DIGITAL AUDIO FOR CABLE TELEVISION

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ABSTRACT

Digtal Audio provides unmatched signal security. Sixteen Bit Linear Pulse Code Modulation offers the highest level of audio performance, but uses the most bandwidth and is the most expensive digital audio system. For a premium audio service, it is still the best choice. Digital audio for video programs is best served by an inband system. Dolby Deltalink Adaptive Delta Modulation offers the best bandwidth efficiency of the companded digital systems and provides very high audio quality as well as low cost. Using Deltalink ADM, BTSC can be replaced with a secure digital transmission. QPSK modulation of the sound carrier is preferrable to in video ASK on a cost basis.

INTRODUCTION

There are two primary applications for digital audio on cable. The audio portion of a video program may be digitized. Premium audio programming independent of video is another ideal application for digital transmission.

Digital audio has two primary advantages over competing analog systems. The signal security provided by numerical encoding techniques is unmatched in analog systems. Digital transmission allows the identical audio quality to be received as is transmitted, independent of cable plant variation. Digital audio services can be much more forgiving of thermal noise and distortion than analog modulated video and audio signals. In other words, service calls will be generated from poor pictures before a digital audio service is affected.

DIGITAL SAMPLING SYSTEMS

Digital sampling systems are numerous. These systems are different methods of converting from a continuous analog audio signal to a digital data bit stream. The basic trade-off in these systems is audio quality vs. data bit rate.

Linear Pulse Code Modulation (LPCM)

LPCM is the sampling technique used for Compact Disks (CDs). The CD sampling rate is 44 KHz with 16 bit binary resolution. There are 65,536 evenly spaced points where each audio sample can occur. The Signal to Noise ratio (S/N) of LCPM is equal to the dynamic range and is 6 dB/bit. For 16 bit LCPM, the CD case, the S/N is 96 dB. The resultant bit rate is 2 channels x 16 bits x 44 KHz sampling rate = 1408 KB/S.

Assuming a 25% data overhead factor is required for control, data framing, and error correction purposes at the PCM receiver, the final transmission bit rate for CD quality LPCM becomes 1408 KB/S x 1.25 = 1760 KB/S. A T1 telephone modem, which carries 24 voice channels, can handle only 1544 KB/S.

Because of the incredibly high bit rate for CD quality LPCM some type of compromise is often considered. Two audio properties should be considered here. Low frequency audio needs greater signal amplitude handling capability than high frequency audio does.

Noise 96 dB below a signal level is inaudible. 30 dB higher noise is also inaudible. CD quality LPCM allows for greater S/N than is required, as well as greater high frequency amplitude handling capability than is necessary.

In an effort to reduce bit rate a usually accepted practice for transmission is to limit the audio bandwidth to 15 KHz, alowing a 32 KHz sampling rate. For 16 bit LPCM the new transmission bit rate becomes 2 x 16 x 32 x 1.25 - 1280 KB/S. This is still a very high bit rate. As a result further techniques are often used to reduce bit rate.

Instantaneous Digital Companded PCM (IDC PCM

IDC PCM or non-linear PCM uses progressively increasing step sizes. The distance between steps for high amplitude signal is much larger than the distance for low amplitude signals. Commonly used curves are μlaw and Alaw. The result of this type of sampling is an increased dynamic range for a given number of bits compared with linear PCM, at the expense of reduced instantaneous S/N. The noise level tracks the signal amplitude, increasing as the signal increases. If this compression technique is extended too far, the noise will become audible. IDC PCM does provide the possibility of reduced cost by allowing lower precision Digital to Analog (D/A) converters. If 10 bits are retained, the transmission bit rate will be 2 x 10 x 32 x 1.25 = 800 KB/S.

Block Companded PCM (BC PCM)

BC PCM uses storage of LPCM for a fixed time interval to determine the maximum bit level that occurs within that time interval or block. Only the desired number of bits downward from the maximum in the block are retained for transmission. Data is carried to the receiver indicating what bit range is to be used for each block. This range data must be extremely well protected against errors, as the errors will have a long lasting effect. If too few bits are retained, the noise will become audible. 10 bits is considered an acceptable limit so the bit rate is 800 KB/S.

Analog Companded PCM (AC PCM)

AC PCM uses an analog compander in front of a PCM Analog to Digital (A to D) converter. Bit rate and cost are saved by using a lower precision A to D converter. Both variable gain and variable pre-emphasis may be used to minimize the apparent noise level. A disadvantage of this system is the transient response errors which analog companders exhibit from changing gain after the signal changes, rather than looking ahead as BC PCM systems do. Some systems (such as the digital audio used on 8MM video tape) combine analog companding and instantaneous digital companding to reduce the bit rate to 8 bits per sample. For 15 KHz audio bandwidth with 8 bits/sample, the transmission data bit rate is 2 x 8 x 32 x 1.25 = 640 KB/S.

Dolby Deltalink Adaptive Delta Modulation (ADM)

Deltalink ADM differs from PCM in that sampling occurs at a rate much greater than twice the maximum signal frequency. Each sample can take on only two levels. The sample or step is really a correction signal which indicates whether the real time average of the preceeding steps is larger or smaller than the present audio level. The audio is recovered by integrating the series of steps which make the data stream. Companding is used to vary the size of the steps. In the Deltalink ADM system the comanding is similar to that of block companded PCM. Storage is used to look ahead so that transients are not distorted. The step size control signal is converted to a delta-sigma data stream and inserted with the audio data stream. This data is evenly weighted and has a very low bandwidth so that it does not require elaborate error protection like block companded PCM.

In addition to step size companding variable pre-emphasis is used to maximize the instantaneous S/N. The variable pre-emphasis control is handled in the same manner as the step size with look ahead storage and data inserted in the main delta mod data stream.

Deltalink ADM has two distinct advantages over PCM systems. The first is lower cost audio reconstruction. Because of the high sampling rate no brick wall filter is required. No precision D to A converter is required. Less logic is required to reconstruct the audio signal. At the receiver no block memory is required. Because all bits are of equal significance in delta modulation, much higher error rates can be tolerated than in PCM systems. With the relatively high C/N available in cable systems by data transmission standards, delta modulation requires no error correction or its associated logic and framing memory.

The most significant advantage of Deltalink ADM over PCM systems is the lower bit rate required. Because error correction is not required, the data overhead factor can be 10% rather than 25%. The data bit rate for a stereo audio channel with 15 KHz bandwidth, 85 dB dynamic range and 60 dB S/N is 2 x 208 KHz x 1.1 = 458 KB/S.

TRANSMISSION TECHNIQUES

There are numerous data transmission systems which could be used for digital audio over cable. Three systems will be considered here which use bandwidth efficiently and are implemented at reasonable cost.

1. QPSK Out of Band

Quadrature Phase Shift Keyed (QPSK) modulation of a carrier which is not part of a video program channel is suitable for premium audio services. QPSK with non-return to zero (NRZ) coding gives two bits per hertz bandwidth efficiency (bit and frame sync are already accounted for in the overhead included in the final bit rate calculations). A 1.5 bandwidth factor for interference to analog signals is practical.

In addition to bandwidth efficiency, QPSK has excellent noise performance. If a QPSK signal is carried 20 dB below video carriers and a noise bandwidth of 1 MHz is assumed, then a worst case situation of 36 dB video C/N would result in a QPSK carrier to noise of 22 dB. The probability is extremely low that a bit error will occur under these conditions due to thermal noise in a life time. Outages are more likely to interrupt a QPSK digital audio service than poor C/N.

NRZ QPSK out of band can handle all the types of digital audio sampling because the bandwidth used need not be backward compatible with analog channel spacings. The most preferrable sampling method in this case is 20 KHz bandwidth 16 bit linear PCM. A channel spacing of 1.5 time Nyquist bandwidth is assumed, giving (1760KB/S ÷ 2) x 1.5 = 1.32 MHz for NRZ QPSK. This would allow only four audio channels inside a video channel, but it would provide CD quality directly into a subscriber's home. Cable is presently the only medium where CD transmission to the home is possible.

Out of band QPSK used for video program audio presents both financial and operational difficulties. A separate addressable digital audio converter cannot compete with a single converter for video, analog and digital audio. The combined converter shares tuning, addressing and display functions, etc. In the future, if all video channels require multi-channel sound and there are numerous channels of premium audio service, the available bandwidth for out of band QPSK will disappear rapidly.

2. Time Division Multiplex (TDM) Vestigal Sideband (VSB) Amplitude Shift Keying (ASK)

TDM VSB ASK may be used in the video Horizontal Blanking Interval (HBI). Assuming 8.25us out of 63.5us would be usable for data and a 7.16 MB/S bit rate (2 tims the color subcarrier frequency) were used with NRZ coding the available effective bit rate would be 930 KB/S. Block companded PCM or Deltalink ADM would be usable with the available bit rate.

A disadvantage of high speed TDM VSB ASK is the stringent frequency and group delay response requirement over the whole transmission path, including the modulator, cable plant and converter demodulator. Another disadvantage is the cost of reconstructing proper video horizontal, vertical and color sync for output to the TV.

3. QPSK Sound Carrier

A QPSK Sound Carrier is possible with an interference bandwidth of 500 KHz so that 4.2 MHz video and -1.25 MHz VSB from the upper adjacent channel are not disturbed. The available bit rate for QPSK NRZ is (500 KHz ÷ 1.5) x 2 = 667 KB/S. According to the previous calculations there are two types of sampling systems which have low enough bit rates. The systems are PCM companded to 8 bits and Deltalink ADM. Deltalink ADM is preferrable to PCM on a cost and error immunity basis.

A desirable feature would be to have an inband digital audio system capable of a stereo channel and a monaural channel simultaneously so that the digital audio could be used as a replacement for BTSC. Further investigation will show that this is only possible for the available bit rate with Deltalink ADM.

CONCLUSION

The two markets for digital audio can best be served by two different digital techniques. A premium audio service should be truly premium, with unsurpassed performance even if it is an overkill. CD compatible PCM should be used for premium audio.

Video program audio should be inband for convenience and cost purposes. Deltalink ADM is the only system which can fully replace BTSC with security and high fidelity.

References

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