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DESIGN OF STATISTICAL TIME DIVISION MULTIPLEXER BASED ON RANDOM PACKET LOSS ALGORITHM

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Abstract

Statistical Time Division Multiplexers (STDM) are used to accommodate more users on typical communication channel than the traditional multiplexers can afford. However, situations do arise when the number of users bandwidth requirement exceed the maximum capacity of a channel bandwidth. In such situations some users are allowed to continue communicating, while others are forced to drop small parts of their speech signals. In the proposed STDM design the excessive users are picked on random basis so that their small packets can be dropped to reduce congestion on the channel. Experimental results indicate that upto 3% of a speech signal can be dropped in addition to Voice Activity Detection (VAD) without affecting much of the quality of the speech for each user. In this design a 64 kbps channel is considered to transmit monologue speech. Each user's speech is compressed at 6.4 kbps. It is experienced that in addition to 10 users, 2 extra users can be further multiplexed on the channel.

Keywords: Statistical Time Division Multiplexer design, Random Packet loss Agorithm

Introduction

STDM is a non-deterministic technique, where allocation of time slot to each user is always made on temporary basis. In this design, allowable number of users on channel is always greater than the normal channel bandwidth. Since more users are allowed on the channel than channel can afford, therefore, there is probability that all users can become active and demand the channel bandwidth at the same time. When activity of users is detected high, some of the users can be denied the access to the channel. In this paper users are randomly selected for discarding smaller amount of the data (packet), so that channel load can be regulated. Is has been observed that using random discarding technique, on top of 10 users, 2 extra users can be allowed on the same link with minimum loss of speech quality.

Voice Activity Detection

Speech signal is a continuous signal, which consists of silences and talksparts, alternatively. The silences can be detected by VAD and utilized in order to increase channel efficiency. Many algorithms for VAD are reported in [1][2][3]. For monologue speech transmission mode, such as radio, or one way speech, these silences are reported to be 20% of the total speech durations [4][5][6][7]. Utilization of these gaps for other active users is a function of (a) channel capacity and (b) number of users allowed on the link [8][9][10][11][12].

Mean Opinion Score

Mean Opinion Score (MOS) that provides a subjective measure of the quality of human speech[8]. The scheme uses subjective tests (opinionated scores) that are mathematically averaged to obtain a quantitative indicator of the system performance. To determine MOS, a number of listeners rate the quality of test sentences read aloud by male and female speakers. A listener gives each sentence a rating as follows: (1) bad; (2) poor; (3) fair; (4) good; (5) excellent.

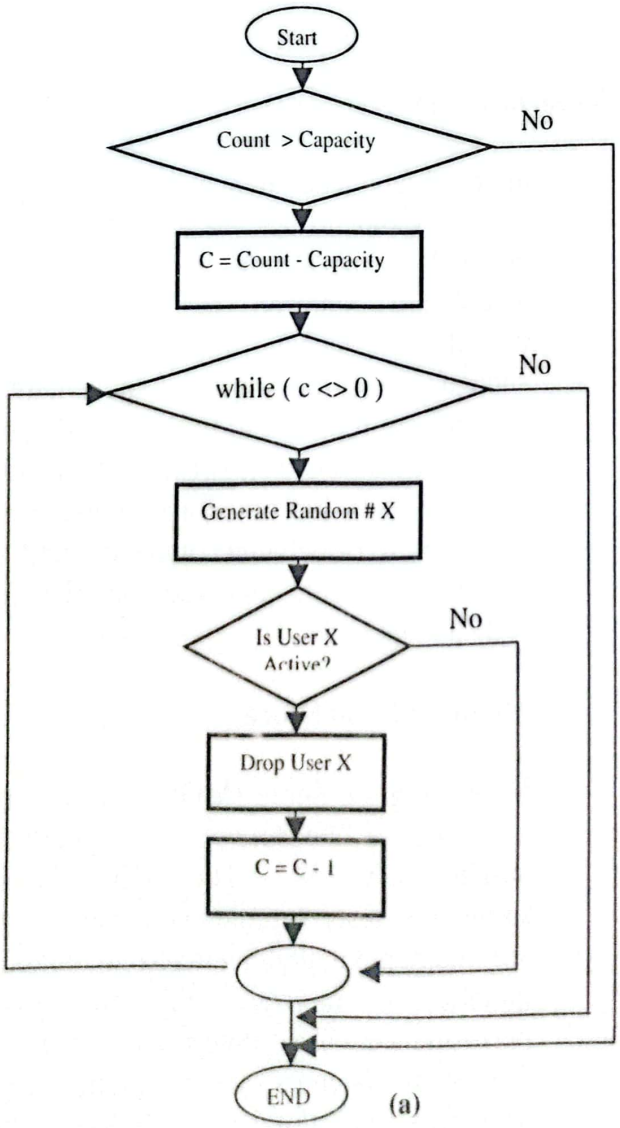
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Random Packet Discarding Algorithm

When total number of users detected active is greater than nine, multiplexer invokes Random Packet Discarding algorithm. For example, 13 out of *N* users want to transmit speech through channel, therefore, any 4 users are required to be discarded for a short duration. In order to pick four users, the algorithm generates four random numbers between 1 – *N*. This algorithm activates each time, whenever the activity of users is higher than the maximum limit of users that is 9 at a time, excessive users are required to drop their packets as follows:

1. A number is generated between 1 and *N* on random basis.
2. If generated number matches with one of the active user's identification and that user is active also, then a packet of that user is discarded. This is same as the user is denied the access.
3. The steps 1 and 2 repeat for required number of times.

The Loss Algorithm's flowchart and pseudo code are shown in Figure 1(a) and 1(b) respectively.



(a)

```
BEGIN
    IF Count > Capacity THEN
      C = Count – Capacity
      While C <> 0
        Generate Random number X
          IF X is Active then
            Drop X
            C = C – 1
          END IF
      END WHILE
    END IF
END
```

(b)

In Figure 1 (a) *count* is the number of active users on the channel, is compared with channel capacity. The loss algorithm will only be activated if *count* is greater than the channel capacity that is nine users at a time. In order to know that how many users needed to be dropped, *capacity* is subtracted from *count* and result is stored in C, from here a loop will start and repeat C times. Each time algorithm generate a random number X between $1 - N$ and checks whether user with identity of generated number X is active or not. If user X is active then C is decremented by one after discarding a packet of user X. This process will repeat until C becomes zero.

Subjective Measurers For Random Packet Loss Algorithm

The MOS test score is presented in Table 1, for this scheme, around 20 experienced subjects were engaged in the test. The speeches compressed at 6.4 kbps were randomly arranged for each participant. The score were obtained on a grading procedure in the range of (5) excellent, (4) good, (3) fair, (2) poor and (1) bad. In this test, three types of speech are experimented. Speech without VAD active, speech with VAD active and no packet discard and speech with VAD active packet discard.

Users	File (a)	File (b)	File (c)	File (d)	File (e)	VAD
–	3.8	4.0	3.8	4.0	3.6	OFF
9	3.5	3.8	3.5	3.5	2.7	ON
10	3.0	3.5	3.0	3.0	2.5	ON
11	2.8	3.2	3.0	2.7	2.2	ON
12	2.5	3.1	3.0	2.2	2.2	ON
13	1.8	2.5	2.5	2.0	2.0	ON
14	1.6	2.0	1.8	1.8	1.5	ON
15	1.5	1.6	1.5	1.4	1.5	ON

Table 1. MOS Test Results for Fifteen Users

MOS results for all speech file (a) – file (e) (i.e. VAD is inactive) is graded between 3.6 to 4.0 as shown in Table 1, that means fair to good speech quality [13][14]. As soon as nine users were allowed on the channel, keeping VAD activity, certainly no one of them is receiving any packet loss, because channel can afford all nine users. In spite of that speech quality for nine users is rated lower than the first case when VAD is in active. The reason behind this loss of quality is obviously VAD itself. Because in certain parts of the signal, VAD fails to detect speech. This loss of quality has been reduced though hangover of VAD by limit for 4 speech frames (120ms). The four frames hangover means, after detecting silence, VAD still declares silence as active speech in order to avoid clipping as mush as possible and that ultimately results in less loss of speech

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quality. The hangover and its advantages and disadvantages are given in [15].

Allowing 10 users on the link, occasionally all ten users become active, in this situation any one of the users is picked up randomly for dropping its frame. Dropped frame is reconstructed at the receiver in spite of that the quality will suffer loss for two reasons (a) packet reconstructing and (b) VAD loss that is inherit in the system due to the limitations of VAD and hangover setting. Moreover, the quality for ten users may get further deteriorated if for next time frame same user is picked up again for speech packet drop. This brings a consecutive loss to same user for two speech frames. The reconstruction mechanism [16], is not able to provide due to its limitations, a best substitute for consecutive frame (packet) loss.

Conclusion

Monologue speech transmission through STDM, the gaps of each speech source care utilized for active users. The full utilization of these gaps is based upon the number of time slots available at the channel. Normally, total capacity requirement of users at the multiplexing end for STDM is more than the channel can afford to provide the service. Therefore, occasionally, more users are demanding slots, in order to control this situation, a random drop algorithm is designed during higher activities, some of the users are forced to drop their packets for speech frame durations i.e. 30ms. The dropped packets are reconstructed at the receiving end to mitigate the effect of dropped packets. The simulation results indicate that in this method the utilization of channel can be increased by the ratio of 12 / 9 = 1.33 i.e. normally 9 users, each have 6.4 kbps capacity are allowed to a 64 kbps channel, a single 6.4 kbps is resumed for signaling information 9 + 1 = 10. MOS results indicate that 12/9 = 1.33 or 3 extra users can be allowed with an acceptable loss of speech quality, with 3 extra users, with little afforded degradation in speech quality.

References

[1] "Voice Activity Detection for Cellular Networks". Technical Report CA 93106 USA Center for Information Processing Research, Departments of Electrical and Computer Engineering University of California.

[2] Freeman, D. Cosier, G. Southcott, C.B. and I.Boyd. "The Voice Activity Detector for the Pan-European Digital Cellular Mobile Telephone Service". In proc. of ICASSP, 369–372 Glasgow, U.K, 1989.

[3] Kondoz A.M. and Evans. B.G. "A High Quality Voice Coder with Integrated Eccho Canceller and Voice Activity Detector for Mobile Satellite Applications". *3rd Int. Mobile Satellite Conf. and Exhibition*, June 1993.

[4] Paul Brady. T. "A Technique for Investigating On-Off Patterns of Speech". *Bell Sys. Technical Journal*, 1-22, Nov. 1965.

[5] Paul Brady. T. "A Statistical Analysis of On-Off Speech in 16 Conversations". *Bell Sys. Technical Journal*, 73-91, Jan. 1968.

[6] Paul Brady. T. "A Model for Generating On-Off Speech Patterns in Two-Way Conversation". *Bell Sys. Technical Journal*, 2445-2472, Sept. 1969.

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[7] John G. Gruber. "A Comparison of Measured and Calculated Speech Temporal Parameters Relevant to Speech Activity Detection". *IEEE Trans. on Communications*, 30(4): 728-738, 1982.

[8] David H. A. Black. "PLC-1: a TASI System for Small Trunck Groups". In *Proc. of IEEE Int. Conf. on communications*, 49.1.1- 49.1.5, Denver, Colorado 1981.

[9] Bullington K. and Fraser, J. "Engineering aspects of TAST". *Bell Sys. Technical Journal*, 38, March 1959.

[10] Martin J. Fischer. "Delay Analysis of TASI with Random Fluctuations in the Number of voice calls". *IEEE Trans.* on *Communications*, 28 (11): 1883-1889, Nov. 1980.

[11] Bullock D. B. Fraser J. M. and Long. N. G. "Over-All Characteristics of a TASI Systems".*Bell Sys. Technical Journal*, 1439-1454, July 1962.

[12] Michael S. Nakhla and David H. A. Black. "Analysis of a TASI System Employing Storage".

[13] Scagliola C. Mc Dermott B. and Goodman. D. "Perceptual and Objective Evaluation of Speech Processed by Adaptive Differential PCM". *Bell Sys. Technical Journal*, 1597-1618, May-June 1978.

[14] Augustine J.R. Gary H. and Markel. J.D "Distance Measures for Speech Processing". *IEEE Trans. on ASSP*, 24(5):380-391, Oct. 1976.

[15] Shah, A. "An Intelligent Multiplexer Architecture for Thin Route Communications".Ph. D. thesis, University of Surrey, Guildford, U.K., 1998.

[16] Kondoz A. Atungsiri, S. Soheili R. and Evans, B. "Effective Lost Speech Frame Reconstruction for CELP Coders". In *Proc. of EUROSPEECH Conf.*, volume 2, 599-602, Italy, 1991.

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