Digital Transmission Fundamentals for Cable Engineers

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ABSTRACT

This paper will examine various techniques as they apply to digital transmission on CATV systems. A brief review of analog to digital conversion techniques, as well as an overview of digital modulation methods, will be presented.

CATV system infrastructure will be examined both from a current capabilities viewpoint and from a forward looking perspective considering the evolution of delivery and networking techniques with fiber, wireless and star-bus bidirectional topologies.

Because analog signal transmission will exist on CATV networks well into the future, techniques for integrating digital capabilities into a broadband hybrid analog/digital network will also be discussed. The evolution of these networks, and their interactions with existing and evolving TDM telephony networks will also be considered.

INTRODUCTION

New cable television system technologies are being introduced at an ever-increasing pace, requiring that technical personnel assimilate the new technologies into their existing operations. Digital signal transmission on cable systems is just such an example. This paper will provide an overview of some of the key technical considerations involved in any kind of digital signal transmission, and more specifically, issues involved in a hybrid analog/digital cable television environment.

Some questions will be raised about specific techniques for cable digital transmission which are the subject of investigations at laboratories both nationally and internationally.

Additional questions posed relate to the deployment of digital technologies in a cable system, especially issues relating to system test and measurement and subscriber location equipment configurations.

ANALOG TO DIGITAL CONVERSION

The first step in preparing an analog signal for transmission through a digital transmission system is the process of analog to digital conversion, often referred to as A/D conversion. The A/D process determines the maximum level of performance that is achievable in a "perfect" digital system, where perfect implies all coding and transmission processes are implemented with zero error.

The A/D conversion process is shown diagramatically in Figure 1. The first step is sampling of the properly band limited source waveform. The Nyquist Sampling Theorem[1] states that a signal of baseband bandwidth 1/T Hz can be completely represented by evenly spaced instantaneous samples at a rate of 2/T samples per second (sps). In order to eliminate ailiasing, or spectral overlap and interference in the signal reconstruction process, it is necessary to strictly band limit the source spectrum to 1/T Hz. Since perfect filters are impractical, sampling is typically performed at rates that are higher than 2/T, or signals are band limited with realizable filters to less than 1/T, or a combination of both techniques is used. As an example, a standard "4 kHz" voice channel for telephony services[2] is sampled at 8 Ksps after the source is band limited by a filter whose 3 dB bandwidth is 3.4 kHz and provides 80 dB of rejection at 4.0 kHz.

The result of the sampling process is a stream of impulses, evenly spaced at a rate of 2/T sps, with amplitudes that exactly represent the amplitude of the signal waveform. The next step in A/D conversion is to map the infinite set of amplitude values for these impulses into a codeword set that simultaneously meets the performance objectives for the channel and the capacity constraints due to the transmission channel. The mapping process is known as quantization, the impairment introduced is referred to as quantization noise. From a practical standpoint, an additional limitation must be considered, that being the availability of practical A/D conversion circuits which cover the bandwidth of the signal. For example, typical A/D circuits for sample rates around 50 Ksps, such as those used for CD audio, are readily available at 16 bit accuracy, with 18 bit units available for premium applications. A/D circuits for video channels in the 10 to 15 Msps range are typically 8 or 9 bit accuracy, with 10 bit units representing the premium or "best currently achievable" level of performance.

The simplest quantization process is linear pulse code modulation, or linear PCM. As depicted in Figure 1, linear PCM represents each impulse of exact amplitude with the code word for the amplitude which provides the closest match from the 2^N-1 values available, where N is the accuracy or resolution of the A/D converter. Linear PCM means that the 2^N-1 values are equally spaced. The total range covered by the available code words must be carefully chosen in order to accurately represent the source at its maximum and minimum instantaneous amplitudes without clipping.

The fundamental degradation inherent in the A/D process is quantization noise, a measure of the average error introduced in mapping the continuous analog amplitude space into the discrete code words available. For a simple sine wave which completely spans the 2^N-1 code words, the quantization noise is given by[3]:

$$S/N=6.02*N+10.8 \text{ dB } (\textit{pk - pk sig/rms noise}) \quad \text{(Eqn 1)}$$

For complex wave forms such as video, standard code word coverage of the amplitude space must account for the peak factor of the composite signal if distortion due to clipping is to be adequately controlled. Signals with large peak factors (peak to rms ratio) will experience larger quantization error if minimal peak clipping is required. Typical values for 8 bit coders (255 code words) are $V_B = 16$ and $V_W = 235$ ($V_{PP} = 219$) for component systems[4] (YUV), V_B=64 and $V_W = 212$ ($V_{PP} = 148$) for NTSC composite system coders, where $V_B = V_{Black}$, $V_W = V_{White}$ and $V_{PP} = V_{Peak\text{-}to\text{-}Peak}$. Coders using 9 and 10 bit code words (2^9-1 = 511 or 2^{10}-1 = 1023 code words) would use comparable ranges.

Referring once again to Figure 1, it should be evident that sampling the wave form at higher rates permits more accurate representation of continuous amplitude wave by the sample stream, albeit at an increase in coder output data

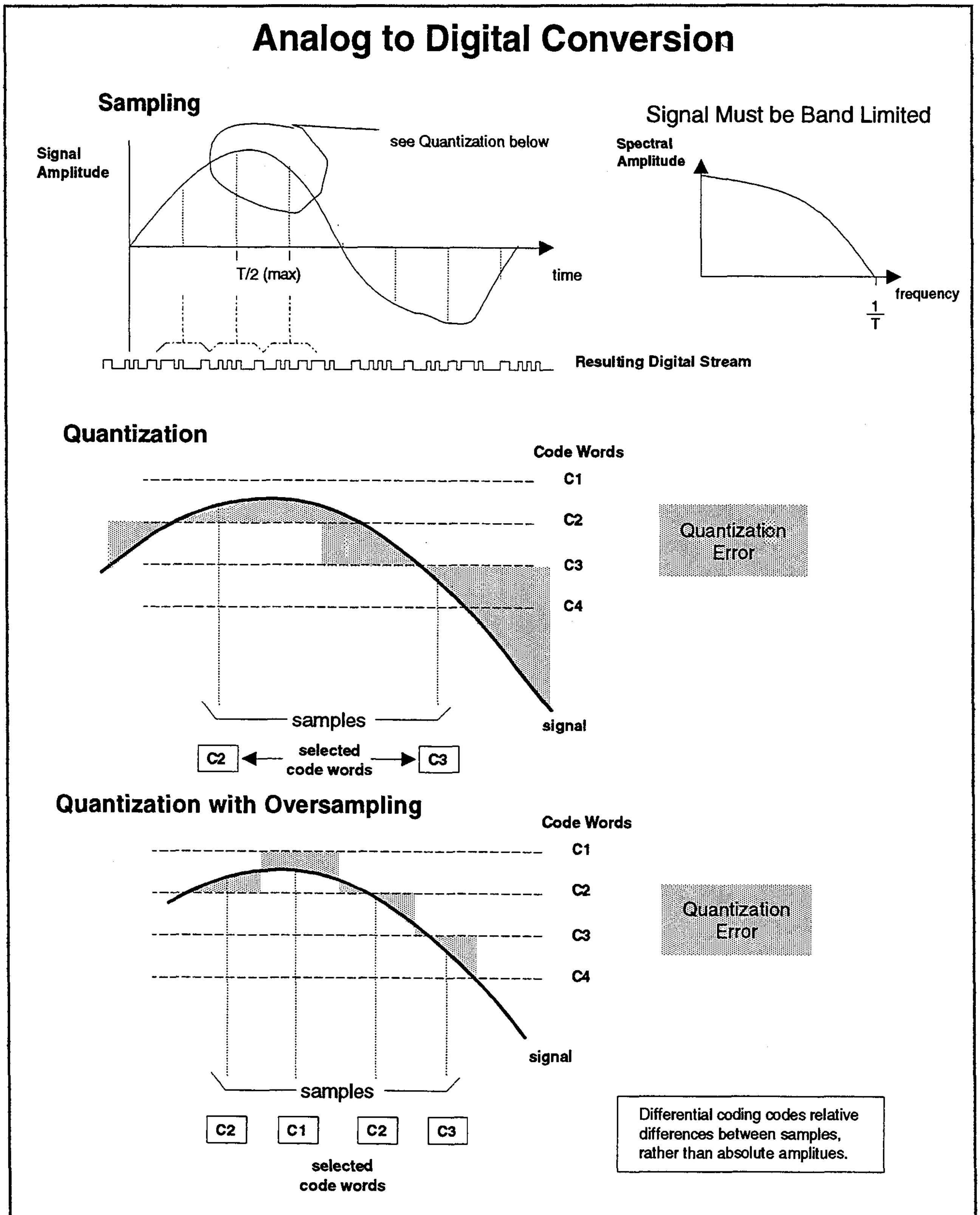

Analog to Digital Conversion
Sampling
Signal Amplitude
see Quantization below
T/2 (max)
time
Resulting Digital Stream
Signal Must be Band Limited
Spectral Amplitude
frequency
1/T
Quantization
Code Words
C1
C2
C3
C4
Quantization Error
samples
signal
C2
selected code words
C3
Quantization with Oversampling
Code Words
C1
C2
C3
C4
Quantization Error
signal
samples
C2
C1
C2
C3
selected code words
Differential coding codes relative differences between samples, rather than absolute amplitues.

Figure 1

rate. Taking into account over-sampling and dynamic range, equation 1 becomes:[5]

$$S/N = 6.02*N + 10.8 + 10*\log(f_s / 2f_v) + 20*\log[(V_W - V_B)/S] \text{ where}$$
$$f_s = \text{ sampling rate}$$
$$\text{and } f_v = \text{bandwidth of the source}$$
(Eqn 2)

<u>BIT RATE REDUCTION</u>

As described above, linear PCM generates a digital bit stream whose bandwidth is large relative to the signal being processed. For example, a "4 kHz" voice grade telephony channel becomes a 64 Kbps digital stream (8 bit linear PCM), an NTSC composite signal (6 MHz nominal) becomes an approximately 100 Mbps stream, depending on detailed coding parameters. In many applications, such as networked telephony, the robustness, ease of processing (switching, echo canceling, etc.) and low incremental cost of digital transmission have made this bandwidth increase not only acceptable but the superior choice from both quality and cost of service perspectives. The same can not be said for wide band signals, such as broadcast or cable video, where the available bandwidth of existing delivery system cannot support a simple analog to digital signaling

	NTSC Composite Signal				CCIR 601 Component YUV
Sampling Rate	12 MHz		14.36 MHz (4*fsub-carrier)		13.5:6.75:6.75 MHz
Bits/Sample	8	9	8	9	8
Rate (Mbps)	96	108	115	129	216
S (pk-pk) (digital words)	148	297	148	297	219
S (pk-pk)/N(rms)	54 dB	60 dB	55 dB	61 dB	58 dB

Table 1
S/N performance of Linear PCM for Video

Table 1 summarizes the S/N results, for typical signals and coder parameters that are in use or of potential interest in digital video. It must be noted that these results are ideal and do not address the degradations due to implementation difficulty. A detailed discussion of sampling and conversion degradations such as aperture effects (finite width versus impulse sampling), jitter on the sampling clock, ailiasing due to imperfect filters and digital noise effects is beyond the scope of this paper but covered well in several excellent references[6,7,8].

conversion.

Bit rate reduction techniques for digital signaling can be placed into two general classes—coding techniques such as Adaptive Differential PCM (ADPCM) and advanced compression techniques such as sub-band coding and transform coding. There is wide overlap among the various approaches, with simple coding improvements yielding reduction factors in the 2 to 4 range and advanced context sensitive techniques yielding improvements of 25 to 50 or more, depending on the source.

From a practical viewpoint, the simplest bit rate reduction techniques are similar to data compression techniques used with computer data where sequences of binary data are replaced with coded sequences. Differential coding techniques focus on transmitting the difference between successive samples (Figure 1) rather than the absolute sample amplitudes individually. Such techniques have achieved bit rate reductions in the 4X to 8X range in typical audio applications. The large (25X to 50x) reductions in coded video images, achievable with significant signal processing, are more fully discussed in Reference[9].

PAYLOAD FORMATTING

After successful conversion from the analog

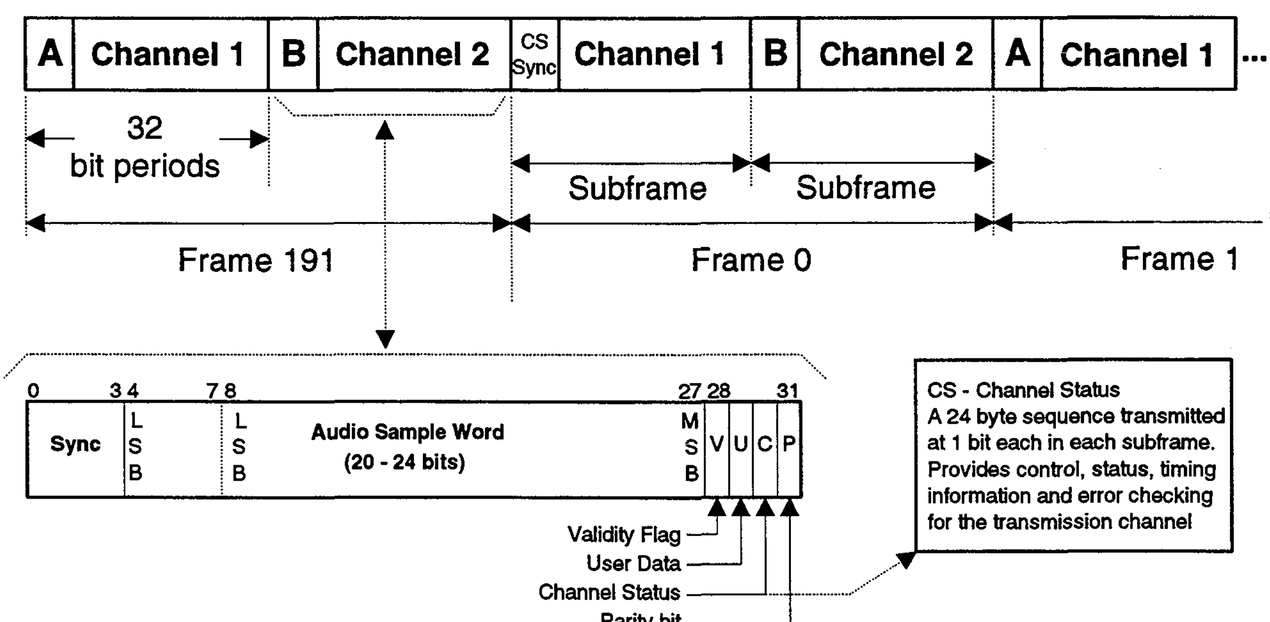


Figure 2 - Payload Formatting Example -- AES/EBU Digital Audio

domain to the digital domain, the next step in processing for transmission is to add to the encoded stream a series of synchronization and control signals which enable the receiver to properly decode the signal stream. Perhaps the simplest familiar payload is the asynchronous character transmission protocol used in typical asynchronous data interfaces. In this instance, an initial synchronization bit ("start bit") and one or two "stop bits" are added to the signal stream along with optional parity bits for error detection.

A far more complex example of a formatted payload signal is shown in Figure 2, showing the byte (8 bit) level structure of the AES/EBU digital transport format[10] used for high performance digital audio channels. In this case, one or two digitized audio channels are formatted into a fixed bit rate digital stream with predefined functional and padding bits such that a common transport format interfaces with the transmission system. The added control information in the 48 bit header field delivers to the receiver all information needed to decode and interpret the serial stream. Once the receiver has recognized the "unique" 12 bit syncword, it is able to read other control bits for decoding and error detection functions.

Formats such as Figure 2 provide for a universally accepted ("standard") framework for the transfer of payload information and associated control information. Similar standards exist for video streams (e.g. D1 parallel and serial interfaces in CCIR 649), data communications (e.g. HDLC (high-level data link control) data packet protocols) and are used extensively in voice communications systems in international telephony systems (e.g. DS0, DS1, etc. for the North American Digital Hierarchy). Strict adherence to open interface definitions makes it possible for equipment from various vendors to interconnect and transfer information effectively. The increase in apparent complexity of formats such as the AES/EBU format of Figure 2, compared to a DS0 telephony channel at 64 kbps, is a direct result of the availability of economical high performance digital processing LSI. An obvious benefit for the user of this format is its ability to adapt to several different specific payloads utilizing the same common interface. There is a general trend in such standard interfaces towards higher complexity, driven by the need for flexibility and enabled by economical real time processing power.

COMBINING AND MULTIPLEXING

In many cases, the available capacity of transmission facilities is significantly greater than that required by any individual service or communications system. In these cases, multiple formatted payloads are combined, or multiplexed, into a higher capacity signal for transport over the facility. In typical applications, the multiplexing system uses a small portion of the transport capacity for its own synchronization and control functions, similarly to the discussion of payload formatting above. The remaining capacity is then divided among the digital streams which share the channel capacity.

Several techniques are used to achieve the multiplexing step, synchronous and asynchronous time division multiplexing being the most prevalent at this time.

Asynchronous Time Division Multiplexing

Each channel operates at a fixed clock rate which is arbitrary (asynchronous) relative to the multiplex system clock. Information is transferred to the multiplex interface at a rate which varies but is always guaranteed (by design) to be less than the "allocated" capacity for that channel. The multiplex interface then bit stuffs or adds additional bits to the channel such that its instantaneous rate matches the allocated capacity. This technique, one of several in use, is commonly called positive bit stuffing. As part of the multiplexing process, control information must be conveyed to the decoder end of the link to enable proper removal of the stuffed bits.

Synchronous Time Division Multiplexing

In the case of synchronous multiplexing, the amount of data transferred between the digital source and the multiplex system is forced to exactly match the allocated capacity for the channel, typically by providing a clock signal from the multiplex system to the signal source (or, more typically, locking both the source and the multiplex to a common clock source). The synchronous multiplex has higher throughput (utilization of capacity for true information bearing applications) but typically requires a more complex timing distribution system.

Statistical Multiplexing

As multiplex processing power increases, it becomes possible to efficiently operate a statistical multiplex. In this case, the multiplex capacity is made available to all input channels on an "as-needed" basis and channels randomly send data to the multiplex. The multiplex system allocates available bandwidth based on a current need basis. This implementation depends on the average statistics of the combined signal sources to not exceed the total capacity of the multiplex. While managing and communicating the continuously varying channel capacity information to the demultiplex end of the system, the benefit of statistical multiplexing is that channels see higher capacity available on an as-needed basis. This capability is particularly effective in situations where the signal sources do not generate continuous information streams and are "bursty" in nature, requiring occasional periods of high capacity relative to the long term average.

Transmission

The transmission of a digital bit stream over a medium is performed by modulating a carrier signal with the digital information in the signal stream. The carrier might be an electrical signal, in the case of metallic media, or an optical signal for fiber optic signals. A simplified view of a digital link or regenerative section is shown in Figure 3. We use simple bipolar (alternate mark inversion) amplitude keying for this example, more extensive analysis of techniques can be found elsewhere[11].

The digital regenerator is comprised of a front-end amplifier, a carrier recovery or timing extraction stage, and a decision circuit which combines the signal and timing information to recreate the digital information. As shown in Figure 3, the transmission media introduces amplitude and edge degradation (in general) in the transmitted stream. Engineering rules specific to the application will specify the allowable loss in the media in order to ensure adequate margin at the decision circuit of the regenerator. Since most real noise sources are statistical in nature, the engineering rules will typically specify a minimum received power in

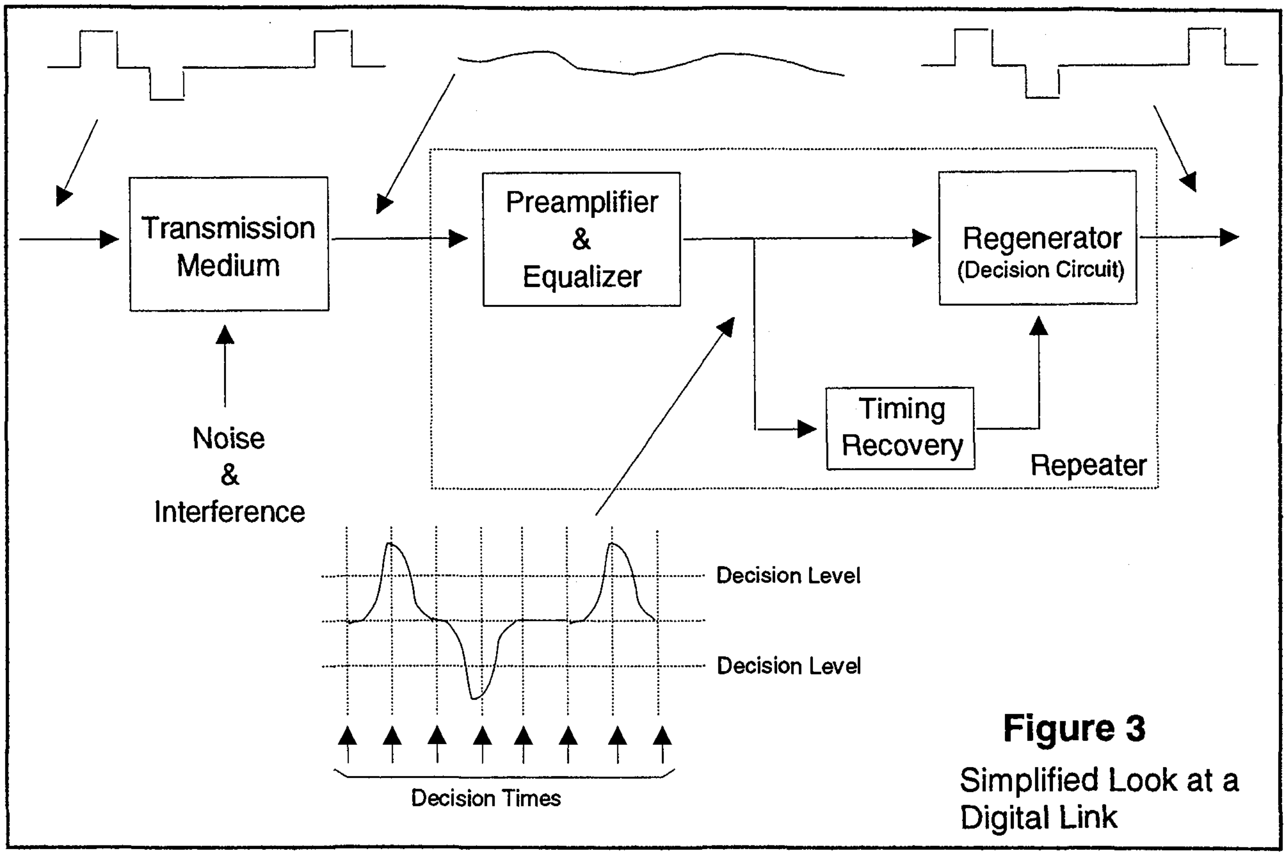


Figure 3
Simplified Look at a Digital Link

order to guarantee a particular error rate or probability of error.

Typical digital transmission systems consist of several concatenated regenerative sections, often with different media (wire, fiber, RF carrier) intermixed in the end to end connection. The dominant performance metric is the probability of error or error rate expected for the link. No link is completely error free (statistically), hence signal sources must be capable of properly handling error occurrences.

Binary baseband signaling has been the predominant format for digital transmission, particularly in fiber optic systems, due to its simplicity and robustness. For cable applications, however, multilevel transmission techniques based on passband channels and MODEM technology are popular due to the nature of the delivery channel and the need for compatibility with existing passband analog services. In a simplified view in Figure 3, we look at 4 level signaling. In this case, two bits of transport information are converted to one of 4 levels ($00_2=0, 01_2=1, 10_2=2, 11_2=3$) and the appropriate level transmitted and detected by a four level decision circuit. Such a system requires significantly better noise performance than the binary signaling system but achieves higher throughput per unit of bandwidth. More on cable applications in the following sections.

DEMULTIPLEXING AND PAYLOAD SEPARATION

The process of recovering individual bit streams at the receive end of the system is performed by continuously scanning the signal stream for the format or synchronization patterns (frame words) inserted in the multiplexing process, using the embedded control information to interpret the bit stream and routing the individual channel streams to their appropriate connections. An important consideration in the demultiplexing process is robustness or immunity to errors in the digital stream.

In systems that use fixed frame word spacing, word detectors typically "fly wheel" on the frame pattern, requiring that several frame words be detected with errors before initiating a new frame search, thus minimizing the effects of random errors. Systems that use variable synchronization spacing generally require a lower (improved) bit error rate performance from the transmission channel in order to achieve equivalent performance.

DIGITAL TO ANALOG CONVERSION

The final step in the process, assuming an analog interface to the end user (speaker, analog TV) is digital to analog (D/A) conversion. This process consists of reconstructing the sampled impulse stream from the digital code words, converting the code words to impulse amplitudes and low pass filtering to recover the original signal envelope.

DIGITAL SYSTEM DEPLOYMENT

With the above digital overview as background, the impacts that digital technology will have on CATV system infrastructures will be examined. The traditional tree and branch system architecture has serviced the industry well for almost four decades. In fact, this point to multi-point network is a basic strength of the cable industry. The delivery of entertainment services has required large amounts of bandwidth in the downstream direction from the headend to the subscriber.

This will continue to be the case in the future as cable networks evolve from just entertainment networks to telecommunications networks. What will change as these networks evolve is that lightwave systems will be increasingly deployed deeper and deeper into the system in a fiber to the service area scenario. This will further reinforce cable system bandwidth expansion, setting the stage for expansion into digital technologies discussed above.

The cable television industry has become comfortable over the years with FDM (frequency division multiplexing) technology as a convenient means to allocate spectrum on a cable system. As digital technologies become available to the cable industry, it seems logical to continue using FDM techniques to apportion spectrum for both analog and digital services. This will allow a hybrid analog/digital structure to exist well into the future to support both types of services.

In a paper titled "The Evolution of CATV to Broadband Hybrid Networks", by Carl J. McGrath, AT&T Bell Laboratories, which is being presented at this conference, the author discusses possible methods of implementing a hybrid analog/digital headend architecture.

The cable system forward bandwidth is divided into a lower frequency portion for analog signals and a higher frequency portion for digital signals (compressed digital video/audio and other data signals). Digital "bandwidth" is added to the system as needed to support digital services, allowing for a modular expansion capability by utilizing digital modem technology.

MODERN MODEM TECHNOLOGY

As discussed earlier under "Transmission", the digital signals must be modulated onto an RF carrier to be carried on a broadband cable television system. The last 10 to 20 years or so have produced an enormous improvement in modem technology in two general classes: power efficient modems and spectrally efficient modems.

Power efficient modems generally are described as those modems which produce less than 2 b/s/Hz (bits per second per Hertz). They are used in power-limited situations such as existing satellite communication systems and digital cellular radio systems.[12]

Of more interest to the cable industry are spectrally efficient modems which produce more than 2 b/s/Hz. In fact, the range is from 2 b/s/Hz all the way to 10 b/s/Hz for the most exotic technology (1024 QAM). (See Table 2.) Likely candidates for cable applications are M-ary QAM, 4-VSB AM, and QPRS (quadrature partial response systems).

Modulation technique	Nyquist rate theoretical efficiency bits/Hz	Practical efficiency bits/s/Hz	C/N Required for Pe = 10 (theoretical)
QPSK	2	1.2-2.0	15.0
4-QAM			
9-QPRS	2	2.0-2.8	17.5
8-PSK	3	2.5-3.0	20.5
16-QAM	4	2.5-3.5	22.5
49-QPRS	4	2.5-4.3	24.5
64-QAM	6	4.5-5.0	28.5
128-QAM	7	4.5-5.5	31.5
225-QPRS	6	5.7-6.3	31.0
256-QAM	8	5.0-7.0	34.5
512-QAM	9	5.5-7.5	37.5
961-QAM	8	6.0-8.0	37.0
1024-QAM	10	7.0-9.0	40.5

Table 2

Spectral Efficiency and C/N Requirement of Wideband Modems

Adapted from K. Feher, "Advanced Digital Communications"

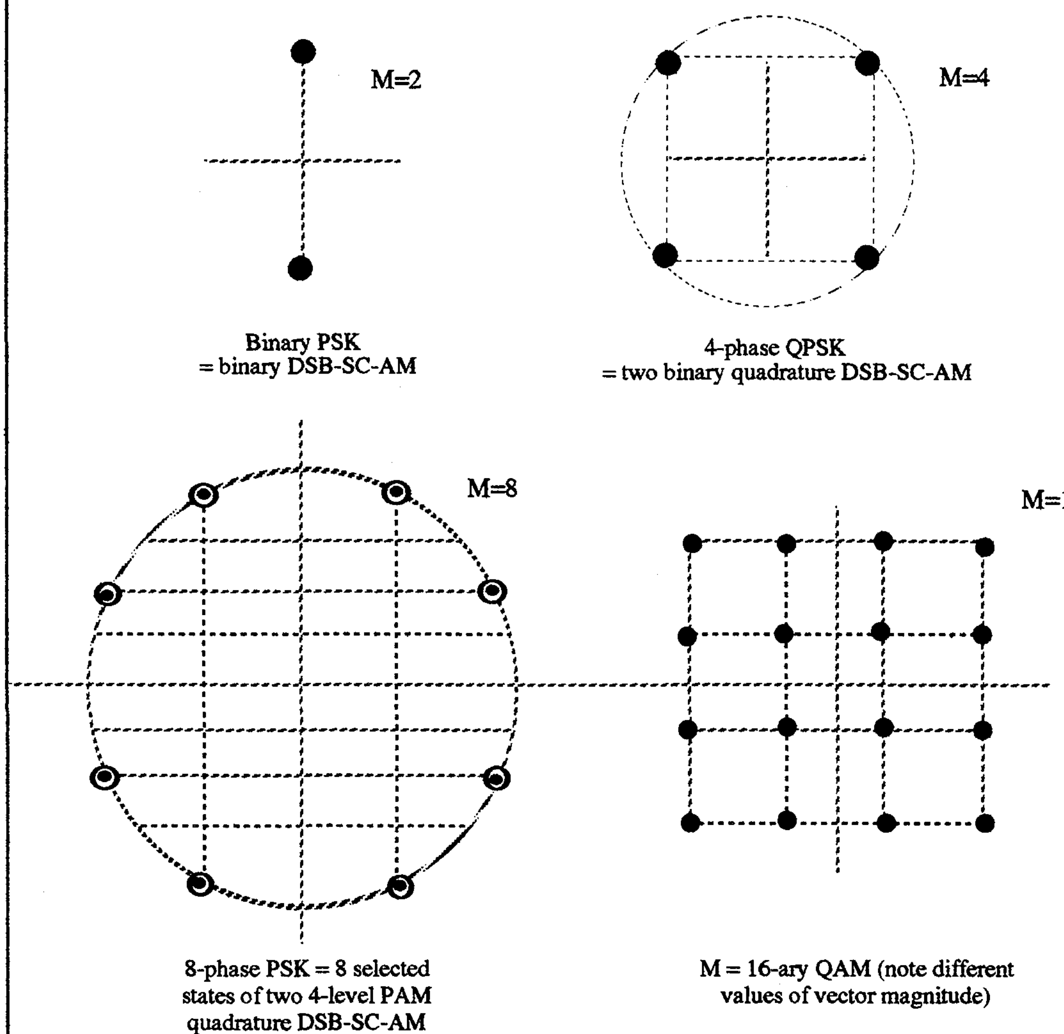


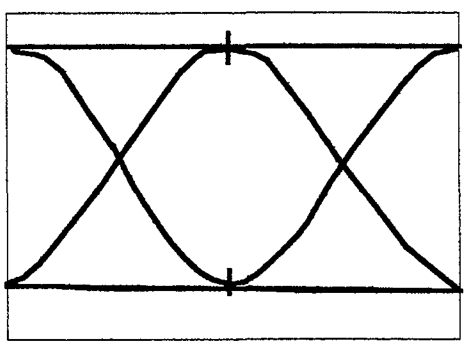
Figure 4

Signal-State Space Diagrams of M-ary PSK and QAM Signals

From K. FEHER "Digital Communications: Microwave Applications"

When dealing with these modem technologies, two of the concepts involved are vector state space diagrams (Figure 4), also known as constellation diagrams, and eye diagrams.[13] A vector state space diagram is a display of all possible trajectories of the carrier — as it passes through each of the possible phase and amplitude states. For example, the constellation for 16 QAM shows 16 possible states of phase and amplitude of the carrier.

An eye diagram is formed by applying the data signal to the Y axis of an oscilloscope, externally triggering with the system clock, with the sweep time set approximately equal to the symbol time. (See Figure 5.) An eye diagram is a useful tool in analyzing system performance since by measuring the percent of eye height in the demodulated received data, one can determine the likelihood of the system meeting the design goal error probability (Pe).



Idealized Eye Diagram

Figure 5

COEXISTENCE/PERFORMANCE ISSUES

Now that the digital signals are packaged for transmission, the question of compatibility arises. Will the digital signals interfere with the analog signals? Conversely, will the analog signals interfere with the digital signals? How robust will various forms of digital modulation be on cable systems in the presence of phase noise and reflections? Should digital channels be multipexed into one high-speed digital stream which requires high performance circuitry at the subscriber location? Should digital channels be packaged individually like analog to ease the decoding costs and complexities in the subscriber terminal? These questions are just some of the issues being investigated by various laboratories both at the national and international levels.

TEST AND MEASUREMENT ISSUES

With the advent of digital transmission on cable systems, there will be some tests required which are not at present in the repertoire of cable system tests. Such tests might include one or more of the following:

1) Phase noise
2) Phase delay
3) BER (Bit Error Rate)
4) Eye height of demodulated data

Some testing will be required at the headend on satellite delivered digital signals as well as on signals delivered by digital headend interconnects. Testing locations should be identified which are representative of subscriber locations which are "deepest" in system cascades whether it is an all coaxial plant or a hybrid fiber/coax system. Some specialized digital test equipment will undoubtedly be required for the cable television analog/digital environment.

The goal here is to maximize the height of the eye diagram for the demodulated data so that the BER is maintained to system design standards. While in analog pictures, phase noise might result in graininess and reflections might result in ghosts, they both can cause the eye diagram to partially close, resulting in difficult digital data detection. As long as the system design goal BER is maintained, the digital transmission process will not degrade the SNR of digital video images as measured at the headend.

It is important to note that as the CNR of an analog picture degrades, it results in a progressively noisier picture. In the digital domain however, once the BER degrades below a minimum threshold value, digital data detection may not be possible. At this point, the subscriber's decoding equipment would most likely result in a freeze frame display of the last digital video frame which was accurately received.

SUBSCRIBER EQUIPMENT ISSUES

Once the analog and digital signals get to the subscriber's location, several questions must be answered about the topology of the subscriber's distribution "system". Depending on what method of securing analog signals is being used in the cable system (positive or negative trapping, interdiction or scrambling) there will be subscriber equipment located at the tap, on the side of the house, or at the TV set.

Are any of these locations the appropriate place to locate the digital circuitry necessary to receive, demodulate, demultiplex, decrypt, and convert from digital to analog the digital signals appropriate to a particular subscriber?

If the tap location is chosen because of possible cost sharing in a multi-output device, there are considerations of powering and sensitivity to temperature swings and other elements of the rather hostile outdoor environment. In some of the newer feeder designs utilizing "superdistribution" techniques, there is no power in the tapped feeder cable so that drop powering would have to be used.

A location on the side of the house would still be subject to all the conditions of an outdoor environment, would be powered from the home, and would service all outlets in the house. The configuration would provide for modular implementation, allowing "service - specific" cards to be plugged in on an as-needed basis.

A variation of the side of the home approach would be a unit designed for an indoor environment in some out-of-the-way location (such as in a closet or in the basement). Obviously, an indoor location would not work for interdiction of analog delivered services since the signals are in the clear on the drop coming into the unit, but eliminates outdoor environmental issues. Descrambling of analog signals could be also performed for all outlets at this central location.

A top of the set or back of the set location requires a box for each TV set to which any secured services (analog or digital) are to be delivered. A multiport equipped TV or VCR could utilize a multiport decoder for secured analog services but a separate digital receiver would be needed for digital services. The output of the digital receiver would be both RF (channel 3 or 4) to accommodate standard TV sets and baseband audio and video to feed a receiver/monitor.

Wherever the digital receiver is located, it would be helpful from a system troubleshooting viewpoint to have a demodulated data port available so a service technician could plug in a BER meter. By comparing the reading to the system design goal BER, the technician could quickly determine if there was a problem at the particular location with the received data.

CONCLUSION

An overview of digital transmission fundamentals has been presented, as well as a discussion of some implications of digital deployment in a hybrid analog/digital cable network.

As fiber gets deployed deeper into the network in various star-bus architectures serving smaller numbers of homes (less than 1,000), the industry will be in a good position to offer near-video on demand services and other personalized communications services.

Modem technologies will be further refined for the cable environment, and coupled with the improved carrier to noise ratios provided in the smaller service areas, will allow higher constellation state digital modulation techniques to be applied (e.g., 64 QAM). The result will be higher bandwidth efficiencies allowing for narrowcast, more personalized services to cable subscribers.

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