SCM — analog modulation for 21st century digital content

Contrary to popular belief, analog modulation isn't going the way of the dinosaur. Signal code modulation, a hybrid digital/analog scheme, may offer a viable alternative.

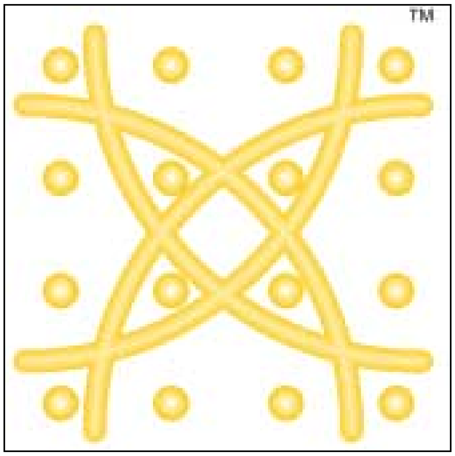
By Eli Pasternak

The industry is in the midst of a digital revolution, where virtually every form of information is either digital — or if it is not already digital (e.g. speech, music and video), it is digitized, stored in digital media, or transmitted digitally over communications channels. Is analog modulation a relic of the past, relegated to a time when AM and FM broadcasts were introduced? Should analog modulation ever be considered for state-of-the-art communications? Are there any clear benefits over digital modulation? The answer is yes and signal code modulation (SCM), a hybrid analog/ digital scheme with unique advantages, is one strong player.

Introduction to signal code modulation

Signal code modulation is a method for transmitting analog information over a noisy channel. SCM provides an analog pipe through which any band-limited waveform can pass, including truly analog information or the output of a digital modem. The operations that SCM performs on the payload signal are simple, as illustrated in Figure 1.

The waveform is sampled and quantized, just like a typical pulse code modulation (PCM) transmission, and the digital signal is then transmitted over the noisy channel using any digital technique, such as quadrature amplitude modulation (QAM). The digital signals are denoted by the symbol D. However, unlike PCM, SCM does not discard the quantization error. This error signal is extracted and transmitted over the noisy channel as an analog symbol, A.



SCM constellation diagram.

The SCM transmission and reception processes are depicted in Figure 2. The transmission channel is divided into two channels. Channel 1 is analog, and channel 2 is digital. In a process essentially identical to PCM, the original analog signal at the system input is sampled at the appropriate rate, based on the sampling theorem, and converted to digital values. The resulting D symbols are transmitted via channel 2 using a digital transmission technique optimized for the channel. Those D symbols represent N bits per analog input sample. To produce the quantization error A, the PCM data is converted back to analog and subtracted from the original input. This A symbol is amplified by a gain of 2^N or any gain that will optimize the voltage swing of the A symbol with that of channel 1.

The SCM receiver performs the opposite operation, combining the A and D symbols into an analog stream replica of the original analog signal. This replica is not a precise copy of the original signal, because noise in the channels could vary the A symbols or cause bit errors in the D symbols. However, the 2^N amplitude gain in channel 1 has provided noise power immunity of 2^{2N} to the A symbols. This is one of the key benefits of SCM.

Its application

This signal processing method is a straightforward approach in implementing a real wireless application, as shown by the following example. A radio channel of bandwidth B is splitting B between channel 1 and channel 2 (see Figure 2). The D symbols are transmitted as digital QAM symbols, and the A symbols are combined in pairs and transmitted as analog QAM. The RF channel bandwidth is divided by time division into the A and D symbols. The actual transmission may consist of a stream of ADADAD... symbols. If a D symbol contains eight bits and the RF channel is suitable for four bits per symbol, the transmitted signal may be arranged as ADDADDADD..., wherein D is a 16-QAM symbol. If, on the other hand, D is a two-bit symbol and the channel is suitable for four-bit symbols, the transmission will be AADAADAAD.... Each particular application will determine the optimum SCM mode in terms of choice of number of bits per D symbol.

The D symbols may be aggregated and encoded using forward-error correction (FEC) techniques, using typical framing and scrambling techniques. The SCM modem design can become a modification of a conventional all-digital QAM modem, using the synchronization and channel equalization techniques with slight modifications; thus SCM is suitable for implementation by existing digital modem techniques. Furthermore, even the A symbols can be processed digitally using digital signal processing (DSP) techniques. The presence of D symbols next to the A symbols simplifies the design of such a modem by having the receiver rely on the digital symbols to calibrate the signal gain and perform

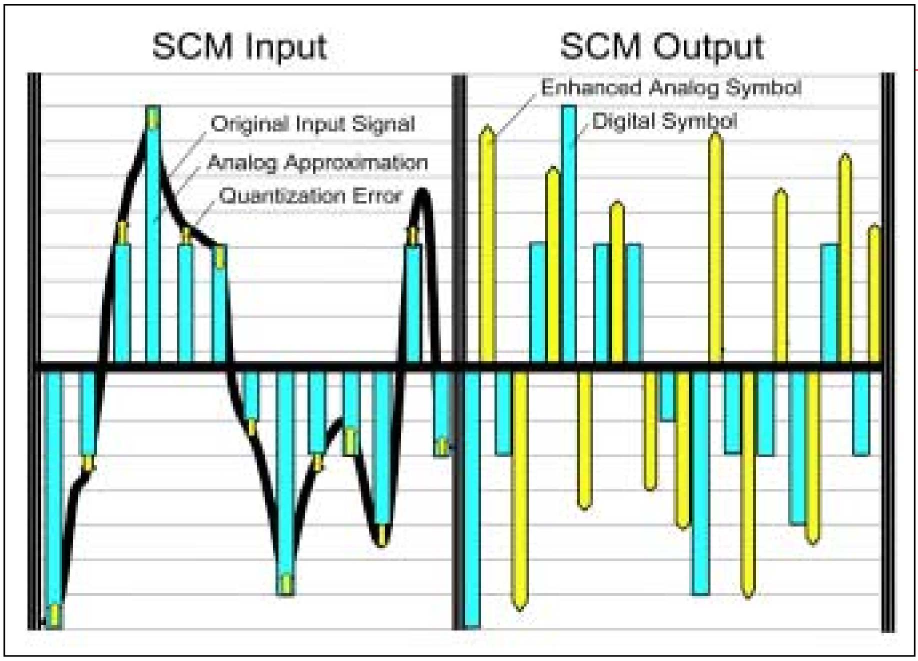


Figure 1. SCM operation on a payload waveform.

the adaptive equalization. An example of a transmission constellation in which the original signal is an unmodulated carrier and the *D* symbols are 16-QAM is shown in Figure 3. (Note: An explanation of Figure 3 appears in Appendix B.)

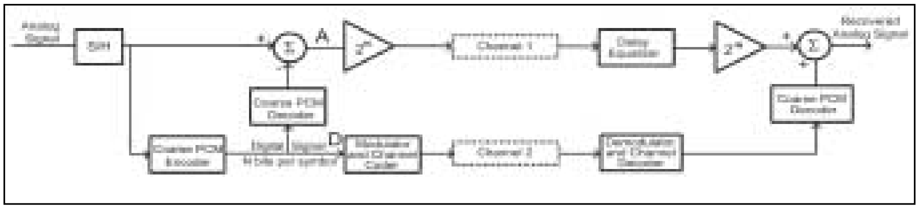


Figure 2: SCM transmission and reception processes.

Performance comparison

Before explaining why SCM is a nearly ideal analog communications method, it is necessary to define the ideal reference and compare it with existing alternatives. A communications link designer faced with an additive white Gaussian noise channel of bandwidth *B* and a limited signal-to-noise ratio (SNR) might choose a digital link as a first choice. The analog samples are converted to digital with a resolution of *M* bits per symbol. It is well-known that, by using an ideal digital error-correction coding technique, the channel can pass the information error-free at a bit rate that is called the channel capacity *C*, where:

Figure 3. SCM constellation with unmodulated carrier input. (An explanation of this figure appears in Appendix B.)

$$C = B \log_2 (1+\mathrm{SNR})$$

If the analog signal-sampling rate is *R*, the number of bits per symbol cannot exceed $M = C \cdot R$; thus *M* is limited and quantization noise is unavoidable.

The designer may consider analog modulation, such as frequency modulation FM. Frequency modulation is known to improve the output signal-to-noise ratio, referred to below as the signal-to-noise ratio destination (SNRd), compared with the signal-to-noise ratio channel (SNRc). FM accomplishes this advantage at the expense of bandwidth increase. The designer will soon find that FM is inferior to PCM at the minimum-channel SNR. This is because FM suffers from a threshold phenomenon in which the performance degrades drastically with channel SNR, several decibels (dB) above the PCM[1]. Shannon, who studied this subject, introduced rate distortion theory, from which the performance of an ideal communications system could be derived. Such system performance will depend on the bandwidth expansion factor b, which is the ratio between channel bandwidth and source information bandwidth.

The numbers

Shannon has derived the following equation[2]:

$$SNRd = \left(1 + \frac{SNRc}{b}\right)^{b} - 1$$

PCM can meet this SNR curve in one point, but the quantization noise remains unchanged as channel SNR exceeds the threshold value. A similar expression can be derived for SCM, as shown in Appendix A. SCM performance is as follows:

$$SNRd = \frac{SNRc}{b}\left(1 + \frac{SNRc}{b}\right)^{b-1}$$

This expression is plotted for *b=2* and *b=4* in Figure 4 for both SCM and the Shannon Bound.

In the threshold corner of SCM, the channel SNR is a fraction of a decibel from the ideal Shannon Bound. If the SCM curve is to follow the Shannon Bound for every channel SNR, SCM must adapt its bit rate and error-coding scheme for each SNR. A practical implementation of SCM is more likely to be optimized for one threshold point only: for example, 23 dB output SNR in Figure 4. The straight line “single point SCM” depicts the resulting performance. The advantage of SCM is now becoming apparent; while optimum PCM performs as well at the threshold channel SNR, SCM continues to improve as channel SNR improves. Any practical communications link operates most of the time at a significant margin above threshold. PCM would remain in threshold performance regardless of SNR, while SCM improves.

Because SCM is essentially an ideal modulation scheme for analog signals, it is difficult to come up with a significantly better scheme. Therefore, SCM is likely to remain useful for the foreseeable future, making it a preferred choice for emerging applications. The obvious question is: Who needs an analog modulation scheme? Surprisingly, most SCM applications are related to digital communications, and this paradox will be resolved in the next section.

Wide range of applications

An SCM-based communications link is basically a transparent, band-limited analog pipe with near-ideal performance in noisy channels. Every analog signal could potentially use SCM because it can outperform other existing modulation schemes. However, SCM has a compelling advantage for digital communications applications as well. For example, SCM can pass digital information by acting as a repeater of a digital channel. This application provides a wireless extension of cable modem digital information.

As illustrated in Figure 5, a cable modem termination system (CMTS) transmits a 42 Mb/s 256-QAM signal in a 6 MHz cable channel shared among the cable modems located at the subscribers' premises. The return upstream path from the cable modems is a 10 Mb/s 16-SQAM signal in a 3.2 MHz cable channel. The signals are carried by a combination of fiber and coax referred to as a hybrid fiber/coax (HFC) network. The fiber delivers a large amount of bandwidth over long distances with strong noise immunity. Coax cables distribute the signal between the fiber and each subscriber.

To reach a business park located beyond the reach of the existing HFC network, the cable operator installs an SCM-based point-to-multipoint wireless access system at any point on the HFC network that has line-of-sight to the business park. All customers located at a particular site share the SCM radio located at that site. The subscribers simply use low-cost cable modems that connect to the SCM radio via a shared coax cable. The wireless subscribers can even share the same cable channels with purely wired subscribers because the wireless link is transparent to the cable equipment.

The significance of SCM in this application is its ability to take a 256-QAM signal and transport it over a wireless link suitable only for a lower modulation scheme, such as 16-QAM. SCM provides significant additional noise immunity, as is depicted in Figure 4 because it uses bandwidth expansion to improve the destination SNR. There is a non-SCM alternative: the 256-QAM signal could first be demodulated back to the original data bits, then modulated as 16-QAM, transmitted over the wireless link, demodulated at the destination, and finally remodulated using 256-QAM. This alternative would be much more costly, given the amount of processing required. It would also add significant latency to the information transported because an efficient channel must perform the error correction of the original signal before transmitting it over the wireless link. Furthermore, because SCM provides a transparent link that is not sensitive to protocol evolution or variations, it is more future-proof and versatile than specific digital standards.

A second example is an application from another discipline: digital audio

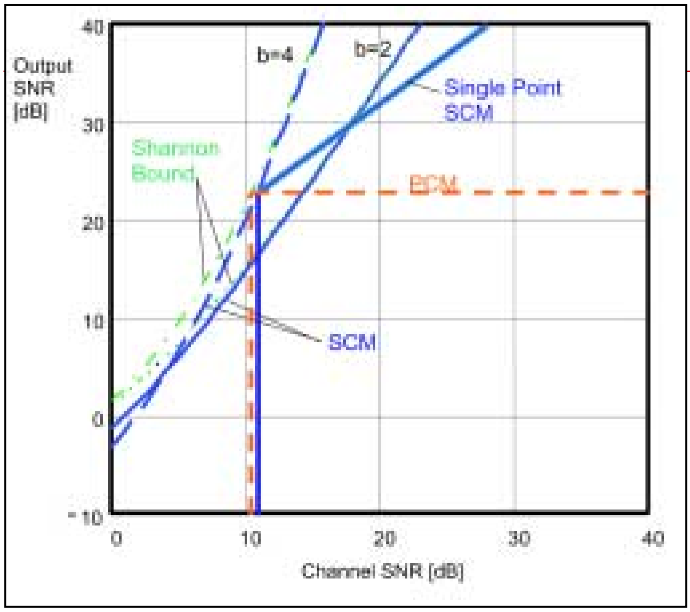


Figure 4. SNR performance of an ideal analog communications channel, PCM and SCM.

recording. A new-generation audio CD could include a digital track identical to and compatible with existing CD tracks, and in addition, have an analog track to provide the enhanced quantization error. Such an analog track would provide audio performance that depends on the quality of the recording and of the disc player. The most discriminating audiophile could use the more sophisticated player for true analog reproduction, while the less discriminating users would enjoy the low-cost CD technology in its current format.

The call

SCM is a versatile, hybrid analog-digital modulation scheme for transmitting analog signals to provide a repeater function for digital signals. The transparent nature of SCM allows it to relay signals without depending on transmission protocols and modulation formats. Such transparent links provide significant cost reduction because they use mass-produced end equipment such as cable modems. The near-ideal performance of SCM makes it a good choice for low-cost transparent links.

References

1. B. P. Lathi, *Modern Digital and Analog Communication Systems*, Third Edition, 1998, page 720.
2. Ibid., *eqn. 15.70a*, page 718.

Appendix A: SNR performance of an ideal SCM channel

Without loss of generality, it can be assumed that the source signal is sampled at the Nyquist rate and is transmitted as a discrete symbol. If b=1, the modulation is simply pulse amplitude modulation (PAM). It is well-known that this link meets the theoretical limit of performance set by the Shannon capacity theorem[i]:

$$SNRd = \left(1 + \frac{SNRc}{b}\right)^{b} - 1 \qquad \text{(Eqn. 1)}$$

SNRd is the output (destination) SNR, and *SNRc* is the channel SNR. Clearly, for $b=1$, there is no gain, and thus PAM meets the ideal performance. For $b>1$ there is a significant performance gain, but how can it be achieved?

Suppose that a band-limited Gaussian channel is used for binary transmission and that by channel coding, the channel capacity is achieved digitally. Such coding is not practical, but today's codes closely approach the theoretical limit. The capacity of a Gaussian channel is:

$$C = B\log_2(1+\text{SNRc}) \qquad \text{(Eqn. 2)}$$

C is the capacity in bits per second of error-free transmission over a channel of width *B*.

Next, consider the following mixed link:

Total bandwidth is *bB*.

It has an analog portion of bandwidth *B*.

It has a digital portion of bandwidth (*b-1*)*B*.

It maintains the same peak power as the original PAM signal. Thus, with the bandwidth increase, the channel SNR is decreased by the factor *b* (i.e., noise power increases by the factor *b*). The available capacity of the digital channel is:

$$C = (b-1)B\log_2\left(1 + \frac{SNRc}{b}\right) \qquad \text{(Eqn. 3)}$$

These bits are used for qualifying the analog symbols in the analog portion. As there are *2B* symbols/sec and *C* bits/sec, there are $M = C/2B$ bits per analog symbol. Now the analog signal in the range [-a, a] is not transmitted in full. Instead, it is divided into 2^M equal segments. We assume that M is an integer; however, we treat M as a continuous variable in the following analysis. For each symbol, only one of the segments contains the analog information. This segment is magnified to the range [-a,a]; i.e., it is amplified by a factor of 2^M and transmitted with PAM modulation. The M bits associated with it are transmitted in the coded digital channel and recovered. The receiver will then take the analog signal, shrink it back by 2^M and translate it to the original level. The signal-to-noise increase is the square of the magnification, thus it equals 2^{2M} =

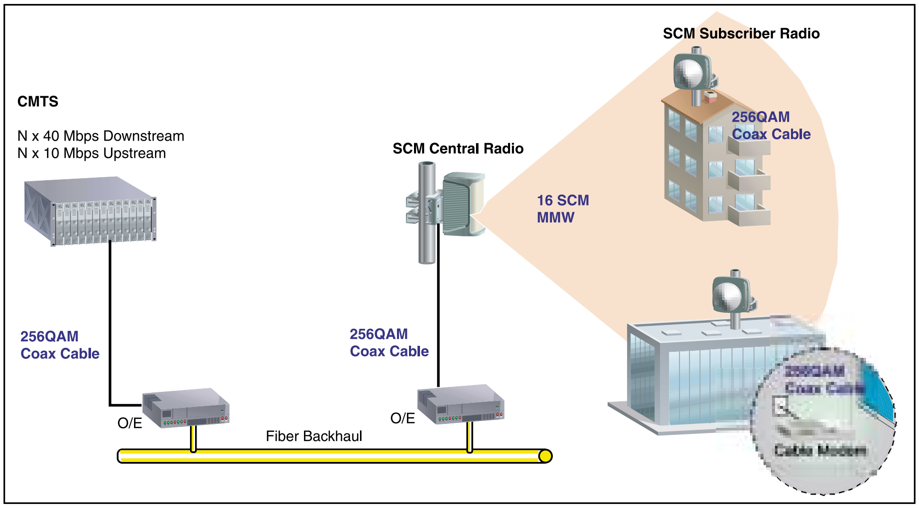


Figure 5.

$2^{C/B}$. Therefore:

$$SNRd = \left(\frac{SNRc}{b}\right)2^{\frac{C}{B}}$$

(Eqn.4)

Substituting Eqn. 3 for C in Eqn. 4:

$$SNRd = \left(\frac{SNRc}{b}\right)2^{(b-1)B\log_2\left(1+\frac{SNRc}{b}\right)}$$

And simplifying:

$$SNRd = \frac{SNRc}{b}\left(1+\frac{SNRc}{b}\right)^{b-1}$$

Reference

1. B. P. Lathi, p. 711.

Appendix B: SCM constellation explanation

How does SCM cause this unusual constellation diagram on page 54?

This diagram represents a specific SCM mode, AAD, wherein the bandwidth expansion is b=1.5, and the input is an unmodulated carrier within the system pass band, although not necessarily at center frequency. I is a 16-QAM symbol carrying four bits, and the SCM conversion uses only a single bit per A dimension. The image is a superposition of many sampling points, each representing a single *A* or *D* point. The collection of all *D* points creates the familiar 16-QAM constellation. When depicted as *I* vs. *Q*, the original analog carrier would produce a circle, a familiar Lissageau waveform. However, SCM, in this example of *N=1* bits per dimension, divides such a circle into four quadrants. The analog difference signal is a quarter of a circle. The four quarters are superimposed in the image, thus forming the symmetrical waveform. By changing the amplitude of the carrier, the quarter circles change their radii, creating different images.

RF

About the author

Eli Pasternak is co-founder, senior vice president and chief technology officer of BridgeWave Communications. Prior to BridgeWave, he served as CTO and chief scientist at Netro Corporation, which he co-founded in 1994. In his role at Netro, he was responsible for most of the IP related to both AirMAN and AirStar products and wrote four fundamental core technology patents in the area of wireless ATM and its associated MAC layer. Prior to Netro, Pasternak was a co-founder of Telestream and a technology consultant in a broad range of technical disciplines, including medical electronics, wireless, microwave products, communications products, system architecture, and food processing. He also served as a consultant in two related areas, as the director of engineering during the first year of P-COM's operations, and as a system architect for Plantronics' wireless infrared headset products. He received B.S.E.E. and M.S.E.E. degrees from the Technion in Israel. He can be contacted at: **www.bridgewave.com.** A demonstration of BridgeWave's technology can be viewed at this site as well.