A Burst Error Reduction Algorithm for VoIP Service in Wireless LAN Network

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Abstract

Abstract In this paper, we propose the burst error reduction (BER) algorithm for VoIP service in the wireless LAN network. In end point device, this BER algorithm can be achieved packet loss bounded QoS provisioning using interleaving in buffering and FEC (Forward Error Correction) through transmitting voice packet. BER algorithm can reduced the voice packet loss rate 5.5% - 60% in VoIP network using wireless LAN.

Index Terms: Voice over IP, Interleaver, wireless LAN

I. INTRODUCTION

In recent years, IEEE802.11 wireless LAN gains in public favor, because not only the price of its equipment has been dropped but also the device for wireless LAN is easy to install and maintain. Wireless LAN is used to data transmission such as e-mail, file transfer, however, it is not used to voice packet transmission frequently. Since the insertion of a wireless link generates distinctly new impairments like fading, noise and corruption and loss of packet at the MAC (Media Access Control) layer. In wireless LAN of transmission rate of 11Mbps and approximate range of 300feet, if a lot of user communicate with each other by using VoIP of bit rate of 8kbps, the network is occurred burst error that a lot of packet suddenly come together in network (in case of using G.729) [1], [9].

In voice network, generally people endure a delay between 150ms and 400ms [1]. But, they do not allow a lot of voice packet loss. So, it has been a lot of effort to resolve a problem of the voice packet loss and QoS of VoIP service. There are Intserv (Integrated services) and Diffserv (Differentiated services). Intserv defines an enhanced service model for expanding Internet services to better meet the needs of diverse applications such as the transport of audio, video, real-time, and classical data traffic within a single network infrastructure. But it provides more powerful service but has serious limitations with respect to network scalability and robustness. To resolve Intserv problem, Diffserv is introduced. It provides scalable service differentiated in the Internet that can be used to permit differentiated pricing of Internet service. It is based on a model where traffic entering a network is classified, possibly conditioned at the boundaries of the network, and assigned to different service classes. But it cannot provide services that are comparable to Intserv. But both methods don't provide the solution of burst error in wireless VoIP network.

For the purpose of these objectives, we propose the packet loss-reducing algorithm in case of burst error in VoIP network over wireless LAN using Interleaving and FEC.

The configuration of this paper is as follows. In section II, the network architecture for VoIP service in wireless LAN. In section III, we explain Interleaving, FEC and design the loss-reducing algorithm. Next, in section IV, we show simulation environment and result using OPNET. Finally, our conclusion and a summary are presented.

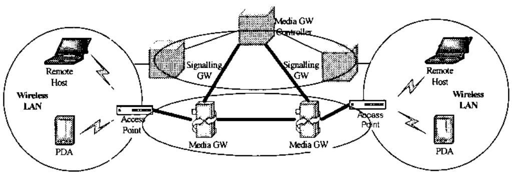
II. NETWORK ARCHITECTURE FOR VOIP SERVICE IN WIRELESS LAN

Recently an emerging trend for implementing VoIP- generally connecting computing devices - is in wireless networks. A wireless LAN is a data system designed to provide location-independent network access between computing devices by using radio waves rather than a cable infrastructure. Wireless LANs give users wireless access to the full resources and services of the LAN across a building or campus environment.

But there are some fundamental problems that wireless LAN is introduced. These problems include a higher frequency of packet losses, larger latency, and more jitter than wire system [1].

- Packet loss: Packets will be dropped under peak loads and during periods of congestion in VoIP network using wireless LAN.
- Latency: This is the time delay incurred in speech by the IP telephony system.
- Jitter: The data will arrive at very inconsistent rates because IP networks cannot guarantee the delivery time of data packets or their order. The variation in inter-packet arrival rate is jitter.

But we consider only packet loss not delay in wireless LAN. Packet losses are caused by the properties of the wireless link. Therefore, the voice packets can be discarded when routers or gateways are congested in VoIP network. Furthermore, considering the backward-adaptive coding schemes of the G.723.1 and G.729 source coders, packet loss results in loss of synchronization between the encoder and the decoder.



[Fig. 1] The existing VoIP network in wireless LAN

[Fig. 1] shows the existing VoIP network in wireless LAN. VoIP network in wireless LAN is consist of remote host (notebook and PDA), access point (or wireless hub), wireless LAN (IEEE802.11x) and existing VoIP network (gateway and its controller). But this network provides function of only wireless connecting computing devices.

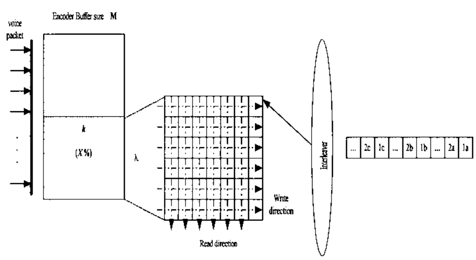
In this network the 802.11x standards focus on the first two layers of the OSI/RM, namely, the Physical Layer and Data Link Layer [2]. And its transmission rate is 11Mbps and approximate range is 300feet. So any wireless LAN application (including VoIP service) of the IEEE 802.11x can run on wireless LAN as easily as they run over Ethernet. But the voice packet loss is happened when that packet is delivered in wireless LAN network. Because wireless link is unstable compared with wired link.

Ⅲ. BURST ERROR REDUCTION (BER) ALGORITHM

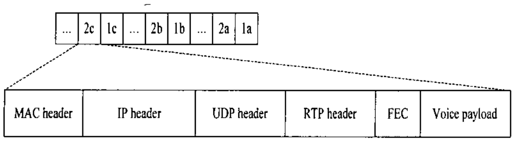
A. Interleaving Mechanism

An Interleaver is a hardware device that takes symbols from a fixed alphabet as the input and produces the identical symbols ant the output in a different temporal order. It is commonly used in conjunction with error correcting codes to counteract the effect of burst errors. Interleaving is a processing in the Interleaver. Namely, Interleaving is a standard signal processing technique used in a variety of communication systems. And this Interleaving is implemented with FEC (Forward Error Correction) that employs error-correcting codes to combat bit errors by redundancy to information packets before they are transmitted in Interleaver. Because Interleaving is to disperse sequences of bits in a bit stream so as to minimize the effect of burst errors introduced in transmission, Interleaving is used with FEC [3].

This paper uses block Interleaver of two kinds of classical Interlearers, which are a block and convolutional Interleaver. In this block Interleaver, the input data is written along the rows of a memory configured as a matrix, and then read out along the column. Therefore, in wireless VoIP network, Interleaver is installed in end point (or end user's device) devices and then each endpoint device executes interleaving when voice packet is transmitted [5].



[Fig. 2] Interleaving operation in wireless VoIP network



[Fig. 3] VoIP packet format in Interleaving

[Fig. 2] shows interleaving operation in end point device of VoIP network in wireless LAN. In Interleaver of the buffer of M size, it writes transmitted voice packet along the rows of a memory configured as a matrix of size k, and then read out along the column. Then, in Interleaver of receiver, it writes and reads this transmitted voice packet into different direction with Interleaver of sender [3], [4]. Then Interleaver forwards this voice packet with FEC to receiver.

[Fig. 3] shows a transmitted VoIP packet format after interleaving in Interleaver of sender buffer. This packet header consists of MAC header, IP header, UDP header, RTP header and FEC field with voice payload field. This paper uses MAC (Media Access Control) protocol because of using IEEE 802.11 wireless LAN.

B. BER Algorithm

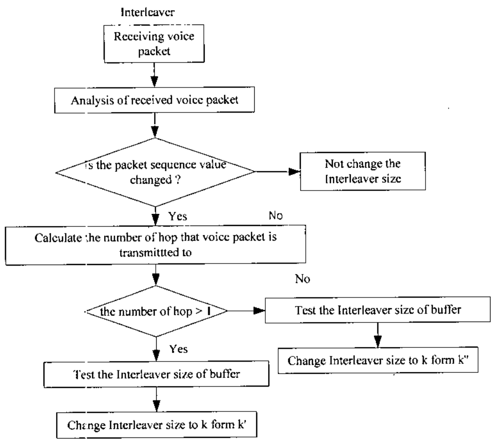
It will explain the algorithm to manage Interleaver size. During transmitting the voice packet, the situation of wireless VoIP network is altered. Because a lot of voice packets joined in wireless VoIP network make busy the device of wireless VoIP network such as gateway, router, media gateway controller, etc. In this case, burst error is occurred, and then this is cause of voice packet loss. Therefore the control of this burst error is needed.

Fig. 4 shows the burst error reduction algorithm for the control in this situation. It uses the Interleaver size change in wireless VoIP network using information of voice packet header.

Each sender and receiver receives the voice packet from each other. Then each of receiver and sender analyzes the received voice packet header. And it finds whether the sequence number of RTP is changed. If the sequence number is changed, then Interleaver calculates the number of hop that voice packet is passed. If the sequence number is not changed, then Interleaver doesn't change its size.

After calculate the number of hop, if the number of hop is larger than N (some value decided by Interleaver), then Interleaver examines the size of Interleaver of the buffer. Then Interleaver changes its size to k' form k. If the number of hop is smaller than N, then the Interleaver change its size to k'' form k.

This modified Interleaver is applied to following transmitting of voice packet. This algorithm has an advantage that it is adapted to various wireless network environments.



[Fig. 4] Algorithm for burst error control by changing Interleaver size

Ⅳ. SIMULATION RESULTS

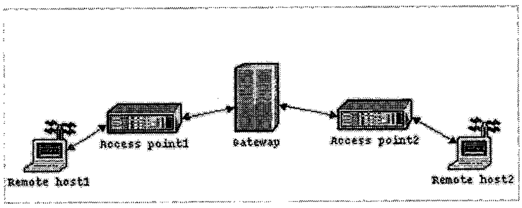
This paper analyzes the performance based VoIP network using wireless LAN configuration of Fig. 1 and simulation results by comparing with the packet loss rate of using BER algorithm and not using BER algorithm as changing the buffer size. But this performance analysis is considered about only voice packet loss not delay.

A. Simulation environment

In this paper, the scenario is considered which consists of a remote host in an IEEE802.11 wireless LAN. Fig. 5 shows this scenario implemented by OPNET tool. The remote host transmits speech packets through the wireless link to the base station that is connected to an Ethernet network. Through the base station and the Ethernet, which they are passed by access point, media gateway, etc., the speech packets reach another remote host. This simulation tool is OPNET [10].

Simulation environment is configured as real VoIP network using wireless LAN like figure 6. In this simulation CODEC is G.729 and G.723.1. At first G.729 is used to support the wireless LAN with bit rate of 8kbps and 10byte frame length. G.723.1 is used to general voice packet compression with bit rate of 5.3-6.3kbps.

And also the simulation time is 10min and the packets are transmitted over a 1Mbps, and the frame size is 10ms. And then this simulation is based on one way traveling delay and it has 125ms long and variable Interleaver size (=buffer size). Input parameter is buffer size and arrival of Poisson distribution, and output parameter is the number of voice packet loss rate.



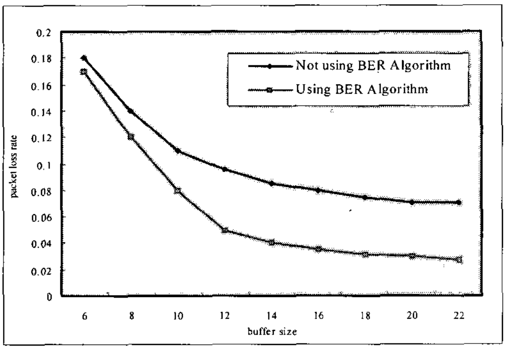
[Fig. 5] Simulation environment of OPNET

B. Measurement Results

The goal of our simulation is to check the voice packet loss rate according to using BER algorithm or not.

The following measurement results present the cases of high error rate that occur on the outer limit of the transmission range or during movement of the mobile terminal. Most of the time we noticed a good wireless channel or no connection could be established. The following plots are based on 300 measurements, each 125ms long and variable Interleaver size.

The two analyzed algorithms were:



[Fig. 6] Packet loss / buffer size

- Voice packet transmission using BER algorithm with Interleaving
- Voice packet transmission not using BER algorithm

[Fig. 6] shows the result of performance analysis of voice packet transmission of VoIP network in wireless LAN. As see this figure, the packet loss rate of using BER algorithm with Interleaving is smaller than not using BER algorithm. In case of not using BER algorithm, the packet loss rate is between 18% and 7%. Otherwise, in case of using BER algorithm with Interleaving, packet loss rate is between 17% and 2.8%. Therefore in the case of using BER algorithm with Interleaving, the packet loss rate can be reduced 5.5% - 60% comparing with not using BER algorithm in wireless VoIP network.

Therefore this algorithm can improve the quality of voice in wireless LAN environment through reducing packet loss by using BER algorithms with Interleaving.

V. CONCLUSION

This paper has started to analyze potential quality improvements of voice in wireless LAN. The experimental results show that it is possible to increase the voice quality in the case of high packet error rates. That is, this paper can reduce voice packet loss rate 5.5% - 60% by using Interleaving and burst error control algorithms in VoIP network over wireless LAN.

Therefore this paper shows to provide QoS for voice packet in VoIP network over wireless LAN by using BER algorithm with Interleaving though wireless link is instable.

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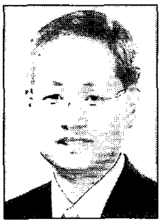

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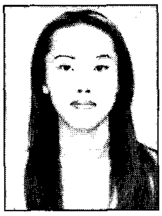

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