AUDIO CONSIDERATIONS IN SATELLITE TRANSMISSION TO CABLE TELEVISION SYSTEMS

John Berry

Group W Satellite Communications

ABSTRACT

Television, by it's name, is usually thought of as the transmission of video images. But much of the information we receive from television comes from the audio portion of the program. As more programming is delivered to cable systems via satellite, it is important for the system engineer to understand the various audio transmission techniques used in the industry. This paper will begin with some backround as to why this subject should be of concern to today's cable system engineer. It will also review some audio basics that the reader should be familiar with. Then the three most common forms of transmission are discussed, pointing out advantages and limitations of each one. Next, we will review the most commonly reported problems and some of the factors that contribute to them. Finally, the paper will discuss types of audio testing normally performed and give a step by step approach to system alignment.

NEW AWARENESS OF AUDIO

In the early days of television, not very much attention was paid to the audio that was accompaning the new pictures being brought into the home. The consumer was too facinated by the new technology to be very critical about it's quality. The TV receivers themselves had small speakers and simple circuitry. And besides that, the only comparison was with the radio or phonograph quality of the day , so all in all television sound was quite acceptable.

The situation today is quite different. Modern day television viewers have become quite aware of the quality of audio they are listening to. The consumer electronics revolution has brought items like high quality stereo systems, CD players, and hi-fi VCR's into a large majority of homes. Even the audio systems in our cars offer sound reproduction far above what was available in the best home stereos 10 years ago. In addition, as the broadcast industry moves rapidly towards implementing BTSC, more stereo capable, and component TV systems are being sold. These consumers, many of them cable subscribers, are becoming more discriminating about the audio signal quality they recieve. Therefore it is imperative that the cable system engineer be as well versed in this subject as possible.

AUDIO BASICS

A sound is produced by waves that cause pressure changes in the human ear. In order to transmit these sounds they must first be changed to electrical signals which we call audio. A transducer, i.e. a microphone, does this conversion and produces a signal that is a complex sinusoidal waveform with a frequency range of 50Hz to 15kHz. This is the range that most humans can perceive. The amplitude of the wave determines how loud or soft we hear it. The goal of any good transmission medium is to allow the signal to be reproduced in as close a quality to the original as possible.

Measurement

Scientific studies show that the human ear's response to sound is not linear, but rather, acts in a logarithmic manner. Thus a doubling of audio power is only perceived as a slight change in volume. In order to handle the math of this relationship, most audio signals are measured using special units. There are two commonly used scales in audio work, the first being dBm. A level of 0 dBm is equal to a power of 1mW delivered into a 600 ohm load. Any good cable engineer should be familiar with the dB measurement system.

The second commonly used scale is the Volume Unit or VU. This is usually measured on a special type of meter that has circuitry to handle the complex waveforms of active audio. It is used to monitor the average program level of an audio signal. While they are related, and often confused with one another, the dBm and the VU are two different measurements.

Average, Peak, and Loudness

We need to understand three other terms used in audio work; Average, Peak and Loudness. Average audio usually refers to measurements over a period of time, as seen on a VU meter. The center of the VU meter is considered 0VU. The "peaks" that are seen on the VU meter are not the peak levels of the audio however. Peak audio level refers to the instantaneous maximum level of an audio signal and is usually measured on a PPM (Peak Program Meter). The scale used is dBm. This is another specially designed audio device that can react fast enough to capture the peak levels of active audio.

Loudness refers to the relative amount of volume that our ear hears. It has a definite relationship to average and peak program level. A program that has been processed to produce a high average audio level will be perceived as louder than one which has higher instantaneous peaks but a lower overall average. Tuning through the FM broadcast band in any major market will demonstate this concept.

Stereo

As the requirement to deliver stereo programming increases, it is important to understand what aspects of stereo can be affected by satellite transmission. In it's most basic form stereo attempts to simulate to the listener the effect of sound coming from two different directions, cleverly refered to as L(eft) and R(ight). Monaural sound is the combination of both, or L+R. Audio engineers spend a lot of time setting up microphones to achieve this perception of Left and Right, known as separation. But the two signals are not mutually exclusive. Part of the Right audio appears in the Left channel and vice versa. As long as both signals remain in the same phase there is no problem. However, should one signal be reversed in phase, a strange effect occurs in the mono channel. The common audio of each channel cancels out, leaving only a very low difference signal. While this reversal can take place at a number of different places in the transmission path, it usually occurs when cabling is connected without regard to polarity.

Distortions

There are three major types of distortion measurement used in evaluating satellite audio transmission.

1) Signal to Noise (S/N) - This is the ratio between the maximum amplitude of the signal and the average noise in the channel, usually expressed in dB.

2) Total Harmonic Distortion (THD) - The amount of harmonics generated by a transmission system as referenced to a pure sine wave. Usually expressed in %.

3) Frequency Response - The amount of output level change from a reference point as the frequency varies from lowest to highest but the input level is held constant.

TRANSMISSION TECHNIQUES

There are three widely used formats for audio transmission in satellite relay to cable systems. These are FM subcarriers, companded subcarriers, and digital. Each will be discussed individually.

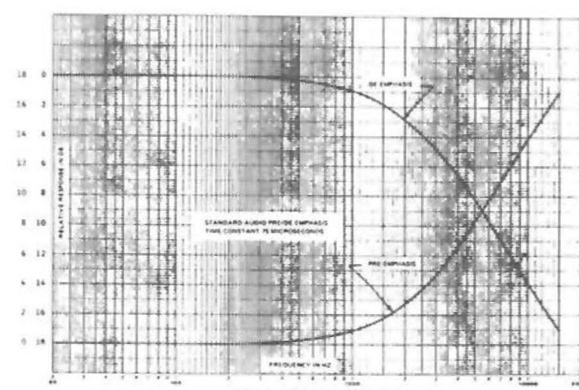
FM Subcarrier

This method is used for carrying the program audio of most non-scrambled cable services. An audio signal is used to frequency modulate a subcarrier located above the video signal, the most common frequency being 6.8MHz. In an FM system, the carrier is deviated from center frequency in relation to the amplitude changes of the modulating signal. Maximum deviation should occur at maximum input level.

Deviation settings used by programmers vary but are typically about 180-240kHz. This converts to an occupied bandwidth of about 400kHz. The total number of subcarriers being used as well as other engineering factors cause this variance. Once the proper deviation is determined, the subcarrier modulators are set up using a Bessel null method. This equipment usually remains quite stable and is monitored closely by the uplink.

An important concept that must be understood in order to evaluate an FM audio system is that of preemphasis. In most audio programming, the higher frequency components occur at lower levels than others, but are very important to the overall quality of the signal. Unfortunately, when these low level frequencies go through a transmission path, they are closer to the noise floor and are therefore received with a worse S/N ratio. The way engineers get around this problem is by boosting or emphasizing the higher frequencies prior to transmission. At the receiving end the signal is then deemphasized by the same amount to get back the original signal. The preemphasis standard most often used is the EIA 75 us curve which produces a boost of 16dB at 15kHz. This curve is also used with the 4.5MHz aural subcarrier of the NTSC TV channel. A chart of that curve is shown below (Fig. 1). What is important to realize is that if your system can only accept a maximum of +10dBm, and there is 17dB of preemphasis at 15kHz, then the highest level test tone you can send at that frequency is -7dBm. There will be more about this in the testing section.

Fig. 1



Companded Subcarriers

While the FM approach uses time tested and well understood techniques, it requires a relativly wide bandwidth and a large injection level into the uplink exciter. This limited early video services to only one or two possible audio channels. However, a requirement arose within the satellite industry to offer more high quality audio services on a single video transponder, both for the programmer's own use as well as to create an additional revenue source. A technique was developed in which the audio is companded using a unique adaptive preemphasis process. This creates an improvement in S/N which allows use of lower deviation and a reduced subcarrier power requirement. As a result, more audio channels can be sent with the video on a single transponder. This technique was pioneered by Wegener Communications, who continue to be the major supplier of both transmit and receive systems.

The major drawback of this technique is that special demodulators must be used that have appropriate circuitry to compliment the processing done at the uplink. Due to the unique systems involved, best results are usually obtained by using a demodulator from the same manufacturer who provides the transmit equipment.

Digital systems

Digital transmission of program audio on satellites is primarily done in conjunction with encryption of the signal. Basically, the audio is sampled at a high rate and each sample is converted to a digital value. This data stream is then scrambled by some algorithm and combined with the video during vertical blanking. At the receive end, the descrambler decrypts the data stream and a digital to analog converter recovers the audio signal. The major advantages of this method are it's immunity to noise and ease of encryption. It also frees up transponder spectrum for other uses, since subcarriers are not necessary. However, it does require a sophisticated demodulation device, in this case, the descrambler.

Combinations

Oftentimes, a programmer may choose to utilize all three techniques on a single transponder. For example, a service that is encrypted offers it's program in stereo via the digital method. They may also use the 6.8 MHz subcarrier for an announcement channel as well as having companded subcarriers for cue tones, second language, or other auxilary uses.

PROBLEM AREAS

The majority of satellite uplink sites used by cable programmers today are high quality, well designed facilities, staffed with trained personnel. The complexity of the systems involved demands that people have a high level of expertise. Most uplink sites and playback operations have audio monitoring equipment at a number of points in the transmission path. They also have available a complement of test equipment to ensure that proper operation is maintained. Again, the dollars being spent to get the signal out demands this kind of attention. In my experience of providing technical support to users of satellite services, I have found that most complaints can be traced to some other source besides the uplink. The two problems most commonly reported are different audio levels between services and distortion or sibilence. We will look at each one separately.

Levels

There are three things that can affect the received audio level at a headend. First, audio level into the transmitter. Second, the modulation setting of the subcarrier modulator (or digital encoder). And last, the output level adjustment of the demodulation equipment. In addition, the deviation control of the cable channel modulator will have an effect on the level the listener at home receives.

As mentioned before, most program origination personnel are quite concious of the need to maintain proper operating levels. Also, the modulation parameters of the companded subcarrier and digital techniques are pretty much set by the manufacturer so not much variation occurs there.

Unfortunately, there is not a hard and fast standard as to what deviation should be used with FM subcarriers. But, with the deviations being used by the major programmers, the variation should be no more than a maximum of 3 dB. This leaves the receiving equipment. The last section of this paper will cover a systematic approach to calibration.

Another important factor to consider regarding the varying level complaint it that of loudness, or the ear's perception of audio strength. This can best be illustrated by an example. A transmission path is aligned for unity gain from end to end. Using the same VCR, two pieces of program material are transmitted and monitored at each end. A steady tone at the beginning of each piece is measured at the same level on the output. The Peak Program Meters indicate that both programs have audio peaks that are just below the maximum allowable. However, after listening to each of them, one tape is noticeably louder. What did we adjust incorrectly?

Nothing! If we also had VU meters to monitor this feed we would have seen that during one program the meter needle was varying widely with the changes in content. However the other one, while having similar content, never seemed to let the needle fall very far down. This material sounds louder because of the higher average program level. This loudness effect is determined in the program's production by the amount of signal processing (compression) that is employed. We will talk about how to handle this in the calibration section.

Distortion

The second most commonly reported problem about satellite reception of audio is that of distortion. And the most frequently described symptom is sibilence. This is a type of high frequency distortion that is manifested as a hissing sound, especially on the "s" sounds. This, and other types of distortion, can come from a variety of sources including defective equipment. But it is usually traced to a situation where the subcarrier is deviated beyond the bandwidth of the reciever. A similar effect can occur if the digital audio modulator is overdriven.

Starting at the beginning of the path, the uplink could be allowing the subcarrier (or digital encoder) to be overmodulated. While this does occur, the rash of complaints from a large number of downlink sites usually causes the problem to be quickly corrected. A second cause can be misadjustment of the downlink receiver, allowing excessive input to the cable modulator. A third cause that has surfaced recently involves the inherent design of some "economy" satellite receivers when used with FM subcarriers.

In an effort to improve audio S/N when used with small antennas, some designers have reduced the audio IF bandwidth by as much as 50%. For example, where a bandwidth of 400kHz is used in most broadcast and cable quality demodulators, some receivers are being sold with bandwidths in the 200kHz range. This means that any modulation beyond this will be sharply removed by the filtering, resulting in distorted audio. Programmers are usually reluctant to compromise their technical parameters to accomodate the users of this equipment.

SYSTEM ALIGNMENT

This section will explain the most often performed audio transmission tests, with descriptions of the measurement equipment needed as well as the test signals used. Then it will outline a step by step approach to system alignment. However, before attempting any audio evaluation it best to check the physical installation of the receiving equipment and it's connection to the cable system headend.

Proper Installation

Wiring - Whenever possible, use shielded audio cable, grounded at one end, to reduce stray pickup. Grounding at both ends can create unwanted ground loops.
Impedance - Most receivers today are designed to operate into a 600 ohm balanced load, such as a cable modulator. A balanced load is one in which neither side is grounded while an unbalanced load has one side connected to ground. Unterminated devices, as well as very high or low impedance loads can cause false readings during testing. If necessary, use matching pads or buffer amplifiers to achieve the proper match.
Phaseing - As was discussed earlier, it is important to keep the phase of the two stereo channels in the proper relationship. Almost every receiver and descrambler is manufactured to maintain proper phase up to it's output terminals. The system engineer must then verify that proper polarity is observed when attaching the connecting cables.

Test Equipment

In order to perform the tests outlined here it is necessary to have some test equipment that is not normally found in a headend, but should be. First, a reliable, accurate method of measuring peak audio levels over a range of at least 70 dB. This can be a good quality Peak Program Meter with built in attenuation or a more sophisticated audio analyzer. A quality VU meter should also be on hand. In order to measure THD however, there are no real shortcuts. You have to have an audio test set that has that function.

Transmission Tests

As part of an ongoing trend of improving service to affiliates, a number of programmers schedule regular test periods. The audio portion usually deals with four areas; calibration of levels, S/N ratio, frequency response, and THD.

Connect the PPM or the audio analyzer to the audio output of whatever device you are testing. Check that impedances are correct.

Level setting- A fixed tone is sent at a reference level, usually 10-14db below maximum modulation. The idea is for each downlink site to adjust for the reference level they need. Set your receive levels as specified by the manufacturer. For most cable modulators, a 0 dBm level from the receiver or descrambler works well.

S/N ratio - For this test a tone is sent at a level that causes maximum modulation. This level is recorded, and then the uplink removes the audio and terminates the line. Remove attenuation as neccessary until a reading is obtained. This is the system noise floor. The two readings are added algebraicly to obtain the audio S/N ratio. For example, if the maximum level is +10dBm and the noise reads -45dBm, the S/N is 55dB.

Frequency Response - This test is designed to measure the flatness of the audio channel throughout the frequency range. A lower than normal input level is used to avoid the effects of preemphasis on the higer frequencies. At the source, a tone is sent at this lower level, typically 20dB below maximum. Then a series of tones are sent at this same level starting at about 50Hz and ending at 15kHz. The level is recorded at each step and then compared to the reference. A desired goal in this test is keep variations within 1 dB.

Total Harmonic Distortion (THD) - This is one test that is simple to perform with the right equipment, but nearly impossible correctly without it. Assuming that an audio analyzer with a THD function is available, it should be connected to the device under test, again verifying the proper impedance matching. When a reference tone is sent, select the THD function and read directly from the meter. A reading of 1-2% is acceptable.

Overall System Alignment

After performing the above tests on a number of services, it is possible to use the PPM and VU meters to adjust the other receiving equipment you have so that the peak levels of all sources are fairly close. When making these adjustments, use the five minute rule. That is, monitor the program for that long to make sure that the signal levels are typical. Remember, a three dB change is just barely noticable. Next, each modulator should be adjusted with a test tone for a peak deviation of 25kHz (NTSC aural carrier; use BTSC deviations where applicable). A number of modulation meters are on the market now to make this adjustment. Also, some modulators have built in metering to set correct deviation.

The combination of matched receive levels and equal deviations should now produce somewhat equal sounding audio as you step through the channels of your cable system. If not, listen to those that are markedly different, and use your VU meter to determine if the problem is program loudness. If most of the programming is that way, you will have to adjust the cable modulator, or the input level from the receiver to get it to match.

ACKNOWLEDGMENTS

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