Mixed Traffic in UMTS Downlink

Antonio Capone, Matteo Cesana, Giovanni D'Onofrio, and Luigi Fratta

***Abstract*—The traffic being transferred within 3G mobile networks will be composed by different information flows with various constraints on the required QoS (bit rate, delays, etc.). In this scenario, flexibility will be a key point for the success of 3G systems. UMTS (Universal Mobile Telecommunication System) offers both circuit switched and packet switched transfer mode, and within each transfer mode, different QoS can be achieved by properly setting physical parameters such as the spreading factor of the physical channels, the power control scheme, the rate of the FEC protecting code, etc. In this letter, we give an evaluation of the downlink performance of W-CDMA UMTS radio interface when providing access to multimedia services. In particular, we analyze through simulations a typical scenario where voice calls and Web-browsing sessions share the same frequency carrier, the former using the dedicated channels (DCH), the latter being transferred on the downlink shared channel (DSCH).**

***Index Terms*—Mixed traffic, Universal Mobile Telecommunication System (UMTS), W-CDMA.**

I. Introduction

THE Universal Mobile Telecommunication System (UMTS) standardization bodies have designed a radio interface highly flexible able to provide different bearer services with different bit rates and different transfer modes [1], [2]. For example, circuit switched and packet switched transfer modes are available. Within each transfer mode different quality of service can be achieved by suitably setting physical layer parameters such as the spreading factors (SF) of the physical channels, the rate of the FEC (Forward Error Correction) code used to protect information bits, the target SIR of the power control procedure and the ARQ (Automatic Repeat reQuest) scheme [3].

The evaluation of the performance of UMTS radio access network in a single service scenario is a widely discussed topic in the literature. However, few works have appeared up to now which aim to analyze the UMTS system with mixed traffic scenarios [4], [5]. In [6] and [7], we have evaluated the impact of some choices on the configuration of UMTS W-CDMA radio interface, when considering packet service only with Web-browsing. It has been proved that the setting of the physical layer parameters has a deep impact on the performance of the radio interface.

Regarding the multimedia service scenario, 3G operators have to decide whether to reserve different frequency carriers to different services or to share a single frequency carrier between different services. In the latter case, when different services contend for the shared resources their performance characteristics may highly differ from the single service case. Therefore, an exhaustive analysis of the radio interface performance is of utmost importance, and can give useful insights to the 3G operators on how to exploit effectively the radio resources [8].

In this letter, we evaluate the performance of a mixed traffic scenario on the downlink of UMTS W-CDMA, where speech traffic and Web-browsing traffic share the same frequency carrier. This letter is organized as follows. In Section II, we describe the simulation scenario and discuss the obtained results, while Section III includes our concluding remarks.

TABLE I
GENERAL SIMULATION PARAMETERS

Topology Model	49 Macrocells, Radius=300m wrap-around domain
Propagation Model	$P_r = P_t 10^{\frac{\epsilon}{10}} L, \epsilon \sim N(0,\sigma^2)$ with $\sigma = 5dB$ $10 \log L = -(128.1 + 37.6 \log r)(dB)$
Power Limits	30dBm on DCH/DSCH 43dBm on Total BS power

TABLE II
VOICE/DATA SIMULATION PARAMETERS

	Data Traffic	**Voice Traffic**
Transport Channel	DSCH SF=4,8	DCH SF=128
FEC Code	R=1/2 Conv	R=1/3 Conv
Power Control	Closed Loop Every 10ms	Closed Loop Every 10ms
ARQ Scheme	Ideal	No ARQ
Traffic Model	ETSI WWW	Poisson Arrival Exp call duration

II. Simulation Analysis

Tables I and II summarize the parameters used in our simulations. In our model, speech traffic at 12.2 Kb/s is delivered in downlink dedicated channels (DCH) with spreading factor 128, while high bit rate Web-browsing sessions share one or more downlink shared channels (DSCH) with spreading factor 8. The choice of such a channel speed has been shown to be optimal [6].

Each cell has an omnidirectional antennas with unit gain located at the center and the generic user is assigned to the Base Station (BS) with the minimum radio attenuation.

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At the receiving side, the Signal to Interference Ratio (SIR) is evaluated for each transmission as

$$SIR = \frac{P_r \times SF}{\alpha I_{intra} + I_{inter} + P_N} \tag{1}$$

where SF is the spreading factor of the physical connection, P_r is the received power defined in Table I, P_N is the thermal noise assumed equal to -99 dBm, I_{inter} is the sum of the signal powers received from the other cells, I_{intra} is the sum of the signal powers received from other users in the same cell, and α is the loss-of-orthogonality factor due to the multipath that, according to [13], is assumed equal to 0.4.

Data and voice blocks are transmitted in 10 ms time-intervals called frames. At each frame and for each transmission the SIR is evaluated and used to derive the Block Error RAte (BLER) from BLER-versus-SIR curves obtained through link level simulation [6]. Each block is then considered correct or erroneous according to a random experiment based on the BLER. For the packet switched traffic only, an ideal ARQ procedure is adopted, i.e., the transmitted block is kept in the transmitting queue in case of error and is cancelled otherwise.

We implemented a typical closed loop power control procedure where, for the sake of simplicity, the power updates are requested every frame instead of every slot. We take into account the lower frequency of the power update by allowing a higher range of feasible updates. Every frame the power update can range from -16 dB to 16 dB, 1 dB for each slot in the frame.

Each voice call enters the system according to a Poisson arrival process and remains active for a period of time which is exponentially distributed with mean 60 s. Calls are offered at a very high rate, and are accepted in each cell up to a given value N so that the number of active calls per cell is almost always N. The traffic model adopted for data users is the WWW model proposed in [13].

The position of users, either voice or data, is uniformly selected in the service domain.

All the presented results have been obtained running steady-state simulations 900 s long. The first 100 s are used as warm-up time. The remaining 800 s are divided into four simulation runs. During each run the results are collected and used to calculate one sample of each statistical quantity used for evaluation. The output results have been tested according to the t-student statistical test. For all the measures reported in the following (throughput, BLER, etc) the confidence interval is under the 5%, given a confidence level of 95%.

The results of our investigations are summarized in Fig. 1, where the system capacity, i.e., the maximum total voice and data throughput, versus voice throughput is represented by the solid line.

In our simulations, voice and data traffic refer to two different QoS requirements: voice must be delivered with an average BLER not exceeding a target value, 10^{-2} in our case, while data blocks are delivered with no error and with an average delay not exceeding 500 ms. For N active voice calls per cell, the system capacity is obtained by increasing data traffic until either the delay constraint or the voice BLER constraint is exceeded.

Two main observations come from the results.

First, the system capacity with data only is 1250 kb/s, 25% higher than that with voice only, 976 kb/s, corresponding to 80 voice calls per cell. The reason is due to the different constraints.

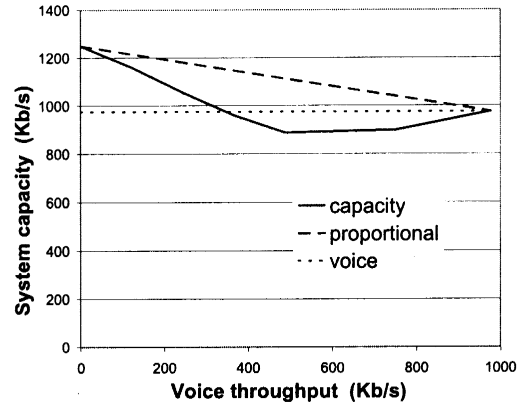

Fig. 1. System capacity as function of the voice throughput.

Voice circuits require a very low BLER value for all their lifetime. As more users are added, the power required to grant the target BLER increases until some users reach the maximum allowed value. The number of users admitted at this point, in a static condition, represents the system instantaneous capacity (IC). IC depends on the position of users and decreases if users suffer high attenuation. It also changes with time as users move or enter and leave the system. In a dynamic scenario, the voice capacity, i.e., the maximum number of users that can be admitted in any condition, is given by the lowest IC observed. In practice, as the lowest IC occurs rarely, to avoid an excessive penalty of the throughput, the maximum number of users is set equal to the IC that is not exceeded 99% of the time.

Similarly, we can define the IC for data as the maximum data throughput allowed in a frame in static conditions. However, data transmissions can operate temporarily beyond capacity, i.e., even if the required power is temporarily not available, since erred transmission can be repeated. As a matter of fact, the highest data throughput is observed with a BLER on the channel as high as 10%. In other words data can face interference fluctuations more effectively than voice due to retransmission.

The second observation is that the system capacity reduces when mixing voice and data. In fact, the capacity line in Fig. 1 is always below the "proportional" line, that is the capacity of the ideal case where mixing traffic does not affect efficiency. The proportional line is given by the straight line that connects the voice alone and data alone capacities.

Furthermore, we observe that, when adding data, the capacity decreases from the voice only point, i.e., below the "voice" line, that represents the capacity if data performed exactly as voice. Therefore, adding a few data requires to drop more voice channels. This behavior is due to the fact that the accuracy of the close-loop power control mechanism is impaired by the burstiness of data. In fact, the power control mechanism is affected by estimation errors when the interference changes, owing to changes in the number of transmissions, their level and their originating point. When power control underestimates the required power, the obtained SIR drops with respect to the target value and errors occur. With voice traffic, interference changes are comparatively rare because transmissions last for a long period of time. On the other hand, with data traffic, transmissions are very short and estimation errors occur more frequently.

In the mixed case, the burstiness of the interference affects voice channels. As voice can not tolerate burst errors, more

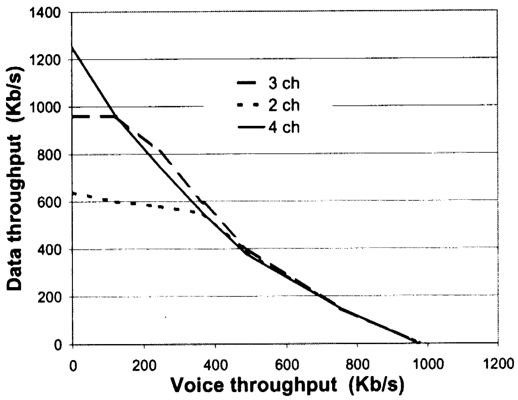

Fig. 2. Maximum throughput reached by data as function of the voice throughput, when using 2, 3, and 4 channels with $SF = 8$ and $R = 1/2$.

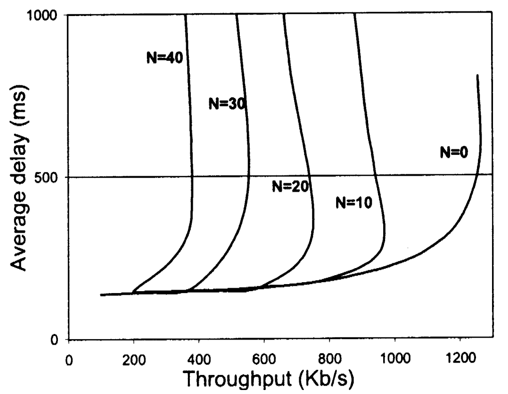

Fig. 3. Delay-throughput curve for data traffic when using 4 channels with $SF = 8$ and $R = 1/2$, whit a different number of voice calls.

power is used to increase the SIR of voice channels and, consequently, less data can be accommodated with respect to the hypothetical case in which data do not increase the burstiness. As data become the dominant traffic, the penalty on voice channels is offset by the intrinsic gain that data transmission presents over voice, and the capacity curve crosses the voice line, though it still remains beyond the proportional line.

More detailed results are represented in Fig. 2. Fig. 2 shows the maximum data throughput when data are transmitted with a different number of channels with $SF = 8$ and $R = 1/2$. The capacity curve of Fig. 1 has been derived by using the maximum data throughput over all cases. We observe that, when two or three channels are used, as voice decreases, the data throughput levels off, because the physical capacity of channels is reached. With four channels data are not limited by the physical capacity of channels. Note, however, that, although the four channels case remains optimum, as it was shown in [6] with data only, the three channels case performs a little better for mixed traffic, due to the decreased data burstiness. The case of two channels is never optimal because the data throughput is strongly limited by the delay constraint.

The average packet delay versus data throughput using four $SF = 8$ $R = 1/2$ channels, when a different number of voice calls are present is shown in Fig. 3. The performance of the downlink shared channel is obviously affected by the presence of voice calls interference, and drops dramatically if the number of voice users grows above 30.

III. Conclusions

In this letter, we have presented a performance evaluation of W-CDMA UMTS radio interface when providing access to speech and data users, obtained by a detailed system simulation.

Although our simulations refer to specific data and voice sources, the result discussion has a more general validity. In summary, we can support the three following statements.

- Data transfer alone is more efficient that voice transmission alone because data can take advantage of ARQ.
- Adding data impairs the performance of voice as higher SIR target must be used for voice to cope with the burstiness of data. In fact, we have verified that the power control mechanism is affected by estimation errors that increase as burstiness increases.
- Given an aggregate channel speed an optimal configuration exists in terms of number of channels per cell, their spreading factor and code rate. This is due to the burstiness and power control effectiveness. In the cases we have considered, the use of $SF = 8$ $R = 1/2$ has shown to be optimum.

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