# OPERATING AND MAINTENANCE
# INSTRUCTION MANUAL

## *"DAVID-II"*
—— MODEL 716 ——

# FM-STEREO PROCESSOR / GENERATOR

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# *DAVID-II*
## ──── MODEL 716 ────

# FM-STEREO PROCESSOR / GENERATOR

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# Section I

# INTRODUCTION

## MODEL 716 PRODUCT DESCRIPTION

**General**    Inovonics introduced the first *"DAVID"* (as in David vs. Goliath) FM-Stereo Processor/Generator in 1992. The original Model 715 was the hallmark of simplicity, and was intended to answer the more basic of FM radio broadcasting needs.

The *"DAVID-II,"* Model 716, includes a redesigned pulse-width-modulation audio processing section. This improved processing circuitry extends the applicability of the *DAVID-II* into broadcast environments where perceived loudness and modulation density are considered germane to the broadcaster's success.

**Features**    Features of the Inovonics *DAVID-II* include:

- Comprehensive stereo audio processing: slow, "gain-riding" AGC — Wideband Compression and Peak Limiting — Independent High Frequency Limiting.

- Exclusive "Bass Enhance" equalizer (defeatable).

- Extended-response low-pass filtering with proprietary overshoot compensation.

- Digital synthesis of pilot and subcarrier for unexcelled performance and freedom from drift and routine adjustment.

- Internal combining for RDS or SCA subcarriers.

- Built-in, adjustable Composite Processor.

## MODEL 716 TECHNICAL SPECIFICATIONS

**Frequency Response** (through appropriate de-emphasis network): ±0.5dB, 20Hz–15kHz; –1dB or less at 16kHz, –60dB or better at 19kHz.

**Stereo Separation** (L→R or R→L): Processing OUT: >65dB, 20Hz–16kHz. Processing IN: >60dB, 20Hz–16kHz.

**Distortion** (with full Compression, 1dB below 100% modulation): <0.25% THD in demodulated audio above 200Hz.

**Noise:** Better than 80dB below 100% modulation in demodulated audio, 20Hz–16kHz. 38kHz residual and "digital" noise above 54kHz, –70dB or better.

**Crosstalk** (M→S or S→M):
Processing OUT: nonlinear crosstalk –55dB or better. Processing IN: overall crosstalk –50dB or better

**Stereo Pilot:**
19kHz, ±1Hz; injection level adjustable between 6% and 12%, relative to 100% modulation.

**Program Line Inputs:**
Active-balanced/bridging; accommodate nominal "0VU" input levels between –15dBu and +15dBu.

**SCA / RDS Input:**
10K-ohms, unbalanced; accepts SCA or RDS subcarrier signal levels between –20dBu and 0dBu for nominal 5% to 10% injection.

**Pre-emphasis:**
Integral with split-band audio processing. May be jumpered for $75\mu s$ or $50\mu s$ characteristic.

**"Bass-Enhance" Equalizer:**
Accentuates "perceived" bass response at user's option. (See Figure 4, Page 10.)

**AGC Amplifier:**
Slow, 0.5dB/sec. correction for long-term input level variations; ±10dB capture range.

**Compressor/Limiter:**
Fast-acting peak limiter with secondary "platform" time constant to compress dynamic range with average-value weighting. Independent high frequency limiter conforms to selected pre-emphasis characteristic. Subsequent program signal clipping is relegated to non-repetitive limiter overshoots 1ms or less in duration.

**Composite (Multiplex) Output:**
Adjustable between 1V p-p and 8V p-p; 75-ohm source impedance.

**Composite Processor:**
Adjustable from 0dB to 3dB clipping of the 100%-modulated multiplex waveform prior to pilot injection.

**Digital Synthesis Sampling Rate:**
608kHz (16X subcarrier oversampling).

**Power Requirements:**
105–130VAC or 210–255VAC, 50/60Hz; 15 watts.

**Size and Weight:**
1¾"H x 19"W x 7"D (1U); 7lbs (shipping).

# BLOCK DIAGRAM

A simplified Block Diagram of the *DAVID-II* is shown on the facing page. Generator circuitry is explained in detail under Circuit Descriptions, Section V, which references Schematic Diagrams found in the Appendix, Section VI.

# PATENT NOTICE

Low-pass filter overshoot compensation circuitry used in the *DAVID-II* is protected under U.S. Patent No. 4,737,725.
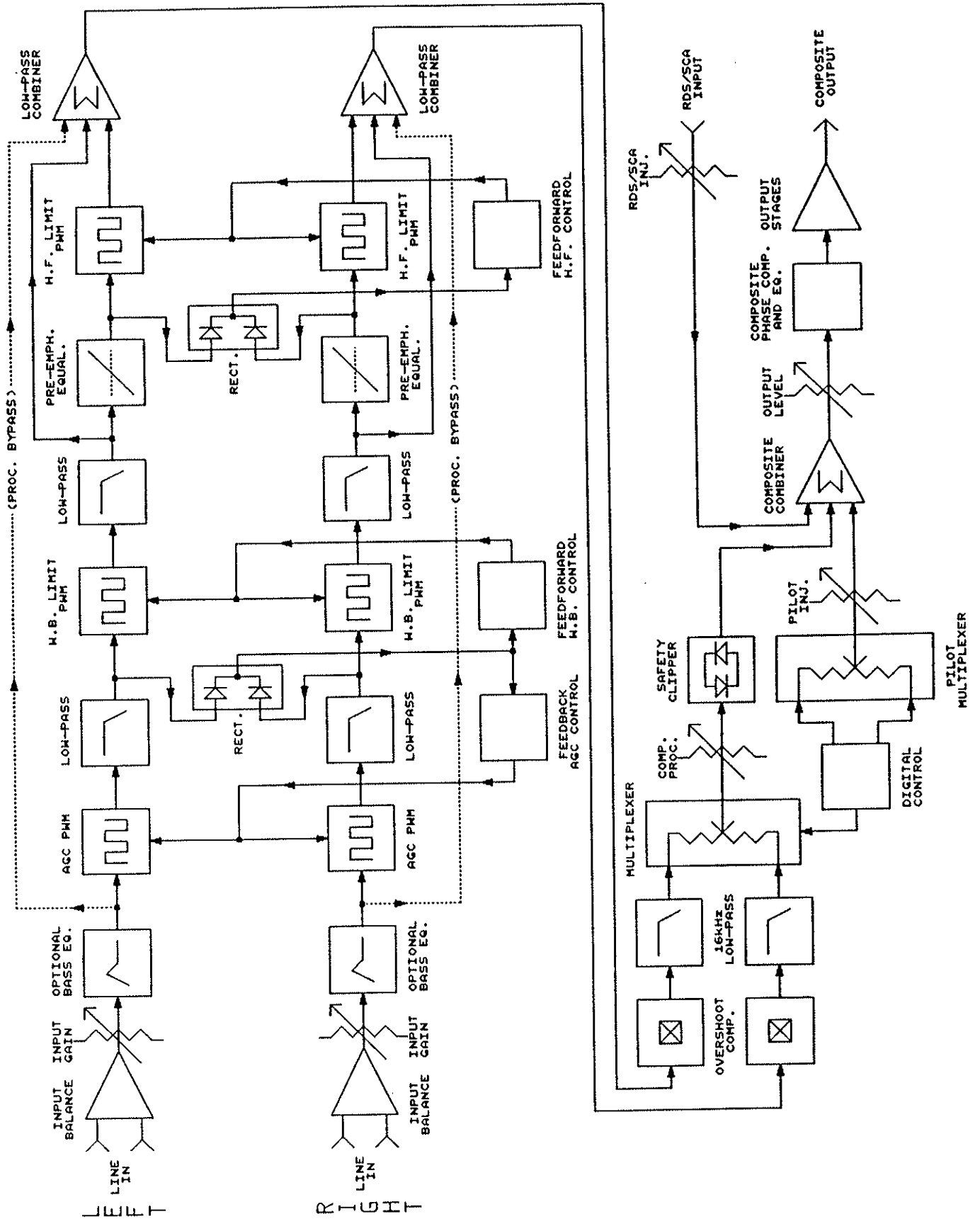
**Figure 1** - Block Diagram, *DAVID-II* FM-Stereo Processor/Generator

# Section II

# INSTALLATION

## UNPACKING AND INSPECTION

Immediately upon receipt of the equipment, inspect carefully for any shipping damage. If damage is suspected, notify the carrier at once, then contact Inovonics.

It is recommended that the original shipping carton and packing materials be saved for future reshipment. In the event of return for Warranty repair, shipping damage sustained as a result of improper packing for return *may invalidate the Warranty!*

**IT IS VERY IMPORTANT** that the Warranty Registration Card found at the front of this Manual be completed and returned. Not only does this assure coverage of the equipment under terms of the Warranty, and provide some means of trace in the case of lost or stolen gear, but the user will automatically receive specific SERVICE OR MODIFICATION INSTRUCTIONS should they be issued by Inovonics.

## MOUNTING

**Rack Requirement**
Inovonics' *DAVID-II* is packaged to mount in a standard 19-inch equipment rack and requires only 1¾ inches (1U) of vertical rack space. The use of plastic "finishing" washers is recommended to protect the painted finish around the mounting holes.

**Heat Dissipation**
Consuming no more power than a small child's night-light, heat generated by the *DAVID-II* is insignificant. The unit is specified for operation within an ambient temperature range of freezing to 120°F/ 50°C. Because adjacent, less efficient equipment may radiate substantial heat, be sure that the equipment rack has sufficient ventilation to keep the temperature below the stated maximum.

## AC (MAINS) POWER

**As Delivered**
Unless specifically ordered for export shipment, the *DAVID-II* is set at the factory for operation from 115V, 50/60Hz AC mains. The rear-panel designation next to the fuseholder will confirm both the mains voltage selected and the value of the fuse supplied.

**Voltage Selector**    A mains voltage selector switch is located beneath the top cover of the unit, adjacent to the AC mains connector on the circuit board. *With primary AC power disconnected,* slide the red actuator with a small screwdriver so that the proper mains voltage (115 or 230) is visible. You must always install an appropriate fuse, and should mark the rear-panel voltage/fuse designation. It is factory practice to cross-out the *inappropriate* marking with an indelible black marking pen. You can remove this strikethrough with solvent to redesignate.

> **BE SURE** that the mains voltage selector setting and primary fuse value are appropriate for the mains supply before plugging the *DAVID-II* into the wall outlet.

**Power Cord**    The detachable power cord supplied with the *DAVID-II* is fitted with a North-American-standard male plug. The individual cord conductors are *supposed* to be color-coded in accordance with CEE standards:

BROWN = AC "HOT"    BLUE = AC NEUTRAL    GRN/YEL = GROUND

If this turns out *not* to be the case, we offer our apologies (cord vendors vary) and advise that U.S. color coding applies:

BLACK = AC "HOT"    WHITE = AC NEUTRAL    GREEN = GROUND

# RADIO FREQUENCY INTERFERENCE (R F I)

**Location**    Although we have anticipated that the *DAVID-II* may be operated adjacent to high-power transmitters, you should exercise care in locating the unit away from *abnormally* high RF fields.

**Ground Loops**    In some installations a mains frequency or RF ground loop may be formed between the input or output cable shield grounds and the AC power cord ground. Use of a "ground-lifting" AC adapter should remedy the situation, though the chassis ultimately must be returned to earth ground for safety. Generally, being screwed-down in the equipment rack will satisfy this requirement.

# LINE INPUT AND INPUT RANGE SELECTION

**Input Connections**    The *DAVID-II* has electronically-balanced (transformerless) left and right channel PROGRAM LINE INPUTS. These are brought out to a screw-terminal barrier strip on the rear panel and include chassis ground connections for cable shields. *Please note that the screw-terminal barrier block can be unplugged from the chassis!* This makes connection a bit easier and permits quicker removal of the *DAVID-II* from the rack should maintenance ever be required.

**Balanced Inputs**    A balanced program audio feed to the *DAVID-II* will use both the + and the − terminals, plus the associated G, for each of the two stereo

channels. Since these are "bridging" (high impedance) inputs, they provide no termination for the console or other equipment which feeds the *DAVID-II*. Please feel at liberty to connect a 600-ohm resistor across the input terminals should you feel this really necessary. Most professional equipment nowadays features low output impedances and high input impedances. The concept of 600-ohm "line-matching" dates from the age of transformer coupling and is rooted in the mystique of telephone practices. More often than not, audio line impedance matching is poo-pooed by today's hip broadcasters (you) and equipment manufacturers (us).

**Unbalanced Inputs**

In rare instances the *DAVID-II* may be fed from "semi-pro" gear with unbalanced outputs. "Disco" mixers fall into this category and are perfectly serviceable for installations on a tight budget. In such cases, the single center conductor of the shielded input lead should be connected to the + terminal, and the shield to G. *In addition,* a jumper wire should be installed between the – terminal and G.

**Input Gain Range**

The *DAVID-II* can accommodate line-level program inputs with a nominal "Zero-VU" value between –15dBu and +15dBu. This 30dB range is divided into two 15dB ranges by internal jumpering.

As shipped, the *DAVID-II* is jumpered for professional level inputs between 0dBu and +15dBu. Most console and STL receiver outputs fall into this range, +4dBu, +6dBu and +8dBu being typical levels.

Lower levels, between –15dBu and 0dBu, may be encountered when interfacing with "semi-pro" gear or with feeds via lossy studio-transmitter telephone line circuits. The extra gain for the low level range is enabled by changing jumpers beneath the top cover.

**Gain Jumpers**

Under the top cover, and just behind the PROGRAM LINE INPUT barrier strip, you will find two jumper pin strips, each with a "push-on" jumper. The strips are identified as JMP3 and JMP5 in the circuit board legend, and each has an H and an L marking to indicate the proper jumper placement for High level and for Low level inputs, respectively. Figure 2 illustrates jumpering options.
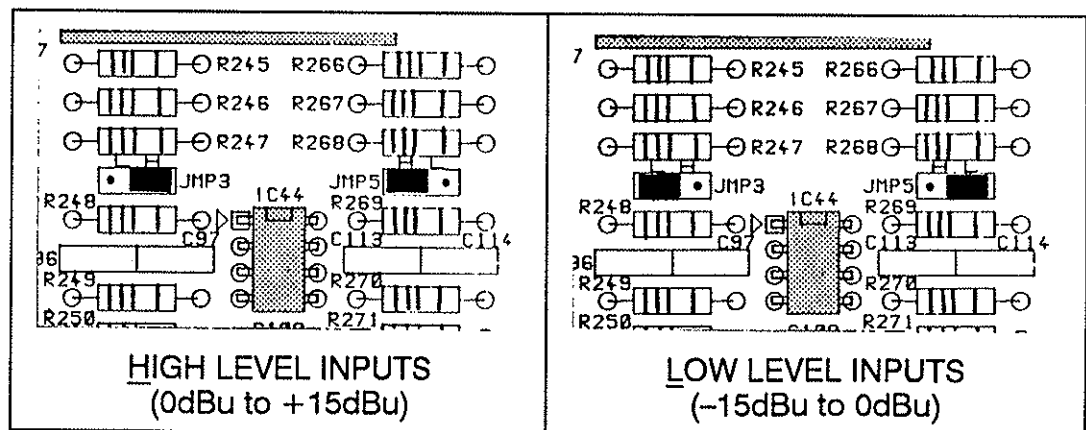


HIGH LEVEL INPUTS
(0dBu to +15dBu)

LOW LEVEL INPUTS
(–15dBu to 0dBu)

**Figure 2** - PROGRAM LINE INPUT Range Selection

## PRE-EMPHASIS SELECTION

The *DAVID-II* supports the 75-microsecond FM broadcasting pre-emphasis standard, common to the Western Hemisphere and parts of the Orient, and the 50-microsecond standard used in Europe and Asia. Pre-emphasis appropriate to the shipping destination is jumpered at the factory, but this is easily changed if necessary.

Beneath the top cover, locate the *second* vertical row of integrated circuits from the right-hand edge of the chassis. This is the row *next* to the one behind the input barrier strip. Between the second and third ICs from the rear panel are two jumper pin strips, each with a push-on jumper. These are identified as JMP1 and JMP2 in the circuit board legend, and each is designated with a 75 and a 50 marking to indicate proper jumper placement for the 75µs and 50µs pre-emphasis characteristics. Figure 3 illustrates the jumpering options



| 75-MICROSECOND | 50-MICROSECOND |
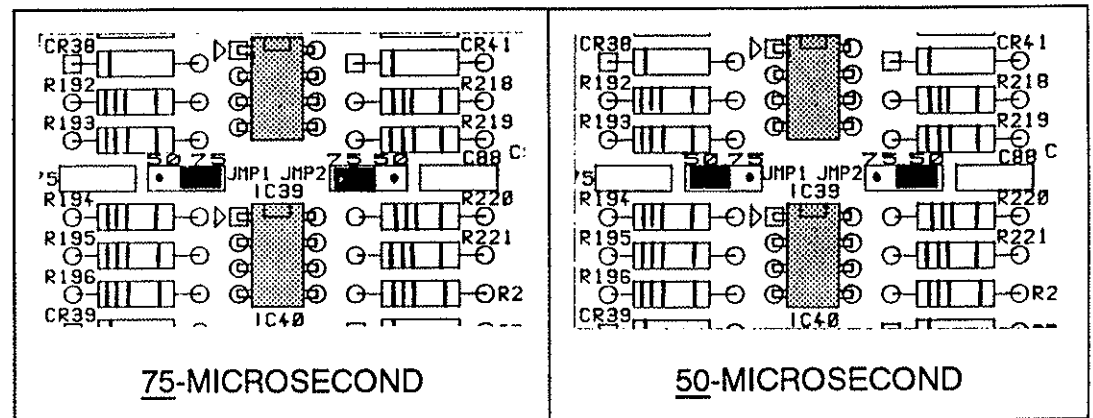
**Figure 3** - Pre-emphasis Jumpering

## "BASS-ENHANCE" EQUALIZATION

The *DAVID-II* incorporates a unique bass-boost equalizer which may be enabled at the user's option.

Normal *DAVID-II* low-end response is flat to within 1dB down to 10Hz. When the equalizer is jumpered into the circuit, response assumes the shape graphed in Figure 3 on the next page. The effect of this equalization is to increase the amount of *perceived* "bottom end," while attenuating much of the *sub-audible* program energy. *Very* low-frequency bass notes are lost in nearly all listening situations, only serving to sabotage modulation efficiency and program density. Even the megawatt subwoofers in cars with blacked-out windows don't respond to frequencies much below 50Hz; rather, they make a lot of noise in the rather narrow region between 60Hz and 80Hz.

The few dB of boost just before cutoff more than makes up for any lost *legitimate* program bass information. This is analogous to the *Rheinfelder Effect*, a phenomenon associated with a novel audio noise-

reduction technique which employs a similar boost-before-cutoff at the high-frequency end of the spectrum.
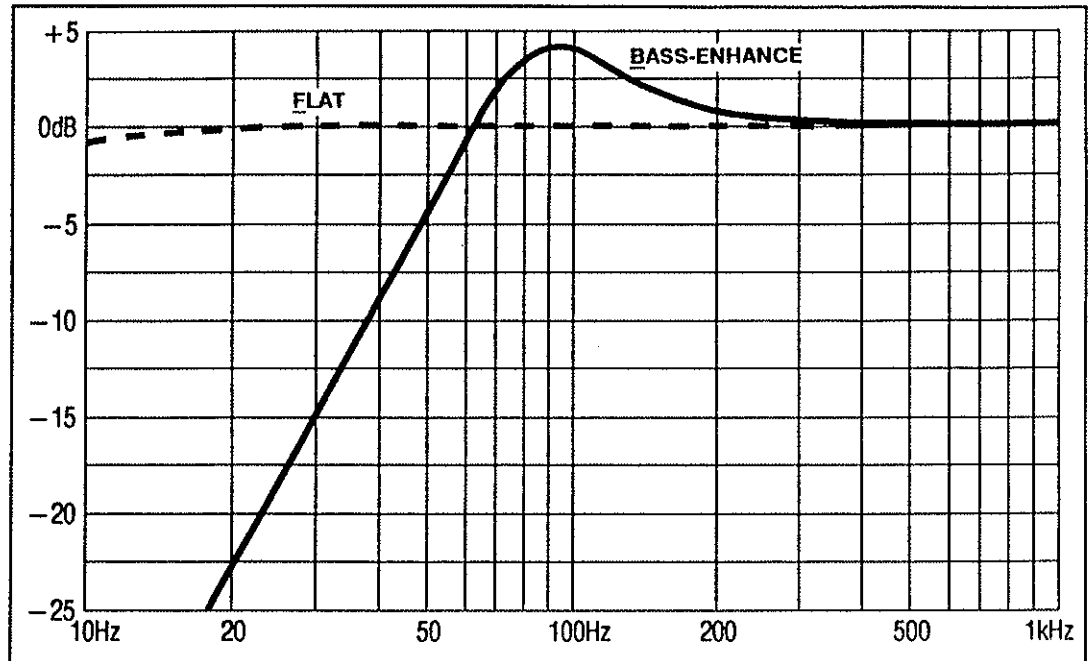


Figure 4 - "Bass-Enhance" Equalization Curve

**"Bass-Enhance" Jumpering**

As delivered from the factory, "Bass-Enhance" equalization is *defeated.* To enable the "Bass-Enhance" equalizer, remove the top cover and locate the first vertical row of integrated circuits closest to the right-hand side of the chassis, the one just behind the PROGRAM LINE INPUT barrier strip. Close to the front panel, and just to the right of the two INPUT GAIN controls, you'll find two jumper pin strips, each with a push-on jumper, labeled JMP4 and JMP6. Each strip is marked with the letters F and B, indicating jumper placement for Flat and Bass-Enhance, respectively. Figure 5 illustrates the jumpering options.



FLAT RESPONSE        "BASS-ENHANCE" RESPONSE
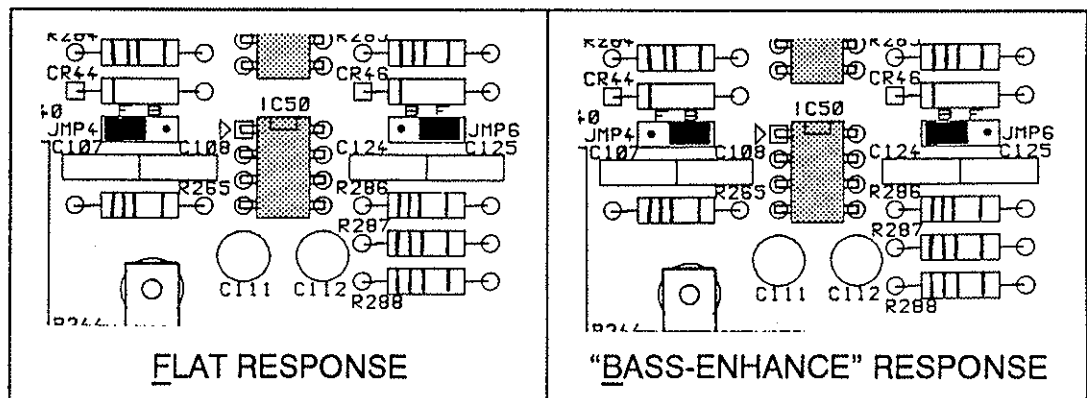
Figure 5 - "Bass-Enhance" Jumpering

# SUBCARRIER INPUT

Internal combining is provided for an RDS or an SCA auxiliary subcarrier. The rear-panel SUBCARRIER INPUT connector is an unbalanced input with a load impedance in the neighborhood of 10K-ohms.

An RDS subcarrier normally accounts for 5% or less of the total composite multiplex output signal, making it 26dB or more below peak program modulation. SCA subcarriers typically have a higher injection level, about 10%, or 20dB below program peak deviation. Subcarrier inputs ranging in level between –20dBu and 0dBu may be fed to the SUBCARRIER INPUT connector, and can be regulated to the proper injection level by the front-panel SUB INJ. level control.

# COMPOSITE OUTPUT

The unbalanced COMPOSITE OUTPUT of the *DAVID-II* has a resistive source impedance of 75 ohms. This permits runs of moderate length (about 50 feet, maximum) to the exciter or microwave link input. If the cable length is more than just a few feet, the cable, itself, should have a 75-ohm characteristic impedance and be terminated in 75 ohms at the far end.

A Stereo Generator must always be connected to an exciter or link (STL) input specifically intended for multiplex stereo. This is a "flat" input, rather than a monaural program input with signal pre-emphasis. Program signal pre-emphasis is imparted by the *DAVID-II*.

**Output Ground Loops**
Because the Stereo Generator output and the exciter input are both unbalanced, it is best to locate the *DAVID-II* near the exciter and power both from a common AC mains circuit. This will help avoid ground loops and mains-related hum in the transmitted signal.

# Section III

# SETUP AND OPERATION

## PANEL CONTROLS AND INDICATORS

A brief description of the front-panel controls and indicators is given here. Do scan this section to at least verify that our terminology agrees with yours.

*PLEASE NOTE* that all front-panel multiturn trim controls require *fifteen to twenty complete rotations* of the adjusting screw to cover their total operating range. Depending on the trim-pot manufacturer, the end of the range may, *or may not,* be identified by a "click-click" sound or other audible/tactile sensation.

INPUT GAIN
These two controls, at the far-right end of the front panel, adjust *DAVID-II* sensitivity for the nominal program level applied to the LEFT and RIGHT channel PROGRAM LINE INPUT barrier strip terminals on the rear panel. The controls have a 15dB range which is multiplied by input gain range jumpering to an input acceptance range of 30dB (see Page 8). The INPUT GAIN controls are adjusted to maintain AGC gain in the center of its operating region.

A G C  GAIN
This series of five LED indicators displays the amount of correction in effect for long-term variations in the program input signal. The AGC circuit is left/right correlated; that is, the overall circuit gain is controlled by the *greater* of the left or right channel signals. This prevents the stereo image from "wandering."

The correction rate of the AGC circuit is intentionally very slow, 10dB of gain correction will take approximately 20 seconds. With a conscientious operator on the console, the indicated AGC GAIN will appear "stuck" at the center, 0dB indicator.

The "safe" operating zone of the circuit is within the central three green LEDs. If either the +10dB or the —10dB red indicator is lighted to full brightness, the input program signal is *out of limits* and the reason should be found and corrected.

GATE OPEN
The *DAVID-II* AGC is "gated." This means that during brief pauses in the program AGC gain is locked to prevent background sounds from increasing to

unnatural proportions. The green GATE OPEN
indicator will remain continuously lighted during most
music programming, but will blink off during pauses
in speech.

**AVERAGE
COMPRESSION**

This is, in essence, a "drive" control at the input of the
compressor/limiter stage, affording the user control
over program density. When fully counterclockwise
(MIN) the broadcast signal will retain most of the
dynamics of the source material. As the control is
rotated clockwise, compression and limiting will
increase, reducing the program dynamic range and
effectively raising the average modulation by the dB
values shown.

A full-clockwise setting of the AVERAGE COMPRES-
SION control will be found tolerable for hard rock
music, but a setting of 6dB to 8dB may sound better
with most other "pop" program material. Classical
music will probably demand an even lower setting.

The effect of the AVERAGE COMPRESSION control is
not instantaneous. There is as much as a one-second
delay between adjusting this control and hearing any
change.

**COMPRESSOR GAIN**

These two LEDs display the gain state of the average
level compression circuitry. With a steady-state tone,
or with no input signal present, the compressor is in
"equilibrium" and both indicators will be lighted to
half-brightness. When the compressor is *reducing*
circuit gain, the red – LED will grow brighter. When
compressor gain is *increasing,* the green + indicator
will brighten. This pair of LEDs shows not only in
which *direction* compressor gain is headed, but can be
interpreted for how *hard* the compressor is working as
well.

**PEAK LIMITING**

The W.B. and the H.F. indicators display action of the
wideband and the independent high frequency limiters,
respectively. Relative limiter activity is directly
proportional to the activity of the LEDs.

**FILTER COMP.**

Proprietary circuitry preceding the primary low-pass
filter inhibits overshoots which would otherwise
compromise full modulation of the carrier. Action of
the overshoot compensation circuitry is displayed by
the FILTER COMP. indicator. It is normal for this LED
to flash almost continuously when "bright" music is
being broadcast.

## COMPOSITE PROCESSING

In the fully-counterclockwise (OUT) position, an output "safety" clipper catches only those very fast peaks and overshoots which sneak by the earlier protection circuitry. As the COMPOSITE PROCESSING control is rotated clockwise, the level applied to the safety clipper is increased by the dB value indicated on the front panel. Composite clipping will increase apparent loudness, but at the expense of generating some spurious harmonic products. The following tabulation is approximate.

| COMPOSITE PROCESSING | PEAK VALUE OF CLIPPING PRODUCTS |
|---|---|
| OUT | >60dB below 100% |
| 1dB | 50dB below 100% |
| 2dB | 45dB below 100% |
| 3dB | 40dB below 100% |

Composite clipping is performed *prior* to 19kHz stereo pilot insertion. This averts clipping or other pilot amplitude modulation. Nevertheless, crosstalk of the program audio into RDS or SCA subcarriers is exacerbated by composite clipping and its inevitable intermodulation and harmonic generation. *Use this feature sparingly!*

## COMPOSITE OUTPUT / PILOT / SUB

The PILOT INJ. and SUB INJ. controls set the injection levels of the 19kHz stereo pilot and an externally-applied SCA or RDS subcarrier. The OUTPUT LEVEL control varies the overall composite output level of the *DAVID-II* delivered to the exciter or STL. OUTPUT LEVEL is a "master" adjustment and does not affect the relationships (level ratios) between the stereo program, pilot and RDS/SCA subcarrier.

## AUDIO PROCESSING

In the IN position, this switch routes the input audio program signal through the AGC and compressor/ limiter processing circuitry. When AUDIO PROCESSING is switched OUT, program audio *completely bypasses* the processing section. The green LED indicates normal operation, the red LED shows that processing has been manually defeated.

The OUT position defeats program audio pre-emphasis as well. However, "Bass-Enhance" equalization (if enabled), low-pass filtering (and its overshoot correction), and composite processing *do remain active* in the program signal path. It is important to leave

the AUDIO PROCESSING switch in the IN position unless outboard provision is made for pre-emphasis and limiting. Without external pre-limiting, program material will be driven hard into the clipping circuits and result in egregious distortion..

PILOT

The PILOT switch lets the user manually turn off the 19kHz stereo pilot for certain tests. Since the 19kHz pilot is necessary for proper system operation, the switch should always be left ON. The green LED indicates normal operation, the red LED lights whenever the pilot has been turned off.

MODE

Under some circumstances it may be advisable to broadcast monaurally. For example, should the station suddenly suffer an emergency power loss, temporarily reverting to mono transmission would help regain some of the lost coverage until full power is restored.

When switched to MONO, the MODE switch removes both the stereo subcarrier and the 19kHz pilot. The transmitted signal will consist of L+R, the algebraic sum of the left and right program channels. The MODE switch does *not* remove an RDS or an SCA subcarrier.

POWER

'Nuf said.

## NORMAL SETUP PROCEDURE

This setup procedure presupposes a normal installation with the *DAVID-II* PROGRAM LINE INPUT fed *directly* from the output of the audio console and a *direct* connection of the COMPOSITE OUTPUT to the wideband input of an exciter/transmitter. Variations from these conditions, such as an intermediate STL (or other link) in either the input or the output path of the *DAVID-II*, may call for considerations not addressed here.

At this point the *DAVID-II* should be installed in the program chain with power applied, and have all front-panel function switches ON. AVERAGE COMPRESSION should be set at 6dB and COMPOSITE PROCESSING turned OUT.

**Input Gain Calibration**

Left (L) and right (R) INPUT GAIN controls are set individually to center the AGC operating range at the nominal "Zero-VU" program line input level. If the following procedure shows control range to be outside adjustment limits, recheck circuit board jumpering for Line Input Range Selection as described on Page 8.

1. To the left channel *only,* apply a 500Hz test tone from the console at a level 2.5dB above normal reference level. This is simply a console VU meter indication of +2.5VU.

2. Adjust the left channel (L) INPUT GAIN control so that *both* the 0dB *and* the –5dB AGC GAIN indicators light to *equal brilliance.* Remember that the AGC circuit responds slowly to level changes, so have patience in performing this step of the procedure.

3. Turn off the +2.5VU test signal to the left channel and apply +2.5VU to the right channel *only.*

4. Adjust the right channel (R) INPUT GAIN control so that *both* the 0dB *and* the –5dB AGC GAIN indicators light to *equal brightness.*

This completes INPUT GAIN adjustment. The *DAVID-II* may now be fed normal stereo program material from the console. If this procedure has been performed properly, and the operator pays reasonable attention to console metering, the AGC GAIN indicators will always remain the green-LED "safe operating" zone between –5dB and +5dB.

**Output Level Adjustment**

Perform this part of the procedure "on-air" under normal operating conditions, and using typical program material. You also must have access to the station's Modulation Monitor. An RDS or SCA subcarrier should *not be connected* until Step #6. Be sure that the COMPOSITE PROCESSING control is OUT; that is, fully counterclockwise.

*PLEASE* be certain that you understand operation of the Mod-Monitor, and that it is properly calibrated and connected to the correct transmitter/exciter RF sample point.

1. Put the Mod-Monitor in its "Total Peak Modulation" measurement mode with peak flasher(s) set to 100%.

2. Turn the AVERAGE COMPRESSION control fully clockwise (10dB). With normal program fed to the *DAVID-II* (at the level which maintains AGC GAIN in the center of its range), adjust the OUTPUT LEVEL control for a Mod-Monitor indication of *frequent* 100%-modulation peaks.

3. Back-down the OUTPUT LEVEL control slightly, so that the 100% peak flasher(s) light only occasionally.

4. Switch the Modulation Monitor to display the injection level of the 19kHz stereo pilot.

5. Adjust the PILOT INJ. control for a 9% Mod-Monitor reading of pilot injection.

6. *(OPTIONAL)* If an RDS or an SCA subcarrier is to be broadcast, connect it to the *DAVID-II* (or other combining point) at this time. Using the Mod-Monitor, or other means of reading the auxiliary subcarrier injection level, adjust the SUB INJ. level control (or other

subcarrier level control) for the proper injection. This is normally about 5% for an RDS subcarrier, 10% for an SCA.

7. Put the Mod-Monitor back into its "Total Peak Modulation" measurement mode and, if necessary, trim the OUTPUT LEVEL control so that the peak flasher(s) indicate only *occasional* 100% peaks.

*NOTE:* Depending on current operating rules and practices, when either an SCA or an RDS subcarrier is broadcast, total peak carrier deviation *may be permitted* to exceed the customary 100%-modulation figure. The station (or consulting) engineer should be aware of legal guidelines in effect and assume responsibility for any adjustment which deviates from this procedure.

## AVERAGE COMPRESSION ADJUSTMENT

Once the OUTPUT LEVEL has been set for proper carrier peak deviation, the user may determine a setting for the AVERAGE COMPRESSION control based on listening tests. A guideline already expressed has stated that "hard rock" formats may use any AVERAGE COMPRESSION setting up to 10dB, probably without ill effect. More conservative practice, particularly on "mellower" formats, would dictate a 6dB to 8dB maximum setting.

## COMPOSITE PROCESSING

This control will have a definite effect on program "loudness," but not without some tradeoff for distortion or probable interference between the main program and any auxiliary subcarriers. Please see the discussion under COMPOSITE PROCESSING on Page 14.

# Section IV

# CALIBRATION

The upgrade to PWM processing has greatly simplified calibration of the *DAVID-II* when compared with its predecessor, the Model 715 *DAVID*. The number of trimmer adjustments under the top cover has been reduced from *nineteen*(!) to only eight.

"Routine" calibration of the *DAVID-II* is never required. The PWM processing and digital-synthesis circuitry is simple and stable. The only justification for confirming any of the internal adjustments is out-of-specification operation, and only after all other possible faults have been considered and eliminated. It's worth a call to the factory to discuss any anomalies before charging-ahead and possibly making things worse.

**Equipment Required**

- **Dual-Trace Oscilloscope** — 5mV sensitivity, 20Mhz bandwidth, with two matched 10:1 probes.

- **Audio Generator** — 10Hz–1MHz, +20dBm output capability.

- **Digital Multimeter**

- **Frequency Counter** — capable of accurate frequency measurement from 1kHz to 2Mhz.

- **"Precision" FM-Stereo Demodulator** — station Mod-Monitor with input for composite baseband signal.

- **Spectrum Analyzer** (optional*) — must have good resolution in the 100Hz to 100kHz (FM composite baseband) display range.

   * The Spectrum Analyzer is *not* required if the "Precision" FM-Stereo Demodulator is capable of resolving crosstalk, stereo separation and 38kHz "residual" measurements to a figure of 70dB or better.

**Power Supply Check**

1. Apply power to the *DAVID-II*.

2. Check that the positive and negative 9-volt regulated supplies are between 8.5V and 9.0V. You can check these on pin 8 (positive) and pin 4 (negative) of any 8-pin IC.

**Clock Set**

3. Use one of the low-capacitance 'scope probes to connect to the input of the frequency counter. Check the clock frequency on pin 11 of IC8.

4. Adjust C16, the trimmer capacitor to the right of the crystal, for exactly 1,216,000Hz.

**Low-Pass Filter Matching**

The low-pass filter for the left program channel is not adjustable; rather, it employs close-tolerance components to meet design parameters. Since close frequency and phase response matching between stereo channels is essential to acceptable Stereo Generator crosstalk performance, trim controls have been included in the right channel filter so that it may be tuned to match left channel response.

1. Set the AUDIO PROCESSING switch to the OUT position. Frequency and phase response matching is optimized with all processing bypassed.

2. Clip the two 'scope probes to the output of the filter buffer amplifier, IC20. The most convenient monitor points are the right-hand ends of R22 and R27, for the left and right channels, respectively. The right-hand ends of the resistors are the ends closest to the COMPOSITE OUTPUT connector.

3. Apply a 10kHz sinewave test signal to both the LEFT and the RIGHT PROGRAM LINE INPUT. This is most simply done with clip-leads if the barrier screw-strip is first unplugged from the chassis. Connect the "hot" side of the oscillator output to *both* + terminals, the grounded side of the Audio Generator output to either G. The − input terminals may be left unconnected for test purposes.

4. Set both 'scope attenuators at 2V/div. Position the left channel signal (from R22) above the right.

5. Adjust the Audio Generator amplitude for a 6-volt (3 divisions) peak-to-peak signal on both 'scope traces. If necessary, trim either the L or R front-panel INPUT GAIN control so that both traces are equal in amplitude.

6. Increase the Audio Generator frequency to the first filter null, approximately 19.1kHz. Observing the upper (left channel) trace, down-range both oscilloscope vertical attenuators and fine-tune the Audio Generator frequency for the deepest null in the left channel signal.

*HINT:* When properly nulled, the signal waveform will be very close to the noise floor. If you find the nulled signal difficult to resolve, try triggering the 'scope timebase directly from the Audio Generator. This will make the waveform "hold still" and stand out from the (random) noise.

7. When the Audio Generator has been fine-tuned to null the left channel, adjust R112 (between IC26 and IC27) for a null in the right channel filter. Both channels should null to the same relative amplitude.

8. Increase the Audio Generator frequency to the second filter null, approximately 21.8kHz. Again, fine-tune the Audio Generator frequency for the deepest null in the left channel signal.

9. Adjust R65 (between IC21 and IC22) for a right-channel null.

*NOTE:* The third filter null is at approximately 34.5kHz. Rather than to match *frequency* response between channels at this point, the third filter adjustment is used to help match *phase* response in the next procedure.

**Crosstalk Trim**  If your station Mod-Monitor includes a true "precision" FM-stereo demod (crosstalk/separation resolution of 70dB or better), it may be used in lieu of a Spectrum Analyzer for this part of the Calibration Procedure. If a Mod-Monitor of *insufficient precision* is used, final Calibration (and performance!) of the *DAVID-II* will reflect the inaccuracy.

1. Connect the COMPOSITE OUTPUT of the *DAVID-II* to the input of the Spectrum Analyzer (or directly to the Composite Input of the Mod-Monitor).

2. Turn the output of the Audio Generator completely off.

3. With the *DAVID-II* MODE switch in STEREO, and PILOT switch ON, adjust either the front-panel OUTPUT LEVEL control (or the Spectrum Analyzer or Mod-Monitor input gain attenuator) for a 19kHz pilot reading of –20dB (10% injection).

4. Adjust R20 (between IC12 and IC17) to null the 38kHz second-harmonic-distortion component of the stereo pilot. The Spectrum Analyzer should display the 38kHz component at the noise floor; the Mod-Monitor should indicate a 38kHz "residual" reading at the bottom of its most sensitive scale (–70dB or better).

5. Next, re-apply the Audio Generator test signal to the LEFT and RIGHT PROGRAM LINE INPUT of the *DAVID-II* as described in Step 3 of the Low-Pass Filter Matching section of this Procedure. The Audio Generator frequency should be set to 1kHz, and the level set the same as before. If in doubt about the level, check for 6-volt p-p signals on the right-hand ends of R22 and R27.

6. Monitor the "sub channel." This is the 38kHz double-sideband, suppressed-carrier component on the Spectrum Analyzer (or an "L–R" Mod-Monitor reading).

7. Carefully adjust the front-panel INPUT GAIN L control for a null in the sub channel.

8. Adjust R157 (to the left of IC35) to further null the sub channel.

*NOTE:* Because the effect of the two preceding adjustments will appear to interact, it will be necessary to repeat Steps 7 and 8 until no further improvement in the sub-channel null can be achieved.

9. Increase the Audio Generator frequency to 10kHz. Adjust R111 (between IC24 and IC25) for the best null in the sub channel at 10kHz.

10. Repeat Steps 7, 8 (at 1kHz) and 9 (at 10kHz) until the sub-channel null at both frequencies is at maximum. You may wish to "split the

difference" to balance crosstalk measurements, sacrificing a null at one frequency to improve that at the other.

When this part of the Calibration Procedure has been successfully completed, the sub-channel measurement should remain below –50dB (relative to 100% modulation) as the Audio Generator is swept from 20Hz to 15kHz.

Because the *DAVID-II* utilizes digital techniques (instead of analog multiplication) to generate the subcarrier, crosstalk is symmetrical; that is, sub-to-main crosstalk will be identical with the main-to-sub, which was optimized in the foregoing Procedure.

**Separation Trim**

The traditional method of adjusting and verifying stereo separation is to visually monitor an oscilloscope display of the composite multiplex signal (one channel driven and stereo pilot off), and adjust for the straightest baseline of the DSB waveform. Although this technique does permit a rough separation calculation, even today's best 'scopes will show baseline distortion when overdriven to the extent necessary to resolve separations of 40dB or more. For the high figures of separation which modern stereo generators can achieve, a precision stereo demodulator will give much greater measurement and, consequently, adjustment accuracy.

It is entirely feasible to trim the equalization and phase adjustments in the *DAVID-II* output circuitry to correct for shortcomings elsewhere in the transmission system; for instance, in a "composite" STL, or even in a long run of coax cable. This procedure does not consider this option, however, assuming instead that, since a chain is only as strong as its weakest link, it's better to fix the weak link than to compensate for it at another point. The procedure which follows will leave the *DAVID-II* adjusted for *its own* best stereo separation.

1. Cable the COMPOSITE OUTPUT of the *DAVID-II* directly to the composite input connection of the Mod-Monitor. Use a short cable of known characteristics, preferably 50-ohm or 75-ohm coax.

2. With the *DAVID-II* in the STEREO mode and with the PILOT switched ON, apply a 10kHz sinewave signal to the left channel input only. The signal level should be identical with that used in previous procedure steps. If in doubt, check for 6-volts p-p on R22.

3. At this point, check the Mod-Monitor's display of Total Peak Modulation. It should read just about 100%. Adjust the *DAVID-II* OUTPUT LEVEL control for an exact 100% reading.

4. Also check the Mod-Monitor indication of Pilot Injection Level. If necessary, adjust the *DAVID-II* PILOT INJ. control for a 9% reading.

5. Switch the Mod-Monitor to display left and right channel levels. The left channel should measure about –1dB. Ideally, the right channel should read –70dB.

6. Adjust the equalization and phase trimmers, R56 and R57, for best stereo separation. There is some interaction between these two adjustments, but there will be only one setting of each which will give best separation. Null one, then the other, and back-and-forth, etc., until no further improvement can be had. The final adjustments should yield best separation at all frequencies between 20Hz and 15kHz, left-into-right and right-into-left.

# Section V

# CIRCUIT DESCRIPTIONS

This section details the circuitry of the Inovonics *DAVID-II*. Circuit descriptions refer to the two pages of Schematic Diagrams contained in the Appendix, Section VI, Pages 39 and 40.

**Component Annotation**

*DAVID-II* Schematics may appear to be annotated in a somewhat haphazard manner, insofar as component reference designations are concerned. Rather than annotate the *schematic* in a logical sequence, we have instead chosen to designate the *components on the circuit board*, top-to-bottom, left-to-right, following the physical placement of the parts in their neat little rows. It is our expectation that this practice will aid any required troubleshooting, making it easier to locate the physical part or test point from an analysis of the circuit diagram.

The first part of this section covers the general subject of Pulse Width Modulation (PWM) as it applies to the audio processing circuitry. Signal path circuitry discussions follow.

## A PWM PRIMER

**PWM Gain Control**

The *DAVID-II* utilizes Pulse-Width Modulation techniques rather than voltage-controlled amplifiers or FET attenuators in its three distinct processing sections. This technology, as applied to broadcast audio processing, was pioneered by Inovonics in 1982. PWM is a simple, colorless, quasi-digital method of linear gain control which avoids the several hassles associated with more expensive, possibly single-sourced VCA ICs or matched and balanced FET devices. Our implementation of PWM utilizes the popular 4000-series CMOS parts which are dirt cheap and available nearly anywhere.

PWM sets the gain of an analog circuit simply by sampling, or "chopping," the analog signal; that is, turning it on and off at a rapid rate. Consider an audio program signal which can be turned on and off with a toggle switch. When the switch is on, attenuation is zero. When off, attenuation is infinite. If we satisfy Dr. Nyquist's requirement and toggle this switch at a rate at least twice that of the highest audio frequency, linear signal attenuation becomes directly proportional to the OFF time as depicted in the chart on the next page.

The *DAVID-II* sampling rate is 152kHz, better than ten times the highest program audio frequency. Since this is well above the Nyquist rate, anti-aliasing and reconstruction filters can be simple and gentle.

| Switch ON | Switch OFF | Attenuation |
|-----------|------------|-------------|
| 100% | 0% | 0dB |
| 50% | 50% | 6dB |
| 25% | 75% | 12dB |
| 10% | 90% | 20dB |
| 1% | 99% | 40dB |

**Feedforward Control**

Both the wideband and the independent high frequency compression/ limiting circuits operate in a feed*forward* mode, rather than using the more traditional feed*back* method. The amplitude of the *input* signal establishes circuit gain to hold the output constant at a 100%-modulation "ceiling" value.

Feedforward control necessitates a conversion factor to give one dB of output *attenuation* for each dB that the input signal *increases*. Put into linear (voltage), rather than log (dB) terms, this function is expressed as $X=1/Y$. Figure 6 graphs this function. X represents the circuit gain needed to hold the output at a constant 1 volt, with a 1-volt to 10-volt input signal Y. This technique becomes a bit precarious at small duty cycles (attenuations of 40dB or more), but in the 0 to 30dB range typically required for audio processing, PWM is easily implemented and quite predictable.
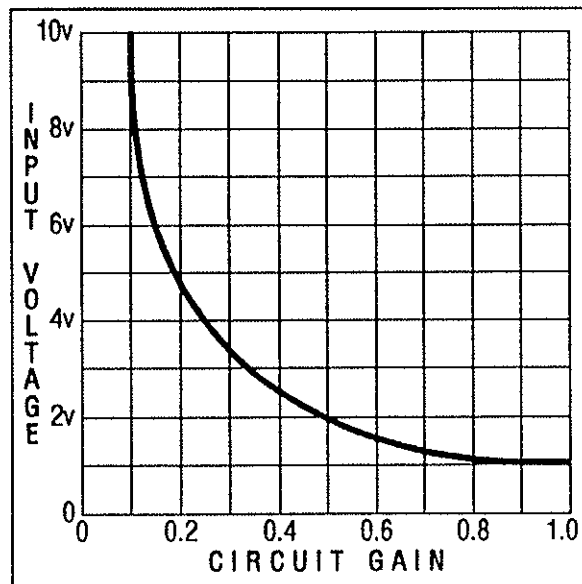


**Figure 6**
Voltage Transfer
Function

Assuming a linear relationship between PWM on/off time and signal gain reduction, the 1/Y expression translates directly to the duty cycle; that is, 0.1 is a 10% on time, and 1.0 a 100% on time.

**"Soft Knee"**

In actual implementation we require a "threshold" level, below which gain will remain at full value. *Just* above threshold, the transfer function will conform to what has become known as a "soft knee"

characteristic. This means that the program signal eases *gently* into compression and limiting, avoiding the abrupt change from a linear to a limited state which characterizes traditional feedback topology. The "soft knee" is normal behavior for feedforward gain control, and is credited for the smoother, less fatiguing sound attributed to this method.

**The PWM .** Figure 7 shows a "ramp" waveform generated by the *DAVID-II* to
**"Ramp"** effect the desired feedforward transfer characteristic. Note the
**Waveform** similarity between this waveform and Figure 6. Allowance has been made for the "soft knee" compression and limiting threshold, and a negative-going "pedestal" has been included to keep the CMOS analog switches active, even when no gain reduction is in effect.



V=1v/div.
H=1μs/div.
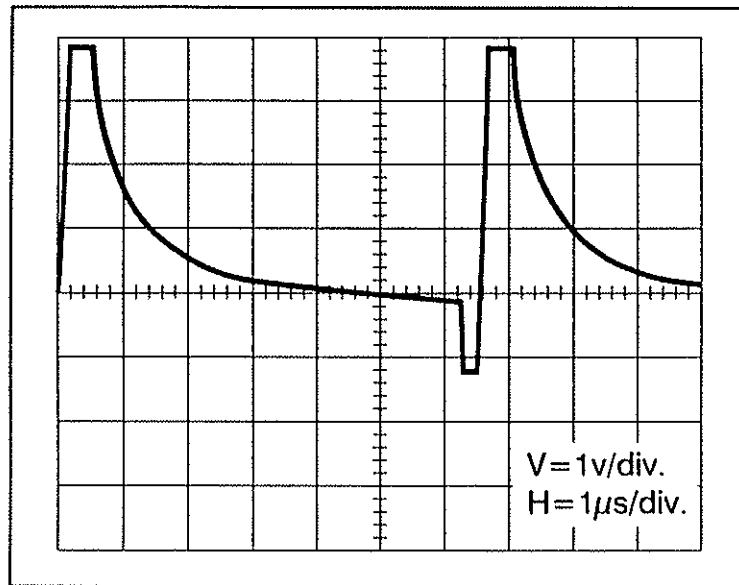
**Figure 7**
Ramp Waveform

This ramp runs continuously at a 152kHz rate. It is applied to one input of a comparator, and a DC voltage derived from the input signal is fed to the other comparator input. The comparator output is the duty-cycle-modulated squarewave which "chops" the program audio signal.


## PWM RAMP CIRCUITRY

**Master Clock** Referring to the second sheet of the Schematic on Page 40, IC8C is the *DAVID-II* master "clock" running at 1.216Mhz, 64 times the stereo pilot frequency. Binary dividers IC7A, IC3A and IC3B derive the 152kHz PWM switching frequency. This is buffered by IC8A and IC8B, and is capacitor-coupled to the base of Q10. Q10, held in saturation by R103, is momentarily turned off with each negative transition of the switching waveform. When Q10 turns off, Q9 turns on for the same brief period. With Q8 serving as an "active pull-up resistor," Q9 delivers a stiff negative-going 200ns pulse at the 152kHz rate.

— 25 —

The negative-going pulse from Q9 triggers a similar event with Q6 and Q7. On the positive transition of the pulse, Q6 is turned off, allowing Q7 to turn on. C49 and R99 establish the off time for Q6, hence the on time for Q7.

**Compressor/ Limiter Ramp**

When Q7 turns on, C52 is charged to very nearly the +9-volt power supply rail through CR18. As soon as Q7 turns back off, C52 begins to discharge through three distinct paths.

The cathodes of CR21 and CR22 are held at two different DC levels by the attendant divider strings. This means that C52 initially discharges through R134, R135, and the series connection of R113 and R114. When the charge falls to about +6 volts, CR21 is biased out of the circuit, removing R135 from the discharge path. As the charge approaches +1 volt, CR22 removes R134, leaving the R113 and R114 to discharge C52 the rest of the way. The effect is a quick initial discharge of C52, slowing as the DC level drops. This yields the modified X=1/Y waveform slope shown in Figure 7.

**"Pedestal"**

Because the LM319 fast comparators cannot deal with an input voltage range which approaches the supply rails, the ramp is taken from the R113/R114 divider. A fixed amount of the negative-going 200ns pulse is coupled through R115 to this same point, creating the "pedestal" mentioned earlier. Q11 is an emitter-follower buffer for the "composite" ramp waveform, applying the ramp to the + inputs of comparators IC42A and IC29A. These generate the PWM squarewave switching waveform for the broadband and the independent high frequency limiters, respectively.

**AGC Ramp**

The AGC function of the *DAVID-II* is a very slow "gain servo" operating in a feed*back* rather than a feed*forward* mode. The AGC requires a PWM ramp of a different shape; not for a proper gain transfer characteristic, but to maintain a constant *rate* of AGC correction.

Each time Q7 turns on, C84 is charged through CR40. Unlike the complex discharge path of C52, C84 has a simple RC discharge through R204 and R234. Since the AGC operates in a feedback mode, the shape of this ramp is far less critical than the one for the wideband and high frequency feedforward limiters. The AGC ramp is applied directly to the + input of the third comparator, IC42B. Though Q7 initiates the two ramp waveforms, CR18 and CR40 ensure that each can assume its characteristic discharge shape.

**Processor Bypass**

S4, the front-panel AUDIO PROCESSING switch, allows the user to completely bypass the *DAVID-II* audio processing system. When S4 is switched OUT, binary divider IC3B is inhibited. Lack of drive holds Q7 off which stops the AGC and limiting PWM ramps. This drives the outputs of all three comparators low, opening the signal path at each PWM attenuator. S4 also switches IC40A and IC40D to route the program input signal around the processing circuitry.

# SIGNAL PATH CIRCUITRY

> *NOTE:* Left and right program channel circuits are identical, so only the left channel will be described here. A skilled technician should find it within his capability to extrapolate for the right channel.

## INPUT STAGES

**Line Input Balancing**

IC44B is an "active-balancing" stage for the left channel program line input, affording rejection of unwanted common-mode signals. When jumper JMP5 connects R268 *into* the network (H), the Line Input accepts "High" program levels between 0dBu and +15dBu. With R268 jumpered *out* (L), "Low" program levels from –15dBu to 0dBu are accommodated. Input range setup is described on Page 8.

**"Bass Enhance"**

IC50B performs the dual function of a variable-gain stage and the exclusive "Bass Enhance" equalizer.

With jumper JMP6 *closed* (F), low frequency response of the stage is "Flat" to at least 20Hz. With JMP6 *open*, the stage becomes a high-Q high-pass filter with a low-end boost. With this jumpering the response of the stage rises below 500Hz, peaking with about 4dB of boost at 80Hz, then falling rapidly. (See Figure 4 on Page 10.) The effect of this special equalization is to accentuate frequencies which are *perceived* as "bottom-end," while attenuating sub-audible program components which can compromise overall modulation efficiency.

The feedback path of IC50B includes the front-panel INPUT GAIN control, R242. This affords the 15dB of continuous level adjustment within each jumpered input gain range.

The "design center" sinewave signal level at the output of IC50B (the level which will maintain AGC gain at 0dB) is –5dBu.

## AGC STAGE

From the output of IC50B, the program signal takes one of two paths. One path bypasses the audio processing section of the *DAVID-II* entirely, with analog switch IC40A re-inserting the program signal downstream of processing and pre-emphasis. The program normally encounters IC46C, the first analog-switch PWM attenuator.

IC47B and IC48B comprise a 4-pole low-pass "reconstruction" filter which removes the 152kHz PWM switching component and restores the program signal to its original (audio) spectrum. The small amount of high frequency "shelving" boost provided by C123 and R282 is important for maintaining compressor/limiter flatness downstream. This boost is canceled by complimentary equalization in a later stage.

**DC Control
Signal Rectifier**

IC49B is a simple unity-gain inverter. Along with diodes CR45 and CR46, it forms a full-wave rectifier for the left channel program signal. The right channel has an identical rectifier, and the resulting rectified voltage represents the *higher* level of the two channels. This voltage is used by the feedback-mode AGC and the feedforward compressor/limiter.

The common-cathode rectifier configuration is the equivalent of an analog OR gate. This causes gain control for the stereo program to respond to whichever channel happens to have the greater energy. Slaving the two channels in this manner is essential to preserving the stereo image.

**"PPM"
Response**

The full-wave-rectified peak level of the program audio is buffered by IC28C. For AGC control, the peak value is captured by CR20 and integrated by R141 and C67. The discharge path of C67 is back to the buffer output through R116. The net response of this circuit approximates the ballistics of the European Peak Programme Meter, or "PPM," generally judged superior to an American "VU" meter in accurately representing program dynamic range. With the AGC emulating a conscientious (but slow-moving!) operator who manually rides gain while watching a PPM, subsequent processing stages are presented a uniform level for much more consistent results.

**Servo Stage**

IC30A compares the integrated peak value of the program signal with a DC level corresponding to "0dB" AGC gain. When the program level is too low, the output of IC30A toggles positive. When the program level is too high, IC30A toggles negative.

Integrator IC30B translates the instantaneous need for more or less AGC gain into a very slow, constant-slope linear ramp. The output of IC30B ramps down for a low program level, and up for a high one. This DC is applied to the – input of AGC PWM comparator IC42B and compared with the 152kHz PWM ramp from C84. The duty-cycle-modulated squarewave from IC42B switches IC46C to set AGC gain. The AGC ceaselessly chases the input program peak value in its attempt to hold the level steady. Though the AGC never actually reaches equilibrium in normal operator, circuit action is so slow that this constant correction is not audible.

**AGC Metering**

The linear DC ramp from IC30B is also routed to IC43. These four op-amp sections drive the five front-panel AGC GAIN indicator LEDs. Each section is configured for gain and offset bias so that the five LEDs fade smoothly, one to the next. This gives the indicator string more display resolution than might be expected for only five LEDs, since intermediate values of gain become proportional to the difference in brightness between adjacent indicators.

**Gating**

AGC in the *DAVID-II* is "gated." When the input program falls below a predetermined level, the AGC returns to its normal 0dB "resting" value, rather than increasing gain and bringing up background noise. The gating threshold is fixed at about –20dB with respect to the

corrected average program level. Moreover, the gating threshold is frequency-weighted for greatest sensitivity in the voice range.

Gating is based on L+R energy. The left and right channels are summed by IC36B through R276 and R255. CR30 and CR29 comprise a full-wave rectifier at the input of comparator IC29B. The output of IC29B is pulled to ground so long as the corrected program level is above the gating threshold. This lights the GATE OPEN indicator and pulls the junction of R138 and R162 to a slightly negative value.

When the program falls below the gating threshold, the output of IC29B goes positive. A positive DC bias from R138 and R162 passes through CR24 to the program peak integrator comparator, IC30A. If the last value of AGC gain is positive, this DC bias toggles IC30A negative to slowly reduce AGC gain.

The divider at the base of Q12 is scaled to turn the transistor on when AGC gain reaches 0dB. This will pull the bias from IC29 toward ground when AGC gain reaches 0dB. If the last value of AGC gain is negative, Q12 inhibits the DC bias from IC29B until gain reaches the 0dB value.

The gain-corrected program signal at the output of IC48B has a nominal (sinewave) level of 0dBu at 400Hz.

## WIDEBAND COMPRESSION AND LIMITING

**Compressor/ Limiter Timing**

As discussed in the description of the AGC circuitry, the output of IC28C is the full-wave-rectified program signal. CR17 passes the peak value of this signal to a fast integrator consisting of R105 and C53. Integration is on the order of 1ms and represents limiter *attack* time.

Assuming the front-panel AVERAGE COMPRESSION control, R145, is turned fully clockwise (wiper to the output of IC28C), integration through R117 causes C54 to charge to the *average* value of the program waveform. The peak integration capacitor, C53, discharges to this average value through R106. This time constant, about 50ms, establishes the initial peak limiter release to the average compression "platform" value. Platform release, back through R117, is about one second. In practice, C53 discharging into C54 elevates the platform when the program material has numerous, repetitive peaks. The resulting program-controlled, "floating platform" release characteristic inhibits program self-modulation, or "pumping," effects.

**AVERAGE COMPRESSION Control**

As R145 is turned counterclockwise, a positive DC bias is added to the platform value. This introduces a variable attenuation which is reflected as program signal gain reduction *ahead* of the compressor/limiter circuit. Front-panel markings relate to an equivalent figure of average level compression for typical program material.

IC28D buffers the DC gain-controlling voltage and delivers it to comparator IC42A. The PWM squarewave from IC42A drives PWM attenuator IC46D. IC45B and IC37A comprise another 4-pole low-pass reconstruction filter to remove the 152kHz PWM switching component.

**Gain Reduction Indicators**

IC28C is a slope detector sampling the wideband compressor/limiter DC control voltage. The output of this op-amp toggles positive, lighting the W.B. LED, when a *rapid* voltage increase indicates the reduction of a program peak.

IC23D performs a similar function for the DC voltage corresponding to the *average* program level. Configured as a linear gain stage, the output normally sits at ground. With no compressor activity, both the – and the + AVERAGE COMPRESSION indicators will be lighted to half-brightness. The slope of the DC voltage derived from the average program level is translated into varying brightness between the two LEDs, indicating the *direction* of compressor gain and, to some extent, the *amount*.

## HIGH FREQUENCY LIMITER

C75, R193/R194 and IC38A constitute an equalizer with a constant 6dB/octave rising characteristic. IC41A sums this equalized signal (via R198) with "flat" program audio (via R197), to impart proper pre-emphasis to the program audio. JMP1 may be jumpered for the 75$\mu s$ (American) or 50$\mu s$ (CCIR) pre-emphasis standard.

Independent high frequency limiting is afforded by feedforward gain control of the equalized program signal. Inverter stage IC39A, with diodes CR38 and CR39, yield full-wave rectification of the equalized signal. The resulting DC is integrated by C70. Limiter attack is established by R181, release by R180. The output of buffer IC36A is fed to comparator IC29A which switches PWM attenuator IC40B. IC23A lights the front-panel H.F. indicator for any appreciable high frequency limiter action.

**Pre- and Post-Limiting Equalization**

Even though the equalized program signal is held constant at a 100%-modulation level, an unavoidable level buildup occurs near the pre-emphasis crossover frequency when it is summed with flat program audio. To restore proper frequency response, C78 and R202 give the summing amplifier a shelving roll-off characteristic. This is complimentary to the shelving boost imparted by IC48B. Net response conforms to the selected pre-emphasis curve *below* the high frequency limiting threshold, and maintains a 100%-modulation level at all frequencies *above* it. This limiter "ceiling" value (at the output of IC41A) is +5dBu.

# FILTER OVERSHOOT COMPENSATOR

**Sources of Overshoot**

All low-pass filters exhibit a certain amount of overshoot and ringing when presented with complex input waveforms. Generally, the sharper the cutoff, the more pronounced the effect. Overshoots result from the elimination of higher-order input signal components which, prior to filtering, helped define the signal peak amplitude. Even a fully phase-corrected filter will exhibit overshoots, and a 7-pole "elliptic" filter, as used in the *DAVID-II*, can overshoot 3dB or more!

Other systems of overshoot control permit the primary low-pass filter to overshoot, then isolate and re-introduce the overshoots to cancel themselves in the signal path. The patented overshoot compensator in the *DAVID-II*, on the other hand, so pre-conditions the limited program signal *ahead* of the filter that there is little or no tendency for the filter to *generate* overshoots.

**Input Clipper**

CR33 and CR34 form a "hard" clipper at the compensator input. These diodes are biased to a point which represents 100%-modulation. Since the program signal has already been limited to this same value, the two diodes rarely clip *legitimate* program waveshapes. Clipping at this point is limited to those fast transients which exceed limiter attack timing, and the clipping of which is judged less obtrusive than a temporary reduction in overall broadband level.

**Phase-Lag and Recombining**

IC34B includes a phase-lag network which time-displaces the fast leading and trailing edges of steep waveforms. This means that the primary characteristic of a program waveform which would normally *excite* filter overshoots is instead added to the waveform *amplitude*. CR32 and CR31, also biased to the 100%-modulation level, "strip" these displaced-and-added components from the program signal. IC32B compares the "stripper" input and output, recovering the stripped-off components. As these contain much of the program harmonic (high frequency) information, we cannot afford to simply throw them away. By recombining these stripped-off program components with the stripped program signal *in opposite phase*, the spectral integrity of the program is maintained. This 180-degree displacement of certain program overtones is not discernible to the listener, but is quite effective in inhibiting filter overshoots.

IC19B combines the stripped-off signal components form both the right and the left channels. This is full-wave rectified by Q8 and Q9 to flash the front-panel FILTER COMP indicator whenever filter compnesator action takes place.

# LOW-PASS FILTER

The 7-pole, elliptic-function (Cauer) low-pass filter is an active version of the classic L-C designs worked-out in Germany during the late 1940s (probably with a slide rule!). The particular active configuration used in

the *DAVID-II* is sometimes called the "FDNR" because each of the legs to ground simulates a Frequency-Dependent Negative Resistance. Referring back to the classic L-C design, resistors in series with the signal replace series inductors, and each of the active circuits to ground replaces an inductor/capacitor series-resonant element.

Components in the left channel filter have fixed values. The resonant frequency of each of the right channel legs is variable over a small range so that the two channels may be precisely matched in amplitude and phase response.

A great "cookbook," with this and other filter circuits which might prove valuable to the broadcaster, is the *Electronic Filter Design Handbook* by Arthur B. Williams, published by McGraw-Hill. The reader is kindly directed to this source for a more informed discussion of how the FDNR circuit works than we could possibly muddle-through here.

IC20B buffers the output of the low-pass filter and includes gain, both to compensate for the 6dB filter loss and to present the proper amplitude to the digital synthesis circuitry. The signal level corresponding to 100% modulation is +10dBu at the output of IC20B.

# PILOT AND SUBCARRIER GENERATION

**Clock**    As previously described in the PWM discussion, IC8C is a crystal-controlled oscillator at 1.216Mhz. IC8D buffers the clock, and IC7A, the first binary divider, furnishes the 608kHz digital synthesis sampling frequency.

**Pilot Generation**    IC6 is an up/down BCD counter clocked at the 608kHz sampling frequency. 1-of-10 decoder IC11, OR gate IC10C, and binary divider IC5A work together to keep IC6 continually counting from zero to 8, back down to zero, etc. Counting logic is decoded by a 1-of-9 de-multiplexer (IC14 and IC16C) which samples a resistor string with sine-weighted values. This generates *one-half* a sinewave for each counting cycle. IC7B reverses the DC polarity applied to the top of the resistor string for *every-other* up/down count, forming the complete 19kHz pilot sinewave from 32 discrete steps. R20 introduces an offset to compensate for any difference between the power supply rails and can be adjusted to null the 38kHz second harmonic component. The segmented Stereo Pilot is buffered by IC15A

**Center-Sampling**    Analog switch IC16A is controlled directly by the 1.216MHz clock, turning on for one-half of one clock period precisely at the *center* of each stepped pilot waveform sample. This charges C19 to the sample voltage value, which is held by buffer stage IC17A until the next center sample is taken. Center-sampling eliminates integration of switching noise which is concurrent with leading and trailing edges of the waveform steps.

**PILOT Switch**

In its OFF position, the front-panel PILOT switch, S3, is arranged to interrupt the pilot sampling sequence and preset IC6 to a zero count. This stops the stereo pilot at its zero crossing.

**Subcarrier Generation**

The FM-multiplex "composite" waveform consists of a "main" channel, conveying the L+R stereo sum in its normal audible frequency range, and a "sub" channel with L–R stereo difference information in the form of a 38kHz double-sideband, suppressed-carrier subcarrier. The 19kHz pilot is added to enable stereo receivers to reconstruct and re-introduce the 38kHz carrier for difference signal demodulation.

Digital synthesis of the composite waveform is similar to the generation of the Stereo Pilot previously discussed, but with sinusoidal *commutation* between the left and the right stereo program channels at a 38kHz rate.

Up/down counter IC4 is clocked at the 608kHz sampling rate. The output is decoded by IC9 and, with gating provided by IC10D and CR7, IC4 counts continuously from zero to 8, back to zero, etc. A Pulse from IC12A presets the counter to its center position (count of 4) each time the pilot reaches the proper phase relationship. This ensures proper and constant synchronization between pilot and subcarrier.

IC13 and IC16D also decode the count, sequentially sampling each tap of the resistor divider string which bridges the left and right program audio signals. Sampled program audio is buffered by IC15B, center-sampled by IC16B, and held between samples by C20 and buffer IC17B. The stereo multiplex signal consists of 16 discrete, sinusoidally-weighted steps.

When the front-panel MODE switch, S2, is set to MONO, the counting sequence is halted with the counter preset at the center position (count of 4). This stops the subcarrier generation, and L+R audio appears at the *DAVID-II* output. In the MONO position, S2 also stops the stereo pilot, regardless of how S3 is switched.

**Composite Processor**

Diodes CR8 and CR9 are biased to the same 100%-modulation level as the clipper diodes in the overshoot compensation circuit. The buffered composite signal, less the stereo pilot, is presented to this clipper through front-panel COMPOSITE PROCESSING control R64. When R64 is set to its fully counterclockwise position (OUT), the two diodes catch only the occasional program peak which somehow has eluded the earlier processing and clipping stages. As R64 is cautiously turned clockwise, program peaks at the 100%-modulation value may be subjected to as much as 3dB of clipping.

Even though the composite waveform is clipped prior to pilot injection, good broadcasting practice demands judicious use of this feature. (See Page 14.)

| | |
|---|---|
| **Combining Amplifier** | IC18A is a combining stage for the main/sub portion of the composite signal, the stereo pilot, and an optional RDS or SCA subcarrier applied to the rear-panel SUBCARRIER INPUT jack. The front-panel OUTPUT LEVEL control, R13, is in the feedback path of IC18A, affording adjustment over the composite level delivered to the exciter. |
| **Output Filter** | Since all components of the composite output are digitally synthesized at a 608kHz rate, only a simple filter is required to eliminate the sampling frequency from the *DAVID-II* output. L2, L1, C31 and C30 form a 4-pole filter with quasi-Butterworth response. This filter is<br><br>exceptionally flat to about 100kHz, but effectively attenuates higher-order digital noise components in the *DAVID-II* output. |
| **Output Phase and Equalization Trimmers** | The output amplifier consists of IC18B, IC19A, and transistors Q1 and Q2. R57 affords variable high frequency tip-up, and R56 varies the phase response. These two controls can be used together to optimize stereo separation throughout the transmission system. As adjusted at the factory, separation is maximized into a resistive load. Using a precision stereo demodulator, the two pots may be re-tweaked to compensate for response or phase irregularities in a composite STL or exciter input circuitry. |

## POWER SUPPLY

*DAVID-II* circuitry operates from a bipolar 9-volt supply. The two supplies are regulated by linear "three-terminal" IC voltage regulators: IC2 for the +9-volt supply, IC1 for the –9-volt supply. The power transformer has dual primary windings which may be switched in parallel or in series for 115V or 230V mains, respectively.

# Section VI

# APPENDIX

The following section of this Manual contains Parts Lists for the Inovonics *DAVID-II*, Schematic Diagrams of all electronic circuitry, and an explanation of Inovonics' Warranty Policy.

# PARTS LIST

## EXPLANATION OF PARTS LISTINGS

This section contains listings of component parts used in the Inovonics *DAVID-II* FM-Stereo Processor/Generator. These are listed either *en-masse,* or by schematic component reference designation, and may, or may not, specify a particular manufacturer. When no manufacturer is called-out, the term "open mfgr." advises that any manufacturer's product is acceptable.

If a particular component is not listed at all, this means that we do not consider it a typical replacement item. Should you need to order an unlisted part, call, write or FAX the factory with a brief description and we'll do our best to figure out what you need and get it on its way to you.

## PARTS LISTING

Unless specifically noted by component reference designation below, **capacitors** are specified as follows:

a) **Under 100pF** are "dipped mica" type, DM-15 (or CM-05 military series) size designation; "P" value is picofarads, ±5%, 200VDC; (open mfgr.).

b) **100pF to 0.47μF** are of the metalized mylar or polyester variety; whole number "P" values are picofarads, decimal values are microfarads, ±5%, 50VDC or better. The style used in the *DAVID-II* is the "minibox" package with lead spacing of 0.2 inch. **Preferred mfgr.:** Wima, MKS-2 or FKC-2 series. **Alternates:** CSF-Thompson IRD series or Roederstein KT-1808 or KT-1817 series.

c) **1.0μF and above** are radial-lead electrolytics, value per schematic, 25VDC; (open mfgr.).

| | |
|---|---|
| C1,2 | Capacitor, Ceramic Disc "Safety" Mains Bypass, .0047μF, 440VAC; Murata/Erie DE7150 F 472M VA1-KC (preferred) |
| C5,6 | Capacitor, Electrolytic, axial leads, 470μF, 35VDC; (open mfgr.) |
| C7,8,9,11,15, 48,50,62,72,95 | Capacitor, Monolithic Ceramic, 0.1μF, 50VDC; (open mfgr.) |
| C16 | Capacitor, Variable, 5-50pF; Mouser 24AA024 |
| C39,40 | Capacitor, Electrolytic, radial leads, 220μF, 6VDC; (open mfgr.) |
| C23,24,25,36, 37,38,43,44, 45,46,47,56, 57,58, 59,60 | Capacitor, "High-Q," .0033μF, 2.5%, 100VDC; Wima FKC-2 (Polycarbonate) preferred, any equivalent *must* have identical characteristics. |

| | |
|---|---|
| CR1-6 | Diode, Silicon Rectifier; (open mfgr.) 1N4005 |
| CR7,10-47 | Diode, Silicon Signal; (open mfgr.) 1N4151 or equiv. |
| CR8,9 | Diode, Schottky; (open mfgr.) 1N5711 |
| F1 | Fuseholder, PC-mounting; Littlefuse 345-101-010 with 345-101-020 Cap for ¼-inch (U.S.) fuses, or 345-121-020 Cap for 5mm (European) fuses. (Fuse is normal "fast-blow" type in value specified on rear panel with reference to mains supply.) |
| FB1 | Ferrite Bead; Amidon 73-801 |
| I1,3,5, | LED Indicator, diffused pastel red, T-1 package; Stanley MVR 3378S |
| I2,4,6 | LED Indicator, diffused pastel green, T-1 package; Stanley MPG 3878S |
| I7,9,11,12, 14,15,16 | LED Indicator, diffused pastel green, T-1¾ package; Stanley MPG 5774X |
| I8,10,13,17 | LED Indicator, diffused pastel red, T-1¾ package; Stanley MVR 5774X |
| IC1 | Integrated Cct.; (open mfgr.) LM337-T (Uses Aavid 574602 B03700 Heat Fin.) |
| IC2 | Integrated Cct.; (open mfgr.) LM317-T (Uses Aavid 574602 B03700 Heat Fin.) |
| IC3,5,7 | Integrated Cct.; (open mfgr.) CMOS 4013B |
| IC4,6 | Integrated Cct.; (open mfgr.) CMOS 4029B |
| IC8 | Integrated Cct.; (open mfgr.) CMOS 4011B |
| IC9,11 | Integrated Cct.; (open mfgr.) CMOS 4028B |
| IC12 | Integrated Cct.; (open mfgr.) CMOS 4081B |
| IC10 | Integrated Cct.; (open mfgr.) CMOS 4071B |
| IC13,14 | Integrated Cct.; (open mfgr.) CMOS 4051B |
| IC15,17-27, 30-39,41,44, 45,47-49,50 | Integrated Cct.; (open mfgr.) LF353N |
| IC16,40,46 | Integrated Cct.; (open mfgr.) CMOS 4066B |
| IC23,28,43 | Integrated Cct.; (open mfgr.) LM324N |
| IC29,42 | Integrated Cct.; National Semi. LM319C |
| J1 | AC Mains Connector, PC-mounting; Switchcraft EAC303 |
| J2 | 6-Position Barrier Block; PCD Co. ELFH06210 PC-mounting header with ELFP06210 plug-in screw-terminal block. |
| J501,502 | Connector, BNC chassis-mounting male; Amphenol 31-221 |
| JMP1-6 | Shorting "Shunt" for 0.1-inch header strip (open mfgr.) |
| L1 | Inductor, fixed, 560µH; ACT Co. PA2-561K |

| L2 | Inductor, fixed, 220μH; ACT Co. C4-221K |
|---|---|
| Q1 | Transistor, PNP; Motorola MJE350 |
| Q2 | Transistor, NPN; Motorola MJE340 |
| Q3,4,8,9,10,11 | Transistor, NPN; (open Mfgr.) 2N3904 |
| Q5,6,7 | Transistor, PNP; (open mfgr.) 2N3906 |
| Q12 | Transistor, NPN; (open mfgr.) 2N5088 |

Unless specifically noted by component reference designation below, **resistors** are specified as follows:

a) **Fixed resistors** with values carried to decimal places implying a 1% tolerance (*example:* 3.01K, 10.0K, 15.0K, 332K) are ¼-watt, 1% metal film type.

b) **Fixed resistors** with values typical of 5% tolerance (example: 220, 3.3k, 10K, 270K) are ¼-watt, 5% carbon film type.

c) **Multi-Turn Trimming Potentiometers** (front-panel adjustable) are Beckman 89PR series, Tokos RJC097P series, or equivalent "cermet" types.

d) **Single-Turn Trimmers** (circuit board) are Beckman 91AR series or Tokos GF06U1 series.

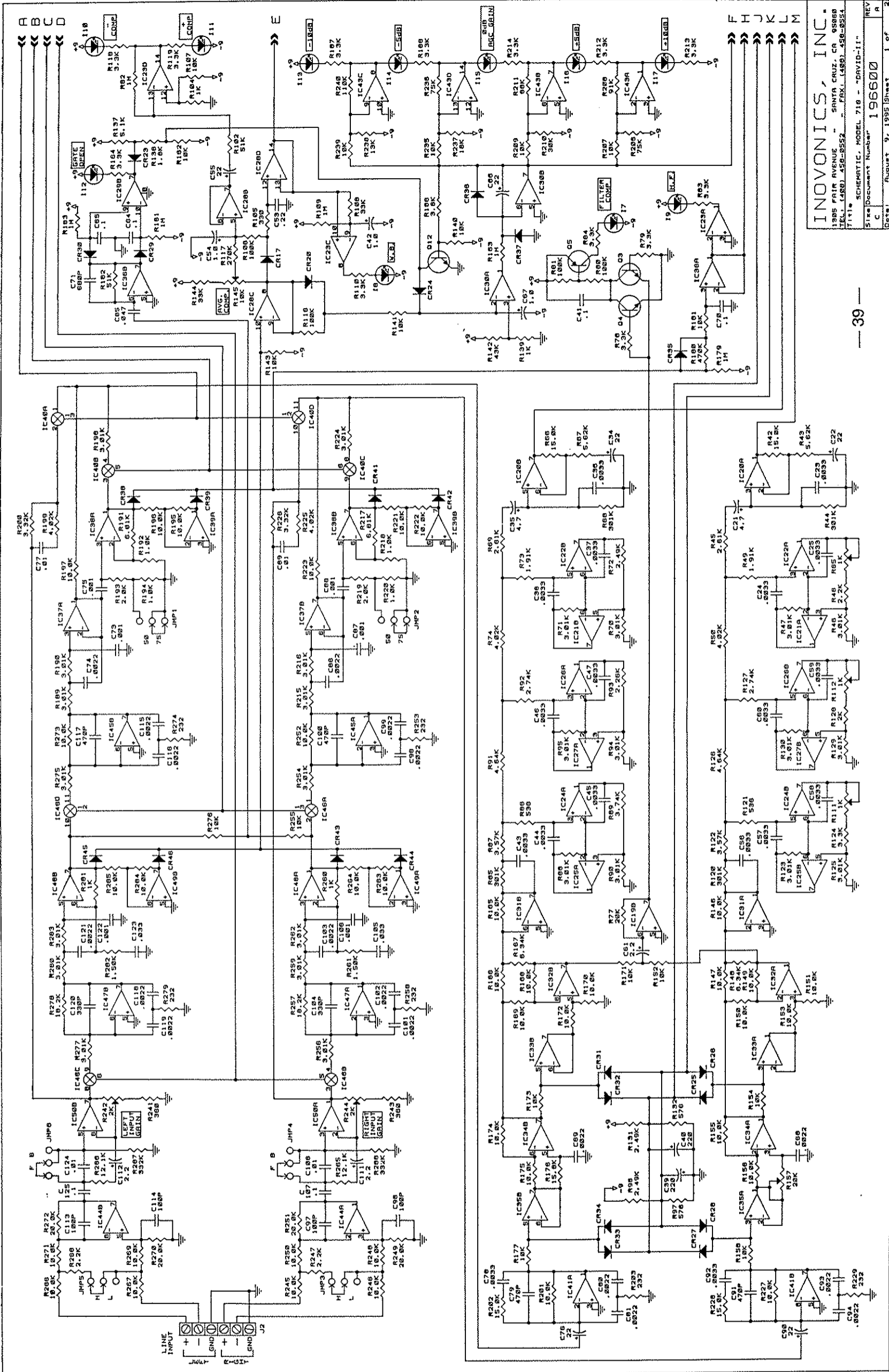| R64 | Resistor, Variable, 5K; Piher PT-15-YB-5K with Fig. 3 spindle. Uses Rogan 1R00A110BD001/PT-0-WL knob. |
|---|---|
| R145 | Resistor, Variable, 10K; Piher PT-15-YB-10K with Fig. 3 spindle. Uses Rogan 1R00A110BD001/PT-0-WL knob. |
| S1 | Switch, DPDT Slide, Voltage Selector; C&K V202-12-MS-02-QA |
| S2,3,4 | Switch, SPDT Miniature Toggle; C&K 7101-M-D9-A-B-E |
| S501 | Switch, Power Rocker; Carling RA 911-RB-O-N |
| T1 | Power Transformer, PC-mounting; Signal LP-20-600 or direct cross-reference |
| Y1 | Crystal 1.216MHz; *SPECIAL - Inovonics Part No. 1242* |

## MAIL-ORDER COMPONENT SUPPLIERS

The following electronic component distributors have proven to be reputable suppliers of both large and small quantities of parts. Any semiconductor, IC, capacitor, resistor or connector used in the *DAVID-II* is *probably* available from one or more of these firms. Each supplier publishes a full-line catalog available free for the asking.
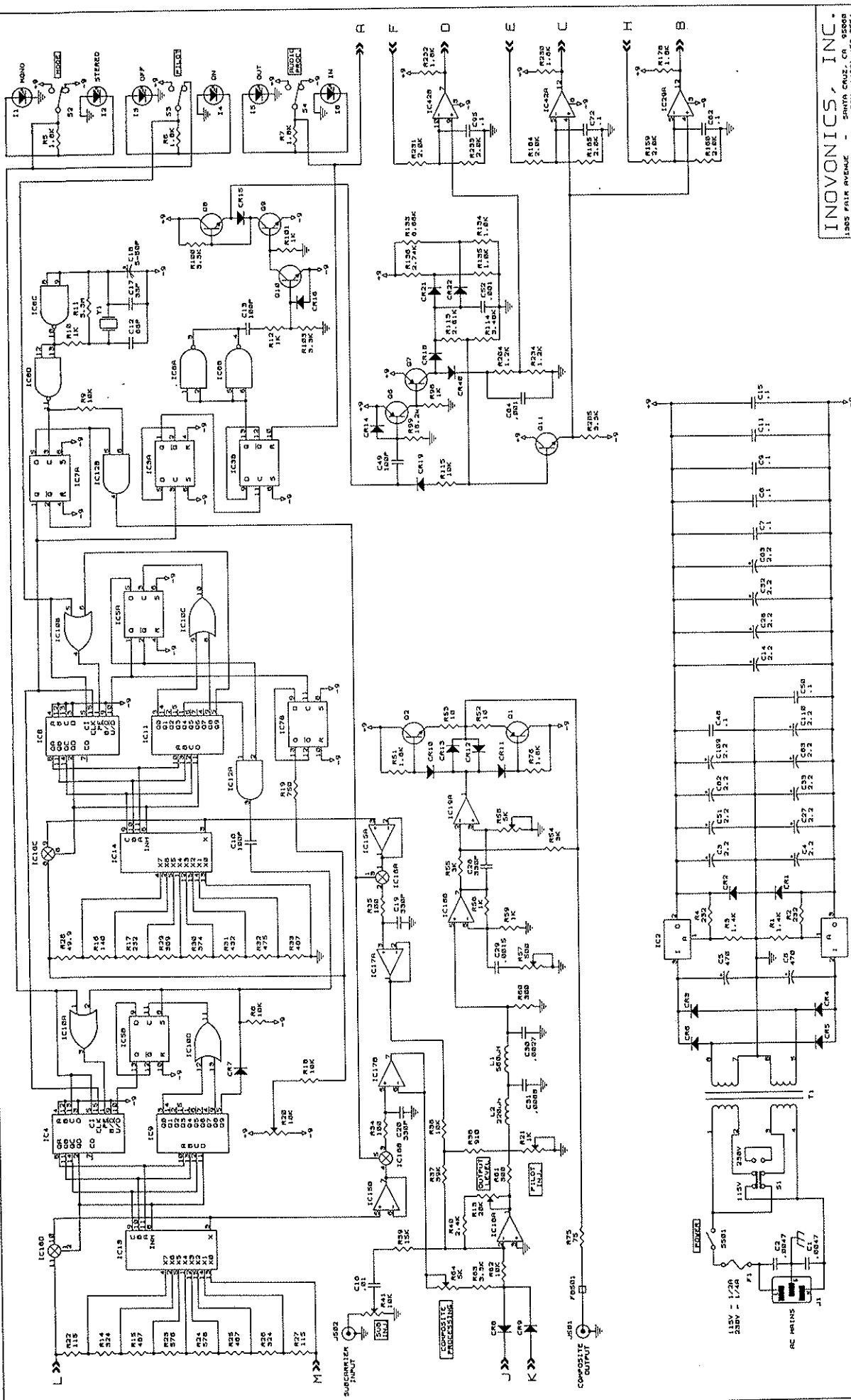
**Mouser Electronics** — Call (800) 346-6873

**Digi-Key Corporation** — Call (800) 344-4539

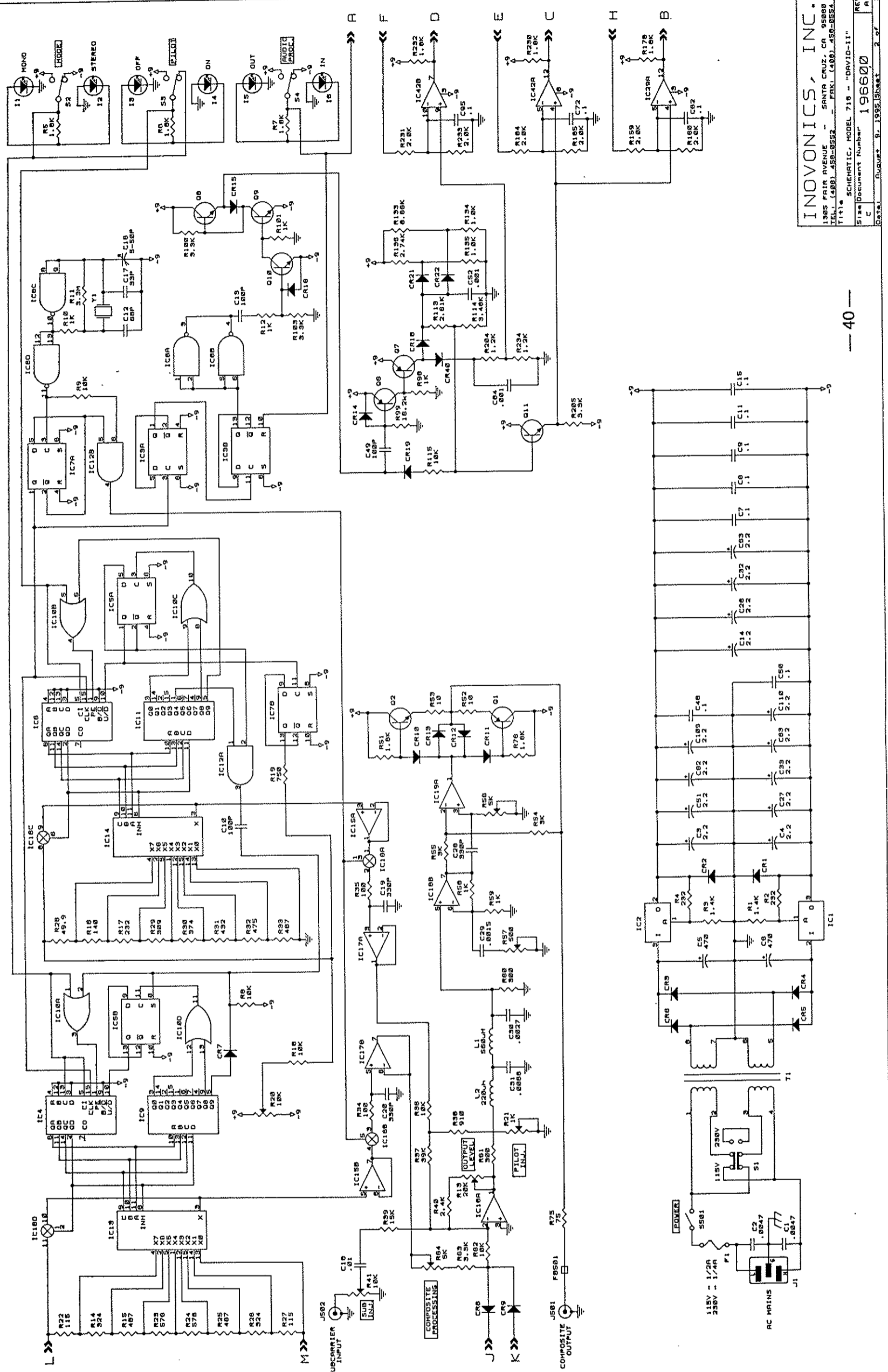**ACTIVE** (div. of Future Electronics) — Call (800) 677-8899
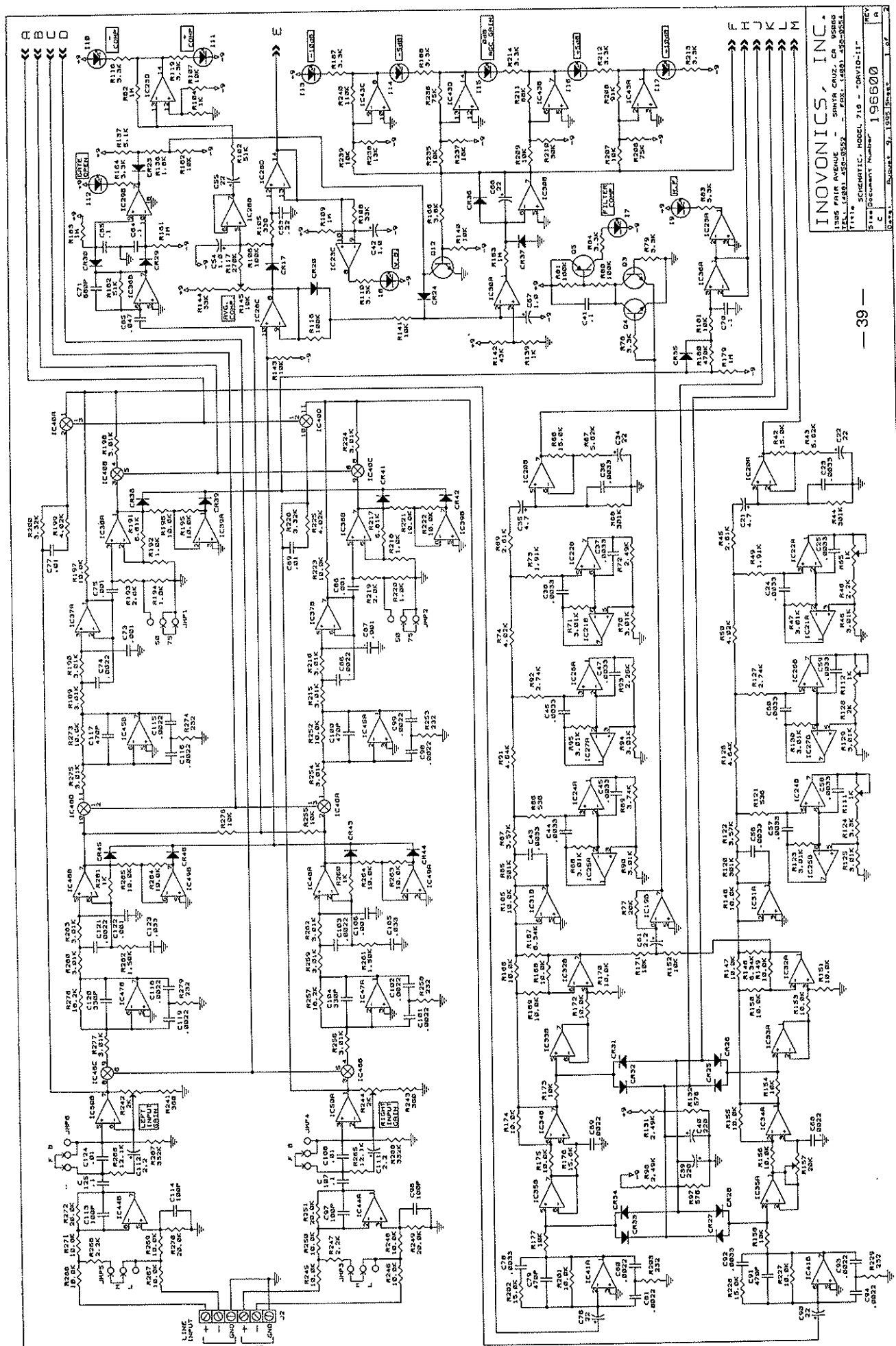
# INOVONICS, INC.

1305 FAIR AVENUE - SANTA CRUZ, CA 95060
TEL: (408) 458-0552   FAX (408) 458-0554

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Document Number: 196600

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Date: August 9, 1995

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LINE INPUT   J2

LEFT   RIGHT

LEFT INPUT GAIN

RIGHT INPUT GAIN

GATE OPEN

AVG. COMP

L COMP

R COMP

-10dB   -5dB   0dB DISC GAIN   -5dB   -10dB

FILTER COMP

H.F.

N.B.

INOVONICS, INC.

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TITLE: SCHEMATIC, MODEL 710 - "DAVID-II"

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MONO  STEREO  PILOT  OFF  ON  OUT  IN  AUDIO PROC.  MODE

SUBCARRIER INPUT

COMPOSITE PROCESSING

COMPOSITE OUTPUT

OUTPUT LEVEL

PILOT INJ.

SUB INJ.

POWER

AC MAINS

115V = 1/2A
230V = 1/4A

115V  230V

— 40 —