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CAUTION: THE BOSE® AM-25P AND AM-30P SERIES II BASS MODULES CONTAIN NO USER-SERVICEABLE PARTS. TO PREVENT WARRANTY INFRACTIONS, REFER SER-VICE TO WARRANTY SERVICE STATIONS OR FACTORY SERVICE.

WARRANTY PERIOD

The Bose Lifestyle® 12, 25 and 30 Series II Speaker Systems are covered by a limited 1-year transferable warranty.

SAFETY INFORMATION

- 1. Parts that have special safety characteristics are identified by the $\langle \cdot \rangle$ symbol on schematics or by special notes on the parts list. Use only replacement parts that have critical characteristics recommended by the manufacturer.
- 2. Make leakage current or resistance measurements to determine that exposed parts are acceptably insulated from the supply circuit before returning the unit to the customer. Use the following checks to perform these measurements:

A. **Leakage Current Hot Check**-With the unit completely reassembled, plug the AC line cord directly into a 120V AC outlet. (Do not use an isolation transformer during this test.) Use a leakage current tester or a metering system that complies with American National Standards Institute (ANSI) C101.1 "Leakage Current for Appliances" and Underwriters Laboratories (UL) 1492 (71). With the unit AC switch first in the ON position, then in the OFF position, measure from a known earth ground (metal water pipe, conduit, etc.) to all exposed metal parts of the unit (antennas, handle bracket, metal cabinet, screw heads, metallic overlays, control shafts, etc.), especially any exposed metal parts that offer an electrical return path to the chassis. Any current measured must not exceed 0.5 milliamp. Reverse the unit power cord plug in the outlet and repeat test. ANY MEASUREMENTS NOT WITHIN THE LIMITS SPECIFIED HEREIN INDICATE A POTENTIAL SHOCK HAZ- ARD THAT MUST BE ELIMINATED BEFORE RETURNING THE UNIT TO THE CUSTOMER.

B. **Insulation Resistance Test Cold Check**-(1) Unplug the power supply and connect a jumper wire between the two prongs of the plug. (2) Turn on the power switch of the unit. (3) Measure the resistance with an ohmmeter be tween the jumpered AC plug and each exposed metallic cabinet part on the unit. When the exposed metallic part has a return path to the chassis, the reading should be between 1 and 5.2 Megohms. When there is no return path to the chassis, the reading must be "infinite". If it is not within the limits specified, there is the possibility of a shock hazard, and the unit must be repaired and re checked before it is returned to the customer.

PROPRIETARY INFORMATION

THIS DOCUMENT CONTAINS PROPRIETARY INFORMATION OF BOSE® CORPORATION WHICH IS BEING FURNISHED ONLY FOR THE PURPOSE OF SERVICING THE IDENTIFIED BOSE PRODUCT BY AN AUTHORIZED BOSE SERVICE CENTER OR OWNER OF THE BOSE PRODUCT, AND SHALL NOT BE REPRODUCED OR USED FOR ANY OTHER PURPOSE.

ELECTROSTATIC DISCHARGE SENSITIVE (ESDS) DEVICE HANDLING

This unit contains ESDS devices. We recommend the following precautions when repairing, replacing, or transporting ESDS devices:

• Perform work at an electrically grounded work station.

• Wear wrist straps that connect to the station or heel straps that connect to conductive floor mats.

• Avoid touching the leads or contacts of ESDS devices or PC boards even if properly grounded. Handle boards by the edges only.

• Transport or store ESDS devices in ESD protective bags, bins, or totes. Do not insert unprotected devices into materials such as plastic, polystyrene foam, clear plastic bags, bubble wrap or plastic trays.

SPECIFICATIONS

General Overview

The digital bassbox is essentially an AM-25P (or AM-30P) powered speaker system with the following new features:

-SPDIF digital signal input.

-the ability to decode Dolby Digital™ digital bitstreams.

-all signal processing preformed with DSPs.

Compared with an earlier Series I, the digital bassbox delivers generally the same output power, consumes the same input power via the same line cord and dual voltage transformer, has the same input impedance (about 3 k in the audio bandwidth) at the audio input connector and the same input signal headroom (2Vrms max). The digital bassbox uses the same headend units

(CD-5, CD-20) as the Series I and responds in a similar manner to the various SmartSpeaker commands issued from the head-end (volume up, mute, etc.).

The digital bassbox uses most of the same parts as the Series I bassbox and is (nearly) physically indistinguishable. New parts include:

-a 13-pin input connector (replaces Series I 8-pin DIN).

-a new set of head-end to bassbox cables.

-a 4-layer DSP PCB (replacing Series I 2-layer Videostage®/eq PCB).

-a 6-channel amplifier PCB (replacing Series I 5-channel amplifier).

-an 18-conductor ribbon cable, connecting the DSP PCB to the Amplifier PCB.

Signal Processing

The signal processing will not be discussed in any great detail at this time. The digital bassbox uses the same user interface (i.e., the keypad on the head-end and on the remote control) as its predecessors, so very few new features have been added.

The most obvious new feature is the ability to receive a digital SPDIF bitstream, either PCM (digitized 2-channel audio) or the AC-3 output from a DVD player. AC-3, also known as Dolby Digital, is a perceptual coding and data compression technique that allows 5 wideband channels and one low frequency bass channel ("5.1") to be compressed into less bandwidth than would be required for 2-channel PCM.

The Videostage used on the Series I expanded a L, R input pair into L, C, R, S and bass. The Videostage 5 used in the digital bassbox does basically the same thing, but adds a few new features; stereo surround channels and a mono to 5 channel mode. The mono to 5 channel mode is a way of processing a mono soundtrack such that all 5 speakers receive parts of the signal.

AC-3, while most typically used to encode 5.1 audio, can be used to encode any number of channels from 1 to 5.1. Thus the Videostage signal processing used on the Series I was expanded to be able to cope with this greater variety of inputs. The rest of the signal processing, (volume control, dynamic eq, speaker eq, tone controls, etc). is analogous to the signal processing performed on Series I, except that in the digital bassbox it's all done in the digital domain.

Power Supplies

There are several sets of internal power supply voltage rails. The AM-25/30P Series II uses the same two transformers as the variants of Series I, (either the dual-voltage 120V/240V or the 100V Japanese). Voltage is applied to the transformer primary via triac D302. The triac is controlled from the head-end via the usual 10V "turn-on" logic signal. Note that this signal is the ONLY way to turn on the bassbox.

As with the Series I, the power amplifiers are supplied with both +17 and +34VDC unregulated rails, as determined by the Class G circuitry. These rails are developed from the transformer secondaries by the usual diode bridge and capacitor bank (B700, B701, C730, C731,etc.)

The digital circuitry on the DSP PCB requires two different regulated voltages, +5V and +3.3V. The +5V regulator is a standard 3 terminal linear regulator in a surface mount package and is located on the DSP PCB. It's powered from the +17V rail via a fusing resistor R704 on the amp PCB, and supplies about 120mA to the DSP PCB, most of which is consumed by the microcontroller and codec. The +3.3V regulator is a switch-mode regulator and must be located under the shield to avoid EMI problems (harmonics of the 100 kHz switching frequency will stray into the AM broadcast band if left unshielded). This regulator is also powered from the +17V rail and supplies about 350mA to the DSPs.

Operation of the 5V linear regulator is very simple. Current is supplied to the regulator via surface-mount power resistors R10 and R11, which drop the voltage supplied to the regulator to avoid excessive power dissipation. The regulator needs a minimum of about 7V at its input terminal to stay in regulation and it should be able to do this even as the +17V rail sags to +10V or so.

Operation of the +3.3V switch-mode regulator is more complicated. In its simplest terms, the unregulated input is "chopped" at about 100 kHz, and the resulting pulse train is low-pass filtered to extract the DC. This DC output is compared to a +3.3V reference, and the duty cycle of the switching waveform is adjusted accordingly. The high-side switch (the switch that connects +17V to the output filter) is integrated into the controller, while the low-side switch (the switch that connects ground to the output filter) is an external diode D500. When the controller is running, you should be able to observe the switching waveform at L500.

Microcontroller

A microcontroller on the DSP PCB is used for the following housekeeping and control functions:

-to interpret SmartSpeaker commands from the head-end.

-to interpret special test mode commands from the functional tester.

-to store and retrieve data from nonvolatile memory (U201).

-to boot and reprogram the codec (U100) as required by the operating mode of the system. -to monitor for

-bass and treble pot position.

-DC offset on the Twiddler™ outputs.

-PCB over-temperature.

-presence/absence of a digital SPDIF signal.

Microcontroller (continued)

-to sequence the amplifier and codec mute lines as required. -to boot and monitor the performance of the DSPs. -and relay information to/from DSPs.

In detail:

1. SmartSpeaker commands are interpreted by watching for digital activity on the serial data input line (J5, pin12). This serial input can be either the "standard" SmartSpeaker protocol, compatible with the outputs of CD-5, CD-20, etc., or it can be 4800 baud RS232 with 5V logic levels. The microcontroller decides which protocol to use based on the voltage level present on the serial data input at boot time. If the line is low, it assumes RS232; if the line is high, it assumes SmartSpeaker. It checks only once, at boot time, so switching from one protocol to the other requires a power cycle to reboot the system.

2. Test mode commands are special commands to facilitate testing, or to retrieve stored information from the nonvolatile memory. Test mode commands are RS232 only, e.g., the revision of the firmware can be queried by sending the command "tv", the answer "2200/2000" might come back on the serial data output line. This would indicate version 22.00 for the microcontroller and version 20.00 for the DSPs.

3. The nonvolatile memory is used to store the following information:

-choice of eq. -system, center, and surround volume settings at power-down. -US or Euro bass tone control preference. -system usage statistics. -number of times system was powered off. -minutes spent powered on.

4. The codec (U100) has a fair number of internal registers, all of which must be set up properly for correct operation, e.g., to switch between the ADC inputs and the SPDIF input (both functions are integrated into the codec), the codec must be reprogrammed. In addition, there are a number of error status bits which are periodically sampled by the microcontroller, the codec can be rebooted if problems are discovered.

5. The tone control potentiometers no longer have analog signals passing through them (since all signal processing is now done by the DSPs). The microcontroller has several ADC inputs, two of which are used to detect the position of each pot. Everytime this position changes, the new position is reported to the DSPs, which adjust their tone control filter coefficients accordingly. One somewhat odd side effect of this is a delay from the time the pot is turned until the time the audible difference is heard. This delay is typically several hundred milliseconds, but is not noticeable to the casual listener unless the pot is turned through a large angle quickly.

6. Other ADC ports are used for the following functions:

-to monitor the gross DC offset on the five satellite speaker channels. Excessive DC. offset is assumed to be an indication of a serious problem and the microcontroller. will respond to this by shutting off the power to the bassbox.

Microcontroller (continued)

-to detect the presence or absence of a digital SPDIF signal. The codec cannot be set up for an analog input and a digital SPDIF input simultaneously and it cannot be reprogrammed from analog to digital (or vice versa) without interrupting the audio signal. There are times when it is helpful to be able to check for the presence/absence of a digital signal. e.g.,

-sequence the mute line. As with the Series I, there are three so-called "speaker modes", 2-channel (bass, left, right); 3-channel (bass, left, right and center) and 5-channel (bass, plus all 5 satellites). Unused output channels are muted, not only at the DSP but also at the power amplifier. The mute inputs to the power amplifiers are 5V logic-level outputs from the microcontroller. -the microcontroller is in charge of booting the DSPs at power-up in the follow ing sequence:

-boots the codec and establishes the correct output signals from the codec (bit_clock, frame_clock, and data);

- enables the 3.3V power supply, and verifies the correct voltage from it;
- -releases the DSP reset line and waits for the DSPs to boot and acknowledge.

The microcontroller is always engaged in dialogue with both DSPs. If the DSPs should fail to respond within a certain time-out, it is assumed that the DSPs have "crashed" and the DSPs are rebooted. The microcontroller must pass some of the nonvolatile information in the EEPROM over to the DSPs at boot time, e.g.: speaker eq, system volume (all volume control in is done in the DSPs), center and surround volume and bass and treble pot position. The microcontroller itself can be reset in two ways, by a hardware reset signal, which is generated by U200 when the 5V supply drops below 4.75V and by the microcontoller itself, once it has already booted successfully (it can pull its own reset line if the 8 MHz clock is running).

Codec

The codec (U100) is a highly integrated device which includes the following functions: -a I 2 C interface for control and status information.

-three channels of 20-bit ACD; only two are used, one for the Left analog input and one for the Right. The ADCs will input signal levels in excess of 1 Vrms.

-six channels of output DAC; all six are used. Maximum output signal level is 1 Vrms.

-a crystal oscillator which establishes the ADC/DAC sampling rate, in this case, it is 11.2896 MHz / 256 = 44.1 kHz.

-a SPDIF receiver. "SPDIF" stands for Sony/Philips Digital Interface Format. Essentially, for every pair of 20-bit audio samples, another 24 bits of status, parity-checking and general housekeeping bits are sent along for the ride. The data is encoded in such a way that

a.) there is no net DC content to the bit steam, allowing transformer or AC coupling and b.) the bit clock can be recovered from the bit steam.

The codec generates three output signals, either from the 11.2896 MHz crystal, or from the input SPDIF bitstream;

-a serial bit stream containing the 20-bit audio data.

-a bit clock, which indicates when to sample the serial bit stream.

-and a frame_clock, which indicates the start of an audio sample.

Codec (continued)

In addition, there is a serial data input (6 20-bit audio samples) to the codec, with timing that corresponds exactly to the timing of the serial data output. These four signals (data out, data in, bit_clock, and frame_clock) are used by the serial ports on the DSPs. The timing of the data flow into and out of the DSP subsystem is driven entirely by the codec; the serial ports on the DSP run asynchronously to the 40 MHz clock which drives the DSPs.

DSPs

The DSPs are Analog Devices 21061L general purpose floating point digital signal processors, each capable of about 40 MIPs of performance. Two are required to provide enough MIPs for:

-AC-3 decoding. -Videostage®.

-bass management (i.e., creating a single bass output from 2 to 6 channels of wideband input).

-six channels of speaker eq.

-tone controls.

The DSPs have no internal ROM; at boot time they load themselves from the external PROM U401. This boot process is more or less automatic; i.e., no intervention from the microcontroller is required (although the microcontroller has control of the DSP reset line). The 21061L processors are designed to be bussed together, which accounts for the relatively large number of pins found on each DSP (240).

For instance, the signals required to connect the DSPs to each other and to the boot PROM include:

-an external data address buss (32 bits, of which 24 are used).

-an external data buss (48 bits, of which 16 are used).

-buss control signals (buss priority, address and r/w strobes, etc).

Also, there are 30 power and 30 ground pins on each DSP, and a 40 MHz clock oscillator (U400, crystal CR400, and associated components) which drives both processors.

Signal Path

The signal path through the system takes the following route(s):

For analog signals, the L/R stereo pair is introduced from the head-end via the control cable and the 13-pin DIN connector J5. Each signal is received (pseudo) differentially by op-amp U105 and associated components and -6 dB of gain is applied to match the maximum signal level of the bassbox (2Vrms, to be compatible with AM-25P) into line with the maximum in/out signal level of the codec (1Vrms).

Signal Path (continued)

For digital SPDIF input signals (which come directly from a SPDIF signal source such as a DVD player and not from the CD-20 head-end), the signal is routed directly to the codec U100 via a network of components (D1 and surrounding) designed to clamp and filter ESD transients.

The codec takes in either the pair of analog inputs, or the single SPDIF digital input and outputs three digital signals (data, bit clock, and frame clock, as described earlier). Since the codec runs on 5V and the DSPs run on 3.3V, any signals passing between them must be levelshifted via the buffers U101 and U102.

Without going into a lot of detail, DSP1 handles (mostly) AC-3 decoding and videostage processing and DSP2 handles (mostly) equalization. With the signal processing complete, DSP2 formats the six output channels into a single serial bitstream and sends this back to the output side of the codec U100. The analog buffers following the codec (U103, U104 and associated components) do three things:

> -remove the 2.3VDC reference voltage from the codec's outputs. The codec is a single supply (+5V) device and references any of its internal analog signals to a reference voltage at approximately 2.3V, about one-half the +5V supply voltage. -reference the signal to the amplifier ground, not the local ground. The power amps are single ended and are referenced to their own local ground on the amp PCB, which is not the same as the local ground at the codec on the DSP PCB. Amplifying the differences between the two grounds will produce audible hum in the output speakers.

-provide enough gain to increase the output signal to levels compatible with the AM-25P output levels. The signals leave the codec itself at about 1Vrms full-scale, this is amplified up to about 6Vrms full-scale by the output buffers.

Power-Up Sequence

The 10V turn-on signal is applied to J5 pin 7, which energizes the opto-coupler U300, which turns on the triac D302, which applies 120VAC to the primary side of the transformer, which charges up the \pm 34V and \pm 17V power supply rails. These rails become (more or less) fully charged after 100msec. The reset IC U200 holds the microcontroller U202 reset line low until the regulated +5V supply stabilizes, by which point the microcontroller's 8 MHz oscillator is already oscillating.

At power-up, all of the microcontroller ports are in a high impedance state, which means that: -the amplifier mute lines are held low, muting the power amps.

-the 3.3V regulated supply is disabled, reducing the demand on the +17V supply until things have stabilized.

-the DSP reset line is held low, holding the DSPs in reset.

Power-Up Sequence (continued)

The microcontroller proceeds to:

-set up its I/O ports. -print out a "reset" message on the TTY output. -program the codec and verify that it responds. -retrieve system variables from nonvolatile memory and print out the "power_cycle" and "eq" variables. -boot the DSPs, and wait for them to respond. -pass over system variables to the DSPs. -print out any appropriate error messages. -unmutes the power amplifiers, then unmutes the DSPs.

At this point, normal signal processing commences. It should take no more than 1.5 seconds from the time power is applied until the time audio appears at the speaker outputs.

Power-Down Sequence

Upon receiving the "off" command from the head-end, the microcontroller;

-mutes the DSPs.

-mutes the power amplifiers.

-powers down the codec and the DSPs.

-stores system variables in EEPROM.

and then monitors the voltage level at the turn-on input. When it sees a low to high transition, it resets itself as if the power had just come on.

Detailed Theory of Operation (see schematic diagram SD251571)

Sheet 1 of 5

Sheet 1 of 5 shows the analog input buffers, codec's power supply, analog output buffers and some "glue logic" around the codec. The analog input buffers consist of op-amp U105 and associated components. The ratio of R102 (and R107) to 100 sets the gain; -6 dB from J5 to the input of the codec. The dual-diodes D100, D102, and D104 are there to clamp ESD discharges into J5. The filter capacitors C100, C105, etc., are present to filter off RF picked up by the input cable.

The output signal from the buffer is ground referenced, but the codec's analog inputs are biased at about 2.3V. C104 is required to AC-couple the signal from the buffer into the codec. R103 and D103 are present to prevent transients from the op-amp $(\pm 12V \text{ rails})$ from overdriving and perhaps damaging the codec (5V rail). R104 and C102 are an EMI filter to prevent any RF leaking from the codec from getting outside the metal EMI shield. The codec is located under the shield, while the input buffer is not.

The SPDIF input to the codec consists of C151, R1, R2, D1, C3, C7 and R5. C151 AC couples the digital SPDIF signal, and R1 terminates the coax with 75 Ohms. R2 and D1 clamp the input (a typical SPDIF input is about 500 mVpp into a 75 Ohm load). C1 AC couples the signal at D1 (which is ground referenced) to the codec input (which will be biased at about 2.3V when the SPDIF receiver is working). R5 and C7 are an EMI filter designed to prevent RF from leaking outside the shield.

Sheet 1 of 5 (continued)

The codec's power supply consists of U1 and associated components. R10 and R11 are power resistors designed to drop the voltage at the voltage regulator's input, for reducing the power dissipation of the voltage regulator itself.

D2, D3, and R4 are used to draw a little current from the +12V regulated supply when the power supplies start to sag severely, as they will at low AC line and heavy amplifier load. The +12V regulator is fed from the +34V unregulated supply and has considerably more headroom than the +5V regulator fed from the +17V supply. Also fed from the +17V supply is the 3.3V switching regulator, which tends to draw more current as its input voltage drops (it delivers a fixed amount of power, at a fixed conversion efficiency). So, as the +17V supply starts to sag, the current load on it actually starts to increase, which tends to make it sag more.

The +5 volts supplied by the regulator is sent to two different components; the codec and the microcontroller and associated components. Since the regulator is located outside the EMI shield and the +5V supply line might be carrying EMI (Electro Magnetic Interference), ferrite beads L2 and L200 were placed in series with each of the supply lines.

L1 and R2 serve to isolate the two different power supply pins on the codec (the "analog" +5VA from the "digital" +5VB). C5 and C6 are bypass capacitors placed directly under the codec.

Codec glue logic

U101 is a 74LCX244, a 3.3V IC that can tolerate 5 volts on its inputs. It's used to convert 5V to 3.3V logic signals.

U102 is a 74ACT244, a 5v part that understands the logic levels used by the 3.3V IC. It's used to convert the 3.3V logic signals to 5V logic signals.

U106 is a flip-flop, used to convert a falling edge (from the codec) to a narrow pulse (to the DSP). This removes a potential timing ambiguity.

R192 (and others) is a 75 Ohm series resistor used to series-terminate some of the high speed logic signals, to preserve edge fidelity at the receiving end.

R6 and R7 comprise a low performance 5V to 3.3V shifter. Using such large series resistances in the presence of even moderate amounts of stray and input gate capacitance slows down the signal's edge, but this particular signal is very low bandwidth.

Analog output buffers consists of U103, U104, and associated components. Gain is set by the ratio of R132 to R131; about 6x. Note that the ratio of R189 to R188 must match. Gain of the output buffers = (codec_out - codec_reference) - amp_ground_sense).

Detailed Theory of Operation (see schematic diagram SD251571)

Sheet 2 of 5 (microcontroller and associated components)

Potentiometers R201 and R204 are unchanged from theSeries I, (to preserve the look and feel). The pots are not in the audio signal path, however. They are driven with +5 volts and the center wipers are connected to a pair of microcontroller ADC inputs. All of the connections between the pots (outside the shield) and the microcontroller (inside the shield) are EMI filtered.

R207 is a thermistor, a temperature sensitive resistor. R207 is mounted on the topside of the DSP PCB. When the DSP PCB temperature rises to about 70 degrees C, the microcontroller starts turning down the output volume by i.e.,programming the codec effectively controlling the maximum temperature inside the bassbox and preventing damage.

R211 connects the signal "CMOUT" to one of the microcontroller's ADC ports. CMOUT is the codec's 2.3V reference voltage. This voltage is monitored as one of the codec's "vital signs", if out-of-bounds, the microcontroller reboots the codec.

R208 connects the 3.3V power supply to one of the microcontroller's ADC ports. The microcontroller has the ability to turn the 3.3V regulator on and off, thus having the ability to check that the 3.3V supply is within limits. The microcontroller needs to be able to turn the 3.3V off at power down to avoid draining the +17V supply faster than the -17V supply. If the upper rails sag faster than the lower rails, the bass amp will unmute causing a "thump" to be heard (about 5 seconds after power down).

R220 through R226 sum together all the satellite outputs and feed them to one of the microcontroller's ADC ports. The microcontroller watches for grossly excessive DC offset at any of the speaker outputs. If DC is detected the microcontroller will shut down the AC power.

U201 is the non-volatile memory. Read/write access is via the 2-wire serial I-squared-C buss.

U200 generates a 250msec reset pulse for the microcontroller whenever the +5V supply drops below 4.75V.

Q202 and Q203 form a simple window comparator. Whenever the signal PROTECT strays more than one diode drop away from ground, one or the other transistor will turn on, which ultimately will turn the AC power off.

D200 forms a simple level translator. The CLIP signal is generated on the amp PCB whenever one of the satellite power amplifiers starts to clip. It's an open-collector output that pulls down to -12V. D200 level shifts this signal from -12V through +12V, to 0V through 5V. This levelshifted signal goes to both the microcontroller and DSP2 (via a 5V to 3.3V buffer). As it turns out, the microcontroller ignores this signal, but DSP2 turns down the system volume until the CLIP signal disappears, then lets the volume drift back up to the original setpoint.

Q204 and Q201 form a simple SPDIF detector. Q204 sets the bias for Q201 such that it is just barely off. An SPDIF signal arriving at Q201 will cause C216 to be discharged, causing the voltage at C216 to drop from 5V to something closer to ground. This voltage is connected to one of the microcontroller's ADC ports and is sampled periodically by the microcontroller.

Detailed Theory of Operation (see schematic diagram SD251571)

Sheet 2 of 5 (microcontroller and associated components continued)

Q200 allows the microcontroller to pull down on the DSP's reset line. When the microcontroller first boots, all of its output ports are set to a high impedance state. Q200 remains turned on by R252, which holds the DSPs in reset, until the microcontroller is ready to boot the DSPs.

R246, R248, etc. connect several of the microcontroller output ports to the MUTE inputs on the power amplifiers. The mute signals are organized into three groups; (bass, left, right), (center) and (left_surround, right_surround), corresponding to the three speaker modes the product supports. Each line is EMI filtered. Pulldown resistors R813, 814 and 815 guarantee that the signals will assume a LOW state i.e., muted) at power-up.

Sheet 3 of 5 (AC power control and associated components)

This circuit is nearly identical to the original Series I. Q300 and Q301 buffer the turn-on signal and drive current through the LED side of the opto-coupler U300. Zener D300 forces the turnon signal to reach some minimum threshold before U300 is energized.

Q302 and Q303 form a latching circuit that will "crowbar" the LED side of U300. Once this circuit latches up and steals all of U300's LED current (thus shutting off the power), it stays latched until the source of the current (the turn-on signal generated by the head-end) is removed.

R308 connects a microcontroller ADC port to a voltage that indicates the state of the "turn-on" signal. It's possible to power the bassbox off, then back up again quickly, without the +5V supply ever going out of regulation. The microcontroller needs to know, one way or the other, that it's just been powered on. The usual way it knows this is that the reset IC U200, issues a reset pulse. The other way it knows that the power is "off" is to watch the state of the turn-on input. If the microcontroller sees the turn-on signal drop low, then go high again, it knows that the power has come back "on" again. It responds to this by issuing its own reset pulse.

Sheet 4 of 5 (DSPs)

There are only two DSPs, but since they each have 240 pins, they've been broken into several different schematic symbols so as not to clutter the page:

-address and data busses.

-buss control, and miscellaneous.

-link ports (sheet 5).

-power and ground (sheet 5).

Without going into any detail, the DSPs are connected to each other and to the boot PROM U401, as follows:

-address buss connected to address buss.

Detailed Theory of Operation (see schematic diagram SD251571)

Sheet 4 of 5 (DSPs continued)

-data buss connected to data buss. -buss control signals connected to buss control signals. -power pins connected to 3.3V power plane. -ground pins connected to ground plane.

The large number of interconnects and the fact that each signal has extremely fast rise times (2 nsec), required the use of a four layer PCB. One of the inner layers is a (more or less) solid ground plane; the other is used for a power plane, plus additional signal interconnect where required.

U400 and associated components form a third-overtone 40 MHz oscillator. There are two buffered outputs; one drives both DSPs and one is connected to the test connector J401, which was used to help debug the prototype PCBs but is no longer loaded.

Sheet 5 of 5

U500 and associated components form a 3.3V regulated supply. C500 and L500 are the regulator's output filter. R500 and C501 set the regulator's switching frequency (about 100 kHz).

C503 is the "bootstrap" capacitor required for establishing an internal bias voltage higher than the supply voltage (for turning on the internal N-channel MOSFET switch). C515, L501, and C505 prevent switching harmonics from leaking back out onto the raw DC input voltage. R501, C507, C508, etc., are feedback and compensation components. C506 is the "soft-start" capacitor, which forces the regulator to start up slowly when power is first applied, thus limiting the inrush current. R510 and Q500 allow the "soft-start" pin to be pulled low, shutting off the regulator. The 3.3V supply is turned off as part of the normal power down sequence.

Detailed Theory of Operation (see schematic diagram SD197228)

Sheet 1 of 2

Bass Power Amplifier

The bass power amplifier is a discrete high efficiency Class-G design. Maximum power is 80W into $4Ω$ at less than $0.1%$ THD. In Class-G operation the amplifier is powered by two different power supplies depending on the amplitude of the signal input. When the audio amplitude is low, the amplifier runs off of the lower supply rails, but during musical peaks it switches to the higher supply rails. Efficiency is typically increased from 20% to 40% and power dissipation is reduced by a factor of 2.5. Detailed operation is as follows:

Referring to sheet 1 of the schematic diagram (197222), audio input is applied to the amplifier PCB at pin8 of connector J700 and is AC coupled through C727. The amplifier is controlled by negative feedback to op-amp U707, which is configured as an inverting amplifier with a voltage gain of 12 (21.6 dB). With no signal applied to the input, all output power devices are biased off. For a negative input signal, pin 1 goes high and conducts driver transistor Q713. Collector current is pulled through R707 and R774 until the voltage drop across R774 reaches about 1 Volt, at which time the high gain darlington transistor Q718 begins to conduct emitter current through power diode D711, which connects to the +17VDC supply.

Detailed Theory of Operation (see schematic diagram SD197228)

Sheet 1 of 2

Bass Power Amplifier (continued)

Collector current from Q718 flows through the speaker load and the voltage at this node is regulated by feedback to the op-amp via resistor R789.

When the audio output voltage approaches the 17VDC power supply rail, output transistor Q718 begins to saturate and conducts much more base current than the normal maximum of 5mA. At approximately 8mA the voltage drop across 75 Ohm resistor R707 exceeds 0.6V and small signal transistor Q714 begins to conduct. This in turn conducts Class-G Darlington transistor Q715, which turns on the 34VDC power supply and reverse biases power diode D711, effectively turning off the 17VDC supply. During this period, the wave form at the collector of Q715 resembles the audio output signal plus the saturation drop of Q718 and Q715 is operated in the active region (not as a switch), thus sharing the power dissipation.

Crossover distortion and switching transients are not an issue due to the relatively low bandwidth of the amplifier (less than 250 Hz) and the ability of the Acoustimass[®] bass module to roll off high frequency distortion products. Crossover distortion is less than 0.5% at 200 Hz, 1 Watt.

Satellite Power Amplifiers

Each of the five satellite amplifiers are operated in Class-G configuration and consist of a 50W, Class AB monolithic integrated circuit (in a multiwatt-15 package, TDA7294). The amplifier is short circuit and thermally protected. External to this IC is a pair of TO-220 Darlington transistors (the same as used in the discrete bass amplifier) to perform the Class-G power supply switching. The following detailed operation is described for the left surround channel, however the other four channels would be the same.

The surround signal is applied to capacitor C717 and couples to the non-inverting input of the TDA7294 amplifier IC. It is configured as a non-inverting amplifier with a voltage gain of 4 (12 dB). The output stage consists of a pair of MOSFET transistors and the positive FET must develop gate drive well above the supply voltage and hence there is a bootstrap cap between pin 6 and 14.

With low amplitude signal, the amplifier runs off of the 17VDC rails through power diodes D704 and D710. The power supply voltage at pin 13 is subtracted by 5.6V zener diode D703 and divided down by the ratio of $1 + R731/R732$. This bias voltage sets the threshold at which transistor Q703 turns on. When the audio or the input to the amplifier exceeds the voltage at the emitter of Q703 by two diode drops, D716 and Q703 conduct. In turn this conducts small signal transistor Q704 which in turn conducts output transistor Q705. A negative feedback loop is established that prevents Q705 from turning completely on and the voltage at the collector of Q705 resembles the audio output wave form plus several volts of saturation headroom. Q705 operating in the active region (instead of as an on/off switch), results in shared power dissipation between the transistor and the power amplifier IC. The phase lag created by the input network R753 and C745 allows the power circuit to switch on slightly ahead of the power amplifier at high frequencies (above 8 kHz) to minimize turn-on glitch.

Figure 1. DSP PCB Block Diagram

Note: Numbers in parentheses correspond to the callouts in Figure 9.

2. Cover Replacement

2.1 Place the cover (2) over the module assembly. Align the cover so that the main PCB input and output connectors are inserted into the holes of the cover.

2.2 Rotate the tab of the cover latch out from the enclosure.

2.3 Slide the cover over the base-plate (4) until the back of the cover snaps over the base-plate tabs.

2.4 Use a flat-head screwdriver or scribe to rotate (to the left) the cover latch (17) back into the enclosure.

2.5 Attach the two tone control knobs (1) by pushing them in towards the module. They are keyed and will only fit one way.

3. Main PCB Assembly Removal

3.1 Perform procedure 1.

3.2 Remove the two screws securing the main PCB (11) to the adapter bracket (13).

3.3 Disconnect the transformer's 5-pin connector from J7, the 8-pin cable from the amplifier, and the flat ribbon cable (20) from connector J8 on the main PCB.

3.4 Release the main PCB (11) from the four snaps of the adapter bracket (13).

4. Main PCB Assembly Replacement

4.1 Place the main PCB (11) onto the adapter bracket (13) component side down. The J5, J9, and J11 input and output connectors should be facing the label side of the module.

4.2 Press the main PCB into the adapter bracket's (13) four snaps.

4.3 Secure the main PCB to the adapter bracket.

4.4 Connect the transformer's 5-pin cable to J7, connect the 8-pin cable from the amplifier's PCB to J10, and the flat-ribbon cable (20) to J8 on the main PCB.

4.5 Perform procedure 2.

5. Amplifier PCB Removal

5.1 Perform procedure 3.

5.2 Remove four silver screws and eight black screws (3) securing the adapter bracket (13) to the base plate (4). Lift the bracket away from the enclosure.

5.3 Disconnect the transformer's 5-pin cable from the amplifier PCB's J1 connector. Disconnect the woofer harness connector (8) from the amplifier PCB's J3 connector. Disconnect the flat ribbon cable (20) from the amplifier PCB's J2 connector.

5.4 Lift out the heatsink amplifier PCB subassembly from the base plate (4).

5.5 Place the subassembly on the workbench with the heatsink (23) face down.

5.6 Place the tip of a flat-head screwdriver into the metal plate's (26) small rectangular slot. Quickly pry the heatsink wall back just enough to release the metal plate from the heatsink. Refer to Figure 7.

Figure 7. Amplifier Assembly

Note: Do not put a permanent bend in the heatsink wall. A large bend in the heatsink wall will not allow reassembly of the heatsink amplifier PCB subassembly.

5.7 Lift out the spring plate (25) that rests on the power devices.

5.8 Remove the amplifier PCB from the heatsink.

6. Amplifier PCB Replacement

6.1 Place the amplifier PCB (24) into the heatsink (23). The PCB should be component side up with the transistors and ICs resting on the inner sides of the heatsink.

Note: Thermal grease should be applied to the heatsink before seating the PCB.

6.2 Position the spring plate (25) into the holes of the PCB. The spring plate can only be inserted one way.

6.3 Position the metal plate (26) so that the rectangular slot is on the same side as the amplifier PCB's J1 connector. The side marked" Outside" Should be facing out. Insert this side of the metal plate into the slot of the heatsink.

6.4 Press down on the metal plate quickly with both palms of your hand. The plate should snap into the slot of the heatsink.

6.5 Place the heatsink amplifier PCB subassembly, metal plate side down, into the module's base plate (4). The large capacitors fit into the recess of the base plate. Make sure that the rubber grommets engage onto the plastic posts.

6.6 Connect the transformer's 5-pin cable to the amplifier PCB's J1 connector. Connect the woofer harness cable (8) back into the PCB's J3 connector and the flat ribbon cable (20) to the amplifier PCB's J2 connector.

6.7 Place the adapter bracket over the heatsink amplifier PCB subassembly and transformer.

Note: Before securing the adaptor bracket and main PCB over the heatsink amplifier subassembly, route the flat ribbon cable underneath the adaptor bracket and around the heatsink.

6.8 Secure the adapter bracket (13) to the base plate (4).

6.9 Redress any wire harness to the adaptor bracket as needed.

6.10 Perform procedure 2.

7. Transformer Removal

7.1 Perform procedure 5 through 5.2.

7.2 Disconnect the transformer's 5-pin cable from the amplifier PCB's J1 connector.

7.3 Lift the transformer from the module's base plate (4).

Note: Numbers in parentheses correspond to the callouts in the Figures referred to in the following procedures. Refer to Figure 9.

8. Transformer Replacement

8.1 Place the transformer (15) into the recess of the base plate (4). Make sure the transformer is positioned so that the primary wires (red, white, brown, orange, black) that connect to the J7 connector are facing the PCB.

8.2 Connect the transformer's 5-pin cable to the amplifier PCB's J1 connector. Connect the woofer harness cable (8) back into the amplifier PCB's J3 connector.

8.3 Secure the adapter bracket to the base plate and redress any wire harness as needed.

8.4 Perform procedure 2.

9. Woofer Removal

9.1 Perform procedure 1.

9.2 Disconnect the woofer harness cable (8) from the amplifier PCB's J3 connector.

9.3 Remove eight black screws (3) that secure the module assembly to the enclosure. Lift the module assembly away from the enclosure.

9.4 Remove four screws (5) from the woofer (6) under repair.

9.5 Lift the woofer up far enough to expose the wires connected to the woofer's terminals.

9.6 Cut the wires as close to the terminals as possible. Refer to Figure 8.

9.7 Remove the woofer from the enclosure.

10. Woofer Replacement

10.1 Strip the ends of the module's wiring harness. Connect the yellow wire to the replacement woofer's (6) + positive terminal and the green wire to the - negative terminal. Refer to Figure 8.

10.2 Line up the woofer's gasket (7) over the woofer's baffle panel hole.

10.3 Place the woofer over the gasket. Make sure it is seated evenly over the gasket and baffle hole.

10.4 Secure the woofer to the baffle.

10.5 Secure the module assembly to the enclosure.

10.6 Perform procedure 2.

Figure 8. Woofer Harness Hookup

11. Satellite Grille Removal

Note: Refer to Figure 10 for the following procedures.

11.1 Place a plastic flat blade tool between the edge of the grille and the edge of the satellite enclosure. With a twisting action, gently release the grille from the catches on the satellite enclosure. Use care not to cosmetically damage the satellite enclosure.

12. Satellite Grille Replacement

12.1 Line up the grille with the catches on the satellite enclosure. Press the grille onto the enclosure so that it snaps into place. The grille with the logo is used on the lower satellite.

Note: The logo is attached to the grille by pressing it's tabs through the holes in the grille frame and secured by bending the tabs over.

Note: The grille is the only replaceable part on the satellite. The Twiddlers™ can not be replaced.

Jewel Cube® Satellite Disassembly/Assembly

Note: Refer to Figure 11 for the following procedures.

13. Grille Assembly Removal

13.1 Swivel the cube array so that the grille assemblies (1 and 2) are not aligned. Pull the grille away from the enclosure by prying off one side of the grille with a small scribe.

Note: Do not lose the small grommets (6) that cover the screws located behind the grille.

14. Grille Assembly Replacement

Note: Be sure the grommets (6) are in place before replacing the grille assembly.

14.1 Align the grille assemblies (1 and 2) with the cube array. The curved edges of the grille must be oriented vertically. Snap the grille into place.

Note: The grille assembly with the nameplate (3) should be on the bottom satellite cube.

15 Twiddler Removal

15.1 Perform procedure 1.

15.2 Remove the four grommets (6) covering the screws that hold the twiddler (4) in place.

15.3 Remove the four screws (5) holding the Twiddler (4) in place. Lift theTwiddle out of the enclosure and cut the wires as close to the terminals as possible.

16. Twiddler Replacement

16.1 Strip the wires and connect them to the replacement Twiddler's terminals as follows:

 16.1.1 If replacing the top Twiddler, connect the black wire to the positive (+) terminal and the yellow wire to the negative (-) terminal.

 16.1.2 If replacing the bottom Twiddler, connect the yellow wire to the positive (+) terminal and the white wire to the negative (-) terminal.

16.2 Place theTwiddler into the enclosure and secure it in place.

16.3 Perform procedure 2.

TEST SET-UP PARAMETERS AND EQUIPMENT

Before performing the tests described in these procedures, use the following test set-up parameters.

Speaker output loading

Equipment Requirements

- 1. Test Cable part number 199527
- 2. RS232 to TTL converter box (B+B Electronics) link to a PC
- 3. 25 to 9 pin serial data cable (you might have to make this, see diagram below)
- 4. A terminal emulator or in windows "95" use "Hyperterm"
- 5. An SPDIF digital signal source, or a DVD player
- 6. The standard equipment needed for testing audio products, i.e., audio signal generator, oscilloscope, dB meter, etc.

Test Set-up Procedure

Using the test cable, part number 199527, plug the 13-pin din connector into the bassbox. Connect the RS232 (B+B TTL232 converter) to the 25-pin D connector on the test cable. Connect the RS232 box to its power supply.

Connect the 25-pin to 9-pin cable to your PC's serial port.

Open "Hyperterm" (used with "windows 95") or the terminal emulator on your P.C.

Set up the serial port for 4800 baud, 8 data, 1 stop, no parity.

Connect an audio signal generator to the RCA input jacks on the test cable 199527.

Connect the SPDIF output from an SPDIF converter, or from a DVD player, to the female RCA on the test cable 199527.

Connect a 10VDC supply to the 3.5mm jack on the test cable 199527. A 9V battery works fine for this purpose.

Diagram of the Test Set-Up

1. Turn-On Test Procedure

1.1 With no signal applied connect the bassbox to a PC using the Test Set-up Procedures on the previous page.

1.2 Connect the 10V supply (9V battery) to the 3.5mm jack and monitor the computer screen.

1.3 You should see a prompt that looks like #0001%01. This boot prompt consists of a number of text strings. Each string is interpreted as follows:

** Is the normal response to being powered on.

> Normal system prompt

#0001 The four digit hex number, following the hash mark "#" indicates the number of times the system has been powered off (AC mains disconnected). The two digit number following the "%" percent sign (%01) indicates the currently programmed equalization curve. The current codes are as follows:

- 01 LS-12 II and LS-25 II systems
- 02 LS-30 II system
- 11 LS-12 II and 25 II 240V systems
- 12 LS-30 II 240V system

1.4 At this time you should not see any error codes such as --12++ which would indicate a problem with DSP1.

2. Gain Test

2.1 Enter the following command just as it is typed, **tn 6,0,0,0,0,0,0,0,0** (bypass mode, this will bypass any signal processing).

2.2 Apply a 100 mV, 200 Hz signal to the left and right inputs.

2.3 Reference a dB meter to the input signal, and measure the gain of the output signal according to the table.

Gain Response Table

3. Bass Module Sweep Test

3.1 Put the module in a well defined state by issuing the following commands:

sk 61,af,ff (select analog source) **sk 31,d0,3f** (select 5-channel mode) **sk 31,0f,ff** (select maximum volume) **sk 31,3f,ff** (un-mute)

3.2 Apply an 80 mVrms ±5 mVrms signal to the left and right inputs.

3.3 Sweep a frequency range of 40 Hz to 300 Hz for a 5 second duration.

3.4 Listen for any unusual buzz, rub, or extraneous noises. Redress any wires that might have buzzed, and repair or replace any defective woofers.

4. Bass Module Air Leak Test

4.1 Apply an 80 mVrms ±5 mVrms 45 Hz signal to the left and right inputs for a 5 second duration.

4.2 Listen for any air leaks from where the amplifier module meets the cabinet. Repair any air leaks that are found.

5. Bass Control Test

5.1 Apply a 40 mVrms ±5 mVrms 100 Hz signal to the inputs.

5.2 Rotate the bass control fully clockwise and counter clockwise.

5.3 Verify that the bass output increases and decreases as the control is rotated. Expect a slight delay in the response of the signal.

6. Treble Control Test

6.1 Connect a satellite cube to the left and right outputs.

6.2 Apply a 100 mVrms ±5 mVrms 8 kHz signal to the left and right inputs.

6.3 Rotate the treble control fully clockwise and counter clockwise.

6.4 Verify that the output increases and decreases as the control is rotated. Expect a slight delay in the response of the signal.

7. Turn-Off Test

7.1 Type an **rs** command to place the unit back to its initial state, and disconnect the 10V turn-on supply (9 volt battery).

7.2 Listen for any popping sounds.

8. Miscellaneous System Tests

8.1 Connect the system according to procedure 1.

8.2 Type a **tv** command and look for a response similar to 2200/2000. The first number is the microcontroller revision and the second number is the DSP revision.

8.3 Type a **vr** command and look for a response similar to: CBLT xxxx (c) 1998 Bose® Corp. S/N xxxxxx Chksum: 0017BAD2.

8.4 Type an **ef ff** to return the module to the factory defaults. Then type an **rs** command to reset the unit. You might see an error code response similar to --16++. This is normal. Retype an **rs** command.

9. SPDIF Digital Signal Test

9.1 Apply a SPDIF signal to the SPDIF input using either a DVD player or a analog to SPDIF converter.

9.2 Type the following commands: **tn 0,0,0,0,0,0,0,0,0 sk 51,af,ff tn 6,0,0,0,0,0,0,0,0.**

9.3 Play a DVD disc or apply an 80 mVrms 200 Hz signal to the left and right inputs. If using a DVD player, connect the audio source to the left and right input.

9.4 Observe the output. If using a DVD player you may see what resembles bursts of white noise at the output instead of an analog signal. In the **tn 6** mode, the AC-3 (Dolby Digital) data is not decoded, its simply passed through to the outputs.

PART LIST NOTES

1. This part is not normally available from Customer Service. Approval from the Field Service Manager is required before ordering.

2. The individual parts located on the PCBs are listed in the Electrical Parts Lists.

3. \angle ¹. This part is critical for safety purposes. Failure to use a substitute replacement with the same safety characteristics as the recommended replacement part might create shock, fire and or other hazards.

4. This part is not interchangeable with the earlier versions of the AM-9P or AM-25P modules.

5. These parts are packaged in quantities of five per system.

MAIN PARTS LIST

(See Figure 9)

Figure 9. Bass Box and Module Exploded Views

Resistors

Resistors (continued)

Capacitors

Capacitors (continued)

Capacitors (continued)

Capacitors (continued)

Diodes

Transistors

Integrated Circuits

Miscellaneous

SATELLITE PART LIST

(See Figure 10)

(See Figure 11)

1 X2 (2)

LS12 II PACKAGING PART LIST

Figure 12. LS12 II Packaging View

LS25 II PACKAGING PART LIST

Figure 13. LS25 II Packaging View

LS30 II PACKAGING PART LIST

Figure 14. LS30 II Packaging View

ACCESSORY PART LIST

Figure 15. Accessory Kit View

Audio Power Amplifier (TDA7294) Bose® part number 170156

EPROM (M27W210) Bose part number 251708

U400, Quad Nand, 3V (MC74LCX00) Bose part number 193858-001

U702, Voltage Comparitor (LM339) Bose part number 187618-001

U100, Codec (CS4226-KQ) Bose® part number 197221

U300, Opto-Triac (MOC3023T) Bose part number 190334-001

U106, JK Flip-Flop (MC74HC73) Bose part number 196670-001

U103 and 104, Quad Op-Amp (TLO74D) Bose part number 186112

Voltage Regulator 37V positive Bose part number 137927

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DSP 1 and 2, (ADSP21061LKS-160) Bose part number 193834

AM-25P II/AM-30P II Digital Bass Module Powered Acoustimass® -25 and -30 Series II Speaker System

This manual is for the Lifestyle® Series II 12, 25 and 30 Systems

SPECIFICATIONS AND FEATURES SUBJECT TO CHANGE WITHOUT NOTICE

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