Development and Performance Analysis of a Lossless Data Reduction Algorithm for VoIP

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ABSTRACT

VoIP (Voice Over IP) is becoming an alternative way of voice communications over the Internet. To better utilize voice call bandwidth, some standard compression algorithms are applied in VoIP systems. However, these algorithms affect the voice quality with high compression ratios. This paper presents a lossless data reduction technique to improve VoIP data transfer rate over the IP network. The proposed algorithm exploits the data redundancies in digitized VFs (Voice Frames) generated by VoIP systems. Performance of proposed data reduction algorithm has been presented in terms of compression ratio. The proposed algorithm will help retain the voice quality along with the improvement in VoIP data transfer rates.

Key Words: **VoIP, Data Reduction Algorithms, QoV, QoS, Compression Ratio.**

1. INTRODUCTION

The Internet is a communication network which carries all types of traffic, including voice, video, and data. In recent time, the Internet is also used as an alternative media to carry voice data compared to traditional PSTN (Public Switched Telephone Network) [1-2]. The technology for transferring voice data is called VoIP. Due to the low costs involved in IP connectivity, the IP telephony comparatively carries low billing rates especially for the long distances [3].

In VoIP technology the voice signals are transmitted over the Internet after converting them into IP packets. This process begins with the sampling of the voice signals at the transmitting end of VoIP system. The sampling rate should be reasonably high so that the original voice signal may be recovered up to the maximum accuracy. Typically, the voice signals are sampled at the rate of 8000 samples per second. Each sampled voice signal is assigned an 8 bit binary code which results into a 64 kb/s bit stream. After applying some digital filtering algorithms, the bit stream is compressed by employing any of the compression standards such as G729 which compresses 64-8kb/s. This process forms VFs of 80 bits long of 10 msec. In order to transfer the sampled voice data, each VF is converted into an IP packet. In the first step, a 12-byte long RTP (Real Time Protocol) header is appended to 80-bit long VF. Next, 8 bytes long UDP (User Datagram Protocol) is added to this data. Finally, a 20 bytes IP header is added to the data to generate the complete IP packet. The VF in the form of IP packet is then sent to the Internet. At the receiving end of the VoIP system, the voice signal is recovered from the

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received IP packet by removing the IP, UDP and RTP headers, respectively. The process of voice-to-IP conversion is summarized in Fig. 1 at the receiving end.

To make Internet as dependable alternate to the traditional PSTN, VoIP system must provide high-quality. VoIP is facing some issues such as: security, global administration, billing, lawful interception, voice quality maintenance and proper bandwidth utilization. Objective of this paper is to present a loss less data reduction algorithm to address the bandwidth utilization issue in VoIP. The proposed algorithm exploits the data redundancies in digitized VF generated by VoIP system. The proposed algorithm may be used as an alternative to the contemporary off-the-shelf data compression algorithm as they degrade the voice quality for high compression ratios. The remaining paper is organized as follows: Section 2 reviews the related work pertaining to the compression issues related to the VoIP. Section 3 presents details of the data reduction algorithm with short VF and improved data reduction algorithm with standard VF size. Section 4 presents the performance analysis of the improved data reduction algorithm based upon experimental results. Finally, the conclusion is made in Section 5.

2. RELATED WORK

To save the bandwidth in VoIP, the compression algorithms have been considered for IP header [4-7]. Mate and Rinne [4] have suggested a header compression scheme for VoIP data on WCDMA (Wideband Code Division Multiple Access) radio transport channels. Yoshimur, et. al. [5] have investigated an RTP/UDP/IP header compression method called MRC (Multiple Reference Compression) [5].

Standard voice compression algorithms such as ITU (International Telecommunication Union) G723 and G729 are used to conserve bandwidth in VoIP systems [8]. However, these compression algorithms may impair the voice quality. Therefore, some researchers have proposed different approaches for voice data compression reported in [9-11]. Data rate compression can also be achieved using voice activity detection techniques as reported in [12-13].

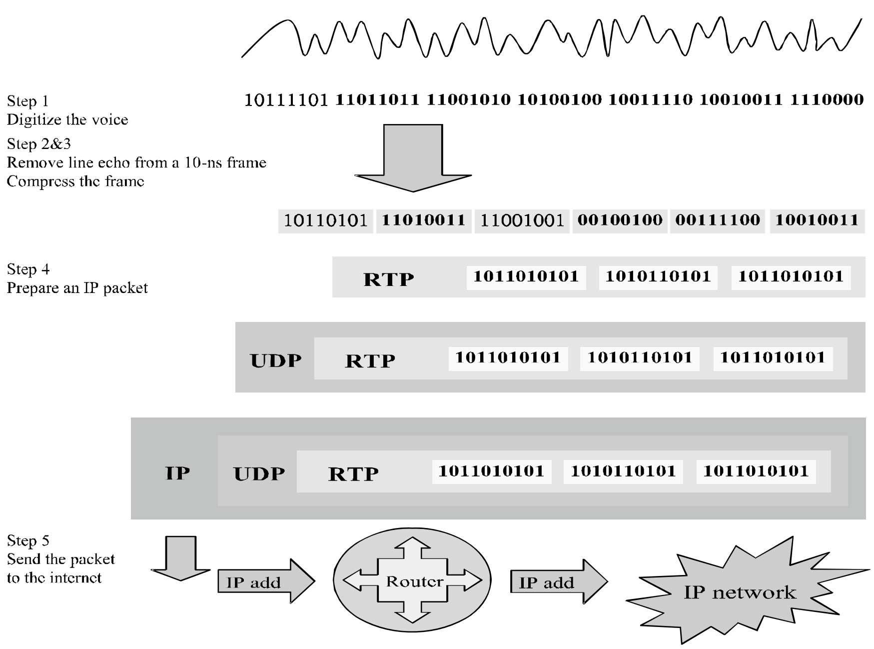

FIG. 1. VOICE TO IP PACKET CONVERSION

Amro and Nour [14] have addressed the issue of data reduction for VoIP utilizing the LPC (Linear Prediction Coding). The suggested scheme maintains a dictionary of the linear prediction coefficients generated by speaking parties at both transmitting and receiving end of VoIP infrastructure. The VoIP data rate is reduced by sending the excitation information and the LP coefficients index instead of transferring the whole coefficients string for a certain frame. Mahlo, et. al. [15] have proposed a variable bit rate speech codec called Speex. This system dynamically adjusts the encoding bit rate and the voice quality depending upon the feedback information about the network congestion, flow priority, and the instantaneous speech properties. Alshakhsi et. al. [16] have investigated the effect of transmission/data and packet size on VoIP quality over WLAN (Wireless Local Area Networks) technology. From their studies, it is concluded that producing different voice packet size will improve the network condition when it is congested. From the simulation studies, it is observed in their work that transmitting packets of big size degrades the voice performance. Furthermore, the effect of loosing large packet is more significant than the effect of loosing a number of packets of small size. Sulovic, et. al. [17] suggested an algorithm which adaptively chooses appropriate CODEC algorithm among G.711, G.729 and G.723 etc to achieve lower compression ratio. The selection is based upon the network conditions, which are reported to the calling party [17].

3. PROPOSED DATA REDUCTION ALGORITHM FOR VOIP

3.1 DR Algorithm Using Short VF

In this section we summarize the DR (Data Reduction) algorithm presented in [18]. The algorithm is based upon the notion that if the voice signal is sampled at relatively high rate, then the equivalent digitized data will have repeated values in a short time window. To gain this repetition, the voice signals should be sampled at the rate of 16000 samples per second. If each sample is assigned a 16 bit code, then 1 msec long VF will contain 256 bits. Each VF is further divided into 4 sub-frames of 64 bits of ¼msec long. The DR algorithm checks for byte repetitions in each frame and appends an 8-bit long CC (Compression Code). The ith bit of CC is set to "1" if the ith byte in the sub-frame is repeated. On the other hand, the ith bit of CC is set to "0" if the ith byte in the sub-frame is not repeated. The process sends the CC and the non-repeated bytes for each SF (Sub-Frame) to the receiving end of the VoIP system. The receiving end reconstructs the original VF with the help of the history of preceding VF it maintains.

3.2 Improved DR Algorithm for VoIP

We realize that the 1 msec duration of VF suggested in [18] is too short and may put some overhead on the VoIP system. To reduce the overhead on the receiving VoIP system, the VF may be considered as 10 msec long. With this length the VF will contain 2560 bits. A one second long VF will contain 100 VFs of 10 msec as shown in Fig. 2.

A 10 msec long VF can be divided into 10 SF of 256 bits each. An ith 256 bit long SF can further be divided into 16 MG (Mini Groups) of 16 bit each (one quantized value) as shown in Fig. 3. Each MG_0-MG_9 represents one 16 bit long quantized level of voice signal.

Two adjacent quantized levels in a VF may have very similar values as shown in Fig. 4. For example, Q_1 and Q_2 levels are much close to each other. Each quantized level may be

2560 bits	2560 bits	2560 bits	●●●	2560 bits
10 m sec	10 m sec	10 m sec		

FIG. 2. ONE SEC LONG VOICE FRAME CONTAINING 100 SUB VOICE FRAMES

represented by 16 bits. If two quantized levels are much close to each other, then their most significant bytes will have similar values. Based upon this observation, an improved DR algorithm is presented here. The algorithm stores the HB (Higher Byte) of the first quantized level in a buffer called HBUF. It will prepare a 16 bit C_0-C_{15} for every 256 bits long SF. First bit of CC is always set to "0". The algorithm compares an ith HB with $(i-1)^{th}$ HB and set C_i to "1" if i^{th} HB is found equal to $(i-1)^{th}$ HB, otherwise C_i is reset to 0 for $i \neq 0$. The algorithm produces modified SVFs which will include 16 LB (Lower Bytes) and the first HB, and optional non-repeated HB's. Modified SF with CC is shown in Fig. 5.

Every complete VF contains 10 SFs and 10 optional 16 bits CC. A sub frame may or may not include the CC depending upon the redundancies of high byte in the quantized voice data.

4. EXPERIMENTAL RESULTS BASED PERFORMANCE ANALYSIS

In order to validate the proposed data reduction algorithm for VOIP, we performed three experiments. In all experiments we recorded the sound for 30 seconds. In the first experiment called Case-1, a regular or heavy conversation was recorded. In the second experiment called Case-2, we recorded the conversation between two parties where in conversation was interrupted intermittently. In the third and last experiment called Case-3, a silence was recorded. A Matlab function was used to record the sound for all three cases. In each case, the voice sample is quantized at the sampling rate of 16 and 32KHz. A 16 bit code is assigned to each digitized sample.

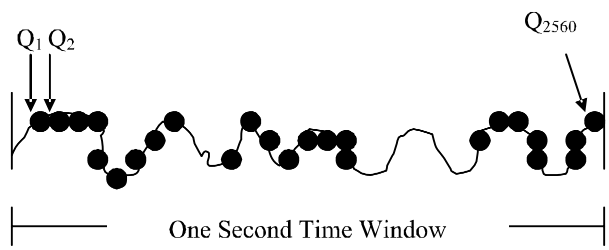

FIG. 4. 2560 QUANTIZED LEVELS OF 10 MSEC VOICE FRAME

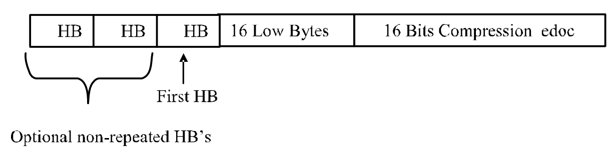

FIG. 5. MODIFIED OR COMPRESSED SUB FRAME

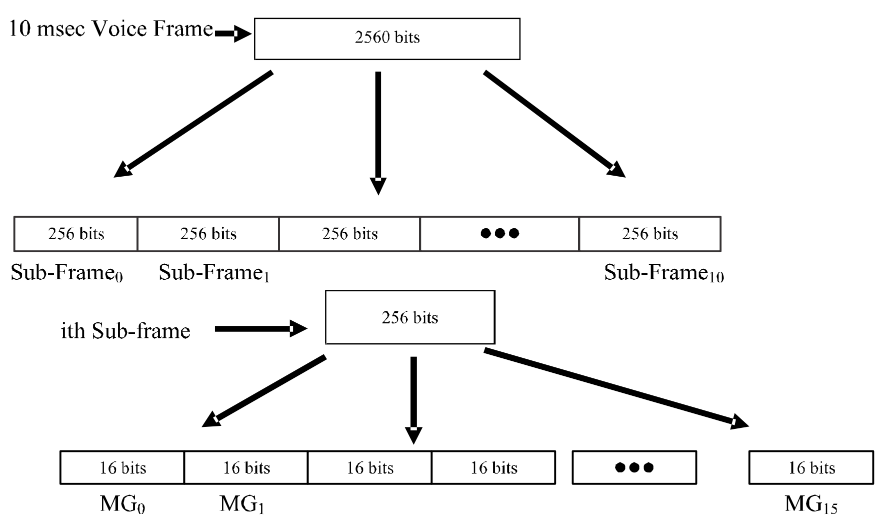

FIG. 3. SUBDIVISION OF VOICE FRAME INTO MINI GROUPS

Table 1 shows a sample of digitized data of Case-1. During regular conversion, the sampled voice signals are close to each other, therefore, the adjacent 16 bits quantized will have redundancies, especially in higher order bytes as reflected in "HB" column in Table 1.

It is obvious from Table 1 that in many 16 bits digitized data, the most significant byte remains constant among consecutive 16 bits data values.

TABLE 1. SAMPLE VALUES OF DIGITIZED VOICE SAMPLE

High Byte	LowByte	High Byte Repeated
01111111	01000000	Not Applicable
01111111	01101010	Yes
01111111	11011100	Yes
10000000	10001000	No
10000001	01011000	No
10000010	00110010	No
10000010	11111010	Yes
10000011	10101010	No
10000100	00111100	No
10000100	10110110	Yes
10000101	00100010	No
10000101	10000000	Yes
10000101	11001110	Yes
10000110	00001000	No
10000110	00101000	Yes
10000110	00110000	Yes
10000111	01001000	Yes
10000110	11110000	No
10000110	10000110	Yes
10000110	00001000	Yes
10000101	01110000	No
10000100	10111010	No
10000011	11100110	No
10000011	00000000	Yes
10000010	00011000	No
10000001	00111100	No

4.1 Discussion of Result

The degree of data reduction obtained as a result of data reduction process is known as CR (Compression Ratio). This ratio measures the quantity of compressed data in comparison with the quantity of original data defineds as:

$$CR = \left(\frac{\text{Length of Original Data String}}{\text{Length of Compressed Data String}} \right) \tag{1}$$

Equation (1) shows that for more effective compression technique, the compression value should be higher than 1. Figs. 6-7 compare the compression ratios of three classes of conversation over VoIP for 16 KHz and 32 KHz quantization rates respectively. These classes are heavy or regular conversation, light conversation and silence. It is noted that the CR for salience is maximum, which is obvious as there will be several redundancies in quantized voice data. Similar observation is also noticed in case of light conversation where pauses during conversation may lead to redundancies. However, in heavy or regular conversation relatively low CR is generated. This situation is probable because each

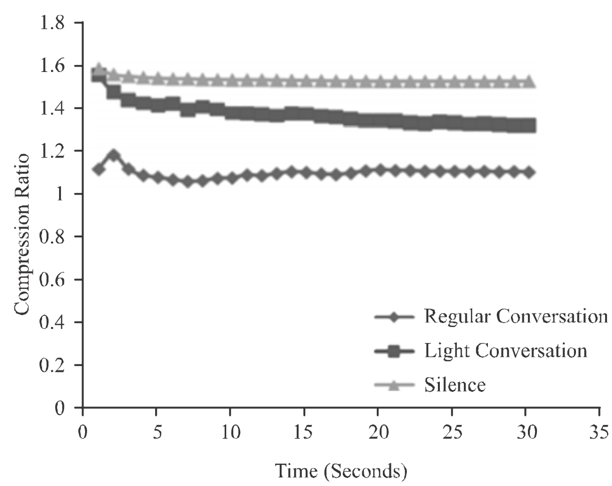

FIG. 6. COMPRESSION RATIO COMPARISON FOR THREE CLASS OF VOIP CONVERSATION FOR 16KHZ QUANTIZATION RATE

digitized voice sample will have less redundancy compared to other samples. Figs 8-10 compare the compression ratios for three classes of conversation for 16 KHz and 32 KHz quantization rates. Obviously, 32 KHz quantization rate gives comparatively a better compression ratio.

5. CONCLUSION

VoIP systems are used to transfer voice data over the Internet after converting them into IP packets. Due to its significant advantages over conventional telephone systems, VoIP is becoming popular medium for voice communication. This paper has proposed a DR algorithm to improve the data transfer rate over VoIP system. The proposed algorithm is executed at both transmitting and receiving ends of the VoIP system. We have done several experiments to validate the proposed algorithm. It is observed that if the voice signal is sampled at relatively higher rate, then there are good chances that the equivalent digital data will have repeated values. The algorithm prepares a CC to indicate the data bits repetition. The proposed algorithm is lossless which guarantees that the transmitted voice data may be

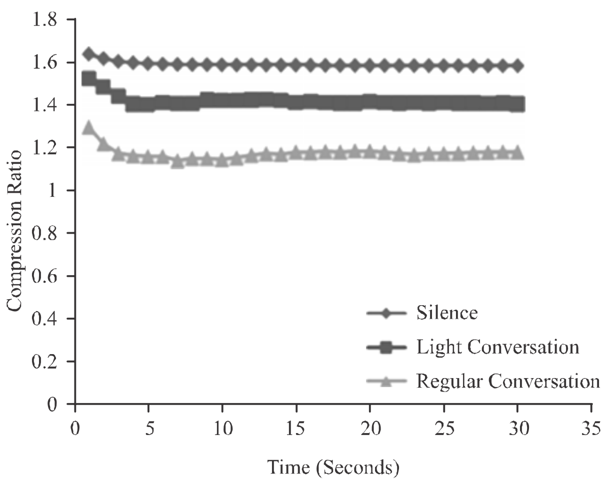

FIG. 7. COMPRESSION RATIO COMPARISON FOR THREE CLASS OF VOIP CONVERSATION FOR 32KHZ QUANTIZATION RATE

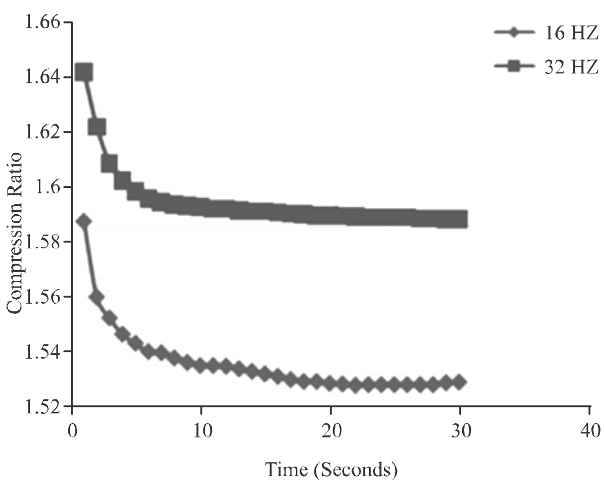

FIG. 8. COMPRESSION RATIO FOR SILENT FOR 16 KHZ AND 32 KHZ QUANTIZATION RATE

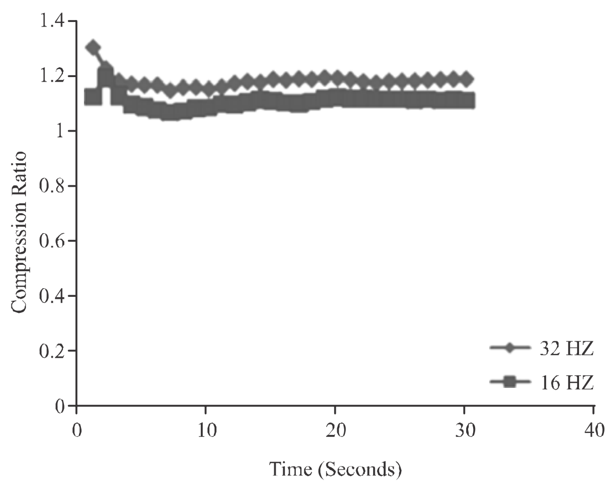

FIG. 9. COMPRESSION RATIO FOR REGULAR CONVERSATION FOR 16 KHZ AND 32 KHZ

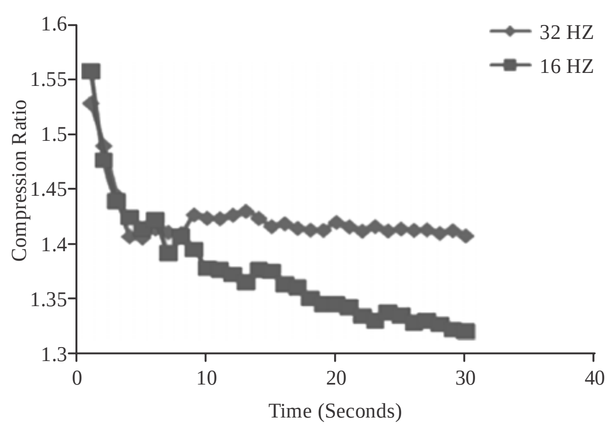

FIG. 10. COMPRESSION RATIO FOR LIGHT CONVERSATION FOR 16 KHZ AND 32 KHZ

recovered up to 100% recovery at the receiving end of VoIP system. Whereas with the use of off-the-shelf standard data compression algorithms, some data components may be lost resulting into impaired voice quality. Therefore, it may be concluded that the proposed algorithm retains the voice quality. The proposed algorithm may be used with header compression algorithms to further improve VoIP data transfer rate.

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