AN-2 STEREO SIMULATOR

User Guide

Issue 3

This User Guide is applicable for serial numbers:
AN2-00251 and later

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Introduction

The Studio Technologies AN-2 Stereo Simulator has numerous applications within recording, live performance, film and broadcast operations because of its ability to create a simulated stereo signal from a monophonic source. The AN-2 is also capable of a number of audio effects centering on its ability to simulate stereo. The stereo signal produced by the AN-2 is completely mono compatible, so that it can be remixed back to mono without the serious signal degradation often encountered when remixing simulated stereo produced by other equipment such as delay lines and harmonizers. In addition, the AN-2 is equipped with a proprietary circuit that randomizes the non-reclusive filtering process, thereby avoiding the harmonic harshness that often results from stereo simulation with delay lines.

Operational Description

To understand the operation of the AN-2, it is useful to have a basic understanding of the way stereo recording techniques can fool the brain into believing it is hearing a number of sound sources.

When two or more microphones are used to record music, two effects are noticed when the recording is played back over two speakers. They are:

- Directional effects: These are due to differences in arrival times at the ears and the intensity differences of the various frequencies that make up the music.
- Spaciousness imparted to the sound: The sound is perceived as originating from an area that is wider than the room in which the listener is sitting. This spaciousness is caused by the interaction of the sounds that reach the microphones directly and those sounds that reach the microphones after being reflected from the surfaces of the recording studio.

The degree of spaciousness of the recorded music coming from the two speakers is dependent on the amount of coherence (sameness or consistency) between the signals fed to the speakers.

If the two channels are perfectly coherent (the same signal fed to both speakers), the sound heard is monaural and the source of the sound seems to be midway between the two speakers. As the signal fed to one speaker begins to vary from the signal fed to the other, the channels become incoherent and the spaciousness of the sound increases. This effect is known as “imaging,” and results in the impression that each instrument or voice is coming from a different point in space. The definition and clarity of the sound sources seems to increase as the imaging increases, up until the point where the two channels become completely incoherent, or different. At this point, the stereo image collapses and there is no spacial effect at all. What is heard, rather, are two mono channels.

The AN-2 is designed to allow you to vary the degree of coherence of the sound (and the resultant amount of imaging) using the depth and width controls, creating a simulated stereo signal from a monophonic source. The two output signals of the AN-2 can be recombined, restoring mono.

Mono Compatibility

In many applications, especially recording and broadcast, it is critical that any stereo signal produced can be summed (L+R) without adverse effects. The AN-2 was designed so that the simulated stereo signal produced is completely mono compatible. No setting of the controls or circuit anomalies ever produce a problem in this area.

The AN-2 uses the classical technique of creating comb filters, as shown in Figure 1. The transfer function of the upper network is $H(z) = 1 - z^{-n}$, resulting in dips at 1/T, 2/T, 3/T, etc. The lower network is merely the complement of the upper; where the upper network has an amplitude peak, the lower has an amplitude dip. If the two outputs are summed, it can easily be seen that the delay line is cancelled out, leaving only the direct, or mono, unaltered signal.

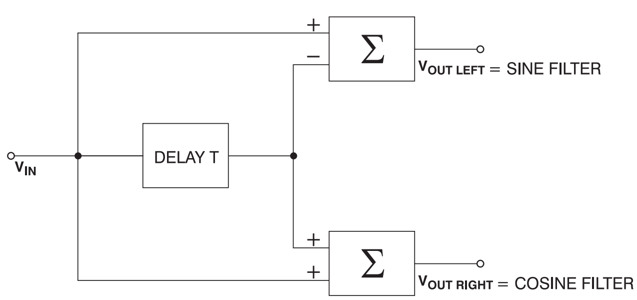

Figure 1

For musical reasons, the AN-2 uses a network after the delay line to randomize the peaks and dips. However, the peaks and dips are still interleaved and the mono sum is also the same as the input signal.

Installation

The AN-2 is rack-mountable, requiring one 1.75-inch (4.45cm) rack space. It weighs five pounds (2.27kg), and operates on either 115Vac or 230Vac. Refer to the Specifications section for more complete electrical and physical specifications.

Placement

If you wish to mount your AN-2 near other equipment, make certain that it is not near the power transformer of another unit. The AN-2 is well shielded, but some of the circuitry is sensitive to high levels of EMF from outside sources.

Connecting the Unit to Power

The AN-2 may be operated from either 100-125 or 200-250Vac power, 50/60Hz. Units shipped to North America are factory selected for 115V operation and are supplied with a 0.5A 3AG fuse. Units shipped outside North America are factory selected for 230V operation and supplied with a 0.25A 5 x 20mm fuse. Before connecting the AN-2 to power, determine the actual line voltage and check to see that the voltage selector switch (located on the back panel of the unit) is set to the appropriate voltage. Please note that an incorrect setting could seriously damage the unit. Should it be necessary to change the voltage selection, insure that the correct fuse value is used: 0.5A for 115V, 0.25A for 230V.

The AN-2 utilizes an IEC standard connector to mate with the line cord. The line cord supplied has a North American standard plug at one end and an IEC connector at the other. In non-North American applications the plug must be cut off and an appropriate plug attached. The wire colors in the line cord conform to the internationally recognized CEE color code and should be wired accordingly.

Connection	Wire Color
Neutral (N)	Light Blue
Live (L)	Brown
Earth or Ground (E)	Green/Yellow

Audio Signal Connections

The input and output connectors of the AN-2 are 3-pin XLR type. The input is female and the outputs are male. Shielded cable should be used when making connections. Proper phasing is very important if you want to be able to remix the simulated stereo back to mono. If the AN-2 is to be used with unbalanced equipment, pins 1 and 2 of the outputs must be shorted together at the connectors which plug into the back of the unit.

Operating Level

The AN-2 is shipped from the factory configured for input and output levels of +4dBm. The unit can be set for –10 or 8dBm operation.

Use with –10dBm Semi-Pro Equipment

If you wish to use the AN-2 with –10dBm equipment, a special switch setting must be made.

1) Disconnect the power cord.

2) Remove the top cover (two Phillips-head screws on each side panel).

3) Locate the 4-position DIP switch on the circuit board (slightly to the right of center and labeled SW1).

4) For use with a –10dBm input, set switches 3 and 4 to the OFF position.

5) To achieve a –10dBm output, set switches 1 and 2 to the OFF position and short pins 1 and 2 of the output connectors together.

To return the unit to +4dBm operation, place all the switches in the ON position and unshort pins 1 and 2 of the output connectors.

Conversion to +8dBm Level

For input and output levels of +8dBm, a qualified technician must remove and replace seven resistors on the AN-2 circuit board. Refer to the schematic for parts values and locations. It is important that the new resistors be 1% tolerance. A resistor kit for –8dBm conversion is included with the unit.

Operation

Using the Controls

There are seven controls and one meter on the front panel of the AN-2.

The Power switch controls AC power to the unit.

The Gain control allows you to control the input level, which is especially useful if you are using a multi-track tape recorder which lacks a gain control of its own. The peak-reading meter (labeled “Level” on the front panel of the unit) monitors the overall gain structure. The gain control should be set so that the meter’s yellow and green LEDs are lit in the presence of a signal, with the red LED lighting only on signal peaks. Attention to the peak-reading meter will give you the best signal-to-noise ratio, while preventing overload of critical circuitry.

The other four controls allow you to tailor the stereo simulation to your tastes.

The Stereo Width control varies the number of combs in each channel (refer to Figures 2 and 3). This changes the apparent spread of the sound source. A single voice or instrument should sound as if it originates from a single point, while a group of voices seems more natural when their sources are apparently separate. Turning the control fully clockwise increases this spreading effect.

The peaks and dips in the frequency response curves shown in Figures 2 and 3 may seem abnormal when compared to other equipment. Keep in mind that the effect you are seeking is a result of the interaction of the direct signal and the signal reflected from the walls of a room. The curves shown in Figures 2 and 3 approximate this interaction.

The Stereo Depth control affects the depth of the combs. When rotated to the mono position (fully counterclockwise), the combs have no depth at all, giving a flat frequency response. Rotated to full stereo (clockwise), the combs reach their maximum depth, giving the maximum stereo effect.

This maximum effect may not be appropriate for your program material. If your signal has few pitch changes (a single speaking voice, for example), the combs may be audible. Setting the stereo depth control to eleven o’clock will give a natural stereo effect without audible comb filter effects. More complex sources, such as a vocal group or full orchestra, will benefit from the enhanced depth resulting from turning the stereo depth control more fully clockwise.

The Modulation ON/OFF switch enables and disables the modulation section. In the out (OFF) position the modulation depth and rate controls do not affect the signal.

The modulation controls (depth and rate) produce an instantaneous change in pitch similar to vibrato. The rate control varies the speed of these changes from a slow “chorus” to a fast warble. The depth control varies the amount of the effect.

The setting of the modulation depth control is related to the settings of the stereo width and stereo depth controls. When the stereo width and depth controls are set to a low position (counterclockwise), the modulation depth control should be raised (clockwise), and vice versa.

The settings of all the controls are closely interrelated. The best way to learn how the unit sounds, of course, is to experiment. For example, a string section might be best enhanced with the stereo width set at two o’clock, a stereo depth setting of one o’clock, modulation depth at three o’clock, and a minimum setting of the modulation rate control. The settings outlined in this and preceding paragraphs will give you a good starting point; your finished product will depend largely on your imagination.

Recording Studio Applications

Although it may be placed almost anywhere in the recording chain, for best results the AN-2 should be between the source and the final stereo mix.

When used with a typical console, the unit should be fed from a track on the tape recorder, or an unused cue or echo bus on the console. The outputs of the unit could then be fed to a pair of console faders panned hard left and hard right. It is neither necessary nor desirable to add any source signal to the mix, as the stereo depth control will accomplish this. Any equalization should be placed upstream of the AN-2 (between the source and the unit).

Although the AN-2 was designed primarily to simulate a stereo signal from monaural sources, it has many practical applications in the recording studio for the enhancement of various vocal and instrumental signals.

For example, it is well known that using a stereo pair of microphones to record a soloist is subjectively more pleasing than using a single microphone. Practical considerations usually make this recording configuration difficult because small movements of the soloist can result in disconcerting jumps when heard over two loudspeakers. The usual solution is to pan the soloist center, but this creates the totally unnatural condition of forcing the listener to “merge” two identical signals coming from different points in space. The AN-2 avoids both of these problems by creating a stable stereo image from a mono source.

The AN-2 can also be used to create a subjectively “louder” signal from a mono source, making it ideal for use with instruments such as bass guitar. How often have you heard the words “more bass” from the producer when the instrument is already panned center and the meter is “in the red”?

Another application for the AN-2 in the recording studio is to convert “tape slap” to stereo; that is, to feed the output of the delay line or tape machine into the AN-2 and mix the outputs of the AN-2 into the stereo buses. The outputs of flangers, chorus devices, echo chambers, and other effects boxes can be converted to stereo in the same way. Not only does the AN-2 enhance the effect, but the result is assured of perfect mono compatibility. This is especially important for commercials, which will most always be heard on radio and TV in mono. This assures that the balance between for example, the reverb and the program, will not change.

Hook-ups using external equalizers, mixers, and filters may yield pleasing results that you may not have expected. One example would use cross-coupled equalizers after the AN-2 outputs. The outputs of both the EQs and the AN-2 are then mixed to form two new output which are fed to the stereo buses. The equalizers should have identical boost and cut characteristics, or a mono signal generated alter will suffer.

In practice, the boosted frequencies would form center images, while the cut frequencies would have more stereo “spread.” Set the stereo depth control fully clockwise for this hook-up.

Remember: When using a non-standard connection, any operation or effect that is performed on one channel must be equally performed on the other. If not, mono compatibility will suffer.

As you become familiar with the operation of the AN-2, you might want to try more experimental uses, such as putting up an ambience microphone in the studio during a session and recording it on an extra track. During mix-down, feed it through the AN-2 into your stereo mix. The result will be a much more “live” feel to the sound.

Broadcast and Film Applications

Television stations that are in the process of converting to stereo can use the AN-2 to process mono program material to stereo, allowing full-time stereo programming. For program material of a general nature containing speech and music, the stereo depth control is best set about mid-range and the stereo width control about three o'clock. Programs consisting primarily of music can benefit from more opening of the controls, while programs that are mainly speech should be processed with the controls less open. The modulation control should not be used for "on air" broadcasting. Leaving the modulation ON/OFF switch in the OFF position disables the modulation section; if you want to be absolutely certain that the modulation section will not be accidently turned on, remove U15 (see schematic) after disconnecting power to the unit.

AM and FM stereo radio stations will find use for the AN-2 in converting old records to stereo or "fattening" D.J. and announcer voices. Again, careful use of the stereo depth and width controls can result in optimum sound. Also, many commercials are received by the station in mono and can be enhanced by conversion to stereo.

The AN-2 can be employed in film studios to remix mono movie sound tracks to stereo. Older music and sound effects libraries can also be upgraded.

Theory of Operation and Troubleshooting

The following paragraphs describe the theory of operation of the AN-2. This information will help you understand where a failure may be located. Please read this material before attempting to troubleshoot the unit.

Theory of Operation

Refer to the schematic diagram of the AN-2 while reading this material.

U9 is a balanced input amplifier with a trim potentiometer to obtain maximum common mode rejection. Two sections of the DIP switch, SW1, are used to select –10 or +4dBm operation.

P5 is the input gain control, which is followed by a buffer amp. This amp sends a portion of the input signal to U5, where the two comb filters are produced by phase addition and subtraction performed on the delayed signal.

A pre-emphasis network precedes U11, which acts as a compressor in a compandor circuit. The compressor attack time is speeded up by a charge pump U14, which reduces transient distortion that is often associated with compandors.

A 6-pole, 20kHz Butterworth filter removes the possibility of audio frequencies aliasing with the clock frequency from U1.

U6 acts as a bucket brigade delay line. The line is driven by U2 and VCO U1, whose frequency is set by the stereo width control, P1.

U15 is an LFO which modulates the VCO via P2 (modulation depth control) and P3 (modulation rate control).

A 5-pole, 20kHz low-pass Butterworth filter removes clock frequency from the output signal.

The expander, U11, restores the dynamic range, using another charge pump, U3, in the expander rectifier.

A proprietary randomizing circuit, labeled N1, assures that the peaks and dips in the combs do not fall on harmonics, preserving the natural sound of the music.

U17 comprises the peak-reading meter circuit; it combines a rectifier and three LED drivers.

U5 is a mixer, interleaving the combs on the left and right outputs. If P4, the stereo depth control, is set fully counterclockwise, any delay signal is removed, leaving only the direct signal. This will also happen if the two outputs are summed externally, since the delay signal will be cancelled. The switches following this IC set the output levels.

The two output stages, U7 and U8, are differential balanced line outputs with cross coupled output sensing feedback. This circuit gives an additional 6dB of headroom for a given supply voltage, while maintaining the output level and keeping the opamp current draw at a safe level, even if one side of the output is shorted.

The power supply is bipolar, using two IC voltage regulators to provide low ripple ±15Vdc. Two additional decoupling stages provide isolation for the clock circuitry.

Equipment Needed for Troubleshooting

The AN-2 is constructed of high-quality solid state components. Given normal use, there is no need for regular preventive maintenance.

If repair does become necessary, you may either have a qualified technician service the unit or return it to the factory. Contact the factory for a return authorization number should you wish factory repair. When returning your unit, be sure to pack it securely and insure it for its replacement value.

If you decide to have a technician service the unit, the following test equipment is required to allow accurate troubleshooting a component replacement:

- high-quality oscilloscope equipped with balanced vertical and horizontal inputs
- low-distortion oscillator (.01%)
- harmonic distortion analyzer
- AC voltmeter with –80dBm sensitivity
- high-quality digital voltmeter
- adjustable attenuator box
- tone burst generator
- frequency counter

Troubleshooting and Alignment

WARNING: The following procedures must be performed by a qualified technician. Operating the AN-2 with the top and/or bottom cover removed exposes the technician to points in the power input section with hazardous voltages. USE CAUTION!

The first step in troubleshooting the AN-2 is to remove the top cover (two Phillips-head screws on each side panel), and check the DC voltages at the test points listed on the schematic. If this technique fails to pinpoint the trouble, connect an oscillator to the input and trace the signal (again, using the schematic) until the source of the problem is found. The following paragraphs will aid in aligning the AN-2 circuits, which will restore proper operation. It is important to perform the alignment procedures whenever you have replaced a component in the unit.

CAUTION: DO NOT attempt to adjust any of the trim pots inside the unit before you understand the entire alignment procedure. Improper or out-of-order adjustment may destroy some of the integrated circuits.

U9

To achieve the best common-mode rejection, feed a 1kHz signal at +4dBm into the input connector: high side to pins 2 and 3 (short 2 and 3 together), and low side to pin 1. Check the output at TP9, and set R89 for a minimum signal.

U3 and U14

Set R110 to 1.34Vdc at TP8 with no input signal. Feed a 1kHz input signal at +4dBm, set gain control to read –10dBm at TP9, and read the AC signal at TP7. Drop the input signal exactly 20dB. The signal at TP7 should drop 15dB. If not, reset R110. Repeat this procedure several times, as R110 also affects the reference level.

U1

The clock frequency of this VCO should vary between approximately 100kHz and 375kHz, depending on the setting of P1 (stereo width) when measured at TP1 (clock set). It should produce approximately 12V, measured peak-to-peak, at TP1. Connect frequent counter between TP2 (clock set) and common. Set P1 (stereo width) fully clockwise. Set SW2 (modulation ON/OFF) to the OFF (out) position. Adjust R97 so as to read 62kHz on the frequency counter. If it is ever necessary to replace U1, use only the part manufactured by National Semiconductor.

U3 and U11

Set R5 to give 1.34Vdc at TP5 with no input signal. Feed a 1kHz input signal at +4dBm, with gain control set as before, and read the AC signal at TP6. Drop the input signal exactly 20dB. The signal at TP6 should drop 20dB. If not, reset R5. Repeat this procedure several times.

U6

Set R46 for minimum THD with P1 (stereo width) set to midpoint. The input for this procedure should be 5kHz at +4dBm.

U11

Using the tone burst generator, feed an input signal to the unit of 1kHz at + 4dBm, and set the gate duration for 2 cycles open and 64 cycles closed. Set R82 for a flat base band. If a tone burst generator is not available, set R82 to midpoint.

Verification of the specifications is accomplished by standard procedures with the exception of the “noise” and “THD” measurements. Since the outputs are combs, P1 (stereo width) should be rotated to give the maximum possible output for the frequency chosen. This measurement becomes the reference level below which the noise should measure 70dB. THD should be measured in the same way.

Specifications

Mounting
One space in a standard 19-inch (48.3cm) rack

AC Mains Requirements
100 to 125Vac or 200 to 250Vac, switch selectable, 50/60Hz, 10 watts

Fuse
0.5A for 115V, 0.25A for 230V; units shipped to North America use 3AG type; units shipped outside North America use 5 x 20mm type

Connectors
Input: 3-pin XLR-type female, pin 3 hot
Outputs: 3-pin XLR-type male, pin 3 hot

Input
Balanced bridging, differential amplifier

Input Impedance
100k ohms, balanced
50k ohms, unbalanced

Maximum Input Level
+20dBm (re 0.775V RMS)

Frequency Response
±1dB, 30Hz to 15kHz, delay section
±1dB, 30Hz to 30kHz, direct section

Dynamic Range
90dB

Noise
70dB below operating level

Gain
8dB

Outputs
Balanced, differential amplifiers

Output Impedance
100 ohms

Maximum Output Levels
+26dBm into 10k ohms balanced (re 0.775V RMS)
+24dBm into 600 ohms balanced (re 0.775V RMS)
+22dBm into 10k ohms unbalanced (re 0.775V RMS)
+20dBm into 600 ohms unbalanced (re 0.775V RMS)

Distortion
Less than 0.2% at maximum operating level

Environment
Operating: 32 degrees F to 122 degrees F
0 degrees C to 50 degrees C
Storage: –4 degrees F to 140 degrees F
–20 degrees C to 60 degrees C

Dimensions (Overall)
19.00 inches wide (48.3cm)
1.75 inches high (4.5cm)
7.00 inches deep (17.8cm)

Weight
5 pounds (2.3kg)

Specifications and information contained in this user guide are subject to change without notice.

Studio Technologies Limited Warranty

The Studio Technologies, Inc. product which you have purchased is guaranteed against defects in material and workmanship under normal use and service for a period of three (3) years from date of manufacture.

During this warranty period, any components of the product that prove to be defective will be repaired (or, at the discretion of Studio Technologies, Inc., replaced) at no charge provided that the product is returned, shipping prepaid, to the manufacturer.

Information on obtaining service is available by contacting the dealer from whom the product was purchased, or by contacting Studio Technologies, Inc.

This warranty does not apply if, in the opinion of Studio Technologies, Inc., the product has been damaged due to abuse, misuse, misapplication, accident or as a result of service or modification by other than an authorized dealer service center.

No other warranties are expressed or implied, including, but not limited to, any implied warranties of merchantability and fitness for a particular purpose. Studio Technologies, Inc. shall not be held responsible for any consequential damages or losses arising from the use of this product.