Evaluating the performance of voice communication in Mobile Adhoc Wireless Networks to achieve perceived voice quality using Coding techniques

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ABSTRACT

The current growth of speedy voice transmission over Mobile Adhoc wireless networks has enabled the deliverance of multimedia broadcasting services to mobile users. They play a vital role in business environments where permanent access to network resources act as key factor.. This document is a survey of the voice communication in wireless networks. The various issues related to implementation of voice communication over the network such as reduced control overhead, minimum ete delay, reduced FLR, QOS, perceived voice quality are taken as a key factor. The scenario of using Automatic Repeat reQuest ***(ARQ)*** retransmission for two-way low-bit-rate voice communications over wireless Adhoc Network. Low delay constraint may require that a corrupted retransmitted packet not be retransmitted again, thus there will be packet-errors at the decoder which results in voice quality degradation. In this report, we illustrate performance results relative to packetization scheme, coding schemes are discussed. In our study we analyze the service of Layered Coding (LC) and Multiple description Coding (MD) for supporting error resilient voice communication in ad hoc wireless networks. Simulation results show that our proposed scheme can effectively reduce the number of packet loss ,reduces end to end delay and achieves QoS.

Keywords

Adhoc Networks, end-to-end delay, frame loss rate, forward error correction, layered coding, multiple description coding, voice quality, multi path.

1. INTRODUCTION

Wireless Adhoc Networks are formed by group of mobile nodes, which transfers packets for efficient communication between the nodes. Wireless Adhoc Networks, typically becoming increasingly popular, but even further elevates the challenges of delay and loss reduction. Degradation of speech quality caused by packet delay and loss of voice traffic is still one of critical technical barrier of the Voice communication system. Furthermore, apart from these limitations AWNs will need to support a large number of concurrent Voice communications since VoIP is spreading rapidly in public spaces. These motivations led us to study the packet loss resilience scheme for supporting voice communication over Adhoc Wireless Networks. The FEC method emphasis to investigate the voice data packets to achieve reduced packet loss rate, to increases the throughput and maintaining an overall good quality. Since these networks can be deployed rapidly and flexibly, they are attractive for numerous potential applications, ranging from emergency and rescue operations to real-time multimedia communications for disaster areas. Real-time multimedia applications can tolerate packet losses to some extent but are highly delay sensitive. In this paper, we discuss mainly on providing voice support over AWNs, because it (voice over an AWN) is potentially a key application in many current and proposed scenarios. When transmitting voice data, continuous delivery with limited packet loss rate is of primary importance than trying to achieve zero packet loss rate by employing retransmission based strategies.

The unique characteristics of the voice communication, such as small payload size (typically 20 bytes) and timely arrival of the packets at the destination, make it very challenging to deploy over AWNs. In wireless networks, in addition to packetization overheads (i.e., header information of the medium access control (MAC) and other higher layers as in wire-line networks), each packet should also include a preamble for synchronization, which typically occurs on a packet-by-packet basis. Also, after receipt of each packet, the receiver needs to send an acknowledgment (ACK) to indicate a successful reception, which further increases the control overhead. Thus, the performance of such networks is poor for small data payloads common in voice communication. In a single-hop network, every node is within the radio range of every other node, and hence, the end-to-end delay is not too large. However, in the multihop scenario, the source and destination may be several hops away and packets need to be forwarded by the intermediate nodes, causing delays at each hop. As a result, the end-to-end delay can become quite large, especially due to packets of a multihop flow contending with each other for the shared channel at successive hops, thereby making voice communication very challenging.

2. LITERATURE REVIEW

The authors of paper [1] proposed the concept of multiple descriptions coding. Multiple descriptions coding (MDC) is a coding technique that fragments a single media stream into n sub-streams $(n \geq 2)$ referred to as descriptions. The packets of each description are routed over multiple, (partially) disjoint paths. In order to decode the media stream, any description can be used; however, the quality improves with the number of descriptions received in parallel. MDC is a form of data partitioning, thus comparable to layered coding as it is used in MPEG-2 and MPEG-4. Yet, in contrast to MDC, layered coding mechanisms generate a base layer and n enhancement layers. This paper has introduced a method for generalized MD coding using correlating transforms. The framework is a generalization of the method proposed by Orchard, Wang, Vaishampayan, and Reibman. This method is very effective in increasing robustness with a small amount of redundancy. The MD scenario provides a good analogy to communication over a lossy packet network. For this reason, "description," "channel," and "packet" have been used interchangeably.

In this paper [2] the authors have proposed about Wireless lane technology. In recent days use of mobile computers, and so many digital devices are emerged which require wireless lane for access internet and sharing files which can be only possible thorough wireless lane or Wi-Fi. The purpose of wireless lane is the same as that of wired or optical Lane: to convey packets of information among the device attached to the LAN through which we can transmit the data. In radio system the choice of transmission technology determines many of the important performance parameter of the system. In this paper we have discussed about various alternatives in wireless LANE design: media choice operating frequency, network topology, operating mode and access method.

In paper [3] the authors discuss about the Multiple description (MD) coders provide important error resilience properties. Specifically, MD coders are designed to provide good performance when the loss is limited to a single description, but it is not known in advance which description. , we combined MD video coding with a path diversity transmission system for packet networks such as the Internet, where different descriptions are explicitly transmitted through different network paths, to improve the effectiveness of MD coding over a packet network by increasing the likelihood that the loss probabilities for each description are independent. The available bandwidth in each path may be similar or different, resulting in the requirement of balanced or unbalanced operation, where the bit rate of each description may differ based on the available bandwidth along its path. We design a MD video communication system that is effective in both balanced and unbalanced operation. Specifically, unbalanced MD streams are created by carefully adjusting the frame rate of each description, thereby achieving unbalanced rates of almost 2:1 while preserving MD's effectiveness and error recovery capability.

In paper [4] and [5] Coding scheme for data transmission over erasure channels which also known as multiple description is coding. The LMMSE prefilter method of Romano is reviewed and generalized to allow three different operational modes of the prefilter. They include the possibility to decrease or increase the number of descriptions to be transmitted. We derive explicitly the Hessian matrix for an efficient calculation of the prefilter. We also study the properties of the distortion measure theoretically. Multiple description coding (MDC) is often linked with a packet oriented transmission scheme like the internet. In the internet, some packets (i.e. descriptions) might get lost. This may e.g. be the case for an internet-router that is congested and its buffers overflow. The problemat the receiver is now to obtain an estimate of the original information from the subset of available packets. But multiple description coding also appears to be a valid tool for an incremental specification of signals. In the Collaborative Research Center SFB 732, methods for incremental specification of speech are investigated. One feature that one would expect from such an incremental scheme is that subsets of different descriptions of the speech signal can be arbitrarily chosen and help to restore a better representation of the original speech sample. A good overview of current techniques for multiple description coding . In this paper we investigate a correlating transform which was introduced in the inspiring paper. After a short introduction into the MDC problem, we will further generalize in section 3. This generalization allows us to handle the case that more descriptions are used after the correlating transform. Thus, our scheme provides the possibility of a redundancy coding. This new operational mode of a MDC correlating transform is very interesting in the case that the channel offers enough bandwidth. It allows us to further minimize the distortion by transmitting redundant descriptions.

Paper [6] discuss about the analysis of the impact of using media–dependent Forward Error Correction (FEC) in VoIP flows over the Internet. This error correction mechanism consists of piggy-backing a compressed copy of the contents of packet n in packet $n + i$ (i being variable), so as to mitigate the effect of network losses on the quality of the conversation. To evaluate the impact of this technique on the perceived quality, we propose a simple network model, and study different scenarios to see how the increase in load produced by FEC affects the network state. We then use a *pseudo–subjective* quality evaluation tool that we have recently developed in order to assess the effects of FEC and the affected network conditions on the quality as perceived by the end–user. In recent years, the growth of the Internet has spawned a whole new generation of networked applications, such as VoIP, videoconferencing, video on demand, music streaming, etc. which have very specific, and stringent, requirements in terms of network QoS. In this paper we will focus on VoIP technology, which has some particularities with respect to other real–time applications, and it is one of the most widely deployed to date. The current Internet infrastructure was not designed with these kinds of applications in mind, so multimedia applications' quality is very dependent on the capacity, load and topology of the networks involved, as QoS provisioning mechanisms are not widely deployed. Therefore, it becomes necessary to develop mechanisms which allow to overcome the technical deficiencies presented by current networks when dealing with real–time applications. Voice–over–IP applications tend to be sensitive to packet losses and end– to–end delay and jitter. In this paper we will concentrate on the effect of FEC on packet loss, and the effect of both on the perceived quality. The effects of FEC on interactive (two–way) VoIP applications are the subject of future studies.

Paper [7] explains the error checking in transmission of signals in wireless network.In recent year's use of mobile and palm tops has emerged. Many of the people use mobile devices, personal digital assistance and portable computers and all that require wireless technology. So in this paper we have proposed about media access protocol for single channel wireless LAN. Using packet-level simulation, we examine various performance and design issues in such protocol. Our main aim is to develop a media access protocol for the use in wireless network infrastructure. While our specific simulation result can be applied PARC's Particular Radio Technology. Our purpose is to re-construct the basic design of MACA and then produced the revised version of them for use in PARC's wireless LAN. The MACA algorithm is proposed in this paper. With this algorithm any station hearing an RTS will defer long enough so that the transmission station can receive the returning CTS. The increasing of use of mobile and computer devices is played important role in our telecommunication. These changes have improved the performance of the media access protocol.

The alternatively ARQ (Automatic Repeat reQuest) can be choosed but, which requires feedback and also for multicasting network it results in worst case but in FEC this can be avoided and it handles multicasting too.

For example, (n , k) = (2 , 3) FEC code and the transmitting two numbers are *a* and *b* and three packets are sended to the network as, packet one as a, packet two as b and packet three as a+b and could be represented as matrix multiplication and can be encoded as

$$\begin{pmatrix} A_1 \\ A_2 \\ A_3 \end{pmatrix} = \begin{bmatrix} 1 & 0 \\ 0 & 1 \\ 1 & 1 \end{bmatrix} \begin{pmatrix} x \\ y \end{pmatrix}$$

The authors in [8] compare MD coding and layered coding (LC) when used with path diversity and observe that MD coding is preferable when the paths are symmetric, while in [9] the authors conclude that LC does better only when the base layer can be transmitted error free or with very low error rates. Path diversity has been shown to have significant benefits over conventional single path transmission in terms of reduced packet loss rate and improved video quality for wireless video transmission in [10].

3. FORWARD ERROR CORRECTION (FEC)

Forward error correction (FEC) or **channel coding** is a technique used for controlling errors in data transmission over unreliable or noisy communication channels. The central idea is the sender encodes their message in a redundant way by using an **error-correcting code** (ECC).

The redundancy allows the receiver to detect a limited number of errors that may occur anywhere in the message, and often to correct these errors without retransmission. FEC gives the receiver the ability to correct errors without needing a reverse channel to request retransmission of data, but at the cost of a fixed, higher forward channel bandwidth. FEC is therefore applied in situations where retransmissions are costly or impossible, such as when broadcasting to multiple receivers in multicast. FEC information is usually added to mass storage devices to enable recovery of corrupted data.

FEC processing in a receiver may be applied to a digital bit stream or in the demodulation of a digitally modulated carrier. Many FEC coders can also generate a bit-error rate (BER) signal which can be used as feedback to fine-tune the analog receiving electronics.

The maximum fractions of errors or of missing bits that can be corrected is determined by the design of the FEC code, so different forward error correcting codes are suitable for different conditions.

Voice communications over IEEE 802.11 based WMNs is challenging because the 802.11 standard is designed primarily for non-real time transfer of data. IEEE 802.11 MAC protocols are designed to minimize collisions and depend on retransmissions to ensure successful transmission, irrespective of the delay incurred by the packet or the number of voice calls supported. Interactive voice communications cannot tolerate large delays (one way end-to-end delay, as specified by the ITU-T recommendation G.114, should be under 200 ms for users to be "very satisfied") and in multi-hop communications, delays occur at each node due to MAC, physical, and network layer protocols. Further, voice quality is affected by packet losses due to bit errors in the wireless channel. The mesh network architecture provides increased robustness by allowing transmission over multiple paths. We look at this key feature of WMNs and investigate multiple description coding as a solution to use multiple paths efficiently.

4. MANETS

The integration of different wireless access technologies combined with the huge characteristic diversity of supported services in next-generation wireless systems creates a real heterogeneous network. The additional benefits of the WLAN interface are, however, likely to be outweighed by its greater rate of energy consumption. This is especially of concern when real-time applications that result in continuous traffic are involved. WLAN radios typically conserve energy by staying in sleep mode. With real-time applications like voice over Internet Protocol (VoIP), this can be challenging since packets delayed above a threshold are lost. Moreover, the continuous nature of traffic makes it difficult for the radio to stay in the lower power sleep mode enough to reduce energy consumption significantly. In this work, we propose the GreenCall algorithm to derive sleep/wake-up schedules for the WLAN radio to save energy during VoIP calls while ensuring that application quality is preserved within acceptable levels of users. VoIP is one of the traditional application scenarios for Mobile Ad Hoc Networks (MANETs) in settings such as emergency response. Ideally, VoIP would be transparent to The network type such that users would not have to worry whether their machine is currently part of a MANET or attached to the Internet.

4.1 Averaging noise to reduce errors

Most telecommunication systems used a fixed channel code designed to tolerate the expected worst-case bit error rate, and then fail to work at all if the bit error rate is ever worse. However, some systems adapt to the given channel error conditions: hybrid automatic repeat-request uses a fixed FEC method as long as the FEC can handle the error rate, then switches to ARQ when the error rate gets too high; adaptive modulation and coding uses a variety of FEC rates, adding more error-correction bits per packet when there are higher error rates in the channel, or taking them out when they are not needed.

By adopting FEC method, it sends k packets in MANET, Reconstruct n packets, Such that we can tolerate k-n losses Called an (n, k) FEC code.

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4.2 Voice Communication

Some of the earlier works have addressed the issue of supporting voice communication over AWNs. These works focus on optimizing packet length, employing forward error control within a packet, reservation policies, bandwidth reuse technique, and retransmission strategies. During voice communication, if the number of lost voice packets is higher than that tolerated by the listener, then either an error control or loss recovery mechanism is required . Typical mechanisms fall in one of the two classes, viz., closed-loop mechanisms and open-loop mechanisms.

4.3 Automatic Repeat Request (ARQ)

Automatic Repeat Request (ARQ) mechanisms are closed-loop mechanisms where the source retransmits lost packets as reported by the destination. ARQ mechanisms are typically not acceptable for interactive speech communication because they increase the end-to-end delay, and thus, the packets might miss the deadline. FEC mechanisms are open-loop mechanisms, where redundant data is transmitted along with the original data so that the lost original data can be recovered from the redundant data. FEC mechanisms can be further classified into two categories: media independent and media specific. In media-independent FEC-based methods, the redundant information can be sent in the form of parity packets.

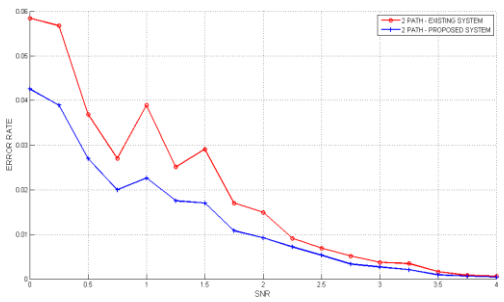

Figure: 1 snr Vs error rate in 2 path

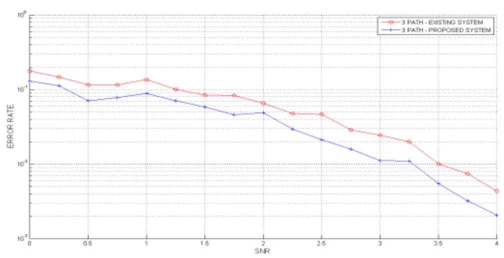

Figure: 2 snr Vs error rate in 3 path

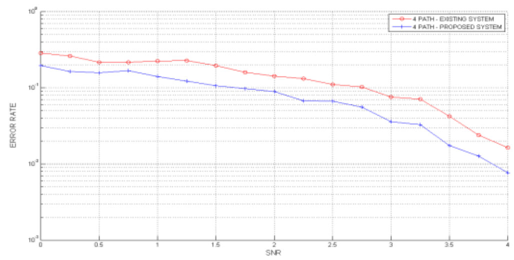

Figure: 3 snr Vs error rate in 4 path

5. CONCLUSION

The effective packetization scheme was achieved using Layered coding and MD coding by adapting FEC mechanism as shown in Figure 1,2 and 3 respectively. . FEC allows error correction without retransmission. A potentially promising approach to reduce the voice packet loss rate is to establish multiple paths between the source and destination of a session and to use speech coding schemes that take advantage of the existence of multiple paths. One such coding scheme is MD coding , in which a voice stream is encoded into multiple independent substreams. These descriptions can be decoded independently to produce a voice stream of basic quality.

It very much supports for multicating. MD coding does not require prioritized transmission as all descriptions have equal importance. When more descriptions are received, the decoder can gradually increase the quality. Since the probability of losing all descriptions is relatively low, it performs better than LC at higher packet loss rates. The combination of inter packet redundancy, MD coding, and path diversity was used to provide speech support over AWNs.

By applying MD coding on the whole voice stream, they obtained two descriptions, respectively, transmitted over two paths set up by the routing module. The mechanisms proposed in our discussions can indeed provide better voice quality and reduced packet loss rate than other related approaches for real-time voice.

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