Coexistence of speech and best effort services in Enhanced Uplink WCDMA

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Abstract— An evaluation of the performance of coexistent voice and best effort data users in Enhanced Uplink WCDMA is studied in this paper. The main focus is on deriving the capacity regions and compare with previous WCDMA releases. It is shown that the Enhanced Uplink yields a large capacity gain in many aspects for all fractions of voice users compared to previous WCDMA releases. It is also shown, by the cumulative distribution functions of noise rise at the capacity limits, that the best effort data users experience bad quality at lower noise rise than voice users. This means that the capacity is in fact limited by the best effort users.

I. INTRODUCTION

The increasing use of data services and the importance of IP based services requires the uplink transmission to manage high speed data rates. Within the 3rd generation partnership (3GPP) a concept for enhancing the uplink transmission, called Enhanced Uplink, is being developed [1]. The overall goal is to improve coverage and throughput as well as to reduce the delay of the uplink. One of the requirements that have been agreed on within 3GPP is that the enhanced uplink channels must be able to coexist with already existing WCDMA releases. For example, the enhanced uplink must not impact seriously on real-time services, such as speech, carried on current WCDMA channels.

The focus of this work is to study the consequences of introducing the enhanced uplink in a WCDMA network. Specifically the impact on quality and capacity of speech carried on release 5 WCDMA channels when introducing the Enhanced Uplink in the same network will be studied. The aim is to demonstrate the trade-off between speech and best-effort performances and the profits of the Enhanced Uplink compared to previous WCDMA releases. The performance evaluations are based on simulations.

The paper is organized as follows. The simulation model is described in Section II. Performance measures and evaluation criteria are defined in Section III. Section IV contains simulation results and evaluations. Concluding remarks are made in Section V.

II. SIMULATION MODEL

This section gives a brief description of the simulation models and assumptions used therein.

A. Propagation Model

The propagation model characterizes the channel qualities by the attenuation of transmitted signals. The attenuation is the inverse of the path gain. The path gain, G, is thus the ratio of the received power to the transmitted power and it consists of four different parts,

\[ G = G_a G_d G_s G_m. \]

Here, \( G_a \) is the antenna gain, \( G_d \) is the distance attenuation, \( G_s \) is the shadow fading and \( G_m \) is the multipath fading. The antenna gains and distance attenuations are given as lookup tables. The lookup table for distance attenuation is calculated using the Okumura-Hata model.

The shadow fading in logarithmic scale is modelled in the simulator by a normal distributed variable, with mean \( \mu = 0 \) and standard deviation \( \sigma = 8 \). The shadow fading is also assumed to be correlated with decorrelation distance 100 meters.

The multipath fading is modelled using the standardized 3GPP Typical Urban model.

B. Simulation Scenarios

1) Traffic Model: The simulated traffic consists of both speech and best-effort data. Both user types are based on Poisson processes for arrival. The speech model includes a voice activity process, where the user is active 60% of the time. The speech calls are modelled with an exponential talk time with an average of 90 seconds.

The best-effort traffic consists of a combined MMS and e-mail traffic model, where MMS occurs with 60% probability and e-mail with 40% probability. The packets are of random size with mean 12.7 kB for MMS and 60 kB for e-mail. The simulator also models the TCP flow control for packet based data.

2) Cell Deployment: The simulator models a two-dimensional environment and maintains positions for base stations and users. The simulation environment consists of seven sites, each with a three sector antenna. This yields 21 cells, forming a uniform hexagonal pattern. To avoid border effects the plan is repeated through a wrap-around technique. The cell radius is set to 500 meters. Users are initially placed randomly throughout the simulated area according to a uniform distribution.
3) Simulation Logging: The simulated time is set to 200 seconds, and the logging starts after 20 seconds when the traffic is assumed stable. Information on the total system is being logged every 2 ms. However information about each user, such as the number of transmitted blocks, is logged only for the total simulated time and not for each time instant, due to memory limitations.

C. System Model

This section briefly describes the most significant assumptions and operations used to model the system.

1) Admission Control: The system load is measured in noise rise, $\eta$. Noise rise is the ratio of the total received power to the background noise power,

$$\eta = \frac{I_{tot}}{N_0}.$$  

When a user requests a channel, the noise rise that will be caused by the user is estimated. The estimate is added to the current system noise rise and if this noise rise estimation exceeds the maximum allowed noise rise of $7\ dB$, the user will not be admitted to the system. For voice users this means that they will be blocked and not given a channel. For best-effort users it means that they will actually be admitted to the system, but not allowed to start the transmission. Hence a best-effort user will actually experience a very bad bit rate because of the waiting time rather than being blocked.

The blocking of voice users will only be used in the voice only simulations with no best effort data users in the system. To model the prioritization of voice users, they are always admitted when letting simultaneous best effort users into the system. This might seem a bit simplified, however it will be shown that the capacity is in fact limited by the best effort users.

2) Fast HARQ: Retransmissions are modelled with a stop-and-wait protocol. This means that no retransmission attempt is done before a negative acknowledgement is received. For the Enhanced Uplink with 2 ms TTI, the simulator uses five parallel queues. The number of parallel queues are set such that the time to go through all the queues are approximately the same as the HARQ round trip time, so when the first queue is handled again, the Acknowledgement (ACK) or Negative Acknowledgement (NACK) has been received by the UE. Also, the time to go through the queues and the round trip time being as close together as possible, reduces the delay.

The outer loop power control model allows four transmission attempts for the Enhanced Uplink, meaning that the probability that the transmission is still incorrect after the fourth transmission is $1\%$. Allowing multiple transmission attempts increases capacity, because UEs need to use less power and retransmissions are done to the Node B which is closer than the RNC and hence faster. If the number of retransmissions exceeds its maximum, a Radio Link Control (RLC) retransmission is triggered, which will perform a retransmission from the RNC.

When errors occur, all blocks transmitted in the current TTI will be retransmitted. The successive attempts are soft combined, using a model for chase combining, where each retransmission is an exact copy of the original transmission. Chase combining is modelled by adding the SIR for each transmission attempt. The transport format and resource combination is not changed if retransmission is required.

3) Node B Rate Control: Node B rate control with a busy indicator is modelled. The node B controls the maximum bit rate, $R_{\text{max}}$, i.e. the maximum number of blocks that a Enhanced Uplink user is allowed to transmit in each TTI. The busy indicator is set when the noise rise exceeds a certain value and unset when it falls below the same value.

If the busy flag is set, a new user with data in the transmit buffer sends a rate request to the Node B. If the busy flag is set the user will not be admitted, but users that already have a radio link will continue the transmission with bit rate $\leq R_{\text{max}}$. However, if the busy flag is set when the rate request is received by the Node B, the maximum bit rate will be decreased and hence the users that are already transmitting are forced to lower their bit rates. Now the new user can be admitted and allowed to transmit with bit rate less or equal to the new $R_{\text{max}}$.

III. EVALUATION METHODS

This section presents the performance measures and criteria used to evaluate the simulations. We will start by defining the requirements for a user to be unsatisfied.

Definition 3.1 (Unsatisfied voice user): A voice user is unsatisfied iff

- The user is blocked

or

- $\text{BLER} \geq 1\%$

Note that the first criterium that the user is blocked will only be used in the voice only simulation. For the mixed service simulations, we only consider BLER.

Best effort users will neither be blocked nor experience block errors. The quality for best effort users will instead be measured in normalized packet bit rate or normalized packet delay. Normalized packet bit rate is the data rate experienced by the user. It is calculated by dividing the size of the message by the time the user spent waiting until the message was transmitted.

$$\text{normalized packet bit rate} = \frac{\text{message size}}{\text{waiting time}} [\text{kbps}]$$

Normalized packet delay is the inverse of normalized packet bit rate. Using these definitions we can also define an unsatisfied best effort user.

Definition 3.2 (Unsatisfied best effort user): A best effort user is unsatisfied iff

- Normalized packet bit rate $\leq 10 \text{ kbps} \iff$ Normalized packet delay $\geq 0.1 \text{ s/kbits}$

Capacity is measured in throughput per cell, where the number of data bits that have been delivered within the simulated time is considered as throughput. Using the above
definitions, we define the capacity measure applied to evaluate the simulation results.

**Definition 3.3 (Capacity):** Capacity is defined as the maximum throughput per cell such that the fraction of unsatisfied voice users < 5% and the fraction of unsatisfied best-effort users < 5%.

The capacity criteria can also be expressed as the normalized packet delay of the 95th percentile of data packets should be less than 0.1 s/kbits and the 95th percentile of voice users should have BLER ≤ 1%

### IV. Simulation Results

Table I shows the most significant simulation parameters that have not already been discussed in section II.

<table>
<thead>
<tr>
<th>TABLE I</th>
<th>SIMULATION PARAMETERS.</th>
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<td>Overall Parameters</td>
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<td>Admission control</td>
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<td>Maximum noise rise</td>
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<td>Noise rise for busy flag</td>
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<td>Best Effort Parameters</td>
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<td>Power Control (PC)</td>
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<td>Enhanced Uplink</td>
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<td></td>
<td>Voice Parameters</td>
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We start by deriving the capacity for the single service cases, i.e. with only one service users in the system at the time. First we consider the voice only case. Figure 1 shows the fraction of unsatisfied voice users versus offered voice traffic load for this simulation. With the evaluation criteria defined earlier, we derive the capacity for voice. The scale is normalized such that the voice capacity = 1. This normalized scale will be used for voice loads throughout the whole paper.

Next we consider the best effort data only case. Figure 2 shows the 95th percentile normalized packet delay versus best effort load for Enhanced Uplink and release 5. Again with the evaluation criteria defined earlier, we derive the capacity for the Enhanced Uplink and the release 5 scenario. The scale is normalized such that the release 5 throughput at the capacity limit = 1. The normalized scale will be used in the sequel of this paper. The enhanced uplink also uses soft combining, which means that data blocks that can not be correctly decoded are saved and combined with later retransmissions of the same blocks to find the correct data. With soft combining the number of retransmissions are reduced. Fast HARQ with soft combining leads to higher capacity and robustness against link adaption errors.

Moving the rate control to the Node B reduces delays which leads to a rapid adaption and a tight control of uplink interference. The fast rate control also allows admission control in the RNC to be more relaxed. A larger number of burst high rate users can be allowed. This in all yields a higher uplink capacity.

Table II shows the mean noise rise at the single service capacity limits. We see that best effort data users carried on release 5 experience bad quality, i.e. low bit rate, for a lower noise rise than the Enhanced Uplink. Voice users manage even more noise rise before experiencing bad quality. The same
 behaviour is seen in the two service case, with coexistent voice and best effort data users. This leads to that the capacity is in fact limited by the best effort data users.

**TABLE II**

<table>
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<tr>
<th>Service</th>
<th>Mean noise rise</th>
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<tr>
<td>Voice</td>
<td>5.96 dB</td>
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<tr>
<td>Enhanced Uplink</td>
<td>3.49 dB</td>
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<tr>
<td>Release 5</td>
<td>1.47 dB</td>
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Now we consider coexistence of voice and best effort data. Deriving the capacity for all simulated voice loads in the same way as before yields the capacities shown in Figure 4.

### A. At the Capacity Limit

This section discusses the differences and gains of introducing the Enhanced Uplink compared to release 5 at the capacity limits.

1) **Best Effort Throughput:** Figure 4 shows the maximum best effort data throughput versus voice load for Enhanced Uplink and release 5 respectively. The areas below the curves define the capacity regions for Enhanced Uplink and release 5 respectively. We see for example that the best effort data carried on release 5 will get no throughput at all approximately above 0.45 of the voice capacity. The same limit for Enhanced Uplink is around 0.80 of the voice capacity.

Figure 5 shows the gain in capacity with Enhanced Uplink, i.e. the gain in total system throughput. The Enhanced Uplink yields a relative gain of approximately 160% compared to release 5 for all voice loads for which it is possible to get a best effort data throughput.

2) **Resource Utilization:** One of the expectations on the Enhanced Uplink is a better resource utilization. We have already seen that the Enhanced Uplink is more noise rise resistant than data carried on release 5 channels. This leads to a higher utilization of the available noise rise. Figure 6 shows the ratio of the mean noise rise at the capacity limits to the maximum noise rise, which in simulations is set to 7 dB. We call this ratio resource efficiency, $\rho$.

$$\rho = \frac{\eta}{\eta_{\text{max}}}$$

Resource efficiency might seem as a bad measure of how good the resource is being utilized. It should be no problem to get a high noise rise without getting a good throughput. However it is a good measure of how much of the resource is being utilized. Assuming that the amount of utilized resource is correlated to a good throughput validates this measure. Figure 7 shows that the Enhanced Uplink in fact uses the caused noise rise even more efficient than data carried on release 5 channels. The figure shows the total system throughput per noise rise (dB) at the capacity limits. We see that the Enhanced Uplink gets a higher throughput per noise rise than release 5 channels.

Figure 8 shows the cumulative distribution functions (CDF) for noise rise. The noise rise CDF for voice is plotted at the capacity limit. The noise rise CDFs for best effort data are plotted at the simulated load just above the capacity limit, so that the noise rise at the capacity limit is actually lower. The figure shows, videlicet that voice users are the most noise rise
resistant, best effort data carried on release 5 channels are most noise rise sensitive and Enhanced Uplink is in between. Hence it also shows that the limitation of not blocking voice users when having coexistent best effort users does not affect the result, since the best effort users are limiting the capacity.

V. CONCLUSIONS

The main focus of this work has been on coexistent voice and best effort data. The work has shown that the Enhanced Uplink has many advantages over previous WCDMA releases. The Enhanced Uplink yields a gain in packet bit rate of approximately 15% at the lowest loads. The Enhanced Uplink also yields a higher throughput for a given noise rise. The work has shown a total throughput increase of approximately 160% for all voice loads compared to WCDMA release 5.

REFERENCES