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Research Article

Class-Based Fair Code Allocation with Delay Guarantees for OVSF-CDMA and VSF-OFCDM in Next-Generation Cellular Networks

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This paper presents a novel class-based fair code allocation (CFCA) protocol to support delay and rate guarantees for real-time flows and to provide fairness for non-real-time flows on the downlink of WCDMA- and VSF-OFCDM-based cellular networks. CFCA not only assigns bandwidth dynamically to different flows but also determines those appropriate OVSF codes whose assignment results in the minimum overhead for code reassignments during dynamic bandwidth allocation. To reduce the overhead of code reassignments, this paper introduces the concept of affinity-mate and enables bandwidth allocation and code placement to interact with each other. A new performance metric, called class-based rate degradation ratio (CRD), is introduced to ensure fairness in providing rate and delay guarantees by measuring the rate degradations of flows based on their traffic types. The simulation results show that code reassignment overhead can be reduced by up to 60% for high network loads. For low network loads, fairness is achieved fully, but for high network loads the average rate requirement is met fairly for 95% of the flows.

1. Introduction

In cellular networks limited radio spectrum is a very important radio resource whose efficient management gets more critical as the bandwidth requirements of new applications increase. A challenging issue in supporting QoS in any wireless cellular network is the time-varying channel conditions due to various types of fading. Employing agile power control alone to counteract variations in channel conditions may cause excessive cochannel interference to other mobile stations in the cell [1]. Also, it is shown in [2] that when compared to fixed-rate power control, adaptive modulation achieves significant throughput advantage. When adaptive modulation is employed instead of power control to counteract the variations in channel conditions, the modulation and coding schemes are varied dynamically based on the varying channel conditions. When channel conditions deteriorate for a user, use of adaptive modulation reduces the data rate achieved by the user because of the use of higher-order modulation and coding scheme. This reduction in data rate impacts the QoS guarantees such as delay and throughput of the user's application. To compensate for the loss in data rate additional bandwidth has to be allocated to the user. Thus, there is a need for dynamic bandwidth allocation. Therefore, an effective dynamic bandwidth allocation algorithm, which dynamically allocates bandwidth with low control signaling overhead to existing mobile users at every time slot based on their channel conditions and delay requirements, is critical in order to meet the QoS requirements and to provide fairness [3]. In this paper, dynamic bandwidth allocation is accomplished by varying the spreading factor assigned to a flow.

Wideband code division multiple access (WCDMA) cellular networks use a CDMA scheme known as OVSF-CDMA [4] to support variable data rates by employing orthogonal variable spreading factor (OVSF) codes. In an OVSF-CDMA-based system, each mobile station is assigned a single OVSF code. Variable data rates are supported by changing the spreading factor (SF). An alternative CDMA scheme known as multicode CDMA (MC-CDMA) [5] assigns multiple

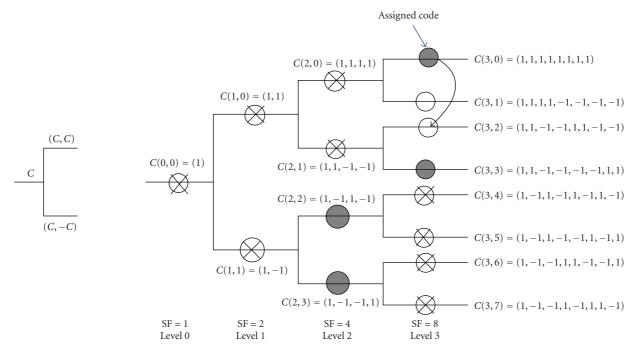


FIGURE 1: Code blocking and reassignment in an OVSF code tree. The filled circle and the crossed circle indicate the assigned and blocked codes, respectively. A free code is indicated by an empty circle. When code C(3,0) is assigned, it blocks the assignment of its all ancestors and descendants, though its descendants are not shown in this figure. To lift the blocking on C(2,0), code C(3,0) can be freed by assigning C(3,2) to the call of C(3,0).

codes of the same spreading factor to a mobile station in order to achieve variable data rates. MC-CDMA requires multiple transceivers to support variable data rates. OVSF-CDMA reduces the hardware complexity of the mobile station because it requires only a single transceiver to support variable data rates. However, OVSF-CDMA suffers from the code blocking problem, as explained next.

OVSF codes are generated recursively in a tree fashion, as shown in Figure 1, using Walsh matrices or applying the following rule recursively: code C(m,k) of level m generates the following two orthogonal codes of level m + 1: C(m+1,2k) = [C(m,k),C(m,k)] and C(m+1,2k+1) =[C(m,k), -C(m,k)], where -C(m,k) denotes the binary complement of C(m,k), m is the level of the code in the OVSF code tree and k is the index of the code at level m, assuming that the root code is at level zero [4, 6]. Any two OVSF codes are orthogonal if and only if one of the two codes is not the ancestor code of the other. For example, in Figure 1 codes C(2,0) and C(3,0) are not orthogonal because C(2,0)is an ancestor code of C(3,0) and therefore they should not be assigned to two different calls at the same time. When a call is initially admitted, it is assigned an OVSF code with the requested rate by an initial code placement algorithm. Code blocking occurs when there is no corresponding free OVSF code for an incoming call, even though the system has sufficient residual capacity to support it. For example, in Figure 1, a new call requiring an SF = 4 cannot be assigned a code since there is no free code with SF = 4. To mitigate code blocking, an existing call may be reassigned a different OVSF code. For instance, to lift the blocking on C(2,0) by freeing

C(3,0), the call of C(3,0) is reassigned with C(3,2), as shown in Figure 1.

Dynamic bandwidth allocation in WCDMA networks involves dynamic assignment of OVSF codes. When dynamic bandwidth allocation is not used, code reassignments are needed to eliminate code blocking only. When dynamic bandwidth allocation is used, code reassignments are needed to dynamically change the data rates assigned to mobile stations as well. The computational overhead can be reduced if the dynamic bandwidth allocation algorithm can easily determine the code to be reassigned for supporting a higher data rate. The control signaling overhead is reduced if fewer number of bits are used to inform the mobile station about the reassigned code. To reduce the code reassignment overhead for a given code, this paper introduces the concept of *x-hop affinity-mate* to find easily another code with the same or higher rate.

This paper presents a class-based fair code allocation (CFCA) protocol to support fairness, rate and delay guarantees while allocating codes with low reassignment overhead in WCDMA. CFCA includes three algorithms: class-based code placement (CBP), class-based code replacement (CBR), and dynamic bandwidth allocation (DBA). Algorithm 1 aims to assign each flow a code whose affinity-mate codes can be easily assigned later to the flow in case of stringent delay requirements or poor channel conditions. If the affinity-mate codes of a code are not available, then Algorithm 2 is invoked to assign an appropriate code to the flow that requires a higher-rate code due to poor channel conditions. Both CBP and CBR also undertake reducing the number of code

```
Input: A new call is admitted to the network because there exists at least one free code to support
the requested data rate. Variable max_hops denotes the maximum x in x-hop affinity-mate.
Output: The new call is assigned a free code with the highest weight W_{i,j}.
(1) Let r denote the number of those free OVSF codes whose SF equals s. Label them 1 to r from
   left to right.
(2) for i = 1 to max\_hops do
       for j = 1 to r do
(3)
(4)
          if (new call is RT (conversational or streaming)) then
(5)
             if (i-hop affinity-mate of the free code j is blocked or being used by an RT call) then
(6)
                 W_{i,j} = x_1
             else if (i-hop affinity-mate of the free code j is blocked or being used only by NRT calls)
(7)
             then
                 W_{i,j}=x_2
(8)
(9)
             else if (i-hop affinity-mate of the free code j is free) then
(10)
                  W_{i,j} = x_3
              endif
(11)
       else if (new call is NRT (interactive or background)) then
(12)
              if (i-hop affinity-mate of the free code j is blocked or being used only by NRT calls) then
(13)
(14)
                  W_{i,j} = x_1
(15)
              else if (i-hop affinity-mate of the free code j is blocked or being used by an RT call) then
(16)
                  W_{i,j} = x_2
              else if (i-hop affinity-mate of the free code j is free) then
(17)
(18)
                  W_{i,j} = x_3
(19)
              endif
(20)
           endif
(21) endfor
(22) endfor
(23) The new call is assigned the free code with the highest weight W_{i,j} among the r free codes
     considered at the previous step. If there is more than one code with the highest weight, then
     choose the code whose index i is the smallest to break the tie. Any further ties are broken
     randomly.
end
```

ALGORITHM 1: Algorithm CBP.

reassignments while eliminating code blocking. Although the existing bandwidth allocation algorithms address rate allocation only without considering code placements and reassignments, Algorithm DBA (see Algorithm 3) enables rate allocation, code allocation and reassignment to interact with each other in order to provide fairness, delay and rate guarantees with low code reassignment overhead.

This paper also introduces a new performance metric, class-based rate degradation (CRD), to schedule the code assignments of flows based on their current rate degradation and traffic class. CRD helps meet the delay and rate guarantees for real-time flows and to provide fairness for non-real-time flows. Hence, the main contributions of this paper are fourfold: (i) the code placement algorithm CBP for reducing the overhead significantly for dynamic bandwidth allocation in WCDMA networks, (ii) the code reassignment algorithm CBR for freeing blocked codes if a cellular network has sufficient residual capacity, (iii) the dynamic bandwidth allocation algorithm DBA that uses the proposed CRD metric to provide delay and rate guarantees for real-time traffic, and fair access for non-real-time traffic, and (iv) the concept of x-hop affinity-mate for reducing overhead in code reassignments during dynamic bandwidth allocation.

While WCDMA-based cellular networks use OVSF codes for channel allocation, variable spreading factor orthogonal frequency code division multiplexing (VSF-OFCDM) has been proposed as the transmission scheme for 4G nextgeneration cellular networks [7–9]. In VSF-OFCDM, spreading is done both in the time and in the frequency domains. The amount of time and frequency domain spreading can be adapted dynamically based on the data rate requirements and channel conditions of the user. OVSF codes can be used to determine the time domain and frequency domain spreading in VSF-OFCDM systems [10, 11]. The amount of time domain spreading can be varied by varying the allocated time domain OVSF code. This is in turn modifies the amount of frequency domain spreading, which is the number of orthogonal subcarriers used for frequency division multiplexing. Frequency domain spreading gives better BER performance when the number of users using the same time domain code is low. However, when the number of users using the same time domain code increases, intercode interference increases. In order to reduce the intercode interference, users are assigned a descendant code of the previous time domain code of higher spreading factor as the new time domain code. This reduces the number of users using the same time **Input**: A new call or an existing call that requires a higher-rate code requests a code of SF *s*. But the network does not have a free code of SF *s*, even though the network has residual capacity to support the call.

Output: An OVSF code of the required SF is freed by reassignment.

Begin

- (1) if a new call then
- (2) Let r denote the number of those blocked codes whose SF equals s, and label them from 1 to r.
- (3) Among the r codes determine the codes that have the maximum weight $W_{i,l}$ for values of i = 1 to max_hops and l = 1 to r.
- (4) Determine the code *j* that has the least number of busy descendant codes assigned to real-time calls among the codes with the same maximum weight. Any further ties are broken randomly.
- (5) **else if** a real-time call requires code reassignment to meet its delay requirements **then**
- (6) Let r denote the number of those codes whose SF equals s. Label them from 1 to r.
- (7) Determine the code j that has the least number of busy descendant codes assigned to real-time calls. Any further ties are broken randomly.
- (8) else if a non-real-time call requires code reassignment to receive fair share of bandwidth then
- (9) Let r denote the number of those codes of SF = s that are free or assigned or blocked by non-real-time calls. If a code of SF = s is not available, search for a free code of the nearest higher spreading factor.
- (10) Determine the code *j* that is assigned to a non-real-time call with the minimum CRD value. Any further ties are broken randomly.
- (11) endif
- (12) Let q denote the number of calls that are already assigned t descendants codes of code j.
- (13) For each call 1 to q, assign a code using the CBP algorithm, if there are more than one code of the required SF s_q for the call. If no code is available of the required SF, then call CBR again to free a code of the required SF s_q .
- (14) Assign code j to the new call or to the existing real-time or non-real-time call requesting code reassignments.

end

ALGORITHM 2: Algorithm CBR.

domain code at least by half and thus reduces the intercode interference. In this paper, we present how the presented fair code allocation scheme can be used in 4G VSF-OFCDM systems to pick an OVSF code that offers flexibility in time and frequency domain spreading.

The rest of the paper is organized as follows. Section 2 presents the related work. Section 3 describes the system model. Section 4 presents the CFCA protocol, including the algorithms CBP, CBR, and DBA, along with the CRD metric, and the concept of *x*-hop affinity-mate. Section 5 presents how the CFCA protocol can be used in VSF-OFCDM systems. Section 6 presents a performance analysis of the CFCA protocol. Simulation results are discussed in Section 7, and the concluding remarks are made in Section 8.

2. Related Work

Code placement schemes [2, 12–25] assign codes to new and handoff calls in such a way that the probability of call blocking is reduced when code reassignments are not allowed in the system. When code reassignments are allowed in the system, the objective of code replacement schemes [26, 27] is to reduce the number of code reassignments by freeing blocked codes. Existing code placement and replacement algorithms do not consider the impact of the code placement on dynamic bandwidth allocation. They focus only on

keeping the code tree as compact as possible so that the number of reassignments that could be needed when a new call arrives is reduced. This is not sufficient, however, for dynamic bandwidth allocation in which the codes of the existing flows may need to be changed because of their poor channel conditions and delay requirements. Therefore, our code placement (CBP) and code reassignment (CBR) algorithms allocate codes to flows by considering the possibility of assigning higher rate codes to the flows when channel conditions are poor or the flows have difficulties in meeting their delay requirements. That is, when CBP or CBR assigns a code to a flow, it ensures that the flow could be reassigned a higher-rate code with a low cost of signaling overhead.

Dynamic bandwidth allocation to support the QoS and fairness in WCDMA wireless cellular networks is studied in [3, 28–34]. Most of the existing bandwidth allocation algorithms ignore the signaling overhead in dynamic bandwidth allocation. In [3, 34], some methods for reducing the signaling overhead are discussed, though the methods in [3] consider the multicode model. In addition, only bursty traffic is considered in [34]. But, when real-time traffic is continuous and non-real-time traffic is bursty, an idle non-real-time flow can accumulate credits and subsequently can receive a higher priority in scheduling. This can affect adversely the rate allocated to continuous real-time traffic, which may result in higher delay for real-time packets. However, this paper considers fairness and QoS

```
Input: A WCDMA-based cellular network with limited number of OVSF codes. Every admitted
flow (or call) f_i is initially assigned an OVSF code, denoted C_i(m, k), and the code C_i(m, k) is
marked "assigned". w_i is given for every flow f_i, and v is common for all flows. count is initially
Output: OVSF codes are assigned to all flows based on their delay and average data rate requirements,
while reducing signaling overhead.
begin
(1) for every frame do
        For those flows that have terminated, mark their codes "unassigned".
(2)
(3)
        Assign every flow f_i its initial code C_i(m, k) even if f_i was assigned a different code during
        the transmission of its last frame.
        count \leftarrow count + 1; for each flow f_i, compute CRD_i if (count \mod w_i) = 0.
(4)
(5)
        for i = 0 to 3 do
           if i is 0 then
(6)
              class ← "conversational".
(7)
(8)
           else if j is 1 then
(9)
              class ← "streaming".
(10)
            else if j is 2 then
              <u>class</u> ← "interactive".
(11)
(12)
            else if j is 3 then
(13)
               class ← "background".
            Let z equal the number of all those flows of <u>class</u> type.
(14)
(15)
            while 0 \le j \le 1 and z > 0 do
              Let f_i denote the <u>class</u> flow with the highest CRD value among those <u>class</u> flows that are
(16)
              not considered yet in this frame.
(17)
               if (CRD_i > 0 then
                 Call ASSIGN_HRC(i, CRD_i, C_i(m, k)).
(18)
(19)
               else
(20)
                 Use the same code C_i(m, k) of flow f_i in this frame as well.
(21)
               endif
              z \leftarrow z - 1
(22)
         endwhile
(23)
(24)
         while 2 \le j \le 3 and z > 0 do
              Let f_i denote the <u>class</u> flow with the highest CRD value among those <u>class</u> flows that
(25)
               are not considered yet in this frame.
(26)
               if (CRD_i > 0) and (Code C_i(m, k)) of flow f_i are not available due to its assignment to a
               real-time flow then
(27)
                 Call ASSIGN_HRC (i, CRD_i, C_i(m, k)).
(28)
(29)
                 Use the same code C_i(m, k) of flow f_i in this frame as well if it is available. Otherwise
                  no code is assigned to flow f_i for this frame.
(30)
(31)
              z \leftarrow z - 1
(32)
         endwhile
(33) endfor
(34) endfor
end
```

ALGORITHM 3: Algorithm DBA.

guarantees for admitted calls of both continuous and bursty real-time and non-real-time traffic. In [33], a joint power and rate adaptation scheme is presented to meet the QoS requirements of traffic belonging to various traffic classes. In [30], a credit-based bandwidth allocation scheme to ensure fairness and minimum rate guarantees under varying channel conditions is presented. In [31], a threshold-based scheme is described to dynamically change the code assigned to a call so that the delay performance of the high QoS traffic is improved. In [32], a packet scheduling scheme for

continuously backlogged traffic is presented. However, in [28, 30–33], only a general bandwidth allocation problem is addressed without addressing code allocation and signaling overhead during dynamic bandwidth allocation. Signaling overhead is the number of bits of control information required to inform the receivers of the mobile stations about the OVSF codes assigned to them during dynamic bandwidth allocation.

In [12–15], the authors address the code assignment and reassignment problem so that the overhead of code

reassignments is reduced while admitting a new or a handoff call. However, they do not consider the impact of code placement and replacement on dynamic bandwidth allocation. Dynamic bandwidth allocation addresses the code assignment problem every time slot for existing calls so that their delay and rate requirements are met, as described in [3, 28, 34]. Hence, no existing work addresses code placement and replacement together with dynamic bandwidth allocation for OVSF-CDMA-based systems. In addition, in 3G networks, the assignment of bandwidth (or code) and power to a non-real-time flow would affect and constrain the power and bandwidth that can be assigned to a real-time flow during dynamic bandwidth allocation. As stated in [34, 35], although non-real-time flows do not have a strict delay bound, it is not desirable either to have too long service times for them. Service providers should provide "enough" bandwidth for all users, leading to more subscribers it can serve, and more revenue they can earn. Therefore, it is necessary to consider the scheduling of nonreal-time traffic along with real-time traffic so that non-realtime flows do not get starved for extended periods of time. This paper proposes the CFCA protocol to address all these issues together in WCDMA networks.

3. System Model

We consider n flows (or calls), f_1, f_2, \ldots, f_n , within a single cell of a WCDMA-based cellular network, where the terms "flow" and "call" are used interchangeably to mean a stream of packets. Any call that is admitted into the system is referred to as a new call regardless of whether it is a hand-off call or is initiated in the current cell. The flows transmit data through wireless channels separated by OVSF codes. Each downlink channel is time slotted such that each time slot is equal to a 10-millisecond WCDMA frame. Control signals such as the transmit power control and rate information are time-multiplexed with the user data in each time slot. We use the control header to transmit the identity of the assigned OVSF code. The code allocations and reassignments are done by a dynamic bandwidth scheduler, based on the power and code resources, the number of traffic flows, and the feedback about the quality of the channels.

We are interested in the downlink control of transmissions in such a way that the flows meet fairly the delay and rate requirements. To achieve this, the rate allocated to a mobile station is dynamically varied by adjusting the spreading factor of the assigned OVSF code [36]. To ensure successful reception of the packetized data at a mobile station (MS), there is a limit on the achieved bit error rate (BER). Depending on the spreading factor, modulation and coding scheme used, a target E_b/I_o should be achieved at the MS so that the limit on BER is not exceeded. E_b/I_o represents the ratio of energy-per-bit (E_b) to interference power spectral density (I_0) . Based on the channel state feedback received from the MS and the spreading factor used, the BS adjusts the power, modulation and coding used for a flow to meet the target E_b/I_o . But, in order not to introduce any additional intercell interference to other cells, the total power at the BS

is constrained. As a result, the power requirements of all the flows may not be met at some instances. In this case, flows are served in their priority order as long as the total transmit power constraint of the BS is not violated.

The third generation (3G) universal mobile telecommunications system (UMTS) describes four traffic types (or QoS classes), namely, conversational (e.g., voice), streaming (e.g., streaming video), interactive (e.g., web browsing) and background (e.g., email). In the proposed code placement and replacement algorithms, the conversational and streaming classes are referred to as the real-time (RT) class and interactive and background classes are referred to as the non-real-time (NRT) class. The four traffic classes of WCDMA are distinguished by the proposed dynamic bandwidth allocation algorithm according to their priorities; that is, conversational traffic is considered first, then streaming, followed by interactive and background traffic classes.

For simplicity, we assume a two-state channel model, according to which the channel can be either in normal state or poor state. Under normal channel conditions, the flow can achieve a data rate equal to its average requested data rate using the OVSF code assigned to it at admission. Under poor channel conditions, a flow still receives data with the same power of transmission, but at a lower rate because of the use of a lower modulation level and lower coding rate. To achieve the average data rate, we assign a higher rate (lower SF) code for real-time flows under poor channel conditions. As shown in [37], for higher spreading factors (SF \geq 32), the additional power needed to achieve the same BER while moving from SF = s to SF = s/2 is of the order of 0.5 dB. The admission control scheme presented in Section 4 ensures that this additional power is always available for all admitted realtime flows under poor channel conditions. It should be noted that the additional power needed to achieve the same BER without changing the SF, modulation and coding scheme is relatively high and is of the order of 3 dB as shown in [38].

We use a simplified E_b/I_o model as the channel model. In a WCDMA network, the E_b/I_o achieved at the mobile k is expressed as

$$\left(\frac{E_{b}}{I_{o}}\right)_{k,SF_{k},MCS_{k}} = \frac{W}{R_{k,t}} \frac{P_{k,t}^{j} L_{k,t}^{j}}{\left((1-\alpha) \times \left(P_{T,t} - P_{k,t}^{j}\right) \times L_{k,t}^{j} + N_{o} + I_{\text{inter},k,t}\right)}, \tag{1}$$

$$\left(\frac{E_{b}}{I_{o}}\right)_{k,SF_{k},MCS_{k}} = SF_{k,t} \frac{P_{k,t}^{j} L_{k,t}^{j}}{\left((1-\alpha) \times \left(P_{T,t} - P_{k,t}^{j}\right) \times L_{k,t}^{j} + N_{o} + I_{\text{inter},k,t}\right)}, \tag{2}$$

where $(E_b/I_o)_{k,SF_k,MCS_k}$ is the E_b/I_o requirement of kth flow assuming a spreading factor of SF_k and modulation and coding scheme MCS_k , $P_{k,t}^j$ is the instantaneous power allocated to flow f_k by base station j at time t, $R_{k,t}$ is the

instantaneous data rate allocated to flow f_k at time t, $L_{k,t}^{j}$ is the power loss on the path from base station j to mobile kat time t, P_T is the total power budget of the base station on the downlink, $P_{T,t}$ is the instantaneous transmit power of the base station on the downlink for all existing flows at time t, N_o is the noise power spectral density, $I_{inter,k,t}$ is the intercell interference at mobile k at time t, W is the chip rate, α is the own-cell orthogonality factor (typically ranges between 0.4 and 0.9), and $SF_{k,t}$ is the spreading factor of the code assigned to flow k at time t. $L_{k,t}^{J}$ is the product of distance-based path loss, slow fading (shadowing), and fast fading (multipath) on the wireless channel from BS j to MS k. Path loss $PL_{k,t}^{j}$ is computed as a function of distance $D_{k,t}^{J}$ as given in (4), where δ is the path loss exponent. Slow fading, $S_{k,t}^j$ is considered to be log-normally distributed around the distance-based path loss $PL_{k,t}^{J}$ with zero mean and standard deviation. Fast fading, $F_{k,t}^{j}$ is generated using a Rayleigh fading distribution with zero mean and standard deviation. Hence, $L_{k,t}^{j}$ and $PL_{k,t}^{j}$ can be written as

$$L_{k,t}^{j} = PL_{k,t}^{j} \times S_{k,t}^{j} \times F_{k,t}^{j}, \tag{3}$$

$$PL_{k,t}^{j} = \left(D_{k,t}^{j}\right)^{-\delta}.\tag{4}$$

4. Class-Based Fair Code Allocation (CFCA) Protocol

This section presents our class-based fair code allocation (CFCA) protocol to assign the appropriate OVSF codes to the traffic flows based on their delay and data rate requirements, channel conditions, and fairness. The objectives of the CFCA are as follows: (i) to assign bandwidth fairly to real-time flows so that their rate and delay requirements are met, (ii) to assign fairly the residual bandwidth among non-real-time flows, and (iii) to reduce the overhead for code reassignments in dynamic bandwidth allocation. CFCA uses three main algorithms, and the list of notations used by these algorithms is shown in Table 1.

CFCA admits a new real-time call to the network if the total network capacity and base station power budget is always capable of supporting all the existing real-time flows under poor channel conditions at which they need higher rate codes. Therefore, there is a constraint on the number of admitted real-time flows to help meet the delay guarantees of real-time flows in the presence of poor channel conditions. It should be noted that a poor channel condition implies a channel state at which a mobile station is still able to receive data with the same power of transmission, but at a lower rate because of the use of lower modulation level and lower coding rate. The acceptable poor channel condition at any location in a coverage area is determined by the cellular service providers by considering path loss, fading, and worst case inter- and intracell interference. Service providers can then use the acceptable poor channel condition as a constraint in determining the optimal locations of base stations in a given coverage area. For example, in [39, 40],

the authors propose optimization models for base station locations considering the signal-to-noise ratio as the quality measure. In [41], the authors propose models for base station location so that the quality of service constraints is satisfied. Once a service provider plans his network for a given poor channel condition, the aim of the CFCA protocol is to provide QoS guarantees to those real-time flows that can at least maintain this poor channel condition by just making use of the power and code resources used to determine their admission.

When a flow f_i is admitted, there are two bandwidth requirements for flow f_i : $B_n(f_i)$ and $B_p(f_i)$. $B_n(f_i)$ is the bandwidth needed for flow f_i under normal channel conditions, whereas $B_p(f_i)$ is the bandwidth needed for flow f_i under poor channel conditions. $B_n(f_i)$ represents the spreading factor (SF) required to meet at least the average data rate of flow f_i under normal channel conditions at which a high-order modulation and coding scheme is used. $B_p(f_i)$ represents the spreading factor (SF) required to meet the average data rate of flow f_i under poor channel conditions at which a lower-order modulation and coding scheme is used. $B_n(f_i)$ is greater than $B_p(f_i)$ and, therefore, $B_p(f_i)$ requires a higher-rate code (code of lower SF) than the code needed by $B_n(f_i)$. The total bandwidth needed by all realtime flows under poor channel conditions cannot exceed the total network capacity. Thus, a new real-time call f_i is admitted if (5) holds:

$$\frac{1}{B_p(f_1)} + \frac{1}{B_p(f_2)} + \dots + \frac{1}{B_p(f_{i-1})} + \frac{1}{B_p(f_i)} \le 1, \quad (5)$$

where $f_1, f_2, \ldots, f_{i-1}$ are the existing real-time flows, and $B_p(f_k)$ is the SF required to meet the data rate of flow f_k under poor channel conditions for $1 \le k \le i$. As for the non-real-time flows, CFCA admits a new non-real-time flow if the total bandwidth requirements of all existing real-time and non-real-time flows under normal channel conditions are less than the total network capacity. Assuming that m-1 is the number of all existing flows (real-time and non-real-time) in the network, a new non-real-time flow f_m is admitted if (6) holds:

$$\frac{1}{B_n(f_1)} + \frac{1}{B_n(f_2)} + \dots + \frac{1}{B_n(f_{m-1})} + \frac{1}{B_n(f_m)} \le 1, \quad (6)$$

where $B_n(f_k)$ is the SF required to meet the data rate of flow f_k under normal channel conditions. This paper assumes that a higher-level modulation and a higher-rate coding scheme are used under normal channel conditions. When channel conditions become poor, both modulation level and coding rate are reduced [42]. For instance, under normal channel conditions, let us assume that the modulation level is 8 (64 QAM) and coding rate is 1/2. When the channel conditions become poor, the modulation level can be lowered to 4 (16 QAM) or 2 (QAM), while the coding rate could be reduced from 1/2 to 1/3 to meet the required E_b/I_o . To compensate for the loss in data rate under poor channel conditions, we assign a higher rate code of lower SF that will run with lower modulation level and coding rate. In [42], the authors show how the achieved data rate can be modified by changing the modulation and coding scheme.

Table 1: Notations.

f_i	Flow i or call i	
$R_{i,t}$	Instantaneous data rate of flow f_i at time t	
$R_{i,avg}$	Requested average data rate of flow f_i	
$SF_{i,t}$	Spreading factor of the code assigned to flow f_i at time t	
C(m, k)	OVSF code at level m with index k	
$W_{x,l}$	Weight of code <i>I</i> based on its <i>x</i> -hop affinity-mate of rate $2_{x-1} \times R$	
$CRD_{i,j}$	Class-based rate degradation ratio of flow f_i at the end of time slot j	
$\mathrm{WRD}_{i,j}$	Window Rate Degradation of flow f_i for window j	
ν	Number of time slots over which CRD is computed	
w_i	Number of time slots for which flow f_i can tolerate its data rate to be lower than average data rate; ν time slots contain one or more windows	
$w_i(C), w_i(S), w_i(I), w_i(B)$	Window sizes of the conversational, streaming, interactive, and background traffic classes, respectively, for flow f_i	
CRD_{th}	CRD threshold for ensuring fairness in rate allocation to NRT flows	
$B_n(f_i)$	The SF needed by flow f_i to meet at least the average data rate under normal channel conditions	
$B_p(f_i)$	The SF needed by flow f_i to meet the average data rate under poor channel conditions	

Therefore, to ensure the availability of a higher rate code of lower SF for real-time flows under poor channel conditions, the following equations should also hold before admitting a new real-time flow. In these equations, we consider only the power required to use a higher rate code of spreading factor $B_p(f_k)$ instead of the normal rate code of spreading factor $B_n(f_k)$. The additional power needed to use a higher rate code is constant and depends only on the amount of reduction in the SF and does not depend on the channel conditions. Based on the analysis in [43], we can first express (2) as an inequality:

$$SF_{k,t} \frac{P_{k,t}^{j} L_{k,t}^{j}}{\left((1-\alpha) \times \left(P_{T,t} - P_{k,t}^{j}\right) \times L_{k,t}^{j}\right) + N_o + I_{\text{inter},k,t}}$$

$$\geq \left(\frac{E_b}{I_o}\right)_{k,SF_k,MCS_k}.$$
(7)

Since this equation is evaluated only at the time of admission for each flow, we can eliminate t as a parameter and replace $P_{T,t}$ with P_{τ} to represent the sum of transmit power assigned to all real-time flows at admission. Equation (7) implies that

$$P_k^j \ge \frac{\left((1 - \alpha) \times P_T \times L_k^j \right) + N_o + I_{\text{inter},k}}{\left(SF_k / (E_b / I_o)_{k, SF_k, MCS_k} + \left(L_k^j \times (1 - \alpha) \right) \right)}, \tag{8}$$

$$P_{k,p}^{j} \ge \frac{\left((1-\alpha) \times P_{T} \times L_{k}^{j} \right) + N_{o} + I_{\text{inter},k}}{\left(B_{p}(f_{k}) / (E_{b}/I_{o})_{k,\text{SF}_{k},\text{MCS}_{k}} + \left(L_{k}^{j} \times (1-\alpha) \right) \right)}, \quad (9)$$

$$P_T \ge P_{\tau} = \sum_{k=1}^{i} P_{k,p}^{j},$$
 (10)

where $P_{k,p}^{j}$ is the power requirement of flow f_k under poor

channel conditions when an SF of $B_p(f_k)$ is used. Before the base station admits a new real-time flow f_i , the base station first ensures that the sum of $P_{1,p}^j, P_{2,p}^j, \ldots, P_{(i-1),p}^j$ of existing real-time flows and $P_{i,p}^j$ of the new flow does not exceed P_T shown in (10).

From (9) and (10), it follows that

$$P_T \ge P_\tau$$

$$= \frac{\sum_{k=1}^{i} \left(\left((1-\alpha) \times P_{\tau} \times L_{k}^{j} \right) + N_{o} + I_{\text{inter},k} \right)}{\sum_{k=1}^{i} \left(B_{p} \left(f_{k} \right) / \left(E_{b} / I_{o} \right)_{k, \text{SF}_{k}, \text{MCS}_{k}} + \left(L_{k}^{j} \times (1-\alpha) \right) \right)},$$

$$(11)$$

$$P_T \geq P_{\tau}$$

$$= \frac{\sum_{k=1}^{i} (N_o + I_{\text{inter},k})}{\sum_{k=1}^{i} (B_p(f_k)/(E_b/I_o)_{k,\text{SF}_k,\text{MCS}_k} + (L_k^j \times (1-\alpha)))}$$

$$/ \left(1 - \frac{\sum_{k=1}^{i} (1-\alpha) \times L_k^j}{\sum_{k=1}^{i} (B_p(f_k)/(E_b/I_o)_{k,\text{SF}_k,\text{MCS}_k} + (L_k^j \times (1-\alpha)))}\right).$$
(12)

Since P_{τ} is a positive quantity, the following feasibility constraint should be met:

$$\frac{\sum_{k=1}^{i} (1 - \alpha) \times L_{k}^{j}}{\sum_{k=1}^{i} \left(B_{p}(f_{k}) / (E_{b}/I_{o})_{k, \text{SF}_{k}, \text{MCS}_{k}} + \left(L_{k}^{j} \times (1 - \alpha) \right) \right)} < 1.$$
(13)

A real-time flow f_i is admitted only if (5), (9), and (10) together hold. Similarly, a new non-real-time flow f_m is

TABLE 2: Protocol CFCA.

Step 1	For a real-time call f_i , determine $B_n(f_i)$, the spreading factor required under normal channel conditions, and $B_p(f_i)$, the spreading factor required under poor channel conditions. Admit the real-time call if (5), (9), and (10) hold. For a non-real-time call, determine only $B_n(f_i)$, the spreading factor required under normal channel conditions, and admit it if (6), (14), and (15) hold.	
Step 2	If a free code with the spreading factor $B_n(f_i)$ is available, go to next step. Otherwise, use Algorithm CBR to free a code with spreading factor $B_n(f_i)$ by doing code reassignments.	
Step 3	Use Algorithm CBP to initially allocate a particular free code of spreading factor $B_n(f_i)$ for the new call.	
Step 4	Run Algorithm DBA to dynamically allocate codes for meeting delay and rate guarantees of all active calls.	

admitted only if the following (14) and (15) hold along with (6):

$$P_{k,n}^{j} \ge \frac{(1-\alpha) \times P_{\gamma} \times L_{k}^{j} + N_{o} + I_{\text{inter},k}}{\left(B_{n}(f_{k})/(E_{b}/I_{o})_{k,\text{SF}_{k},\text{MCS}_{k}}\right) + \left(L_{k}^{j} \times (1-\alpha)\right)},$$
(14)

$$P_T \ge P_{\gamma} = \sum_{k=1}^{m} P_{k,n}^{j},$$
 (15)

where $f_1, f_2, ..., f_{m-1}$ are the existing real-time and non-real-time flows, $B_n(f_k)$ is the SF required to meet the data rate of flow f_k under normal channel conditions for $1 \le k \le m$, $P_{k,n}^j$ is the power requirement of flow f_k under normal channel conditions, and P_{γ} is the sum of transmit power assigned to all flows at admission.

CFCA is implemented in four steps as shown in Table 2. In Step 1 of CFCA, the SF of a new call is determined. A real-time call is admitted depending on whether (5), (9), and (10) hold or not. Similarly, a non-real-time call is admitted depending on whether (6), (14), and (15) hold or not. It should be noted that the SF required under poor channel conditions $(B_p(f_i))$ is only used to determine the admissibility of a real-time call in Step 1 of the CFCA protocol using (5). Once it has been determined that enough code resources meet the rate requirements of a real-time call even under poor channel conditions, only a code whose SF is equal to that required under normal channel conditions $(B_n(f_i))$ is assigned to a real-time call. During the life time of a call, if the channel conditions become poor, then a higher rate (of lower SF $B_p(f_i)$) code is assigned to meet the delay requirements of a real-time call. In the second step of CFCA, if a free code of the required spreading factor is not available even though the system has enough capacity to support the new call, algorithm CBR is invoked to free a code of the required spreading factor. The code that is made free is then assigned to the new call. In the third step, when a new call arrives, it is assigned an OVSF code of the required spreading factor using the CBP algorithm. In the fourth step, the call can use its initial code that is assigned by the CBP and CBR algorithms or a higher-rate code, and this decision is made every time slot by the DBA algorithm. When a higher-rate code is used, the mobile station is informed about the higher-rate code using control channel signaling. We use in-band control channel signaling mode [26], the control header is decoded by the mobile station using the initially assigned code. If the control header has control

information suggesting the use of a different higher-rate code to decode the data segment, the data segment is decoded using the higher rate code. In the next frame, the control header is again decoded using the initially assigned code and the process continues. The CFCA protocol uses the concept of x-hop affinity-mate to keep the control channel signaling overhead low when higher rate codes are assigned to calls.

Before presenting the algorithms CBP, CBR, and DBA, we now introduce the definitions for CRD and x-hop affinitymate next. We consider the last v time slots of flow f_i , each of which corresponds to a frame transmission time. We assume that w time slots constitute a window, so that v time slots have v/w windows. Let $R_{i,rcv}$ denote the average rate that flow f_i receives over a window, while $R_{i,avg}$ denotes the requested average rate. The value of $R_{i,rcv}$ depends on the modulation and coding scheme used. For example, for a symbol rate of 100 symbols per second, QPSK modulation scheme (BPS = 2) and 1/2 convolutional coding (CR = 1/2) give an information bit rate of 100 bits per second. On the other hand 64-QAM modulation scheme (BPS = 6) and 3/4 turbo coding (CR = 3/4) give an information bit rate of 450 bits.

Definition 1 (class-based rate degradation (CRD)). CRD_i represents the average rate degradation for which flow f_i experiences over the last ν time slots consisting of ν/w windows during which it receives less rate than the requested average data rate $R_{i,avg}$. CRD_i is expressed as

 CRD_i

$$= \frac{w_i}{v} \sum_{j=1}^{v/w_i} \left(\frac{R_{i,\text{avg}} - \min\left(R_{i,\text{avg}}, \sum_{k=1}^{w_i} R_{i,(t-k-w_i \times (j-1))}/w_i\right)}{R_{i,\text{avg}}} \right). \tag{16}$$

CRD_i (i.e., CRD of flow f_i) basically refers to the ratio of $(R_{i,avg} - R_{i,rcv})$ to $R_{i,avg}$ over v/w_i windows, when $R_{i,avg} \ge R_{i,rcv}$. The value of v is the same for all types of flows, whereas the window size w_i gets larger as the class priority of flows decreases. The value of v depends on the minimum time interval during which average rate requirements of non-real-time flows must be met. On the other hand, w_i is determined based on the delay requirements of f_i to indicate the number of consecutive time slots that flow f_i can tolerate its data rate to be lower than average data rate. In Figure 2, the value of v is chosen as 20 because it is the minimum time interval during which the average rate requirement of the

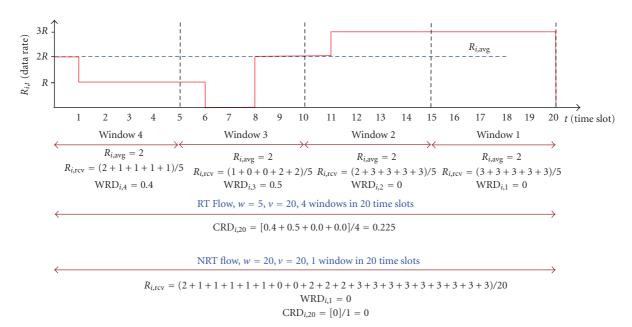


FIGURE 2: The data rates of a flow that are achieved within 20 time slots are the average data rate 2R in slot 1, the data rate R in slots 2 to 6, no data transmission in slots 7 to 8, and the average rate 2R or more in slots 9 to 20. (a) If the flow is a real-time (RT) flow with a window size of w = 5 over v = 20 time slots, then $CRD_{i,20}$ is the average of the window rate degradations $WRD_{i,1} = 0.0$, $WRD_{i,2} = 0.0$, $WRD_{i,3} = 0.5$, and $WRD_{i,4} = 0.4$. That is, $CRD_{i,20}$ is (0.4 + 0.5 + 0.0 + 0.0 + 0.0)/4 = 0.225. (b) If the flow is a non-real-time (NRT) flow with a window size of w = 20 over v = 20 time slots, then CRD equals zero.

non-real-time flow must be met. The ν time slots consist of v/w_i distinct windows, each having w_i time slots. CRD_i is computed as follows: (a) for each window j from j = 1 to v/w_i , compute the average $R_{i,rcv}$ for all rates $R_{i,(t-k-w_i*(j-1))}$ over w_i time slots, where $R_{i,(t-k-w_i*(j-1))}$ refers to the received data rate at time slot $(t - k - w_i * (j - 1))$, (b) determine the minimum of $R_{i,avg}$ and $R_{i,rcv}$ to find out whether $R_{i,rcv}$ is below the requested average rate $R_{i,avg}$, (c) compute the window rate degradation, denoted by $WRD_{i,j}$, for window j by subtracting the minimum of $R_{i,avg}$ and $R_{i,rcv}$ from $R_{i,avg}$ and then by dividing the resultant by $R_{i,avg}$ (i.e., WRD_{i,j} = $(R_{i,avg} - \min(R_{i,avg}, R_{i,rcv}))/R_{i,avg})$, and (d) finally find the average of all v/w_i windows' degradations and then call it CRDi. At a given time slot, CRDi is determined based on the received rates of the flow at the last ν time slots. The window sizes of all the flows in a traffic class are the same, and depend on the priority of their traffic class in the sense that a higher priority traffic class has a lower-size window. That is, if $w_i(C)$, $w_i(S)$, $w_i(I)$, and $w_i(B)$ denote the window sizes of the conversational, streaming, interactive, and background traffic classes, respectively, then it follows that: $w_i(C) < w_i(S) < w_i(I) < w_i(B)$. Figure 2 shows how CRD_i is computed. Note that CRD_i is computed by (16) in a sliding window manner with a period of w_i time slots. This implies that, for flow f_i , CRD_i is computed after every w_i time slot. CRD is somewhat similar to Degradation Ratio (DR) in [44] that determines whether a new call can be admitted by degrading the rates of existing flows.

This paper uses CRD to support delay requirements of real-time traffic by ensuring that the average requested rate is met at variable window sizes of frames. That is, if a flow is more delay sensitive, then the window size during which the requested average rate should be met is made smaller in the computation of CRD. In order to determine whether a flow meets the delay and rate requirements, we employ CRD for all flows and traffic types as described earlier. The value of CRD_i for flow f_i ranges from 0 to 1 to indicate "no degradation" and "maximum degradation", respectively. Specifically, if CRD_i equals zero, then flow f_i meets its both delay and rate requirements. However, if CRD_i is greater than zero, it indicates that flow f_i has experienced a degradation in rate and delay requirements, and the CRDi value represents the amount of degradation. In [44] the objective of the degradation metric is to intentionally degrade the QoS of existing flows in order to admit new flows. However, the objective of this paper is to support the rate and delay guarantees by keeping the degradations at a low value.

The authors in [3, 34] present scheduling algorithms to dynamically assign OVSF codes to mobile users on a timeslot-by-timeslot basis based on a credit-based mechanism. A credit-based mechanism assigns credits to a flow every time slot based on its requested average rate and deletes credits from a flow every time slot based on the rate allocated to that flow. Flows with higher credits have higher priority in scheduling and code allocation. The algorithms provide average data rate guarantee for bursty data traffic. Though the algorithms can be used for real-time traffic, it is more appropriate for non-real-time traffic because of the following issue. When there exist more than one flow with the same traffic type, these flows are scheduled based on their CRD values such that the flow with the highest CRD value is scheduled first to prevent it from having further degradation.

Table 3: The *x*-hop affinity-mates and data rates of some codes shown in Figure 1, for $1 \le x \le 3$, where *R* represents the data rate of any code with SF = 8. The affinity-mates of any other code can be determined similarly. This table is stored at the base station for determining the affinity-mate codes faster.

Code	1-hop affinity-mates	2-hop affinity-mates	3-hop affinity-mates
[C(3,0),R]	[C(2,0),2R],[C(3,1),R]	[C(2,1),2R],[C(3,2),C(3,3),R]	[C(2,2), C(2,3), 2R], [C(3,4), C(3,5), C(3,6), C(3,7), R]
[C(3,1), R]	[C(2,0),2R],[C(3,0),R]	[C(2,1), 2R], [C(3,2), C(3,3), R]	[C(2,2), C(2,3), 2R], [C(3,4), C(3,5), C(3,6), C(3,7), R]
[C(3,2), R]	[C(2,1),2R],[C(3,3),R]	[C(2,0),2R],[C(3,0),C(3,1),R]	[C(2,2), C(2,3), 2R], [C(3,4), C(3,5), C(3,6), C(3,7), R]
[C(3,3), R]	[C(2,1),2R],[C(3,2),R]	[C(2,0),2R],[C(3,0),C(3,1),R]	[C(2,2), C(2,3), 2R], [C(3,4), C(3,5), C(3,6), C(3,7), R]
[C(2,0),2R]	[C(2,1),2R]	[C(2,2), C(2,3), 2R]	-
[C(2,1), 2R]	[C(2,0),2R]	[C(2,2),C(2,3),2R]	_

In addition, to help reduce the degradation of a flow with high CRD value, our algorithm DBA attempts to increase the data rate of the flow above the requested average rate to meet the delay requirements. Increasing the rate requires the assignment of a higher rate code. The CFCA protocol uses the concept of *x*-hop affinity-mate whose definition is presented next, to keep the overhead during the assignment of higher-rate code low.

Definition 2 (x-hop affinity-mate). Two OVSF codes are referred to as x-hop affinity-mates if one of the following two conditions holds: (i) their nearest ancestor has the distance of exactly x hops to one of them and the distance of at most x hops to the other one, or (ii) one of the codes is the ancestor of the other one and the distance between them equals x hops, where a hop corresponds to an edge in the OVSF code tree.

The first part of the definition basically says that two codes are x-hop affinity-mates only if none of the two codes is at a distance greater than x hops from their common ancestor and at least one of the two codes is at a distance of x hops from their common ancestor. Table 3 shows a table for the 1-hop, 2-hop, and 3-hop affinity-mates of all the codes in the code tree illustrated in Figure 1. Because the lowest SF in WCDMA is 4, the codes C(0,0), C(1,0), and C(1,1) are not included in the table. In determining the code to be assigned to a new call, the CBP algorithm in Step 3 of the CFCA protocol chooses the code that has the best x-hop affinitymate of 2^{x-1} times higher rate. If more than one free code with the best 1-hop affinity-mate is available, CBP attempts to find a code with the best 2-hop affinity-mate and so on until x-hops. The determination of the best x-hop affinitymate is explained in the next section.

4.1. Class-Based Code Placement (CBP) Algorithm. When a new call arrives, there may be more than one free code that could be assigned to the call. To choose the free code that can cause the minimal overhead of code reassignments in case of poor channel conditions, we introduce the concept of weight for free codes as follows. For a new call of traffic type u, a free code j with SF = s and rate R, and its i-hop affinity-mate code with the data rate of $2^{i-1} \times R$, the weight $W_{i,j}$ of the code j is determined by a number x_k depending on the traffic type u and the status (e.g., free, busy with a real-time call, or busy with a non-real-time call) of the i-hop

affinity-mate code, for $1 \le k \le 3$ and $0 < x_1 < x_2 < x_3$. When the new call is a real-time call, the most desirable free code is the one whose i-hop affinity-mate code is free (lines 9-10), so that the call may be reassigned easily the free affinity-mate code of higher rate to meet its delay and rate requirements. The second most desirable free code is the one whose affinity-mate code is currently being used by a non-real-time call (lines 7-8) because the real-time call may be assigned the affinity-mate of higher rate that is currently being used by a non-real-time call when the real-time call requires a higher data rate. When the new call is a non-realtime call, the most desirable free code is again the one whose affinity-mate code is free (lines 17-18) because the call may be assigned the higher rate affinity-mate to increase its rate. But, the second most desirable free code is the one whose affinity-mate code is currently being used by a real-time call (lines 15-16) because the real-time call can use the codes assigned to this non-real-time call to improve its rate. If there are more than one free code with the same highest weight, the code with the smallest index *i* is chosen. Any further ties are broken randomly.

Example 1. In Figure 3(a), an OVSF code subtree is shown such that the code C(0,0) refers to an OVSF code with SF = X and the data rate of 8R, for $4 \le X \le 64$. Let $x_1 = 1$, $x_2 = 2$, and $x_3 = 3$. The OVSF code subtree is assumed to have initially a single RT call, namely, Call 1, that is assigned code C(3,0). When a new NRT call, namely, Call 2, requiring a code with SF = 4 is admitted to the network, there are three free codes with SF = 4: C(2,1), C(2,2), and C(2,3). Algorithm CBP determines the weights $W_{1,i}$ (lines 12–20) of the three free codes as 2, 3, and 3, respectively. Because the free codes C(2,2) and C(2,3) have the same weight $W_{1,i}$, we need to determine their $W_{2,j}$ values to try to choose a code with a higher weight. But, their $W_{2,j}$ values happen to be the same as well and, therefore, CBP chooses C(2,3)randomly to break the tie (line 23). Now, a new RT call (Call 3) requiring an SF = 8 is admitted to the network. There are five free codes, C(3, 1) to C(3, 5). Algorithm CBP determines the weights $W_{1,j}$ (lines 4–11) of the five free codes as 1, 3, 3, 3, and 3, respectively. CBP computes the $W_{2,i}$ values of codes C(3,2) to C(3,5) as 1, 1, 2, and 2, respectively, and then chooses one of C(3,4) and C(3,5) randomly (line 23). A new RT call (Call 4) requiring an SF = 4 is assigned code C(2,1) as it is the only code available with that SF. Now, a new RT call (Call 5) is assigned code C(3,5) since it has

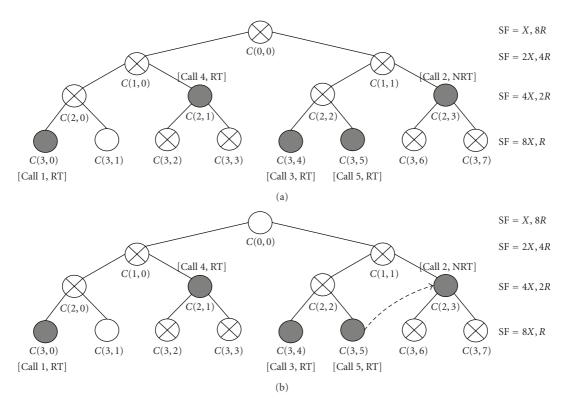


FIGURE 3: An example for showing the operation of Algorithm CBP over an OVSF code subtree, where the code C(0,0) refers to an OVSF code with SF = X and the data rate of 8R, for $4 \le X \le 64$. Let $x_1 = 1$, $x_2 = 2$, and $x_3 = 3$. (a) The OVSF code subtree is assumed to have initially a single real-time (RT) call, namely, Call 1, that is assigned code C(3,0). When a new non-real-time (NRT) call, namely, Call 2, requiring a code with SF = 4 is admitted to the network, there are three free codes with SF = 4: C(2,1), C(2,2), and C(2,3). Algorithm CBP determines the weights $W_{1,j}$ (lines 12–20) of the three free codes as 2, 3, and 3, respectively. Because the free codes C(2,2) and C(2,3) have the same weight $W_{1,j}$, we need to determine their $W_{2,j}$ values to try to choose a code with a higher weight. But, their $W_{2,j}$ values happen to be the same as well and, therefore, CBP chooses C(2,3) randomly to break the tie (line 23). The initial assignments of codes to calls 3, 4, and 5 are done similarly. (b) When calls 1 and 5 experience poor channel conditions during dynamic bandwidth allocation, they are assigned the higher rate codes C(2,0), and C(2,3), respectively, to compensate the data rate decline due to deteriorating channel conditions.

a higher $W_{2,j}$ over the code C(3,1) (note that C(3,5) and C(3,1) have the same value of $W_{1,j}$). Figure 3(b) illustrates how dynamic code allocation can be implemented when Call 1 and Call 5 experience poor channel conditions. In this case, Call 1 and Call 5 are assigned the higher rate codes C(2,0) and C(2,3), respectively, to compensate the data rate decline due to deteriorating channel conditions.

4.2. Class-Based Code Reassignment (CBR) Algorithm. The objective of the CBR algorithm is to find and free the most desirable code (as mentioned in Section 4.1), when a new or existing call needs code reassignments. To achieve this, the CBR algorithm performs reassignments under three cases, namely, to assign code to a new call, a real-time call requesting a higher-rate code, and a non-real-time call to ensure fairness. Lines 1 to 4 handle the first case, where a new call requiring an SF of s requires code reassignments because of code blocking. On line 2, all blocked codes of required SF s are found. On line 3, the algorithm first finds all the codes of SF s that have the maximum weight $W_{i,l}$ as defined in the CBP algorithm. On line 4, the algorithm then finds the code j that has the least number of descendant codes

that are assigned to real-time calls. This leads the number of reassignments to be low for real-time calls. Lines 5 to 7 handle the second case, where a code for reassignment is chosen when a real-time call needs to increase its data rate and when none of its x-hop affinity-mate is available. In this case, a code that is assigned or blocked by only non-real-time calls is reassigned so that the other real-time calls are not affected by this reassignment. If there are more than one code available for reassignment, code *j* that has the least number of descendant codes that are assigned to real-time calls is chosen for reassignment. Lines 8 to 10 handle the third case, where code reassignments are performed to ensure fairness in bandwidth allocation for non-real-time calls. In this case, a non-real-time call with a high CRD value is assigned the code of a non-real-time call with a low CRD value. The busy descendant codes of code j are then freed by doing code reassignments on lines 12-13, and finally the code j is assigned to the incoming call on line 14. Figure 4 shows an example for the operation of Algorithm CBR.

4.3. Dynamic Bandwidth Allocation (DBA) Algorithm. Before describing the algorithm DBA, we first explain the CRD

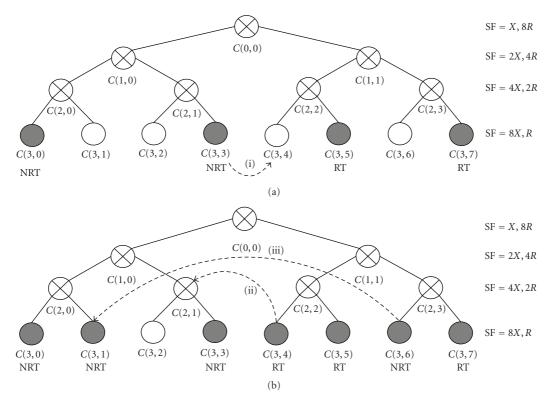


FIGURE 4: An example for showing the operation of CBR Algorithm. (a) It is assumed that codes C(3,0), C(3,3), C(3,5), and C(3,7) are already assigned to the NRT, NRT, RT, and RT calls, respectively. When a new RT call arrives and requests an SF = 4, the NRT call assigned to code C(3,3) is reassigned to code C(3,4) as shown by arrow (i), and code C(2,1) can now be assigned to the new call (lines 1–4). (b) It is assumed that all codes except C(3,2) are already assigned some RT and NRT calls, and that the call that is assigned C(3,4) needs a higher rate code to meet its delay requirements. Arrow (ii) shows that the RT call of code C(3,4) is reassigned a new code C(2,1) (lines 5–7). NRT call assigned to code C(3,3) will not receive any code assignment for the current frame. In case, an NRT flow assigned to code C(3,6) requires a fair share of the bandwidth, reassignments are performed as shown by arrow (iii). The NRT call assigned to code C(3,6) is assigned the code C(3,1) assigned to another NRT call whose CRD value is low (lines 8–10).

threshold for non-real-time flows. CRD threshold, denoted by CRD_{th}, refers to the maximum amount of rate degradation that a non-real-time flow can tolerate because of a real-time flow assigned to one of its x-hop affinity-mates. The DBA algorithm does not allow a non-real-time flow to use a code other than the x-hop affinity-mate codes if $CRD_i \leq CRD_{th}$. However, if CRD_i becomes greater than CRD_{th} , any code can be assigned to a non-real-time flow to ensure a fairness bound in rate assignment.

Lines 14 to 23 of the DBA algorithm assign codes to real-time flows by first considering conversational flows only and then streaming flows. On line 16, a real-time flow f_i with the highest CRD value is picked up. If CRD_i of flow f_i is greater than zero, the flow is assigned a higher rate code by calling procedure $ASSIGN_HRC$. Thus, when the difference between the network capacity and the total aggregate capacity of the real-time flows is more than the new rate requirement of flow f_i , lines 17 and 18 of the algorithm ensure that the delay requirement of a flow is always met. Lines 24 to 32 of the algorithm assign codes to non-real-time flows. On line 26, if a non-real-time flow with a high CRD value has its CRD value greater than 0, and if its code $C_i(m,k)$ is not available due to its assignment to a real-time flow, procedure $ASSIGN_HRC$ (see Algorithm 4) is called to

improve the CRD value of the flow. This step ensures fairness among non-real-time flows. The basic idea behind procedure *ASSIGN_HRC* is to first increase rate of the flow by assigning a higher-rate affinity-mate code (lines 1 to 5) and then to assign a higher rate non-affinity-mate code if an affinity-mate code is not available (6 to 14).

5. Class-Based Fair Code Allocation (CFCA) Protocol for OFCDM-Based 4G Wireless Systems

Variable spreading factor orthogonal frequency division multiplexing (VSF-OFCDM) has been proposed as the air interface by NTT-DoCoMo [7] for 4G broadband cellular networks. 4G networks are expected to support data rates of up to 100 Mbps for vehicular users and up to 5 Gbps for pedestrian users. VSF-OFCDM uses two-dimensional spreading of data bits over frequency and time domains to control the multicode interference while taking advantage of the frequency diversity gain [7–11]. Basically, in VSF-OFCDM, each data symbol of call i is first spread over the time domain using a time domain spreading code $C_{i,\text{time}}$ of spreading factor $SF_{i,\text{time}}$ and then each time domain symbol

```
Procedure ASSIGN_HRC(i, CRD_i, C_i(m, k))
Input: A flow f_i with CRD<sub>i</sub> and C_i(m, k). The x-hop affinity-mate codes of C_i(m, k) are known for
1 \le x \le max\_hops. The CRD threshold for non-real-time traffic is denoted by CRD<sub>th</sub>.
Output: Flow f_i is assigned a higher rate code.
begin
 (1) for j = 1 to max\_hops do
        Compute CRD<sub>j</sub> for all flows assigned to j-hop affinity-mate codes of C_i(m, k).
        if there exists an available j-hop affinity-mate of the required rate that is not assigned to a
        real-time call code for which CRD_i = 0 then
           Assign this j-hop affinity-mate code to flow f_i. Exit ASSIGN HRC.
 (4)
  (5) endfor
 (6) if (none of the above affinity-mate codes are available) and (flow f_i is real-time) and (the residual
     network capacity can support the average rate of flow f_i) then
        Call CBR to assign a free code to f_i by doing code reassignments among non-real-time flows,
         so that CRD_i is reduced.
 (8) else if (flow f_i is non-real-time) and (code C_i(m, k) of flow f_i is not available) and (there is residual
     network capacity left) then
        if (CRD_i > CRD_{th}) then
           Call CBR to assign a free code to f_i by doing code reassignments among non-real-time flows,
            so that CRD; is reduced.
 (11) endif
 (12) else
        Use the same code C_i(m,k) of flow f_i in this frame as well if it is available. Otherwise no code
         is assigned to flow f_i for this frame.
 (14) endif
end
```

ALGORITHM 4: Procedure ASSIGN_HRC.

is spread over $SF_{i,freq}$ orthogonal subcarriers in the frequency domain. The overall spreading factor, SF, used to spread each call's data symbol is therefore given as $SF_i = SF_{i,time} \times SF_{i,freq}$.

Increasing time domain spreading reduces intrauser multicode interference, whereas increasing frequency domain spreading increases frequency diversity. Thus a trade-off between reduction of multicode interference and increased frequency domain spreading can be achieved by varying SF_{i,time} and SF_{i,freq}. Especially, when the number of users using the same time domain code increases, the loss in signal quality due to multi-code interferece exceeds the gain in signal quality achieved through frequency diversity gain. For example, as shown in Figure 5, OVSF code tree can be used to allocate the time domain and frequency domain spreading codes to calls for two dimensional spreading. Calls A, B, and C are assigned OVSF codes C(3,0), C(2,1), and C(3,6), respectively. In Figure 5(a), calls A and B use code C(1,0) as the time domain spreading code. In order to satisfy the requirement $SF = SF_{time} \times SF_{freq}$, the frequency domain spreading codes of A and B are C(2,0) and C(1,1), respectively. The frequency domain codes are determined by considering the time domain code as the root code of the OVSF tree as shown by the dotted triangle in the figure. In Figure 5(b), calls A and B use codes C(2,0) and C(2,1) as the time domain codes, respectively. This increases time domain spreading and also reduces intrauser multicode interference as the time domain codes are used by only one call. The frequency domain codes are again determined by considering the time domain codes as the root codes and they would be C(1,0) and C(0,0) for calls A and B, respectively.

The constraints on OVSF code assignment in VSF-OFCDM would be as follows: (i) each call i should be assigned an exclusive OVSF code C(m, k) of spreading factor SF_{2^m} ; (ii) the time domain code $C_{i,\text{time}}$ of call i should not be the ancestor of the time domain code, $C_{j,\text{time}}$ of any other call j; however, $C_{i,\text{time}}$ code of two or more users can be the same, (iii) the frequency domain spreading factor $SF_{i,\text{freq}}$ is determined so that $SF_i = SF_{i,\text{time}} \times SF_{i,\text{freq}}$.

Once an OVSF code of spreading factor $SF_i = SF_{i,time} \times SF_{i,freq}$ is assigned, $SF_{i,time}$ and $SF_{i,freq}$ can be varied as long as constraint (ii) is satisfied. However, just varying $SF_{i,time}$ is not sufficient to meet rate and delay requirements of users experiencing poor channel conditions. Hence in this case, the assigned OVSF code of spreading factor SF_i has to be varied just like in the case of WCDMA. In the following subsections we will show how CFCA protocol can be used with VSF-OFCDM-based 4G systems to provide delay and rate guarantees under time varying channel conditions.

5.1. Class-Based Code Placement for VSF-OFCDM. As shown in Section 4.1, the basic idea behind algorithm CBP is to choose a free code for a new call that can cause the minimal overhead of code reassignments in case of poor channel conditions. Algorithm CBP accomplishes this by assigning an RT call a code adjacent to a free code or a code assigned to an NRT call. This is useful even in the case of VSF-OFCDM systems, because when a higher-rate code is assigned to an RT call during poor channel conditions, the intrauser multicode interference will also be reduced as the NRT calls assigned to the affinity-mate codes will not share the time

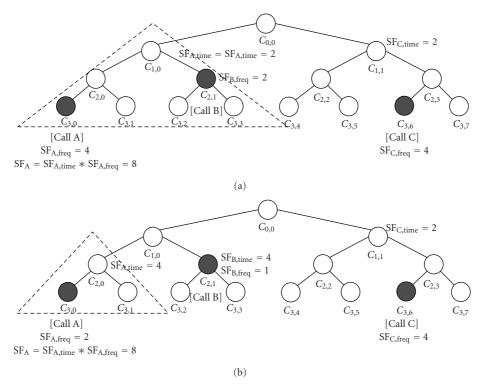


FIGURE 5: An example of code assignment in VSF-OFCDM. Calls A and B share the time domain code C(1,0) in (a). (b) shows how descendant codes of C(1,0) are assigned to calls A and B to reduce the multicode interference.

domain code used by the RT call. Algorithm VSF-OFCDM-CBP enhances algorithm CBP by also choosing the time domain code and frequency domain code for the new call. The time and frequency domain codes are chosen so that the call can take advantage of frequency diversity gain while keeping the intrauser multicode interference low on the time domain code as shown in Algorithm VSF-OFCDM-CBP in Algorithm 5 .

Algorithm VSF-OFCDM-CBP first calls algorithm CBP to assign an available free code C(m,k) for call i on line 1. On lines 2 to 18, the time domain and frequency domain codes are determined. On lines 2 and 3, an initial assignment of time domain code is made so that it is an ancestor code of C(m,k) that is either already used as a time domain code for some other call. If there is no such ancestor code that is already used as a time domain code for other calls, then on line 5, an ancestor code that does not have any descendant codes already used as time domain codes is used as a time domain code. This is done so that the constraint (ii) mentioned above that the time domain code $C_{i,\text{time}}$ of call i should not be the ancestor of the time domain code $C_{j,\text{time}}$ of any other call j is satisfied.

On lines 7 to 17, the load L on the time domain code is checked to see that the intra user multicode interference does not exceed the threshold $L_{\rm thresh}$. If L exceeds the threshold, the time domain spreading is increased to reduce the intra user multicode interference. On lines 10 to 14, the time domain spreading factor of any other calls using code C(l,k) as the time domain code is also reduced. This is repeated until

either the load L is less than the $L_{\rm thresh}$ or the time domain spreading factor cannot be increased any further. The load L can be a simple function of the number of calls using the same time domain code.

5.2. Class-Based Code Replacement for VSF-OFCDM. This is similar to the CBR algorithm for WCDMA networks except that whenever a code is reassigned, the time domain code has to be determined in a similar fashion as done in the Algorithm CBP-VSF-OFCDM.

5.3. Dynamic Bandwidth Allocation for VSF-OFCDM. The algorithm proposed for WCDMA networks is used without any modifications. Any changes to the time domain and frequency domain spreading are done by the VSF-OFCDM-CBP and CBR algorithms. Whenever the DBA algorithm assigns a higher-rate code, the number of users sharing the time domain code decreases as some of the NRT calls assigned to affinity-mate codes will not be scheduled. This not only reduces frequency diversity gain but also reduces intrauser multicode interference.

6. Performance Analysis

When channel conditions are normal during the whole data transmission, all real-time flows meet their delay requirements because the code placement algorithm CBP assigns those codes that provide $R_{i,avg}$. However, when channel

Input: A new call is admitted to the network because there exists at least one free code to support the requested data rate.

Output: The new call is assigned a free code C(m, k). The time domain code C(m, k) and the frequency code C(m, k) and C(

Output: The new call is assigned a free code C(m,k). The time domain code $C(m_t,k_t)$ and the frequency domain code $C(m_f,k_f)$ are also determined for the call.

begin

- (1) Call Algorithm CBP to determine a candidate code C(m, k) of $SF_i = 2^m$ for call i.
- (2) **if** (an ancestor code of C(m,k) is already being used as a time domain code for some other call) **then**
- (3) Determine the ancestor code C(l, u) of code C(m, k) such that it is already used as a time domain code for some other call.
- (4) else
- (5) Determine the ancestor code C(l, u) of code C(m, k) such that none of the descendants of C(l, u) are being used as a time domain code for any other call.
- (6) endif
- (7) Let C(l, u) be the time domain code. $SF_{i,time} = 2_l$.
- (8) Compute the load L on the time domain code C(l, u).
- (9) while $(L > L_{\text{thresh}} \text{ and } SF_{i,\text{time}} < SF_i)$ do
- (10) Let r be the number of calls using C(l, u) as the time domain code. Label them 1 to r from left to right.
- (11) **for** j = 1 to r **do**
- (12) $SF_{i,time} = SF_{i,time} \times 2$.
- (13) $SF_{j,freq} = SF_j/SF_{j,time}$.
- (14) endfor
- (15) Determine the descendant code of code C(l, u) that can be the new time domain code for call i, l = l + 1 and $u = k/2^{(m-l)}$.
- (16) Recompute the load L on the new time domain code C(l, u).
- (17) endwhile
- (18) Assign C(l, u) as the time domain code $C(m_t, k_t)$ for call i. Determine the frequency domain code $C(m_f, k_f)$ for call i so that $m_f = m m_t$ and $k_f = k k_t \times 2^{m_f}$

end

ALGORITHM 5: Algorithm VSF-OFCDM-CBP.

conditions deteriorate, the instantaneous data rate of flow f_i may drop to data rates below $R_{i,avg}$. Let $R_{i,worst}$ denote the lowest data rate that real-time flow f_i can achieve under poor channel conditions using OVSF code $C_i(m,k)$. Transmission of data at this lower data rate under poor channel conditions is made possible through the use of adaptive modulation and coding schemes. But, lowering the data rate may lead to delay violations. A higher rate code can achieve a data rate of $R_{i,avg}$ under poor channel conditions because of the reduction in spreading factor. Therefore, the algorithm DBA aims to meet the delay guarantee of each real-time flow by assigning a higher data rate OVSF code under poor channel conditions.

Lemma 1. Consider a real-time flow f_i with the average data rate $R_{i,avg}$, the poor-channel data rate $R_{i,worst}$, and CRD_i . Assume that when flow f_i is admitted to the network, it is assigned initially a code supporting the data rate $R_{i,avg}$ under normal channel conditions, which is sufficient to meet the deadline guarantee of f_i . When the data rate of flow f_i later drops to $R_{i,worst}$ due to poor channel conditions, the algorithm DBA can still meet the delay guarantee of flow f_i , provided that the channel conditions are not worse than the expected poor channel conditions in Step 1 of CFCA.

Proof. See the appendix.

As for the non-real-time flows, when the admitted real-time flows do not experience poor channel conditions, CFCA allows the residual network capacity to be used fairly by non-real-time flows. The following lemma indicates that the fairness bound among non-real-time flows is derived by showing that the CRD difference, denoted by CRD $_{\rm diff}$, of any two non-real-time flows is bounded by CRD $_{\rm th}$.

Lemma 2. Consider n continuously backlogged non-real-time flows, f_1, f_2, \ldots, f_n , for $n \ge 1$. The algorithm DBA guarantees that the difference between the CRD values of any two non-real-time flows f_i and f_j with the same channel conditions does not exceed a positive constant CRD_{th} , that is, $CRD_{diff}(t) = |CRD_i(t) - CRD_j(t)| \le CRD_{th}$.

Proof. See the appendix.

7. Simulation Results

This section presents the simulation results for the performance of the CFCA protocol using a simulator written in *C*. The simulation environment consists of 100 mobiles uniformly distributed in a cell of radius 3000 meters. Multiple simulation runs are done for a duration of 5000 frames. The call arrival rate follows the Poisson distribution with a mean arrival rate of 1 to 10 calls per second. The traffic types of the

calls are generated using a uniform distribution so that there are equal number of calls for each traffic class (conversational class 0 to background class 3). The requested average rate of calls is uniformly distributed between 75 and 1200 Kbps. The call holding time is exponentially distributed with a mean holding time of 10 seconds (i.e., 1000 frames for a frame time of 10 milliseconds). During the call holding time, a number of packets are generated using an exponential distribution with mean packet size of 500 bits.

A simplified E_b/I_o model as given in (2) is used as the channel model. Intracell orthogonality factor is set to 0.4. Path loss PL_i^j is computed as a function of distance D_i^j as given in (4), where δ is chosen as 4. Slow fading, S_i^j is considered to be log-normally distributed around the distance based path loss PL_i^j with zero mean and a standard deviation of 8 dB. A random way-point mobility model is used. Fast fading, F_i^j is generated using a Rayleigh fading distribution with zero mean and a standard deviation of 12 dB. The value of v in (16) is chosen as 20 frames, and the values of w for the four traffic classes are made equal to 2, 5, 10, and 20, respectively. CRD_{th} is set equal to 0.4 for the purpose of simulations. The value of x in x-hop affinity-mate is set to 3.

For the purpose of comparison with the CFCA protocol, we choose four other schemes that are distinguishable from the CFCA protocol in the choice of the code placement algorithm and/or the dynamic bandwidth allocation algorithm. We call the first scheme the first come first served (FCFS). The FCFS scheme makes use of the leftmost first code (LFC) placement algorithm that assigns the leftmost available free code to a new call. The FCFS scheme serves the flows during dynamic bandwidth allocation in the order of their arrival; that is, there is no special mechanism neither to increase the data rate of flows experiencing delay violations nor to prioritize the flows based on fairness conditions. The second scheme is called as the (CFC-DBA) scheme, and as the name implies, this scheme makes use of the crowded first code placement algorithm presented in [14] and uses the DBA algorithm proposed in this paper to prioritize and schedule the flows dynamically. However, the DBA algorithm in the (CFC-DBA) scheme chooses an arbitrary code to increase the bandwidth allocation instead of a higher-rate code that is not an *x*-hop affinity-mate.

The third scheme we used for comparison is scheme 3 of the dynamic bandwidth allocation algorithm presented in [3]. We have chosen scheme 3 of [3] because of its better performance results over the schemes 1 and 2 presented in [3]. We call this scheme CHAU in this paper. This scheme uses a credit-based scheme to prioritize real-time and non-real-time flows. The dynamic bandwidth allocation presented in [3] tries to assign a flow with the highest priority (credit) a code to meet its requested peak rate, which is greater than the requested average rate of the flow. If a code of the requested peak rate is not available, a descendant code of highest possible rate is assigned. As a result, a flow with lower priority may not get a code assignment in a given time slot. The fourth scheme we used for comparison is the one presented in [34]. We call this scheme KAM. In this scheme,

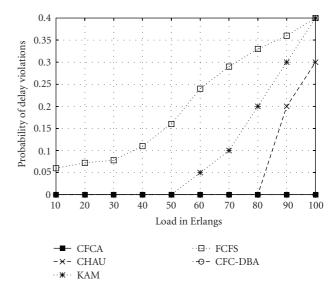


FIGURE 6: Probability of delay violations versus load.

flows are again prioritized using a credit-based scheme and flows with highest priority are assigned higher rate ancestor codes as long as they have back logged data greater than the rate supported by the higher-rate code and enough credits equivalent to the credits corresponding to the higher-rate code.

In the simulations we assume that (i) none of the realtime flows experience worse channel conditions than the expected poor channel conditions, and (ii) a packet has a delay bound of 1 packet transmission time, which is equal to the size of the packet divided by the requested average rate of the flow. With these assumptions, Figure 6 shows the probability of delay violations experienced by real-time flows. In case of CFCA and (CFC-DBA), the number of delay violations of real-time packets is 0 at all loads. The admission control presented in Section 4 and the DBA algorithm help CFCA and (CFC-DBA) achieve this result. Since (CFC-DBA) does not use the concept of x-hop affinity-mate, it will incur more signaling overhead, which is shown in Figure 8. FCFS scheme has a delay violation probability at all loads because FCFS does not do any dynamic bandwidth allocation to compensate for the loss in data rate due to poor channel conditions. At very low loads of up to 50 Erlangs, CHAU and KAM achieve a very low delay violation probability of 0. At low loads, the OVSF code tree has free capacity and CHAU and KAM allocate this free capacity to real-time flows with poor channel conditions. As the load increases, the delay violation probability increases for CHAU and KAM. CHAU, and KAM do not distinguish between real-time and non-real-time flows during prioritization. As a result, realtime flows may not receive a higher priority sometimes in code allocation. The reason for KAM to have higher delay violations than CHAU is due to the fact that KAM only tries to assign a higher rate ancestor code and two real-time calls assigned to sibling codes might block each other. Figure 7 shows the average delay experienced by real-time packets. At low loads, CHAU and KAM have lower average delay than

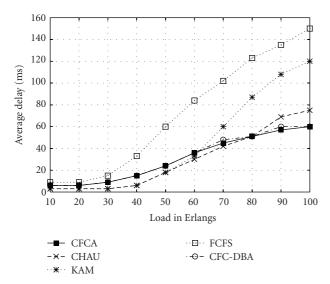


FIGURE 7: Average delay versus load.

CFCA. This can be attributed to the fact CHAU and KAM always try to assign a higher-rate code when the OVSF code tree has enough capacity, unlike CFCA that assigns a higher-rate code only when the value of CRD becomes greater than 0

While any code placement scheme can achieve a delay performance comparable to that of our scheme using our DBA algorithm, the drawback of such schemes is that they do not consider the overhead of code allocation in dynamic bandwidth allocation. This is evident in Figure 8 where (CFC-DBA) has a much higher overhead when compared to CFCA, though they both use the same DBA algorithm. The FCFS scheme has no overhead, but it incurs significant delay, rate, and fairness violations. In case of CHAU and (CFC-DBA) any code assignment is informed using the entire branch and layer numbers of the code. Therefore, the signaling overhead is high. CFCA and KAM use fewer bits to inform code assignments during dynamic bandwidth allocation because they almost always use ancestor codes during dynamic bandwidth allocation. KAM has a lower signaling overhead when compared to CFCA because KAM only uses its ancestor codes for reassignments, whereas CFCA sometimes assigns a code that is not an x-hop affinity-mate. In our simulations we found the reduction in the control overhead because of the CFCA protocol that makes use of the concept of x-hop affinity-mate and KAM to be 60% less than the (CFC+DBA) and CHAU scheme as shown in Figure 8.

To measure the fairness for non-real-time traffic types from the perspective of satisfying average rates, we employ the satisfaction index parameter SI defined in [30, 45] as SI = $(1/n)((\sum_{i=1}^n x_i)^2/\sum_{i=1}^n x_i^2)$, where x_i is equal to 1 if a flow receives at least the average requested data rate and is equal to $R_i(t)/R_{(i,avg)}$ otherwise. If SI equals 1, 100% of the flows meet their average rate requirements fairly. Thus, this serves as a measure of how fairly the average rate requirements of flows are met. As can be seen from Figure 9,

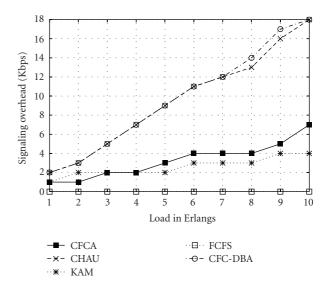


FIGURE 8: Code reassignments overhead in dynamic bandwidth scheduling.

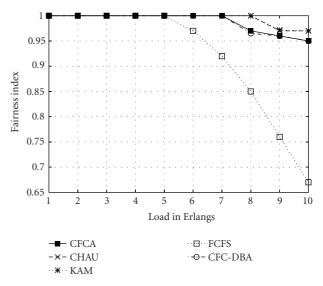


FIGURE 9: Satisfaction index to measure fairness.

the CFCA, CHAU, and KAM achieve better user satisfaction. For CFCA, at a load of 100 Erlangs, 95% of the flows meet their average rate requirements fairly. CHAU and KAM have a slightly higher satisfaction index than CFCA because, CHAU and KAM treat real-time and non-real-time flows in the same fashion, whereas CFCA assigns a higher priority to real-time flows over non-real-time flows. This results in some unfairness to non-real-time flows because real-time flows may use the codes originally assigned to non-real-time flows.

8. Conclusion

The performance of WCDMA- and VSF-OFCDM-based cellular networks depends on the proper utilization of OVSF codes. OVSF codes suffer from the code blocking problem.

Hence, OVSF code allocation and reassignment algorithms have significant impact on the performance of WCDMAbased cellular networks. Dynamic bandwidth allocation, which is done to meet rate and delay requirements in WCDMA systems involves dynamic OVSF code assignments. But, dynamic code assignments involve significant control overhead because of code blocking. Therefore, code allocation should be designed with dynamic code assignment in mind so that signaling overhead of DCA is low. This paper has proposed a code allocation algorithm, CBP, that allocates codes to flows by considering the possibility of assigning higher rate codes to flows when they experience poor channel conditions. The CBP algorithm reduces the overhead of code reassignments through the concept of x-hop affinity-mate. The proposed code reassignment algorithm, CBR, reduces the number of reassignments experienced by real-time calls, while assisting the DBA algorithm in meeting the delay and rate requirements of calls. The CRD metric prioritizes the flows for dynamic bandwidth allocation and ensures fairness in terms of both delay and rate. Delay and rate guarantees for real-time flows are met as long as their channel conditions are not worse than the expected poor channel conditions. Simulation results show that the proposed CFCA protocol reduces the control overhead for reassignments by 60% at high network loads for WCDMA systems. The proposed VSF-OFCDM-CBP algorithm, determines the time domain and frequency domain spreading factors so that the frequency domain spreading is maximized while keeping the intrauser multicode interference on time domain codes low.

Appendix

Proof of Lemma 1. In order to show that the algorithm DBA meets the delay guarantee of flow f_i , it suffices to show that flow f_i does not achieve a data rate less than its average data rate in more than w_i consecutive time slots out of v consecutive time slots. When flow f_i experiences poor channel conditions, its CRD_i value increases above zero. Line 17 of the DBA algorithm assigns a higher-rate code to a real-time flow, when it finds the CRD_i value of the flow to be greater than zero. Thus, in the worst case, CRD_i of flow f_i would be greater than zero in only one window out of v/w_i windows. Therefore, as long as higher-rate code and required power budget can be allocated to the real-time flow f_i , the average data rate would not be met in only w_i time slots out of v time slots.

This indicates that, in order to prove the lemma, we need to show that a higher-rate code can be assigned to a real-time flow when its CRD_i becomes greater than zero. Assignment of a higher-rate code for a frame is possible as long as the network has residual network capacity after allocating $R_{i,avg}$ for all real-time flows. Step 1 of CFCA ensures that this residual network capacity is available as long as the channel conditions are not worse than the expected poor channel conditions. To prove that a higher-rate code is available and not blocked, there are three cases to be considered in ASSIGN_HRC, depending on the status of the affinity-mate codes of flow f_i 's code.

Case 1. In this case, it is assumed that the x-hop affinity-mate codes of f_i 's code are free, for $1 \le x \le \max$ hops. In lines 1 to 5 of ASSIGN_HRC, flow f_i is assigned such a free affinity-mate code that makes CRD_i equal to zero.

Case 2. In this case, it is assumed that the x-hop affinity-mates of f_i 's code are not free, but at least one of them is assigned to a non-real-time flow. Because f_i has higher priority than the non-real-time flow, the higher rate affinity-mate code of f_i is assigned to f_i to make CRD $_i$ zero.

Case 3. In this case, it is assumed that all the x-hop affinity-mate codes of f_i 's code are already assigned to real-time flow. In lines 6 to 7 of ASSIGN_HRC, flow f_i is assigned a code by doing code reassignments in the algorithm CBR, so that CRD $_i$ is made equal to zero.

Proof of Lemma 2. Consider two continuously backlogged flows f_i and f_i that have identical channel conditions. The worst case condition for the fairness bound occurs when one non-real-time flow f_i is always assigned its requested data rate whereas another flow f_i is not assigned any data rate. In this case, CRD_{diff} (i.e., the difference in the CRD values of f_i and f_i) becomes the maximum. Recall that a CRD value ranges from 0 to 1, so that the maximum difference between two CRD values could be 1. Let us assume that CRD_{diff} = CRD_{th} at the current frame, where CRD_{th} < 1. In addition, because a CRD is computed over ν time slots (or frames), the value of a CRD can increase or decrease by at most 1/vover a frame time. Note that if the CRD values of both flows decrease or increase at the same time, then CRD_{diff} does not change. Therefore, the only case where CRD_{diff} exceeds CRD_{th} at the next frame is that the CRD value of one flow increases while the CRD value of the other flow decreases. We will show next that the algorithms DBA and ASSIGN_HRC do not allow this case to occur.

Without loss of generality, let us assume that $CRD_i = 0$ and $CRD_j = CRD_{th}$ at current frame time. Because CRD_j equals CRD_{th} , line 25 in the algorithm DBA chooses flow f_j due to its higher CRD value for code allocation. Because $CRD_j > 0$, line 27 in the algorithm DBA invokes ASSIGN_HRC to assign a higher-rate code to flow f_j . If line 10 in ASSIGN_HRC assigns a free code to flow f_j , CRD_j is reduced and, therefore, CRD_{diff} does not exceed CRD_{th} . On the other hand, if line 10 in ASSIGN_HRC cannot assign a free code to flow f_j , CRD_i and the CRD value of any other flow including f_j increase, thereby keeping CRD_{diff} the same. (Note that if f_j cannot be assigned a code, the other non-real-time flows cannot be assigned a code either because f_j has the highest priority due to its highest CRD value.) It follows that CRD_{diff} does not increase beyond CRD_{th} at any frame time

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