Studying the Effects of FEC on Voice Traffic Using PSQA

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In recent years, the growth of the Internet has spawned a whole new generation of networked real–time multimedia applications, such as VoIP, video-conferencing, video on demand, music streaming, etc. which have very specific, and stringent, requirements in terms of network QoS. VoIP in particular has become very widespread lately, as attested by several commercial offerings, and software packages that implement it. It is well known that the Internet was not designed with real–time applications in mind, so the quality of these services tends to be very variable in a best–effort network context. Several techniques have been developed in order to improve the perceived quality. In this contribution, we extend the results published in [1] and study the performance of one of those techniques, namely media-dependent Forward Error Correction [2]. This error correction mechanism consists of piggybacking a compressed copy of the contents of packet n in packet $n+i$ (i being variable), so as to mitigate the effect of network losses on the quality of the conversation[1]. To evaluate the impact of this technique on the perceived quality, we use a $./M/1/H$ 3–class queue to model the network, and study different scenarios to see how the increase in load produced by FEC affects the network state. We then use a *pseudo–subjective* quality evaluation tool that we have recently developed in order to assess the effects of FEC and the affected network conditions on the quality as perceived by the end–user.

Assessing the quality of a media stream as perceived by the end–user is not an easy task, as quality is a very subjective concept. There are basically two ways to perform this assessment, namely *subjective tests*, *à la* ITU-P.800, and *objective tests* such as PAMS, PESQ or MNB2 (cf [3] for references on this subject). We are interested in being able to perform these assessments in real–time, as we are currently working on real–time quality control mechanisms for VoIP. However, subjective tests (which provide the best assessments, normally in the form of Mean Opinion Scores, or MOS) are not usable in real–time, nor are most of the objective ones, which require access to the original sample, in order to perform a comparison with the received one. Only the ITU E-Model is able to predict the perceived quality without accessing the original sample, but unfortunately, its results do not correlate well with those of human observers. This is why we use *Pseudo–Subjective Quality Assessment*, a hybrid approach [4], [5] we have recently developed, which does not require access to the original sample, and provides results that correlate well with human perception (i.e. subjective tests). This approach also outperforms the other objective techniques for VoIP [3], so it is worth using even if real–time assessment isn't strictly necessary (as is the case in this contribution).

In order to use PSQA, we need to consider a set of parameters which *a priori* affect the perceived quality, such as the network loss rate, the codec used, the packetization interval, etc. Let this set be $\mathcal{P} = \{\pi_1, \ldots, \pi_P\}$. We then choose several representative values for each parameter, and use those values to obtain several *configurations*, i.e. tuples of the form $\gamma = \{v_1, \ldots, v_P\}$, where v_i is one of the chosen values for π_i. The number of possible configurations is usually very large, so we choose a subset of those, and use them to generate several distorted voice samples, by using a test-bed or a simulator. These distorted samples are later assessed in a subjective test, so that we have an MOS value associated to each configuration considered. This mapping is used to train a statistical learning tool (after trying several tools, we found that Random Neural Networks, or RNN [6] work best in this context). Being that RNN generalize very well, it is possible, if the configurations used were properly chosen, to provide very good quality assessments for all of the configuration space.

As mentioned before, we use a 3–class[2] $./M/1/H$ queue to model our network, and we consider three classes of traffic, namely background traffic (file transfers, e–mail, WWW, etc.), audio traffic without FEC, and audio traffic with FEC. Voice traffic is not a very significant portion of the overall Internet traffic, but this is likely to change as VoIP gains momentum. This is why we consider scenarios with VoIP traffic fractions ranging from 5% to 50% of the total traffic. Simple Markovian analysis of our model allows us to derive loss rates and mean loss burst sizes (MLBS) for each class of traffic, which we'll use to perform the quality assessment. Since we are working

[1] It should be noted that as VoIP traffic has very stringent delay constraints, it is generally not possible to re-send a lost packet, as the replacement packet would arrive too late to be played out.

[2] We define one class for background traffic, another one for VoIP traffic which does not use FEC, and a third one for VoIP traffic using FEC.

with traffic aggregates, we assume that packets of all classes have exponential inter–arrival times. This would not be the case if we worked on a per–flow basis, in which case an ON/OFF model with CBR traffic during ON periods might be more appropriate to model VoIP streams. It should be noted that any network model can be used as long as these parameters (and any other parameters that may be considered, such as delay and jitter, for instance) can be derived, either analytically, or through simulation.

The results obtained with this enhanced model confirm the results obtained in [1], namely, that the use of FEC is advisable in all cases, since there is a marked difference in perceived quality between streams with no redundancy added, and FEC protected streams. It is interesting to see that when fewer streams use FEC, the results are better for those streams. However, as the number of FEC users increase, quality slightly degrades for everybody. This is a consequence of an increased MLBS, which in this model is given for each class i by $MLBS = 1 + \lambda_i/\mu$, where λ_i is the arrival rate of class i packets, and μ is the service rate of the network's bottleneck. The increase in MLBS implies a decrease in FEC efficiency, unless the FEC offset is adapted.

Figure 1, shows this phenomenon for a case where voice traffic accounts for 50% of the total network traffic. It can be seen that as the number of flows using FEC increases, the perceived quality of the FEC–protected flows decreases. As most flows will be likely using FEC, some mechanism such as the dynamic adjustment of the FEC offset in response to varying MLBS values should be implemented in order to achieve the best possible quality. However, increasing the FEC offset needs to be done carefully, since it induces and increase in end-to-end delay, which has a negative impact on interactive streams. We are currently studying these issues, as part of a wider project to use PSQA in automatic quality control for VoIP and other multimedia flows.

Concluding Remarks. We have presented an extension of the work presented in [1], which shows that the effectiveness of FEC diminishes as more streams use it. Having a tool like PSQA allows us to perform this kind of studies, both off-line as in this case, and in real–time, in which case they can be used for control purposes. Part of our ongoing work [7] uses these results and analogous ones in order to provide control mechanisms for QoS in a (future) wireless home network. In this context, many devices are connected to the Internet via a broadband connection, and multimedia services such as VoIP, Internet Radio and digital TV use the network as their distribution medium. We are using PSQA to provide quality assessments for the different services, and to evaluate different real–time control mechanisms which aim to improve the perceived QoS.

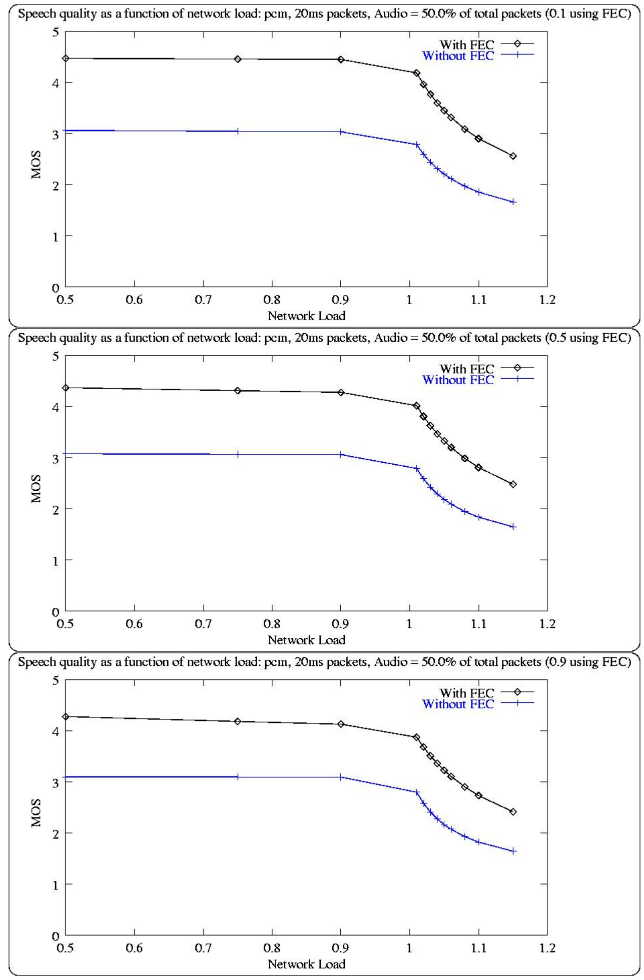

Fig. 1. MOS as a function of network load for different fractions of the voice traffic using FEC (offset = 1 packet)

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