Performance of Voice Calls with Rate Matching Optimization in UMTS Networks

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Abstract—Two transmission modes were standardized in UMTS: a Circuit Switched (CS) and a Packet Switched (PS) mode. In both cases, an efficient transmission framework was defined to allow the transmission of multimedia information with the required reliability, in spite of the medium characteristics. Multimedia calls are delivered in several transport channels - one for each media related to the call - properly multiplexed in one or more physical channels. These transport channels are designed to attain the required levels of QoS using two main techniques: Channel Coding and Rate Matching. In this paper we discuss the creation of transport channels for voice calls in both CS and PS modes and present a new methodology to optimize the Rate Matching parameters for a given channel coder and a set of QoS objectives. With the proposed methodology the maximum $E_b/N_0$ assigned to the transport channels is minimized, thus reducing the transmitted power.

Index Terms—UMTS, Rate Matching, Power control, Optimization

I. INTRODUCTION

In recent years, Internet and cellular applications have emerged as the major driving forces behind new developments in the area of telecommunications networks. Furthermore, there is a great interest in using IP technology to provide multimedia services in general.

Cellular systems are evolving from a purely telephony system to a multimedia service based on the IP technology which has been incorporated in 3G and particularly in the UMTS specifications [1],[2]. In fact, the UMTS system is in continuing evolution. There have been several releases, each of them providing a new version of specifications for vendors and operators to be used in system implementation. Presently, a reference architecture specifies the main network components of UMTS including interfaces between access network, core network and PSTN and also signaling network.

Two domains were delimited in the UMTS architecture: a Circuit Switched (CS) and a Packet Switched (PS) domains. 3GPP Release 5 introduced the IMS-(IP Multimedia Subsystem), intended to support multimedia traffic in general.

The main difference between UMTS and the second generation mobile CDMA system is the ability to offer simultaneous flexible services at various rates as high as 2 Mbit/s (in dedicated channels), that were not possible in the conventional cellular system. These services consist of voice, video, circuit switched data and packet switched data services. Different combinations of multiple services are offered by using multiple transport channels, each having their own characteristics such as CRC code length, channel coding schemes and coding rate. One very important part in UMTS that is responsible to provide flexible services is the Rate Matching process that is part of the transport channel coding and multiplexing mechanism. When combining two or more different services, it is necessary to adjust the channel transmission parameters in order to fulfill different Quality of Service (QoS) requirements. To sort this matter out, the Rate Matching unit is used to repeat or puncture coded bits, in order to achieve the necessary balancing effect.

Several papers have been published seeking to clarify, detail or discuss the UMTS next-generation architecture, in particular, the all IP scenario [3],[4]. A general description of the present UMTS architecture may be found in [5]. Concerning Rate Matching, there are several papers in the literature [6],[7],[11] considering the subject, but none of them gives a reasonable treatment to the aspect of optimization process. Furthermore, 3GPP does not consider any optimization methodology in the standardization documents related to the Rate Matching unit.

Here, we intend to address some important aspects related to the performance of voice calls in UMTS. We introduce a discussion on the relative performance of the end-to-end voice calls developed in CS and PS modes considering the air interface effect and the heavy signaling burden in the PS mode. In this context we focus on the rate matching process for which we propose an optimization procedure. This is capable to give the exact number of bits to repeat or puncture, in order to achieve the best commitment between the maximum required power for transmission and the required levels of QoS for each service. To the best of the author’s knowledge, the approach here investigated to optimally allocate repetition/puncturing bits has not yet been addressed in the technical literature.

This paper is organized as follows: Section 2 discusses the creation and multiplexing of transport channels. Section 3 presents the voice transmission process in CS and PS modes. Section 4 briefly describes the Rate Matching process while Section 5 formally introduces the related mathematical formulation for the optimization problem which leads to a set of features for the optimal solution. Section 6 shows two application examples based on practical cases related to UMTS specifications and finally section 7 presents the conclusions.

II. CREATION AND MULTIPLEXING OF TRANSPORT CHANNELS

The UMTS system is capable to simultaneously work with several media at varying rates and different reliability levels.
With the concept of “transport channels”, the information to be transmitted is conveniently allocated to one or more transport channels, each with its proper QoS level. Both the Circuit Switched (CS) and the Packet Switched (PS) modes use this concept.

Next, the Rate Matching block performs two processes: spans one, two, four or eight radio frames of 10 ms duration. Frame interleaving is then performed on a time window that frames, when more than one radio frame is transmitted. Inter-segmentation have the purpose of adapting the data to the radio 1/3 turbo coding (TC) or no coding. The RF equalization and convolutional coding (CC2), 1/3 convolutional coding (CC3), is performed, with four different coding possibilities: 1/2 limits specified for the block sizes. Next, the channel encoding either concatenated or segmented, in order to fit the allowable QoS needs. Both the Circuit Switched (CS) and the Packet Switched (PS) modes use this concept.

Figure 1 shows the process of creation and multiplexing of transport channels in the case of uplink transmission [8]. For each channel, after the CRC attachment, the data blocks are either concatenated or segmented, in order to fit the allowable limits specified for the block sizes. Next, the channel encoding is performed, with four different coding possibilities: 1/2 convolutional coding (CC2), 1/3 convolutional coding (CC3), 1/3 turbo coding (TC) or no coding. The RF equalization and segmentation have the purpose of adapting the data to the radio frames, when more than one radio frame is transmitted. Inter-frame interleaving is then performed on a time window that spans one, two, four or eight radio frames of 10 ms duration. Next, the Rate Matching block performs two processes:

- First, the sum of the instantaneous bit rates of all transport channels is adapted to the physical channel capacity using bits repetition or puncturing;
- Second, the mechanism of repetition and puncturing is used to differentiate the channels in respect to individual QoS needs. For example, in one situation, the channels with more stringent requirements will have assigned more repetition bits;

Every 10 ms, the rate matched outputs of all the transport channels are multiplexed, forming the Coded Composite Transport Channel (CCTrCH). After intra-frame interleaving and the modulation phase, the CCTrCH is allocated for transmission in one or more physical channels [8]. Note that the physical channel bit rate is determined by CDMA spread factor which may assume values from 4 to 512.

III. V OICE T RANSMISSION IN CS AND PS M ODES

In this section the process previously described will be detailed for the case of voice transmission in PS and CS modes of a FDD UMTS system using the AMR codec at 12.2 kbit/s.

In CS mode we have a total of four channels, as defined in [9]:

- **Channel A**: Carrying the most sensitive information bits (81 bits) of the AMR codec, with Error Probability (EP) of \( 5 \times 10^{-4} \). The 1/3 Convolutional Code (CC3) is used;
- **Channel B**: Carrying the medium sensitive information bits (103 bits), with Error Probability (EP) of \( 10^{-3} \). The 1/3 Convolutional Code (CC3) is used;
- **Channel C**: Carrying the least sensitive information bits (60 bits), with Error Probability (EP) of \( 5 \times 10^{-3} \). The 1/2 Convolutional Code (CC2) is used;
- **Channel D**: For signalization purposes (148 bits), with Error Probability (EP) of \( 10^{-4} \). The 1/3 Convolutional Code (CC3) is used;

Figure 2 illustrates the frame structure for these transport channels (a spread factor of 64 is assumed). The tail bits are used for the initialization of the respective coders. Note that the final number of bits depends of the Rate Matching unit.

![Fig. 2. Transmission of voice signal (CS mode)](image-url)
Voice call signaling packets: for telephony signaling purpose; they carry the same information of the channel D specified before.

For the sake of simplicity, let us consider a VoIP transmission with the most favorable condition, i.e. voice packets with header compression, simultaneously with voice call signaling packets. Figure 3 describes this situation.

![Fig. 3. Transmission of voice signal (PS mode)](image)

Here, we have channel A carrying voice multiplexed with channel B (like channel D of CS mode). Before the rate matching unit we have 639 bits (510+129) to be accommodated in physical channel. There are two options:

- The first one is to use a spread factor of 32, corresponding to the rate of 120 kbit/s. Then, to fit 639 bits in a radio frame of 10ms (1200 bits) we will have 561 bits (1200-639) to use as repetition bits, distributed over the two channels;

- The second one is to use a spread factor of 64 (like the CS case), corresponding to the rate of 60 kbit/s. Now we will have to use puncturing, for fitting 639 bits in radio frames of 600 bits.

IV. THE RATE MATCHING PROCESS

The receiver can take advantage of the repeated bits in order to improve the error rate. Denoting \(P_S(\epsilon)\) and \(P_D(\epsilon)\) respectively the bit error probability without and with bit repetition, then for the situation of \(N\) coded bits entering the rate matching unit and considering the addition of \(\Delta N\) repeated bits forming a sequence of \(N + \Delta N\) bits, the overall bit error probability is given by:

\[
P(\epsilon) = (\frac{\Delta N}{N}).P_D(\epsilon) + (1 - \frac{\Delta N}{N}).P_S(\epsilon)
\]

As an example, Figure 4 shows the error probability versus \(E_b/N_0\) for different relations of \(\Delta N/N\) in a typical BPSK channel strictly subjected to additive Gaussian white noise. There we can clearly see the improvement achieved in \(P(\epsilon)\) with the mechanism of bits repetition.

Suppose we have a multimedia transmission consisting of \(n\) channels to be multiplexed, one for each media, with the respective QoS objectives, expressed as a certain required value of BER. To cope with these BER values, each channel has to be transmitted with a proper \(E_b/N_0\) ratio. If the physical channel can accommodate more bits than the total of the \(n\) channels, it would be possible to distribute these extra bits (spaces) to the channels, duplicating one or more bits in each channel. Therefore it is possible to assign the number of repeated bits in each channel, as a way to manipulate the required \(E_b/N_0\). This is the Rate Matching objective. On the other hand, when the total number of bits surpasses the total bits for the \(n\) channels, bits must be removed (punctured) instead. The number of bits to be repeated (or punctured) is given by the upper levels of the protocol.

Unfortunately when a convolutional or turbo encoder is included, there is no closed form for BER calculation, being necessary to resort to computer simulation.

We emphasize that although the examples here presented involve coding and additive white gaussian channel, this approach is by no means limited to this case. On the contrary, any channel model (pedestrian, vehicular, indoor, etc..) that can be simulated or field measured, can provide the BER data that are needed.

Throughout this text we assume that the BER curves are functions of the original number of bits \(N\), the number of repeated/punctured bits \(\Delta N\) and the value of \(E_b/N_0\). These functions are assumed to be numerically available for all kind of encoders discussed. Appendix I presents one technique capable of generating BER curves from collected BER data.

Having these considerations in mind, a question may be posed: how should we distribute the repetition/puncturing bits among the transport channels in the best possible way. Certainly channels with stringent QoS levels deserve a special attention. On the other way, a precise definition of "best" solution that meet practical requirements should be given.

Next section is devoted to develop a formal solution of this problem. We present an alternative technique to solve it that is fairly simple from the computational viewpoint.

V. THE OPTIMIZATION PROCEDURE

Consider the situation where we have \(M\) transport channels. We wish to transmit \(N_i\) bits per radio frame on the \(i\)-th channel

![Fig. 4. BER Curves for BPSK Signal Corrupted by Additive Noise](image)
where the sum $N_T = \sum N_i$ is different than $N$, the total number of bits that are capable to be transmitted. Hence $\Delta N = N - N_T$ bits can be selected for repetition/puncturing, thus reducing/increasing the power requirements without sacrificing QoS. We define as $\Delta N_i$ the number of repeated/punctured bits in the ith-channel. Each channel is also characterized by its bit error probability $P_b$, since the QoS demands require that $\frac{\Delta N}{N}$ be minimized. Since the requirements of the above restrictions. Hence we can formulate our problem as one of optimizing $\Delta N_i$.

Throughout this work, we will assume that $P_i$ is a monotonically decreasing function on both arguments which is a fairly reasonable assumption. For the sake of notation’s simplicity, let us define $x_i = \Delta N_i$ and $y_i = (E_b/N_0)_i$.

These quantities should obey the following restrictions:

1. $x_i + N_i \geq 0$ and $y_i \geq 0$ for all $i \in \mathbb{N}$
2. $x_1 + x_2 + \ldots + x_M = \Delta N$
3. $P_i(x_i, y_i) \geq \alpha_i$ for all $i \in \mathbb{N}$

Let us define a set $\Omega$ made of all vectors of the form $(x, y) = (x_1, x_2, \ldots, x_M, y_1, y_2, \ldots, y_M)$ that obey the above restrictions. Since this set is not a singleton, one might argue which element of $\Omega$ is “the best”.

Among all possible ways of defining a quality criterion, one that seems more suitable for the problem in hand, is the minimization of the maximum $E_b/N_0$ value that meets the requirements. Hence we can formulate our problem as one of optimizing in which we seek the minimization of an artificial variable $w$ subject to:

1. $y_i \leq w$ for all $i \in \mathbb{N}$
2. $(x, y) \in \Omega$

In the optimization theory parlance, we are faced with an $(2M + 1)$ dimensional mixed-integer problem with linear objective function but with linear and nonlinear constraints. Problems of this sort tend to require sophisticated computer programs to be solve and to be time consuming.

We will drop for a moment the integrality condition imposed on the $x$ variables. Later we will discuss how this condition should be reinstated.

The main goal of this paper is to show that this problem can be solved by an alternative algorithm that does not require conventional optimization tools at all. On the contrary, this algorithm requires only simple procedures that can be implemented in any computer language with low programming cost and computational effort.

Let us start rewriting the optimization problem in a more convenient way. By defining a new decision variable as $\bar{x} = (x_1, x_2, \ldots, x_M, y_1, y_2, \ldots, y_M, w)$, the problem can be written as:

$$\min_{\bar{x}} z = f(\bar{x}) = w$$

subjected to:

$$h(\bar{x}) = \sum_{i=1}^{M} x_i - \Delta N = 0$$

and for $1 \leq k \leq M$:

$$g_k(\bar{x}) = x_k + N_k \geq 0$$

$$g_{k+M}(\bar{x}) = w - y_k \geq 0$$

$$g_{k+2M}(\bar{x}) = \alpha_k - P_i(x_k, y_k) \geq 0$$

(4)

Let $x^*$ a regular point of $\Omega$. The well known theorem of Karush-Kuhn-Tucker [10] states that if $x^*$ is a local minimum then there exists an scalar $\nu \in \mathbb{R}$ and non-negative vectors $\mu, \lambda, \eta \in \mathbb{R}^M$ such that:

$$\nabla f(x^*) - \nu \nabla h(x^*) - g^T \nabla g(x^*) = 0$$

$$\nu h(x^*) = 0$$

$$g^T \nabla g(x^*) = 0$$

where $g^T = [\mu^T, \lambda^T, \eta^T]$.

This theorem when applied to the current optimization problem reveals that, besides the fact that $\sum_{i=1}^{M} \lambda_i = 1$, the following relations must hold for each $k$ in the range $1, 2, \ldots, M$:

$$-\mu_k + y_k P_{ka}(x_k^*, y_k^*) - \nu = 0$$

$$\lambda_k + y_k P_{kb}(x_k^*, y_k^*) = 0$$

$$\mu_k (x_k^* + N_k) = 0$$

$$\lambda_k (w^* - y_k^*) = 0$$

$$\eta_k (\alpha_k - P_i(x_k^*, y_k^*)) = 0$$

$$\lambda_k, \eta_k \geq 0$$

where $P_{ka}$ and $P_{kb}$ represent the derivatives of function $P_i$ respectively in relation to $x$ and $y$.

The important conclusion from the findings shown in Appendix II, is that if $x_j^* + N_j > 0$ then $P_j(x_j^*, y_j^*) = \alpha_j$ and $y_j^* = w^*$. Obviously if $x_j^* + N_j = 0$ then $P_j(x_j^*, y_j^*) \leq \alpha_j$ and $y_j^* = w^*$.

From a practical point of view, this optimization problem can be reduced to a one-dimensional search. If we sweep the value of $w^*$ on a certain range and if we have means of finding $x_j^*$ by solving $P_j(x_j^*, w^*) = \alpha_j$, then we only need to verify if the condition $\sum_{j=1}^{M} x_j^* = \Delta N$ holds. The well-known bisection method is an effective way of implementing this search.

The following procedure was conceived to find the optimal bit allocation:

1. Define $[w_{MIN}, w_{MAX}]$ as the search region for $w$.
2. Let $w = \frac{w_{MIN} + w_{MAX}}{2}$
3. For all $j \in \mathbb{N}$ find $x_j^*$ satisfying $P_j(x_j^*, w) = \alpha_j$
4. Let $S = \sum_{j=1}^{M} x_j^*$
5. If $S < \Delta N$ then do $w_{MAX} = w$.
6. If $S > \Delta N$ then do $w_{MIN} = w$.
7. If $w_{MAX} - w_{MIN} > \text{tolerance}$ then go to 2.
8. Optimal solution $w^* = w$ is found.

At this point it seems appropriate to discuss some facts concerning the feasible region $\Omega$ and the optimal solution. Figure 5 illustrates an example involving 4 transport channels.
There we can see the xy-plane and all the isocurves associated with the equation \( P(x, y) = \alpha_k \). Notice that these curves are compatible with the fact that function \( P(x, y) \) is monotonic decreasing in both arguments.

One can conclude from the results obtained that each optimal solution \((x_k^*, y_k^*)\) must always be in the corresponding isocurve. Those solutions in which \( x_k^* + N_k > 0 \) must have \( y_k^* = w^* \) but those in which \( x_k^* + N_k = 0 \), the variable \( y_k \) can have values to the right of the isocurve and to the left of \( w^* \). The value on the isocurve corresponds to the minimum energy case and therefore should be the one chosen.

In this illustration, the optimal value \( w^* \) indicates nonzero \( \Delta N \) values for transport channels 2 to 4. Channel 1 does not require extra bits because its isocurve does not intercept the vertical line indicated by \( w^* \).

If no extra bit is given to any of these transport channels, the minimum \( E_b/N_0 \) values that meets the desired BER are respectively, 2.147 dB, 1.983 dB, 1.733 dB and 2.612 dB, this last value corresponding to the most severe case. Note that 84 extra bits can be distributed over these channels.

A MATLAB script that implements the allocation bit procedure here discussed was written. When this algorithm is run, the result tell us that 13 extra bits should be given to channel 1 and 74 extra bits should be given to channel 4, thus reducing the maximum \( E_b/N_0 \) to the value of 1.982 dB. Consequently, the energy saving for the worst channel is 0.630 dB but the global savings (sum of savings for all channels) is 0.795 dB. The processing time for this example in a Pentium IV-2.6MHz computer was less than a second.

![Fig. 5. Feasible region and optimal solution](image)

It is now opportune to discuss how the integrality condition can be reinstated. We are well aware that in mixed-integer programming the optimal solution very often is not the truncated or rounded version of the optimal real solution. Nevertheless, due to the smoothness of function \( P(x, y) \), all simulation results have revealed that the variation of this function between two consecutive integers were not superior to 2%. Although we cannot strictly speak of optimality if truncations is the process to produce integer solutions, we are fairly confident that results produced this way will not deviate more than a few per cent of the optimal solution. Therefore, from a practical point of view, this type of solution seems to attend our purposes.

### VI. EXAMPLES

In order to illustrate the proposed methodology, two examples are presented. In the first one, we have four transport channels that share a frame 600 bits long with characteristics described in Table I.

This is the real case already considered at the beginning of Section III, where we have voice transmission at 12.2 kbit/s by an AMR codec. There are three transport channels (1,2 and 3) carrying bit sequences with different QoS requirements and an associated signaling channel (4). A spreading factor of 64 is used, thus resulting in a physical channel of 600 kbit/s. The BER values are those stated in [9].

The function \( P(x, y) \) is obtained by means of the methodology described in Appendix I applied on data generated by computer simulation.

In the previous example, only repetition bits are needed to optimize the Rate Matching operation. This second example shows the more general case of using repetition as well as puncturing. For this purpose, refer to the PS mode case also presented in Section III. Considering the second option, where we have 639 bits to be fitted in a frame of 600 bits, Table II shows the results of the algorithm application. It is a rather interesting situation, because, in the contrary of the natural expectation of suppressing (puncture) 39 bits, the algorithm shows the need of simultaneous puncturing and repetition of bits in the two channels.

Table: Parameters of the Transport Channels in PS mode

| Chnl | Coder  | Bits  | BER     | \( E_b/N_0 \) | \( R/P \) | \( E_b/N_0 \) |
|------|--------|-------|---------|--------------|--------|--------------|
| 1    | CC1    | 129   | \( 10^{-4} \) | 1.762       | 45     | 2.529        |
| 2    | CC1    | 129   | \( 10^{-4} \) | 2.756       | 46     | 1.982        |
| Total|        | 639   |         | 2.529        |        | 2.529        |

In conclusion, this paper discussed the performance of voice calls in UMTS networks considering some important aspects of the transmission system in CS and PS mode. The structure of the transport channel have been described for both modes and a performance analysis focusing on the rate matching operation have been carried out. In addition, a method has been proposed for allocating/puncturing bits in transport channels of an UMTS communication system in order to meet QoS levels and minimize the maximum \( E_b/N_0 \) required in each one of.
these channels. A mathematical analysis of this problem has revealed that this minimization can be produced by a simple algorithm that is easily programmable and computationally cheap. To the best of the author’s knowledge, the approach here investigated to optimally allocate repetition bits has not yet been addressed in the technical literature.

where $E_b/N_0$ is expressed in dB. Each one of the $a’$s coefficients can be estimated from data in the LS sense and they can be further modelled by an expression of the form:

$$a_i(N,\Delta N) = \sum_{j=0}^{N_2} b_{ij}(N,\Delta N)^j$$  \hspace{1cm} (6)

Again each one of the coefficients $b’s$ was estimated from data in the LS sense and they were further modelled by an expression of the form:

$$b_{ij}(N) = \sum_{k=0}^{N_1} c_{ijk} \cdot N^k$$  \hspace{1cm} (7)

Therefore the probability in questions was modelled by:

$$P(N, \Delta N, E_b/N_0) = 10^{f[N, \Delta N, E_b/N_0]}$$  \hspace{1cm} (8)

where:

$$f[N, \Delta N, E_b/N_0] = \sum_{i=0}^{N_1} \sum_{j=0}^{N_2} \sum_{k=0}^{N_3} c_{ijk} \cdot (E_b/N_0)^i \cdot (\Delta N)^j \cdot N^k$$  \hspace{1cm} (9)

Figure 6 shows the results of this model when applied to the rate 1/2 convolutional coding used in the UMTS system. Data were obtained by computer’s simulation. The approximation has shown to be fairly acceptable for the following parameter values: $N_1 = 2$, $N_2 = 3$ and $N_3 = 3$. Although not shown here, similar good results were also obtained for the 1/3 convolutional coding and 1/3 turbo coding, thus indicating that this fitting function seems to be a nice model for fitting data.

**IX. APPENDIX II**

**Proof of the Equivalent Optimization Condition**

Since physical evidence allows us to believe that function $P$ is monotonic in both arguments, the derivatives mentioned above are strictly negative in the region of interest. Hence $\lambda_k > 0 \implies \eta_k > 0$.

Let $\Omega_\lambda$ be the subset of $\{1, 2, ..., M\}$ in which $i \in \Omega_\lambda \implies \lambda_i > 0$ (and consequently $\eta_i > 0$). Note that this set is non-empty because the sum of $\lambda’s$ is one. Therefore if $j \in \Omega_\lambda$, then $y^j = w^*$ and $P_j(x^*_j, y^*_j) = \alpha_j$. Consequently $\nu + \mu_j < 0$ thus implying that $\nu < 0$.

Now let us analyze those $j’s$ that do not belong to $\Omega_\lambda$. In this case, $\eta_j = 0$, implying that $\nu + \mu_j = 0$. Since $\nu < 0$, we conclude that $\mu_j > 0$, which leads to $x^*_j + N_j = 0$.

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