Voice Message Silence Compression

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Abstract Silence compression algorithm for voice message compresses sound files. In This paper software shows greater efficiency for the sound files having number of silence runs. It can be used in telecommunication for sending voice messages in compressed form the advantage of context adaptive voice message Silence compression is that it frequently smooths out the data, which makes it even more suitable [4]. In This paper we get an extra unexpected benefit from lossy compression, above and beyond the compression itself. Silence compression software has successfully been tested for 25 wav files sampled at 8000hz, 16 bits .

Keywords Context Adaptive Lossless Image Compression (CALIC), Lossless Compression

1. Voice Compression

Manipulation of sound by computers is possible since the birth of digital computers, but only in the last five years or so has inexpensive hardware brought this to the average user's desktop. Now the ability to play digitized sound is expected to be an integral part of the "multimedia revolution." The use of multimedia focuses the issue of data compression for users. Computer graphics in particular quickly take up all available disk space. Digitized audio is far less voracious in its storage requirements, but even so it can quickly swallow up all free space on the average user's hard disk. Fortunately for computer users, the world of telephony has used digitized audio since the 1960s, and extensive research has been done on effective methods of encoding and compressing audio data. The world's telecommunications companies were intensely aware of the cost of transmission bandwidth and made efforts to reduce expenses in this area [1].

2. Voice Compression Techniques

2.1. Lossless Compression of Sound

In this technique sound is compressed without any lossy in data, the compressed data can be decompressed to its actual

form. Lossy compression of sound The very word "lossy" implies that when using this type of compression we are going to give up a certain amount of precision [4]. This would certainly not be acceptable when compressing the data or text files we use on our computers.

2.2. In Lossy Compression

In lossy compression we accept a certain loss in fidelity. The signal we get after going through a compression/expansion cycle will not be identical to the original, but we can adjust the parameters to get better or worse fidelity, and likewise better or worse compression. Lossy compression is not necessarily an end to itself. We frequently use lossy compression in a two-phase process: a lossy stage followed by a lossless stage [4]. One advantage about lossy compression is that it frequently smooths out the data, which makes it even more suitable for lossless compression. So we get an extra unexpected benefit from lossy compression, above and beyond the compression itself.

2.3. Context Adaptive Voice Message Silence Compression

Context adaptive algorithm of Silence compression on sound files is the equivalent of run-length encoding on normal data files. In this case, however, the runs we encode are sequences of relative silence in a sound file. This is a lossy technique because we replace the sequences of relative silence with absolute silence [3].

3. Digital Audio Concepts

For modern computers to manipulate sound, they first have to convert it to a digital format. The sound samples can then be processed, transmitted, and converted back to analog format, where they can finally be received by the human ear.

Digitization of sound began in earnest in the early 1960s. Virtually all new switching equipment installed by telephone companies today is digital. But analog switching is still found in older installations and in the smaller PBX and key systems installed in businesses. Of course, the final subscriber loop between the telephone company and the end

user is still persistently analog. Digital audio is now coming of age in the highly visible consumer electronics arena as well. The digital compact disk has nearly completed its displacement of analog LP records [5]. It remains to be seen whether digital audio tape will do the same thing to analog cassette tape, but it seems likely that someday most recorded music will be distributed in digital format.

The X axis in this diagram represents time. The Y axis represents a voltage measured at an input device, typically a microphone. The microphone attempts to faithfully reproduce changes in air pressure caused by sound waves traveling through it. Some human ears can hear sounds at frequencies as high as 20,000Hz and nearly as low as DC. The dynamic range of our hearing is so wide that we have to employ a logarithmic scale of measurement, the decibel, to reasonably accommodate it. This presents a unique set of requirements for digitization[9].

A waveform like the shown in Figure 1 is typical of audio sample. It isn't a nice, clean sine wave that has a regular period and can be described as a simple mathematical function. Instead, it is a combination of various frequencies at different amplitudes and phases. When combined, we see something that looks fairly irregular and not easy to characterize.

Figure 1 A typical audio waveform as it might be displayed on an oscilloscope.

This particular “snapshot” shows about 5 milliseconds (ms) of output. Notice that the largest recognizable components of the waveform appear to have a period of roughly two milliseconds. This corresponds to a frequency of about 500Hz, a fairly characteristic frequency found in speech or music.

Figure 2 A typical audio waveform being sampled at 8KHz.

The first step in working with digital audio is “sampling.”

Sampling consists of taking measurements of the input signal at regular times, converting them to an appropriate scale, and storing them. Figure 2 shows the same waveform sampled at an 8KHz rate. This means that 8,000 times per second a measurement is taken of the voltage level of the input signal. The measurement points are marked with an “x” on the waveform[11].

In most computer systems, this first step of digitization is done with an analog-to-digital converter (ADC). The ADC takes a given voltage and scales it to an appropriate digital measurement. An eight-bit ADC, for example, might have a “full scale” input voltage of 500 millivolts (mv)—it would output an eight-bit value of 255 if the input voltage were 500mv and zero if the input voltage were zero. A voltage between these values would be scaled to fit in the linear range of zero to 255. Since audio signals are AC in nature, the ranges are usually adjusted so that a zero voltage signal falls in the middle of the range. For the previous example, the range would be adjusted to between -250mv and +250mv. Outputs from the eight-bit ADC would range from -128 to +127. The stored sample points then represent a series of voltages that were measured at the input of the ADC. Figure 3 shows the representation of those voltages overlaid with the input AC signal[12]. Note that since the sample points in this case are occurring many times more frequently than the period of the waveform, the digital samples themselves trace the analog signal very accurately.

Figure 3. shows the representation of those voltages overlaid with the input AC signal

Sample voltages overlaid with the input AC signal Now that the sound has been digitized, it can be stored via computer using any number of technologies, ranging from fast storage, such as main processor RAM, to off-line slow storage on magnetic tape. The actual speed of the storage medium is relatively unimportant with digital sound, since the bandwidth needed to accurately store the sound is relatively slow compared to most digital media [13]. Eventually, the sound needs to be played back. This is done via another electronic component that is the converse of the ADC: the digital-to-analog converter (DAC). The DAC is responsible for taking a digital value and converting it to a corresponding analog signal. To be effective, the conversion process needs to be the mirror image of that performed when converting the analog signal to digital. While the exact voltages Produced at the output of the DAC do not need to be

identical to those seen at the input, they do need to be proportional to one another so that one waveform corresponds to the other. In addition, the samples need to be output at exactly the same rate that they were read in. Any deviation here will cause the output frequencies to be shifted up or down from the input, generally not a good thing [13].

Figure 4 shows the output of the DAC when given the same set of samples produced in Figure 2. At first glance, it seems that this is a radically different waveform. All the nice, smooth shapes shown in the earlier figures are gone, replaced by this stair-step, rectangular, artificial-looking creation.

Figure 4. shows the output of the DAC when given the same set of samples produced in Figure 2.

DAC output Fortunately, Figure 4 is not that far removed from Figure 1. Mathematically, the sharp jumps that occur when we move from sample to sample represent high-frequency components in the output signal. These can (and must) be eliminated from the signal by means of a low-pass filter that lies between the output of the DAC and the final amplification stage of the audio output.

A low-pass filter is a network of electrical components designed to let frequencies below a certain value pass through it unhindered, while attenuating frequencies above that point.

An ideal low-pass filter used with the samples shown here would completely stop any frequency above 4KHz and let frequencies below 4KHz pass through with no attenuation [14].

In practice, low-pass filters don't work perfectly, but even a low-budget filter can take Figure 4 and create a nearly indistinguishable copy of Figure 1. Without the filter, the sound sample will still be intelligible, but it will be filled with distracting highfrequency "noise" that is part of the reproduction process.

Figure 5, Sampling at a much higher rate.

Figure 5 shows the same figure when the sampling rate has been stepped up to a much higher rate. This increase in sampling rate clearly does a more accurate job of reproducing the signal. The next section discusses how variations in these parameters affect the output signal.

3.1. Sampling Variables

When an audio waveform is sampled, two important variables affect the quality of the reproduction: the sample rate and the sample resolution. Both are important factors, but they play different roles in determining the level of distortion produced when a sample is played back. The sample resolution is simply a measure of how accurately the digital sample can measure the voltage it is recording. When the input range is -500mv to +500mv, for example, an eight-bit ADC can resolve the input signal down to about 4mv. So an input signal of 2mv will either get rounded up to 4mv or down to 0mv. This is called a quantization error [10]. Figure 6 shows the results of quantization error when sampling a waveform. In some cases the sample point has a larger magnitude than the audio signal, but in other places it has less. When the digitized signal is played back through a DAC, the output waveform will closely track the sample points, resulting in a certain amount of distortion.

Figure 6, Quantization error when sampling a waveform

It might seem that eight bits should be enough to accurately record audio data, but this may not be the case because of the large dynamic range of audio the human ear can detect. If our 500mv range example were used, we might find that our input signal magnitudes range from 1mv to 500mv in a single recording session. The crash of drums in an orchestra could push the ADC to its limits, while a delicate violin solo may never go outside 5mv. If the minimum digital resolution is only 5mv, a very noticeable level of distortion will be introduced during this part of a recording session [15].

The sampling rate plays a different role in determining the quality of digital sound reproduction. One classic law in digital signal processing was published by Harry Nyquist in 1993. He determined that to accurately reproduce a signal of frequency f , the sampling rate has to be greater than $2 \cdot f$. This is commonly called the Nyquist Rate. The audio signal in Figure 7 is being measured at a considerably slower rate than that shown in the previous examples, with noticeably

negative consequences. At several places in the waveform it is not even sampled a single time during an excursion above or below the center line.

Figure 7. A slower sampling rate.

Figure 8 shows the waveform we could expect after playing back the digitized samples stored from Figure 7. Clearly, after the digitized output is filtered, the resulting waveform differs quite a bit from that shown in the previous figure. What has happened is that the high-frequency components of the waveform have been lost by the slower sampling rate, letting only the low-frequency parts of the sample through.

Figure 8. The waveform after playing back digitized samples.

The human ear hears sound up to 20KHz, which implies that we need to sample audio to 40KHz or better to achieve good reproduction. In fact, the sampling rate used for digital reproduction of music via compact disk or digital audio tape is 44KHz, using sixteen-bit samples. The quality of sound achieved at this sampling rate is generally acknowledged to be superior. This does not mean that all digital recordings have to be done at 44KHz rates [17]. Virtually every digital phone system in the world uses an 8KHz sampling rate to record human speech, with generally good results. This means that the phone system is unable to pass any signal with a frequency of 4KHz or higher. This clearly does not render the system useless—millions of long-distance calls over digital lines are made every day. The average speech signal is composed of many different frequencies, and even if everything above 4KHz is discarded, most of the speech energy still makes it through the system [18]. Our ears detect this loss as a lower-fidelity signal, but they still understand it quite well. The ultimate test of all this is how the audio output sounds to our ears. It is difficult to quantify a “quality

of sound” measurement in strictly mathematical terms, so when discussing audio output, it is always best to temper judgments with true listener trials.

4. Voice Message Silence Compression

1. Determine threshold value that can be considered silence, even though it is not pure silence.
2. Extract the data from the sound file to be compressed pass through threshold check if it is below the threshold (considered to be silence) make it pure silence.
3. Run length coding to the manipulated data.
4. Store it as a compressed file.

5. Flow Chart of Context Adaptive Algorithm of Silence Compression

6.1. Experiments and Results

File Name	Size in bytes	Bytes compressed
1)Original1.wav	640440	1462
2)Original2.wav	499100	5000
3)Original3.wav	360100	4088
4)Original4.wav	986320	117781
5)Original5.wav	398600	45

Sampled at 8000 hz,16 bit

Original 2.wav

Original 2. wav after Silence compression /decompression

6.2. Software Implementation Snapshots

6.2.1. Compression

6.2.2. De-compression

7. Conclusions

Voice message Silence compression system that improves data compression in a digital speech storage device, such as a digital telephone answering machine, without undue clipping of voice signals [19]. Instead of employing only real-time compression, the inventive silence system analyzes and compresses or re-compresses digital speech samples stored previously, when the voice messaging system is off-line or otherwise in a low priority state. A method of Context adaptive algorithm of Silence compression comprises receiving real-time speech samples, storing the same in memory, and analyzing the stored speech samples at a later time to determine thresholds for periods of silence [20]. The periods of silence are then compressed, and the silence compressed voice message is restored in memory. In this fashion, the processor is not required to make a silence period determination on-the-fly simultaneous with encoding and compression of the real-time voice message, and thus is not subjected to heavy processor loads typically encountered in real time. This enables more efficient compression of speech samples, lighter duty processors, and improved voice quality upon reproduction by eliminating undesired clipping of the voice signal encountered in prior systems after periods of silence. Voice message Silence compression speech samples is stored in a storage device for subsequent playback [21].

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