PERFORMANCE OF PUNCTURED CONVOLUTIONAL CODES OVER THE MOBILE RADIO CHANNEL AND ITS IMPACT ON THE DESIGN OF A CALL SET-UP PROTOCOL

Performances de codes convolutifs perforés sur le canal radio mobile et leur impact sur la conception d'un protocole de mise en communication

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RÉSUMÉ

Cette contribution fait suite à un travail effectué à l'ETSI (European Telecommunications Standards Institute) pour l'établissement de la norme du système TETRA (Trans European Trunked RAdio).

Les performances, obtenues par simulation, d'une chaîne de communication numérique modélisant les caractéristiques de base du système TETRA sur le canal radiomobile sont présentées. Différents schémas de codage sont étudiés. Ils servent à évaluer les performances de divers protocoles de mise en communication. Chaque protocole implique le séquencement des messages ainsi que leur format et leur codage. Le choix final est justifié par le trafic écoulé et par le temps de mise en communication en fonction du trafic moyen et pour une perte de trafic donnée.

ABSTRACT

This paper is subsequent to studies that were run by ETSI (European Telecommunications Standards Institute) in order to establish the standard of the TETRA system (Trans European Trunked RAdio).

Simulated performances of a digital communication system summarising the basic characteristics of TETRA on the mobile radio channel are presented. Different coding schemes are studied. They are used to evaluate the performances of various call set-up protocols. Each protocol implies the message sequences along with their format and coding scheme. Its choice is determined by the throughput and by the call set-up time against average incoming traffic and for a given loss of traffic.

1. Introduction

For land mobile radio systems, apart from spectrum efficiency, *throughput efficiency*, and *quality of service* are usually key parameters to optimize.

Throughput efficiency is the maximum number of communications per time unit that can be set-up, maintained and released, with a given limited signalling capacity and under given traffic conditions.

Quality of service is specified as a minimum guaranteed probability that a "quality requirement" is met (this "quality requirement" may be e.g. "the net bit error rate should be less than 1%" so to guarantee an acceptable speech quality).

The following sections focus on the optimisation of throughput efficiency and quality of service.

2. Brief overview of the TETRA system

TETRA is a wide ranging standard which comprises three distinct areas dealing with mixed voice plus data, packet data optimised and direct mobile to mobile operation. This paper considers only the mixed voice plus data operation.

For the first generation of the TETRA systems, the general features are: Time-Division Multiple Access (TDMA) multiplexing up to 4 TDMA channels on the radio path, Slotted Aloha access technique, 400 MHz carrier frequency-band (a second generation will exist in the 900 MHz band), $\pi/4$-DQPSK modulation with a gross rate of 36 kbit/s and a 25 kHz channel spacing.

For most of TETRA customers, the *call set-up time*, i.e. the time spent between a first access attempt by a user and the actual allocation of a resource to this user, is one of the major quality of service requirements.

3. TETRA TDMA frame structure

Basically, one *timeslot* is a time interval of 14.17 ms (510 bit duration), where bursts are transmitted. One timeslot can carry up to 432 encoded bits. As one carrier may be time-shared by up to 4 communication links, one *TDMA frame* comprises 4 timeslots.

Assuming that a fixed capacity, e.g. one timeslot per frame, is assigned to signalling at the call set-up, the mean number of call set-up requests on one frame defines the *average traffic*. One timeslot can a priori be divided into an integer number N of *subslots*: in this way, one signalling message could be transmitted in one *subslot*, and there would be N message transmission opportunities per timeslot. In order to maximize the throughput and minimize the call set-up time, which is "the optimum" value for N?

On the one hand, the transmission of only one message per timeslot (N=1) would allow a low-rate error control scheme with a good robustness against fading, interferences and noise. In that case, the number of required message re-transmissions would be low, in average: the capacity required for a call set-up procedure to succeed would be accordingly low, for the benefit of throughput efficiency and call set-up time performance.

On the other hand, for high traffic densities, choosing "N=1" would, by design, restrict the maximum throughput (since there cannot be more than one successful transmission of random access message per frame). From this point of view, throughput and call set-up time would be degraded.

In the following, the possibilities of N=1 and N=2 are investigated. The case "$N > 2$" is not envisaged here mainly because this would constraint the random access message (i.e. the first message of the call set-up procedure) to be very small, and imply that an extra-message is transmitted, hence some extra-call set-up delay, even for a lightly loaded system.

4. Error control simulation

The generic error control scheme, for signalling channels, is displayed on Figure 1. The k information bits of one signalling message are first encoded by a (n, k) block code, with $n = k + 16$, which is primarily aimed at error detection. This is a standard code [1] derived from the cyclic block code with generator polynomial $g(x) = x^{16} + x^{12} + x^5 + 1$.

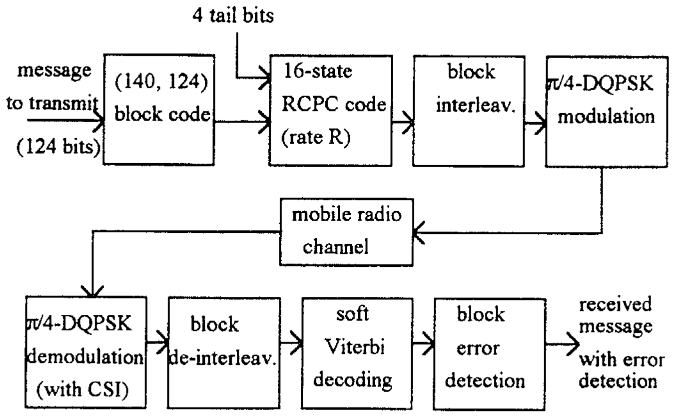

Figure 1 : Error control scheme for signalling channel

In the TETRA system, one signalling message carries 124 information bits, i.e. (n, k) = (140, 124). Block-coding is followed by *Rate-Compatible Punctured Convolutional* (*RCPC*) coding [2]. The number of bits before RCPC-coding is 144 (including 4 tail bits). Therefore, the choice of N=1 or N=2 subslots per timeslot leads to the use of a R=1/3- or a R=2/3- rate code, respectively. In the following simulations, we shall consider the 8/24- and the 8/12- rate RCPC code as defined in [2, p. 392]. The RCPC-coded bits that encode one message are then interleaved together using block interleaving.

The simulations that follow assume a 1-path Rayleigh fading channel, i.e. the time-varying received signal, z(t), can be written in complex notations as a function of the transmitted one, u(t), as z(t) = r(t) u(t) + w(t), where both r(t) and w(t) are complex, Gaussian processes. r(t) is affected by a maximum Doppler frequency of 18.5 Hz, which corresponds to 50 km/h for 400 MHz carrier frequency. This is simulated using Jakes modelling [3].

For the purpose of these simulations, demodulation is performed with Channel State Information (CSI): an estimate of r(t), denoted by $\hat{r}(t)$, is calculated by a recursive least mean square algorithm, and the ratio z(t)/$\hat{r}(t)$ is provided for de-interleaving. As indicated on Figure 1, de-interleaving is followed by soft-decision Viterbi decoding [4] of the RCPC code and block error detection.

Figure 2 shows a series of error rates as a function of E_S / N_0 , where E_S denotes the energy of one transmitted symbol, i.e. of two transmitted bits.

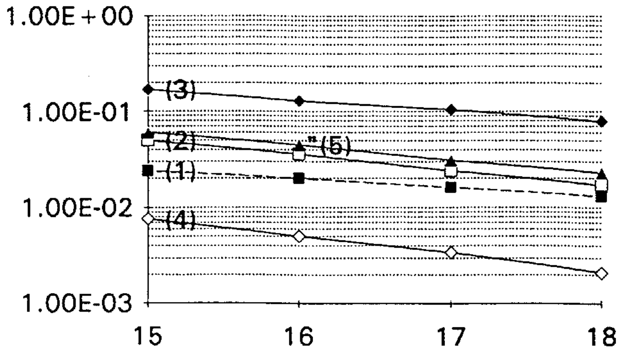

Figure 2 : Performance of RCPC codes

Curve (1) (in dashed lines) shows the Bit Error Rate (BER) as obtained after demodulation. Curves (2) and (3) respectively show the BER and the Message Error Rate (MER) after Viterbi decoding for $N = 2$ subslots per timeslot (i.e. a 2/3-rate RCPC code was used with block interleaving of 216 bits). Curves (4) and (5) respectively show the BER and the MER after Viterbi decoding for $N = 1$ subslot per timeslot (i.e. a 1/3-rate RCPC code was used with block interleaving of 432 bits).

In TETRA, the reference receive quality corresponds to a BER after demodulation in between 1% and 2%, hence to $E_s/N_0 \approx 17$dB. At this reference, the considered channel coding schemes do not provide significant BER improvement; actually, for BER performance at $E_s/N_0 = 17$dB, the coding gain is negative on curve (2). This is because the low Doppler frequency implies a very strong auto-correlation of the fading in the short term.

Indeed, for signalling purpose, the main motivation for channel coding is the ability of the decoding scheme to regroup errors into bursts in order to keep the MER low, quasi-independently of the Doppler frequency. On the opposite, other simulations have shown that for higher Doppler frequencies, an uncoded scheme would provide prohibitively high values of the MER from the protocol perspective.

5. Call set-up protocol

In the following, an intra-cell call set-up is considered: a mobile M_1 is calling a mobile M_2 via a base station B, both mobiles being located in the cell covered by base station B. A description of the message sequence is displayed on Figure 3. Figure 4 describes the call set-up procedure by means of a finite-state diagram.

Assume that mobile M_1 generates an access request (*u-setup* message) during TDMA frame number n. This message will be transmitted in the timeslot assigned to signalling of the TDMA frame $(n+1)$.

This message may collide with another *u-setup* message (issued by another mobile). If so, mobile M_1 will not receive its response from the base station B. It may re-transmit the *u-setup* message in the control slot of TDMA frame number x, where x is a random integer number chosen by M_1, uniformly distributed between $(n+1+T_0)$ and $(n+1+T_1)$. The time interval where the *u-setup* message can be re-transmitted is called "Aloha frame". T_0 and T_1 are integer numbers called "Aloha frame parameters", with $T_0 \le T_1$. Note that no more than N_u *u-setup* transmissions are allowed; beyond this number, the call is lost.

Whenever the *u-setup* transmission is successful, base station B sends to both mobiles M_1 and M_2 a unique message. For the former, this message acts as an acknowledgement and is called *d-call proceeding*; for the latter, it acts as a paging message and is called *d-setup*. The transmission of the *d-call proceeding* message may fail. If so, mobile M_1 would have to re-transmit the *u-setup* message again.

Whenever the *d-setup* transmission is successful, mobile M_2 may transmit the *u-connect* message to base station B. This message just acknowledges the *d-setup* message. Note that no more than N_d *d-setup* transmissions are allowed; beyond this number, the call is lost. The transmission of the *d-setup* message may fail. If so, base station B would have to re-transmit the *d-setup* message to mobile M_2.

After reception of the *u-connect* message, base station B may send to both mobiles M_1 and M_2 a unique message telling which traffic channel they should use to continue their call: this message is called *d-connect* for mobile M_1 and *d-connect ack* for mobile M_2.

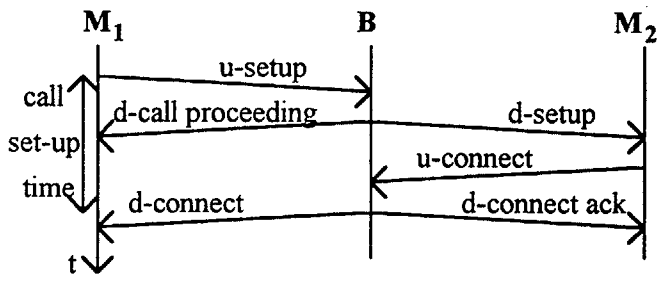

Figure 3 : Time sequence of call set-up messages

The call set-up time is defined as the time between the initial access attempt (*u-setup* message) by mobile M_1

and the first transmission of the *d-connect* and the *d-connect ack* messages by base station B.

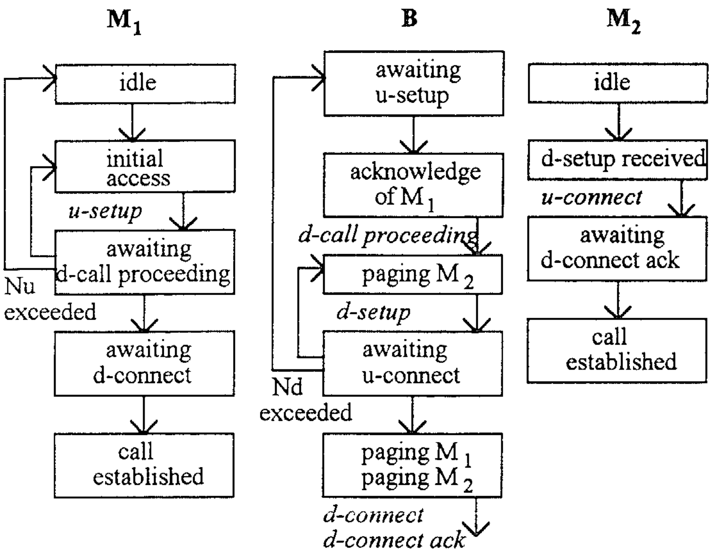

Figure 4 : States diagrams for call set-up procedure

6. Protocol simulation with channel coding

Under ideal conditions, the TETRA air interface is designed so to guarantee a call set-up time of less than 300 ms. By definition, when taking into account the traffic and radio constraints, 90% of the successful call set-up do succeed after a time of at most the "90% call set-up time".

As for the general assumptions and definitions:
- the cell is assumed to be a discus, where mobiles are uniformly distributed;
- path loss is calculated as the sum of the average path loss following [5] and of a slow shadowing term assumed as log-normal with 5 dB standard deviation;
- the average $E_s / N_0 = 22$ dB at the edge of cell;
- both parameters N_u and N_d are set to 5 attempts;
- Aloha parameters T_0=6 and T_1=12;
- 1 timeslot per TDMA frame is assigned to signalling for both "*N*=1" and "*N*=2" scenarios;
- instantaneous traffic follows a Poisson distribution;
- throughput is the average call set-up success per TDMA frame;
- traffic loss is that part of the average traffic for which the call is lost = 1 - throughput / average traffic.

For *N*=1 and for *N*=2, Figure 5 shows the throughput (curves *N*.1), the traffic loss (curves *N*.2) and the 90% call set-up time in terms of number of TDMA frames (curves *N*.3), versus the average traffic. Values on the Y-axis should be read in terms of frame count for *N*.3 curves and in terms of traffic percentage for other curves.

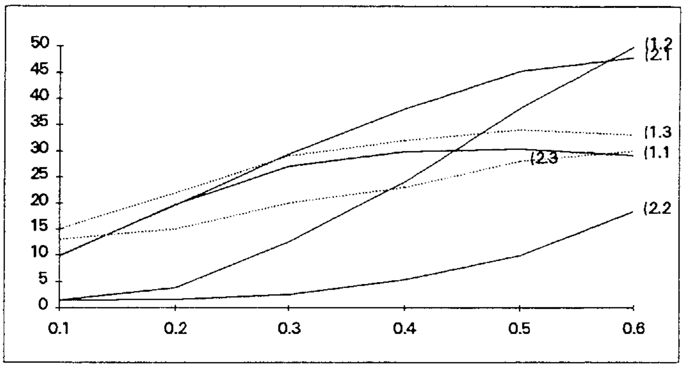

Figure 5 : Initial access simulation results

For an average traffic exceeding 20%, protocol limitations, by opposition to radio limitations, become predominant. All key quantities are in the favour of solution "*N*=2". Throughput reaches a maximum when the average traffic is around 50% (*N*=1), and 60% (*N*=2).

For a maximum acceptable traffic loss of 10%, solution "*N*=2" allows to operate with an average traffic of 50% versus only 28% with solution "*N*=1". For this operating point, the 90% call set-up time is about 28 TDMA frames duration (1.6 s) in both cases.

7. Conclusions

With regard to the application of the TETRA protocol:
- channel coding is essential in order to keep message error rate low enough with any Doppler frequency;
- but performance of low rate coding scheme is weakened by tight interleaving constraints and by the effect of collisions on the random access. This explains the final choice of *N*=2 for the TETRA system.

References

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