Audio Quality for a Simple Forward Error Correcting Code

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***Abstract*— The aim of this paper is to study the audio quality offered by a simple Forward Error Correction (FEC) code used in audio applications like Freephone or Rat. This coding technique consists in adding to every audio packet a redundant information concerning a preceding audio packet which belongs to the same audio flow. We show that the audio quality depends not only on the number of FEC flows and the utility function associated to the quantity of information received, but also on the traffic conditions. Indeed, no improvement in the audio quality can be obtained for a smooth traffic whereas a marginal improvement can be observed for a bursty traffic. A significant increase of the audio quality is reached for a heavier bursty traffic. We also show that increasing the offset between the original audio packet and the packet bearing its redundancy does not improve significantly the audio quality.**

Keywords: FEC, VoIP, audio quality, Markov chain.

I. Introduction

Recent years have seen a growing use of audio and video applications in the Internet. Unlike file transfer applications like FTP or HTTP, these applications have strong real-time constraints. Yet, even though there is an increasing demand of a more predictable service, the current Internet offers only a best effort service without any performance guaranties on delay variations (jitter) and packet losses for instance.

The compensation for jitter can be accomplished through adaptive playout algorithms [1]. As for the packet losses, they can be handled through a variety of forward error correction (FEC) algorithms and local repair at the receiver. Based on parity codes [2], Reed-Solomon codes [3], [4] or redundant speech codecs [5]–[7], these FEC algorithms send redundant information to compensate for loss.

In this paper, we focus on a simple FEC scheme which has been standardized by the IETF [8]. This scheme has already been used in audio tools like Freephone [9] and Rat [10], and has been generalized in [11]. It consists in adding a low quality copy of the original packet n to packet $n + \phi$. If packet n is lost in the network, it can be recovered and played out by the receiver if packet $n + \phi$ is correctly received. The redundant copy is usually obtained with a lower-bandwidth rate, lower-quality encoding technique such as LPC or GSM. The spacing between the original packet and its redundancy represented by the offset ϕ is a compromise between loss recovery and interactivity. Indeed, a large value of the parameter ϕ is expected to reduce the impact of correlated losses (as it is usually the case in the Internet [11]–[13]), but increases the delay of recovery and the jitter. A high delay would deteriorate the interactivity of a conversation and a high jitter would affect the fluidity of the speech. Few studies of this scheme have been reported so far.

The performance of this FEC scheme has been first evaluated in [14] by extensive simulations. The authors used a rate-distortion metric to measure the audio quality. Their results have shown that the efficiency of this scheme is strongly dependent on the traffic mix. They conclude that it may be useful to use this scheme if the use of redundancy is carefully controlled. Analytical models for the computation of the performance of this FEC scheme are so far limited to smooth traffic conditions (Poisson processes): [15], [16] have used a simple $M/M/1/K$ queuing system to study the audio quality for a single audio flow. The metric used is derived from the packet loss rate using an *utility function* as proposed by [17]. This work has been extended in [18] by using a $M/G/1/K$ queuing system to study the audio quality for an audio flow which was multiplexed with an exogenous flow. It has been shown that the benefit of this simple FEC scheme depends on the number of FEC sources implementing it, and also on the shape of the utility function which is not necessarily linear for multimedia applications.

In the present paper, we continue the work realized in [15], [18] and adapt it to the source model which is described in [19] in order to study the audio quality obtained from this FEC scheme under various conditions of traffic. We observe the behavior of the audio quality when a single audio flow is multiplexed with several input flows generating an exogenous traffic in a more or less bursty way. We also analyze the case where all flows become FEC audio flows. In addition, we study the effect of the offset ϕ on the audio quality. We conclude that the audio quality depends not only on the number of FEC flows and the utility function associated to the quantity of information received, but also on the traffic conditions. Indeed, no improvement in the audio quality can be obtained for a smooth traffic whereas a marginal improvement can be observed for a bursty traffic. A significant increase of the audio quality is reached for a heavier bursty traffic. We also show that increasing the offset between the original audio packet and the packet bearing its redundancy does not improve significantly the audio quality.

The rest of the paper is organized as follows. In Section II, we explain the analytical framework used. The modelization of the packet loss rate is proposed in Section III. We define the metrics used to evaluate this FEC scheme in Section IV. The numerical results obtained from these metrics are reported in Section V and compare in Section VI with results obtained by simulations. Finally, conclusions are drawn in Section VII.

II. Analytical Framework

In this section, we recall the markovian model introduced by Oguz and Ayanoglu in [19]. We briefly describe the overall topology of the system and the Markov chain.

A. Topology of the System

We consider a queue of capacity B packets in FIFO mode. The traffic is generated by one FEC audio source multiplexed with a cross-traffic modeled by $N-1$ independent sources. The FEC audio source generates tagged packets (foreground traffic). The other $N-1$ sources generate a background traffic which interferes with the foreground traffic. At each slot, 0 up to N packets are generated by the N sources depending on their activities and are sent to the queue. Moreover at the beginning of each slot, one packet is served (if the queue is not empty). We assume that the service time of each packet is one slot even if the size of the packets belonging to the audio and the exogenous flows are not necessarily the same (see Section II-D for a discussion of this assumption). We study in the following the tagged traffic generated by the FEC audio source.

B. Source Model

The source model used in this paper is a discrete-time On/Off model which alternates between active state and idle state periods. Let α and β be respectively the idle-to-idle and active-to-active state transition probabilities. At each slot, the state moves from the active to the idle state with probability $1-\beta$ and from the idle to the active state with probability $1-\alpha$. We assume that a state transition takes place just prior to the end of a time slot and that a packet is generated at the beginning of a slot if the new state is the active state.

The stationary probability of the active state is the *normalized load* offered by one non-FEC source and is equal to $\rho_1 = (1-\alpha)/(2-\alpha-\beta)$. We identify a particular source (the tagged source) generating audio traffic at rate ρ_1.

Let ρ_{et} be the load offered by an audio FEC source and r be the ratio between the redundancy volume and the original packet volume (in general $r \in [0,1]$). We have:

$$\rho_{et} \quad = \quad \rho_1(1+r) \ .$$

As r increases, the load offered by the FEC source becomes heavier than the load offered by one non-FEC source. This effect is obtained by increasing the active-to-active transition probability, that is, the average size of the bursts. This new transition probability is equal to:

$$\beta' \quad = \quad 2-\alpha-(1-\alpha)/\rho_{et} \ .$$

The *normalized aggregate load*, which is defined as the total load generated by N non-FEC sources is then computed as $\rho = N\rho_1$, since the sources are identical in the absence of coding. When the traffic generated by one FEC source is mixed with $N-1$ non-FEC sources, the *aggregate load* becomes equal to $\rho = \rho_{et} + (N-1)\rho_1$.

C. The Markov Chain

The Markov chain evolves in a state space $\mathcal{S}_E$ where each state is represented by a triple (b,t,u). For a given slot, $t \in \{0,1\}$ is the state (0 if idle, 1 if active) of the tagged audio source in the slot, $u \in \{0,\ldots,N-1\}$ is the number of non-tagged sources which generate a packet in the slot and $b \in \{0,\ldots,B\}$ is the number of packets contained in the queue after the queueing of the $t+u$ generated packets in the slot. Lost packets in a slot are chosen randomly. $\mathcal{S}_E$ can be partitioned into three subsets: states of $\mathcal{S}_D$ where the tagged FEC source is idle (therefore no tagged packet is lost), and states where the tagged FEC source is active and the tagged packet is lost ($\mathcal{S}_L$) or not ($\mathcal{S}_S$):

$$\mathcal{S}_D \quad = \quad \left\{s=(b,0,u)\right\} \ ,$$

$$\mathcal{S}_S \quad = \quad \left\{s=(b,1,u) \text{ and the tagged packet is saved}\right\} \ ,$$

$$\mathcal{S}_L \quad = \quad \left\{s=(B,1,u) \text{ and the tagged packet is lost}\right\} \ .$$

Moreover, the transition probability q_{ij} from state $i = (b,t,u)$ to state $j = (b',t',u')$ is readily computed from α and β. Details can be found in [19].

D. On the Validity of the Model

While discrete-time models have been naturally proposed for ATM networks (this is the case of [19]), their use in the modeling of IP networks is less common. The strong assumption here is that all packets are assumed to have the same size. We believe that the present model is appropriate for the following reasons. First, it has been observed that the majority of the traffic volume in routers tends to be made of packets with a size of 1500 bytes (the Maximum Transmission Unit in most networks). Therefore, modeling a background traffic in discrete-time (with one time slot per packet) may capture essential features of the system. Second, while the audio traffic is typically made of much smaller packets, we will consider situations where this traffic has a small load, compared to that of the cross traffic. Accordingly, we expect that approximating the real packet size by the packet size corresponding to one time slot will have a small impact on the performance. This fact allows us to handle, with the same model, situations where the audio packet sizes are different due to the addition of a proportion r of redundancy. We validate these modeling assumptions through simulations in Section VI. Finally, we shall also consider the situation where the modeled router is devoted to the audio traffic. The discrete-time model is natural for this case.

Observe that models able to handle arbitrary packet size distributions as well as elaborate traffic sources are much more complicated to analyze and rare in the literature. The usual alternative is to assume an exponentially distributed packet size. The fixed-size packet assumption is closer to reality in many respects.

To conclude on the practical validity of the model, we acknowledge that it has been shown (see [20], [21] and several other recent publications) that the presence of *long range dependence* in the cross traffic may lead to higher loss probabilities than predicted by short-memory models. On the one hand, our model can be extended to account for more persistent packet flows, at the expense of increasing the number of states in the Markov chain. On the other hand, taking such correlation phenomena into account is likely to decrease the efficiency of forward error correction, thereby reinforcing our conclusions. This should be the topic of forthcoming studies.

III. Loss Rate Modeling

We propose in this section a novel recursive formula to compute the packet loss rate, using the Markov chain introduced in Section II.

For this purpose, we define by $g_i^{(\phi)}(m,l)$ the probability that tagged packet n is lost (if $m=1$) or saved (if $m=0$) and that tagged packet $n+\phi$ is lost (if $l=1$) or saved (if $l=0$) when the system is in the state i. Likewise, $g_i^{(\sigma)}(l)$ is the probability that the σ-th next tagged packet is lost (if $l=1$) or not (if $l=0$).

Conditioning on the first transition of the Markov chain, we obtain the recursive formulas (1), (2) and (3) where $1_{\{A\}}$ is the event-indicator function which is equal to 1 if condition A is true and is equal to 0 otherwise.

$$\begin{aligned} g_i^{(\phi)}(m,l) &= \sum_{j\in\mathcal{S}_D} q_{ij} g_j^{(\phi)}(m,l) + \sum_{j\in\mathcal{S}_S} q_{ij} g_j^{(\phi-1)}(l) 1_{\{m=0\}} \\ &+ \sum_{j\in\mathcal{S}_L} q_{ij} g_j^{(\phi-1)}(l) 1_{\{m=1\}} \ , \end{aligned} \tag{1}$$

$$g_i^{(\sigma)}(l) = \sum_{j\in\mathcal{S}_D} q_{ij} g_j^{(\sigma)}(l) + \sum_{j\in\mathcal{S}_S\cup\mathcal{S}_L} q_{ij} g_j^{(\sigma-1)}(l) \ , \tag{2}$$

where $\phi > \sigma \geq 1$, and:

$$\begin{aligned} g_i^{(0)}(l) &= \sum_{j\in\mathcal{S}_D} q_{ij} g_j^{(0)}(l) + \sum_{j\in\mathcal{S}_S} q_{ij} 1_{\{l=0\}} \\ &+ \sum_{j\in\mathcal{S}_L} q_{ij} 1_{\{l=1\}} \ . \end{aligned} \tag{3}$$

The loss of an audio packet generated by the tagged source depends on the type of the arrival state j reached at the next slot in the Markov chain. If the tagged source is inactive in state j, then there is no generation of an audio packet. The value of $g_j^{(\phi)}(m,l)$ (respectively $g_j^{(\sigma)}(l)$, $g_j^{(0)}(l)$) should therefore be computed depending on the tagged audio packet not yet generated (respectively packet n, a packet different than packet n and packet $n+\phi$, packet $n+\phi$). On the other hand, if the tagged source is active then a tagged packet is generated. This packet could be:

1) the audio packet n. We then compute (1):
   a) In the case where the tagged packet is saved ($j \in \mathcal{S}_S$), we compute $g_j^{(\phi-1)}(l)$ provided that $m=0$.
   b) In the case where the tagged packet is lost ($j \in \mathcal{S}_L$), we compute $g_j^{(\phi-1)}(l)$ provided that $m=1$.
2) the audio packet $n+\phi$ (i.e. $\sigma=0$). We obtain (3).
3) an audio packet different than packet n and packet $n+\phi$ (i.e. $1 \leq \sigma < \phi$). In this case, we compute $g_j^{(\sigma-1)}(l)$ using (2) without any additional condition.

For all $i \in \mathcal{S}_E$ and $m,l \in \{0,1\}$, once the values of $g_i^{(\phi)}(m,l)$ are computed, the probability to lose ($m=1$) or save ($m=0$) packet n and also to lose ($l=1$) or save ($l=0$) packet $n+\phi$ is given by:

$$G^{(\phi)}(m,l) = \frac{1}{\rho_{et}} \sum_{i\in\mathcal{S}_S\cup\mathcal{S}_L} p_i g_i^{(\phi)}(m,l) \ ,$$

where p_i is the stationary probability to be in state i of the Markov chain.

IV. Metrics

Let Z_n be a random variable that gives the status of packet n issued from the audio flow:

$$Z_n = \begin{cases} 0 & \text{if packet } n \text{ is not lost,} \\ 1 & \text{if packet } n \text{ is lost.} \end{cases}$$

A. Packet Loss Rate Before FEC Decoding

We define the Packet Loss Rate (PLR) metric before FEC decoding. When the Markov chain is stationary, each packet has the same loss probability before FEC decoding:

$$PLR(r) = EZ_n = \sum_{l=0}^{1} G^{(\phi)}(1,l). \tag{4}$$

B. Audio Quality

We denote by $Q(\phi,r)$ the average audio quality obtained after the reconstruction of the lost original packet from the audio packet bearing the redundant information concerning the lost packet. Since audio quality does not necessarily increase linearly with the volume of data in a packet [17], Altman *et al.* [18] have introduced an utility function U which depends on the quantity of information r, and is such that $U(0)=0$ and $U(1)=1$. As shown in [18], we have:

$$\begin{aligned} Q(\phi,r) &= P(Z_n=0) \\ &+ U(r)\, P(Z_n=1)\, P(Z_{n+\phi}=0|Z_n=1) \ . \end{aligned} \tag{5}$$

Assuming that the Markov chain is in the steady state, the probability to lose packet n coincides with the stationary probability to lose any packet, in other words, with the PLR computed by (4). Next, the probability $P(Z_{n+\phi}=0|Z_n=1)$ is the probability to lose packet n and also to save packet $n+\phi$. Consequently, we have:

$$P(Z_{n+\phi}=0|Z_n=1) = G^{(\phi)}(1,0) \ .$$

Finally, substituting these values in (5) leads to:

$$Q(\phi,r) = 1 - PLR(r)\,(1 - U(r) G^{(\phi)}(1,0)) \ .$$

Observe that for all $\phi \geq 1$, $Q(\phi, 0) = Q(0) = 1 - PLR(0)$.

In the experiments below, we use four utility functions U_0, U_1, U_2 and U_m taken or adapted from [18] which represent extreme cases:

- $U_0(r) = r$, $U_1(r) = \sqrt{r}$, $U_2(r) = r^{\frac{1}{10}}$,
- $U_m(r) = 0$ if $r = 0$, 1 if $r > 0$.

Utility functions U_1 and U_2 obviously lie between the utility function U_0 and U_m. The function U_0 represents the case of an utility proportional to the quantity of information. The function U_m represents an ideally optimistic case where only a small amount of information provides the maximum level of utility, and gives the best upper bound on the maximum audio quality that can be obtained. Since $\lim_{r \to 0^+} U_m(r) = 1$ and since in this case quality appears to be decreasing with r (see next section), then it is possible to quantify the maximum audio quality Q^* that can be obtained by:

$$Q^*(\phi) \;=\; 1 - PLR(0)\,(1 - G^{(\phi)}(1, 0)),$$

and the maximum improvement by:

$$Q^*(\phi) - Q(0) \;=\; PLR(0)\,G^{(\phi)}(1, 0).$$

Admittedly, this utility model needs to be validated against subjective quality measurements. If such a correspondence is at all possible, one may safely assume that the perceived quality depends on correlations between losses. The metric defined in (5) uses two consecutive packets as a first approximation of the complete process.

We show in Section V that for certain conditions of traffic we can obtain significant improvement of the audio quality by increasing slightly the redundancy r.

V. Numerical Results

In this section, we present numerical results showing the variation of the audio quality according to the redundancy parameter r and the offset ϕ. We set the offset ϕ such as $\phi \in \{1, 5\}$. Note that the use of large offset ϕ is not feasible for interactive conversations because of the added delay. Similar conclusions can be drawn for audio streaming since a large offset may not be much better than retransmissions.

The numerical results are obtained for a buffer size $B = 25$ packets, for $N = 16$ sources and for different types of traffic depending on the configuration of the coefficients α and β for the FEC audio source and for the $N - 1$ other sources generating the cross-traffic.

We examine in Section V-A the audio quality obtained from an audio FEC source that generates a smooth traffic, when the nature of the exogenous traffic varies. This would be the situation at an access point of the network. As in [22], we also observe in Section V-B the impact of FEC on the audio quality when the FEC audio source generates a more bursty traffic. This would be the situation when the network introduces jitter in the flow of audio packets. The audio quality obtained when all flows are audio FEC sources is studied in Section V-C.

A. *Smooth Audio Traffic*

We consider in this section a single audio FEC source set up in such a way that $\alpha = 0.995$ and $\beta = 0$ and that generates packets bursts of size 1. The normalized load of this source is $\rho_1 = 0.00497$ and its behavior is similar to audio sources in the Internet.

We vary the cross-traffic by modifying the configuration of the $N - 1$ other sources. In order to obtain a bursty traffic which gives for each of these sources a normalized load of $\rho_1 = 0.05$ (Fig. 1(a)), we set up $\alpha = 0.995$ and $\beta = 0.905$. The total load of the cross traffic is 0.75. In this case, an excellent audio quality of about 94.72% is observed in the absence of redundancy for $\phi = 1$ (Fig. 1(a)). However, it is possible to improve slightly the audio quality according to the utility function for small values of r ($r \leq 0.2$). For instance, for $r = 0.1$ and for utility function U_m, the audio quality equals 94.91% when $\phi = 1$.

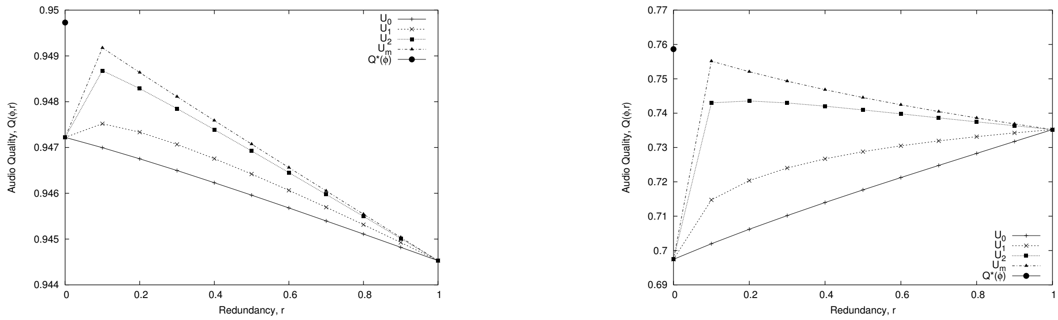
Figure 1(b) illustrates the case where $N - 1$ sources are set up in such a way that $\alpha = 0.99$ and $\beta = 0.91$ (hence $\rho_1 = 0.1$) so as to obtain a heavier bursty traffic for the cross-traffic with a load of 1.5. In this case, we observe as expected that the audio quality obtained from the FEC source is lower than the one obtained from the configuration presented in Fig. 1(a). Nevertheless, it is possible to improve the audio quality by increasing the amount of redundancy contained in every audio packet. We notice that the quality audio increases uniformly with r for the utility function U_0 and U_1. This is not the case for the utility functions U_2 and U_m. However, for small values of r, the audio quality can be improved. For instance, for $\phi = 1$ and $r = 0$, $Q(\phi, r) = 69.74\%$ whereas the audio quality equals 75.51% for U_m when $r = 0.1$. Notice that for $\phi > 1$, the increase of the offset between the packet bearing the original information and the packet bearing the redundant information allows only to obtain a marginal improvement of the audio quality as reported more extensively in [23].

Finally, we set up $\alpha = 0.96$ and $\beta = 0.24$ (and therefore $\rho_1 = 0.05$) for the $N - 1$ sources generating the cross-traffic so as to obtain a *smooth traffic* with bursts of average length 1.3 and idle periods of average length 25 In this case, the audio quality obtained is close to 100% when $r = 0$. It is not necessary to add redundancy in the audio packets since the audio quality decreases with r. There is no need to increase the offset ϕ either since the audio quality is already excellent for $\phi = 1$.

B. *Bursty and Heavier Bursty Audio Traffic*

In this section, we consider that the FEC source is set up in the same way as the $N-1$ sources generating the cross-traffic.

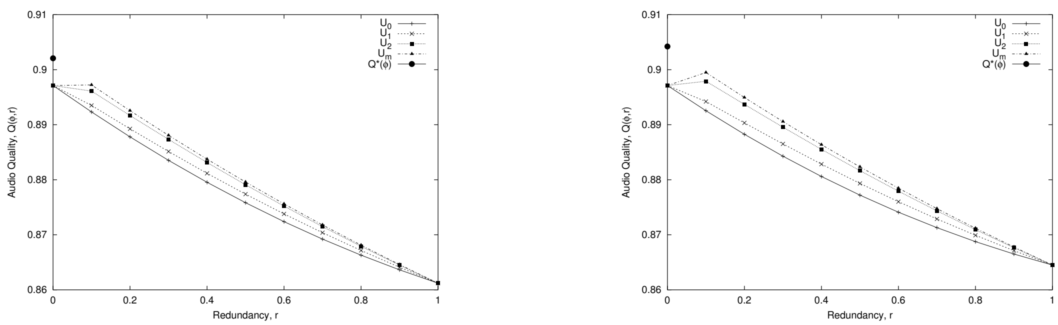
The bursty traffic generated by one FEC source and $N - 1$ identical cross-traffic sources (Fig. 2(a) and Fig. 2(b)) gives an aggregate load $\rho \in [0.8, 0.85]$. Hence we set up $\alpha = 0.995$ and $\beta = 0.905$ for the N sources. We can observe that the curves of the utility functions U_0 and U_m are close to each other. However, the audio quality for U_0 decreases monotonously with r, whereas the function U_m improves the audio quality for small amount of redundancy ($r < 0.2$). This means that



(a) Bursty Traffic for the Cross-Traffic.

(b) Heavier Bursty Traffic for the Cross-Traffic.

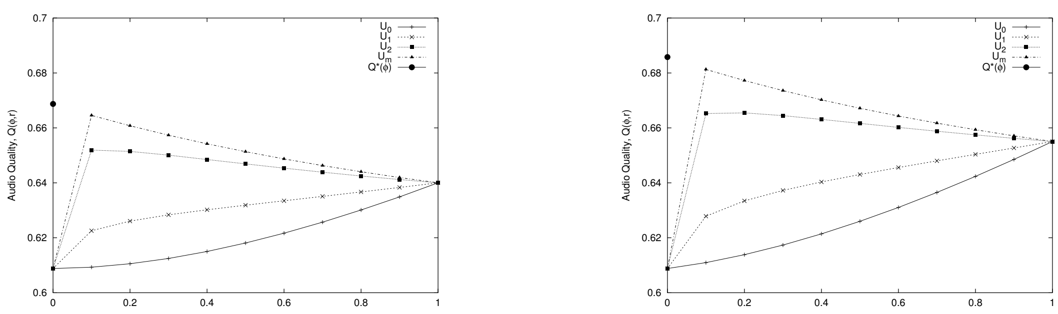
Fig. 1. Audio Quality for a Smooth Audio Traffic with 1 FEC Audio Source and $\phi = 1$.



(a) 1 FEC Source, $\phi = 1$.

(b) 1 FEC Source, $\phi = 5$.

Fig. 2. Audio Quality for a Bursty Traffic with Different Offsets.



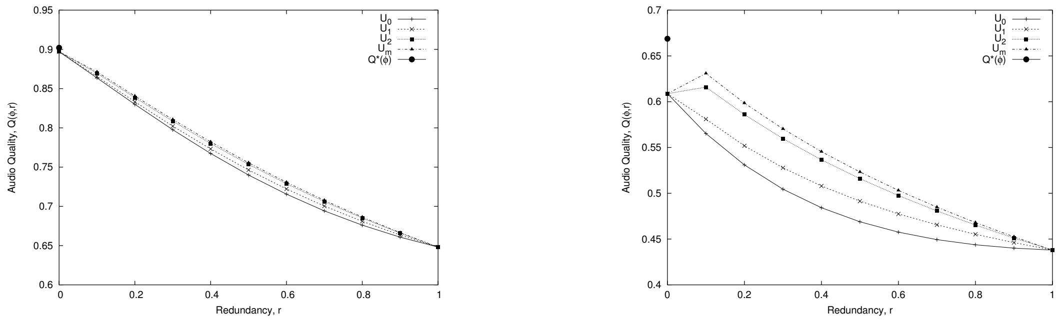
(a) 1 FEC Source, $\phi = 1$.

(b) 1 FEC Source, $\phi = 5$.

Fig. 3. Audio Quality for a Heavier Bursty Traffic with Different Offsets.

the addition of a small amount of redundancy in every audio packet (specifically using a LPC codec) can slightly increase the audio quality provided that the utility is close to U_m. This improvement peaks at 90.2% for $\phi = 1$. Furthermore, we can observe in Fig. 2(b) that the increase of the offset ϕ is not effective for both utility functions.

The heavier bursty traffic generated by one FEC source and $N-1$ identical cross-traffic sources (Fig. 3(a) and Fig. 3(b)) overloads the system with an aggregate load $\rho \in [1.6, 1.7]$ (we have $\alpha = 0.99$ and $\beta = 0.91$). In this case, we obtain obviously a lower audio quality as compared to the audio quality obtained from a bursty traffic. However, it is possible to increase the audio quality and to obtain an important improvement as compared to the one observed in the case of a bursty traffic. For the function U_0, the audio quality increases uniformly with r. This means that the audio quality can be improved by the use of a more powerful audio coder that can generate the redundancy contained in each audio packet. On



(a) Bursty Traffic.

(b) Heavier Bursty Traffic.

Fig. 4. Audio Quality for N FEC Audio Sources and $\phi = 1$.

the other hand, the function U_m increases significantly the audio quality when a small amount of redundancy is added in every audio packet by the use of LPC or GSM codecs. Moreover, it is more beneficial to increase the offset ϕ in this case than in the case of a bursty traffic. This suggests increasing the offset ϕ as much as the application allows. On the other hand, further experiments, not reported here, show that the improvement with respect to ϕ quickly reaches a plateau. Increasing ϕ above 5 is not useful anymore.

C. Several FEC Audio Sources

For the configuration of Fig. 4(a) where all sources are audio FEC sources (the aggregate load is here equal to $\rho = N\rho_1(1+r)$) and generate a bursty traffic, we obtain a heavier aggregate load since $\rho \in [0.8, 1.6]$. In this case, the curves for U_0 and U_m are almost identical and therefore the dependence on the utility function becomes less important than in the case of the configuration of Fig. 2(a) to Fig. 2(b). Moreover, the audio quality decreases with r more quickly than the curves in Fig. 2(a) to Fig. 2(b) (for instance, for $\phi = 1$, it decreases from 0.897 for $r = 0$ to 0.648 for $r = 1$), and the increase of ϕ does not improve significantly the audio quality.

When all sources become audio FEC sources and generate a heavier bursty traffic (Fig. 4(b)), the aggregate load is $\rho \in [1.6, 3.2]$. The addition of redundancy allows an increase of the audio quality for the function U_m for small r ($r \leq 0.2$ when $\phi = 5$). This is not the case for the function U_0 since it decreases continuously with r. Finally, it should be noted that in spite of a heavy network load, it is possible to improve the audio quality by increasing the offset ϕ.

VI. Validation

We describe in this section the validation of the modeling assumptions made in Section II. The issue is whether the discrete-time assumption, corresponding to a fixed network packet size and synchronous sources, is accurate enough. We have performed simulations under ns [24] with the network topology represented in Figure 5. The source S_1 is the (tagged) audio source generating an average of 50 packets per second. The 15 remaining sources are On/Off UDP sources, with a

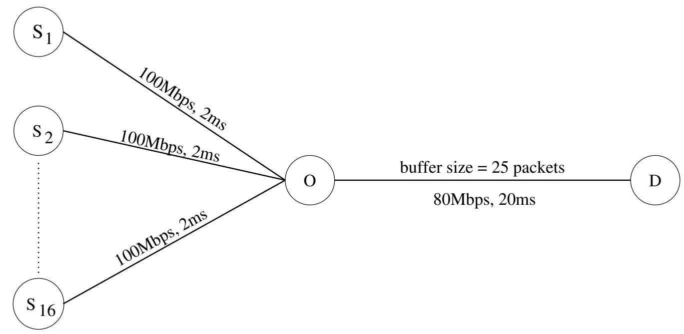


Fig. 5. Network Topology.

packet size of 1000 bytes and a peak rate of 80Mbps. The characteristics (bandwidth, latency) of the links are given in the figure. In particular, the output link has a bandwidth of 80Mb/s, which corresponds to a time slot of 0.1ms in the model.

The simulations reported in Table I correspond to different scenarios where: a) the audio traffic differs: Constant Bit Rate (CBR) or Poisson process (corresponding to the model under the smooth traffic conditions of Section V-A), b) audio packet sizes vary, and c) the cross traffic conditions vary. The parameters for the On/Off cross traffic sources are: $\alpha = 0.995$, $\beta = 0.905$ for the bursty traffic (which gives average idle and burst times of 20ms and 1.05ms respectively), and $\alpha = 0.99$, $\beta = 0.91$ for the heavier bursty traffic (which gives average idle and burst times of 10ms and 1.11ms). We have measured the Packet Loss Rates (PLR) for the audio source and for the cross traffic (Table I). Each estimate and its 95% confidence interval result from 100 independent simulations over 300 seconds.

The results show clearly that loss rates are not very sensitive to the audio packet size, nor to the nature of the audio traffic (CBR or Poisson). The PLR statistics also match quite well the values predicted by the model: 0.0527 for a bursty cross traffic, and 0.3025 for a heavier cross traffic (see Figures 1(a) and 1(b)). We conclude therefore that the model reproduces

TABLE I
COMPARISON OF LOSS RATES FOR DIFFERENT TRAFFIC SCENARIOS.

Type of the Audio Traffic	Type of the Cross-Traffic	Audio Packet Size (bytes)	PLR Audio	PLR Cross-Traffic
Constant Bit Rate	Bursty Traffic	180	$0.05577 \pm 4.02\,10^{-4}$	$0.09710 \pm 2.04\,10^{-4}$
		1000	$0.05607 \pm 3.74\,10^{-4}$	$0.09824 \pm 1.97\,10^{-4}$
	Heavier Bursty Traffic	180	$0.31889 \pm 8.10\,10^{-4}$	$0.36276 \pm 2.99\,10^{-4}$
		1000	$0.30490 \pm 7.77\,10^{-4}$	$0.36415 \pm 1.88\,10^{-4}$
Poisson Process	Bursty Traffic	180	$0.05623 \pm 3.92\,10^{-4}$	$0.09738 \pm 2.14\,10^{-4}$
		1000	$0.05666 \pm 3.79\,10^{-4}$	$0.09795 \pm 2.14\,10^{-4}$
	Heavier Bursty Traffic	180	$0.30385 \pm 7.47\,10^{-4}$	$0.36278 \pm 2.20\,10^{-4}$
		1000	$0.30526 \pm 7.15\,10^{-4}$	$0.36411 \pm 2.02\,10^{-4}$

adequately the features of the system when the audio traffic represents a small proportion of the traffic.

VII. CONCLUSION

In this paper, we studied the performance of a simple FEC scheme implemented in recent audio tools like Rat [10] and Freephone [9]. This scheme consists in adding a low quality copy of the original audio packet n to the audio packet $n+\phi$. Our model is based on the Markov chain presented in [19] and on a novel recursive formula which computes the probability to lose or to save packet n and packet $n+\phi$.

We showed that the FEC scheme can improve the audio quality depending not only on the number of FEC flows and the utility function as shown in [15], [18] but also on the traffic conditions. But a significant improvement of the audio quality is only observed in the case of a heavy bursty traffic for the background traffic. This traffic condition illustrates that unlike [18], the audio quality can increase in presence of a linear utility function. Even though the increase of the audio quality for the case where all sources are FEC depends on the utility function as shown in [18], it is only for a heavier bursty cross-traffic for the background traffic that a significant improvement is obtained. Furthermore as in [18], we observed that quality increases with ϕ. However, it is more interesting to increase ϕ for a heavier bursty traffic rather for a bursty or a smooth traffic. Yet, an increase above a certain threshold gives a marginal improvement. In addition, we showed that no improvement in audio quality can be obtained for a smooth traffic. This confirms results of [18]. However, the quality observed in the absence of redundancy is close to one in that case. In any case, the quality generally decreases with r and FEC should be used with as few redundancy as possible, depending on the actual shape of the utility function.

In conclusion, the FEC scheme does not seem to be very efficient for most traffic conditions. However, it appears to be more interesting when the cross-traffic is a heavy bursty traffic.

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