A Novel Fuzzy Inference-Based Technique for Dynamic Link Adaptation in SDR Wideband Waveform

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***Abstract*— A new dynamic link adaptation technique based on a fuzzy inference system for hybrid multicode code division multiple access/time division multiple access wideband networking waveform of software defined radio is proposed. A constrained optimization problem is formulated under the quality of service, signal-to-noise ratio, and throughput constraints. The optimization problem is solved using the fuzzy inference system by dynamically changing the modulation technique and the number of multicodes assigned to each user. The proposed algorithm provides maximum possible data throughput as required by the user or application by reducing the packet retransmission overhead through the use of optimum waveform parameters decided through the fuzzy inference system. A novel contribution of the proposed algorithm is that it reduces the computational complexity and thus the power consumption by restricting the throughput to the required value even if the channel conditions are fair enough to allow higher throughput. Simulation results are presented to demonstrate the effectiveness of the proposed algorithm in achieving better throughput by efficiently reducing the packet retransmissions overhead.**

***Index Terms*— Multicode CDMA, link adaptation, fuzzy inference, SDR.**

I. Introduction

THE MODERN wireless networks demand high speed communication by ensuring specific Quality of Service (QoS) requirements. A substantial research has been carried out to investigate the potential of applying the Software Defined Radio (SDR) approach to achieve these goals [1]. Due to these requirements, the developments for the modern SDR-based wireless networks are moving toward wideband and digital signal based networking capable of providing adaptive and high speed communication [2].

To optimize the use of scarce SDR resources, efficient algorithms for resource allocation/utilization are required. This involves adapting the transmission parameters of the SDR waveform to changing channel conditions, QoS and data rate requirements. This process is called link adaptation. The requirement of link adaptation increases when the transmissions are in the form of network packets. A low value of bit error rate can drastically increase packet error rate and thus packet re-transmissions, if the waveform parameters are not dynamically changed according the channel conditions and tolerable bit error rate. A well-known link adaptation strategy is adaptive modulation and coding (AMC) [3], [4]. In AMC, the channel coding rate and modulation technique are changed according to the changing channel behavior. Many varieties of AMC strategies are possible [5]–[13]. Orthogonal multicode transmission which is primarily used to enhance the data rate in the 3rd Generation Partnership Project (3GPP) standard [3], [14], has also been used for link adaptation.

A link adaptation for High Speed Packet Data Access (HSPDA) is presented in [15] by adaptive modulation & coding, multicode transmissions and Hybrid Automatic Repeat Request (HARQ). The paper compares the throughput of MC-CDMA and DS-CDMA technologies using the proposed method. The average bit error rate performance of the AMC and multicode scheme for Nakagami fading channel is studied in [16]. A fundamental scheme for achieving variable data rates by changing the set of spreading sequences in multicode Code Division Multiple Access (MC-CDMA) is proposed in [17]. The expressions for Multiple Access Interference (MAI) have also been derived. This paper lacks the scheduling algorithm for multicode transmission. A scheduling algorithm for both the AMC and multicode transmission is proposed in [18], which maximizes the Carrier-to-Interference Ratio (CIR) to increase the throughput. For uplink CDMA system, the problem of maximizing the total throughput under a bit error rate constraint is investigated in [19]. The realization of variable data rate is achieved by parametrizing the number of signature waveforms (multicodes) and constellation points in Quadrature Amplitude Modulation (QAM) for each user. The solution is optimal and potentially complex. A sub-optimal approach of deriving the expressions for optimal resource allocation based on single user is proposed in [20]. The single user solution is then extended to form a sub-optimal sequential optimization procedure for multiple users. A new scheme for optimum selection of decode and forward (DF) relay jointly with link adaptation using adaptive modulation, coding and transmit power by fuzzy logic is proposed in [21]. The spectral efficiency is maximized by adapting the

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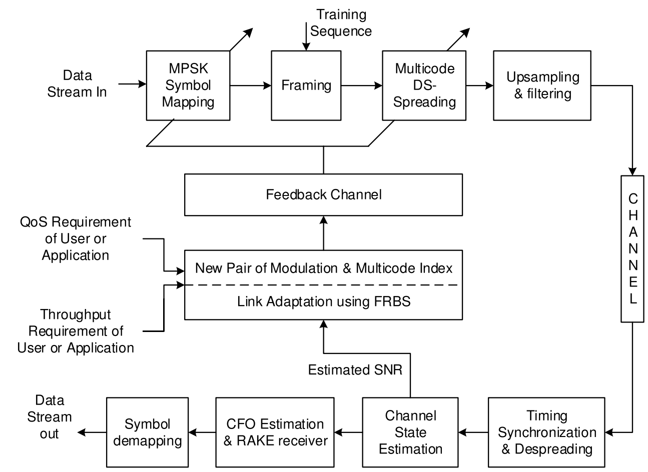

Fig. 1. Proposed link adaptation model using fuzzy rule based system.

power of the relays, modulation index and code rate by keeping BER less than the accepted level. Two adaptive modulation and coding schemes using fuzzy logic are proposed in [22] and [23] for OFDM systems. The required throughput is not considered as input to the fuzzy inference system.

In this paper, a novel algorithm for link adaptation using fuzzy rule based system (FRBS) for packet-based wideband networking waveform of SDR is proposed. The waveform uses multicode CDMA and adaptive TDMA as multiple access schemes. To reduce the packet re-transmissions overhead, the configurable system parameters need to be changed dynamically according to the channel conditions. Moreover, the varying QoS requirement of various applications is fulfilled by link adaptation by varying the system parameters at the physical layer and/or adaptive TDMA. The focus of this paper is on the first part with multicode CDMA at the physical layer. A novel scheme based on fuzzy inference is presented which is capable of selecting the most suitable parameters based on the heuristics. A constrained optimization problem with QoS and throughput constraints has been solved by dynamically changing the modulation technique and the number of multicodes. The proposed algorithm reduces the complexity and thus power consumption by restricting the throughput to the value required by user or application, even if the channel conditions are fair enough to allow higher throughput. Results have been presented to demonstrate the effectiveness of the proposed algorithm. It has been shown that the proposed link adaptation scheme achieves better throughput by efficiently reducing the packet re-transmissions overhead.

The proposed link adaptation scheme is shown in Figure 1. All the transmissions are in the form of bursts. At the receiver side, after the detection of each valid burst, multicode despreading operation is performed. Then channel estimation block includes both the estimation of Signal-to-Noise Ratio (SNR) and channel state, which are estimated for each burst. The proposed link adaptation algorithm generates a new pair of modulation and multicode indices for the next transmission through FRBS based on the estimated received SNR, QoS and throughput requirements.

II. SYSTEM MODEL

Considering a maximum of K users at the physical layer of wideband waveform, the spreading waveform corresponding to the m^{th} code assigned to k^{th} user is given by

$$\pi_{k,m}(t)=\sum_{n=0}^{G-1} c_{k,m}[n]g(t-nT_c) \quad m=0,1,...,I_{M,k}-1 \quad (1)$$

where $c_{k,m}[n]$ is the n^{th} sample of the m^{th} spreading code of k^{th} user, $I_{M,k}$ is the multicode index (i.e. number of codes) assigned to k^{th} user. The transmitted signal of k^{th} user is given by

$$s_k(t)=\sum_{i}\sum_{m=0}^{I_{M,k}-1} d_k[I_{M,k}(i-1)+m]\pi_{k,m}(t-iT) \quad (2)$$

where $d_k[.] \in e^{j2m_k\pi/M_k}$ are the M-PSK symbols of k^{th} user, M_k is the modulation index and $m_k=0,1,...,M_k-1$. After passing through the multipath fading channel, the received composite continuous-time baseband signal is

$$\begin{aligned} r(t)=&\sum_{k=1}^{K} e^{j2\pi\Delta f_k t}\cdot\sum_{i}\cdot\sum_{m=0}^{I_{M,k}-1}\cdot\sum_{l=1}^{\gamma_k}\alpha_{kl}d_k \\ &\times[I_{M,k}(i-1)+m]\pi_{k,m}(t-iT-\tau_{kl})+w(t) \quad (3)\end{aligned}$$

and $f_{D,kl}$, α_{kl} and τ_{kl} are the Doppler spread, constant complex path gain (considering $BT \le T_{coh}$, where B is the number of symbols per burst, T is the symbol duration and T_{coh} is the coherence time) and delay spread corresponding to the l^{th} path and k^{th} user, respectively, γ_k is the number of multipath and Δf_k be the k^{th} user's Carrier Frequency Offset (CFO). Let the waveform $q_{k,m}(t)$ for k^{th} user's m^{th} spreading code be defined as

$$q_{k,m}(t)=\sum_{l=1}^{\gamma_k}\alpha_{kl}\pi_{k,m}(t-iT) \quad (4)$$

so that considering first burst of data, (3) becomes

$$\begin{aligned} r(t)=&\sum_{k=1}^{K} e^{j2\pi\Delta f_k t}\cdot\sum_{i=0}^{B-1}\sum_{m=0}^{I_{M,k}-1}\cdot\sum_{l=1}^{\gamma_k}\alpha_{kl}d_k \\ &\times[I_{M,k}(i-1)+m]q_{k,m}(t-iT)+w(t) \quad (5)\end{aligned}$$

After sampling the received signal given by (5) at N_s/T_c, timing and frequency synchronization is achieved. Efficient algorithms for the timing and frequency synchronization of the proposed wideband waveform have been proposed in our earlier work [24], [25]. The goal of the proposed link adaptation scheme is to dynamically change the modulation index M and multicode index I_M through FRBS which takes QoS and throughput requirements and estimated received SNR as inputs. The first two inputs (i.e. QoS and throughput) are either specified by user or as required by a specific application, whereas the received SNR needs to be estimated at the receiver. Several SNR estimation algorithms are present in the literature, e.g. see [26]–[28]. The proposed link adaptation algorithm uses higher modulation index M up to 16. Due to this reason, a moment-based algorithm for estimating SNR for higher order modulation given in [28] is used in this paper. The estimation performance of this estimator is superior to other existing moment-based algorithms.

III. Optimization Problem

The optimization problem for link adaptation is presented. First, we present the mathematical framework for the derivation of analytical formula for the throughput of the proposed wideband waveform. Using the available RF bandwidth B_{RF}, the maximum allowed sampling rate (f_s) (chips/sec) is calculated. Note that f_s is the rate at which the spreader block generates the data. The burst detection block takes the input data at the same rate. The sampling rate (f_s) is given as

$$f_s = \frac{2B_{RF}}{B_{norm}N_s} \tag{6}$$

where N_s is the upsampling factor and B_{norm} is the normalized bandwidth calculated from the normalized two-sided Power Spectral Density (PSD) of the transmitted data. Once f_s is found, the overall throughput for k^{th} user is given as

$$R_k = \frac{f_s I_{M,k}\log_2(M_k)}{G}.\frac{N_d}{N_0 + N_d} \tag{7}$$

where M_k is the modulation index and $I_{M,k}$ is the multicode index of the k^{th} user, respectively, N_d is the number of data symbols/burst, N_0 is the number of symbols in training sequence and R_k is the throughput of k^{th} user. The effective throughput of any waveform depends on the both the waveform parameters and channel conditions. The waveform parameters include RF bandwidth, modulation scheme, number of samples per symbol, forward error correction etc. For a wideband waveform based on multicode CDMA, these parameters also include spreading gain and multicode index. Wideband networking waveforms suffer heavily by re-transmissions overhead due to erroneous received packets in case of harsh channel conditions. This overhead reduces the effective throughput. The aim of a good link adaptation scheme should be to provide maximum possible data throughput as required by a user or application by reducing the packet re-transmissions overhead through the use of optimum waveform parameters. The computational complexity and thus the power consumption need to be reduced by restricting the throughput to the required value even if the channel conditions are fair enough to allow higher throughput. Thus, the problem of link adaptation at the physical layer of wideband networking waveform is formulated as

$$\max R_k = \frac{f_s I_{M,k}\log_2(M_k)}{G}.\frac{N_d}{N_0 + N_d} \tag{8}$$

subject to the constraints

$$\text{BER} \le \text{BER}_{max}$$

$$R_k = \begin{cases} R_{max}, & \text{if } R_{max} \le R_{k,req} \\ R_{k,req}, & \text{if } R_{max} > R_{k,req} \end{cases}$$

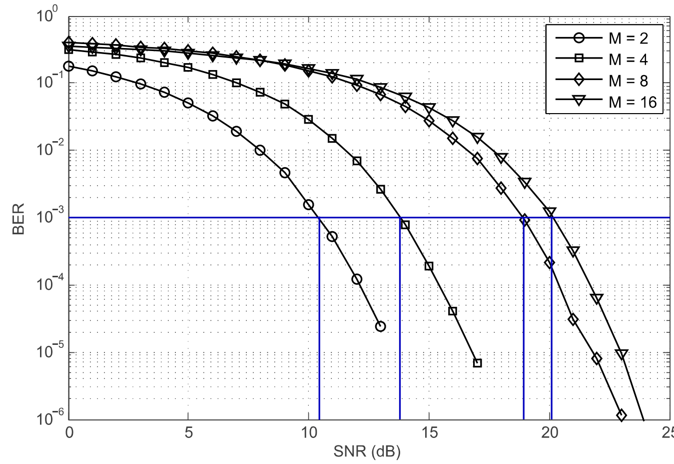

Fig. 2. BER Performance for various values of multicode index $I_M = 8$ and modulation indices (M) varying from 2 to 16.

where

B_{RF} = RF bandwidth in MHz

BER = Bit Error Rate

BER_{max} = Maximum allowed BER as per QoS

$R_{k,req}$ = Throughput required by k^{th} user/application

R_{max} = Maximum possible throughput

We have assumed $10^{-4} \le \text{BER}_{max} \le 10^{-1}$ and 0.25Mbps $\le R_{k,req} \le$ 17.5Mbps. Note that the effective throughput depends on packet error rate. As packet error rate increases, the effective throughput is decreased due to the increase in number of packet re-transmissions.

IV. Data Acquisition for Fuzzy Rules Formation

The first step of the proposed algorithm is the acquisition of data from BER performance curves of the system for all the possible modes of operation. BER performance of the system is analyzed for all the possible pairs of modulation and multicode indices. Figure 2 shows an example with a set of BER curves for a multicode index $I_M = 8$ and modulation indices (M) varying from 2 to 16. In similar way, BER curves for all other pairs are obtained through simulation. The data is then acquired from the set of BER curves by drawing horizontal line (for a specific QoS requirement) for each BER curve and noting the intersection point. This will give the minimum SNR that guarantees the BER to be within maximum allowable value along with the achievable throughput for a specific Modulation and Multicode Index (MMI) pair. This process is repeated for the complete set of BER curves obtained through simulation to obtain the data acquisition table I. Table I shows the minimum SNR values (in dBs) that guarantee the BER to be within maximum allowable value. Note that the subscript k is dropped for simplicity.

V. Proposed FRBS for Link Adaptation

In this section, we present the procedure for creating the proposed FRBS which selects the optimal pair for the

TABLE I
DATA (SNR VALUES) ACQUIRED FROM THE BER CURVES

I_M	BER_{max}	$M=2$	$M=4$	$M=8$	$M=16$
1	10^{-1}	-	0.35	5.02	5.6
	10^{-2}	1.28	4.87	9.85	10.68
	10^{-3}	3.76	7.07	12.2	13.2
	10^{-4}	5.31	8.63	13.72	14.8
2	10^{-1}	-	3	7.6	8.27
	10^{-2}	3.9	7.43	12.42	13.45
	10^{-3}	6.38	9.7	14.82	15.97
	10^{-4}	7.83	11.21	16.43	17.48
4	10^{-1}	1.06	5.23	9.95	10.53
	10^{-2}	6.15	9.67	14.74	15.7
	10^{-3}	8.55	12.02	17.14	18.2
	10^{-4}	10.3	13.45	18.68	19.78
8	10^{-1}	2.85	7.1	11.73	12.46
	10^{-2}	8.01	11.55	16.58	17.64
	10^{-3}	10.41	13.8	18.93	20.15
	10^{-4}	12.2	15.41	20.4	21.72
16	10^{-1}	4.2	8.4	13.03	13.85
	10^{-2}	9.38	12.92	17.89	19.1
	10^{-3}	11.81	15.18	20.3	21.59
	10^{-4}	13.32	16.56	21.77	23.14

next transmission. The fuzzy logic depends heavily on human thinking in spirit and is much closer to natural language as compared to conventional logic systems. By adjusting the input signal, the fuzzy logic controller executes similar actions as a human operator executes. A typical fuzzy system comprises of these stages; Fuzzification, Rule Base and Defuzzification. In Fuzzification stage, these inputs are first converted to fuzzy numbers. The next stage, rule base, generates fuzzy number of the compensated output signal by using the fuzzified input variables. The fuzzy numbers corresponding to the compensated output signal are converted to crisp values in the defuzzifier stage. Collectively, these three stages are referred to as Fuzzy Inference System (FIS). The conventional adaptive modulation systems using ordinary hardware decision making circuits are mostly inefficient to decide or change modulation scheme and/or other parameters according to given conditions. The reason is that the input values must be exact for the selected output parameters to be accurate. On the other side, using fuzzy logic in decision making interface makes the system more efficient and simple as the inputs are now treated as vague. For example, low SNR, high throughput, normal QoS etc. The performance is improved because each output MMI pair is selected by combining the effect of more than one membership functions of each input variable by using implication and aggregation. In link adaptation without fuzzy logic, the parameters are selected based on a one-to-one mapping.

A. Selection of Fuzzy Sets

After completing the data acquisition from BER curves, fuzzy sets and the corresponding Membership Functions (MF) are now selected to cover the complete range of inputs and outputs. The three inputs to the proposed FIS are QoS requirement (taken as negative logarithm of BER, denoted as $nLogBER$) throughput requirement ($R_{k,req}$) and estimated SNR. The generated output is MMI pair. Triangular membership function is

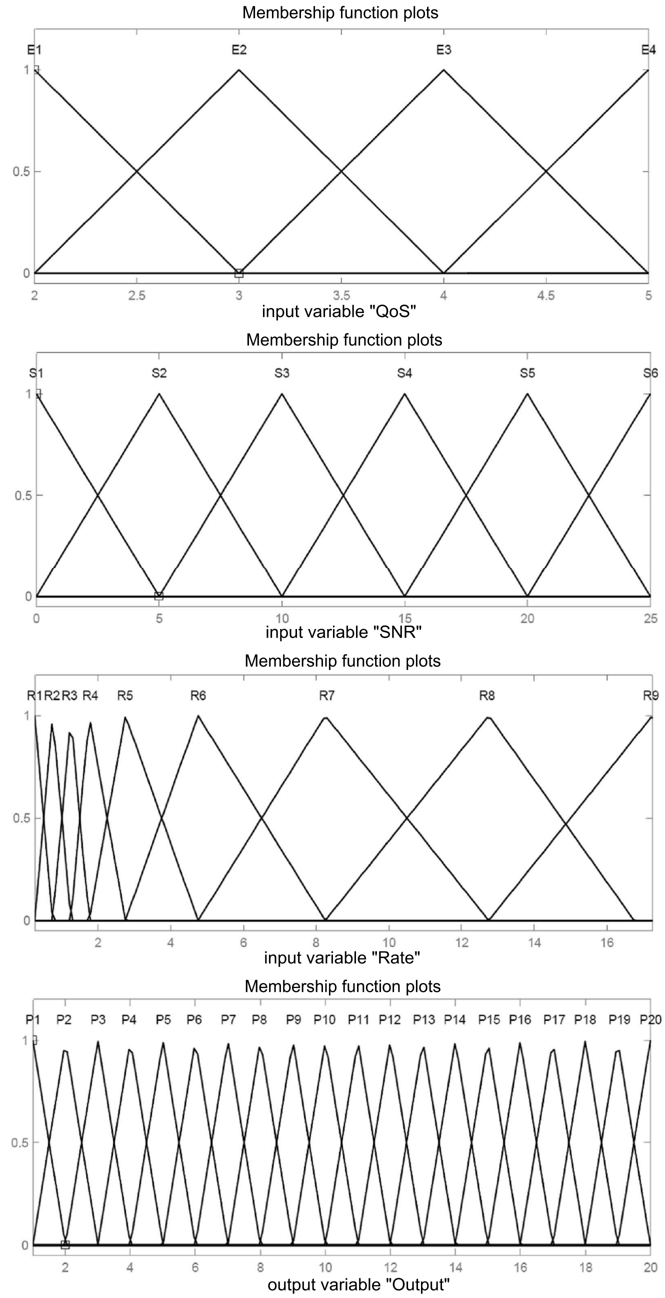

Fig. 3. Membership functions of the inputs and output of the proposed FIS using fuzzy systems toolbox.

used with min-max (and-or) as implication and aggregation operation. The triangular membership function $M(x)$ is with endpoints $(a, 0)$ and $(b, 0)$ and the high point (c, α) is given as

$$M(x) = \begin{cases} \alpha\left(\frac{x-a}{c-a}\right), & a \le x \le c \\ \alpha\left(\frac{x-b}{c-b}\right), & c \le x \le b \\ 0, & \text{otherwise} \end{cases}$$

The third input ($R_{k,req}$) uses non-uniform triangular MF, whereas the other two inputs and the output use the standard uniform triangular MF. The MFs of the inputs and output of the proposed FIS are shown in Figure 3. Sufficient number of fuzzy sets are used to represent the inputs and output.

TABLE II

OUTPUT MMI PAIRS (MODULATION INDEX, MULTICODE INDEX) FOR THE 20 FUZZY SET VALUES

Value	MMI Pair	Value	MMI Pair	Value	MMI Pair	Value	MMI Pair
P_1	(2,1)	P_6	(4,2)	P_{11}	(8,4)	P_{16}	(16,8)
P_2	(4,1)	P_7	(8,2)	P_{12}	(16,4)	P_{17}	(2,16)
P_3	(8,1)	P_8	(16,2)	P_{13}	(2,8)	P_{18}	(4,16)
P_4	(16,1)	P_9	(2,4)	P_{14}	(4,8)	P_{19}	(8,16)
P_5	(2,2)	P_{10}	(4,4)	P_{15}	(8,8)	P_{20}	(16,16)

The input variable $nLogBER$ is simply calculated as

$$nLogBER = -\log(\text{BER})$$

The number of fuzzy sets used for the inputs are 4, 6 and 9 for $nLogBER$, received SNR and $R_{k,req}$, respectively. For the output MMI pair, the possible values of M are 2, 4, 8, 16 and the possible values of I_M are 1, 2, 4, 8, 16. So, the maximum 20 possible fuzzy sets are used for the output MMI pair.

B. Fuzzy Rule Matrix for Modulation & Multicode Indices Selection

The Fuzzy Rule Matrix (FRM) is the main processing stage of any FIS. It is based on a collection of logic rules in the form of IF-THEN statements. The 'IF' part of the rule is called 'antecedent' and the 'THEN' part of the rule is called 'consequent'. As mentioned earlier, the FIS of the proposed system takes three inputs (QoS requirement, throughput requirement and estimated SNR) and generates an output Modulation and Multicode Index pair. Let the total number of fuzzy rules be N_R and $\{E_1, E_2, E_3, E_4\}$, $\{S_1, S_2, \ldots, S_6\}$, $\{R_1, R_2, \ldots, R_9\}$ and $\{P_1, P_2, \ldots, P_{20}\}$ be the fuzzy set values for the required $nLogBER$, SNR, $R_{k,req}$ and the output MMI pair, respectively.

The output MMI pairs for the 20 fuzzy set values are shown in Table II. Based on the number of fuzzy set values used for the inputs and output of the FIS, a total number of $N_R = 216$ fuzzy rules are formed using the acquired data from the BER curves. Some rules have same antecedent (IF part) but different consequents (THEN parts). Such rules are called conflicting rules. For conflicting rules, the consequent resulting in higher throughput is selected. Similarly, if two or more consequents result in same throughput, the consequent with lower value of modulation and/or multicode indices is selected since it will result in lower computational complexity and thus lower power consumption. If some input/output pairs are not available in the acquired data, then those parts are filled by human intuition or expert knowledge. For example, if an MMI pair fulfills the given specifications at a lower SNR, then it is certainly valid for higher SNR. If the throughput specification is not achievable by the available MMI pairs under a given QoS requirement, then the MMI pair resulting in the highest possible throughput is selected. The complete FRM is given Table III.

C. Defuzzification

In the fuzzification stage, all the three inputs are converted into fuzzy numbers $\mu_{nLogBER}$, μ_{SNR} and $\mu_{R_{k,req}}$. The fuzzy

TABLE III

HUMAN INTUITION BASED FRM FOR MMI PAIR SELECTION

QoS	SNR	Required data rate ($R_{k,req}$)								
		$\boldsymbol{R_1}$	$\boldsymbol{R_2}$	$\boldsymbol{R_3}$	$\boldsymbol{R_4}$	$\boldsymbol{R_5}$	$\boldsymbol{R_6}$	$\boldsymbol{R_7}$	$\boldsymbol{R_8}$	$\boldsymbol{R_9}$
$\boldsymbol{E_1}$	$\boldsymbol{S_1}$	P_1	P_5	P_5	P_5	P_5	P_5	P_5	P_5	P_5
	$\boldsymbol{S_2}$	P_2	P_6	P_9	P_{13}	P_{17}	P_{17}	P_{17}	P_{17}	P_{17}
	$\boldsymbol{S_3}$	P_3	P_4	P_7	P_8	P_{11}	P_{18}	P_{18}	P_{18}	P_{18}
	$\boldsymbol{S_4}$	P_3	P_4	P_7	P_{10}	P_{12}	P_{15}	P_{16}	P_{19}	P_{20}
	$\boldsymbol{S_5}$	P_3	P_4	P_7	P_{10}	P_{12}	P_{15}	P_{16}	P_{19}	P_{20}
	$\boldsymbol{S_6}$	P_3	P_4	P_7	P_{10}	P_{12}	P_{15}	P_{16}	P_{19}	P_{20}
$\boldsymbol{E_2}$	$\boldsymbol{S_1}$	P_1	P_1	P_1	P_1	P_1	P_1	P_1	P_1	P_1
	$\boldsymbol{S_2}$	P_1	P_5	P_5	P_5	P_5	P_5	P_5	P_5	P_5
	$\boldsymbol{S_3}$	P_2	P_3	P_{10}	P_{13}	P_{17}	P_{17}	P_{17}	P_{17}	P_{17}
	$\boldsymbol{S_4}$	P_2	P_4	P_7	P_8	P_{11}	P_{18}	P_{18}	P_{18}	P_{18}
	$\boldsymbol{S_5}$	P_2	P_4	P_7	P_8	P_{12}	P_{15}	P_{16}	P_{19}	P_{20}
	$\boldsymbol{S_6}$	P_2	P_4	P_7	P_8	P_{12}	P_{15}	P_{16}	P_{19}	P_{20}
$\boldsymbol{E_3}$	$\boldsymbol{S_1}$	P_1	P_1	P_1	P_1	P_1	P_1	P_1	P_1	P_1
	$\boldsymbol{S_2}$	P_1	P_1	P_1	P_1	P_1	P_1	P_1	P_1	P_1
	$\boldsymbol{S_3}$	P_5	P_6	P_6	P_6	P_6	P_6	P_6	P_6	P_6
	$\boldsymbol{S_4}$	P_2	P_4	P_7	P_{10}	P_{14}	P_{14}	P_{14}	P_{14}	P_{14}
	$\boldsymbol{S_5}$	P_2	P_4	P_7	P_8	P_{11}	P_{15}	P_{18}	P_{18}	P_{18}
	$\boldsymbol{S_6}$	P_2	P_4	P_7	P_8	P_{11}	P_{15}	P_{16}	P_{19}	P_{20}
$\boldsymbol{E_4}$	$\boldsymbol{S_1}$	P_1	P_1	P_1	P_1	P_1	P_1	P_1	P_1	P_1
	$\boldsymbol{S_2}$	P_1	P_1	P_1	P_1	P_1	P_1	P_1	P_1	P_1
	$\boldsymbol{S_3}$	P_1	P_5	P_5	P_5	P_5	P_5	P_5	P_5	P_5
	$\boldsymbol{S_4}$	P_2	P_3	P_{10}	P_{10}	P_{17}	P_{17}	P_{17}	P_{17}	P_{17}
	$\boldsymbol{S_5}$	P_2	P_4	P_7	P_8	P_{11}	P_{18}	P_{18}	P_{18}	P_{18}
	$\boldsymbol{S_6}$	P_2	P_4	P_7	P_8	P_{11}	P_{15}	P_{16}	P_{19}	P_{20}

inference stage generates a fuzzy output value Δu by using the FRM. This fuzzy value is then defuzzified in to a crisp value in the form of MMI pair number. For defuzzification, Centroid of Area (COA) method [29] is used in the proposed FRBS. For discrete number of fuzzy rules the defuzzified output $\Delta u(k)$ by COA method will be

$$\Delta u(k) = \frac{\sum_i \mu_{R_i}(\Delta u)\Delta u(R_i)}{\sum_i \Delta u(R_i)} \tag{9}$$

where $\Delta u(R_i)$ is the representative crisp value corresponding to the peak value of the membership degree of the fuzzy set which is an output from the FRM for the rule R_i.

VI. SIMULATION RESULTS

This section presents the simulation results of the proposed link adaptation algorithm. The RF bandwidth is taken to be $B = 8$ MHz. The roll-off factor (α) for the pulse shaping filter is 0.65, whereas the spreading gain (G) is 16. The proposed algorithm is applicable to the systems with a maximum of K active users at the physical layer in the current time slot, where $K \leq G$. Note that if $K = G$ in a single time slot, then I_M will always be unity which results in a maximum data throughput of 1.08 Mbps. The aim of the proposed link adaptation algorithm is to provide either

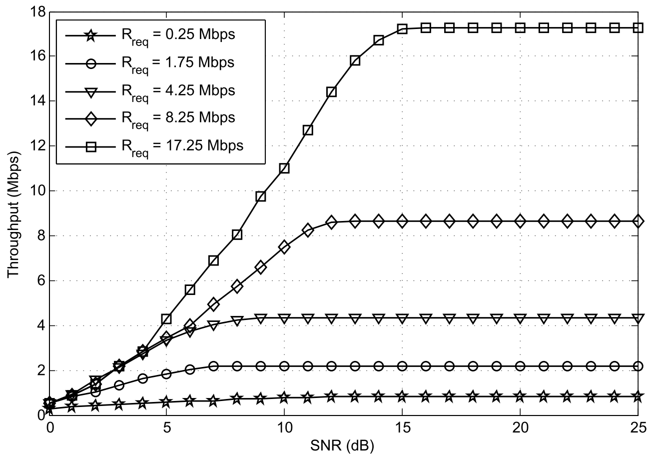

Fig. 4. Throughput performance of the proposed algorithm for QoS demand of 10^{-1}.

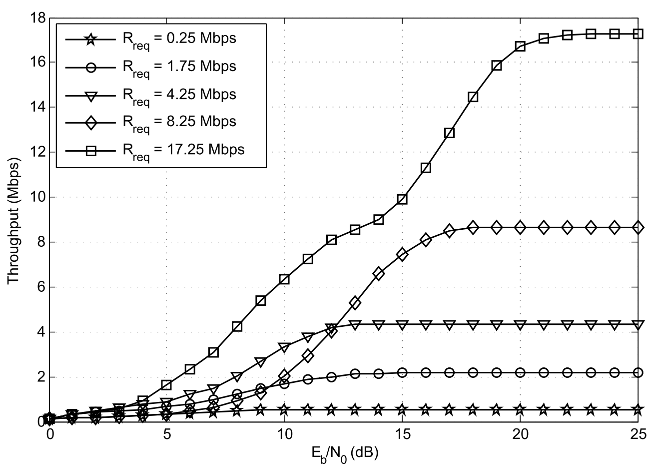

Fig. 5. Throughput performance of the proposed algorithm for QoS demand of 10^{-3}.

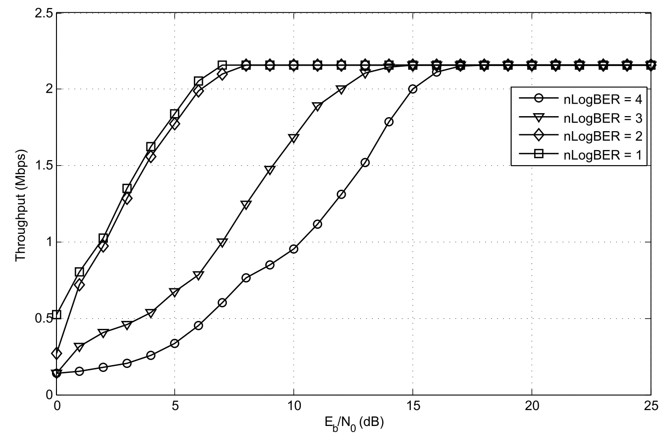

Fig. 6. Throughput performance of the proposed algorithm for various QoS requirements and $R_{req} = 1.75$ Mbps.

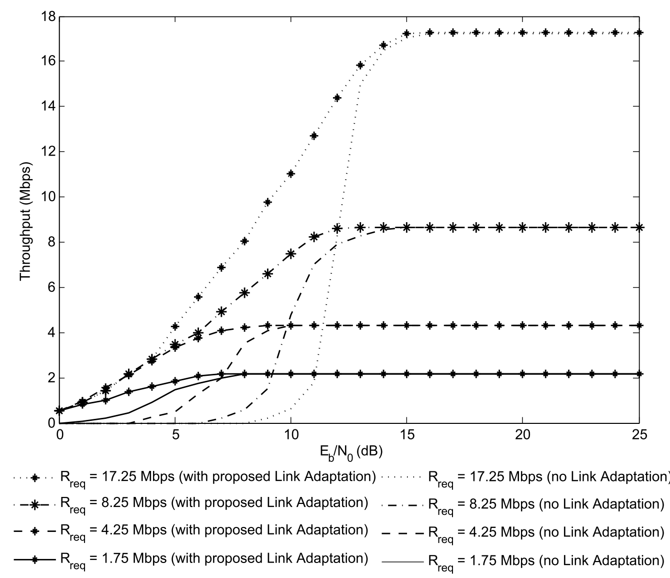

Fig. 7. Throughput with and without proposed link adaptation (QoS $= 10^{-1}$ and $R_{req} = 17.25$ Mbps).

maximum possible data rate (R_{max}) depending upon the QoS and throughput requirements (if $R_{max} \leq R_{req}$) or restrict the data rate to that required by the user (if $R_{max} > R_{req}$). The data rate is restricted to the throughput required by the user or a specific application to reduce the computational load and thus the power consumption and allow more active users to communicate during the current time slot.

Figure 4-5 show the resulting throughput versus E_b/N_0 for QoS requirements of 10^{-1} and 10^{-3}, respectively, plotted for R_{req} ={0.25, 1.75, 4.25, 8.25, 17.25} Mbps. The Stanford University Interim channel model is used in the simulations to investigate the performance in the presence of Doppler spread and multipath fading. SUI channels are a set of 6 channel models representing three terrain types and a variety of Doppler spreads, delay spread and Line of Sight (LOS)/Non Line of Sight (NLOS) conditions. The detailed Power delay profile and specifications of all SUI channel models is given in [30]. We have used SUI-4 channel model in our simulations which models a terrain having moderate to high tree density and weak LOS. K-factor represents the ratio of LOS component to NLOS components. For NLOS case, K-factor is zero. It can be seen that the proposed algorithm achieves the required throughput for all the QoS and data rate requirements at various E_b/N_0 values by maintaining the BER less than or equal to BER_{max}. The higher throughput requirements are met at higher values of E_b/N_0 because of the selection of higher modulation and multicode indices. However, maximum possible throughput is achieved at lower values of E_b/N_0 by the selection of appropriate MMI pair through FRBS.

Figure 6 compares the effect of QoS demands on the maximum achieved data throughput. As expected, R_{req} is achieved at lower values of E_b/N_0 for relatively lower QoS demands. The QoS demands are taken in $nLogBER$ format defined earlier in the paper. For $BER_{max} = 3$ and 4, the effect on throughput is small. This is due to less difference in BER performance between $M = 8$ and $M = 16$, as shown in figure 2.

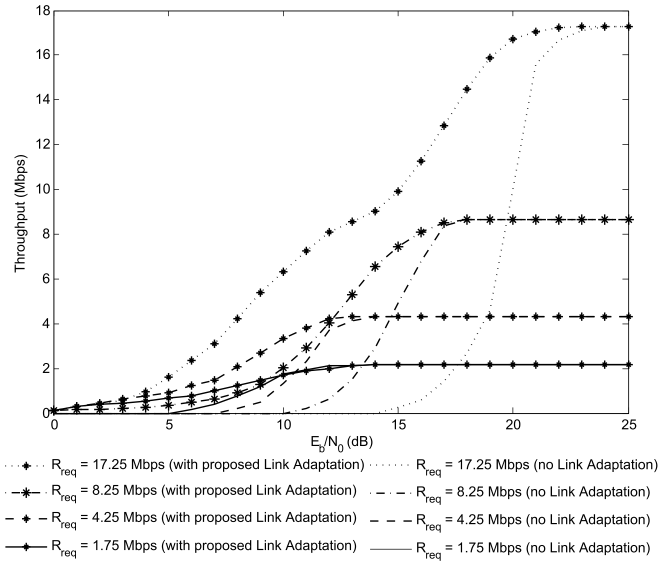

Fig. 8. Throughput with and without proposed link adaptation (QoS $= 10^{-3}$ and $R_{req} = 8.25$ Mbps).

Mostly, the wideband networking waveforms use network packet-based communication. In the absence of link adaptation, a lower value of BER drastically increases the packet error rate, thereby increasing the packet re-transmission rate. The proposed link adaptation algorithm reduces packet re-transmissions rate by switching the modulation scheme and multicode index to provide maximum possible throughput for lower E_b/N_0 values. Figure 7-8 show the throughput comparison with two different QoS requirements with and without the proposed link adaptation algorithm for $R_{req} = \{1.75,\ 4.25,\ 8.25,\ 17.25\}$ Mbps. It can be seen that for all the throughput requirements without link adaptation, the modulation scheme and modulation index are chosen which provide the required throughput in perfect channel conditions. In the absence of link adaptation, the throughput drastically reduces at lower E_b/N_0 values because almost every network packet needs re-transmission due to very high packet error rate. On the other side, the proposed link adaptation provides maximum possible throughput for lower E_b/N_0 values and restricts throughput to the required value in good channel conditions.

VII. Conclusion

A novel algorithm for link adaptation using fuzzy rule based system for adaptive TDMA/multicode CDMA based wideband networking waveform of SDR is proposed. A constrained optimization problem is formulated and then solved using FRBS by changing the modulation technique and the number of multicodes assigned to each user. The proposed algorithm selects optimum pair of modulation and multicode indices to provide possible maximum or desired throughput, depending upon the throughput required by the user or application. The restriction of throughput to the required value through proposed algorithm reduces the computational complexity and thus the power consumption. It is shown that the proposed link adaptation scheme achieves better throughput by efficiently reducing the packet re-transmissions overhead.

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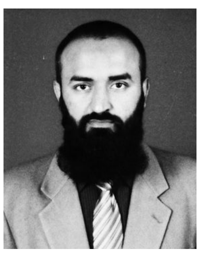

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