PRE-DIGITAL CONTROL and VIDEO GAIN CONTROL

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The digital revolution has brought about many changes in the way signals are stored and transmitted. With this new age of DIGITAL SIGNAL TRANSMISSION come new problems that must be solved.

The world around us is an analog one. The human body experiences everything with infinite levels. Light is not just on and off, but comes in varying levels of brightness. We hear changing levels of Sound, Heat, Pressure, Motion all come in infinite levels. This article will focus on Audio and Video signals and the problems associated with digital transmission.

All Audio Video signals begin their journey as an analog signal. All audio and video signals start out as changing sound levels and changing light levels, both of which are analog in nature.

The first process that occurs in any digital processor is the conversion of the analog signal into a digital signal. This is accomplished with an Analog-to-Digital converter (A/D). The A/D is an integral part of every digital product and is directly connected to any incoming analog signal. The A/D converter samples the incoming analog signal and assigns a digital number to each level it detects, provided the level does not exceed the highest number that can be assigned.

The designers of digital devices must choose a sampling rate and maximum sample size that best fits the signal type. They balance this against the complexity and cost of manufacturing the product. This means that they must make a trade-off when designing the equipment. Often that trade-off causes the equipment to over-load with real world signals.

This brings us back to the analog signal. In the real world these signal levels vary greatly; occasionally the levels are extrodinarily high and at other times very low. If the designers built the equipment to handle the extrodinarilly high signals that occasionally occur, the price and complexity of the digital process would be too expensive to sell.

So occasionally, the digital equipment will fail to transmit the signal momentarily when the level is extraordinarily high. This is caused by an over-load created by the D/A converter in the digital equipment.

One example of video digital overload is seen as a "comet tail" or “streaking” to the right side of the bright part of a video picture, diminishing as it moves to the right side of the screen. The bright spot in the video caused the digital system to reach its maximum number and over flow the highest number possible. The recovery time produces the "comet tail". In other equipment video overloads or "number overflow" will cause break up of the video picture.

This is especially true with any video compression system. You can experience "tiling" or complete failure of video transmission during these high level periods.

If you set the input signal down below the standard input level to allow for more headroom before overload, then your signal will have greater noise all of the time. Some times the signal will be a very low level, which causes the signal to be closer to the digitizing noise floor, and makes the signal much noisier. To solve this, the incoming level should be automatically raised to nominal level during “low level” inputs.

Similarly the level should be automatically lowered during “high level” signals to prevent the overload. This level control must be done before the signal enters the digital equipment, before the A/D converters inside the digital equipment. This is referred to as PRE-DIGITAL CONTROL.

The basis of PRE-DIGITAL CONTROL is to regulate the audio and video as analog signals before the digital system receives the signal so that all signals presented to the digital equipment will have regulated, controlled levels. This will prevent digital noise caused by too low a level and eliminate over-load due to high level signals.

Level control features built-in to the digital equipment cannot prevent the overload and under level noise problem, because the A/D converter is ahead of any control features.

Using True Pre-Digital Control will help to maintain the digital signals integrity and prevent the most common failures of the digital transmission system.

Requirement for an audio and video PRE-DIGITAL CONTROL system should include:

AUDIO:

Automatic control of audio loudness over 30 dB range.
Automatic Gating to prevent "pumping" of background sounds.
Program-Dependent Time Constants to prevent "ducking".
Dual-Band control system for both high and low frequencies.
High frequency overload control system to prevent "S-ing".

VIDEO:

Handle signals video from 0.5 Vpp to 2.0 Vpp.
Automatic SYNC control to 40 I.R.E. units.
Automatic Luminance control to 100 I.R.E. units.
Automatic Chrominance Equalization.
Automatic DC Restoration (Clamping).
Automatic 60 Hz elimination (Hum Bucking).

To start with an analog signal requires a certain amount of bandwidth to transmit the signal. A digital signal requires at least 5 times as much bandwidth to transmit the same signal.

ARE DIGITAL SYSTEMS NECESSARILY BETTER?

Good digital disc players can play back recorded music with fidelity unrivaled by any other home type tape or record system. In fact the improvement over LP discs or commercial tapes borders on the spectacular. No wonder then that many people are lead to believe that if it is digital, it is necessarily better and that it would follow that if digital recording is better, then digital transmission must also be better than analog transmission. But is this really so, are recording and transmission considerations the same? It turns out that for the transmission of music, nothing could be further from the truth.

RECORDING vs TRANSMISSION

Why should transmission be any different than recording when deciding between analog and digital processes? Both are degraded by noise introduced within the medium, but in recording no one really cares what power density or bandwidth is required to lay down and retrieve the music from the record medium, whereas the power density and occupied bandwidth are a price concern when transmitting the music on cable systems.

DIGITAL vs ANALOG TRANSMISSION OF AUDIO

To compare digital with analog transmission, you must first construct a level playing field, this can be done by requiring each process to:

1. Occupy the same time duration to play a given piece. Here we will consider music played in real time, but "time" expansion and contraction would be OK as long as the same rules applied to both digital and analog.

2. Signal power and injected noise level must be the same for each.

3. The occupied bandwidth must be the same for each process.

Examining past comparisons between analog and digital transmission systems, we find that where a digital system was declared to be greatly superior, we also find the digital system typically occupying one or more orders of magnitude greater bandwidth than the analog system it is being compared with.

BACK TO FUNDAMENTALS

When it comes to comparing various transmission processes, information Theory is as fundamental as you can get, and is the best place to start. This paper is not intended to be a tutorial on the subject of information theory, so I will not trot out statistical theory or even write down one dazzling formula, instead I will extract a fundamental theorem that applies directly to the subject at hand, to wit:

Given that each of two transmission processes are constrained to the same power level, and
Given that each must encounter the same noise level in transmission (equal Carrier-to-Noise Level), and also
Given to occupy the same transmission band width, and also further
Given that the two processes have equal signal energy distribution efficiencies, then
Consequently both systems will have an equal demodulated signal to noise ratio.

In other words, on a level playing field, there is no "digital advantage" at all, and in fact neither system is superior, merely because of the type of process. We must be more discerning and look carefully at the specific advantages and disadvantages of each system.

PRACTICAL CONSIDERATIONS

What we are left with is a choice between two systems that are basically equal, given equal transmission parameters. However, the digital system is by nature a wide band beast, while analog systems are at home in either narrow band or wide band channels. Let us explore the performance of the two systems in WIDE BAND and NARROW BAND transmission facilities.

WIDE BAND CHANNELS are hereby defined as transmission channels two or more orders of magnitude wider than the base-band bandwidth of the signal to be transmitted.

Digital transmission is in its natural home in the wide band channel, for transmission in such a channel, the signal can be sampled sufficiently higher than the minimum Nyquist rate to ensure transmission of the highest modulating frequencies without aliasing distortion and also a sufficient number of sampling levels to reduce quantizing noise and minimize distortion at low modulation levels. At these bandwidths the digital systems are capable of Super Quality audio transmission.

Analog frequency modulation can also spread the base-band bandwidth efficiently over the same bandwidth as the digital systems, and will result in similar received signal to noise ratios. What differences could be measured between the two processes would only reflect the differences in evenness of energy distribution over the pass band that each system produced. The system that spread the signal energy most evenly over the allocated bandwidth would produce the best signal to noise ratio. Both transmission systems are capable of similar energy distribution, so differences would be minimal.

NARROW BAND CHANNELS are hereby defined as being less than two orders of magnitude wider than the modulating signal (but not less than the base-band bandwidth for real time transmission).

Analog transmission has special advantages in narrow band channels, since there need be no impairment of the signal as there will be in digital systems that must use sub-optimal analog-to-digital (A/D) encoding to narrow transmitted bandwidth appreciably.

Once constrained to occupy a narrow band-width a digital system must sacrifice some signal fidelity, while an analog system does not need to limit signal fidelity.

Typical techniques employed in digital systems to reduce occupied bandwidth include:

1. Bit compression, wherein one transmitted bit does not represent one of two data states, but one of four states, or one of eight states, etc. The greater the compression the less bandwidth required to transmit the signal. However there is a price and that is that the power of the signal must be increased to overcome the greater noise susceptibility of the multistate transmission process that is required for this bit compression. There goes the level playing field.

2. Use fewer sampling levels, and use a shorter word length. However this causes increased quantizing noise and also greater distortion at low music levels, where only a few quantizing steps must accurately reproduce the complex audio waveform.

3. Use a lower sampling rate. The nyquist rate (of twice the highest frequency to be transmitted) sets the absolute lower limit, but in practice almost all digital systems already use the lowest practical rate anyway, so very little is to be gained by applying this maneuver.

4. Use some form of digital companding. This usually takes the form of gradually increasing quantizing step sizes, with very small steps near zero voltage and rather large steps near 100% modulation. This has the advantage of reducing the word length needed to describe a sampled voltage, thus reducing the band width required to send the signal, while at the some time keeping the quantizing noise and low volume level distortion low during soft music passages.

Unfortunately there are side effects. For one, distortion at high music levels is now greater due to the much larger quantizing steps near 100% modulation. Second, low level, high frequency components riding on low frequency high level signals (that being the condition of most musical signals) are only reproduced near zero crossings, with the high frequencies chopped off as the low frequency sound nears the negative and positive peaks. This is because the sample step size exceeds the amplitude of the high frequency ripple, and is not recognized by the sampling process. The result? The violins are turned on and off at the low frequency rate. Not exactly what the composer had in mind.

5. Use analog companding followed by linear encoding. This process holds some promise of success by combining some of the analog advantages with digital processes. The author is not aware of any operating systems of this type.

6. Stop trying to define the exact voltage at each sample interval with a binary word. In other words abandon pure digital transmission for a process such as Delta Modulation. This is a sort of halfway house between pure digital and pure analog.

In this process the delta modulator waits until the audio waveform departs from its previous voltage by some specified delta increment (or decrement). At this time a pulse is sent to increment or decrement the voltage at the receiving point by that amount. The faster the waveform changes in time the more pulses must be sent. Many variations on this basic process are possible that help to reduce the total bandwidth.

The best known delta modulation scheme is the Dolby Digital Audio System. This system is very complex, in that the signal must be carefully analyzed prior to encoding, resulting in a very expensive transmitter, but a fairly inexpensive receiver. Super Quality Audio can be achieved with this process at bandwidths intermediate between narrow band and wide band. This process is not entirely digital and not entirely analog, so it is left to the reader to decide which category it belongs in.

7. Reduce the redundancy in the music prior to transmission and hope that the missing pieces of redundancy won't be missed. On a coarse basis, there is much redundancy to be exploited in music, so perhaps a great reduction in bandwidth could be obtained this way. It all depends on how much you think you can tinker with the signal without also changing the timbre and fine nuances in the music. Certainly the melody and basic harmony would remain, but I doubt that the composer would be pleased with the result if enough processing of this sort were done to reduce the bandwidth significantly.

CONCLUSIONS

Information theory tells us that if occupied band width, Carrier-to-Noise Ratio and signal energy distribution are equal, all modulation processes result in the same basic signal to noise ratio.

If very wide transmission bandwidth is used (approximately 100 times the base-band band width) either digital or analog modulation processes can provide Supper Quality Stereo.

Where it is desirable to transmit stereo over narrow band channels (approximately 10 times the base-band band width), it is necessary to somehow modify the digital or analog modulation technique to accommodate the bandwidth reduction.

The techniques available that enable narrow band transmission of digital signals all operate on the instantaneous shape of the musical wave form, and tend to be perceived as some form of distortion to the ear on the continuing basis. Depending on the digital bandwidth reduction system employed, either background (digitizing) noise increases, high level distortion or low level distortion increases, or some change in the character of the music is noticed by the listener. The grater the bandwidth reduction, the grater the degree of distortion perceived. This is true even for digital bandwidth reduction systems that seem to test very well with simple sine wave test.

The bandwidth reduction techniques available, using modern analog companding, operate on the RMS or average level of the music waveform, with no wave shape alteration during passages of relatively constant amplitude. There is no constant level of distortion as would be perceived in a digital bandwidth reduction system. Instead, the artifacts of analog bandwidth reduction occur only during musical level changes, not for the entire duration of the music passage. Therein lies the principal advantage that analog has over digital in narrow band transmission systems.

The human ear is particularly sensitive to distortion products that are continuously present, such as those caused by digital bandwidth compression, whereas analog bandwidth reduction processes induce very short term distortion products with a duration less than 20 milliseconds, which are difficult, if not impossible for the ear to detect. Modern analog companding systems operate within this acoustic "deaf spot", so the analog companding artifacts will not be perceived.

This article shows that given the same bandwidth availability and equal noise and signal power both analog and digital transmission systems will generate similar signal-to-noise ratio and distortion performance.

As the noise level increases in an analog transmission system, the overall signal-to-noise ratio also increases linearly with a consequent decrease in quality. Analog systems tend toward graceful failures, whereas a digital transmission system suffering a similar increase of noise increase in the transmission system will suffer a sudden cataclysmic total loss of signal (if squelched) or even worse, extreme high level crashing sounds (if not squelched). Digital systems are either "perfectly fine" or crash cataclysmically upon exceeding a certain threshold of noise.

Given equal transmission considerations, the principle difference between analog and digital performance is that the digital system fails cataclysmically while the analog system fails gracefully. We have the solution to this problem!

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