COMPANDORS

In general, the human voice consists of compound waves of different frequencies and intensities. A large dynamic range is observed when a human voice is converted to electrical signals in ordinary communication systems, say a telephone system; that is, amplitudes of these electrical signals can swing between very wide ranges. In addition, different speakers with different spoken words or syllables captured by different voice sensors also result in considerable variations in signal levels. The intensity variations between a loud syllable of a loud speaker and a soft syllable of a soft speaker could be as large as 70 dB (1). On average, the dynamic range among speakers is about 30 dB to 40 dB (1).

Such large variations in speech signal levels may have an unfavorable effect on the design of a speech transmission system. The primary problems attendant to the large dynamic range of speech signals are additive noises introduced during speech transmission, crosstalk between different users in multichannel communication systems, and overload effects of strong speech signals (1). From an additive noise point of view, strong noise signals may corrupt speech syllables at a low signal level and reduce the intelligibility of the speech signal. On the other hand, a strong speech signal is less affected by additive noises, but will raise the crosstalk problem. Crosstalk is interference by a strong neighboring channel, which has sufficient power to affect the desired speech channel. A large speech signal level will also overload a system. The overload will not only damage a system, but will also result in serious distortion due to nonlinearity in a communication system. Therefore, there is a difficult tradeoff between increasing and reducing speech signal levels. All these problems can be solved by the use of a compandor. The word compandor is a contraction of the words "compressor" and "expandor."

According to whether a compandor operates on an analog signal or a digital signal, it can be categorized as an analog compandor or a digital compandor. An analog compandor operates on a continuous time analog signal and compresses the dynamic range of a speech signal continuously at a syllabic rate. A digital compandor operates on a discrete time signal and adapts to the variations in sampled speech signal levels dynamically by varying the quantization level. According to different operating mechanisms, there are three kinds of digital companding techniques: instantaneous companding, syllabic companding, and hybrid companding.

ANALOG COMPANDING

In the past, speech signal levels were adjusted manually by a telephone operator according to a volume indicator (1). However, it was inconvenient to employ an operator to monitor the progress of a speech transmission and follow the rapid variation of speech signal levels. Moreover, noises occurring during silent periods are very annoying to listeners. Therefore, analog compandors are developed to adjust speech signal levels automatically and achieve noise reduction during silent periods.

Operating Mechanisms of an Analog Compandor

An analog compandor consists of two electric circuits; (1) a compressor to compress the variation in speech signal levels and (2) an expandor to restore original speech signal levels. Both compressor and expandor contain a full-wave or a half-wave rectifier, an LPF, a variable loss device, and an amplifier. Functional block diagrams of a 2 : 1 syllabic compressor and a 1 : 2 syllabic expandor are shown in Fig. 1 (2).

A full-wave rectifier is used to calculate the absolute value of the input signal. If the input signal levels do not change

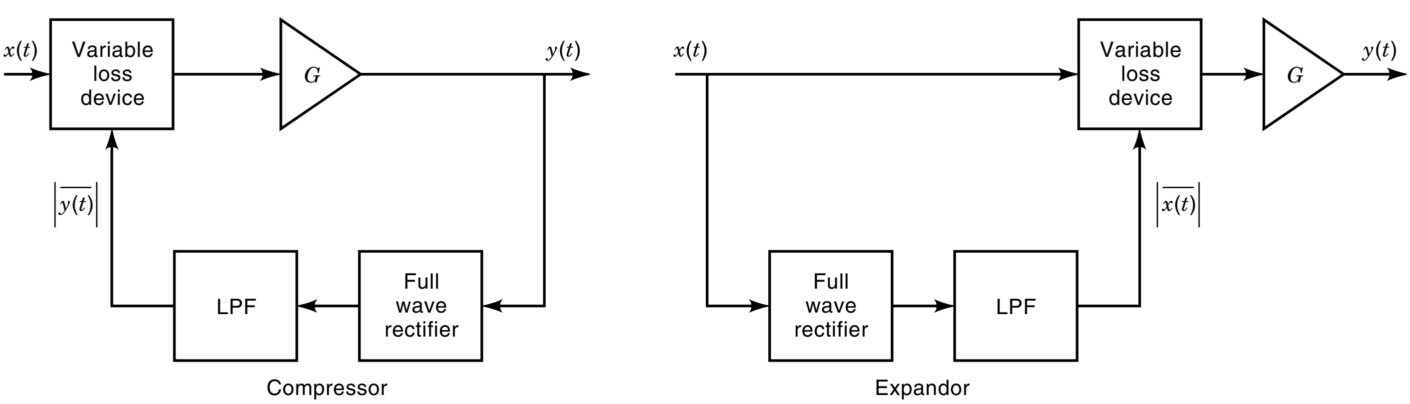

Figure 1. The basic building blocks of a compandor, including a variable loss device, an amplifier, a rectifier, and an LPF.

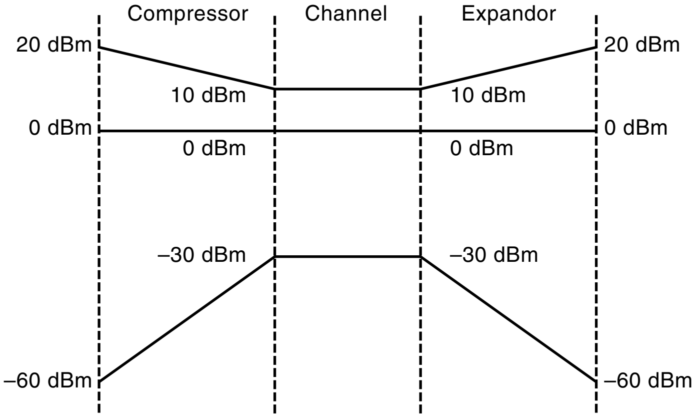

Figure 2. The operation of a compandor with a input dynamic range from 20 dBm to −60 dBm.

fast, a half-wave rectifier can be used instead. A low pass filter (LPF) is used after the full-wave rectifier to average out the envelope of the input signal over a short period of time—that is, at a syllabic rate with a 10 ms to 20 ms time constant. At the heart of a compandor is a variable loss device through which both compression and expansion are performed. At a compressor, the input speech signal to the variable loss device is divided by the output of the LPF. At an expandor, the input speech signal to the variable loss device is multiplied by the output of the LPF. An amplifier is then used to adjust the output signal level in either circuit.

Quantitative Description. The relationships between the input and output signal levels of a compressor and expandor are described by Eqs. (1) and (2):

$$\{\overline{y^2(t)}\}^2 = K_1\overline{x^2(t)} \tag{1}$$

$$\overline{y^2(t)} = K_2\{\overline{x^2(t)}\}^2 \tag{2}$$

where the average is typically taken over a 10 ms to 20 ms time window. By taking the decibel on both sides of Eq. (1), it is readily observed that every 2 dB change in the input speech signal level results in a 1 dB change in the output speech signal level for a compressor; that is, it has a 2 : 1 power compression ratio. Similarly, Eq. (2) shows that an expandor has a 1 : 2 power expansion ratio. According to the decibel versions of Eqs. (1) and (2), the operation of a compandor with a 2 : 1 power compression ratio and a 1 : 2 power expansion ratio on various input power levels is shown in Fig. 2.

The 0 dB point in the figure is referred to as a focal point, which is a reference power level at which both compression and expansion are not functioning. It was assumed that the input speech signal level varies between +20 and −60 dBm. The compression operation is shown on the left side, and the expansion operation is on the right side. If the input signal level is +20 dBm, it is compressed to 10 dBm by the 2 : 1 compressor. If the input signal level is −60 dBm, it is compressed to −30 dBm. Thus, with a full 80 dB input dynamic range, the output dynamic range of a 2 : 1 compressor is 40 dB. Through the expandor, the compressed speech signal is expanded to the original 80 dB dynamic range.

It is instructive to explain the I/O characteristics of a compressor by a curve of the type $y = x^{1/n}$ (3), where n is called the compression ratio. Let y_1 and y_2 be two output speech signal levels corresponding to two input speech signal levels x_1 and x_2; thus we have the following equations:

$$\begin{aligned} 10\log\left(\frac{y_2}{y_1}\right) &= 10\log y_2 - 10\log y_1 \\ &= \frac{10\log x_2 - 10\log x_1}{n} = 10\log\left(\frac{x_2}{x_1}\right)^{1/n} \end{aligned} \tag{3}$$

These equations indicate that if we express signal levels in decibels, the variations in the output signal levels are smaller than the variations in the input signal levels by a factor of n. For an expandor, the I/O characteristics is a curve of the type $y = x^n$, where $1/n$ is called the expansion ratio. Thus, the variations in the output signal levels are larger than the variations in the input signal levels by a factor of n. Compandors with $n = 2$ are commonly used and are called 2 : 1 compressors and 1 : 2 expandors.

Characteristics of an Analog Compandor

An analog compandor is mainly characterized by the following three parameters (1):

1. The compression and expansion ratio.
2. The companding range.
3. The attack and release time.

The Compression and Expansion Ratio (1). The compression ratio is always larger than 1, while the expansion ratio is always smaller than 1. If the compression ratio is too large, the dynamic range of compressor outputs will be squeezed and a small noise induced fluctuation in signal levels might cause undesirable distortion. If the compression ratio is too small, little compression effect is achieved. For general telephone applications, a compression ratio of 2 (2 : 1 compressor) and an expansion ratio of $\frac{1}{2}$ (1 : 2 expandor) are used and they provide satisfactory performance. Compandors with higher compression ratios have been proposed by Greefkes et al. (3).

The Companding Range (1). The full dynamic range of an input signal that a compandor can operate upon is called the companding range. The companding range must be wide enough to accommodate a full range of input speech signals to prevent distortion. In general, a companding range of 60 dB (1) is sufficient. As described above, both compression and expansion operate around a reference point, called the focal point. Maximum noise advantages can be achieved when the focal point coincides with the maximum input signal level of the companding range. On the other hand, the focal point could be smaller than the maximum input signal level to decrease the mean power of the compressor output and alleviate the overloading effect (1).

The Attack Time and Release Time. It is observed on the left-hand side of Fig. 3 that a 12 dB increase in compressor input signal level results in a 6 dB increase in compressor output signal level. The transient period for the output to settle within 1.5 times of its final level is called the attack time. We also observe on the right-hand side of Fig. 3 that there is a 6 dB decrease in compressor output signal level when there is

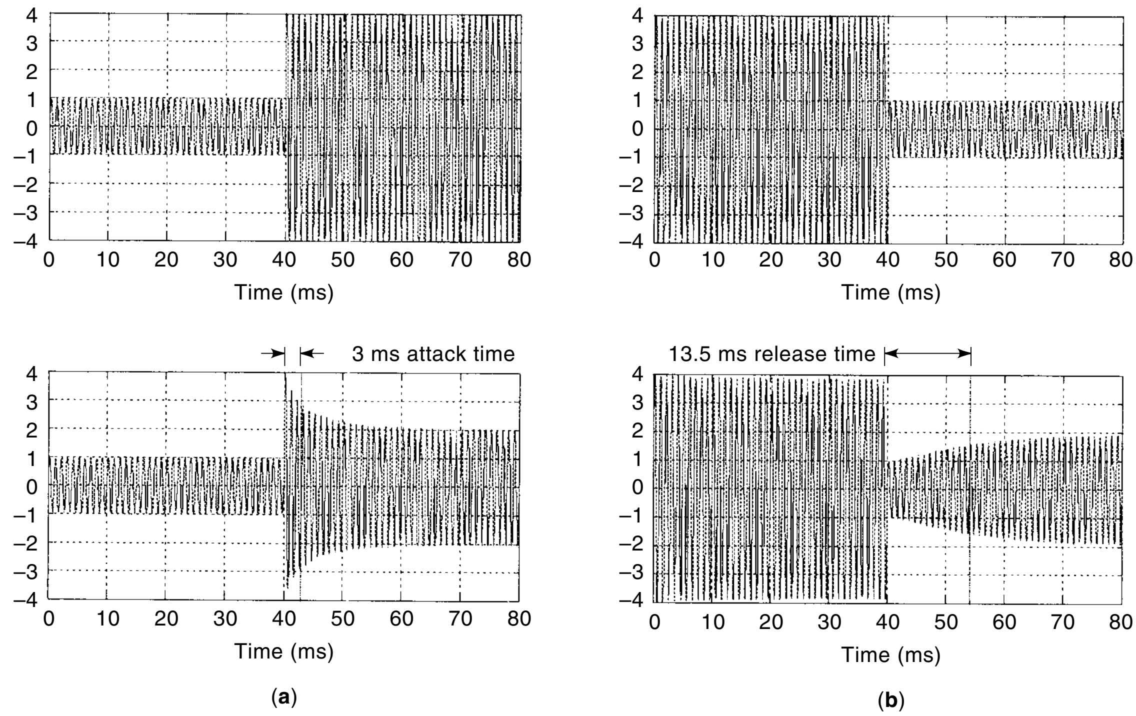

Figure 3. The attack time (a) and release time (b) characteristics of a 2 : 1 compressor.

a 12 dB decrease in compressor input signal level. The transient period for the output to come out at 1.5 times of its original level is called the release time. These two parameters are defined by the CCITT (4), and the recommended values are 3 ± 2 ms for attack time and 13.5 ± 9 ms for release time. They are primarily determined by the time constant of the LPF in Fig. 1 and reflect the response time of a compandor when there are variations in input speech signal levels.

Advantages of an Analog Compandor

Basically, the use of a compandor has two advantages: noise advantage and crosstalk advantage (1).

Noise Advantage. It is suggested that the noise power level should be more than 20 dB below the weakest speech signal level for intelligible communications (1). When a compandor is not used, weak speech signals are vulnerable to additive noises when their power levels are significant. When a compandor is used, the dynamic range is compressed toward the focal point. Therefore, the levels of weak signals are raised and the power level differences between weak signals and additive noises are increased. After the expandor, the power level differences between signals and noises are further extended. This is the noise advantage of a compandor.

Crosstalk Advantage. In a multichannel communication system, the large signal level of neighboring channels might interfere with a desired channel. By the use of a compressor, the peak power of each channel is reduced with respect to the weak signals and the probability of crosstalk interference to adjacent channels is also reduced. In addition, an expandor further reduces the effect of crosstalk during a silent period. Since an expandor extends the dynamic range of speech signal levels, the relatively weak crosstalk signals are greatly attenuated when speech signals are absent. This causes a muting effect during the silent periods.

Applications of Analog Compandors

Two applications of analog compandors are introduced here, which include a frequency modulation scheme and a single sideband amplitude modulation scheme (4), both for mobile radio voice communications. The frequency modulation scheme is based on the Advanced Mobile Phone Service (AMPS) (5), which is the first generation analog mobile phone standard used in North America. The single sideband amplitude modulation scheme is similar to the Amplitude Companded Side Band (ACSB) system proposed by B. Lusignan at Stanford (6).

Frequency Modulation Scheme. AMPS is an analog cellular mobile radio telephone system with a carrier frequency band around 800 MHz, a 12 kHz peak frequency deviation, and a 30 kHz channel spacing. The block diagram of transmitter audio processing in AMPS is shown in Fig. 4. The input speech signals are first bandpass-filtered between 300 Hz and 3000 Hz and then sent to a 2 : 1 syllabic compressor. After the 2 : 1 syllabic compressor is used, a preemphasis circuit with +6 dB/octave frequency response between 300 Hz and 3000 Hz is used to amplify the high-frequency components of speech signals, which are usually relatively weak but important for intelligibility. After the preemphasis circuit is used, a peak clipping circuit is used to limit the instantaneous peak frequency deviation to 12 kHz. Speech waveforms are normally clipped at a level 10 dB below their instantaneous peak level. A bandpass filter is used after the clipper to suppress the out-of-band signals generated by the nonlinear clipping

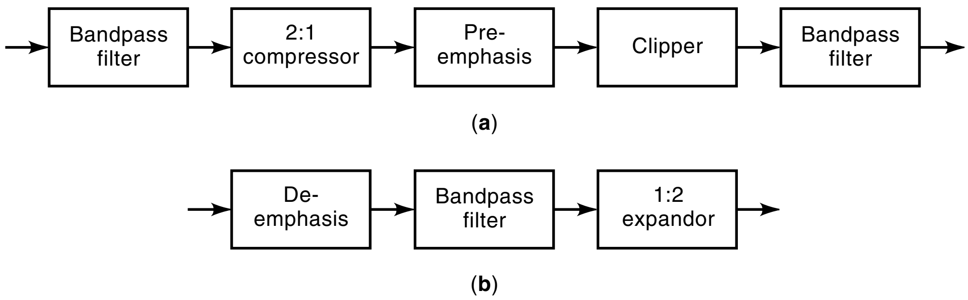

Figure 4. The block diagrams of transmitter (a) and receiver (b) baseband audio processing in the AMPS system.

operation. The block diagram of receiver audio processing in AMPS is also shown in Fig. 4. The operation of a receiver is just the reverse of a transmitter. The discriminator output speech signal is first deemphasized and then bandpass-filtered to suppress out-of-band receiver noises and the effect of random frequency modulation (FM) caused by a fading channel. Finally, a 1 : 2 expandor is used to restore the output speech signal level.

Single Sideband Amplitude Modulation Scheme. The block diagram of transmitter audio processing for the single sideband amplitude modulation (SSBAM) scheme is shown in Fig. 5. The transmitter audio signal processing in the SSBAM system is similar to that in AMPS, except that a 4 kHz pilot tone is added before a second-stage 2 : 1 syllabic compressor.

The use of the pilot tone is described as follows:

1. At the transmitter, the pilot tone provides a threshold for the second-stage 2 : 1 syllabic compressor. If the pilot tone power is larger than the input speech power, the pilot tone power will dominate and the second-stage compressor will not sense any variations in its input signal level. Therefore, no second-stage compression occurs. If the input speech power is larger than the pilot tone power, the second-stage compressor will sense variations in its input signal level and another level of 2 : 1 compression is obtained. The relationship between output syllabic power and input syllabic power with a pilot-added two stage compressing scheme is shown in Fig. 6. An additional 5 dB of compression of speech signal is obtainable through the second-stage compressor when the pilot tone power is set at 10 dB below the peak syllabic power of speech signals. The compressing effect of the second-stage compressor is automatically recorded in the output pilot tone level.
2. At the receiver, the pilot tone provides a reference for coherent demodulation through an antifading technique called feed-forward signal regeneration (FFSR) (7).

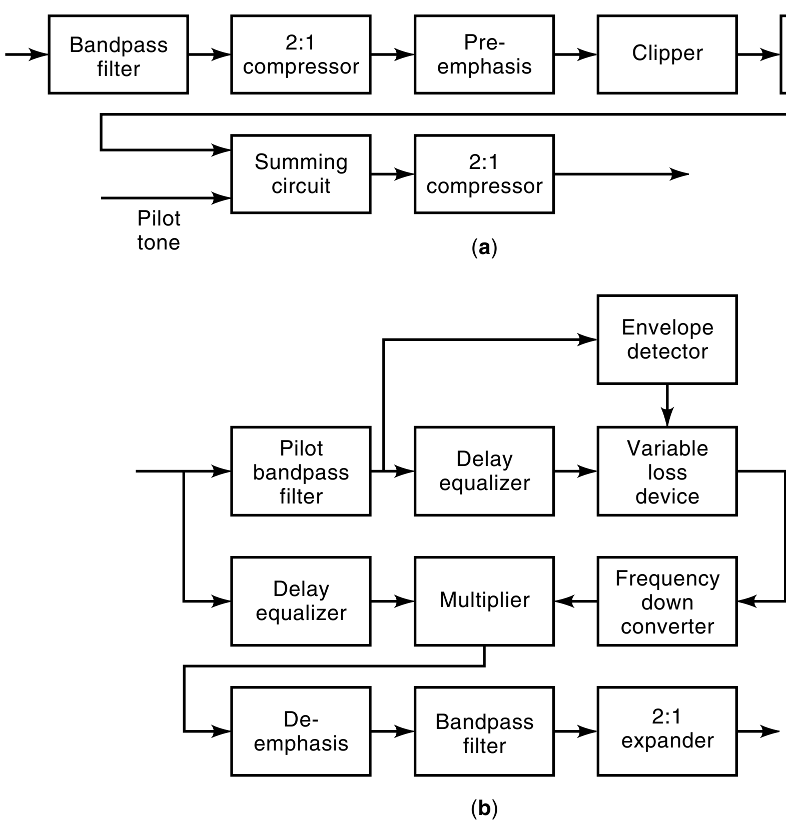

Figure 5. The block diagrams of the transmitter (a) and receiver (b) audio processing for the SSBAM system.

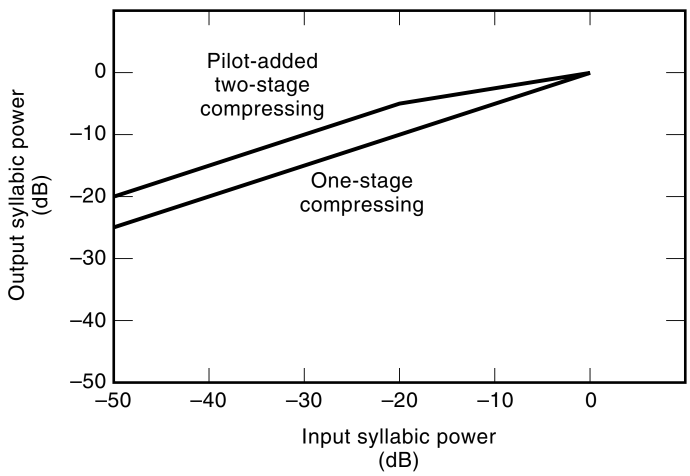

Figure 6. The relationship between output syllabic power and input syllabic power. Both a one-stage compressing scheme and a pilot-added two-stage compressing scheme are shown for comparison.

The block diagram of SSAMP receiver processing is also shown in Fig. 5. This receiver implements the FFSR technique. The faded pilot tone is extracted by a pilot bandpass filter. The filtered pilot tone is then envelope-detected. At the same time, the filtered pilot tone is delay-equalized and sent to a variable loss device whose gain is inversely proportional to the square of the detected pilot envelope. The output is then down-converted to compensate for the pilot tone frequency offset (i.e., 4 kHz). The received signal is delay-equalized and multiplied with the processed pilot tone for coherent demodulation. The multiplication not only compensates for fading in the channel but also compensates for the second-stage compression performed at the transmitter. Afterwards, the regenerated speech signal is processed in the same way as in the AMPS receiver.

DIGITAL COMPANDING

Digital companding techniques can be classified into three categories: instantaneous, syllabic, and hybrid companding. For instantaneous companding, effective quantization levels are adapted at every sample time. There are various instantaneous companding algorithms (8–10), such as the A-law/μ-law algorithm used in pulse code modulation (PCM) systems and the algorithm used in Jayant's adaptive delta modulation (ADM) system with one-bit memory (11). For syllabic companding, effective quantization levels are adapted at a syllabic rate. A good example of syllabic companding is the algorithm used in continuous variable slope delta modulation (CVSD) (12). For hybrid companding, effective quantization levels are adapted according to both an instantaneous algorithm and a syllabic algorithm. Since both instantaneous and syllabic signal level information are used, better speech coder quality can be obtained. Hybrid companding is first proposed by Un and Magill in their residual excited linear prediction (RELP) vocoder system (13). Other hybrid companding schemes are used in hybrid companding delta modulation (HCDM) (9), adaptive differential pulse code modulation (ADPCM) (14), and controlled adaptive prediction delta modulation (CAPDM) (15).

Instantaneous Companding

The goal of instantaneous companding is to adapt to speech waveform at every sample time. In one approach, speech samples are quantized using a fixed nonuniform quantizer. In another approach, a speech waveform is tracked by adjusting the effective quantization levels at every sample time. Both approaches can accommodate a very wide dynamic range in input speech signals.

Nonuniform Quantization. It is known that the quantization noise power of a uniform quantizer with a step size Δ is $\Delta^2/12$. Therefore, the signal-to-quantization-noise ratio (SQNR) decreases as the input signal level decreases. In order to achieve a constant SQNR over a wide input signal dynamic range, a nonuniform quantizer must be used.

In general, high-intensity speech samples are much less frequent than low-intensity speech samples. Furthermore, high-intensity speech samples are less affected by quantization noise from a human ear perception point of view. Therefore, we can use small quantization levels for low-intensity speech samples and large quantization levels for high-intensity speech samples.

A nonuniform quantizer can be implemented as a combination of a compressor and a uniform quantizer. The SQNR of a nonuniform quantizer can be expressed as

$$\begin{aligned}\text{SQNR} &= \frac{\text{Speech power}}{\text{Quantization noise power}}\\ &= \frac{3L^2}{x_{\max}^2}\frac{\int_{-\infty}^{\infty} x^2 P_X(x)\,dx}{\int_{-\infty}^{\infty} P_X(x)\left\{\dfrac{dc(x)}{dx}\right\}^{-2} dx}\end{aligned} \tag{4}$$

where L is the total number of quantization levels, $P_X(x)$ is the probability density function (pdf) of the input speech samples which has a maximum value $x_{\max}$, and $c(x)$ is the compressor input–output characteristics (16). In order to obtain a constant SQNR, the slope of the compressor transfer curve, $dc(x)/dx$, has to be inversely proportional to x:

$$\frac{dc(x)}{dx} \propto \frac{1}{x} \tag{5}$$

It is suggested from Eq. (5) that a logarithmic function can be used. Two commonly used nonuniform quantizers are A-law and μ-law. The compression characteristics of a μ-law quantizer is shown in Fig. 7, and its equation is (see Ref. 16)

$$c(x) = x_{\max}\frac{\ln\left(1+\dfrac{\mu|x|}{x_{\max}}\right)}{\ln(1+\mu)}\,\text{sgn}(x) \tag{6}$$

Instantaneous Quantization Level Adaptation. Another example of instantaneous companding is the algorithm used in Jayant's ADM with one-bit memory (9), commonly called constant factor delta modulation (CFDM). The block diagrams of both encoder and decoder are shown in Fig. 8. The difference between the input speech sample and its prediction is quan-

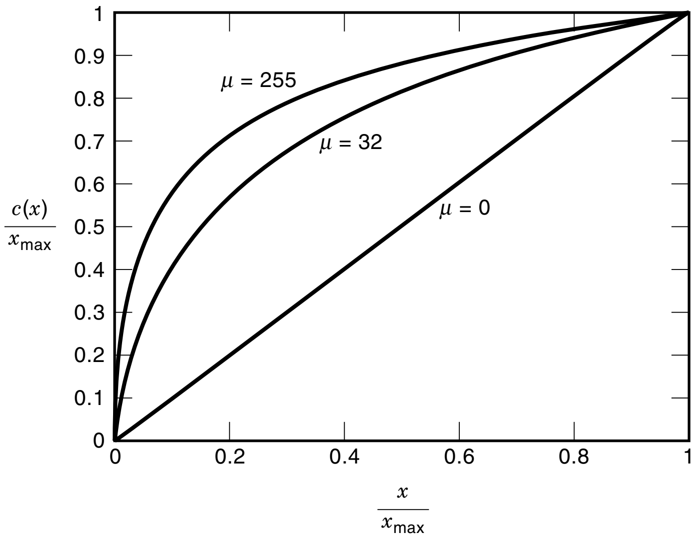

Figure 7. The compression characteristic of a μ-law quantizer. The input and output coordinates are both normalized by the maximum value of the input signal. It is noted that, for $\mu = 0$, the compression curve is just a linear function.

tized by a 1-bit quantizer. The input speech sample is predicted by a "leaky" integrator equation

$$\hat{X}(n) = h\hat{X}(n-1) + b(n)\Delta(n) \tag{7}$$

where h is a leaky factor with value equal to or less than 1.

The effective quantization level—that is, a step size $\Delta(n)$—is adapted from the following equation:

$$\Delta(n) = \begin{cases} P \cdot \Delta(n-1) & \text{if } b(n) = b(n-1) \\ Q \cdot \Delta(n-1) & \text{if } b(n) \neq b(n-1) \end{cases} \tag{8}$$

When two consecutively encoded bits are the same, the coder is slope overloaded and the step size is multiplied by a factor P ($P > 1$). When two consecutively encoded bits are different, the coder is slope underloaded and the step size is multiplied by a factor Q ($Q < 1$). It is necessary that $P \cdot Q \leq 1$ for stability reasons. The optimum values of P and Q are 1.5 and 0.6 (11), in order to achieve good SNR at 24 kbit/s coding rate (11).

The decoder of CFDM is the same as the feedback path of the encoder. An LPF is used to suppress out-of-band noise due to oversampling. Through the use of instantaneous companding, both slope overload noise and granular noise can be effectively controlled by a delta modulator at a moderate sampling rate.

Syllabic Companding

For syllabic companding, effective quantization levels are adapted at a syllabic rate, about every 5 ms to 20 ms. A good example of syllabic companding is the algorithm used in the CVSD. This speech waveform coding scheme is widely used in military communications due to its robustness to channel errors. The block diagrams of CVSD encoder and decoder are shown in Fig. 9. From the figure, $\alpha(n)$, a sample time-dependent variable is generated by the equation

$$\alpha(n) = \begin{cases} 1 & \text{if } b(n) = b(n-1) = b(n-2) \\ 0 & \text{otherwise} \end{cases} \tag{9}$$

and the quantization step size is estimated by

$$\Delta(n) = \beta\Delta(n-1) + \alpha(n)\Delta_0 \tag{10}$$

where Δ_0 is a constant, which is usually the minimum step size, and β is a control factor with value less than 1. If three consecutively encoded bits are of the same sign, the adaptation logic generates a positive signal to excite a leaky integrator. Otherwise, the quantization step size decays gradually at a rate controlled by β.

The input speech sample is predicted from

$$\hat{X}(n) = h\hat{X}(n-1) + b(n)\Delta(n) \tag{11}$$

where h is a leaky factor. The decoder operates in the reverse order of the encoder. In general, CVSD operates at 16 kbit/s or 32 kbit/s. The performance of 32 kbit/s CVSD is similar to that of 48 kbit/s Log-PCM, and the performance of 16 kbit/s CVSD is similar to that of 32 kbit/s Log-PCM (16).

Hybrid Companding

Since human speech signals are nonstationary and have a very wide dynamic range, neither instantaneous companding nor syllabic companding alone work well. It is therefore sug-

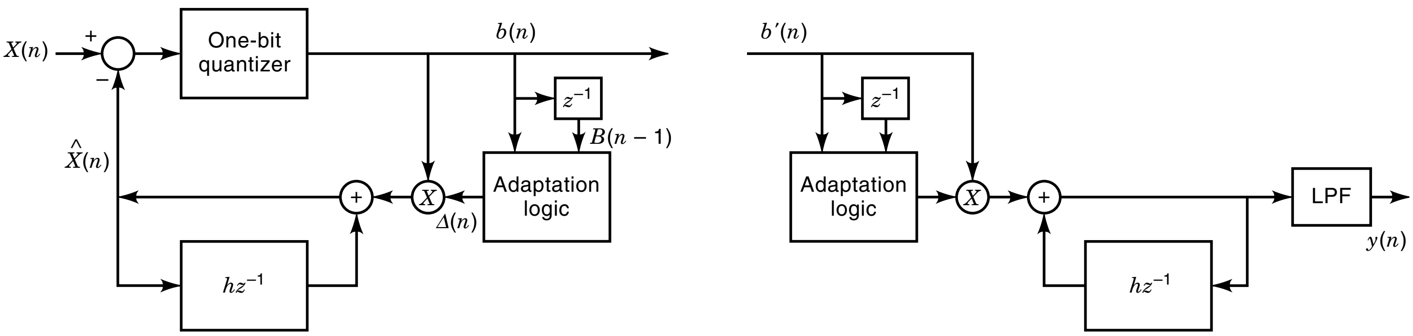

Figure 8. The block diagrams of CFDM encoder and decoder with one-bit memory. The step sizes are adjusted instantaneously.

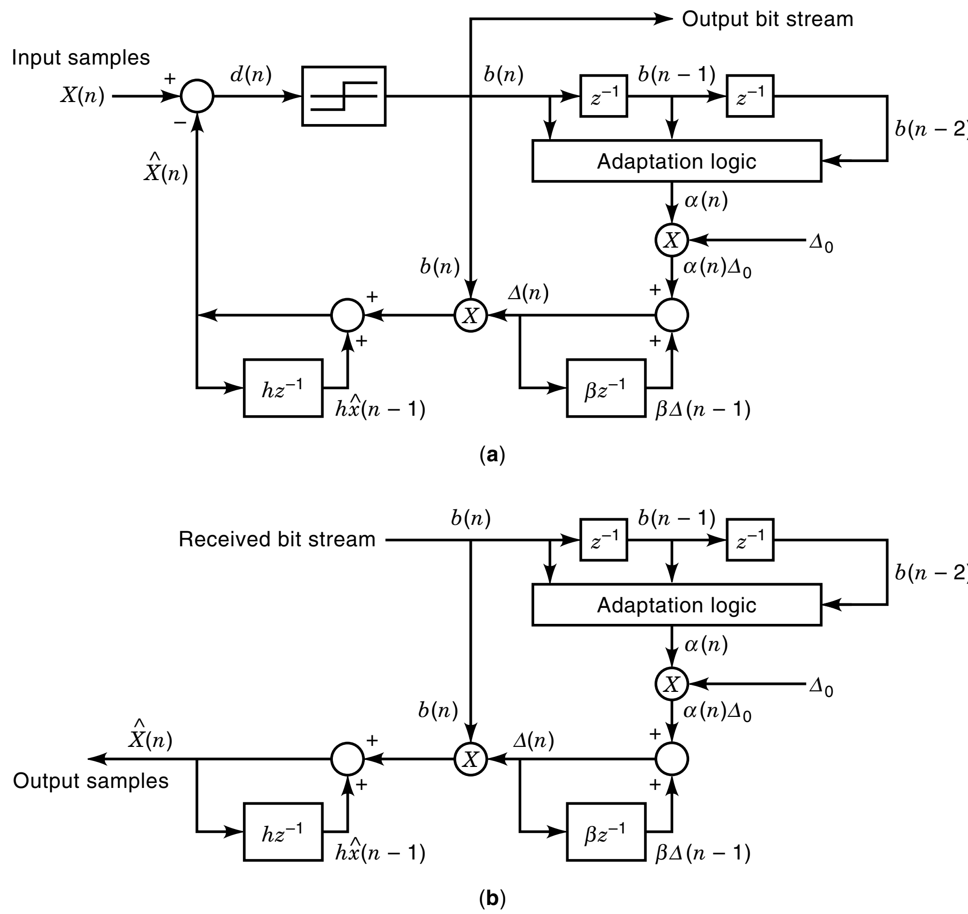

Figure 9. The block diagrams of CVSD encoder (a) and decoder (b). The step sizes are adjusted at a syllabic rate.

gested to combine both instantaneous and syllabic companding. The HCDM employs both instantaneous and syllabic companding algorithms (9).

The block diagrams of HCDM encoder and decoder are shown in Fig. 10. Syllabic companding is achieved by measuring speech signal energy over a time window of about 5 ms to 10 ms. The measured speech signal energy, E, is used to calculate the basic step size, δ_0, using a scaling factor, α (9):

$$\delta_0 = \alpha E \tag{12}$$

The basic quantization step size is updated once for every time window and held constant during that time.

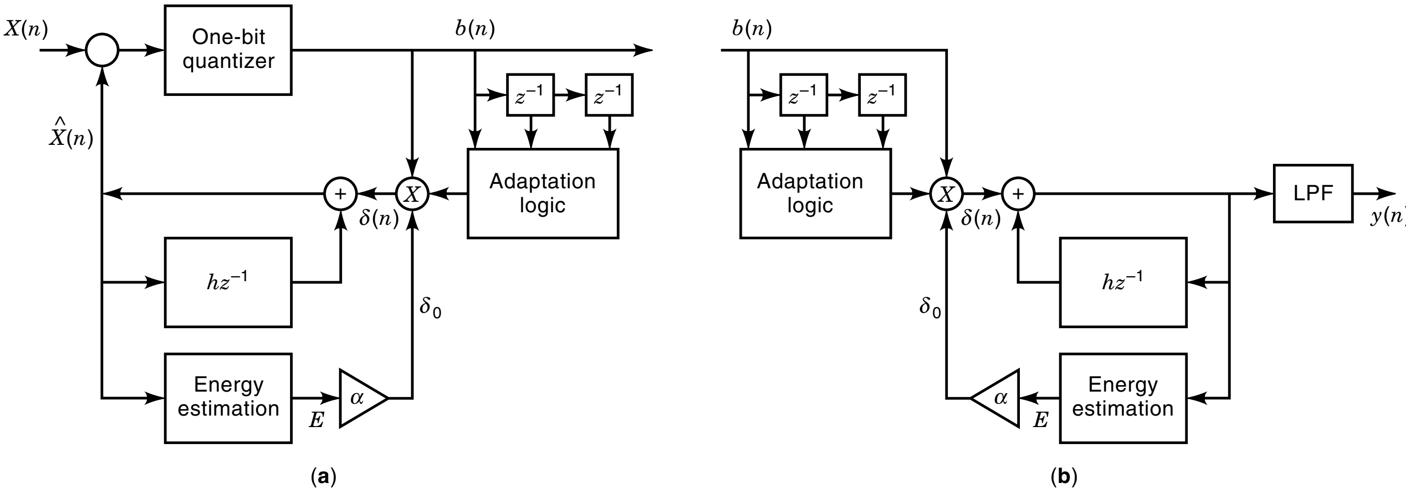

Figure 10. The block diagrams of HCDM encoder (a) and decoder (b). It is evident to see that HCDM combines the features of instantaneous and syllabic companding

$b(n-2)$	$b(n-1)$	$b(n)$	Step size multiplier
1 0	1 0	1 0	1.5
1 0	0 1	0 1	1.0
1 0	1 0	0 1	0.66
1 0	0 1	1 0	0.66

Figure 11. Instantaneous step size adaptation table of HCDM. The adaptation of step size is according to the current and the last two bits. The coder states are classified into overload, transient, and underload state.

The hybrid step size, $\delta(n)$, is adapted as follows (9):

$$\delta(n) = \gamma(n) \cdot \delta_0 \tag{13}$$

$$\gamma(n) = M(n) \cdot \gamma(n-1) \tag{14}$$

where $\gamma(n)$ is an instantaneous step size gain and $M(n)$ is a step size multiplier determined from the table in Fig. 11.

CONCLUSION

Companding is an effective technique to process both analog and digital speech signals with a wide dynamic range. For analog companding, the dynamic range of a speech signal is compressed by a compressor and restored by an expandor. Because compressed signals have a much smaller dynamic range, the transmitted signals are protected against noise added during transmission. The signal-to-noise ratio can be further improved by an expandor at a receiver. For digital companding, the effective quantization levels of a waveform coder can be adapted instantaneously, syllabically, or in a hybrid manner to accommodate a wide input dynamic range. Hybrid companding algorithms are used effectively in sophisticated waveform coders, such as ADPCM (14) and CAPDM (15).

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