PRACTICAL CONSIDERATIONS FOR BTSC STEREO IN THE CATV PLANT

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ABSTRACT

Presented is a summary of significant experience with BTSC stereo, with elaboration on areas where operators must make decisions critical to system performance. The acquisition of audio source material in the headend is possible by several means. Formats include various satellite subcarrier systems, scrambled satellite transmissions, and locally supplied audio feeds. Operators often need automatic switching for redundancy or commercial message insertion. Several methods are available for the transportation of stereo from the earth station to the headend. Some cable operators use FM links or AM links to connect various distribution systems, which may pose special problems when stereo is used. Audio input levels to the encoder must be set. Apparent loudness, peak deviation, and average levels are explained. The role of audio signal processing is discussed. Manufacturers provide multiple options for interfacing the stereo encoder with the headend modulator. Sources of error and performance degradation are cited, with techniques of minimizing them. CATV scrambling systems can affect separation and noise performance. Subscriber equipment can cause the most degradation to the BTSC stereo signal. Performance characteristics for various consumer decoders are described. There are several useful techniques for system evaluation. Complete checks require interruption of service, though basic operation can be verified while on line.

I. INTRODUCTION

BTSC stereo is appearing on more and more cable systems. The question of "when to go stereo" is giving way to the question of how to best make it work.

This paper is a distillation of experience gained in the first year of BTSC stereo on cable. The discussion is divided into eleven topics, ordered to roughly correspond to the signal path through the plant. Extra focus is placed on the characteristics of audio systems because audio is a new concern for many cable operators. Familiarity with the headend is assumed in the discussions. Grateful acknowledgement is made to the dozens of cable operators and technicians who have installed stereo in their systems and brought to light many of the problems and solutions described here.

II. BRINGING LEFT- AND RIGHT- CHANNEL AUDIO INTO THE HEADEND

Most of the services with which cable operators use BTSC stereo are satellite delivered.

If satellite transmissions are scrambled with the Videocipher system, audio information is sent in an encrypted digital format. The headend descrambler provides three audio outputs. These are "Left," "Right," and "Mono." [1] See Figure 1.

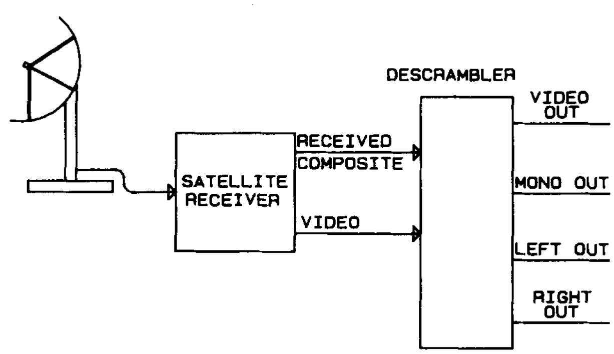


FIGURE 1: RECOVERING AUDIO FROM A SCRAMBLED SATELLITE TRANSMISSION

If satellite transmissions are in the clear, program audio is found on subcarriers transmitted with the video. These subcarriers are located between 5.0 MHz and 8.5 MHz. Usually two or three video related subcarriers are sent with each satellite delivered service. Each subcarrier is frequency modulated with one audio channel. See Figure 2.

The satellite programmer may provide "left," "right," and "mono" on three subcarriers. This is called the "discrete" channel format. When the discrete format is used, programmers often use special noise reduction with the audio signals for the left and right channels. The subcarrier can then be frequency modulated with less deviation, and bandwidth is conserved. The subcarrier demodulator at the headend has signal processing to complement the noise reduction system. See Figure 3a.

Instead of using the discrete format, programmers may use the "matrixed" format. One subcarrier sends "mono" and another subcarrier sends a "difference" channel. With the matrixed format, the left and right channels are recovered by properly combining the "mono" and "difference" signals. The "de-matrixing" circuit is usually located with the subcarrier demodulator circuits. Many receivers contain all this circuitry and provide left and right channel audio signals at their output. Most matrixed systems do not use any noise reduction other than a standard pre-emphasis/de-emphasis. See Figure 3b.

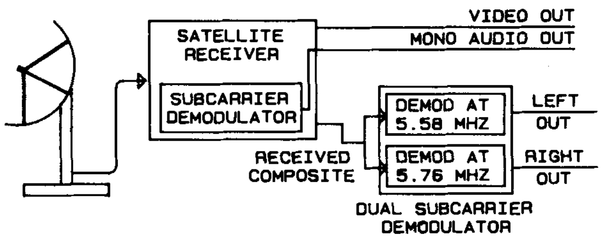


FIGURE 2: RECOVERING AUDIO FROM A SATELLITE CHANNEL WITH AURAL SUBCARRIERS

BTSC stereo may be used with locally originated program material from public access facilities or broadcast studios. A pair of FM links or leased phone lines bring audio into the headend. See Figure 4. Highest fidelity results when the left and right channels travel over identical paths. If their paths are not the same, fidelity on mono receivers may suffer, and stereo imaging may be affected.

III. CARRYING LOCAL BROADCASTERS IN BTSC STEREO

Local broadcasters can be put on the cable system in several ways. A signal processor can receive an off-air signal and place it on the cable system. The processor provides control of aural carrier level and rejects undesired adjacent signals. Many cable systems use signai processors to successfully carry local stereo broadcasts. See Figure 5. However, not all processors perform equally well. The path through which the aural carrier travels is critical. This path must be sufficiently wide to pass the sidebands that BTSC stereo creates. The amplitude and delay responses must be reasonably flat. If the aural carrier path through the processor is not wide enough, a BTSC signal will suffer increased distortion. If the path is not symmetric around the carrier frequency, or amplitude response is not flat, there may be excessive FM to AM conversion. AM on the sound carrier is recognized by some set top terminals as a cue to descramble. When a

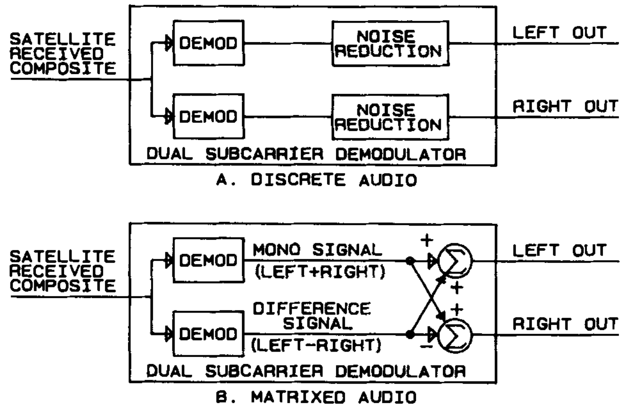


FIGURE 3: "MATRIXED" AND "DISCRETE" AURAL SUBCARRIER DEMODULATION

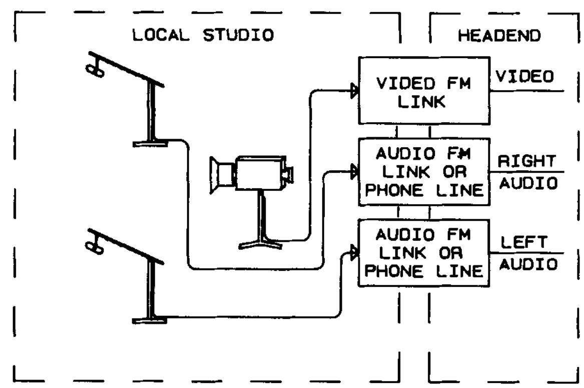


FIGURE 4: LOCAL STUDIO FEED

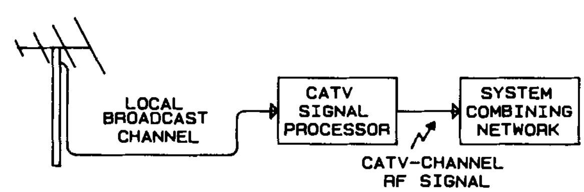


FIGURE 5: USING A SIGNAL PROCESSOR FOR A LOCAL STATION

local broadcaster uses the BTSC Second Audio Program (SAP) service or Pro channel, the carrier receives even wider deviation, and FM to AM conversion becomes more critical. Most processors provide adequate performance. Consult the manufacturer if there is any doubt.

Some cable operators place local broadcasters on their systems after demodulating the off-air signal. Baseband video and audio from the demodulator are routed through patch bays or switchers for control purposes. A modulator then places the signal on the system.

See Figure 6. Using such a demod-remod process is not recommended if the carriage of BTSC stereo is desired. The demodulator and modulator may not have sufficient audio frequency response to pass the entire BTSC signal. Typically the BTSC pilot tone will make it through, but the difference channel information that is required for true stereo performance is lost. Even if the demodulator and modulator can pass the entire BTSC baseband composite signal, stereo separation will severely suffer if deviation sensitivity of the modulator does not perfectly match the output level of the demodulator.

The demod-remod process will pass BTSC stereo if the 4.5 MHz aural carrier output from the demodulator is used instead of the baseband audio output. The aural carrier is sent to the modulator without ever being demodulated. This method preserves the critical BTSC parameter of aural carrier deviation setting. See Figure. 7.

For highest fidelity, some broadcasters provide special feeds of video and audio to nearby headends. In this case, Figure 4 again applies. Baseband left- and right-channel audio signals are brought to the headend.

IV. AUDIO LINE IMPEDANCE AND BALANCED VS. UNBALANCED LINES

Various devices in the headend will call for balanced or unbalanced lines, and for high impedance or low impedance loads.

Unbalanced Audio Sources

An audio source is said to be unbalanced if its output is presented on one terminal referenced to ground. The signal voltage on this terminal swings above and below ground.

An unbalanced source may have an "active" output or it may be "transformer-coupled." If a device has an active output, the final amplifier is wired directly to the output terminals. If the device is transformer-coupled, the output amplifier drives the primary side of a transformer. Two leads from the secondary side of the transformer provide the output signal. One of the leads is connected to ground. The other lead then has the signal voltage on it. The signal voltage swings above and below ground. See Figure 8.

Balanced Audio Sources

The output of a balanced audio source appears across two non-grounded terminals. Additionally, a third terminal at ground potential is usually available. The voltage on one terminal swings above and below ground as the signal varies. The voltage on the other terminal swings above and below ground in equal amounts, but in the opposite direction. Thus the voltages on the two terminals are "balanced," as shown in Figure 9.

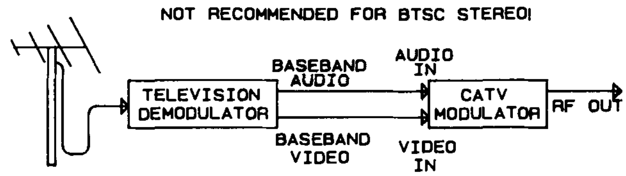


FIGURE 6: DEMOD-REMOD FOR A LOCAL STATION

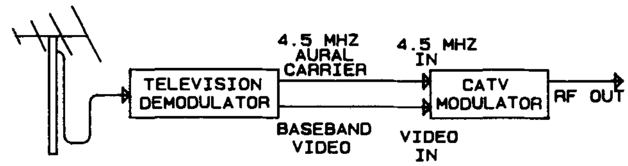


FIGURE 7: DEMOD-REMOD WITH 4.5 MHZ AURAL CARRIER

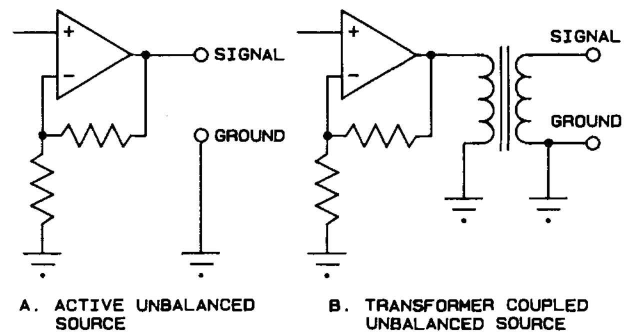


FIGURE 8: UNBALANCED SOURCES

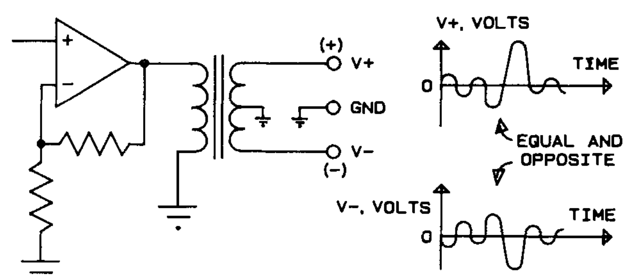


FIGURE 9: TRANSFORMER COUPLED BALANCED SOURCE

Different methods are available to manufacturers for providing balanced outputs. The output can be transformer-coupled, as in Figure 9. The audio output amplifier in the source drives the primary side of a transformer. The secondary side of the transformer has a center tap. The audio signal appears across the end leads. If the center tap is connected to ground, the DC voltage of the secondary is held at zero. The end leads of the transformer now provide a true balanced output: the AC voltages on these terminals with respect to ground are equal and opposite. If the center tap is not connected to anything or there is no center tap, the output winding is said to be "floating" and not balanced.

An "active balanced" output does not use a transformer. Instead, transistor amplifiers or integrated circuit op amps are used to create the output signals directly. See Figure 10. One amplifier drives one output terminal. Another amplifier produces an inverted version of the same signal and drives the other output terminal. A true balanced output again results: the signal terminals contain equal and opposite voltage waveforms.

Balanced sources can usually be identified by the markings near the terminals or by reading the manufacturer's data sheet. If the output terminals are labeled (+) and (-) this is probably a balanced source. Whenever you use a balanced source, be sure that the (+) and (-) output signal lines are never shorted to ground. This may damage the audio source and will disturb operating levels.

If necessary you can usually convert a balanced source to an unbalanced one. For equipment with active balanced outputs, simply connect the signal wire to the (+) output and the ground wire to the source ground. Do not make any connection to the (-) output terminal. On equipment with transformer coupled outputs, disconnect the center tap from ground. Connect the (-) terminal to ground instead. The (+) terminal is now the signal line. See Figure 11.

Audio Inputs

When an audio input is driven by a signal line, that input is said to "present a load" to the line. The load may be balanced or unbalanced; high impedance ("bridging") or low impedance.

A balanced input has two signal input terminals in addition to a third terminal for ground. The ground terminal is used only to connect shielding on audio cables. The balanced input operates by amplifying the difference between the voltage signals that are applied across the two input terminals. When driven by a balanced source, the desired signal appears as a difference-mode voltage across the input terminals.

Noise may be picked up on the audio wires, often as hum or radio interference. Usually this noise is induced equally on the (+) and (-) wires. Such common-mode signals are rejected by the balanced input.

Balanced inputs may be driven by unbalanced sources. The ground wire from the unbalanced source should be connected to the (-) input terminal. The signal line from the

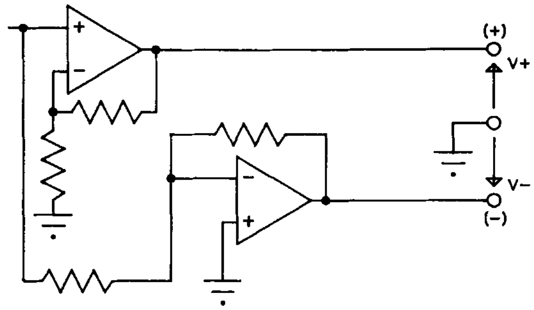


FIGURE 10: ACTIVE BALANCED SOURCE

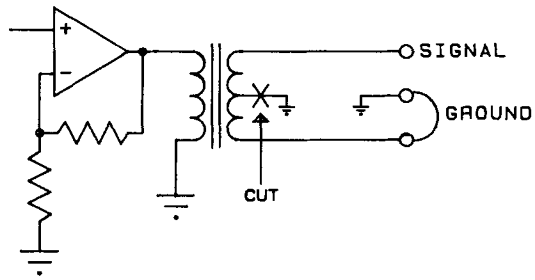
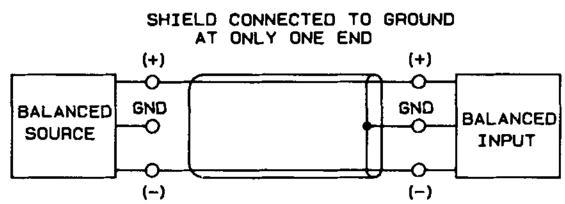
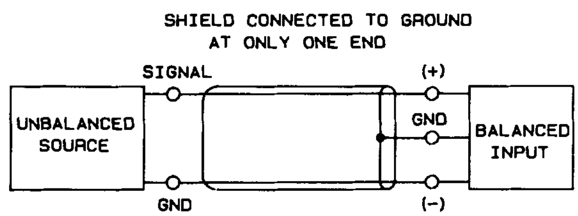


FIGURE 11: CHANGING A TRANSFORMER COUPLED BALANCED SOURCE TO AN UNBALANCED SOURCE



A. FROM A BALANCED SOURCE



B. FROM AN UNBALANCED SOURCE

FIGURE 12: CONNECTING TO A BALANCED INPUT

unbalanced source should be connected to the (+) input terminal. Hum and noise rejection may no longer be as effective as with a balanced system. See Figure 12.

An unbalanced input has one signal input terminal. It is referenced to a ground terminal. The advantage of an unbalanced audio system is that fewer wires must be used for signal routing and patching. The disadvantage is that hum may easily be caused by imperfect grounding of either the audio or AC power system. An unbalanced input should not be driven from a balanced source.

Transformer coupled inputs can be driven from balanced or unbalanced sources. They provide common-mode noise rejection and can break ground loops that cause hum.

The standard source, line, and load impedance for professional audio equipment is 600 ohms. However, many products have input impedances higher than 600 ohms. These "hi-z," or "bridging" inputs are usually 10 kilohms to 50 kilohms. If audio cables are short, several hi-z loads may be driven by one 600 ohm source. It is recommended that the load furthest from the source present a terminating impedance of 600 ohms.

In most headend applications it is permissible to leave audio lines unterminated. But for lines longer than several dozen feet, termination preserves frequency response flatness and reduces induced noise. When lines are terminated, be sure that they are terminated only once. Depending on how low the source impedance is, signal amplitude will be affected by changing the load impedance. The lower the source impedance is, the heavier load (lower load impedance) it may drive. Most sources will not perform well if they must drive more than one 600 ohm load.

Connections to the (+) and (-) terminals on sources and loads should be consistent. If the (+) and (-) connections are reversed somewhere, the left and right channels will end up out of phase with each other at the receiver. Monaural receivers will have no sound output. Mono compatibility is assured when the left and right channels are wired in phase with each other.

V. SIGNAL SWITCHING FOR LOCAL MESSAGE INSERTION

Figure 13 shows one system that detects cue tones and uses a video tape player to insert messages. The tone detector listens for control signals provided by the programmer. These control signals precede pauses into which local messages are inserted. On some services a separate subcarrier is transmitted over the satellite to carry the control signal. On others, the "mono" program audio channel has control tones.

When tones are detected, the switcher selects video from the tape deck and sends it to the modulator. At the same time, a relay contact closure commands the stereo encoder to use its alternate audio inputs. When the taped message is over, program video is switched back in and the encoder is commanded to accept "main" audio.

VI. TRANSPORTING STEREO FROM A REMOTE EARTH STATION TO THE HEADEND

Separate FM links can be used to deliver video and audio to the headend. The satellite receiving system contains all descrambling or demodulating circuits to produce baseband video, left audio, and right audio outputs. Coaxial cable connects the earth station to the headend. An FM modulator uses a 14 MHz wide channel to transmit the video signal to the headend. Two separate FM modulators use channels about 200 kHz wide to transmit the left and right audio signals. [2] Three FM receivers in the headend provide baseband video, left, and right audio outputs. The stereo encoder and modulator create the BTSC television signal. See Figure 14.

Figure 15 shows a method that is not recommended. A BTSC stereo encoder is located at the earth station. Its output is a modulated

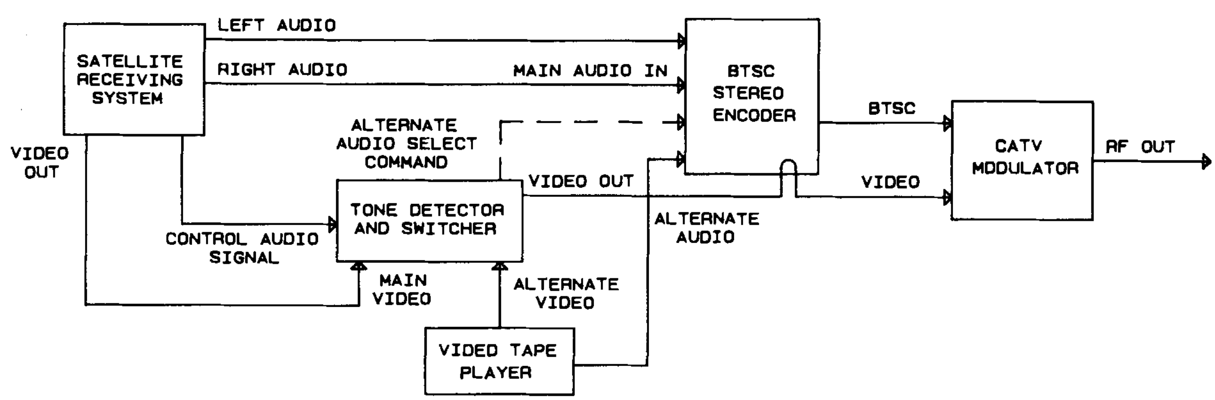


FIGURE 13: SIGNAL SWITCHING FOR LOCAL MESSAGE INSERTION

4.5 MHz aural subcarrier. Baseband video combined with the 4.5 MHz aural carrier is sent over a video FM link. This scheme may result in two serious problems: visible intermodulation beats and noisy audio. The visible beat between the aural carrier and color carrier is likely because the FM link's channel is only wide enough for quality transmission of video. When the subcarrier at 4.5 MHz is added, FM sidebands farther away from the carrier frequency become more important. But distant sidebands are attenuated in bandpass filters in the transmitter and receiver. The receiver produces an output that shows intermodulation distortion because important FM sidebands were cut out. [3] The audio is noisier because of the nature of frequency modulation systems: higher modulating frequencies suffer more noise at the receiver. The 4.5 MHz carrier at the output of the video FM receiver is surrounded by much more noise than it was at the input. The BTSC stereo format itself involves frequency modulation, with special noise reduction for its higher modulating frequencies. The degradation of carrier-to-noise ratio that the 4.5 MHz signal suffers in the FM link will combine with other system noise on the way to the subscriber. Intolerable noise performance may result.

Figure 16 illustrates the use of a composite video FM microwave link. Two FM subcarriers are modulated with the left and right audio signals. These subcarriers are combined with the baseband video to create a composite signal. An FM transmitter uses a 22 MHz wide channel in a microwave radio band. A receiving system in the headend produces baseband video, left, and right audio outputs. Because this system uses a wider channel than the simpler video FM link, it is capable of carrying subcarriers above the video with less intermodulation distortion. In addition, these subcarriers employ wider deviation or special noise reduction to make up for degraded carrier-to-noise ratios at the microwave FM receiver output. High quality audio and video transmission is achieved.

VII. HEADEND TO HUB-SITE TRANSPORTATION

When a hub-site is fed over a properly operating AM microwave link, BTSC stereo signals are not adversely affected. See Figure 17. However, in complicated hub-sites the received signal may not be routed directly to the trunk amp. Instead, signal processors or demodulator-modulator pairs may be used to create flexibility in channel assignment, scrambling, and control. The quality of received stereo depends critically on the quality of the path through which the aural carrier travels. Though any single block of the cascade may pass BTSC stereo, the combined effects of several processors, demodulators, and modulators will add up. In a complicated distribution system performance figures for stereo are more easily measured than predicted.

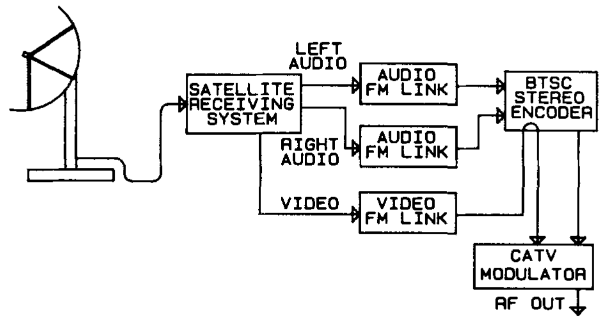


FIGURE 14: EARTH STATION TO HEADEND: TRANSPORTING LEFT, RIGHT, AND VIDEO ON SEPARATE FM LINKS

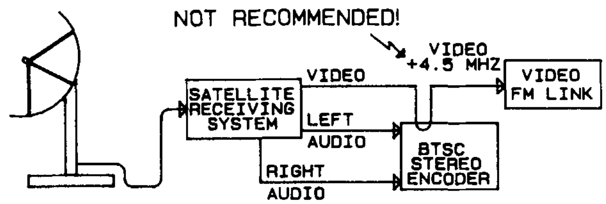


FIGURE 15: EARTH STATION TO HEADEND: TRANSPORTING VIDEO +4.5 MHZ

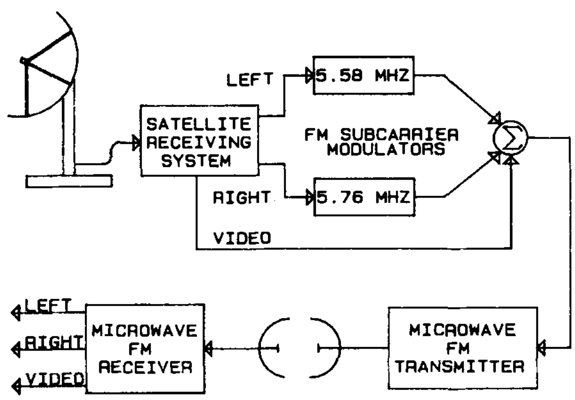


FIGURE 16: EARTH STATION TO HEADEND: COMPOSITE VIDEO ON FM MICROWAVE LINK

Some straightforward rules help preserve stereo separation:

* Do not carry baseband BTSC composite multiplexed signals more than the few feet needed between stereo encoder and television modulator.

* Once the BTSC-modulated aural subcarrier has been created, do not demodulate it to baseband and remodulate it again. Leave it in the form of an FM carrier and let the subscriber's equipment demodulate it.

* Highest fidelity comes from transporting discrete video and left- and right-channel audio, and using a BTSC encoder and television modulator at each hub site.

* Do not add the 4.5 MHz aural subcarrier to an FM link designed to carry video only.

VIII. AUDIO LEVELS

Some Terms Used In Discussions of Audio Processing

"Loudness level" is a subjective property of audio program material. Loudness is not easily measured. Instruments are available that attempt to simulate the response of human hearing and give a numerical measure of loudness. "Ideal" measurements are not achieved because of the complexity of ordinary sounds, and differences among listeners. Human perception of loudness is a non-linear, frequency-dependent phenomenon. Statistical data has been collected to determine how people judge loudness of single tone signals. The "Fletcher-Munson" curves resulted from early studies, and later researchers have produced other similar curves. [4]

"Peak level" describes the maximum excursion of signal voltage above or below zero Volts. Some waveforms will produce positive peaks of different value from the negative peaks. The human voice is one of these. Peak levels are easily measured.

"Average level" describes the average of the magnitude of the signal waveform over some time interval. This may be measured by full-wave rectifying the waveform and averaging it with a low pass filter. In such a circuit the averaging interval depends on the charge and discharge time constants of the storage network. The root-mean-square (rms) value of the signal may be measured to get a different kind of average. In this case the signal is multiplied by itself, which results in a wave that is greater than or equal to zero. The waveform is integrated over a time interval. A logarithmic amplifier may be used to provide an output proportional to a decibel reference. Otherwise a circuit may be used to simulate the square-root function and provide an output linearly proportional to rms level.

The "dynamic range" of a signal or channel is the ratio of the peak signal level that is possible to the lowest acceptable signal above noise. The "signal-to-noise" is the ratio of the present signal power to the noise in the channel. "Headroom" is the difference between the amplitude of the biggest undistorted sine that can be carried by the channel, and the nominal, average, or "operating" signal level.

Loudness control is achieved by various methods. The most common is compression. "Compression" implies gain control too slow to limit peaks, yet fast enough to substantially reduce the amplitude differences between passages with small and large average energies. Broadcast and studio equipment commonly employs frequency selective compression, which allows many aesthetic improvements over ordinary compression. Most important among these is the avoidance of loudness modulation of one frequency band by energy in a different band. Unfortunately it also allows opertors to drastically change the dynamics of recordings. A "loudness controller" contains circuitry to evaluate the spectral content of a signal. Program material is split into various frequency bands, compressed, and possibly clipped. Filters for each band get rid of harmonics caused by clipping. The bands are recombined and sent to the output.

Phase shift networks are used to make assymmetric peaks symmetric. This allows a broadcaster to fit a higher average energy within the same peak levels. It is widely

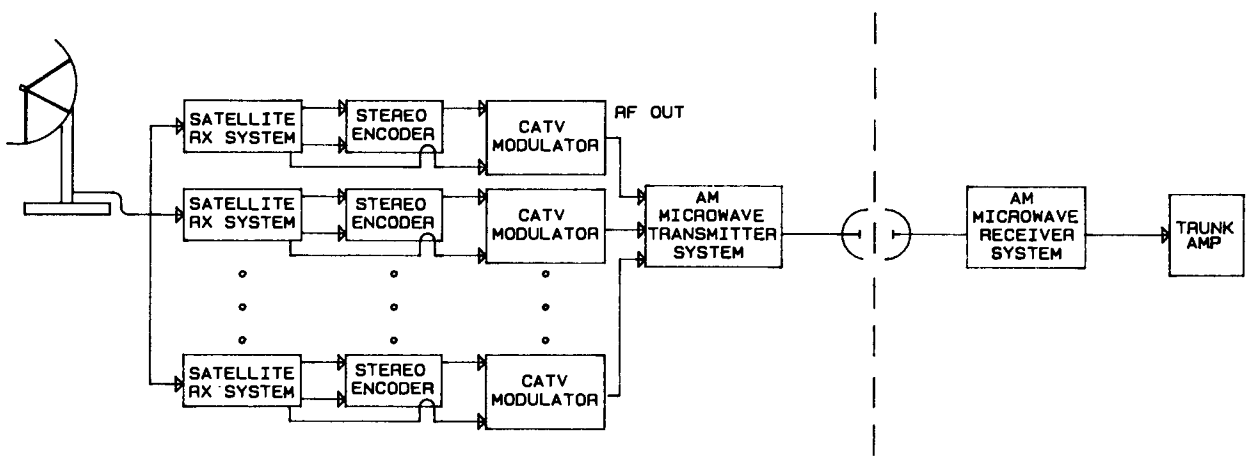


FIGURE 17: HEADEND TO HUB-SITE: SEVERAL PROGRAMS ON AM MICROWAVE LINK

accepted that most phase delays are not audible, as long as the left and right channels are treated equally. Fixed phase shift networks have been used in broadcasting for over 25 years.

Loudness Variations

Loudness levels among television channels vary greatly. Several things contribute to this.

* Some audio program material is "dense" or loud because of the program content: gunfights on horseback; chords played on an electric guitar. Other material is quiet or sparse; soft conversations or background sound effects.

* Modulation levels on some channels are set higher than on others.

* The original recording or transmission levels for programs vary.

* Some channels are compressed and limited somewhere in the transmission path. This may raise their loudness level compared with unprocessed channels without changing peak modulation.

* Some programs are compressed and limited during original production.

Audio Levels and Headend Practice

When adding BTSC stereo to a system, care must be employed when setting the left and right input levels on the stereo encoder. Most satellite-delivered programs are provided with audio that has not been compressed and limited to meet the needs of CATV transmission. Some programs may have quiet soundtracks with few loud passages, and others may have moderate to loud sound all the way through. Either situation can occur even though the peak levels indicated on the stereo encoder's meters appear the same. This is because different program producers use different audio processing when creating their soundtracks. If a program has been compressed and limited, it will have a higher perceived loudness for the same peak voltage levels. FM broadcasters take advantage of this fact: different FM stations have different loudness due to the audio processing they may or may not use, but they all have the same peak deviation of their carriers. Many people regard the over-use of these techniques as a degradation of program fidelity.

With mono program material, cable operators set levels to achieve nominally equal perceived loudness from channel to channel, and to avoid too much flashing of the "over deviation" light on the modulator. This is an effective but very imprecise method. Some stereo encoders have true peak reading meters on the front panel. The tendency is now to set levels to avoid peaks (keep the meters out of the red). This may result in a perceived loudness on stereo channels that is less than the loudness on mono channels. This difference will be heightened on satellite-delivered programming that has not been compressed and limited. If loudness levels are a problem, the operator has several options.

* The left and right input levels on the stereo encoder can be turned up some. This will drive the level meters into the red more often, which will mean that the aural carrier is being deviated more widely than called for in the FCC specs. Broadcasters are not allowed to do this, and usually don't. However, cable operators commonly overdeviate their aural carriers. Remember that excessive level can increase distortion, especially if the stereo encoder has clipping circuits to protect against overdeviation. Excessive overdeviation will cause audible distortion in recievers and visible interference on the TV screen.

* A more expensive option is to install a stereo compressor/limiter before the input to the stereo encoder. This way the stereo programming can have its peak-to-loudness ratio brought more in line with what local broadcasters have. The operator may view this as too much trouble and expense. Setting up most stereo processors is a non-trivial problem that requires the subjective judgement of the individual at the controls.

* Another option is to leave the levels set correctly and send the subscriber sound with its original characteristics. Wider dynamic range results, but the operator may have to deal with complaints about inconsistent loudness levels.

IX. ENCODER-MODULATOR INTERFACE

The stereo encoder may be connected to the modulator in many different ways. Figures 18, 19, and 20 are generalized block diagrams that show most of the configurations available.

Stereo Encoder Inputs and Outputs

Figure 18 shows the inputs and outputs of a "generic" BTSC stereo encoder. The left and right audio signals are supplied by satellite receiving equipment or local origination sources. Program video is looped through the stereo encoder so that the pilot tone may be locked to the horizontal scan frequency.

The stereo encoder provides several different kinds of output. The BTSC multiplexed baseband composite signal consists of the sum channel from 50 Hz to 15 kHz, the pilot tone at

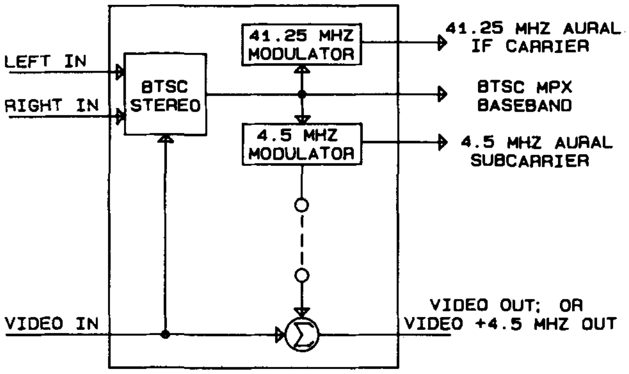


FIGURE 18: BTSC STEREO ENCODER INPUTS AND OUTPUTS

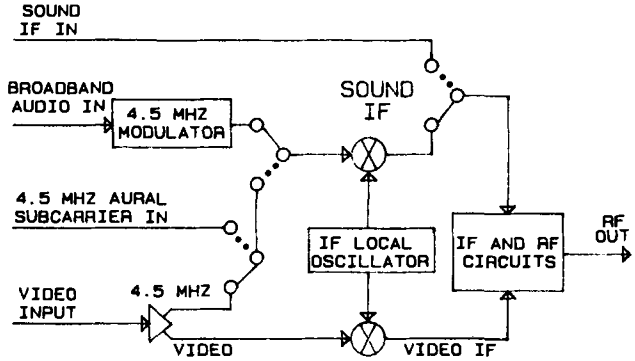


FIGURE 19: TELEVISION MODULATOR INPUTS AND OUTPUTS

15.734 kHz, and the difference channel which has been multiplexed to the 30 kHz-wide band centered at 31.468 kHz. If the Second Audio Program (SAP) channel is used, additional energy is present in a 20 kHz-wide band around 78.670 kHz. [5] A 4.5 MHz aural subcarrier modulator accepts the BTSC composite signal as its input. It frequency modulates a 4.5 MHz carrier and makes it available as an output. Some encoders allow the 4.5 MHz carrier to be combined with the video baseband signal. A 41.25 MHz modulator is also available for some stereo encoders. It produces a frequency modulated carrier at the standard sound IF.

Television Modulator Inputs and Outputs

Figure 19 shows various inputs and outputs for a "generic" CATV modulator. The video input circuit on some modulators has a splitter. This allows the introduction of video only or video plus the modulated 4.5 MHz aural subcarrier. Some modulators have a separate input for a modulated 4.5 MHz aural subcarrier. A simple modification can usually create this input on modulators that do not already have it.

The audio input on older modulators is not capable of accepting the BTSC multiplexed composite signal. Manufacturers offer upgrades and new modulators that do accept this type of input. Modulators configured for traditional monaural broadcasts include a 75 microsecond pre-emphasis circuit that is standard for noise reduction. When a modulator accepts the BTSC baseband input, its pre-emphasis must be defeated. The stereo encoder now performs the pre-emphasis. The modulator must also be able to deviate the sound carrier with frequencies up to 100 kHz.

CATV modulators usually provide some way to access the sound IF and video IF circuitry. For certain modulators, the aural carrier path does not pass BTSC stereo. In these cases the sound IF input may be the only way to successfully inject a BTSC stereo signal.

Connecting Stereo Encoders to Television Modulators

Figure 20 shows four connection schemes for BTSC stereo systems. Scheme 1 uses the BTSC multiplexed composite baseband output of the stereo encoder. The modulator must be able to accept a broadband input at its audio terminals, with no 75 microsecond pre-emphasis engaged in the modulator. This configuration requires that the deviation sensitivity of the modulator be precisely set. If deviation sensitivity is misadjusted, stereo separation degrades. Calibration tones in the stereo encoder and an accurate deviation indicator in the modulator will greatly simplify the set-up process. Specific instructions can be obtained from manufacturers supporting the interface at baseband. When correctly installed, the interface at baseband provides excellent BTSC stereo performance.

In Scheme 2 the modulated 4.5 MHz aural subcarrier comes from the stereo encoder. The television modulator has a separate input to accept the 4.5 MHz carrier. This set-up minimizes the adjustments that the operator must make upon installation. The deviation sensitivity of the 4.5 MHz modulator is calibrated at the factory, not in the field. For some modulators, a modification must be performed to create a separate input for the 4.5 MHz carrier.

Scheme 3 uses the modulated sound carrier at the intermediate frequency of 41.25 MHz. This system is best when using modulators that cannot be modified to accept either baseband input or a modulated 4.5 MHz carrier. Some modulators may be limited in their ability to pass the BTSC signal because of various filters in their signal processing chain. The 41.25 MHz modulator bypasses most of this processing. It is important that the manufacturer of a 41.25 MHz modulator provides sufficient frequency accuracy and low phase noise to maintain a quality signal.

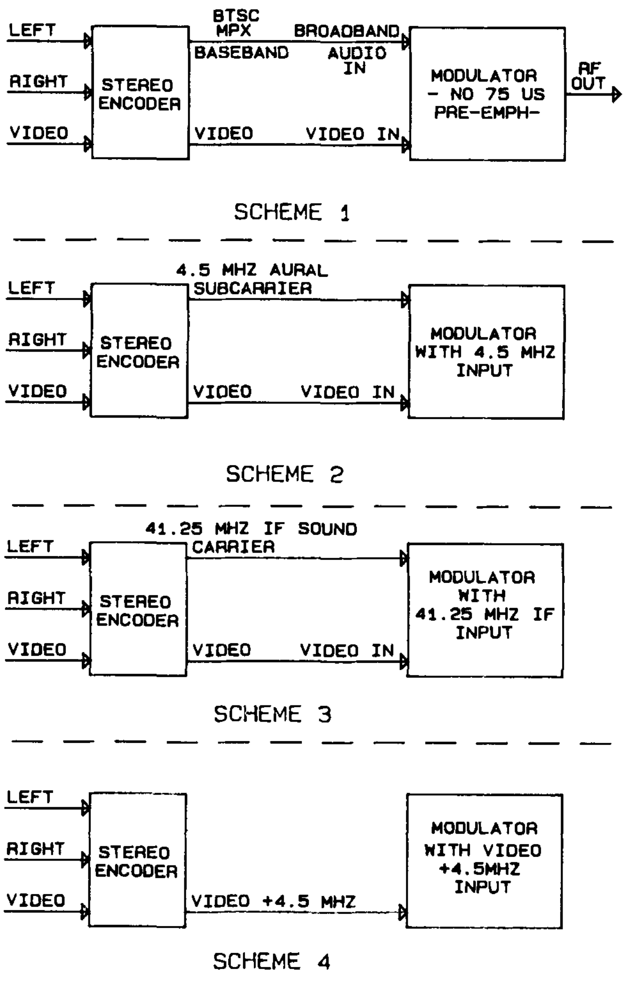


FIGURE 2Ø: CONNECTING BTSC STEREO ENCODERS TO TELEVISION MODULATORS

Scheme 4 sends video plus the modulated 4.5 MHz aural subcarrier over one coaxial cable to the video input of the modulator. This scheme is not optimum. In the modulator, the 4.5 MHz subcarrier is split from the video. Once they are there, removing video artifacts from the aural carrier band is not possible. Therefore it is best to lowpass filter the video to eliminate energy above 4.2 MHz before it is combined with the 4.5 MHz subcarrier. Video artifacts may be amplified by the aural carrier limiter stages of the modulator. The result is increased buzz in the audio. Typically the audio signal-to-noise ratio may suffer a 2 to 3 dB decrease, depending on video content. However, many operators use Scheme 4 to take advantage of existing patch bay systems in their headend. Performance for mono or stereo can be improved by avoiding video overmodulation. Additionally, the 4.5 MHz signal from the stereo encoder to the modulator may be increased above the 0.1 Volt level that is standard. This provides more immunity from video noise. Distortion is not likely to be created unless the modulator's limiter circuits are overloaded.

Because of the many connection combinations possible for BTSC stereo, it is best to consult the manufacturer of the encoder and modulator while determining the choice for a headend. Factors such as cost, switching requirements, routing requirements, redundancy, and service make it impossible to say that one way is best for all cases.

X. STEREO AND SCRAMBLING

CATV scrambling systems affect television audio if the aural carrier is used to transmit descrambling information. Television audio modulates the frequency of the aural carrier. Descrambling information modulates the amplitude of the aural carrier.

RF Scrambling Systems

RF scrambling systems disguise the television signal's synchronization information. The amplitude of the television carrier is modulated up and down by the scrambler. The amplitude of the aural carrier is modulated with coded pulses or sine waves. The subscriber's set top terminal receives the AM information riding on the aural carrier. From this it determines how to recover the synchronization information. The set top changes the amplitude of the television carrier up and down as necessary to restore its original shape. In the set top, the video carrier and aural carrier travel through the same path. When the video carrier is restored to its correct shape, the aural carrier is amplitude modulated with this correction too. This leaves quite a bit of amplitude modulation on the aural carrier.

An ideal FM detector ignores all AM on the carrier and correctly reproduces the original audio program. However, the demodulator in the subscriber's television receiver may be sensitive to AM. The descrambling information will then be heard in the audio output. Scrambling interference is heard around harmonics of 30 Hz, 60 Hz and/or 15734 Hz, depending on the scrambling scheme. It also has energy at twice the horizontal scan rate, which is 31468 Hz. These frequencies are for the most part out of the range of what most loudspeakers will reproduce, or out of the hearing range. Thus scrambling buzz has not been a big concern with monaural television audio.

When the subscriber is taking advantage of BTSC stereo, scrambling interference is more important. Scrambling interference has two effects: audible buzz and degraded stereo separation. Audible buzz is worse than that for mono systems for two reasons. The most obvious reason is that now the subscriber is listening to the audio more closely, and with better loudspeakers. The second reason is that the scrambling interference includes energy centered at twice the horizontal scan rate. This is right where BTSC stereo has its "difference" channel. If interference energy is present in the difference channel, it is translated into noise in the left and right audio outputs. BTSC stereo separation depends on the phase of the pilot tone at 15.734 kHz, the horizontal sweep frequency. One of the interference components from scrambling may appear here. If the interference component is large enough, it can add an apparent phase shift to the pilot tone. The stereo decoder will then produce incorrect difference channel information. Stereo separation also depends on the noise reduction system used for the difference channel. Buzz added to the demodulated difference channel may cause the noise reduction to mistrack, creating slight amplitude errors in the difference channel. These amplitude errors are translated to separation errors when the left and right channels are recreated. Scrambling systems that change the video carrier by 10 dB create more interference than 6 dB systems.

The degradations described here are not necessarily very severe. It must be emphasized that these effects are due mostly to AM detection in receivers, and that most receivers with stereo decoders do not suffer very much at all from this. As of this writing many systems are up and running in BTSC stereo with various RF scrambling schemes.

Baseband Scrambling Systems

Baseband scrambling systems, like RF scrambling systems, disguise the television signal's synchronization information. The baseband converter demodulates the video signal down to baseband from its RF carrier. While the video is at baseband, the converter restores the original synchronization information to it. At the same time, the FM sound carrier is demodulated to recover the audio baseband signal. While the audio is at baseband in the converter, the subscriber may use a volume control to adjust its level. The converter then remodulates video and audio onto RF carriers in the NTSC broadcast format. The subscriber's television set is tuned to this RF channel.

Baseband converters may introduce a slight amount of buzz into the BTSC stereo signal, but they always affect stereo separation. Buzz may occur as the video and audio are remodulated onto RF carriers in the converter's output circuits. Video harmonics that were created during envelope detection may spill into the bandwidth reserved for the aural carrier. The subscriber's TV will produce a little more buzz because of these harmonics. However, a quality baseband converter will not add any significant buzz to the audio signal.

Baseband converters affect stereo separation because of the volume control that is available to the subscriber. This volume control (and the demod-remod process in the converter) allows the user to change the deviation of the aural carrier. This changes the amplitude of the pilot tone that reaches the BTSC stereo TV, and more importantly, changes the absolute amplitude of the stereo difference signal. Proper operation of the BTSC system depends on the absolute amplitude of the carrier deviation. As the user changes the position of the volume control, the noise reduction circuit in the BTSC stereo decoder mistracks. Stereo separation is reduced. At the extremes of the volume control range, the stereo pilot may no longer be detected by the subscriber's receiver.

Reduction of stereo separation is tolerable in varying degrees to different people. It has been suggested that acceptable stereo signals are delivered for a useful range of baseband converter volume control levels. [6] The audio from a baseband converter no longer conforms to the BTSC standard. Still, it may well satisfy many subscribers' desire for stereo.

XI. CONSUMER DECODER PERFORMANCE

Consumer decoders can be evaluated on the basis of the stereo separation they can provide, the frequency response of their left and right channel outputs, and their susceptibility to buzz under different audio and video modulation conditions.

Many cable operators intend to test various consumer decoders so that they can provide subscribers with advice about stereo equipment. When testing BTSC stereo equipment, it is extremely important that laboratory quality test instruments be used. This especially applies to stereo encoders or decoders that are used as references for the measurements. Inaccurate or imprecise BTSC test equipment can make the stereo separation of devices under test look much better or much worse than it really is.

Consumer decoders are limited in stereo separation by the precision required to implement the noise reduction system that is used for the difference channel. Filters in the decoders also affect separation. Errors in the noise reduction tracking translate to degraded stereo separation. Stereo separation for some decoders is as good as 25 or 30 decibels, or as poor as 12 decibels. These variations are explained by the newness of BTSC stereo. Improvements occur as manufacturers learn about the unique demands of the format.

Limits on frequency response come from the filter requirements of a multiplex system. Complicated low pass filters prevent the sum and difference channels from corrupting each other in the decoding process. To extend the frequency response of these filters to 15 kHz while preserving separation calls for more expensive designs. As of this writing in February 1987, the left and right channel frequency response of consumer decoders extends to about 13 kHz. Broadcasters provide BTSC stereo with content up to 15 kHz, and some CATV quality BTSC stereo generators also meet this specification. It is expected that consumer decoders with frequency response to 15 kHz will be available as more experience with the format is accumulated.

Buzz performance varies widely among decoder models. It is likely that for the home audience video-related buzz and interference from some types of scrambling will be more of an issue than stereo separation. Manufacturers of decoders and television receivers must use quality receiver design and alignment to minimize interference. Headend operators can prevent severe buzz problems by maintaining correct video modulation depth.

XII. MEASURING BTSC STEREO ON THE CATV SYSTEM

Critical parameters of BTSC stereo performance include signal-to-noise ratio, signal-to-buzz ratio, frequency response, stereo separation, and relative phase between the left and right signals.

On-Line Checks

There are several on-line checks of stereo performance that you can do without interrupting service. These on-line checks provide qualitative indications of performance, and allow you to subjectively evaluate various parameters. Use these procedures to help troubleshoot system failures and recognize major changes in stereo performance.

To perform the on-line checks you need a BTSC stereo decoder, a pair of headphones, and an oscilloscope that can produce an X versus Y display. The BTSC stereo decoder may be of consumer quality. The better the decoder is, the more useful the measurements will be. The decoder must have a "mono/stereo" control. If you are working in a loud environment, headphones with some isolation from outside noises are recommended.

Signal-to-noise may be evaluated by listening to the signal through the headphones. Place the stereo decoder in the "mono" mode. Listen to the dynamics of the program audio. Observe program video. Buzz in the audio may be video dependent. Listen for changes in noise as scenes change. Adjust video modulation depth for possible improvements. Avoid video overmodulation. Note that some noise will originate in the program source material.

Buzz is most easily analyzed during quiet passages or passages with very good stereo separation. Listen to the stereo decoder while switching the decoder between its "mono" and "stereo" modes. If buzz is audible only when listening in the stereo mode, the noise is in the difference channel. Difference mode buzz may be caused by some scrambling systems or by interference from video.

Some frequency response problems will be audible in the headphones. Listen to the stereo decoder in mono mode or to a conventional monaural television receiver. If the audio sounds very tinny and the high frequencies are distorted, the aural carrier modulator may be adding pre-emphasis to the BTSC composite signal. This can only occur in a system where the stereo encoder provides baseband BTSC composite audio to the television modulator. The aural carrier modulator should be set to provide a flat frequency response, not the pre-emphasized response used in monaural transmissions. Pre-emphasis is already provided by the BTSC stereo encoder. Alternatively, check the audio input connections to the stereo encoder. Improper grounding or incorrect terminations may cause frequency response degradations or hum.

The left and right channel should be in phase with each other. Listen to the stereo decoder. Switch it between the mono and stereo modes. The apparent loudness between the modes should not change very much. If the loudness drops severely when listening in the mono mode, the left and right channels may be out of phase. Check the audio path at or before the stereo encoder inputs.

Stereo separation can be checked with the oscilloscope. Use the oscilloscope in the X-Y display mode. Connect the left channel output of the decoder to the "Y" input of the oscilloscope. Connect the right channel output of the decoder to the "X" input of the oscilloscope. Set the input sensitivities to be equal. The scope will now display a "Lissajous" pattern. When the signals on the left and right channels are the same, as during a mono transmission, the dot will move to create a diagonal line on the scope screen. The line will be angled 45 degrees from horizontal, travelling from the bottom left of the screen to the top right. See Figure 21a.

If the left and right channels have equal but opposite signals, they are said to be "out of phase." The display will now show a diagonal line angled 135 degrees from horizontal, travelling from the top left of the screen to the bottom right. Mono receivers will produce little audio output under this condition. See Figure 21d.

During stereo broadcasts, the left and right channels will often have unequal signals. The display will show a "scribbly" pattern, usually tilted toward the axis of in-phase mono programs. See Figure 21e. If the program is supplied in "synthesized" stereo, the lissajous pattern will usually be circular. When the left and right channels are out of phase during a stereo broadcast, the "scribbly" pattern will be tilted toward the axis of the out-of-phase mono programs. Mono receivers will produce little audio output.

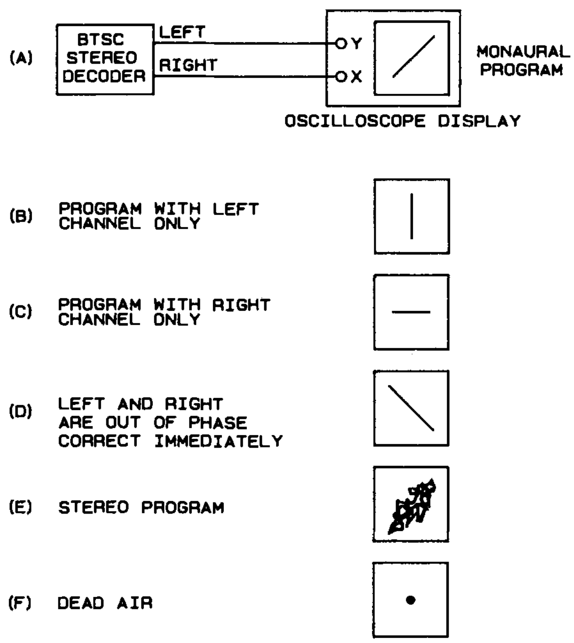


FIGURE 21: LISSAJOUS PATTERNS ON AN OSCILLOSCOPE

A different check for stereo separation in the BTSC system may be performed. It disturbs on-line performance but provides some confidence. Listen to the stereo decoder, or look at the X-Y oscilloscope display. Disconnect the right channel audio input from the BTSC encoder in the headend. The right channel audio should become greatly attenuated as heard in the headphones, while the left channel remains largely unaffected. The oscilloscope display should resemble Figure 21b.

BTSC Proof-Of-Performance Measurements

A complete proof-of-performance requires a precision BTSC stereo decoder, driven by an accurate aural carrier demodulator. A low-distortion audio oscillator and distortion analyzer are necessary. [7]

The proof of performance verifies frequency response, distortion, stereo separation, subchannel crosstalk, deviation calibration, noise performance, and pilot amplitude. A comprehensive proof usually includes tests of the stereo encoder by itself, followed by tests through the entire signal chain. Individual procedures depend on the test instruments being used. In most cases precision stereo decoders are accompanied by detailed instructions for their use in this application.

XIII. CONCLUSION

As cable operators install BTSC stereo on their systems, it is important to have access to experience at other plants. Success in implementing stereo depends on an understanding of its performance limits and where they originate, and knowledge of measurement and troubleshooting techniques.

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