can adapt to changes in the environment. Specifically, we consider adaptive coding techniques where the encoder is switched according to temporal-only changes in the environment, as in conventional adaptive coding schemes [4][5]. In the proposed adaptive coding scheme we introduce reconstruction of the statistically and temporally changing environment with FSM to orient encoding switching. There are two benefits of our proposed scheme. Firstly a FSM can orient the maximum likelihood trellis encoder estimation and switching thereby estimating the encoder transitions without using the supplementary information. Secondly statistically steady and dynamic channels can be foreseen so as to switch the encoder and predict its transitions easily. This is the layered FSM.

So far, adaptive coding schemes require either additional information or that the decoder estimation is based solely on reliability. On the other hand, the aim of our study is to minimize the DSE. Conventional systems use additional information in order to select the right decoder. As these techniques imply a certain redundancy, systems have been developed, that operate without such supplementary information. This can for example be achieved by using reliability-based selection obtained with all Viterbi decoding results, as we have published recently [6][7][8]. However, with such a system it is hard to increase the distance between two encoders. Thus, we now propose the use of a trellis diagram to estimate which encoder was used. By using a FSM for the encoder selection we further reduce the estimation error.
Our evaluation shows that one should avoid using supplementary information.

At the same time there are problems in FSM. One problem is the gradual transition of the channel condition. It is not clear whether the proposed method of limiting the encoder transition by FSM is appropriate to support estimation of dynamic channels. The other problem is when there is no transition of channel condition. In the worst case, taking for granted that the channel changes from the best state to the worst state suddenly, and moves between two states. This method of limiting the encoder transition will deteriorate the BER and Frame Error Rate (FER) performance compared with the unrestricted encoder transition. We propose the layered encoder transition countermeasure for these problems. The layered encoder transition scheme selects several encoder transitions of FSMs that can adapt to changes in the channel condition. The proposed layered FSM system can select the optimum encoder by switching several FSMs, each with different encoder transitions.

Computer simulation is employed to evaluate the DSE rate, BER and throughput of the proposed scheme. The performance is compared with a conventional adaptive channel coding scheme based on reliability in Viterbi decoding with supplementary information about the used encoders.

2. ADAPTIVE CHANNEL CODING TECHNIQUE USING A FINITE STATE MACHINE

In this section, we propose an adaptive coding scheme using a FSM and Viterbi estimation for SDR.

2.1 System Model

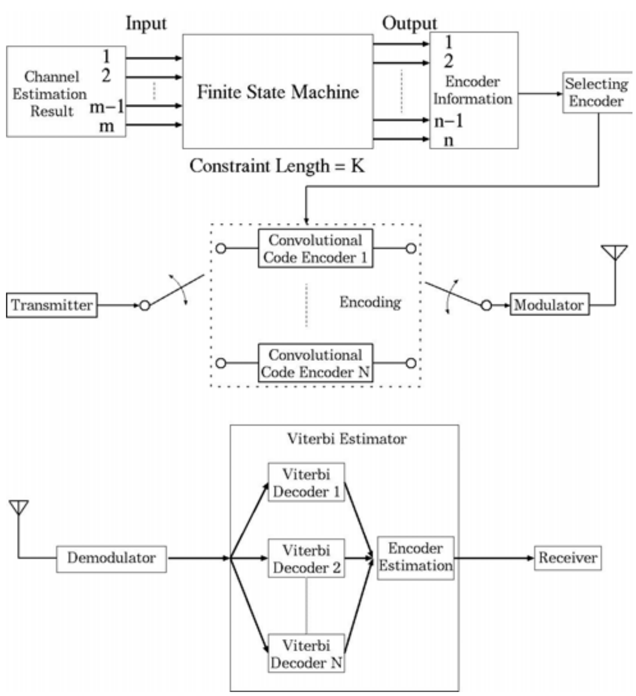


Fig. 1 System Model of Proposed Coding and Decoding Scheme

Fig. 1 shows an overview of our proposed system: The adaptive coding system uses N convolutional codes and Viterbi decoding. The upper part of Fig. 1 shows the method for switching encoders that depends on the channel condition. The bottom part shows the method to used estimate encoder transition. We assume the channel to be slow fading and the modulation to be Quadrature Phase Shift Keying (QPSK). Symbol rate is uniform (=600ksysmbol/s). Fading is assumed uniform in the frame of switching the encoder. The receiver side has knowledge of the number of encoders and the kind of encoder. The suggested FSM (Input/Output) selects the encoder from the estimated channel conditions.

Fig. 2 shows a flowchart of our proposed system: On the transmitter side, we transform the channel estimation results into useful channel information. This information is input to the FSM, which generates the encoder information to select the optimum encoder and gradual transition for the channel condition.
On the receiver side, we decode the incoming signal with all possible Viterbi decoders in parallel and then based on this decoding, the receiver estimates which encoder has been used. Additionally we use Viterbi estimation in the trellis estimation of the used encoder.

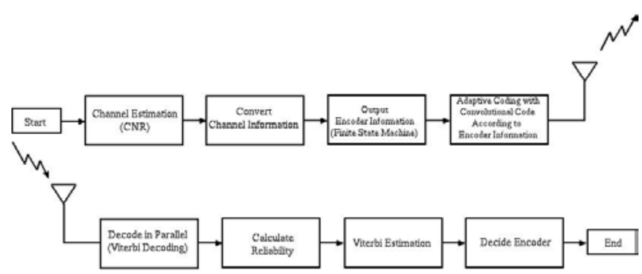


Fig 2 Flowchart of Proposed System

2.2 Estimation of Used Encoder Using Viterbi Decoding

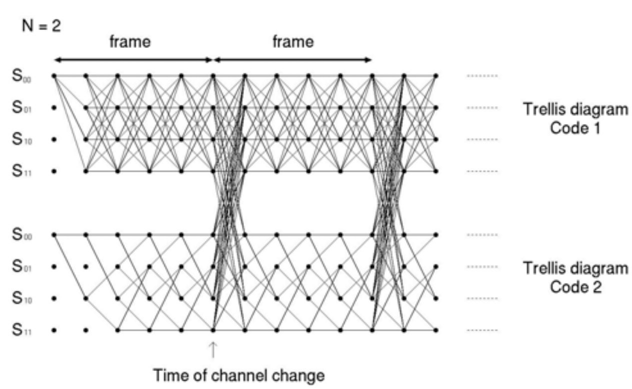


Fig 3 Trellis Diagram

This section describes how to estimate which encoder has been used when decoding, assuming one needs to know what kind of encoders have been used on the receiver side. Fig. 3 shows the trellis diagram for decoding the proposed system. We assume that the number of encoders is two. A large decoder which is the combination of all possible decoder's trellis diagrams for every switching section is used for decoding using the Viterbi algorithm. In this way, it can estimate which encoder has been used automatically. We assume that each encoder has the same number of states because the trellis diagram must combine every switching encoder section.

2.3 Estimation of Encoder Transition Using Viterbi Algorithm

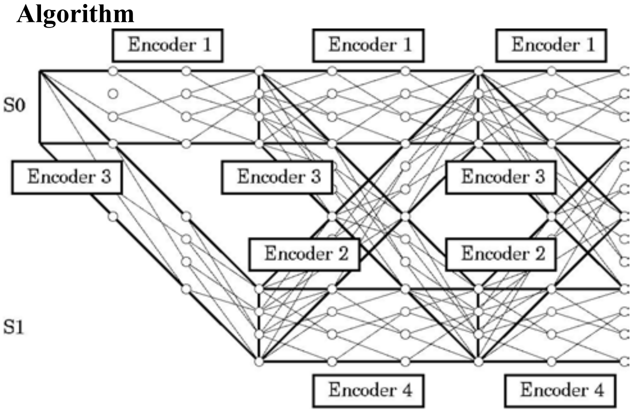


Fig 4 Trellis Diagram of Encoder Transition (Four Encoders)

This section explains how we estimate the encoder transition using the Viterbi algorithm. The encoder estimation result obtained by the proposed Viterbi decoding is converted to an encoder information estimate, called the observation, which corresponds to the encoder information at the output of the FSM. Using the observation, Viterbi estimation is used to estimate the maximum likelihood trellis encoder transition. The number of encoders is more than the number of changes in the channel level; it means that the path number of encoder transition is more than the number of channel transition. This fact turns out that it can increase the minimum distance of any pair of paths on encoder transition. Therefore, this scheme can increase quality and system performance better than an adaptive coding scheme which estimates the encoder using the Viterbi decoding with the same number of changing the channel level and encoders. When using a FSM, one must be careful that the switching encoder does not adapt to changing environments. Therefore, we must design the encoder transition of the FSM to adapt to changing channel conditions using the state transition table.

Fig. 4 that is according to three state transition diagrams of the Fig. 10 shows the trellis diagram of the encoder transition using these FSM (Fig.7, 8, 9). We assume that there are four encoders. The trellis decoding process is 3-dimensional, in the sense that the incoming signal is processed by interconnected trellises of the individual decoders. FSM can show the trellis diagram of the state transition and estimate the maximum likelihood encoder transition using the Viterbi algorithm. S0 and S1 are related to inside state 0 and 1. This trellis diagram can estimate the maximum likelihood encoder transition using the Viterbi algorithm and select the appropriate decoder.

3. LAYERED FINITE STATE MACHINE SYSTEM

In this section, we propose a Layered FSM system.

3.1 Problems in Finite State Machine

It is not clear whether this method of limiting the encoder transitions by FSM is an appropriate way to support the changing channel condition. In the worst case, taking it for granted that the channel changes from the best state to the worst state suddenly, and moves between two states. This method of limiting the encoder transition deteriorates the performance of BER and FER compared with an unrestricted encoder transition.

As a solution, we propose switching FSMs which have different encoder transitions to each other. This is the layered encoder transition scheme which selects several encoder transitions of FSMs to adapt to changing channel conditions. The method of the layered encoder transition is achieved using the FSM for switching, depending on the channel condition.

3.2 System Model

This section describes the system model of the layered FSM.

3.2.1 Transmitter Side

Fig. 5 shows the proposed layered finite state machine system. We assume that the number of channel levels and selecting encoders has a one-one relation on the transmitter side. This system can select the optimum encoder by switching between several FSMs which have different encoder transitions respectively and switch the several FSMs by using the FSM which is for selecting FSM from the estimated channel conditions. We transform the channel estimation results into useful channel information. This information is input to the FSM for selecting FSM, which generates the FSM information to select the optimum FSM. Then, we input the channel information to the selected FSM, which generates the encoder information to select the optimum FSM.

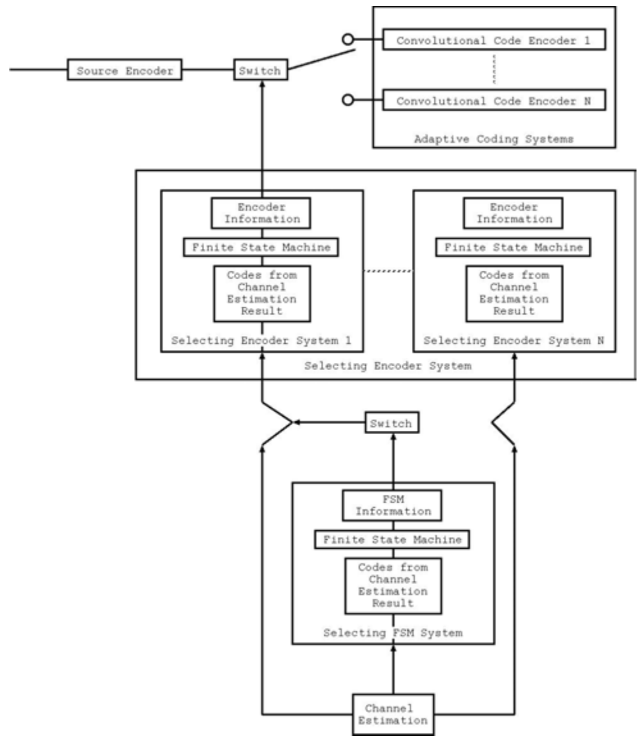


Fig 5 System of Controlling FSM and Encoders

3.2.1 Receiver Side

Fig. 6 shows the triple Viterbi algorithm which is used to decode the convolutional code, and that estimates the encoder transition and the FSM transition on the receiver side. Firstly, in Viterbi decoding, we use a trellis diagram, which is combined with several trellis diagrams of the codes. Then, it can estimate which encoder has been used automatically. Secondly, the encoder estimation result obtained by the proposed Viterbi decoding is converted to an encoder information estimate, called the observation, which corresponds to the encoder information at the output of the FSM. Using the observation, Viterbi estimation is used to estimate the maximum likelihood trellis encoder transition. Thirdly, the FSM transition estimation result obtained by using Viterbi estimation is converted to a FSM information estimate, called the observation, which corresponds to the FSM information at the output of the FSM for selecting FSM. Using the observation, Viterbi estimation is used to estimate the maximum likelihood trellis FSM transition.

4. NUMERICAL RESULTS

In this section, we investigate the performance of the proposed system. We evaluate the computer simulation of our proposed system.

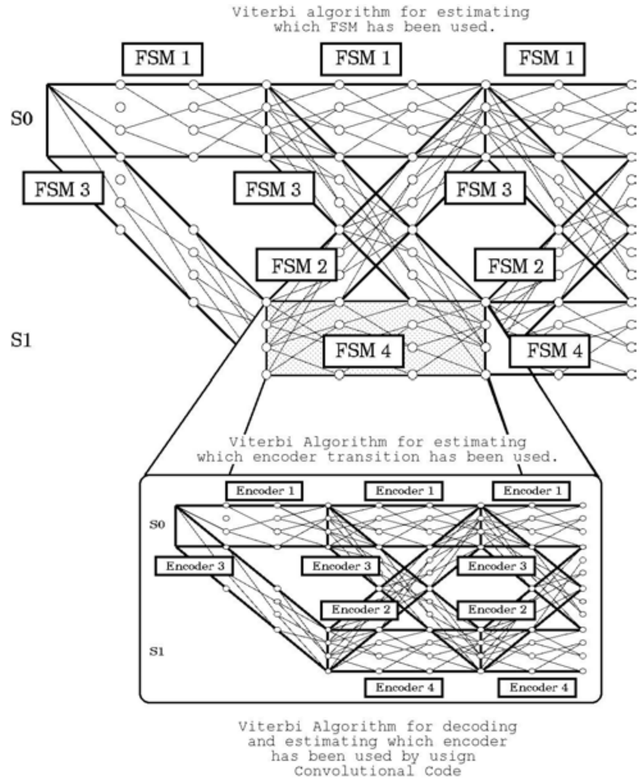


Fig 6 Estimating Scheme by Using Viterbi Algorithm

4.1 Compared Schemes

In order to evaluate the performance of the proposed scheme, we describe two compared schemes in this section. From here on, we mention especially in case that each scheme uses four encoders. However, those following schemes are applied to the cases of more encoders.

4.1.1 Method of Using the Supplemental Information (Comparison 1)

The method of using supplementary information is as follows. First, the supplementary information of all frames are gathered and transmitted. The supplementary information is encoded by the code employed to protect frames. In the receiver, the supplementary information is decoded and the received sequences are then decoded by switching trellises according to the decoded supplementary information. A frame is formed containing a preamble made up of repeated supplement information vectors (containing information on the encoder used), following by the actual encoded information. The receiver detests which encoder is used from this supplement information and decodes the information accordingly. The

DSE can be controlled by altering the order of redundancy with which the supplement information is transmitted, so as to match the DSE performance of our proposed scheme. This allows a good measure of throughput gain in the proposed scheme.

4.1.2 Method of Combined with Several Trellis Diagrams of the Codes in Viterbi Decoding (Comparison 2)

The method of combining several trellis diagrams of the codes in Viterbi decoding is as follows.

The trellis diagram, which is a combination of trellis diagrams of the codes, selects the decoder depending on the decoding result. This is the same technique utilized in the middle of the proposed system.

4.2 Conditions of Simulation

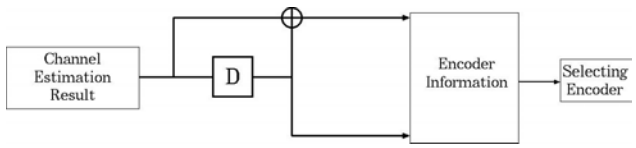


Fig 7 Circuit of Finite State Machine (Four Encoders)

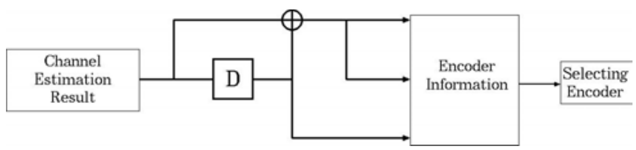


Fig 8 Circuit of Finite State Machine [Number of Output Data Increased] (Four Encoders)

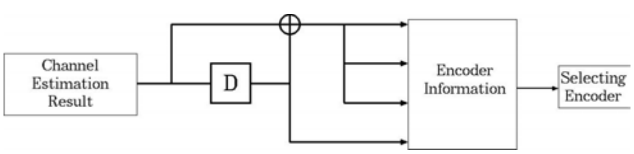


Fig 9 Circuit of Finite State Machine [Number of Output Data Increased] (Four Encoders)

Performance comparison of the proposed system uses the two schemes described. For our proposed system, switching the encoder uses the FSM in Fig. 7, 8, 9. The algorithm of the switching encoder using FSM is now described (is shown in Fig. 7, 8, 9). The circuit of the FSM used in our system. These three circuits select four encoders each. The input data is a code associated to the channel estimation results. The output data is the encoder specification. The FSM (Input/Output) therefore selects the encoder from the estimated channel conditions. If the output bits of FSM in Fig. 7, 8, 9 are increased, then the bits that are input to estimate the encoder transition at the receiver will be increased to 2, 3, 4 bits. Therefore, refer to Fig. 4, Fig. 10; the Hamming Distances between encoders are increased to 2, 3, 4 respectively. Similarly, the minimum Hamming Distance between the paths on trellis diagram of encoder transition is increased to 3, 5, 7. This affects accuracy of estimating encoder transition. Fig. 11 and Fig. 12 show the difference between the DSE performances corresponding to Fig. 7, 8, 9. Fig. 10 shows the state transition diagram. The Input and Output data of the FSM depend on the channel condition level or the channel condition and the inside state. Each states Input data is [0, 1]. The Output data for each state is [00, 11, 10, 01], [000, 111, 101, 010], [0000, 1111, 1011, 0100].

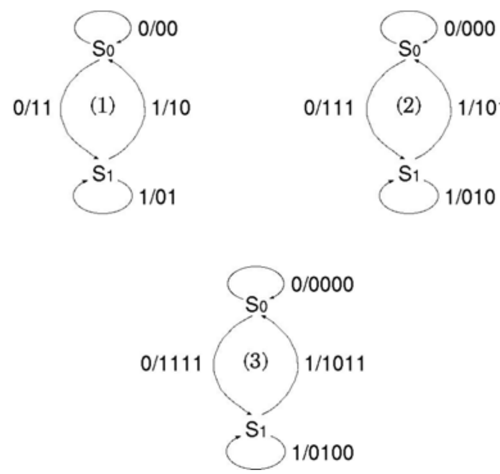


Figure 10 State Transition Diagram

The trellis diagram, which is a combination of trellis diagrams of the codes, selects the decoder depending on the decoding result.

It is over flat fading channel model. QPSK modulation and four convolutional encoders with different coding rates are used. The length of data frame is 300 bits. The encoders are switched every 300 encoded bits constantly. In each data frame therefore, the length of the encoded bits is same to the length of data frame. In the simplest terms, information is encoded to the same length (300 bits) with selected encoders which coding rates are different each other. Therefore, there is no delay between paths in Viterbi decoding and estimation of our proposed system. If DSE occurs, it might cause the out of synchronization of the information data, however, we leave this problem out of consideration in our paper.

There are limitations to consider both in the proposed scheme and the comparison schemes on both the transmitter and the receiver sides.

Symbol rate which inputs both the encoder and decoder is uniform (600ksymbol/s).

The receiver side is assumed to know what kind of encoder is used on the transmitter side

Fading level is constant in a frame of switching the encoder.

The channel level is known on the transmitter side.

We consider that the computational complexity can be achieved concerning the real processing time.

4.3 Decoder Selection Error Performance

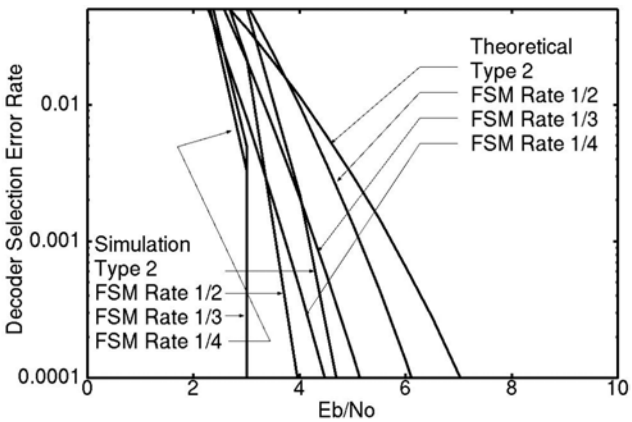
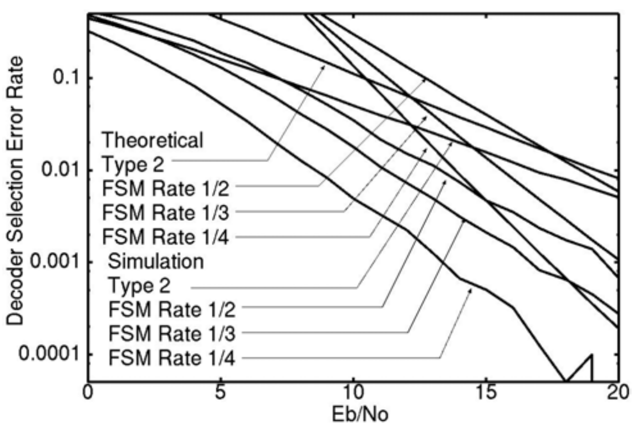


Fig 11 Theoretical Upper Bounds of DSE Under AWGN Conditions

Fig 12 Theoretical Upper Bounds of DSE Under



Rayleigh Fading Conditions

In this section, the DSE performance of the proposed scheme is investigated. If DSE occurs, the number of decoded data bits is not the same as the number of original input bits and burst errors are caused. Therefore, the performance of the proposed scheme depends on the DSEs. DSE rate is defined as:

DSER=NDSE/NSD (1)

NDSE is the number of DSE. NSD is the number of selecting decoders. The number of samples in this simulation is four million.

4.3.1 Upper Bound of DSE Performance

The theoretical upper bounds of DSE rate as well as simulation results are shown in Fig. 11, 12 in the case of AWGN and Rayleigh Fading conditions. From this result, it is confirmed that we can reduce DSE by using FSMs.

5. CONCLUSIONS AND FUTURE STUDIES

There are many different types of adaptive coding systems.

However, these systems have the problem of DSE.

We proposed an adaptive coding and decoding system using a Finite State Machine and Viterbi Estimation for Software Defined Radio.

Our simulations showed, that the proposed system achieves lower DSE rates than those based on reliability only. With respect to both BER and Throughput, our system out-performs the conventional solutions.

In our future studies, we shall concentrate on reducing the computational complexity of the scheme, to meet real time processing demands in future implementations. Moreover, we shall add other Adaptive Coding System, to enable adaptation to a wider range of scenarios.

6. REFERENCES

[1] Ryuji Kohno: ``Signal Processing and ASIC's for ITS Telecommunications--Spread Spectrum, Array Antenna and Software Defined Radio for ITS--," IEICE Transactions on Fundamentals Vol. E85-A No.3 pp.566-572 2002 3.

[2] J.Mitola: ``Globalization of Software Radios," IEEE Communications Magazine, vol. 37, no. 2, Feb 1999.

[3] D.L. Goeckel: ``Adaptive Coding for Time-Varying Channels Using Outdated Fading Estimates," IEEE Communications Magazine, pp. 844-855, vol. 47, no.6, June 1999.

[4] Hidehiro Matsuoka, Seiichi Sampei, and Norihiko Morinaga: ``Adaptive Modulation System with Punctured Convolutional Code for High Quality Personal Communication Systems," IEICE Transactions on Communications}, Vol.E79-B, No.3, pp. 328-334,1996 3.

[5] K. IKEMOTO, K. UMEBAYASHI, S. ISHII and R. KOHNO, ``Adaptive Coding and Decoding Schemes Using Finite State Machine for Software Defined Radio," Technical Report of IEICE, SR01--01, pp.1--8, (2001--04)

[6] K. IKEMOTO and R. KOHNO, ``Adaptive Coding Schemes Using Finite State Machine for Software Defined Radio," The 24th Symposium on Information Theory and Its Application (SITA'01), pp.-, (2001--12) (in Japanese)

[7] K. IKEMOTO and R. KOHNO, ``Adaptive Coding Schemes Using Finite State Machine for Software Defined Radio," Technical Report of IEICE, IT00--98, pp.165--172, (2001--03) (in Japanese)