MULTIMEDIA NETWORKING

Overview of Data Compression

The benefits of data compression in high-speed networks are obvious. Following are those that are especially important for the compressed version of data.

- Less transmission power is required.
- Less communication bandwidth is required.
- System efficiency is increased.

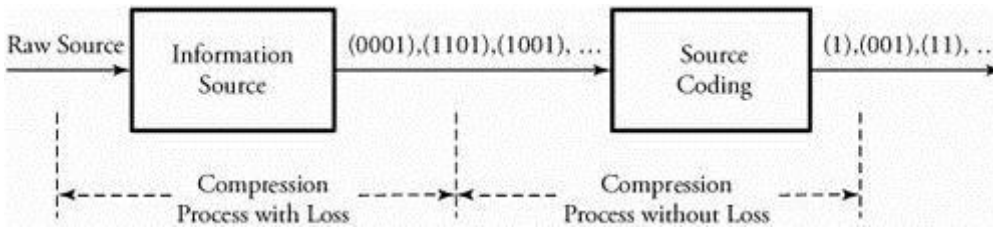
There are, however, certain trade-offs with data compression. For example, the encoding and decoding processes of data compression increase the cost, complexity, and delay of data transmission. Both of the two processes of data compression are required for producing multimedia networking information: compression with loss and compression without loss.

In the first category of data compression, some less valuable or almost similar data must be eliminated permanently. The most notable case of compression with loss is the process of signal sampling. In this category, for example, is voice sampling (Section 7.2). With data compression without data loss, the compressed data can be recovered and converted back to its original form when received. This method of compression is typically applied to digital bits after sampling.

Figure 7.1 shows the basic information process in high-speed communication systems. Any type of "source" data is converted to digital form in a long

information-source process. The outcome is the generation of digital words. Words are encoded in the source coding system to result in a compressed form of the data.

Figure 7.1. Overview of information process and compression in multimedia networks



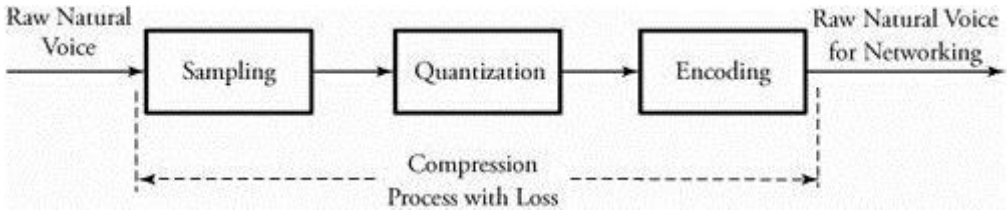
Digital Voice and Compression

Our discussion starts with the voice as a simple real-time signal. We first review the fundamentals of voice digitization and sampling.

7.2.1. Signal Sampling

In the process of digitalizing a signal, analog signals first go through a sampling process, as shown in Figure 7.2. The sampling function is required in the process of converting an analog signal to digital bits. However, acquiring samples from an analog signal and eliminating the unsampled portions of the signal may result in some permanent loss of information. In other words, the sampling resembles an information-compression process with loss.

Figure 7.2. Overview of digital voice process



Sampling techniques are of several types:

- Pulse amplitude modulation (PAM), which translates sampled values to pulses with corresponding amplitudes
- Pulse width modulation (PWM), which translates sampled values to pulses with corresponding widths
- Pulse position modulation (PPM), which translates sampled values to identical pulses but with corresponding positions to sampling points

PAM is a practical and commonly used sampling method; PPM is the best modulation technique but is expensive. PWM is normally used in analog remote-control systems. The sampling rate in any of these schemes obeys the Nyquist theorem, according to which at least two samples on all components of the spectrum are needed in order to reconstruct a spectrum:

Equation 17.1

$$f_S \geq 2f_H,$$

where f_H is the highest-frequency component of a signal, and f_S is the sampling rate.