PERFORMANCE OF THE GPRS RLC/MAC PROTOCOLS WITH VOIP TRAFFIC

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Abstract - GSM/GPRS networks have been designed to carry both voice and data traffic. Our aims are to study the behaviour of the GPRS mechanisms (RLC/MAC layer and radio resource scheduler) when the application layer carries packetized voice, and determine if the quality of service provided by the network is suitable for the VoIP service. The paper includes a discussion about the voice source model, an analysis of the effects in capacity of a GPRS carrier when feed with VoIP packets, measurements of statistical parameters of the voice traffic stream and a discussion of the influence in the quality of the service.

Keywords - GPRS, VoIP, QoS.

I. INTRODUCTION

Compared to traditional circuit-switched networks, the use of a packet-switched bearer for voice transmission would increase substantially the utilisation of the scarce radio resources, thanks to the effect of packet multiplexing. This is why next generation wireless systems will be based on packet-switching techniques, extending uniform end-to-end IP transport over both the radio interface and wired Internet. The main challenge for network engineers is then to achieve a good trade-off between efficient spectrum usage and good quality for real-time services such as voice and video delivering. A first approach to analysing the GPRS capacity (for general traffic) can be found in [2,11].

We study the capacity, in number of simultaneous VoIP calls, of the GPRS system. We also examine the effect of multislot transmission and different scheduling strategies at RLC/MAC layer. Finally, we comment the results obtained for different measures of interest such as end-to-end GPRS radio interface transmission delay and its variation (jitter) and PDCH utilisation or throughput. We also discuss the trade-off between the number of simultaneous users and the Quality of Service requirements recommended by international regulations.

II. DESCRIPTION OF THE SCENARIO AND THE GPRS SIMULATION MODEL

Voice transmission, as any real time service, has a very strict requirements of quality of service: a low transfer delay (about 200 ms for a acceptable quality, reaching 400 ms in some circumstances) and a loss ratio less than 5 % of packets, according to TIPHON group from ETSI [3]. We will use these criteria to determine the real capacity of voice

calls in a GPRS scenario with a BER (Bit Error Rate) on the radio interface as bad as 10^{-2} (good radio channel) and a big delay due to the scarce radio resources. Another important feature is the amount of header information in the global RTP/UDP/IP header. Using the G.729B codec the header length supposes the 67 % of the packet length. In this paper we assume the utilisation of a header compression mechanisms such as CRTP and ROCCO [4].

We use an ON/OFF model based on real voice traces coded with the ITU-T G.729 standard [8], at a rate of 8 Kbps and with a subjective quality (MOS) comparable to the 32 Kbps ADPCM. To characterise the ON and OFF periods in terms of mean, variance and distribution, we use the Kolmogorov – Smirnov (KS) and χ^2 goodness-of-fit test (for more details, see [1]). Using the plots of the probability function and the first and second moment, we can observe that the variability of the ON period can be fitted by a exponential distribution, but the OFF period presents a tail that is heavier than the exponential case. These results are validated in [9]. Table 1 summarises the parameters of our voice model.

Table 1
Parameters of the model proposed after analysis of real VoIP traces.

Parameter	Mean	Typical Deviation	PDF
ON period	0.4011 sec	0.3637 sec	Gamma
OFF period	0.5775 sec	1.1774 sec	Weibull

Our GPRS simulator models the radio interface and implements the basic functions of the RLC/MAC and LLC layers using the ETSI recommendations [5,6,7]. Our assumptions are:

- For each ON period, the mobile station (MS) generates a random access burst (RACH channel) to get resources from the BS resource controller. The access time (transmission of the random access burst plus the response and the process time) is modelled as a uniform distribution with a minimum value of 20 ms and a maximum value of 40 ms. It is assumed that this is the time needed for the resolution and assignment of resources for a MS.
- The scheduler algorithm has some parameters such as the number of time slots assigned to a call (2 or 4) and the slots of the TDMA frame assigned to GPRS (we assume in all simulations 7 PDCH traffic channels and 1 PRACH control channel). The implementation of the

scheduler balances the load in all PDCH and minimizes interference between different MS.

- The propagation model is the TU-3 from ETSI (Typical Urban Scenario – 3 Km / hour).
- Two of the four GPRS coding schemes have been used: CS-2 and CS-3, since they seem to be the typical elections of the operators. CS-2 has an important effect over the voice codec, since G.729B generates a frame of 20 bytes every 20 ms, to which we add the encapsulation and compressed RTP/UDP/IP header and the GPRS LLC/RLC header. Then, two RLC blocks are needed for a transmitting a single IP frame of 20 bytes. This technique reduces the redundancy due to multiple headers, but has the disadvantage of bringing an extra 20 ms delay.

III. SIMULATION RESULTS

Our simulations replicate a typical GPRS scenario offered commonly by mobile telephony operators: 2 - 4 time slots and coding scheme 2 - 3. The study could be extended to more combinations of channel coding scheme and timeslots, but due to space limitation we will comment only the aforementioned scenarios, focusing on the capacity of the GPRS system in terms of number of MS or voice calls, delay and loss.

A. Delay

The delay introduced by the GPRS air interface is mainly caused by three factors: the delay access on each ON period (*variable*), the delay of the multiplex process (*variable*) and the transmission delay (*fixed* ≈ 20 ms). Figures 1 and 2 show the mean and maximum delay of the GPRS air interface for CS-2 and CS-3 using two voice models: Model 1, exponential distributions in both periods and Model 2, the model presented in section II. We observe an inflexion point when the number of calls reaches 9 – from that point on, the delay increases substantially. The increasing delay is correlated with the multiplexing gain effect: when the offered traffic load is low the 7 traffic slots can be assigned to single calls, but when traffic is over 7 simultaneous calls, users have to share slots. Under low load conditions, the delay is due only to the access and reservation mechanisms modelled with a uniform distribution.

The choice of the voice model has a strong influence on the delay analysis. Traditional voice models (with exponential distributions for ON and OFF periods) lead to optimistic results in terms of delay, as we can see clearly in Figure 2 with the maximum delay, since they underestimate the burstiness of the traffic. This may cause underestimation of the resources needed for a certain QoS level, and can lead to buffer overflow. There are not noticeable differences using different channel coding schemes and CS-3 performs better because only one RLC block is needed for each IP packet, as commented before.

Fig. 1. Mean GPRS air interface delay.

Fig. 2. Maximum GPRS air interface delay.

Fig. 3. Delay excess bound probability.

We can determine a delay bound from this data. Our criteria are a mean delay of 150 ms or lower, and a maximum delay not greater in a % of a upper bound. Figure 3 shows the

probability that an IP packet suffers a greater delay than a certain upper bound using Model 2 and CS-2. This data can be used for dimensioning purposes. As we can see, for 11 MS (or calls) we obtain that the 24 % of the IP packets present a delay greater than 100 milliseconds and the 6 % greater than 400 milliseconds. From the data presented on the figures and the table, we can set a limit between 9 or 11 simultaneous calls in a 7 PDCH channel using the G.729B voice codec, depending on the delay target assumed by the mobile company.

The influence of the delay distribution is a critical aspect when dimensioning the system. The bound of 9 calls is an inflexion point on the statistic of the stream due to multiplexing effects. This shapes the tail of the distribution, making it more heavy-tailed. To prove this result, we calculate the α parameter (characterize the hyperbolically decaying of the distribution) using two typical methods: A minimal square regression (MSR) over the logarithmic probability function and the Hill estimator.

Table 2

Estimated Values of the hyperbolically decaying parameter in the delay distribution.

Number of MS	MSR	Hill
7	5.36	5.39
9	2.60	2.78
11	1.39	1.67
12	1.53	1.93
13	1.72	2.88

The results presented in Table 2 show a clear increase of the distribution delay variability (in the range of $1 \leq \alpha \leq 2$ a distribution presents theoretically infinite variance and is called heavy-tailed). This tendency is truncated when the traffic offered to the system (13 mobile stations) is near the maximum throughput of the system. The presence of heavy tailed distributions in the delay series can have severe repercussions in the design of smoothing algorithms at the destination point due to the presence of frequent bursts with high delay values. With these results, we can conclude that the multiplexing scheduler has an effect not only on the mean and max delay, but also on the distribution of this delay. As we mentioned before, when the system is unloaded, the delay distribution presents a uniform behaviour due to the access and resource reservation mechanisms and when the number of simultaneous mobile station increases, the distribution changes to a more variable one.

The use of 2 or 4 time slots (for the same load) also has effects on the delay. Using 4 timeslots reduces the blocking delay (wait until get a free resource – a non-used slot), and the effect is a reduction of the delay, and more specifically, the maximum delay, as shown in Figure 4.

Fig. 4. Delay comparison 2 TS – 4 TS.

B. Jitter

Jitter (variation of delay) has an important influence in the statistical properties of the real-time traffic. The jitter distribution found in our simulations shows the same trends as found in delay distribution. When multiplexing interference is low (less calls than available timeslots) the jitter presents a deterministic distribution but, when the number of call increases we observe a change towards a heavier-tailed distribution. Using 4 timeslots the multiplexing interference is higher even with less calls, since each call uses more slots. In this situation the jitter can decrease, because the chance of transmitting a RLC block is high (no queuing delay blocking) but the variability of the delay is higher (Figure 5). We observe the same inflexion point detected in the delay analysis.

Fig. 5. Delay jitter comparison 2 TS – 4 TS.

C. IP Packet Loss

We apply the criteria of a maximum loss of 5% of packets. In a VoIP call, packets can be lost by two main causes: an

erroneous RLC block which affects the whole IP packet, or an IP packet with a delay greater than a certain bound (and therefore, unusable). We proceed with the analysis of both possibilities.

- Error probability: Using the TU-3 model for characterisation of the radio channel CI and the BLER (Block Error Ratio) we obtain some results for IP packet error probability for CS-2 and CS-3. Figure 6 shows the evolution of packet error versus carrier-to-interference ratio (C/I). It should be noted that when using CS-2 each IP packet is transported in two RLC blocks, while in CS-3 only one RLC block is needed for each packet. This is the reason why the error probability is greater using CS-2 than CS-3. In terms of BER (Bit Error Ratio) we have to consider that an IP packet using CS-2 contains 40 bytes and using CS-3 20 bytes. In this situation, the effective error probability (quantity of information bits lost) is less using CS-2 but is necessary a rigorous study to determine the repercussions on the subjective quality for voice perception. From the data presented in the figures we can conclude that 19 - 20 dB is the minimum C/I for guaranteeing a low probability of error in IP packets.
- Delay: When the delay is greater than a certain upper bound, the packet may be considered as lost, since its payload has already out of date. These results are shown in Figure 3 and determine the maximum number of simultaneous VoIP calls as we can observe as the acceptable delay is very dependent on the number of simultaneous calls. The bound is found at 9 - 11 mobile stations to assure that less of the 5 % packets are greater than 400 ms.

Fig. 6. IP packet error probability versus C/I.

D. PDCH utilisation

Finally, we note that with 9 to 11 calls the system works at 50 - 75 % of the real GPRS throughput. This waste of

resources is necessary for maintaining a certain level of QoS, in delay terms.

E. Conclusions and discussions about delay and loss results

Considering the delay, we can establish an upper bound of 9 to 11 calls in a GPRS carrier with 7 PDCH. Compared with GSM, which could only offer 7 simultaneous calls per carrier, the use of VoIP over GPRS brings an increment of 28 % to 57 %. However, this increase of capacity is only achieved with good conditions in the radio channel ($C/I > 20$ dB), because of the fatal influence of a high error rate.

The fact that heavy-tailed distributions have appeared in several parts of the analysis is of the greatest interest for the correct dimensioning of the shaping buffers used for reconstruction of the real time voice stream, which cancel the delay distortion introduced by the network. The design of those buffers faces a trade-off between delay and loss probability.

IV. CONCLUSIONS

This work explores the possibility of offering new voice services over GSM networks, exploiting the new features of packet switching from GPRS. In general, packet voice on GPRS will not be a replacement for GSM, but opens a way to implement VoIP in future 3rd generation wireless networks. The results show that GPRS can increase the capacity of a traditional circuit switched system to 125 - 150%, but at the price of some controlled degradation of the quality of voice caused by increased packet losses and higher delays. Regarding delay, we have found an upper bound of 10 calls in a GPRS with 7 PDCH traffic channels.

With regard to the voice model, the G.729 Annex B codec seems better suited for simulation studies than the traditional ON/OFF model based on pure exponential distributions. Simulations fed with his model bring most precise results than the exponential, especially in the behaviour of the delay distributions and the effects of long-range dependence in self-similarity [10].

This report is a first approach to the problem, which requires additional efforts. More work is needed, with the aim of characterising the trade-off between quality of service and capacity, since it seems clear that this could lead us to defining different service levels which could be mapped to several commercial offers. These new VoIP services could coexist using capacity on demand techniques with circuit-switched ones, offering more competitive prices at lower performance (in terms of speech quality).

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