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ANALYSIS AND SIMULATION OF A DETERMINISTIC PACKET DISCARDING TECHNIQUE IN LOW BIT RATE MULTIPLEXING ARCHITECTURE FOR RURAL TELEPHONY

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ABSTRACT

Providing cheap telephonic service to the rural area customers has become possible due to (a) Recent development in low bit rate speech coding (b) multiplexing statistically many low bit speech source to out going channel, and employing Digital Speech Interpolations (DSI) to further maximize the utilization of the channel. In this paper a low bit rate (thin route link) speech based Statistical Time Division Multiplexer (STDM) architecture analysis and simulation results are presented. To control the congestion of speech packets generated by the users at the input of multiplexer the user frames are discarded in a deterministic way. The discarded information is reconstructed at the receiver and transmitter. The DSI advantage, that is "the ratio of input sources to out going links" has been presented.

Key words: Packet discarding technique, low bit rate multiplexing architecture.

Introduction

DSI systems permit number of "N" input sources to be multiplexed over a limited number of out going links "C" available for sharing. By employing Vice Activity Detection (VAD) for each input speech source the silence periods can be utilized for active users. Study indicates that silence detected for conversational speech is 60% and for commentary type speech silence gaps are 20%.[2] [3].

Achieving the highest possible Digital Speech Interpolation (DSI) advantage only with VAD is not possible. However Lost Speech Frame Reconstruction (LSFR0 can play an effective role for bandwidth saving. In Section 1 simplex schematic of a rural telephone system is presented. In Section 2 a detailed analysis of the traffic load of the proposed multiplexer architecture is presented. The mutliplexer architecture is discussed in Section 2. In section 4 the frame loss statistic is presented. Finally in section 5 the conclusion and results are presented.

Rural Multiplex Equipment

The Rural Multiplexing Equipment (RME) switch calls both within the rural community and out to the wider (PSTN via the thin route trunk. A major bottle knack for such a system is the limited capacity of the thin route connection with the out side word.

Traditional STDM multiplexer architectures exploits speech interpolation to maximize trunking efficiency. The efficiency of such a trunk therefore is a combined function of the bandwidth required per mutliplexed users and the activity ratio. For a digital trunk, VAD and Discontinuous Transmission (DTX) replace speech interpolation whilst bandwidth requirement translates to bit rate requirement per user. The LBR speech codecs itself has a potential of increasing the number of users that can be multiplexed onto the out going trunk. In a traditional multiplexer this number $N \leq W/B$, where W is the bit rate of the outgoing trunk and B is the required bit rate per user.

The proposed lossy multiplexer is derived from the nature of contemporary LBR speech codecs. Low bit rate speech coders encode and transmit segments of frames of speech in packets. During transmission, some of these packets may be lost due to channel errors or outgoing link congestion. It has been shown that such codecs can still provide reasonable speech quality even at a random frame loss rate (FLR) of 3% by employing lost speech frame reconstruction (LSFR) at the decoder [1]. When the trunk from the multiplexer is congested, individual input codecs are forced

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to drop some packets. By employing an effective LSFR technique at the receiver, speech quality is still guaranteed for each active channel, provided the FLR for each channel does not exceed 3%.

Analysis of a Proposed Multiplexer Parameter

Formulation:

$Pk = \backslash m! / k! (m-k)! * (q)^k (1-q)^{m-k}$

Pk is the probability of k users active, m is total number of users on the link, q is the probability of each user being active that is 80% (monologue) speech). Moreover, channel capacity is assumed 64 kb/s, and each input source of generating 6.4 kb/s digitally encoded speech.

The user k Activity	80%
1	.00000000196
2	.0000000550
3	.000000954
4	.0000114
5	.0001007
6	.0006717
7	.00345
8	.027638
9	.02579
10	.1031823
11	.18760
12	.25013
13	.23089
14	.13194
15	.00703

However, the output link capacity is assumed to carry 9 parallel channels, each of 6.4 kb/s capactiy, a single 6.4 kb/s is reserved for signaling inforation required to safe gaurd data bits being transmitted.

$\varphi p = 1/m \sum m (k = c+1) (k-c) b(k,m,q)$

φp is the average pack loss, c is the number of channels for sharing and b(k,m,q) is the binomial situation.

Cyclic Frame discarding Multiplexer Algorithm Design

Adjacent frame losses have been encountered in the random scheme [4]. By regulating that loss better performance can be achieved. The problem of loss of adjacent frames is tackled by a frame discarding mechanism is which the loss from control mechanism goes through al the active user first and only then returns to discard the next frame from users. Two control parameters are set to accomplish this task: one to monitor the activity and another to discard frames as follows:

A loss pointer is set at the first user at the beginning. As soon as higher activity is inframed to the pointer, a single frame is discarded.

After discarding the current positioned user frame, the loss pointer jumps to the next active user in the queue and discards a frame and so on, until the required number of frames is discarded.

Frame Loss Statistics for Cyclic Design

A comparison of frame loss statistics for analysis and simulation is shown in Fig. NO. 3. Although for 11 users 3% frame loss is received which is higher that the random technique for the same number of the users. But this scheme is still expected to perform better than the the random schemes. That is due to the fact that no adacent lossed attempted within given parameters.

The DSI advantage of 1.33 as shown in Fig. No. 4 can be achieved at the expense of 3% frame loss. High advantage can also be obtained for additional loss if the quality of speech can be maintained.

Conclusion

The analysis and simulation of proposed multiplexer architecture was carried out and presented. The analysis and simulation resuts indicate that at 3% frame loss the speech quality can still be acceptable to listeners. A gain of 1.33, (DSI advantage) can be achieved. Beyond 3% to loss with the proposed architecture frame-discarding mechanism can not be afforded and further loss causes severe effects on quality.

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Discussion

Among multiplexing techniques, statistical multiplexing is better in performance. The Analysis and simulation graphs as shown in Fig. 3 can become close by running simulation for a longer duration. The expected DSI advantage for monologue speech is 1.25 without anty loss of frames. This is only possible to achieve, when all the gaps are filled in right time with other users talkspurts. Practically this ideal situation can not be attained. Since each user is independently generating silences and the talkspurts and this purely a random phenomenon.

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