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ASSEMBLY & OPERATING INSTRUCTIONS (version 2.0)

for the

W9GR DSP-3 DIGITAL SIGNAL PROCESSOR

GREAT RECEPTION

The W9GR DSP-3 is an improved low cost digital signal processor for radio communications use. It features 18 DSP functions, a LED display, and 13 bit audio precision.

To use these instructions and build this kit, you should be capable of constructing a PC board from a parts list and schematic diagram, recognize electronic component parts, identify integrated circuit pin numbers, and solder. Potential kit builders lacking these skills are referred to **The Radio Amateur's Handbook**. This is not a kit for beginners!

The PC board is double sided with silk screen and solder mask. The "silk screen" is the set of white markings (such as "C14" and "R5") on the PC board which guides you in inserting the parts at the correct location. The components mount on the side with the silk screen.

The dark colored solder mask will help to avoid "solder bridges" between adjacent leads. It also provides insulation between the traces on the board and the exposed metal parts on the bottom of some of the IC sockets. Because of this, be careful not to scratch the solder mask; otherwise, there could be shorts or intermittent shorts between the IC sockets and the traces which run underneath the IC sockets.

The installation of parts progresses from the shortest parts, such as IC sockets and resistors, to the tallest parts such as the electrolytic capacitors and heatsink.

This double sided board has plated-through holes. At various places on the board, the traces on the top side need to connect to traces on the bottom side. A plated-through hole called a "via" performs this function. If you look at the board, you can see several examples of "vias" under IC1, the largest IC on the board. Notice that the vias have somewhat smaller tinned copper pads than the pads where the parts mount. When you install parts, do not install them in these "via" holes, which may be close to the hole where the part is supposed to go. Most of the parts have silk screened lines which will help you avoid making this error.

If you find you made a mistake and need to unsolder and remove a part, be careful so as to not remove the plating or the pads from the plated-through hole. Use a minimum amount of heat, and either a vacuum solder removal device or wire braid to remove the solder from the hole.

The electrolytic capacitors are polarized, and must be installed in a particular direction. There is a "+" mark on the PC board where the "+" or positive end of the capacitor must be installed. Sometimes capacitors have the "-" end marked instead of the "+" end. In this case, insert the unmarked end, which will be the "+" end, into the "+" hole on the PC board. As an additional check on polarity, the "+" lead on the polarized capacitors will usually be longer than the "-" lead.

Similarly, the two diodes have a circular band at one end; this marked end should match up with the markings on the silk screen.

The integrated circuits must also be inserted in the correct direction. Each IC will have either a notch or a "dimple" at one end. This end of the chip must match the silk screen markings on the board.

The integrated circuits used in this kit are susceptible to electrostatic damage. When handling the ICs, use standard procedures to avoid static damage. In brief, ground the circuit, the ICs, and most importantly yourself before handling static sensitive parts.

When you insert the ICs into their sockets, it is very important to check that the pins are all straight and that the IC is pushed into its socket straight. Otherwise, an IC pin can easily be bent under the chip as you insert it, and visual inspection will not reveal the problem!

If you need to remove a chip from its socket, insert a small screwdriver between the socket and the chip alternately at one end then the other, and back the chip out straight so as not to bend its pins.

Review the following parts list and identify all parts *before* soldering them into the board. Notice that one of the 20 pin IC sockets resembles a wire-wrap type, with its pins bent 90 degrees to accommodate the LED bargraph display.

Some parts may be marked with a number that indicates the value indirectly. According to this convention, the first two digits are followed by a number of zeros which is indicated by the third digit. "103" on a disc ceramic capacitor would indicate 10 followed by 3 zeros: 10000 pF or equivalently, 0.01 uF. The 22 pF capacitors may be marked "220J03." The "220" part of the marking means "22 followed by no zeros." So this means 22 pF, *not* 220 pF! The 22 pF capacitors may also be marked simply "22."

Parts List

Unless otherwise specified, all resistors are 1/4 watt 5% and capacitors are 10% tolerance. Some of the capacitors may be marked with a three digit number where the last digit represents the number of zeros to follow the first two digits. For example, "103" would translate to 10000 picofarads, which would be the same as 0.01 uF (0.01 microfarad). Sometimes these numbers are combined with other digits or letters.

Any of the 74XX series TTL parts (marked with a * below) may be supplied in any the following variations: 74XX, 74LSXX, 74SXX, 74FXX, 74ALSXX, 74HCXX, 74HCTXX, 74AHCTXX, 74HCTLSXX, etc. For IC5, a HCMOS type is recommended (e. g. HCT, HCTLS, etc.).

Quantity	Reference Designator(s)	Description
2	C13, C14	22 pF silver mica or disk ceramic capacitor (may be marked "22" or "220J03" or "22PJ3")
1	C7	0.001 uF ceramic 20% (may be marked "102" or "102Z")
2	C5, C8	0.01 uF ceramic 20% (may be marked "103" or "103Z")
12	C6, C10, C11, C15, C17, C18, C20, C21, C22, C23, C24, C25	0.1 uF monolithic ceramic, 20% (may be marked "104," "104M," ".1Z," ".1M," or "104Z")
3	C2, C3, C4	1.0 uF electrolytic (16V or greater)
2	C1, C9	10 uF electrolytic (16V or greater)
3	C12, C16, C19	220 uF electrolytic (16V or greater)
1	R14	2.7 ohms (red-purple-gold-gold)
1	R16	10 ohms (brown-black-black-gold)
1	R13	39 ohms (orange-white-black-gold)
11	R8, R21, R22, R23, R24, R25, R26, R27 R28, R29, R30	330 ohms (orange-orange-brown-gold)
1	R7	820 ohms (gray-red-brown-gold)
1	R15	1000 ohms (brown-black-red-gold)
1	R2	1800 ohms (brown-gray-red-gold)
2	R10, R12	3000 ohms (orange-black-red-gold)
2	R4, R5	6.2 K ohms (blue-red-red-gold)
6	R1, R3, R17, R18, R19, R20	10 K ohms (brown-black-orange-gold)
2	R6, R9	100 K ohms (brown-black-yellow-gold)
1	R11	100 K ohm pot
2	D1, D2	1N4001 silicon diode (diodes with higher voltage ratings such as 1N4002, 1N4003, 1N4004 etc. may be supplied)
1	X1	20 MHz fundamental crystal
1	IC1	TMS320P15 DSP CPU
1	IC2	TCM320AC39 audio codec (TCM320AC38 may be substituted)
2	IC3, IC4	74299 shift register (*)
1	IC5	7400 quad NAND gate (HCMOS type recommended) (*)
3	IC6, IC7, IC11	7474 dual flip flop (*)
1	IC8	74368 hex inverter (*)
1	IC9	74139 decoder (*)
1	IC10	74374 octal flip flop (*)
1	IC12	LM380 audio power amplifier
1	VR1	7805 or LM340T-5 TO-220 voltage regulator
2	L1, L2	10 uH RF chokes
1	LED1	10 segment LED bargraph, Radio Shack 276-081B or equivalent
1	J1	Coaxial DC power jack (2.1 mm)
2	J2, J3	1/8" audio jack
1	J4	1/8" stereo headphone jack
2	S1, S2	DPDT push-push switch
2		Switch caps for S1 & S2 (1 red; 1 gray)
1	S3	16 position binary rotary encoder switch
1	S4	SPDT toggle switch (for BIO function)
1		TO220 heatsink
5		14 pin IC socket
2		16 pin IC socket
4		20 pin IC socket
1		40 pin IC socket
1		20 pin right angle socket (for LED bargraph)
2		knob
1		PC board
1		set of #4 hardware for VR1 & heatsink

The W9GR DSP-3 is a digital signal processor which operates on audio signals. It connects between your receiver's audio output and your loudspeaker. It provides filtering functions for noise reduction, automatic notch filtering, FSK filters, a SSTV filter, narrow SSB filters, CW filters, a DTMF decoder, and a CTCSS decoder. A Product Review of this DSP appeared in the August 1995 issue of **QST** (pages 73-75). This DSP is also featured as a construction project in the 1996, 1997, and 1998 editions of the **ARRL Handbook**.

SOME CONSTRUCTION HINTS

❏ Choose a place to work where you can leave the kit sit for a while if necessary. The workplace should be uncluttered, and stray parts from other projects should not wander into the area.

❏ Use a 15-50 watt soldering pencil with a clean tip.

❏ Use only rosin core solder intended for electronic use.

❏ Use bright lighting. A magnifying lamp or bench-style magnifier may be helpful.

❏ Do your work in stages, taking frequent breaks to check your work.

❏ Carefully remove wire cuttings so they do not create shorts.

❏ For the best appearance and finesse, locate parts so that the component marking is visible when viewed from the direction that the silk screen marking is read. In the case of resistors, insert them so that the color bands all read in the same direction; left to right when viewed from the direction that the silk screen marking is read.

❏ One source of further information on kit building is the article *Kit Builder's Primer (Yeah, I built that)* by Mike Bryce, WB8VGE which appeared in the December 1994 issue of **73** magazine.

PC BOARD ASSEMBLY

❏ **1**. First install and solder each of the fixed resistors.

❏ **2**. There are three pads at jumper position JP1 which are used to configure the DSP for the right CPU chip. Take one of the clipped-off resistor leads and solder it between the center pad at jumper JP1 (MC/MP-) and the "1" pad. There is a silk screened line between the two pads. The other position is used only for special purposes - if you intend to provide your own firmware in the 57C43 EPROMs at IC13 and IC14, and your own TMS320C10 or TMS320C15 CPU chip at IC1 (The "0" position tells the TMS320C10/15 processor to look for firmware in the external EPROMs at IC13 and IC14. The "1" position tells the TMS320P15 processor to use its internal program.)

❏ **3**. If you do not want the headphone jack to mute speaker audio when you plug in a pair of headphones, then install a jumper wire at JP3. If you leave the jumper wire out, then when you plug in a pair of headphones, the speaker audio output at J3 will mute.

❏ **4**. Install and solder the two 1N4001 silicon diodes (D1 & D2). Be sure to match the polarity bands on the diodes to the ones on the silk screen.

❏ **5**. Install and solder the IC sockets. **Do not install sockets at IC13 and IC14. These ICs are only used for special applications.**. Be sure to align the notch at the notched end of the socket with the notch on the PC board silk screen. Sometimes sockets will have a beveled mark, a dot, a "1," or some other distinguishing characteristic at pin 1 instead of a notch. (Pin 1 is the lower left hand corner viewed from the top with the notch on the left; pin numbers increase counterclockwise from pin 1.) (Note: if you look at the board from the "front" end, where the LED bargraph is located, all the notched ends go to the left.)

❏ **6**. Install and solder the two RF chokes, L1 and L2.

❏ **7**. Install and solder the 0.001 uF ceramic disc capacitor at C7.

❏ **8**. Install and solder the 0.01 uF ceramic disc capacitors at C5 and C8.

❏ **9**. Install and solder the twelve 0.1 uF monolithic ceramic capacitors.

❏ **10**. Install and solder the two 22 pF capacitors at C13 and C14. These may be either silver mica or ceramic types.

❏ **11**. Install and solder the rest of the electrolytic capacitors, being careful to match the polarity markings on the board with those on the capacitors. The electrolytic capacitors are polarized, and must be installed in a particular direction. There is a "+" mark on the PC board where the "+" or positive end of the capacitor must be installed. Sometimes capacitors have the "-" end marked instead of the "+" end. In this case, insert the unmarked end, which will be the "+" end, into the "+" hole on the PC board. As an additional check on polarity, the "+" lead on the polarized capacitors will usually be longer than the "-" lead.

❏ **12**. Install and solder the 20 MHz crystal at X1. If you leave a little lead length on this component, it will not break off if inadvertently bent over. The crystal supplied may have a third wire soldered to the metal case. If it does, cut the wire off.

❏ **13**. In this step, the voltage regulator (VR1) is mounted together with a heatsink to the PC board, and then the leads are soldered to the board. Use a small amount of thermal grease (not supplied) between VR1 and the heat sink. (Thermal grease is not absolutely essential. If you do not have any, just make sure that the voltage regulator and heatsink are clean and free of debris so that the best thermal contact can be made without thermal grease.) If the heatsink has solder tabs which would interfere with the voltage regulator leads, cut them off. Use the #4 hardware supplied to mount VR1 and the heatsink to the board.

❏ **14**. In this step you will install the two audio connectors J2 and J3 at the rear of the PC board. But first you will need to bend the socket pins slightly. If you look at each connector you will see that there is a "kink" in each of the three pins. Using a needle nose pliers, squeeze each pin so that it becomes flat instead of kinked. There are two plastic pins that go through corresponding holes on the PC board. Insert the two plastic pins and the three de-kinked metal leads through the holes on the PC board. The plastic pins establish the socket's height above the PC board. Adjust the sockets so that they are parallel to the PC board surface before soldering them in place. This will avoid mechanical interference problems later when the board is installed in the enclosure.

❏ **15**. Install the coaxial DC power jack at J1 on the rear of the board.

❑ **16**. Install the DPDT push-push switches at S1 and S2 on the front of the board. Be sure to orient the actuators towards the edge of the board, and make sure that the switches are seated firmly on the board before soldering, to avoid mechanical problems later. Install the red switch cap on S1 (ON/OFF) and the gray switch cap on S2 (IN/OUT).

❑ **17**. A right angle 20 pin IC socket is supplied for the LED bargraph display. In most applications the PC board will mount horizontally and the LED bargraph will poke through an opening in your front panel. Make sure that the socket is firmly seated against the board before soldering, so that mechanical problems can be avoided later.

❑ **18**. Install the stereo headphone jack at J4. Let the plastic pins stay on the board surface and establish the socket's height above the PC board.

❑ **19**. Locate the 100 k ohm AF GAIN control. Look at the control with the shaft side facing you. If there is a metal locating tab beside the shaft, bend it outward so that it will not interfere with the front panel. Install the AF GAIN control at R11 on the front of the board.

❑ **20**. Locate the 16 position rotary encoder switch. Look at the encoder with the shaft side facing you. If there is a metal locating tab beside the shaft, break it off with a pair of pliers. Install the encoder at S3 on the front of the board. Make sure that the encoder is firmly seated against the board before soldering, so that mechanical problems can be avoided later.

❑ **21**. Using three short pieces of hookup wire (not supplied) (about 2 inches) connect the SPDT toggle switch S4 to the three pads at JP2. The middle pad goes to the middle terminal of the switch, the "1" pad goes to the bottom switch terminal, and the "0" pad goes to the top switch terminal. This way, when the switch is up, BIO will be set to a logic one.

❑ **22**. Do not insert the integrated circuits or the LED display into their sockets just yet! A "smoke test" will be performed to make sure that the supply voltage applied to the logic integrated circuits is not excessive. For the "smoke test" it is best to use a 12 volt power supply which is **current limited** to approximately 1 ampere. If you do not use a current limited power supply, there is the danger of burning traces off the board if there is a short. With **none** of the integrated circuits installed in their sockets (except for VR1, the soldered-in voltage regulator), apply power and turn on the switch S1. You can apply power either at the pad E2, or via the 2.1 mm coaxial DC power connector at J1. The center post is positive, and the outside shell is negative at J1. Using a voltmeter, connect its negative lead to ground. While viewing the board with the LED bargraph at the right side, at VR1 the leftmost pin should have approximately 12 volts present. (Depending on your power supply, the voltage may be as low as 8 volts or as high as 16 volts.) There should be zero volts at the middle pin. The rightmost pin must have 5 volts +/- 0.5 volts. If there is more than 5.5 volts at the rightmost pin of VR1, **do not install the socketed integrated circuits**. If you do, they may be destroyed and the warranty will be voided. Check the installation of components and correct the problem before proceeding. After this test is completed, disconnect the power.

❑ **23**. After you have verified that there is not excessive voltage on the nominally 5 volt output of the voltage regulator IC VR1, disconnect power and insert the integrated circuits into their sockets, being careful to insert them straight and with the proper orientation. On most ICs it is usually necessary to bend the leads *slightly* inward so that they will go into the socket straight. Bend the leads as a group against a flat surface. Your kit may be supplied with "LS" (low power Schottky) parts, "HCT" (high speed CMOS) parts, or some other compatible version. So for example, a 74LS139 may be installed at IC9 where the board is labeled "74139." Similarly, a 74HCTLS00 may be installed at IC5 where the board is labeled "7400."

❑ **24**. **IMPORTANT:** Your DSP-3 kit uses three dual D flip flop ICs of the 74xx74 family. You may find included with your DSP kit two 74HCT74 variants of this device, and one 74LS74 variant (bipolar). To make sure that the DSP works properly over temperature extremes, be sure to install the bipolar 74LS74 device at IC6, and install the two 74HCT74 devices at IC7 and IC11. If your kit is supplied with three 74LS74 chips, then simply install them at IC6, IC7, and IC11.

❑ **25**. Insert the LED bargraph into its socket. The markings, if any, are usually along the pin 1-10 edge, which goes down in this application. If the bargraph does not light up you can try reversing it; we have found a few bargraphs with the lettering along the pin 11-20 (top) edge.

INSTALLATION IN YOUR CIRCUIT & ENCLOSURE

If you purchased the custom made box for the DSP-3, it is installed by first removing the panel nuts and washers from the AF GAIN control (R11) and from the rotary encoder (S3). Insert four #4x1/2 screws through the holes in the cabinet bottom. Put a 1/4" aluminum spacer on each screw. Slide the front of the DSP board into the cabinet so that the LED bargraph, switches, headphone jack, and controls poke through the appropriate holes. Attach the board with four #4 lockwashers and nuts. S4, the "BIO" switch connected to the pads at JP2, installs on the back panel of the enclosure. Make sure that the switch does not interfere with JP1, the MC/MP- wire jumper, or C13. If your kit was supplied with silver mica capacitors, you may have to bend C13 out of the way. Put the control washers and nuts on the AF GAIN and mode select controls. *Do not over tighten the nut on the AF GAIN pot or it may bind.* Put a knob on both of these controls. Do not push the knob all the way on to the AF gain pot or it may bind.

The lid is attached to the bottom with four #6 sheet metal screws. If your enclosure is supplied with screws that are 3/8" long, you will also find four #6 flat washers and finishing washers. The sheet metal screws should each go through a finishing washer, then a flat washer. This will stop the screw from protruding too far into the cabinet and damaging the PC board.

If you did not purchase the custom made box, install the PC board in your enclosure. (If there is room to spare, you may want to install it in your speaker cabinet.) Any of the switches, jacks, LED bargraph display and volume control may be removed from the PC board and installed on your front or rear panels. If you remove parts, make sure that the wires you use to attach the components make the right connections.

In case you choose not to use the DC and audio connectors supplied, there is a second set of I/O (input/output) connections next to the heatsink at VR1. The four PC board pad connections are GND, +12 volts, audio in, and audio out.

High speed CMOS, as used in this project, produces EMI in prodigious quantities. As with most any project designed for use in a ham environment, it is highly recommended that you shield the PC board by installing it in a metal cabinet and bypass signals going into and out of the cabinet. Some of this EMI bypassing is already provided on the PC board itself. If you do not shield the digital signal processor, at least its autonotcher firmware will help clean up some of its own radiated birdies which you might tune across!

TESTING THE DSP

To "bring up" the DSP, you will need the following:

1. A source of audio, which may be taken from your receiver.
2. A loudspeaker.
3. A 12 volt DC power supply. One recommended power supply is the Radio Shack 273-1653A (12VDC, 1A).

Do not apply power yet. First, make sure that the on/off switch S1 is off, which is the "out" position. Apply the audio from your receiver or any other source which is capable of driving a speaker to the 1/8" input connector at the rear labeled J2 and "AF IN." Alternatively, you may apply the audio at pad E3. Similarly, connect your speaker to the 1/8" rear output connector at J3 labeled "AF OUT." You should now be hearing "straight through" audio passing through the turned-off DSP. Next, set the mode switch S3 to the 12 o'clock position (knob pointer straight up or flat on shaft straight down). Next, apply DC power. If you use the coaxial DC power connector at J1, the center pin is positive. Turn the power on by pushing S1 in. When you first turn the power on, you should see a pattern of LEDs light up and quickly sweep back and forth across the LED bargraph. Then you should be seeing some LEDs light up in response to the input audio. Try increasing and decreasing the audio to see how the bargraph display responds. If you push S2 in, you will be listening to the digitally processed audio. With S2 out, the DSP is bypassed and the signal only passes through the AF gain control and analog audio amplifier stage. Finally, with the in/out switch S2 in, rotate the mode switch through its various modes, and you will be able to hear the effect of the various DSP modes.

OPERATION

The DSP will normally connect between your receiver and your loudspeaker. The DSP requires a source of 12 volt DC power to operate. There are three connections to be made:

1. Audio in - from your receiver
2. Audio out - to your speaker
3. DC power - a source of reasonably clean 12 volt DC power. One recommended power supply is the Radio Shack 273-1653A (12VDC, 1A).

The front panel controls, from left to right, are:

❑ **Mode switch (S3)**. This 16 position rotary switch selects the DSP's operating mode.

❑ **On/off switch (S1)**. This switch turns the power on and off. When this switch is pushed in, the power is on; when the switch is out, the power is off. When the DSP is turned off, the audio input is connected to the audio output. This will bypass the DSP when its power is off.

❑ **In/out switch (S2)**. This switch enables or bypasses the digital signal processing function. When this switch is pushed in, the output signal has been digitally processed; when the switch is out, the digital processing is bypassed and the signal only passes through the analog speaker amplifier. This switch does not affect the LED display. In other words, when the switch is out, bypassing the DSP, the LED display will continue to indicate audio level or decoded tones (depending on the setting of the mode switch S3).

❑ **AF GAIN control (R11)**. This controls the audio output level from the DSP.

The DSP includes a 10 segment LED bargraph display. In most of the operating modes, the LED bargraph displays audio level. The audio level display is a peak reading type with a delayed recovery characteristic. Each LED represents a 3 dB change in audio level. The maximum input level without clipping is 5 volts peak to peak, which is about 1.8 volts rms.

Although the hardware and software have been designed to accept and process audio across a wide range of input levels, it is best to operate the DSP so that the audio level makes most of the LEDs on the bargraph display illuminate on peaks. The recommended method of operation is to set the receiver audio gain so that on the strongest signals, most or all of the LEDs occasionally come on. Once the receiver gain is properly adjusted, do not touch it; instead use the AF gain control on the processor to adjust speaker volume. This procedure will keep the A/D and D/A precision highest, and the quantizing noise lowest. Most modern receivers have good AGC characteristics with a flat AGC slope, the result being consistent audio level for both strong and weak signals. If your receiver's audio level is inconsistent, you may experience occasional overloads of the digital signal processor, producing clipping distortion. If this happens, simply turn down the audio input a tad.

The "BIO" switch (S4) selects DSP configuration options and activates special functions. It may be thought of as a "mode" switch. "BIO" stands for "binary I/O."

The DSP functions included in the DSP-3 are arranged with function #1 being at the "12 noon" position of mode selector switch S3. From the straight up #1 position, the functions are numbered clockwise. The 16 positions and the 18 DSP functions are as follows:

❑ **1. Combined automatic notch and denoiser**. This mode simultaneously automatically notches carriers and reduces noise. This mode is recommended for most HF SSB operation. When the BIO switch is set to "1" in this mode, audio AGC is enabled, and audio level will be digitally brought up to full level if the input level lights at least one LED of the bargraph display.

These functions both use the Widrow-Hoff LMS adaptive filtering algorithm. The noise reducer mode is most effective against hiss and thermal noise but also reduces impulse noise and static crashes. Noise reduction reduces listener fatigue and is recommended for long-term monitoring. The automatic notch function eliminates multiple carriers very quickly, within a few milliseconds. Tuner-uppers, CW interference, carriers, and other forms of undesired audio tones are quickly eliminated. If a carrier comes on your frequency, all you will hear will be a subtle "click" as the automatic notch acquires.

❑ **2. Denoiser**. This mode simultaneously reduces noise with somewhat greater effectiveness than the denoiser included in mode #1. This mode is recommended for weak voice signals, including HF SSB, VHF SSB, and FM. When the BIO switch is set to "1" in this mode, audio AGC is enabled, and audio level will be digitally brought up to full level if the input level lights at least one LED of the bargraph display.

❑ **3. Automatic notch**. This mode automatically notches carriers with somewhat greater effectiveness than the automatic notch included in mode #1. This mode is recommended for HF SSB operation where interfering carriers are the most objectionable problem. When the BIO switch is set to "1" in this mode, audio AGC is enabled, and audio level will be digitally brought up to full level if the input level lights at least one LED of the bargraph display.

❑ **4. 2.1 kHz narrow voice FIR filter**. This mode is a fixed "brick wall" 100th order narrowband voice FIR filter, intended for rejecting adjacent overlapping SSB signals. There is no adaptive noise reduction or automatic notching included in this mode. The theoretical filter specifications are as follows:

passband: 300-2100 Hz +/- 0.3 dB ripple
<150 Hz attenuated >50 dB; >2300 Hz attenuated >70 dB

❑ **5. 1.8 kHz narrow voice FIR filter**. This mode is a fixed "brick wall" 100th order narrowband voice FIR filter, intended for rejecting adjacent overlapping SSB signals. There is no adaptive noise reduction or automatic notching included in this mode. The theoretical filter specifications are as follows:

passband: 300-1800 Hz +/- 0.3 dB ripple
<150 Hz attenuated >50 dB; >2100 Hz attenuated >70 dB

❑ **6. RTTY filter**. This is a 100th order linear phase bandpass filter. This mode is intended for 170 Hz shift frequency shift keying (FSK) signals such as Baudot, AMTOR, etc. When the BIO switch is in the "0" position, North American RTTY tones are selected (2125/2295 Hz). When the BIO switch is in the "1" position, European RTTY tones are selected (1275/1445 Hz). Theoretical specifications are as follows:

BIO=0 (North American RTTY tones)
passband 2075-2345 Hz +/- 0.3 dB ripple
<1875 Hz attenuated >60 dB; >2545 Hz attenuated >60 dB

BIO=1 (European RTTY tones)
passband 1225-1495 Hz +/- 0.3 dB ripple
<1025 Hz attenuated >60 dB; >1695 Hz attenuated >60 dB

After changing the position of the BIO switch to change between high and low tones, this mode must be reselected with the mode switch S3 before the new mode option takes effect.

❑ **7. HF packet filter or SSTV filter**. This is a 100th order linear phase bandpass filter. This mode is intended for 200 Hz shift 1200 baud HF packet FSK signals or SSTV signals. When the BIO switch is in the "0" position, the HF packet mode is selected (1600/1800 Hz). When the BIO switch is in the "1" position, this mode becomes a SSTV filter (1200-2300 Hz). Theoretical specifications are as follows:

BIO=0 (HF packet)
passband 1550-1850 Hz +/- 0.4 dB ripple
<1350 Hz attenuated >65 dB; >2050 Hz attenuated >65 dB

BIO=1 (SSTV)
passband 1075-2350 Hz +/- 0.3 dB ripple
<875 Hz attenuated >70 dB; >2550 Hz attenuated >60 dB

After changing the position of the BIO switch to switch between HF packet and SSTV, this mode must be reselected with the mode switch S3 before the new mode option takes effect.

❑ **8. DTMF Decoder**. This mode will decode "dual tone multi frequency (DTMF; also known as "Touch-Tone") signals, using the LED bargraph to indicate the decoded tone pairs. You can use this mode to test DTMF encoders, troubleshoot autopatches, monitor patch access, etc. The LED bargraph does not indicate audio level in this mode because it must be used to indicate decoded DTMF tones. To use this mode, first use one of the other modes to set the input level so that at least three LEDs light up when a DTMF tone pair is received. When you first select this mode, a unique pattern of two illuminated LEDs at each end indicates that no valid DTMF tones have yet been received. When a valid DTMF tone pair is received, the corresponding LED pattern will illuminate and stay illuminated until another valid DTMF tone pair is received. The LED patterns are as follows:

DTMF signal	LED pattern
Unknown	❑❑■■■■■■❑❑
1	❑■■■■■■■■■
2	■❑■■■■■■■■
3	■■❑■■■■■■■
4	■■■❑■■■■■■
5	■■■■❑■■■■■
6	■■■■■❑■■■■
7	■■■■■■❑■■■
8	■■■■■■■❑■■
9	■■■■■■■■❑■
0	■■■■■■■■■❑
*	❑❑❑❑❑■■■■■
#	■■■■■❑❑❑❑❑
A	❑❑■■■■■■■■
B	■■❑❑■■■■■■
C	■■■■■■❑❑■■
D	■■■■■■■■❑❑

❑ = LED illuminated
■ = LED not illuminated

NOTE: if you set the IN/OUT switch to the IN position, you may notice a chopping noise in the DTMF mode. This is a normal artifact of the DSP's DFT algorithm which is used to search for DTMF tones. If you want to listen to the received audio in this mode, simply set the IN/OUT switch to OUT.

The DTMF decoder may occasionally "trip" on voice audio. This is normal and due to the momentary presence of audio energy at the DTMF tone frequencies.

You can also use the DTMF mode to "play back" DTMF sequences. The DTMF mode has a 16 tone memory. To play back the last 16 tones heard, simply toggle the BIO switch (in other words, change the switch's position, either from "1" to "0" or "0" to "1"). The previous tones (up to 16 maximum) will be played back in the order that they were received. This form of memory is called "first-in first-out" (FIFO). The LED patterns will light up one by one on the LED display at the rate of about one per second. The display will blank between patterns to make it easy to discern repeated digits. NOTE: if you switch the DSP out of the DTMF mode, the DTMF playback memory will be erased.

If a DTMF sequence is received and you want to make sure it is not overwritten in playback memory, simply disconnect the audio input to the DSP or turn off the receiver feeding the DSP. Otherwise, any new DTMF tones received will be added to the memory, and the sequence you want to review may be pushed off the end of the FIFO memory.

❑ **9. CTCSS Decoder & Squelch**. This mode is useful for decoding subaudible frequency "continuous tone-coded squelch system" (CTCSS) tones. This mode will decode any of the 38 subaudible tones (also known as "PL" tones) used for repeater access, tone squelch, etc. This mode is useful when testing your own CTCSS transmissions, for determining the CTCSS tone frequency by monitoring stations on a repeater’s input frequency, etc.

IMPORTANT NOTE: this mode may not work if your receiver cuts off the low frequency range below 300 Hz! If your receiver removes the tones, then the DSP cannot detect them!

You can often get around this problem by (a) increasing the coupling capacitor sizes within the audio section of your receiver, or (b) driving the DSP from the "discriminator output" of the receiver. In any event Quantics will not be liable for any damage done incidental to making modifications to any receiver for this purpose.

Another way to ensure the integrity of the subaudible tones is to use a receiver which includes a "wide FM" mode such as the ICOM R7000. Often the "narrow FM" mode will roll off low audio frequencies, and the "wide FM" mode will not.

Intermodulation distortion can create low frequency signal components which can be falsely interpreted as CTCSS. If the signal you are monitoring has too much deviation for the receiver you are using, then distortion will result. Distortion can also be caused by overdeviation, off-frequency transmitters and/or receivers, speech clipping, etc.

Proper operation of the CTCSS mode depends on having a low distortion signal path with adequate low frequency response. Remember the adage: "garbage in, garbage out!"

The LED bargraph does not indicate audio level in this mode because it must be used to indicate decoded CTCSS tones. To use this mode, first use one of the other modes to set the input level so that at least three LEDs light up on signal peaks. A pattern of one illuminated LED at each end indicates that no valid CTCSS tone is being received. When a valid CTCSS tone is received, the corresponding LED pattern will illuminate and stay illuminated only as long as the CTCSS tone is present. The LED patterns are as follows:

CTCSS signal	LED pattern
NO CTCSS	❑■■■■■■■■❑
67.0	❑❑■■■■■■■■
71.9	■❑❑■■■■■■■
74.4	■■❑❑■■■■■■
77.0	■■■❑❑■■■■■
79.7	■■■■❑❑■■■■
82.5	■■■■■❑❑■■■
85.4	■■■■■■❑❑■■
88.5	■■■■■■■❑❑■
91.5	■■■■■■■■❑❑
94.8	❑❑❑■■■■■■■
97.4	■❑❑❑■■■■■■
100.0	■■❑❑❑■■■■■
103.5	■■■❑❑❑■■■■
107.2	■■■■❑❑❑■■■
110.9	■■■■■❑❑❑■■
114.8	■■■■■■❑❑❑■
118.8	■■■■■■■❑❑❑
123.0	❑❑❑❑■■■■■■
127.3	■❑❑❑❑■■■■■
131.8	■■❑❑❑❑■■■■
136.5	■■■❑❑❑❑■■■
141.3	■■■■❑❑❑❑■■
146.2	■■■■■❑❑❑❑■
151.4	■■■■■■❑❑❑❑
156.7	❑❑❑❑❑■■■■■
162.2	■❑❑❑❑❑■■■■
167.9	■■❑❑❑❑❑■■■
173.8	■■■❑❑❑❑❑■■
179.9	■■■■❑❑❑❑❑■
186.2	■■■■■❑❑❑❑❑
192.8	❑❑❑❑❑❑■■■■
203.5	■❑❑❑❑❑❑■■■
210.7	■■❑❑❑❑❑❑■■
218.1	■■■❑❑❑❑❑❑■
225.7	■■■■❑❑❑❑❑❑
233.6	❑❑❑❑❑❑❑■■■
241.8	■❑❑❑❑❑❑❑■■
250.3	■■❑❑❑❑❑❑❑■

❑ = LED illuminated
■ = LED not illuminated

When the IN/OUT switch is set to OUT, the DSP passes the input audio to its output. But when the IN/OUT switch is set to IN, this mode can be used as a simple tone squelch. When the switch is set to IN, then the audio will be muted unless any CTCSS tone is detected. In other words, this mode will pass the audio if **any** of the 38 CTCSS tones is received.

You can also use the CTCSS mode to "play back" detected CTCSS tones. The CTCSS mode has a 16 tone memory. The CTCSS memory will only record a new tone if it is different from the previous detected tone. To play back the last 16 different tones heard, simply toggle the BIO switch (in other words, change the switch’s position, either from "1" to "0" or "0" to "1"). The previous tones (up to 16 maximum) will be played back in the inverse order that they were received (this is the opposite of the DTMF playback order). In other words, the last tone received will be the first one displayed. This form of memory is called "last-in first-out" (LIFO). The LED patterns will light up one by one on the LED display at the rate of about one per second. NOTE: if you switch the DSP out of the CTCSS mode, the CTCSS playback memory will be erased.

If a series of CTCSS tones is received and you want to make sure they is not overwritten in playback memory, simply disconnect the audio input to the DSP or turn off the receiver feeding the DSP. Otherwise, any new CTCSS tones received will be added to the memory, and the sequence you want to review may be pushed off the end of the LIFO memory.

❑ **10. 400 Hz CW filter, 100 Hz bandwidth**. This is a fixed linear phase FIR CW filter of 229th order. Theoretical specifications are as follows:

passband 350-450 Hz @ -3 dB
<270 Hz attenuated >60 dB; >530 Hz attenuated >60 dB

A short 400 Hz "dit" is sent when this filter is selected, to indicate the center of the passband. This filter may be tuned anywhere from 400 Hz down to 280 Hz.

❑ **11. 400 Hz CW filter, 50 Hz bandwidth**. This is a fixed linear phase FIR CW filter of 229th order. Theoretical specifications are as follows:

passband 375-425 Hz @ -3 dB
<320 Hz attenuated >50 dB; >480 Hz attenuated >50 dB

A short 400 Hz "dit" is sent when this filter is selected, to indicate the center of the passband. This filter may be tuned anywhere from 400 Hz down to 280 Hz.

❑ **12. 600 Hz CW filter, 100 Hz bandwidth**. This is a fixed linear phase FIR CW filter of 229th order. Theoretical specifications are as follows:

passband 550-650 Hz @ -3 dB
<470 Hz attenuated >60 dB; >730 Hz attenuated >60 dB

A short 600 Hz "dit" is sent when this filter is selected, to indicate the center of the passband. This filter may be tuned anywhere from 600 Hz down to 420 Hz.

❑ **13. 750 Hz CW filter, 200 Hz bandwidth**. This is a fixed linear phase FIR CW filter of 229th order. Theoretical specifications are as follows:

passband 650-850 Hz @ -3 dB
<575 Hz attenuated >60 dB; >925 Hz attenuated >60 dB

A short 750 Hz "dit" is sent when this filter is selected, to indicate the center of the passband. This filter may be tuned anywhere from 750 Hz down to 525 Hz.

❑ **14. 750 Hz CW filter, 100 Hz bandwidth**. This is a fixed linear phase FIR CW filter of 229th order. Theoretical specifications are as follows:

passband 700-800 Hz @ -3 dB
<620 Hz attenuated >60 dB; >880 Hz attenuated >60 dB

A short 750 Hz "dit" is sent when this filter is selected, to indicate the center of the passband. This filter may be tuned anywhere from 750 Hz down to 525 Hz.

❑ **15. 750 Hz CW filter, 50 Hz bandwidth**. This is a fixed linear phase FIR CW filter of 229th order. Theoretical specifications are as follows:

passband 725-775 Hz @ -3 dB
<670 Hz attenuated >50 dB; >830 Hz attenuated >50 dB

A short 750 Hz "dit" is sent when this filter is selected, to indicate the center of the passband. This filter may be tuned anywhere from 750 Hz down to 525 Hz.

❑ **16. 1000 Hz CW filter, 100 Hz bandwidth**. This is a fixed linear phase FIR CW filter of 229th order. Theoretical specifications are as follows:

passband 950-1050 Hz @ -3 dB
<870 Hz attenuated >60 dB; >1130 Hz attenuated >60 dB

A short 1000 Hz "dit" is sent when this filter is selected, to indicate the center of the passband. This filter may be tuned anywhere from 1000 Hz down to 700 Hz.

❑ **Tuning the CW filters**. To tune any of the CW filters, perform the following steps:

1. Turn the power off.
2. Set the BIO switch S4 or jumper wire connected to the pads at JP2 to "1."
3. Set the mode selector switch S3 to the CW filter position you want to tune.
4. Turn on the power. The DSP will produce a slowly declining audio tone at the center of the CW filter bandpass. When the audio tone reaches the frequency you want, turn the mode selector switch S3 to any other position. The filter is now tuned to the note that you heard when you turned the mode selector switch. If you do not switch the mode selector during the declining tone, the CW filter tuning will terminate at a limit of -30% detuning.

After this process, all of the CW filters will be tuned down by the same percentage. So if you tuned the 1000 Hz filter to 900 Hz, for example (-10%), then the 750 Hz filters will be at 675 Hz, the 600 Hz filter will be at 540 Hz, etc. This condition will remain until you turn the power off. Even if you select one of the voice modes and then return to the CW filters, the filters will remain tuned to the same frequency unless the power is turned off. When you select one of the CW filters after the tuning process, the "dit" sent will be at the new center frequency of the filter.

This process does not affect or tune the other filters (RTTY, SSTV, narrow SSB, etc.).

❑ **BIO switch (S4) function summary**.

Modes 1,2,3:	BIO=1: digital AGC enabled
	BIO=0: digital AGC disabled
Modes 4,5:	BIO has no effect
Mode 6:	BIO=1 selects European (low) RTTY tones
	BIO=0 selects North American (high) RTTY tones
Mode 7:	BIO=1 selects SSTV filter
	BIO=0 selects HF packet filter
Modes 8:	BIO toggling initiates DTMF tone playback
Modes 9:	BIO toggling initiates CTCSS tone playback
Modes 10-16	BIO=1 enables CW filter tuning
	BIO=0 disables CW filter tuning

Note: in modes 6 and 7, the BIO switch does not have an immediate effect. You must re-select the desired mode *after* changing the BIO switch. If, for example, you are using the RTTY filter and you want to switch from high tones to low tones, you must re-select the RTTY mode after changing the BIO switch position. This is because the firmware only reads the BIO switch when mode 6 or 7 is first selected.

In modes 1, 2, and 3, the BIO switch *does* have an immediate effect (selecting or de-selecting digital audio AGC).

When the DSP is powered up with the BIO switch in the "0" position, a quick LED test sweeps back and forth across the display four times (two "round trips") with one LED at a time illuminated. When the BIO switch is in the "1" position at power up, the same LED test is performed, followed by another two "round trips" of LED sweeps where multiple LEDs light up. This enables you to tell the position of the BIO switch, which may be located on the rear panel, without having to look at it.

❑ **Digital AGC and Reading the LED bargraph audio level display**. In most of the DSP's modes, the LED bargraph display indicates audio level. In these modes, each LED corresponds to a 3 dB change in input level. Since there are 10 LEDs, the range displayed is 30 dB. The "meter" is peak reading.

When the digital AGC function is selected (BIO=1 in modes 1, 2, and 3), the AGC range is limited to 30 dB so that noise, hum, etc. are not brought up to full output amplitude in the absence of a signal. So if you use AGC in modes 1, 2, or 3, as long as at least one LED comes on, the audio output level will be brought up to full amplitude.

The AGC function is entirely DSP based, and it exists in software. In other words, there is not a separate analog AGC chip in the DSP hardware. The recovery time constant is very slow, taking about 30 seconds to adapt to a new (lower) signal level. The time constant was made deliberately slow to avoid "pumping" and "breathing" artifacts. Attack time is fast, virtually instantaneous. If the input level suddenly drops, and you do not want to wait up to 30 seconds for the AGC to adapt, you can make it re-initialize by turning the mode switch S3 away from the mode you are using, then moving it back. Then the AGC will instantly re-initialize to the new signal level, and the output of the DSP will be at full amplitude.

❑ **The LMS "denoiser" algorithm and different types of noise**. The LMS "denoiser" algorithm discriminates for or against signals depending on their degree of autocorrelation or repetitiveness. Thermal or atmospheric noise generally has no repetitiveness whatsoever. An interfering carrier, on the other hand, has each cycle identical to the next, and therefore has a high degree of repetitiveness. Speech lies somewhere between these two extremes.

The denoiser works by automatically forming bandpass filters around the most significant spectral lines in the signal, allowing the repetitive components of voice signals to pass, while greatly reducing random signals like thermal or atmospheric noise.

The automatic notch works by rejecting those parts of the signal which are highly repetitive. The adaptive filter automatically forms narrow bandpass filters around the undesired audio tones, which are then subtracted from the signal.

If the signal you are trying to listen to is occasionally buried by strong impulse noise or static crash QRN, the denoiser will not work any miracles. But if you have background QRN, such as essentially "white" thermal noise, it will help a lot. It is most effective with noise that is more or less constant (like hiss) rather than spiky noise (the kind conventional noise blankers take out). But whereas a noise blanker works best if the noise is very strong, the LMS algorithm works best if the noise is moderate to weak. If the noise you are experiencing comes in bursts of an amplitude such that during the noise bursts the instantaneous signal to noise ratio is still at least a small positive number (6 dB or preferably more) then the LMS algorithm DSP filter will probably help. But if the impulse noise is strong, then a conventional noise blanker would work better, because it has access to the wideband RF signal prior to IF filtering.

When people say "noise" or QRN it can mean a lot of things. If the noise consists of strong impulses related to 50/60 Hz power line frequencies or to an automotive ignition rate, then the DSP will not do much, but a conventional noise blanker will. On the other hand, if the noise is closer to white noise or maybe a background "frying" noise, as opposed to a "buzz," then the DSP will provide some relief.

❑ **A few notes on the LMS automatic notch algorithm**. The LMS automatic notch algorithm forms extremely narrow notch filters at frequencies where it finds continuous audio tones. In order to stop the automatic notch from attacking voice signals, it is necessary to make it sensitive only to pure audio tones, and to make the notch quite narrow. This means that if the tone has any sort of "wobble" or frequency modulation on it, it will not receive as much attenuation. If your receiver has AC hum on its local oscillator or beat frequency oscillator (BFO) which makes any FM, you may notice that the effectiveness of the notch is reduced. Similarly, particularly on 40 meters, where a tone may be due to several co-channel foreign broadcast stations' carriers, the resulting complex phase modulation may produce enough "wobble" in the audio tone that the effectiveness of the notch filter is reduced.

❑ **Bandwidth limiting in the DSP modes**. When the DSP is switched in, you may notice a reduced audio bandwidth. This should not be noticeable when you are driving the DSP with SSB type audio, which normally has about a 3 kHz bandwidth. But if you are driving the DSP with a wideband audio source, such as a broadcast FM radio, you will hear the bandwidth reduction. This is normal. U2, the audio codec chip, includes switched capacitor lowpass filters for antialiasing and reconstruction. The filters in U2 cut off at about 3.2 kHz. These filters ensure that the Nyquist criterion is not exceeded, which requires that the highest audio frequency must be less than half of the sampling rate. The sampling rate in this DSP is approximately 7700 Hz.

❑ **A few words about the CW filters**. Although some amateurs use the noise reduction mode for CW, the dedicated CW filters are better in most applications. The noise reduction modes work by automatically forming little bandpass filters around the most significant spectral lines in the signal, and on CW, the bandpass filters must recreate themselves with each dit and dah. But in the case of CW, we have *a priori* knowledge of the desired signal's spectrum, so we can design a fixed filter which is already optimal and which does not need to adapt itself. Furthermore, the fixed CW filters are linear phase (no time delay distortion) but the adaptive filters formed by the LMS algorithm will not in general be linear phase. Where it would make sense to use the noise reduction mode for CW would be in the case where you are tuning the band and want a wide (SSB type) bandwidth to hear more signals. But when you want to hear only a single signal, it is probably best to use the dedicated CW filters.

On the subject of "ringing" there is a lot of misinformation. All bandpass filters "ring" - they must do so in order to work! But there are two causes of ringing. The first cause of ringing is the fact that the bandwidth has been reduced. All bandpass filters ring due to this mechanism. The second cause of ringing is nonlinear phase, or phase distortion. Conventional filters incur this additional ringing due to nonlinear phase. A FIR filter is linear phase, and so it does not ring because of this mechanism. In other words, for a given bandwidth, a FIR filter will ring less than a conventional analog filter, but it will still ring. The CW filters in this DSP are all FIR types, so there will be only minimal ringing.

Most radios set the sidetone so that if you are transmitting on the same frequency as the CW station you are working, the note of the received signal will be at the same frequency as your sidetone. But if your radio's sidetone does not match the received signal's audio frequency, then the CW sidetone will be outside the passband of the digital CW filter, it will not be heard! There are several ways around this problem.

First, it may be possible to modify the transceiver so that the sidetone is centered within the DSP CW filter bandpass you use (400, 600, 750, or 1000 Hz). This may be a matter of changing parts values in an RC oscillator timing circuit. But if the sidetone is derived by dividing down from a crystal timebase, then it might not be possible to change its frequency.

Another way is to tune the CW filter to the right frequency as outlined in these instructions.

Yet another way would be to bypass the DSP when transmitting, either by switching it out manually or by using a relay in place of the switch at S2.

❏ **Power supply considerations**. The DC power required is 12 volts at 400 ma maximum. It should be fairly well filtered, meaning not more than a few tenths of a volt of AC ripple. Many "wall transformer" supplies do not have adequate filtering even though they may be rated to supply the necessary 400 ma. However, they can be bolstered with the addition of about 5000 uF of additional filtering capacitance. So, if you experience AC hum, try adding a 4700 uF, 5000 uF, or larger capacitor across the DC power supply. One recommended power supply which does not cause a hum problem is the Radio Shack 273-1653A (12VDC, 1A).

❏ **Receiver AGC effects**. Keep in mind that your receiver's AGC takes place prior to the DSP. So if the DSP is filtering out a strong carrier, or strong QRM on CW, the audio output level of the DSP may drop significantly. However, this will be due to the receiver's AGC gain reduction action because the QRM is present in the receiver.

❏ **Further information**. An explanation of the algorithms used in this digital signal processor and a description of the hardware are in the September 1992 ***QST*** magazine article "Low Cost Digital Signal Processing for the Radio Amateur" by Dave Hershberger, W9GR. For those looking for a thorough mathematical treatment of the firmware noise reduction and autonotcher algorithm, refer to "Using the LMS Algorithm for QRM and QRN Reduction" by Dr. Steven E. Reyer, WA9VNJ, and David L. Hershberger, W9GR, in September 1992 ***QEX*** magazine.

HARDWARE DESCRIPTION

+5 volt power and ground connections are not shown on the schematic. They are:

	ground	**+5 volts**		**ground**	**+5 volts**
IC1	10	30	IC8	8	16
IC2	16	5	IC9	8	16
IC3	10	20	IC10	10	20
IC4	10	20	IC11	7	14
IC5	7	14	IC13	10	20
IC6	7	14	IC14	10	20
IC7	7	14			

The DSP-3 hardware starts with a 13 bit digital audio codec chip, IC2. 13 bits of precision is theoretically equivalent to an 80 dB audio dynamic range. This is 30 dB more range than the original W9GR DSP which only used 8 bits of audio precision. The 13 bit dynamic range, together with improved DSP software, makes setting audio levels much less critical. It also makes it possible to do much better CW filtering, since a wide dynamic range is necessary to filter out a weak CW signal amidst much stronger QRM. For voice applications, given that the signal to noise ratio on a relatively strong SSB signal might be 30 dB, there is 50 dB of S/N "headroom" left with 13 bits! In many cases, the S/N ratio will be quite a bit worse than 30 dB. Since 80 dB is much greater than 30 dB, 13 bits is more than adequate for amateur applications.

The audio codec chip IC2 uses switched capacitor technology for antialias lowpass filtering. It also includes a switched capacitor highpass filter. The lowpass filter cuts off at about 3.2 kHz. IC2 accepts the analog input at pin 18, and it outputs a serial digital audio bit stream at pin 13. Shift register chips IC3 and IC4 convert the serial output from IC2 to a parallel word which is can be read via the parallel data bus by the DSP CPU chip IC1. The sampling rate is approximately 7700 Hz.

The CPU chip IC1 contains a OTP (one time programmable; not erasable) PROM memory which has been pre-programmed with the DSP software or firmware. The CPU executes the instructions in its internal 8k bytes of program memory to perform the various DSP functions. IC1 is a 16 bit digital signal processor CPU with a 32 bit accumulator. This processor executes DSP type instructions at a rate of 5 MIPS.

For special applications, the CPU (IC1) can be replaced with one that does not contain internal program memory, such as a TMS320C10 or TMS320C15. These processors require an external program memory, which can be provided by using 57C43 high speed EPROMs at IC13 and IC14. This capability is provided for advanced experimenters who want to do their own DSP software development. Normally, IC13 and IC14 are not used in this DSP.

The rotary encoder switch at S3 outputs a 4 bit binary code which is periodically read by the CPU via the tristate inverter IC8. The setting of S3 determines which DSP function is executed by IC1.

IC5D is connected as an inverter; this gate produces a power-on reset pulse. When the power is turned on, this short pulse initializes the CPU (IC1), the shift registers (IC3 and IC4) and some of the various remaining flip flops. The output of IC5D should go low for a fraction of a second when the power is turned on.

IC7A accepts the 5 MHz clock output from IC1 via IC8 and produces a 2.5 MHz clock which is necessary to run the audio codec chip IC2. IC5A, IC5B, IC5C, IC7B, and IC6 all work to produce the proper timing signals to interface the audio codec chip and shift registers to the CPU.

IC9 is a port decoder, which produces active low outputs to read the binary encoder mode switch S3, to read the A/D converter's output from the shift registers, to write parallel digital audio words to the shift registers, and to write data to the LED bargraph display.

When the CPU writes to the LED display, the data word controlling the individual LEDs is written to IC10 and IC11. IC10 holds the "bottom" (leftmost) 8 bits of the LED display, and IC11 holds the top (rightmost) 2 bits. A logic 0 at any output turns the corresponding LED on, and a logic 1 turns the associated LED off.

After the CPU performs the desired filtering operation on the input audio, it outputs parallel audio words to the shift register chips at IC3 and IC4. These shift registers are bidirectional; in addition to converting the serial A/D data to parallel, they also convert the parallel digital audio data back to serial again so that it can be sent to the D/A in IC2.

The processed serial digital audio data enters IC2 at pin 8. Within IC2, it is converted back to analog again. IC2 also includes a switched capacitor lowpass filter for analog reconstruction and sin(x)/x correction. The filtered analog output is taken from IC2 pin 2.

At this point the only thing that is needed is AF GAIN control and audio amplification. R11 is the AF GAIN control, and IC12 provides over 1 watt of audio output power.

The 12 volt DC power supply powers the audio power amplifier IC12. The 12 volt supply is dropped to 5 volts to power the audio codec IC2 and the digital circuitry by linear voltage regulator VR1.

In the interests of tradition, a small PC board "cartoon" is also included with the DSP hardware at no additional charge.

IN CASE OF DIFFICULTY

❏ **1**. Check your soldering for unconnected pins, cold solder joints, and/or solder bridges. ***This is a very important troubleshooting step! Almost every kit which has been returned to us for repair has not worked because of simple soldering defects!*** The odds are that you can save yourself a lot of time and possible embarrassment if you carefully check your soldering work for unsoldered connections, "cold" joints, solder bridges, and scratched off PC board traces. Use a magnifying glass to find suspicious connections, and re-solder them. Don’t rely on the solder mask to be an infallible insulator against sloppy soldering. Remove solder blobs which encroach over the solder mask or onto nearby "vias." Use a soft brush to remove metallic and conductive solder dust and debris. In several cases, the soldering problem was found *underneath* an IC socket, necessitating removal and replacement of the socket! Sometimes capillary flow can draw solder up through vias and holes to the other side of the board. If this happens underneath a socket, it could be very hard to find the problem. It’s really true what Heathkit used to say in their manuals: 90% of the problems with Heathkits were due to bad soldering.

❏ **2**. Check all of the integrated circuits to see if pins are bent underneath the chips. With the power off, use an ohmmeter to check for continuity from the IC pin to the pad underneath the board. Or, you can remove each chip for visual inspection and then reinstall them.

❏ **3**. Use an oscilloscope if you have one available to look for "sick" looking (i. e. lack of full logic swings) digital waveforms. If you find a sick looking logic waveform, it may be shorted to a neighboring trace. Look for bad soldering somewhere on that node. Bear in mind that many of the logic signals on the board are "tri-state" meaning that the drivers go high impedance at certain times. You may see what looks like "sagging" voltages; this is normal on some nodes such as the data bus. Look for the logic swings to go all the way to a valid logic 1 and a valid logic 0 at least some of the time. With the power off, use an ohmmeter to check for shorts to nearby traces and IC pins.

❏ **4**. From our statistical experience so far, there is at least a 95% probability that any given non-working kit has *soldering defects*. To search for soldering defects, begin by first going around each chip with an ohmmeter. Check to see if pin 1 is shorted to pin 2, pin 2 to 3, etc. on each chip. If you find pins shorted together, refer to the schematic to see if they are supposed to be connected; sometimes that will be the case.

❏ **5**. If you do not find a short that way, then take your ohmmeter and go around each chip and check not only adjacent pins on each IC, but *every* other pad on the PC board. Each time you find continuity, check against the schematic to see if the two points which you found to be connected are in fact supposed to be connected. Although this may sound time consuming, it may be the fastest way to get your kit working.

❏ **6**. Some of the shorts we have found have been due to very small dendrites of solder which worked their way under the solder mask. You might have to look very closely, with a magnifying glass, to find them!

❏ **7**. We have learned of several failures which have occurred after weeks or months of normal operation. These failures were due to residual debris and flux from soldering working through the solder mask and contacting nearby traces. These failures may not be "zero ohm" shorts but may measure in the tens, hundreds, or thousands of ohms. If you experience such a delayed failure, it may be cured by using a small brush and chemical flux remover. Alternatively, you might try *carefully* and *gently* scraping off the flux and soldering residue from around each pad with a small screwdriver or awl, being careful not to scrape off the traces.

❏ **8**. Make sure that all grounds are connected together among the devices that connect to the DSP: the 12 volt power supply, the audio source (receiver), the DSP unit itself, and the speaker. Make sure that the "hot" and "ground" wires in cable connectors are not reversed.

❏ **9**. Check the two diodes at D1 and D2; the banded ends should be oriented towards the front (LED bargraph) end of the board.

❏ **10**. Do some of the LED segments in the bargraph display fail to light up? This is usually NOT due to a defective bargraph. It is usually caused by solder shorts on the data bus, and sometimes due to static damage to IC10 (which drives the 8 leftmost LED segments; the two on the far right are driven by IC11). The chip used at IC10 has proven to be the most static sensitive of any of the integrated circuits used in this kit. You can verify whether the LED bargraph is defective by unplugging it and moving it one position to the left or right (one pair of pins at the end of the bargraph will then not be plugged into the socket). If the same LED does not illuminate, then the problem may be a bad bargraph display. But if the unlit LED stays with the same position of the socket, rather than moving with the bargraph, then there is something wrong with the circuitry driving the bargraph. First check for soldering problems. Then if the problem is one or more segments within the 8 leftmost LED segments, try replacing IC10.

❏ **11**. Does the mode switch (S3) appear to be intermittent or inoperative? If so, the bipolar 74**LS**74 chip is probably not installed at IC6. Check to make sure that a bipolar type 74**LS**74 is installed at IC6. (Not a CMOS 74**HCT**74 device!) (Refer to assembly step #24.)

❏ **12**. Do the noise reduction modes seem to increase the noise level rather than reducing it? Check the setting of S4. It should be set to "0" (AGC off). When S4 is set to "1," then AGC is enabled in the noise reduction modes. The AGC can increase the signal level up to 30 dB. When switching the DSP in and out with S2, it can sometimes seem like the noise is increased, but this is only because the overall gain is higher when the DSP is in when the AGC is on. When comparing the output to the input, always make sure that S4 is set to "0," so that the "DSP in" and "DSP out" settings of S2 will have the same gain.

Also, when evaluating the noise reduction capability of the DSP-3, be sure to use a noisy signal - as opposed to pure noise, with no desired signal present. Make sure that there is at least some signal present, to give the algorithm a correlated signal to acquire.

❏ **13**. Check the DC voltages at various points throughout the circuit. With power on but no signal applied, you should measure the following DC voltages:

IC5 pin 11	5.0
IC2 pins 2,3,4,17,18,19,20	2.5
IC12 pins 1,8	approximately half of the +12 volt power supply (about 6 volts)
IC12 pin 14	12 volts (supply voltage)
IC10 pin 20	5 volts (V_{cc})

❑ **14**. Short out the 10 k ohm resistor at R1 with a clip lead. With the resistor shorted, the voltage at IC5 pin 11 should go to zero. When you remove the clip lead, the voltage should return to 5 volts. This tests the reset circuit. When IC5 pin 11 is at 0 volts, the DSP is being reset. If IC5 pin 11 remains at a low voltage, the DSP cannot operate.

❑ **15**. If you have an oscilloscope, or audio signal tracer, you can trace audio the signal through the unit. Input audio is applied to the junction of R5 and R7. Input audio will also appear at pin 19 of IC2, riding on a 2.5 volt DC bias. At this point the audio is digitized inside IC2 leaves in a serial format from IC2 pin 13. The digital audio is sent to the DSP CPU (IC1) via shift registers IC3 and IC4. Serial digital audio returns from the CPU via shift registers IC3 and IC4, and arrives at IC2 pin 8. After being converted to analog, the audio appears at IC2 pins 2, 3, and 4. From this point the audio is applied to the AF GAIN control R11 and then to the audio power amplifier IC12. Finally, the speaker output is driven from IC12 pin 8.

❑ **16**. Does the bargraph display seem to indicate audio presence? If so, audio is getting to the A/D converter (IC2) and into the CPU (IC1). The problem is probably at the D/A converter IC2, S2, or IC12. If the bargraph does not indicate anything when audio is applied, or if it flashes randomly, look for problems in the digital part of the circuitry.

❑ **17**. Of the units that have been returned for "warranty service," about 95% of the problems have been *defective soldering*. One non-soldering problem that has occurred several times has been PC board defects. So in the great majority of cases, the problems have been attributable to defects in kit building. We say this not to impugn anybody's building skills but to point out that there have been only about 3 defective electrical components discovered (out of over 200,000 components shipped as parts of kits).

❑ **18**. **Although the great majority of kit builders have completed the W9GR DSP kit successfully, there have been significant numbers of "warranty" returns, the vast majority of which are anything but that. Over 95% of the kits returned for "warranty service" do not work because of *bad soldering*. The warranty covers "defects in materials and workmanship" of Quantics, and *does not* cover the workmanship of the kit builder, over which we have no control.**

❑ **19**. **We are spending so much time correcting soldering defects that it is slowing down kit shipments. Therefore, effective immediately, any kit returned for "warranty repair" must be accompanied by a check for $40.00 which is our "flat rate" for getting defective kits to work and returning them to you (abuse excepted). If in our judgment the failure is due to a defect in our components or workmanship, your $40.00 payment will be returned to you with the repaired kit. On the other hand, if the failure is due to improper assembly or soldering defects, the flat rate fee will be retained.**

❑ **20**. In summary, the warranty covers defects in components supplied by Quantics (for example PC board manufacturing flaws). The debugging service, which is not free, covers kit building errors (for example soldering defects). This policy supersedes and clarifies any previous statements.

❑ **21**. As of November 1994, the statistics regarding kit problems are as follows:

Defective soldering:	96
Chips plugged into sockets incorrectly:	2
PC board flaws:	3
Bad parts supplied by Quantics:	2

❑ **22**. If you are still unable to resolve the difficulty, you may either write for assistance or return the unit for repair. Any returned kit **must be accompanied by a check for $40.00**. If in our opinion the problem is covered by the warranty, then your check will be returned to you with the kit. The warranty is not intended to cover defective soldering, so please check your soldering before returning kits for repair. Support is handled by correspondence; at the present time we are not able to provide support via telephone. However, problem reports, suggestions, comments, and questions are welcomed and should be directed to:

QUANTICS
P. O. Box 2163
Nevada City, California 95959-2163

If you need a reply to a question, a self-addressed stamped envelope would be appreciated.

Internet users can send questions to **dave@w9gr.com** - this is the fastest way to get a response. E-mail is normally answered the same day unless we are out of town. Our web site is located at: **http://www.w9gr.com**.

UPS shipping address for returned kits:

David L. Hershberger, W9GR
10373 Pine Flat Way
Nevada City, California 95959-9136

Parcel post shipments (U. S. mail) should be directed to the P. O. Box address above.

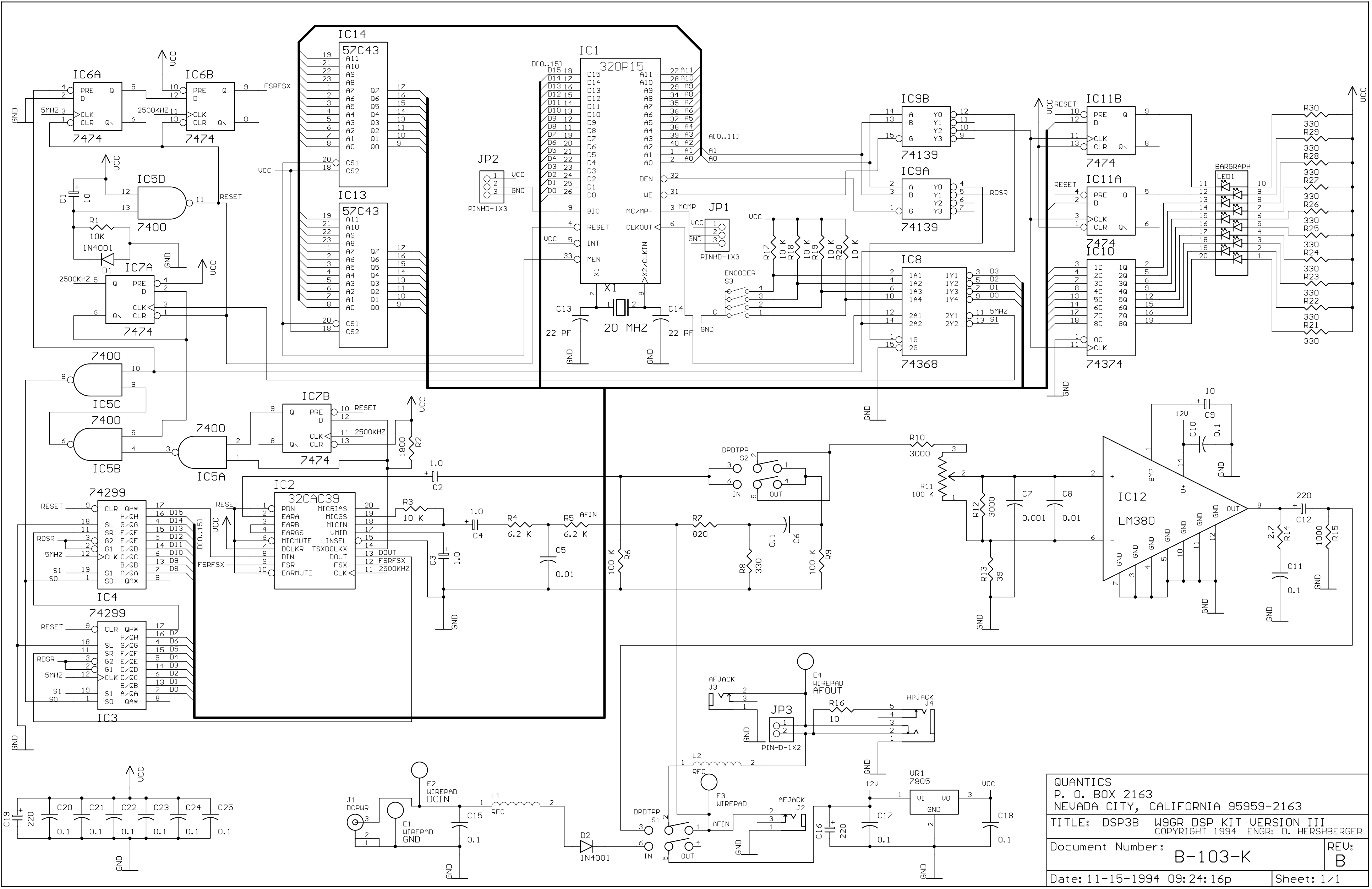
LIMITED WARRANTY

This product is warranted to be free of defects in materials and workmanship for a period of thirty days from the date of purchase. In the event of notification within the warranty period of defects in materials or workmanship, the seller will, upon return of the product, repair or replace (at its option) the defective parts. The remedy for breach of this warranty shall be limited to repair or replacement and shall not encompass any other damages, including but limited to loss of profits, special, incidental, consequential or other similar claims. This warranty does not cover any damages due to accident, misuse, abuse or negligence on the part of the purchaser. This warranty does not cover other equipment or components that a customer uses in conjunction with this product.

THE SELLER SPECIFICALLY DISCLAIMS ALL OTHER WARRANTIES, EXPRESSED OR IMPLIED, INCLUDING BUT NOT LIMITED TO IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE. Some states do not allow the limitation or exclusion of liability for incidental or consequential damages, or the exclusion of implied warranties, so the above limitations and exclusions may not apply to you. This warranty gives you specific legal rights and you may also have other rights which vary from state to state.

Thank you for your purchase of the W9GR Digital Signal Processor!

73, *Dave Hershberger*, **W 9 G R**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