

# Intelligent Multiplexer

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**Abstract:** Multiplexing is a way of accommodating many input sources of a low capacity over a high capacity outgoing channel. Statistical Time Division Multiplexing (STDM) is a technique that allows the number of user to be multiplexed over the channel more than the channel can afford. The STDM normally exploits unused time slots by the non-active users and allocates those slots for the active users. In this way STDM normally utilizes channel bandwidth better than traditional Time Division Multiplexing (TDM).

In this paper we are presenting an intelligent architecture of STDM for speech users that does not only exploits silence gaps of user but it also accommodate the high surge of the active users by applying buffering concept. Also in this paper we are proposing some techniques for buffering voice packets when ever the number of active users is more than the maximum users supported on the medium. It is believed that with a limited channel bandwidth quit a high number of users can be accommodated and efficient bandwidth utilization can be achieved for communication channels.

## Introduction

In the history, speech is the most important and conventional medium of communication between human beings. Speech patterns in telephone conversation are characterized by random durations of talk spurts that are followed by silence periods.

Bradey[1] presented experimental measurements of the average duration of speech and silence periods and transition rates between these states from a study of telephone conversations. Typically average speech activity is found to range from 28 to 40% in Dialog communication and it is found 80% in monolog communication. Average length of talk and silence spurts is in the range of 0.4-1.2s and 0.6-1.8s.

Speech communication system normally transmits both silence and talk spurts through a communication channel. Silences are signal parts that may be characterized as noise portion of the signal. However these are to be transmitted in order to keep speech communication smoother. There are systems and algorithms both hardware/software that can easily detect silence out of speech. By detecting silences, the need of their transmission can be avoided, thus enhancing the medium bandwidth efficiency up to some extent. The phenomenon of talk spurts can be modeled by Voice Activity Detector (VAD). Once silences are separated from talk spurts, the two state scenarios are exhibited by

VAD. By using VAD, medium bandwidth utilization enhances to 2.5 times more in dialog and 1.25 times more in monolog communication.

Statistical Time Division Multiplexer can carry voice packets from one end and forward them to other. When we use STDM multiplexer on a voice communicating network, it will transmit silence gaps along with talk spurts thus wasting a considerable amount of bandwidth of medium as shown in Fig 1.



Fig.1

Fig.1 shows there is capacity of carrying maximum of eight users on medium , among these eight 5 users are having talk spurt while remaining 3 are silence thus wasting 3 time slots in this case. A VAD can be kept here for avoiding the transmission of silence gaps between the talk spurts. So for utilizing these free time slots we can accommodate three more stations as shown in Fig.2.

Now medium has a maximum capacity of 8 users but we have connected 11 stations with it. Transmission can work fine until the number of talk spurts on medium is equal to or less then eight.



Fig.2

But the problem arises when more than eight users are having voice packet to transmit as shown in Fig.3.

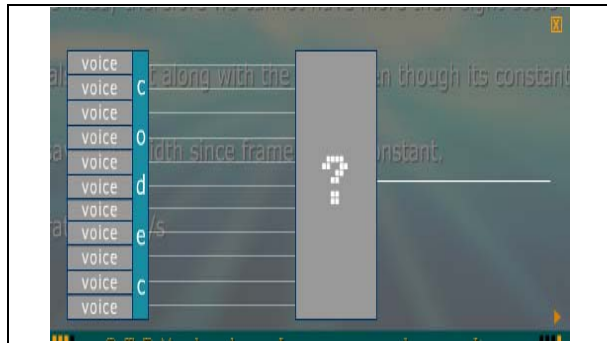


Fig.3

We have to overcome with this problem. This paper is intended to propose some possible solutions to this problem including buffering some packets, dropping some packets and using variable bit rate (VBR). Packet dropping concept and VBR are already discussed [2]. This paper emphasize on the concept of buffering the packet at multiplexer layer when ever there is more traffic than the medium can afford.

**Buffering at Multiplexer:** Statistical Time Division Multiplexer (STDM) can be modeled as a queuing system with finite buffer space. The Statistical Multiplexing Gain (SMG) is an important performance metric that quantifies the multiplexing efficiency. The SMG may be calculated as the ratio of number of variable bit rate sources that can be multiplexed on a fixed capacity link under a specified delay or loss constraint and the number of sources that can be supported on the basis of peak rate allocation[4]. We can improve SMG by buffering some packets coming from different sources, when link capacity is less than the required capacity for transmitting all voice packets simultaneously. In Fig.2 we have seen that we have to accommodate 11 users on a medium with a capacity of supporting maximum of 8 users, so in this case we can buffer 3 extra talk spurts and transmitting remaining 8. Once the 8 talk spurts are completely transmitted, we can send the remaining 3 talk spurts as shown in Fig.4.

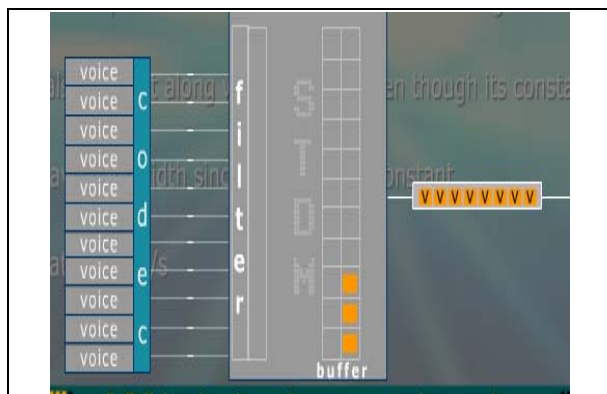


Fig.4

The above discussion shows we can enhance utilization of medium bandwidth by buffering the packet coming from different source when medium capacity is less than the required capacity to transmit all the packets to destination. The question arises, how to perform buffering?

Following three types of buffering are proposed in this paper.

**Random Packet Buffering:** The simplest approach to packet buffering is random packet buffering. During the higher activities, that is, when activity of the users is higher than maximum capacity of the channel, some of the users are randomly selected and their packets are buffered periodically until medium gets enough capacity to transmit these packets. In case of random packets buffering, during momentary overloading consecutive packet buffering can occur, and same user can suffer adjacent packets to be buffered thus facing a long delay which reduces the quality of service.

**Cyclic Packet Buffering:** In this technique we can make sure that packets which are buffered do not belong to same source rather we can pick packet for buffering one from each source starting from first to nth source, where n is number of sources connected with multiplexer.

**Criterion Packet Buffering:** Instead of going through random and cyclic buffering we can set some criteria for the packet that should be buffered. The one way to put criteria for this purpose is to consider the Signal-to-Noise Ratio (SNR). The packets with lower SNR should be considered more eligible for buffering than those which are having high SNR.

### Effects of Buffering on Voice Quality

The Voice Quality (VQ) can be defined as a way of describing and evaluating speech fidelity, intelligibility and characteristics of voice signal itself [5]. Voice Clarity (VC) is also an important parameter in VQ. In context of VQ testing, clarity describes the perceptual fidelity, the clearness and non-distorted nature of the particular voice signal. Clarity can also be described as speech intelligibility, indicating how much information can be extracted from a conversation. However it is possible to understand what is said during voice conversation but still experience poor clarity. For example, voice that is distorted and not easily heard can still be understood. According to our proposed solution i.e. buffering talk spurts can introduce some sort delay in voice communication. Delay does not affect on VQ directly but instead affects the characters of conversations [5]. Below 100ms, most users will not notice the delay. Between 100ms and 300ms users will noticed slight hesitation in their partners response. In this situation conversation seems cold. Interruptions are more frequent and conversation gets out of beak. Beyond 300ms the delay is obvious to users, and they start to

back off to prevent the interruptions. At some point, conversation is virtually impossible. Obviously shorter delay results in better conversation and better perceived over all VQ see Fig.5.

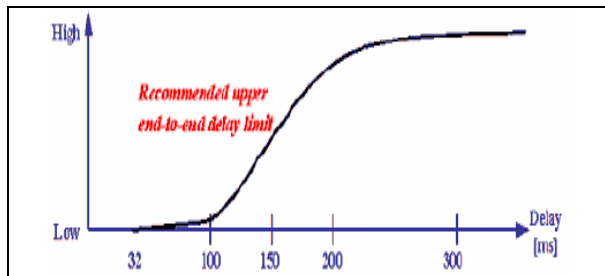


Fig.5

### For what time the talk spurts can be buffered

The VoIP gateways and VoIP terminals contribute significantly to end-to-end delay as result of signal processing at both sending and receiving of link [5]. This processing includes the time required for codec to encode the analog voice signal into the digital signal and decode the digital voice signal back to analog. Some codec also compress the voice signal, there by extracting the redundancy, which further increases the delay due to necessary computation. At transmitting side, packetization delay is another factor. Packetization delay is the time needed to fill a packet with voice data. On the receiver side, voice packet must be delayed to compensate for the variation in packet interval time, known as jitter. Even packets generated with constant spacing in time will generally arrive at receiver with randomly spaced distribution as result of different and queuing times packet experience and varying transmission routes in the IP network. Jitter smoothing using jitter buffers is required because speech codec needs a constant flow of data without gaps.

No matter how well VoIP devices and networks are designed a fundamental delay exists that simply can't be eliminated. Some delay will always be introduced as a result of physical limits of packetization, processing time and propagation time.

Consider an example in which IP packets, each containing 20ms of voice data. It takes 20ms to fill (packetize) the very first packet, assume that codec imposes a further delay of 10ms for framing and computation. A jitter buffer size of at least one frame (20ms) can be expected at the receiving end of the link. Add transmission times, router processing times, and

other miscellaneous sources of delay and one arrives at 60ms. It is clear that 30ms (packetization plus codec framing) is a fundamental lower limit on end-to-end delay in this example, the delay can not be made smaller than this. When we include other sources of delay it becomes the total of 60ms. This is approximate delay that can occur without buffering talk spurts at multiplexing layer. VQ can be reasonable, if packet transmit-receive delay is between 100-300ms. This shows we can buffer a voice packet at multiplexer for approximately 240ms to maintain the VQ at a reasonable level when number of sources are more than the maximum capacity of medium.

**Discussion:** The efficient utilization of medium bandwidth for voice communication can be achieved by avoiding the transmission of silence gaps. More users can be connected on a medium then the maximum number of users supported on that medium. When the number of active users becomes more then the medium supports, the extra packet can be buffered periodically. Buffering can be either random, selective or criterion based.

**Conclusion:** Buffering can be done at multiplexer end, when number of users is more then maximum capacity of medium. This can introduce some delay in transmission. A delay between 100 and 300ms range is acceptable, so if buffering is done without exceeding this delay range, more users can be accommodated on medium without affecting the quality of voice.

### References

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