THE PERFORMANCE OF SIR–BASED HYBRID LINK ADAPTATION ALGORITHMS IN MOBILE RADIO NETWORKS

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SUMMARY

In this article we have described and simulated new SIR-frame based link adaptation algorithms. Algorithms were designed to increase efficiency of data transmission among user equipment and base stations (uplink). Simulation results of hybrid adaptation (power and modulation BPSK, QPSK, 16-QAM, 64-QAM) are compared and expressed as data throughput and outage probability for different simulation environments (pedestrian channel with mobile subscriber speed 10 km/s and vehicular channel with speed 120 km/h).

Keywords: *data transmission efficiency, link adaptation algorithm, hybrid adaptation*

1. INDRODUCTION

We have implemented new hybrid link adaptation algorithms (modulation schemes BPSK, 16-QAM, 64-QAM and enhanced power control) in to the model of radio chain [4, 5]. This model is main part of the WCDMA (3G technology [1, 3, 8]) mobile radio network simulator [10], which was created in the Matlab environment. The simulator allows comparing various modifications of SIR-based hybrid link adaptation algorithms and efficiency of high-order modulation schemes using. The present evolution status of 3G mobile network proves that using high-order modulation schemes is possible. The HSDPA (High Speed Downlink Packet Access) is one example of using 16-QAM in real WCDMA system [11, 12].

The simulator of WCDMA network consists of optional number of base stations, which serve to traced mobile stations ("served MSs") [6]. In each cell there are generated also other mobile stations ("non-served MSs"), which cause intercell and intracell interference. OVSF codes for served mobile stations are from the same branches of the code tree. On the contrary, OVSF codes from different branches are chosen for non-served mobile stations (better cross-correlation functions [5, 7]).

2. SIR-FRAME BASED ALGORITHMS

The SIR-frame based adaptation algorithm was derived from 3GPP specifications for UMTS system [1] and also new block of decision, which modulation schema will be used for data transmission, was added (Fig. 1). Input parameters are initial (actual) modulation scheme MOD_A and required one, MOD_R. The Open loop power control (OpLPC) sets up the initial output power for the mobile station transmitter P_{MS_out} [dBm]

$$P_{MS\_out} \geq P_{BS\_out} - \left(L_{PATH-LOSS} + L_{RAYLEIGH} + L_{SHADOW}\right) \qquad (1)$$

where P_{BS_out} [dBm] is power of Dedicated Physical Control Channel, $L_{PATH-LOSS}$ [dB] is the path loss of selected environment, $L_{RAYLEIGH}$ [dB] is the Rayleigh fading represented by Clark's model [4] and L_{SHADOW} [dB] is log-normal shadow fading. Simulation results of time slots transmission among MSs and appropriate BSs are as follow: actual BER (BER_A), intracell and intercell interference and actual SIR (SIR_A) values. The Closed loop power control (CLPC) is used for MS transmission power adjustment after time slot transmission. It can be expressed by next part of algorithm program code:

```
if(SIR_A≥SIR_R) then P_MS_out - ΔP;
else if (SIR_A<SIR_R) then P_MS_out + ΔP;
end;
```

where ΔP [dB] is the power control step. The Outer loop power control (OuLPC) adjusts SIR_R value of used modulation or can set the higher-order (lower-order) modulation scheme after frame transmission according to SIR_A. This can be expressed by next part of algorithm program code:

```
if (BER_A>BER_R) and (P_MS_out = P_MS_max) then set lower-order modulation scheme as MOD_A;
else if (SIR_A > SIR_R_act-ΔSIR) and (SIR_A <SIR_R_act + ΔSIR) then do not change MOD_A;
else if (SIR_A ≤ SIR_R_act - ΔSIR) then set lower-order modulation scheme as MOD_A;
else if (SIR_A ≥ SIR_R_act - ΔSIR) then set higher-order modulation scheme as MOD_A;
end;
```

where SIR_{R_act} [dB] is required SIR for actual modulation scheme. The adjusting of modulation scheme is according to actual radio channel conditions (Non-forced algorithm). SIR_R is adjusted according to:

```
if (BER_A >BER_R) then
      if (MOD_A ≤ MOD_R) then SIR_R + ΔSIR;
      else SIR_R - ΔSIR;
      end;
else if (BER_A < BER_R) then
        if (MOD_A=MOD_R) then keep SIR_R in
           the actual values (+/-ΔSIR);
        else if (MOD_A > MOD_R) then
                     SIR_R - ΔSIR;
              else   SIR_R + ΔSIR;
              end;
        end;
end;
```

We have organized SIR-frame based algorithms (Fig. 1) into next groups:

1. **Non-forced** - the modulation scheme is adjusted by the successive changing of SIR_R value. This algorithm was described in detail at the beginning of chap. 2.
2. **Forced** - the modulation is changed by forced switching. We have developed four methods of forced modulation switching:

   - **Forced Soft** – if BER > BER_R, $P_{MS\_out} < P_{MS\_max}$ and MOD > MOD_R then SIR_R is continuously (soft) decreased by ΔSIR value, but if $P_{MS\_out} = P_{MS\_max}$ the modulation scheme is forced switched to more robust one:

```
if (BER_A > BER_R) then
   if (P_MS_out = P_MS_max) then
         set lower-order modulation
         scheme  as MOD_A;
   else if (MOD_A ≤ MOD_R) then
                  SIR_R + ΔSIR;
         else     SIR_R - ΔSIR;
         end;
   end;
else if (BER_A ≤ BER_R) then
       if (MOD_A = MOD_R) then
             keeps SIR_R in the actual
             values (+/-ΔSIR);
       else if (P_MS_out = P_MS_max) then
              if (MOD_A ≤ MOD_R) then
                 set lower-order
                 modulation scheme
                 as MOD_A;
              end;
              SIR_R - ΔSIR;
   else if (MOD_A > MOD_R) then
                 SIR_R - ΔSIR;
        else  SIR_R + ΔSIR;
        end;
   end;
end;
```

   - **Forced Hard** – if BER > BER_R, $P_{MS\_out} < P_{MS\_max}$ and MOD > MOD_R then the modulation is switched to MOD_R. The algorithm for the condition BER < BER_R is the same as in forced soft algorithm.

```
if (BER_A > BER_R) then
     if (P_MS_out = P_MS_max) then
           set lower-order modulation
           scheme  as MOD_A;
     else if (MOD_A ≤ MOD_R) then
                SIR_R + ΔSIR;
          else set initial MOD_R;
          end;
     end;
end;
```

   - **Forced Soft Return** and **Forced Hard Return** - in these algorithms we have developed the tool for return of modulation to requested one (MOD_R), if the channel is rapidly changing (the transmission is longer or the mobile speed is higher). On the other side, if MS achieved the initial requested modulation scheme, BER > BER_R and $P_{MS\_out} < P_{MS\ max}$, the MOD_R is locked in specified interval (MOD_R +/- optional safety interval) until one of the inequalities is broken. The forced soft and the forced hard algorithms keep only last set required modulation (according to actual radio channel state). This OuLPC condition is used to keep required modulation scheme as long as possible and limits too often modulation schemes switching.

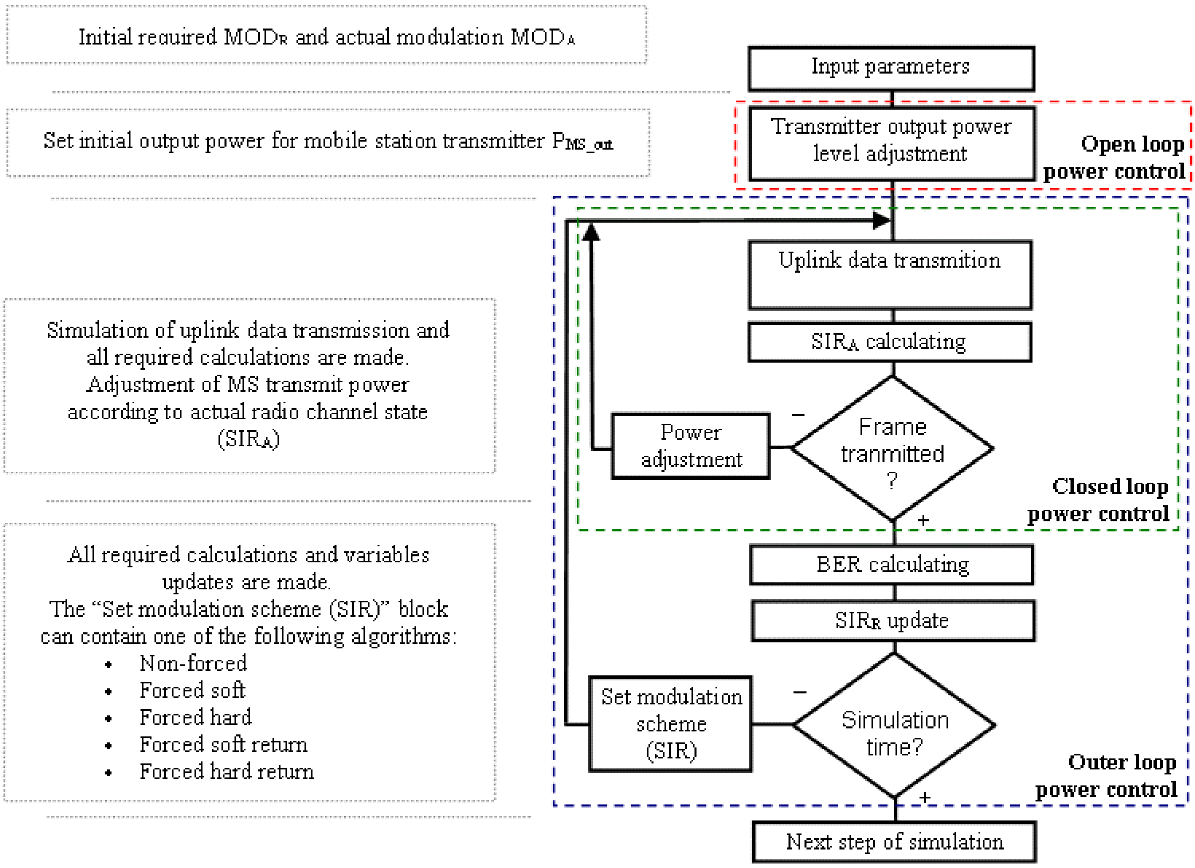

Fig. 1 Diagram of SIR-based algorithm

3. RESULTS

The simulated mobile radio network was created by 9 cells. Inside of each cell were randomly placed one traced MS and 10 interfering MSs. Two types of environments were used [3, 6]:

1. **Pedestrian environment** – average MS velocity was 10km/h and Pedestrian (A&B) channel model was used [3].
2. **Vehicular environment** – average MS velocity was 120km/h and Vehicular (A) channel model was used [3].

The simulator was designed to save all parameters of uplink radio channel and mobile stations when the first simulation is finished. Next simulations can use saved parameters with different algorithms. This property ensures comparability of simulation results [10]. We have compared algorithms by two parameters: modified data rate R_{mod} and satisfied user outage probability P_{out_su} [3]. Modified data rate is average data rate with regard to satisfied user:

$$R_{\mathrm{mod}} = \frac{\sum N_{b\_\mathrm{mod}}}{t_{sim}} \qquad (2)$$

where N_{b_mod} is the number of correct transferred bits except time interval t_{err}:

$$t_{err} = \sum_{i=1}^{M} t_{0,042\_i} \qquad (3)$$

where M is the number of time intervals $t_{0,042\_i}$ [s]. Duration of the time interval $t_{0,042\_i}$ is $0,042 t_{sim}$ with BER > $\mathrm{BER_R}$. Time interval $t_{0,042\_i}$ is set by satisfied user requirement [3], in which the user is satisfied, if he is not interrupted. The session is interrupted if BER > $\mathrm{BER_R}$ during the time longer than t_{dropp}:

$$t_{dropp} = \max\left(5, \frac{10}{R_t \cdot \mathrm{BER_R}}\right) \qquad (4)$$

We supposed (for real time services) mean session duration 120 sec [3], so t_{dropp} is 5/120 = 0.042 %. The second result of simulations is the probability of outage with regard to satisfied user P_{out_su}:

$$P_{out\_su} = \frac{t_{err}}{t_{sim}} \qquad (5)$$

The results of simulations are shown on Fig. 2 and 3.

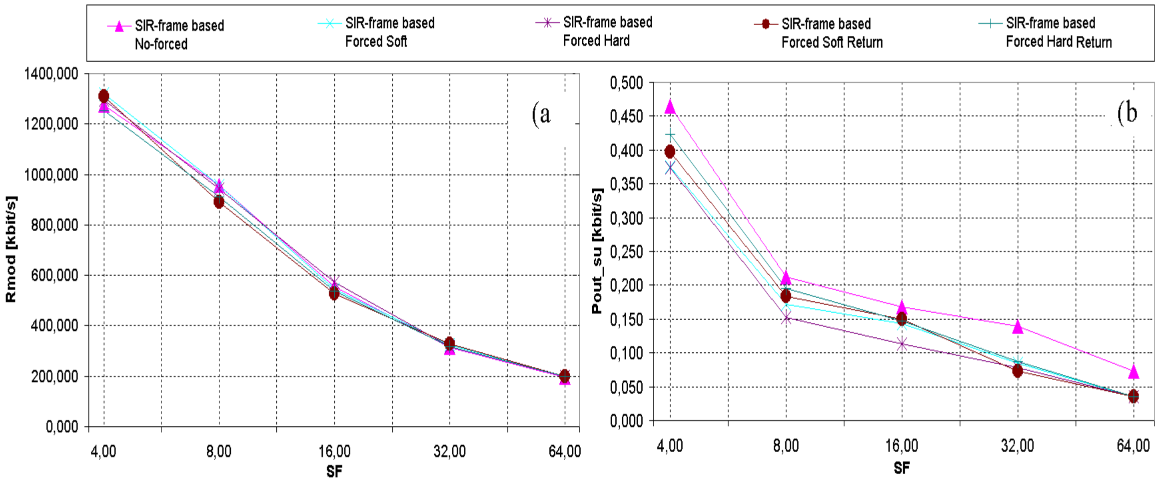

Fig. 2 a) R_{mod} for Pedestrian environment b) P_{out_su} for Pedestrian environment

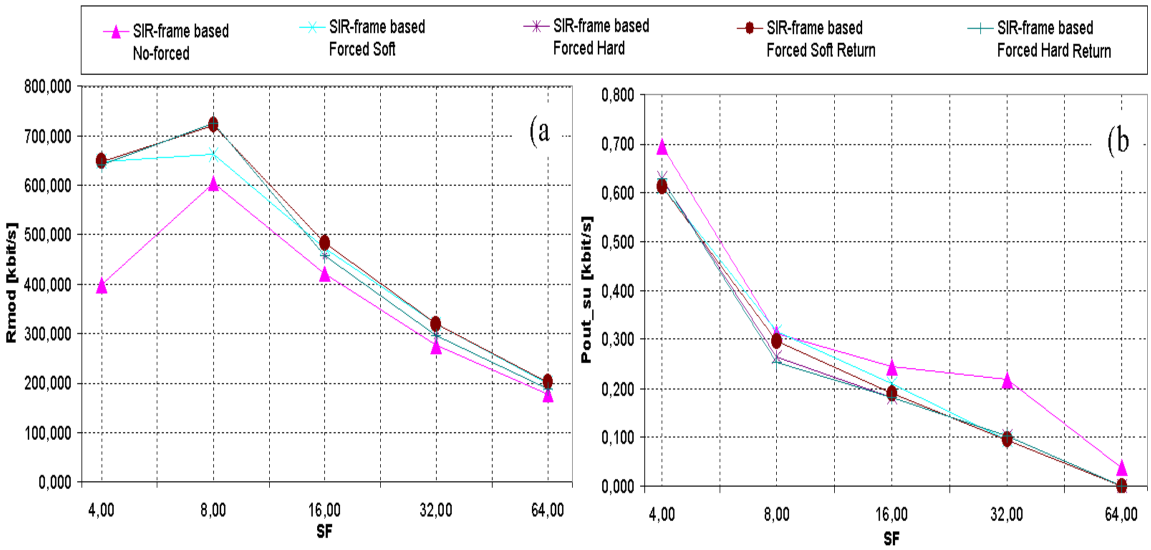

Fig. 3 a) R_{mod} for Pedestrian environment b) P_{out_su} for Pedestrian environment

4. CONCLUSION

The simulation results show that the data rate is almost the same for each link adaptation algorithm in Pedestrian environment (Fig. 2a), but the P_{out_su} is better for algorithms where mechanism for return to required modulation scheme is implemented (Fig. 2b). The average difference between P_{out_su} for non-forced and P_{out_su} for forced with return algorithm is 0.045. The similar results were achieved for Vehicular environment (Fig. 3a and 3b). The R_{mod} values are lower due to mobile subscriber speed (120km/h), but there is bigger difference between non-forced and forced with return algorithm (average deviation is 80kbit/s). The average difference between P_{out_su} for non-forced and P_{out_su} for forced with return algorithm is 0.078. If we are comparing results for algorithms simulated in Vehicular environment, we can observed that forced with return algorithms achieved higher data rate than non-forced algorithm, and simultaneously P_{out_su} is lower for forced with return algorithms than non forced algorithm. The improvement is visible on the Fig. 3a and 3b.

The placement of actual modulation scheme control block to the outer loop power control (SIR-frame based algorithm) allows change of modulation (required SIR) according to actual channel state (radio channel fading) once per radio frame (10ms). According to this property, the user equipment should transmit data with higher-order modulation, if radio channel state is insufficient until whole radio frame is transmitted. Therefore is logical to place actual modulation scheme control block to the closed loop power control (SIR-slot based algorithm). The modulation schema can be changed once per time slot (0.667ms). We expected improvement of Rmod (increasing) and P_{out_su}

(decreasing), because in the case, when channel state is changed from the sufficient state to the insufficient one after the first slot transmission, only data contained in the second time slot are lost. The third time slot is transmitted by using lower-order modulation scheme (and data from the second slot can be restored by using channel and source decoder [7] or by using ARQ method [2, 9]). We suppose that information about uplink channel state is available immediately.

Our next goal is to make our model more realistic with regard to transmission delays (information about radio channel state) and to the wrong transmitted data recovery capacity.

ACKNOWLEDGEMENTS

The authors gratefully acknowledge support from the VEGA project No. 1/0140/03 "Effective radio resources management methods in next generations of mobile communication networks" and State project No. 2003 SP 51/028 09 00/028 09 10 "Communication Networks and Services of New Generations" in conjunction with the preparation of this paper.

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BIOGRAPHIES

Vladimír Wieser was born in Púchov in 1954. He received the M.S. degree in electrical engineering and communication from Military Academy Brno, Czech Republic, in 1978 and Ph.D. degree from Military Academy Liptovský Mikuláš in 1996. Since 2001 he works as Ass. Prof. in Telecommunication Department of University of Žilina, Slovak Republic. His research includes mobile communication networks, especially power and rate adaptation, radio resource management.

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