Generation II

User Guide

Generation II
Stereo Simulator

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Introduction

General Description

The Studio Technologies Generation II Stereo Simulator is designed to create a simulated stereo signal from a monaural input. It is intended for use in applications such as television and radio broadcasting, film production, and audio restoration. Specialized circuitry converts the incoming signal into a time-domain randomized audio field which provides an excellent stereo "feel."

In television applications, the Generation II is appropriate for both on-air and production use. The MTS television broadcaster is faced with the reality of having a large amount of mono audio material to broadcast. Using the Generation II Stereo Simulator can greatly improve the listener's appreciation of "stereo" TV. The output is mono-compatible, ensuring that monaural (L+R) listeners are not negatively impacted by the use of the Generation II.

In film and video production use, the Generation II is invaluable in converting mono audio into realistic stereo. The two modes of operation, music and music & voice, allow for correct processing in all audio scenarios. In the music & voice mode, a specialized filter is used to give greater simulation to all frequencies except those in the voice band. This ensures that excellent voice centering is maintained.

In some cases, it is desirable to disable the front panel controls. An authorized person may want to optimize the "sound" of the Generation II, and then restrict the changing of those settings. A 3-position switch, located inside the unit, allows the operating status to be set. In the fixed—music setting, the music mode is activated, the front panel controls are disabled, and the stereo intensity trim potentiometer is active. In the fixed—music & voice setting, the music & voice mode is activated, the front panel controls are disabled, and the stereo intensity trim pot is active.

Careful circuit design provides excellent audio performance and operational characteristics to match the special requirements of most facilities. The line input stage is electronically balanced, maintaining excellent common-mode rejection, low noise, and low distortion. The audio output stages use advanced circuitry to provide high output level, low noise, and low distortion. Two switches allow the input and output levels to be set for nominal 0, +4, or +8dBu operation. Two 3-section LED meters, along with a meter select switch, provide monitoring of the input and output levels.

Installation

Overview

In this section you will:

- Remove the top and, if required, the bottom cover
- If required, install an input load resistor
- Set the input, output, and system status switches
- Replace the cover(s)
- Mount the Generation II
- Connect the audio signals
- Connect power
- Adjust the input level

Remove the Cover(s)

The following procedures must be performed by a qualified technician. Operating the Generation II with the top and/or bottom cover removed exposes the technician to points in the power input section with hazardous voltages.

Ensure that the power cord is removed from the power connector on the back panel.

Remove the top cover. Four Phillips head screws hold the top cover to the side rails.

If an input load resistor is going to be installed, remove the bottom cover. Four Phillips head screws hold the bottom cover to the side rails.

Please Note: Do not touch any potentiometer on the circuit board or factory calibration may be required.

Input Load Resistor

The input impedance of the Generation II is 25k ohms. In some cases, an input termination resistor, with a value such as 150 ohms or 600 ohms, may be desired. Provision has been made for an input resistor to be installed in the Generation II's circuit board. The resistor is shown on the schematic connected to pins 2 and 3 of the input connector. It is labeled R2 on the circuit board's legend. Carefully insert the desired resistor into the circuit board and solder the connections. Using diagonal cutters, trim the excess resistor lead. You may want to label the back panel with the revised input impedance.

Configuration Switches

The Generation II is designed to accept nominal audio signal levels of 0, +4, or +8dBu. Set the desired input level using the 3-position switch located on the left side, toward the back of the circuit board.

The Generation II is designed to produce nominal audio signal levels of 0, +4, or +8dBu. Set the desired output level using the 3-position switch located in the center, toward the back of the circuit board.

The system status switch sets the way the Generation II operates. This switch is located on the left side, near the front of the circuit board.

In the front panel position, the front panel controls are enabled.

In the fixed — music position, the front panel controls are disabled, and the unit operates in the music mode.

In the fixed — music & voice position, the front panel controls are disabled, and the unit operates in the music & voice mode.

Replace the Cover(s)

Carefully replace the top and, if removed, bottom cover using the screws that were previously removed.

Mounting

The Generation II Stereo Simulator is rack-mountable, requiring one standard rack space (1.75 inches) in a standard EIA 19.00 inches rack. Keeping the Generation II relatively cool will enhance operating life. Most electronic equipment failures are power-supply related. Power supplies tend to generate heat which, when not adequately controlled, toast the power supply, dry out electrolytic capacitors, etc. Keeping all equipment relatively cool will reduce the likelihood of problems occurring.

Signal Connections

All signal connections to and from the Generation II are made via 3-pin XLR-type connectors located on the back panel; one female for line input, and two male for left and right line output. For hum, noise, and radio frequency pickup rejection, shielded cable must be used. The shield should be connected to pin 1 of the input and output connectors. Pin 1 is electrically in common with the Generation II's power supply ground, chassis ground, and power cord ground leads. Following the European convention, pin 2 is audio high and pin 3 is audio low. Maintaining consistent audio input and output polarity is important for correct performance. Failure to maintain consistent left and right output polarity will make the output audio not mono-compatible.

The Generation II uses electronically balanced input and output circuitry. Best performance is achieved if the equipment sending signal to and receiving signal from the Generation II is operating in a balanced mode. The input and output stages will operate correctly in an unbalanced mode, but selected performance characteristics will be sacrificed. On the input side you will lose the ability to reject common-mode signals that balanced operation affords. On the output side, you will lose 6dB maximum output level. This is not a technical fault but is inherent in electronically balanced output stages. If unbalanced input operation is required, strap the audio low to ground (pin 3 to pin 1). Connect input audio high to pin 2 and signal ground to the combined pins 1 and 3. If unbalanced output operation is required, strap pin 3 to pin 1. Connect the output audio high lead to pin 2 and signal ground to the combined pins 1 and 3.

In broadcast settings, it is recommended that the Generation II's input and output signals be connected via patch points. Installation, testing, and servicing procedures will be greatly improved if the unit can be easily taken off line. Make sure that the Generation II can be "patched around" during testing and maintenance.

The Generation II's line outputs are low impedance, electronically balanced, line level. They are capable of driving virtually all line inputs (low or high impedance, transformer or transformerless).

In most cases, transformer coupling between audio equipment is neither required, nor desirable. If the equipment sending signal to, or receiving signal from the Generation II requires the isolation given by a transformer, but does not contain internal transformers, external transformers can be added. Refer to the Technical Notes section of this manual for recommended transformers.

Connecting to Power

The Generation II may be operated from either nominal 115Vac or 230Vac, 50/60Hz. Units shipped to North America and Japan are factory selected for 115V operation and are supplied with a 0.50A 3AG fuse. Units shipped outside North America or Japan are selected for 230V operation and are supplied with a 0.25A 5x20mm fuse. Before connecting the unit to power, determine the actual line voltage and check to see that the voltage selector switch located on the back panel is set to the appropriate voltage. If the voltage selector switch is set for 230V, a 0.25A fuse is required; 115V requires 0.50A.

Caution: An incorrect voltage setting and/or incorrect fuse could seriously damage the unit.

The Generation II uses 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
Protective Earth (E)	Green/Yellow

The Generation II does not contain a power switch. The unit will start to operate as soon as power has been connected.

Input Trim Adjust

Once the unit has been installed and is operating, the input trim potentiometer may be adjusted. This control is accessible from the back panel, directly next to the input connector. The trim pot allows the input level to be adjusted over a small range, approximately ±0.75dB. Use this control only if you need to precisely match the signal levels in your audio chain. In most installations, the pot can be set at 50 percent rotation, and then forgotten about. The Generation II's electronically balanced output stages have slight output level changes as a function of load impedance. It is normal to find a 0.35dB drop in output level when driving a 600 ohm load, versus driving a 20k ohm load. The input trim pot can be set to "make up" this loss. Do not use a sine wave input signal when attempting to perform this, or any other Generation II adjustment. A single frequency input signal produces output signals that have level characteristics very different from those produced by normal audio input signals. When testing, it is best to use normal program audio or pink noise.

Operation

Using the Controls

There are two switches and one potentiometer on the front panel of the Generation II. In addition, there is a trim potentiometer accessible via an opening in the front panel. There is no power switch; the Generation II is designed for continuous operation. Not having a power switch means that the unit cannot be accidently turned off.

Mode Switch

The mode switch controls the overall sound of the Generation II. The mode switch is active only when the internal system status switch is set to the front panel position. In the music position, the full audio bandwidth of the input signal is simulated. This mode is appropriate for audio material with little or no voice only content. In the music & voice position, the input signal is sent through a band-reject filter prior to getting sent to the simulator circuits. This creates stereo simulation over the low and high frequency range, while limiting simulation in the voice band. This mode is appropriate for audio material that contains voice only content, such as a film track or television show.

Meter Switch

The meter switch selects which signals are monitored by the two LED meters. In the input position, the left meter displays the line input signal and the right meter is not active. In the output position, the left meter displays the left line output and the right meter displays the right line output.

Stereo Intensity Control

The stereo intensity control determines the amount of stereo simulation that is produced. The stereo intensity control is active only when the internal system status switch is set to the front panel position. In the fully counterclockwise position, no simulated stereo is produced. The mono input signal is sent equally to the left and right outputs. As the control is turned clockwise, the amount of stereo simulation increases. The size of the stereo image that is produced is directly related to the setting of the stereo intensity control.

Stereo Intensity Trim Control

To the lower left of the stereo intensity control is a small opening in the front panel. This opening allows access to a trim potentiometer. This trim pot controls the stereo intensity when the internal system status switch is set to the fixed — music or fixed — music & voice positions. This trim pot allows the sound of the Generation II to be adjusted, but “hides” the control from unauthorized personnel.

Panel Controls Disabled LED

On the left side of the front panel is a yellow panel controls disabled LED. This LED is lit when the internal system status switch is set to the fixed — music or fixed — music & voice positions. It alerts the user that the front panel mode switch and stereo intensity potentiometer functions are not active. This also indicates that the stereo intensity trim potentiometer is active.

Mode LEDs

The operating mode of the Generation II is displayed by two red LEDs labeled music and music & voice. If the panel controls disabled LED is lit, the fixed operating mode will be displayed via one of the two LEDs. If the front panel controls are active, the LEDs will follow the operation of the mode switch.

Audio Signal Level LEDs

Two sections of the LED meter display the audio input and output levels. These meters are not intended to be calibrated, VU- or peak-type, but strictly to provide a relative level indication. The green LEDs show the presence of low level signal in the Generation II. The yellow LEDs show the presence of moderate signal levels. The red LEDs light when the signal level approaches, within about 10dB, the internal clipping level.

Normal Operation

You should now feel confident that the Generation II Stereo Simulator has been carefully installed and is ready to go to work. With normal audio input signals, the green and yellow LEDs should light most of the time. On peaks, the red LEDs may light. However, if they light steadily, you should reduce the level of the input signal. If you have any questions concerning the unit, now is a good time to give us at Studio Technologies a call. Otherwise, set the controls for your desired simulated stereo sound and that’s it.

Technical Notes

Definition of Level

Studio Technologies has opted to use the dBu designation as it seems to be quite rational. Using dBm was fine when all audio line outputs were terminated with 600 ohm loads. In this way, it was easy to say that 0dBm is 1 milliwatt dissipated in the known load (i.e., 0dBm across 600 ohms will measure 0.7746V). In contemporary situations, an audio output line is rarely terminated in 600 ohms; generally it's 20k ohms or higher. The dBu designation is a better reference because it refers to dB referenced to 0.7746V, with no reference to load impedance. This takes into account today's audio scene with load impedances varying greatly. When the Generation II's specifications refer to the maximum output level in dBu, this would translate to dBm only when the output is terminated with 600 ohms.

Transformer Coupling

As discussed in the Installation section of this manual, there may be cases where transformer isolation of the Generation II's line input and/or output signals may be required. As the unit has excellent electronically balanced input and output circuitry, interfacing with most other equipment should not require transformer isolation except in the strangest of cases. If you do find a case where you need them, we suggest using transformers from Jensen Transformers Incorporated, 10735 Burbank Blvd., North Hollywood, CA 91601, telephone (213) 876-0059. They make excellent parts and we recommend their JE-11-DMCF, a 600 ohm to 600 ohm output type transformer. This should give very good results on both the input and output sides.

Resistor Loading

Some audio and broadcast operations maintain as standard practice termination resistors on all audio lines. This is usually done at the input stage of all pieces of audio equipment. This is not necessary with the design of most contemporary audio equipment. Current practice calls for low source impedances and high input impedances. The Generation II follows this practice. A problem can occur if you choose to load the line input and output signals connected to the Generation II. Double loading can occur if you have wired the Generation II via a patch bay and you patch around the unit. The output of the device driving the Generation II will now be driving the two line inputs of the device connected to the outputs of the Generation II. This double loading may reduce the audio level being sent to the next piece of audio equipment. The best solution is to load neither one. In fact, loading the Generation II's line outputs does nothing to improve performance. It basically just wastes output current. Be aware of this situation if you decide to use load resistors.

Non-standard Input and Output Levels

Studio Technologies designed the Generation II to match the audio operating levels of most facilities. If you are the creative type who likes to experiment, or the unlucky one who inherited a facility and has an operating level different from 0, +4, or +8dBu (ref. 0.7746V), don't despair. The 0, +4, or +8 settings are used to optimize the Generation II's performance, and using a slightly different operating level will make only minor differences in performance. If you are within 2dB of one of the Generation II's preset levels, set the input and output switches to the closest value. If you are exactly in between two of the choices, go for the lower one. An example would be a facility running +6dBu. Set the input and output levels for +4. If you are running an operating level below –2dB or above +10dB, contact Studio Technologies for details on simple operating level modifications.

Proof of Performance Testing

Proof of performance tests on a broadcast facility commonly use sine waves of different frequencies and levels to check such things as frequency response, noise, and distortion. Very unusual results can occur if Generation II performance tests are made using sine wave or constant frequency signals. These results are due to the way the unit creates simulated stereo, taking a mono input and delaying, randomizing, and gyrating the signal into a good stereo image. Simulating stereo from a fixed frequency input will result in different left and right output levels that will appear to change randomly as the input frequency is varied slightly. This is completely normal and expected. Remember that the usual input signal is complex music, voice or other natural sound.

To get rational proof of performance data, either take the unit out of the signal path, or reduce the stereo depth control to the minimum, fully counterclockwise position. If you wish to check The Generation II with stereo simulation taking place, you need to use pink noise. The randomness of pink noise will allow left versus right frequency response observation.

Circuit Description

In this section, we will review the Generation II's circuitry. This information will assist you in understanding the operation of the unit, as well as providing a guide to troubleshooting any problems that may arise. Operational amplifier sections of an integrated circuit will be referred to as an "op amp section," or as an "op amp."

Power Supply

A step down transformer converts the incoming line voltage to nominal 40Vac. The primary of the transformer is tapped for either 115Vac or 230Vac input. A switch on the back panel selects which tap is connected. A fuse in series with the primary protects the unit in case of a component malfunction. The Generation II contains no power switch. The power line is connected directly, via the fuse, to the transformer. This prevents the unit from being accidently turned off. The secondary of the transformer has a center tap, which acts as the circuit common, as well as being strapped to the metal chassis and line cord ground wire. The secondary of the transformer connects to a full wave bridge rectifier. The full wave bridge produces nominal ±30Vdc. From the unregulated ±30Vdc two fixed voltage 3-terminal regulators produce ±15Vdc. A resistor and zener diode combination produces –22Vdc. The audio circuitry uses ±15Vdc, and the delay line oscillator FET switch uses –22Vdc. An LED indicator, in series with a resistor, is connected to +15Vdc to provide an indication that the unit has incoming power.

Meter Circuit

Two, three-stage LED level meter circuits provide the user with an indication of several Generation II internal operating levels. The meters are not intended to give a precise reading, but simply to provide general level indications. The right meter displays the level of the right channel signal just prior to the right channel line driver stage. The input to the right meter comes, via the meter select switch, from the output of the op amp that serves as a right channel difference/gain section. The left meter is switch-selectable to display the level of the input signal or the level of the left channel signal just prior to the left channel line driver stage. The input to left meter comes from either the output of the line input section, or the op amp that serves as the left channel summing/gain section.

Op amps configured as half wave rectifiers convert the incoming meter audio into DC. This DC voltage is fed to three sections of op amp which are configured as voltage comparators. Voltage dividers provide reference voltages for the comparators. The comparator outputs act as current sinks to light the LEDs. The LED current is resistor limited to 10mA for the green and red LEDs, and 15mA for the yellow LEDs. The different currents compensate for the different LED efficiencies.

The green LEDs have a low lighting threshold, making them act as “signal present” indicators. The yellow LEDs are set to light with moderate signal levels. The red LEDs are set to light when internal signal levels approach clipping.

Line Input

The purpose of the line input stage is to take a balanced audio input signal, reduce its level by a fixed amount, and convert it to unbalanced. For best operating performance, the internal operating level of the Generation II is –6dBu. This level combines a good signal to noise figure with ample peak signal headroom for circuits operating with ±15Vdc power sources. Since different facilities run different average operating levels, the line input stage is switch-selectable for 0, +4, or +8dBu operation. This translates to attenuating the input signal by 6, 10, or 14dB.

Signal enters the Generation II via a female XLR-type connector. A resistor position has been left vacant on the circuit board if an input load resistor is desired. The input signal is direct coupled to a differential line receiver integrated circuit. This IC is truly optimized for audio applications, providing many fine features, including excellent common-mode rejection, low distortion, and low noise. The now unbalanced signal enters a passive attenuator section consisting of a 3-position switch, fixed resistors, and a trim pot. This attenuator reduces the signal level to the nominal –6dBu internal operating level from the nominal input levels of 0, +4, and +8dBu. The trim pot allows the input level to be adjusted over a ±0.75dB range. One section of op amp buffers the attenuated signal to correctly drive the band-reject filter, the summing/gain and difference/gain sections, and the system status switch.

Band-Reject Filter

The band-reject filter is made up of four sections of op amp. The 3dB points of the filter are 400Hz and 2.2kHz, with an 11dB dip at 1.1kHz. This filter is set to attenuate signals in the voice band, while leaving low and high frequency audio signals unaffected. This filter is the reason why the Generation II in the music & voice mode adds simulation primarily in the non-voice region of the audio spectrum. The signal leaves the band-reject filter and proceeds to the system status and front panel mode switches.

Status Switch

Spread out over several of the schematic pages is the system status switch. It is a 4-pole, 3-position slide switch. What it does is really quite simple. In the fixed—music & voice position: the output of the band-reject filter is sent to the input of the compressor, the bucket brigade delay's clock oscillator is set to the high frequency position, and the wiper of the stereo intensity trim pot is sent to the summing/gain and difference/gain op amps. In the fixed—music position: the output of the line input stage is sent to the input of the compressor, the oscillator is set to the low frequency position, and the wiper of the stereo intensity trim pot is sent to the summing/gain and difference/gain op amps. In the front panel position: the mode switch and stereo intensity potentiometer on the front panel are active.

Front Panel Mode Switch

If the status switch is set to the front panel position, the front panel mode switch is active. The switch selects if the band-reject filter is in or out of the signal sent to the compressor, and controls the BBD's clock oscillator frequency. This allows the users to select if the Generation II operates in the music or the music & voice mode.

Audio Delay

A large part of the Generation II's circuitry involves time delaying the audio input signal. The "heart" of the time delay is an integrated circuit that implements a bucket brigade delay (BBD) line. A BBD is an analog technique that samples a signal and stores an analog representation of its level. This analog signal is then passed, "bucket to bucket" to achieve a delay. The more buckets, and the more time the signal is stored in each bucket, the longer the time delay. The length of time the signal sits in a bucket is inversely related to the frequency of its clock signal. The higher the clock frequency, the shorter the time delay. An anti-aliasing filter is required to prevent the BBD from receiving input signals greater than one half the lowest clock frequency. A low-pass filter on the output of the BBD prevents clock signal from being present in the audio output. A compandor scheme is used to improve the BBD's signal to noise ratio. The input signal is compressed prior to being sent to the BBD, and then expanded afterwards. Charge pumps help the compressor and expander circuits respond to signal transients.

Compressor

Audio enters the delay circuitry via an inverting buffer op amp associated with the compressor portion of an integrated circuit compandor. A simple resistor/capacitor pre-emphasis network precedes the inverting buffer. The compressor attack time is speeded by a charge pump, which reduces transient distortion that is often associated with compandors.

Input Low-pass Filter

The compressed signal connects to an anti-aliasing low-pass filter. Three sections of op amp form a 6-pole, 20kHz Butterworth low-pass filter. This reduces the possibility of audio frequencies aliasing with the BBD's clock signal.

BBD Clock

An oscillator integrated circuit provides three signals for the BBD: clock 1, clock 2, and a voltage reference. Clock 1 and clock 2 are identical square wave signals 180 degrees apart in phase. The frequency of clock 1 and clock 2 is set by a resistor-capacitor (RC) combination, in association with an FET-controlled frequency select circuit. In the music mode, the FET is biased to the non-conducting state. In the music & voice mode, a shorter delay is required and the FET is turned on, reducing the time constant of the RC circuit and raising the clock frequency. The clock is set for 64±1kHz in the music mode and switches to approximately 119kHz in the music & voice mode.

Output Low-pass Filter

The time delayed audio enters a 5-pole, 20kHz Butterworth low-pass filter created using three sections of operational amplifier. This filter removes clock signal from the BBD's output signal.

Expander

The signal was compressed prior to being delayed. The expander portion of the compandor integrated circuit is used to recreate the original dynamic range. Again a charge pump is used, this time in the compandor rectifier. The audio signal has now been delayed and is ready to go to work!

Randomizing Network

The delayed signal enters a proprietary randomizing circuit which is in the form of a sealed module. Frankly, the great sound of the Generation II's simulated stereo is created by this network. The randomizing circuit modifies the delayed signal to ensure that the peaks and dips in the soon to be created comb filter do not fall on objectionable harmonics, preserving the natural sound of the audio. The output of the network connects to both the front panel and trim stereo intensity potentiometers.

Summing/Gain and Difference/Gain

The simulated left and right signals are created by two sections of op amp. The left channel is created by summing the output of the front panel or trim potentiometer stereo intensity control with the direct input signal. This creates a comb filter which breaks the audio signal into hundreds, or thousands, of little pass-bands. The right channel is created by producing the difference between the output of the front panel or trim potentiometer stereo intensity control and the direct input signal. Again, combs are created, but with every comb 180 degrees out of phase with those created in the left channel. These combs are what give the listener the illusion of space, i.e., stereo! Both op amp sections add gain according to the position of the 3-position output level switch.

The mono-compatibility of the Generation II's simulated stereo can be easily understood by a careful study of how the stereo is actually made. The output of network module is really the "stereo information." The left channel is created by adding some of the stereo information to the incoming mono signal; the right channel is created by subtracting the same amount. The amount added and subtracted is directly related to the setting of the front panel or trim stereo intensity controls. When listening to the simulated left and right output signals in stereo, a feel of "space" is perceived. When listening in mono, the left and right signals are summed (added together), dropping out the stereo information and giving the original signal. The Generation II's stereo information exactly cancels out! What is added to the left channel is subtracted from the right.

This process holds true in the music and music & voice modes. In the music mode, full bandwidth delayed input audio is added and subtracted. In the music & voice mode, band-passed delayed input audio is added and subtracted. In either case, perfect mono compatibility is maintained.

Line Output Sections

The output stages use sophisticated, balanced line driver integrated circuits. These ICs are optimized for audio applications such as the Generation II. Unbalanced signal enters the ICs via the summing/gain section for the left channel, and difference/gain section for the right channel. The nominal level fed to the ICs is –6, –2, or +2dBu, depending on where the output level switch is set. The line driver IC adds 6dB of gain, giving final nominal output levels of 0, +4, or +8dBu. The line driver ICs are capable of driving high signal levels into 600 ohm or greater loads, remaining stable even when driving unbalanced loads. This means that if one side of a line output is grounded, the IC is not harmed and the other side of the line output still functions correctly. The output signals are direct coupled to the XLR-type line output connectors.

Troubleshooting

The following procedures must be performed by a qualified technician. Operating the Generation II with the top and/or bottom cover removed exposes the technician to points in the power input section with hazardous voltages.

Equipment Required

- Low Distortion Audio (Sine Wave) Generator, must have tone burst capability
- Qty 2, AC Voltmeter, high input impedance type
- Frequency Counter (200kHz minimum measuring ability)
- Audio Distortion Analyzer, Sound Technology 1710A or equal
- DC Voltmeter, high input impedance type
- Oscilloscope, Tek 465 or equal
- Test Probes for above
- Audio Cables as required

Power Supplies

The first step in troubleshooting the Generation II is to remove the top cover (two Phillips head screws on each side panel). It is common for failures associated with contemporary audio equipment to be power-supply related. The components most likely to fail in the power supply are the 3-terminal regulators (one each for +15 and –15) and the large filter capacitors.

Power Supply Voltages

With the DC voltmeter common lead connected to COM test point: check +15 test point for +15±1Vdc, –15 test point for –15±1Vdc.

Analog Problems

If the power supplies are not at fault, use standard troubleshooting procedures to locate the problem. A simple technique is to connect an audio signal to the line input and trace the signal (again, using the schematic) until the source of the problem is found.

Alignment

Alignment is required after any repair involving the compandor, BBD, or associated circuitry. Alignment is not required after repairs are made to the line input, band-reject filter, summing/gain, difference/gain, meter, or line driver circuits. The following paragraphs will aid in aligning the circuits, which will restore proper operation. It is important to perform the alignment procedures whenever you have replaced any components in the aforementioned circuits. Alignment can be tricky and, if possible, factory repair and realignment is recommended.

Caution: DO NOT attempt to adjust any of the trim pots inside the unit before you understand the entire alignment procedure and are equipped to go through the entire process.

All signal measurements are made in reference to test point COM. This test point is common to the +15, –15V, and –22Vdc power supplies, pin 1 of the three connectors, chassis ground, and earth ground via the power cord.

Procedure

1) Apply power to the Generation II. **Be Very Careful:** High voltage is present in the power entry area.
2) On the Generation II's front panel: set the mode switch to the music mode, and the stereo intensity control to the fully counterclockwise (min) position.
3) On the Generation II's circuit board: set the system status switch to the front panel mode, input level switch to 0, output level switch to 0, and input trim pot R1 (accessible via the back panel) to 50 percent the (middle) position.
4) Connect the signal generator output to the line input, located on the back panel.
5) Set the signal generator frequency to 1kHz, with output level to OFF so that no signal is getting sent to the line input. The signal generator signal must be a sine wave.

Initial Compandor Adjustments

1) Again, ensure that no audio is coming into the unit.

2) With the DC voltmeter common lead connected to test point COM: check test point COMP for 1.35±0.5Vdc. If not within this range, set R85 to give 1.35Vdc at test point COMP.

3) With DC voltmeter common lead connected to test point COM: check test point EXP for 1.35±0.5Vdc. If not within this range, set R88 to give 1.35Vdc at test point EXP.

BBD Clock Frequency

1) Connect common lead of frequency counter to test point COM. Check test point CLK for 64±2kHz. Remember, you must have the unit in the music mode. If not within this range, set R63 to give 64±2kHz.

Incoming Audio Test Signal

1) Connect the AC voltmeter common lead to test point COM.

2) Adjust the audio generator output level to give a –10dBu reading at test point AUDIO 1.

Compressor

1) Move the AC voltmeter lead to test point AUDIO 2. Note the reading that you observe. It should be –10±2dBu.

2) Reduce the generator level by 20dB.

3) Observe the level at AUDIO 2. It should have dropped 15±0.3dB. If not, adjust R85 to get this condition.

4) Raise the generator level 20dB. Observe level at AUDIO 2.

5) Reduce the generator level 20dB. Observe level at AUDIO 2. It should drop 15±0.3dB. If not, again adjust R85 to get this condition.

BBD Distortion

1) Set the audio generator's frequency to 5kHz; adjust its level to give –10dBu when measured at test point AUDIO 1.

2) Set the front panel stereo intensity control to give –10dBu when measured at test point AUDIO 3.

3) Connect distortion analyzer to measure at test point AUDIO 3.

4) Adjust R13 to give minimum distortion (THD + Noise). The acceptable minimum distortion is 0.35 percent or lower. The usual minimum is 0.10 to 0.25 percent.

Expander

1) Set generator frequency to 1kHz; adjust its level to give –10dBu at test point AUDIO 1.

2) Measure level at AUDIO 3. You should have left the front panel stereo intensity control where it was set a few steps previously.

3) Reduce the generator level 20dB. Level at AUDIO 3 should drop 20±0.3dB. If not, adjust R88 to get this condition.

4) Raise generator level 20dB. Observe level at AUDIO 3.

5) Drop generator level 20dB. Observe level at AUDIO 3. It should drop 20±0.3dB. If not, again adjust R88 to get this condition.

Expander Transient Response

1) Adjust the generator level to give 0dBu when measured at test point AUDIO 1. Set the generator to the tone burst mode: two cycles on, 64 cycles off. In this way, two cycles of audio pass to the Generation II, then 64 cycles of no audio, etc.

2) Observe AUDIO 3 on the oscilloscope. Adjust R34 to give the flattest base band signal. The signal should not over or under shoot.

Gain Variation Trim Potentiometer

1) Set the audio generator frequency to 1kHz; adjust its output level to give –10dBu when measured at test point AUDIO 1.

2) Adjust R98, Gain Variation Adjust, to give –10dBu when measured at the output (pin B) of the Network Module. It is easier to measure this signal at the clockwise connection of the stereo intensity controls.

Specifications

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

AC Mains Requirements
115Vac or 230Vac, 10%, switch selectable, 50/60Hz, 10 watts

Fuse
0.5A for 115V, 0.25A for 230V; units shipped to North America and Japan contain 0.5A 3AG type; all others shipped with 0.25A 5x20mm type

Connectors
Input: 3-pin XLR-type female, pin 2 high

Outputs: 3-pin XLR-type male, pin 2 high

Input
Electronically balanced, direct coupled, load termination resistor optional

Input Level
0, +4, or +8dBu, switch selectable

Input Level Adjust
Approximately ±0.75dB

Input Impedance
25k ohms, nominal

Input Common Mode Rejection
100dB @ DC and 60Hz, 70dB @ 20kHz, 62dB @ 40kHz (typical)

Output
Electronically balanced, direct coupled

Output Level
0, +4, or +8dBu, switch selectable

Output Level at Clipping (0dBu = 0.7746V)
+25dBu into 600 ohms, balanced

Frequency Response
Better than ±0.1dB, 20Hz to 20kHz (no simulation added)

Signal to Noise Ratio (Simulator In Circuit)
76dB

Distortion (THD + Noise, 20Hz to 20kHz)
Simulator In Circuit: less than 0.40%

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

Weight
4.2 pounds (1.9kg)

Specifications and information contained in this instruction manual are subject to change without notice.

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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.

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