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ANALYSIS OF PACKET LOSSES OF CONVERSATIONAL VOICE CALLS ON A THIN ROUTE COMMUNICATIONS CHANNEL

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Abstract

An analysis of packet losses on a low capacity communication channel is presented, considering channel capacity of 64 kb/s, each voice source activity is 40% and silence is 60%. The analysis results indicate that on top of 9 users extra 11 users can be accommodated, with a 5% packet loss, i.e 20 users can be allowed on the channel, where each voice source is coded at 6.4 kb/s. Thus 20/9 = 2.27 a bandwidth advantage can be achieved.

Key words: Packet discarding technique, low bit rate, thin route, single channel per carrier

Introduction

The Single Channel Per Carrier (SCPC) satellite channels or a 64 kb/s leased telephone lines are commonly known as thin route channels [1]. The transmission capacity of these channels is limited because of low bandwidth offered by the channels. In normal telephonic conversation, the average speaker is active 50% of the time. When silences or pauses between talkspurts are accounted for, it turns out that the average speaker is active approximately 40% of the time and the rest 60% of the time, the speaker is silent [2] [3] [4]. The communication channel during the periods of silences can be utilized for additional number of users through multiplexing techniques. However, it is simply not possible that all silences occur simultaneously of equal sizes, so that they can be utilized for another set of active users. The scenario of activities and silences is a random phenomenon and different to predict that, which user is going to generate silence and talkspurt at what time. In order to fully utilize the silence durations various techniques can be employed with voice Activity Detector [5]. In this paper packet loss statistics and analysis is presented. This alternatively lead to more than double bandwidth on same channel that is 20 instead of 9 users on the same link.

Speech frame losses

Considering 20 speech sources are contending for channel, each one is alternating between talkspurts and silences. Further assuming that these sources transmit frames (packets) at the same constant rate (one every 30 ms) while in talkspurts and that no frames are transmitted during silences. All frames are assumed to contain an equal number of bits. Let these sources deliver frames to a single multiplexer (switch) and share, with equal priority, a single output line whose capacity is such that a maximum of 9 frames can be transmitted every T_f (30 ms).

In a single T_f, up to 20 frames can be produced by the sources and up to 9 frames can be sent out on the channel. In some T_f more than 9 frames can be generated, and a queuing discipline must be established to deal with the resultant excess frames. Since a large buildup of speech delay is intolerable, some limit on queue size or queuing time should be imposed. A scheme that will be considered here, limiting the duration of queuing such that no frame remains in the system for time longer than T_f (30 ms). If $k > c$ frames are produced in T_f, $k - c$ of these frames can be chosen among these k active

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talkers, and will be discarded. If $k \leq c$ frames are produced in an interval, all will be transmitted. The fractional frame loss associated with this scheme is easy to derive and provides an over bound to the frame loss incurred in a scheme where overflow of frames occur. Given the above assumptions, the fraction of frame loss can be calculated quite easily. Over long time duration MT_f, where M is number of links, the total number of speech frames offered by the n talkers will be given by the ratio.

$$F_{offered} = \sum_{k=0}^{n} k(k,n,p) \quad \text{equ:(1)}$$

Eqn. (1) gives the number of frames offer and Eqn. (3) can compute the resulting cutout fraction of frame losses.

$$F_{discarded} = \sum_{k=c+1}^{n} (k-c)(k,n,p) \quad \text{equ:(2)}$$

$$F_{loss} = \frac{1}{np} \sum_{k=c+1}^{n} (k-c)(k,n,p) \quad \text{equ:(3)}$$

$k-c$ ($c=9$)	Probabilities of F_{loss}
1	0.1171416
2	0.1419898
3	0.1064923
4	0.0582522
5	0.0242718
6	0.0077670
7	0.0018878
8	0.0003384
9	0.0000423
10	0.0000033
11	0.0000001
$F_{discarded}$	0.4581865
Average F_{loss}	0.0572733

Table: 1 Fraction of Frame loss of 64 kb/s

$F_{discarded}$ used for the single user equ:(2) and F_{loss} used for the average loss.

It can be emphasized that the above analysis describes packet loss through a single switching (multiplexing) node, and do not deal with frame losses which might occur at additional intermediate nodes between the sources and the destination. Assuming the link capacity of 64 kb/s and each source of 6.4 kb/s, such as PRELP [6]. The sources are all using VAD, to detect and utilize silences in conversational mode each source is active for 40% and silent for 60% of the time [7]. Then the analysis can be carried out by using Equ: (4).

$$P_k = \frac{n!}{k!(n-k)!}(p)^k (q)^{n-k} \quad \text{equ: (4)}$$

where
n = total numbers of users
k = number of active users
p = probability of activity
$q = 1 - p$ = probability of silence

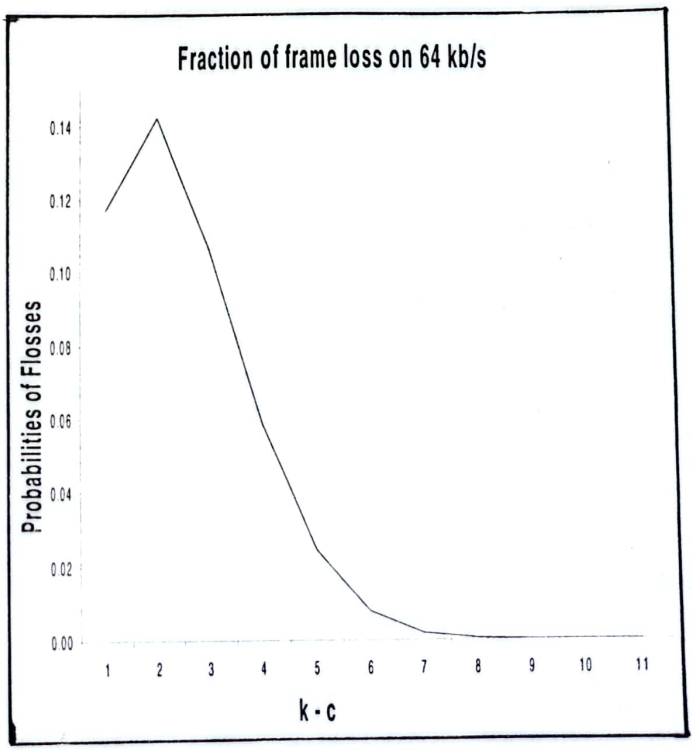


Figure: 1 Fraction of Frame loss of 64 kb/s

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Discussion

In Table 1, k is the number of users greater than c (9) channel links available. $F_{discarded}$ is the total discarded packet probabilities, whereas, Average F_{loss} is the packet loss received by each user. The k is calculated from the equ: (3). The analysis of voice calls of normal conversation for channel of a bandwidth of 64 kb/s is carried out with each user having 6.4 kb/s.

The analysis results indicate that allowing 20 users on the link that can carry 9 users in normal circumstances, is possible only when (a) silences are detected and utilized and (b) 5% packet loss can be allowed. This 5% loss may bring loss to speech quality but lost packets can faithfully be reconstructed to mitigate the effects. This alternately bring 20/9 = 2.27 gain over bandwidth of the channel.

Conclusion

The analysis indicate that at 5% packet loss and 60% detection of silences can allow 20 users of same capacity on the channel that can carry only 9 users in normal circumstances. The bandwidth advantage of 20/9 = 2.27 can be achieved as indicated by analysis.

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