Active Buffer Management Algorithm for Silence Suppressed Voice Applications

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This paper proposes silence drop first algorithm(SDF) for the active buffer management. This algorithm finds and drops silence packet rather than talkspurt packet in the queue for resolving buffer overflow of queue. This algorithm can serve more simultaneous user while maintain voice service quality with same link capacity. Simulations with voice codec of G.711 and G.729a are performed in this paper. The simulation results shows that SDF can serve more users than drop-tail and drop-head, whose are general active buffer management algorithms for voice application used in commercial and military router. By using proposed SDF algorithm, the voice capacity is increased by 84.21% with G.729a and 38.5% with G.711. Also, SDF algorithm loosens the silence packet inter-arrival time limit to service target number of users.

***Index Terms*—Active Buffer Management, Active Queue Management, Silence Suppression, Voice Communication**

I. Introduction

Technical advance in communication system changes operation concept of military communication because it provides fast and precise information transport to soldiers. In this situation, data communication is increased in the military communication system. However, voice communication is still important in the military communication, because it is simple to the users and it can quickly send emergency information or situations without typing keyboard or manipulate complex device. Also, the operation concept is still in changing so users simultaneously operate previous voice based military network system and next network system. Therefore, voice communication still occupies certain level in military communication.

The target of voice system is providing service to voice users. However, bandwidth or channel resource limit occur by communication equipment, frequency allocation plan, jamming, etc. It cause limitation of number of simultaneous voice users. Therefore, researches for voice system targets to providing more simultaneous users, i.e. voice capacity, with limited bandwidth and channel resource.

Voice communication uses voice codec to encode and decode analog voice signal. The codec periodically generate encoded packet and send it to receiver by network. The silence suppression is introduces to increase network efficiency by not sending in wired network or by sending small packet in wireless network, when user does not speak, because previous voice codecs generate and send packets periodically although user does not speak. Military voice communication requires efficient communication therefore many voice terminal includes silence suppression function.

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Talkspurt packet generally includes main information such as voice or sound that be transmitted, therefore talkspurt packet is more important than silence packet. So, there occurs large difference in the packet size of talkspurt packet size and silence packet size at the application layer. However, in the point of network elements view, the size difference between two types of packet does not large because of packet headers. So, if the silence packet be dropped in front of talkspurt packet, then the system can carry more important information to users with same network performance. Base on this intuition, this paper propose active buffer management(ABM) algorithm for voice application.

The analysis on the talkspurt and silence packet size at network element is described in the chapter 3. This paper proposes new active buffer management algorithm in chapter 4. Performance analysis results are presented in chapter 5, and this paper concludes in chapter 6.

II. Related Works

Commercial router generally uses drop-tail and random early detection(RED) based ABM algorithm[1][2]. Routers for military network are physically stronger than commercial one, for maintain its performance in combat situation. However, ABM algorithm in the military router does not different in commercial router because it is based on the commercial one.

Drop-tail algorithm drops an incoming packet if the packet causes overflow of queue. RED algorithm and its related algorithms are proposed to accelerate congestion control in TCP. It drops packet with probabilities before congestion occurs. However, multimedia applications such as voice or video does not perform congestion control, therefore RED does not perform good performance for voice applications.

The drop-head introduced in [3] to reduce queuing delay by dropping a packet in the head of queue instead of incoming packet. It reduces average queuing delay of queue as traffic intensity increases over 1.0[4].

[5] proposed ABM algorithm for MPEG video. In [5], the algorithm assigns different priority by frame type of packet, and discards less important packets first. The proposed algorithm in our paper is based on intuition on [5]. However, [5] does not give the packet type decision methodology in network element for detecting type of frames.

III. Analysis on the Silence Packet in Lower Layer

Voice traffic can be modeled by a cycle with talkspurt period and silence period. Voice codecs with silence suppression generate talkspurt packet in talkspurt period and silence packet in silence period. Talkspurt packet includes voice or sound information of human speech while silence packet includes signal power level of background noise only. So, silence packet is less important than talkspurt packet. Therefore, sometimes, system does not send silence packet. However user may experience discomfort voice call and can misjudge that call is disconnected without silence packet transmission. Also, the silence packet is used to maintain resource allocation status or session connection in the wireless network, because wireless network release long vacant channel resource for channel efficiency. So, silence packet transmission is recommended in wireless network.

In the application layer, size of silence packet is very smaller than that of talkspurt packet. For example, the talkspurt packet size is 10 bytes while silence packet size is 2 bytes with G.729a codec as shown in Table 1. In this case, size of talkspurt packet is same as 5 times of size of silence packet. So, it can be possible that silence packets can be ignorable in the voice system. However, as header added into the packets at each layer, the ratio between two packet sizes get smaller than that in application layer. For example, talkspurt packet size is 50 bytes and silence packet is 42 bytes in the IP layer as shown in Table 1. Therefore, the silence packets occupy many portion of link capacity. If the network element can distinguish talkspurt packet and silence packet and give priority to talkspurt packet, then performance of network system can increase while maintain user requirements.

Table 1. Packet Size of Voice Codec in each Layer(G.729a)

Layer	Talkspurt(T)	Silence(S)	Ratio(T/S)
Application Layer	10 Bytes	2 Bytes	5
Transport Layer (RTP+UDP)	30 Bytes	22 Bytes	1.36
IP Layer	50 Bytes	42 Bytes	1.19

IV. Proposed Active Buffer Management Algorithm

This paper proposes silence drop first(SDF) algorithm which is an ABM for voice application. SDF algorithm drops silence packets rather than talkspurt packet to schedule more talkspurt packet than conventional ABM algorithms.

```
silence_ptr = -1
queue_size = 0
buffer_size = MAX_BUFFER_SIZE

while(1)
  if(service_completed)
    queue_size = queue_size - 1
    silence_ptr = max(0, silence_ptr - 1)
    continue
  endif
  if(packet_arrived)
    while(insert_packet(arrived_packet) == false)
      if(silence_ptr == buffer_size && queue_size > 0)
        drop_packet(0)
        queue_size = queue_size - 1
        continue
      endif

      drop_packet(silence_ptr)
      queue_size = queue_size - 1
      while(silence_ptr < buffer_size)
        if(silence_packet_check(silence_ptr) == true)
          break
        endif

        silence_ptr = silence_ptr + 1
      endwhile

      queue_size = queue_size + 1
      if(silence_packet_check(packet) == true)
        silence_ptr = min(silence_ptr, queue_size - 1)
      endif
    endwhile
  endif
endwhile
```

Figure 1. Silence drop first algorithm for active buffer management

Figure 1 shows the pseudo code of SDF algorithm. In this algorithm, if arriving packet causes overflows in the queue then algorithm drops silence packet nearest to the head of queue. After, algorithm finds next nearest silence packet for next dropping. The algorithm performs dropping and searching silence packet until the space for arrived packet appears. Then the new packet is inserted into the queue. If there is no silence packet in the queue, algorithm drops packet in the front of the queue, as same as drop-head mechanism. This drop-head mechanism helps to reduce end-to-end delay.

V. Performance Analysis

1) *Simulation Configuration*

The simulation using OPNET is performed for the performance analysis of proposed SDF algorithm. Network architecture is formed as Figure 2. In this simulation, VoIP sessions are generated and send/receive voice traffic through bottleneck link. Voice codecs used in this simulation are G.729a and G. 711, and voice frames per packet is 2. The default inter-arrival time of silence packet is set as 30ms. Link speed of bottleneck link is 10Mbps. This paper use mean opinion score(MOS)[7] as primary performance metric and analyze that how the MOS appears by using other performance metrics.

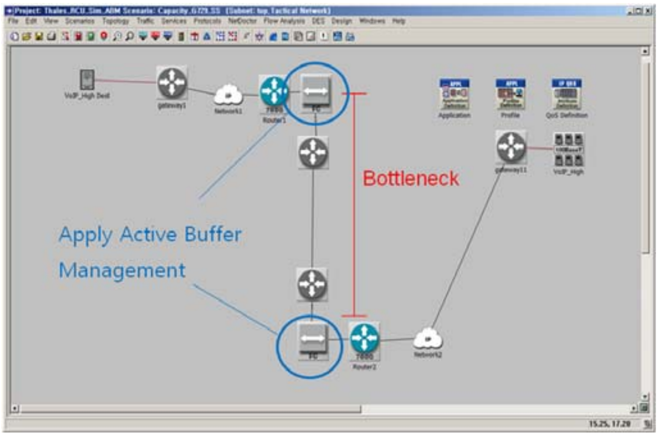

Figure 2. Network architecture for the simulation

2) *Results*

Figure 3 shows the average MOS with G.729a voice codec with respect to the number of users in the system. In the result, SDF algorithm shows better MOS than that of drop-tail and drop-head. If the target MOS is 3.0, then SDF algorithm can service up to 1000 users while drop-tail and drop-head service up to 570 users. In the result, our algorithm can increase capacity by 84.21% by reducing the talkspurt packet loss ratio in the network when target MOS is 3.0.

The difference of MOS comes from the packet loss ratio of talk-spurt packets in network. Figure 4 shows the talk-spurt packet loss ratio in the network with respect to the number of simultaneous users with G.729a voice codec. Drop-tail and drop-head increases talkspurt packet loss ratio when users are larger than 550. SDF has no or very small talk-spurt packet loss ratio in the network. However, MOS decreases as number of users larger than 1000 because packet loss ratio increases as number of users increases. Figure 5 shows the mouth-to-ear delay with G.729a voice codec. As shown in the graph, mouth-to-ear delay increases as number of user increases, and it cause degradation in MOS.

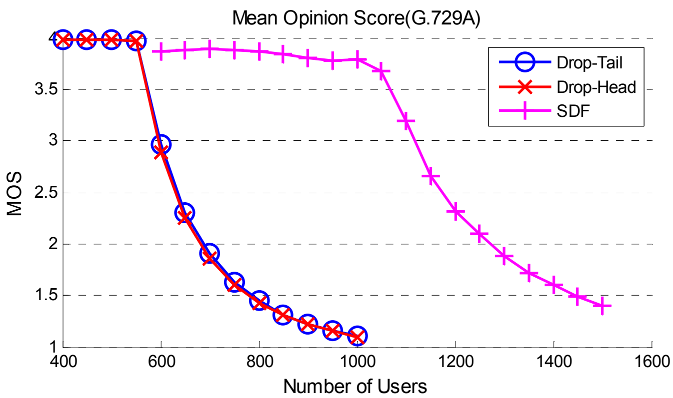

Figure 3. MOS respect to the number of simultaneous users with G.729a voice codec

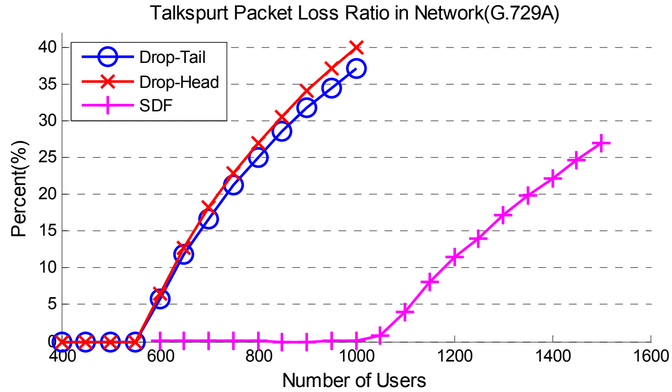

Figure 4. Talk-spurt packet loss ratio in the network respect to the number of simultaneous users with G.729a voice codec

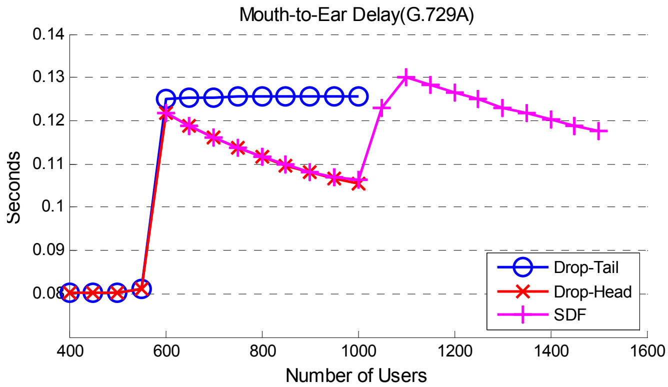

Figure 5. Mouth-to-ear delay respect to the number of simultaneous users with G.729a voice codec

Figure 6 shows the average MOS with G.711 voice codec. In this case, SDF algorithm can service up to 360 users while drop-tail and drop-head service up to 260 users. In the result, our algorithm can increase capacity by 38.5% in this case. This result comes from talkspurt packet loss ratio in the network as same as result with G.729a as shown in Figure 7. Figure 8 shows the mouth-to-ear delay. In the low number of user, SDF shows similar delay pattern with drop-head because it drops packet nearest to the front of buffer. However, delay jumps at 340 users as jitter buffer delay increases, because codec increases jitter buffer size to reduce packet loss ratio caused by jitter buffer as packet loss ratio increases. Still, main reason of performance difference is network packet loss ratio.

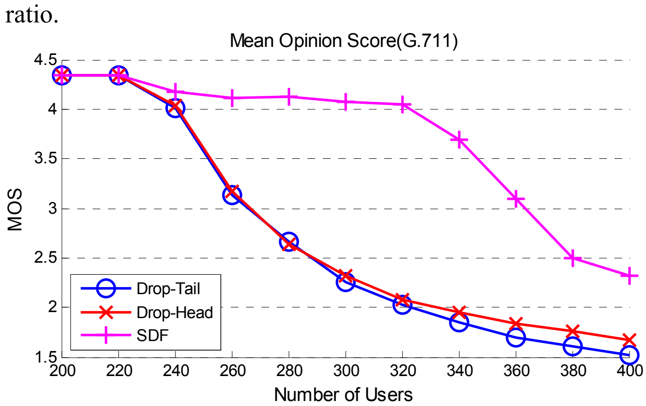

Figure 6. MOS respect to the number of simultaneous users with G.711 voice codec

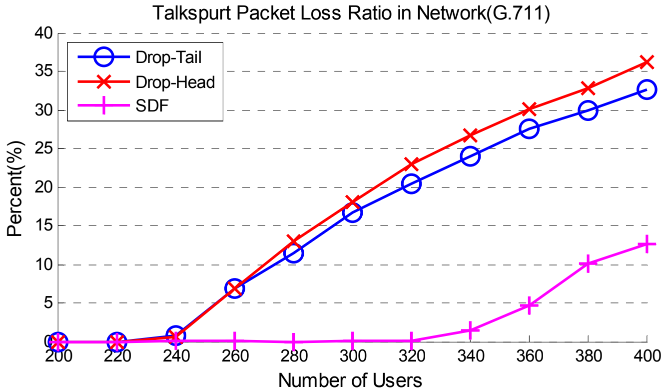

Figure 7. Talk-spurt packet loss ratio in the network respect to the number of simultaneous users with G.711 voice codec

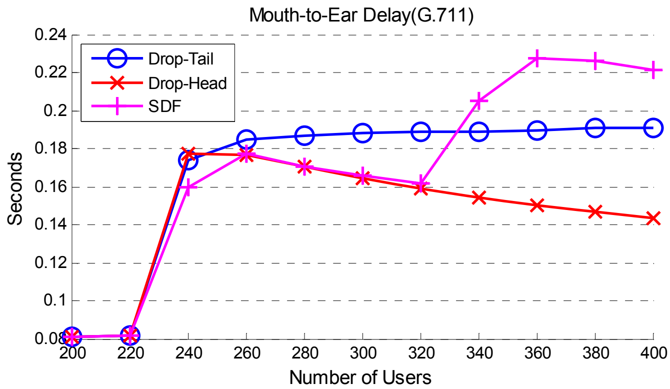

Figure 8. Mouth-to-ear delay respect to the number of simultaneous users with G.711 voice codec

Silence packet inter-arrival time(SPIT) can change the performance of voice system. The rate of silence packet generation decreases when SPIT increases. So, the system can provide more simultaneous users by increasing SPIT. However, some system requires small SPIT so maintain resource allocation or session status. Therefore, SPIT is one of important parameter in the voice system.

Figure 9 shows MOS with respect to the SPIT with G.729a codec. Figure 9 does not include MOS with drop-head because it has similar result with drop-tail. In the graph, drop-tail and SDF can provide voice service with MOS of 3.0 or more when SPIT is larger than 0.06 and simultaneous users is 800. Also, SDF can provide service with SPIT of 0.04 seconds. However, when number of users becomes 1000, then drop-tail cannot provide service with MOS of 3.0 within 0.16 seconds of SPIT although SDF can service with SPIT of 0.04 seconds or more.

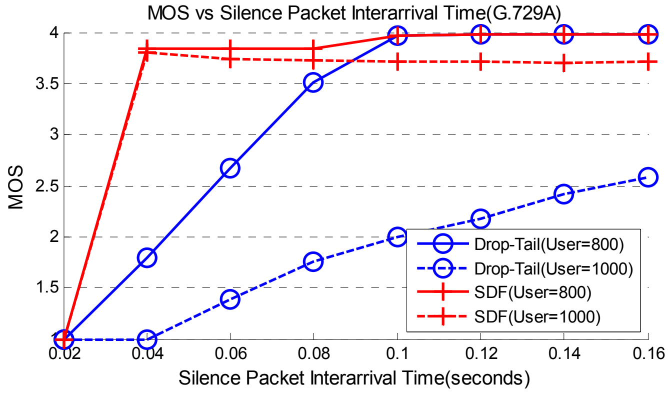

Figure 9. MOS respect to the silence packet inter-arrival time with G.729a codec

VI. Conclusion

This paper proposes silence drop first algorithm for the active buffer management. This algorithm finds and drops silence packet rather than talkspurt packet therefore it can service more simultaneous user while maintain same service quality. Simulation with G.711 and G.729a is performed in this paper and the results shows that SDF can service more users then drop-tail and drop-head. The voice capacity is increased by 84.21% with G.729a and 38.46% with G.711.

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