W9GR DSP II
Audio Filter

RCE

Introduction

Thank you for purchasing the W9GR DSP II Audio Filter. The W9GR DSP II is manufactured in the United States by j•Com to the highest quality standards.

Dave Hershberger, W9GR, has applied the latest Digital Signal Processing techniques to the problem of interference from both man-made and natural sources. The results of his efforts are embodied in the DSP II Audio Filter. With a DSP filter, a single circuit can be made to function as any number of different filters by changing only the software algorithms executed by the microprocessor CPU.

The DSP II has been designed to operate in 10 different modes. Four filters are optimized for reducing interference to SSB phone signals from CW, heterodynes and random noise interference. Four filters operate as "brick-wall" CW bandpass filters attaining bandwidths and phase linearity impossible with analog devices. Two filters are designed for RTTY and HF packet radio operation.

A single front panel switch allows the user to easily select the filter he wants to use at any time.

Installation

Power connection

The W9GR DSP II filter requires 9 to 16 volts DC. A maximum of 250 mA is drawn at full volume. In many cases, your rig will operate from a 13.8 volt power source, and you will be able to power the filter from one of the accessory jacks, or wire it directly to your rig's power supply. Consult your transceiver manual for information on how to make this connection.

We recommend that shielded cable be used to reduce RF pickup by the power supply leads. It is always wise to check the voltage and polarity at the power plug before you plug it into the W9GR DSP Filter.

ALWAYS DISCONNECT YOUR RIG FROM THE AC LINE BEFORE MAKING ANY CONNECTIONS!

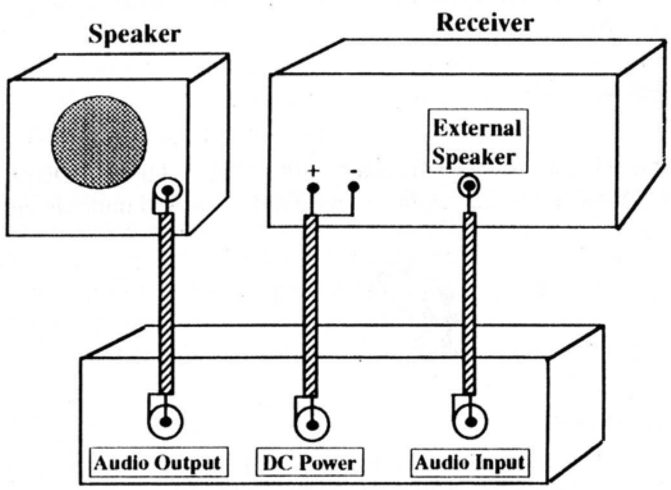
Audio Connections

The W9GR DSP Filter installs between your receiver and an external speaker. Both audio connections to the filter are made with 1/8 inch mono phone plugs connected to a shielded audio cable as shown:

If your speaker is supplied with a 1/8 inch mono phone jack, you can use the audio cable supplied with the W9GR DSP Filter. First, unplug the cable from your external speaker, and plug it into the AUDIO INPUT jack on the rear panel of the filter. Second, plug the supplied audio cable into the AUDIO OUTPUT jack on rear panel of the filter. Finally, plug the other end of this cable into the speaker's input jack. If your rig requires different connectors, it will be necessary to cut the audio cable and solder appropriate connectors to the ends. A sufficiently long cable has been provided to make this possible.

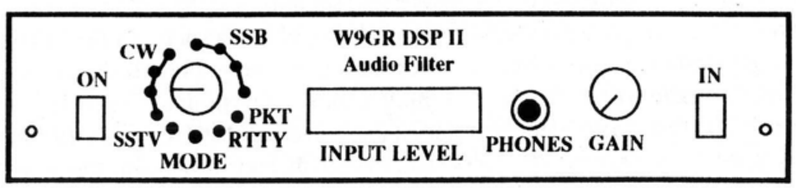
The filter must be connected to the speaker output of your receiver. In most cases, the headphone output or the record audio output will not provide sufficient drive to operate the filter correctly. In addition, these outputs will probably be a different impedance from the 8 ohms expected by the filter, resulting in some distortion.

Typical W9GR DSP II Filter Conections



W9GR DSP II Audio Filter

W9GR DSP II Audio Filter Front Panel



Front Panel Controls

ON	Power On/Off switch.
MODE	Mode selection switch.
INPUT LEVEL	Bargraph audio level indicator.
PHONES	0.25 inch stereo/mono headphone jack.
GAIN	Audio output gain control.
IN	Filter In / Filter Bypass switch.

Initial Operation

After connecting the power and audio cables, turn on your receiver and depress the ON switch on the front panel of the filter. The LED display should light up, display a few bars, and then go blank.

Mode Selection

The Mode switch is used to select the desired filter type. Set the MODE control to the 12 o'clock position for the combined noise reduction and multiple automatic notch filter.

In/Out Bypass Switch

The switch at the right labelled **IN** is used to select whether the filter should be used or bypassed. When this switch is depressed, the filter is used to filter the signal. When the IN switch is out, audio bypasses the filter itself and is fed directly to the speaker or headphones through the filter's internal amplifier.

Depress the IN switch so that the filter is being used. Slowly increase the volume on your receiver, the noise level or received signal should cause the LED bars to light up from the left. For proper operation, the level should be set such that the LED at the far right flashes occasionally, but never stays lit for very long. If this level is exceeded, the audio will be distorted. Simply reducing the volume control on your receiver will correct the problem. The automatic gain control (AGC) action of most modern receivers will keep the level constant enough that no further adjustment will be required.

Adjust the GAIN control on the filter for comfortable listening on your speaker or headphones.

SSB Operation

There are four filters designed to improve the reception of SSB or speech signals. Although we have designed the filter for use by amateur radio operators, these filters are also very useful for increasing the intelligiblity of any speech signal with a lot of noise.

SSB - Combination Denoiser and Automatic Notch Filter

The filter which will probably be the most useful for general SSB reception is the combination Denoiser and Automatic Notch Filter. Turn the Mode Selection knob to the 12 o'clock position to select this filter.

Because this filter is a compromise between the optimized denoiser and the optimized automatic notch filter, it will not be quite as effective as the optimized filters. However, it is still very effective, and the efficiency of having both functions operate automatically makes it a good default operating mode. If you are plagued by excessive atmospheric noise, or heterodyne interference below the threshold of the compromise filter, you can always switch to the optimized modes.

SSB - Optimized Denoiser

This filter has been optimized for maximum denoising performance using the Widrow-Hoff LMS (Least mean squares) adaptive filtering algorithm. The digital signal processor adapts the transfer of each frequency in the spectrum to the output according to an algorithm which was designed to enhance the intelligibility of speech signals and reduce the noise content of the signal.

While a conventional filter uses *frequency* as its criterion for passing or rejecting a signal, the filters designed with the LMS algorithm use *correlation*, or repetitiveness, as a method for discriminating against noise or interference. The filter samples the audio signal at regular intervals. Signals which are correlated from one sample to the next are considered likely to be speech, and are passed to the output. Signals which are more or less random in nature are considered to be noise, and are filtered out of the output. This method of operation is considered to be adaptive because the characteristics of the filter will change from one instant to the next as it adapts to the changes in the input signals.

This kind of denoiser will be most effective against white noise, the background hiss resulting from random fluctuations in the signal path due to atmospherics and the background noise from your preamplifier. It will also be useful in reducing QRN from line noise, ignition noise, and other pulse noises.

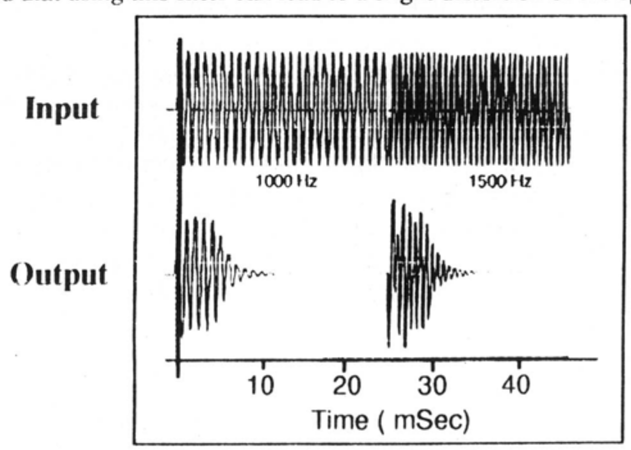
SSB - Optimized Automatic Notch Filter

This filter monitors the incoming audio for constant frequency interference. Should a signal remain constantly on a single frequency for more than a few milliseconds, the digital signal processor will notch it out by not transferring that signal to the output. Depending on the strength of the heterodynes and the other signals, the Optimized Automatic Notch Filter will be effective against 3 or 4 heterodynes appearing simultaneously.

The notch filter is actually the reverse of the noise filter. In this case, the interference is more correlated than the desired signal. A constant frequency sine wave is about

SSB - Weak Signal Automatic Notch Filter

as correlated as you can get. The filter parameters are set such that the most correlated signal is filtered out, and the less correlated signal is passed to the output. The threshhold level of the automatic notch filter in the Combination mode, and in the Optimized Automatic Notch filter mode is set so that it minimizes distortion of the speech signal while still providing relief from strong heterodyne interference. Sometimes, it is desirable to notch out weak carrier signals which are at a level below the threshhold designed into the other two filters. The Weak Signal Automatic Notch filter has been included to do this. However, under certain conditions, you will find that using this filter can lead to a slight distortion of the speech signals.



The graph shows the effectiveness of the automatic notch filter on a 1000 Hz and a 1500 Hz sine wave. The interference is removed in less than 0.01 Sec.

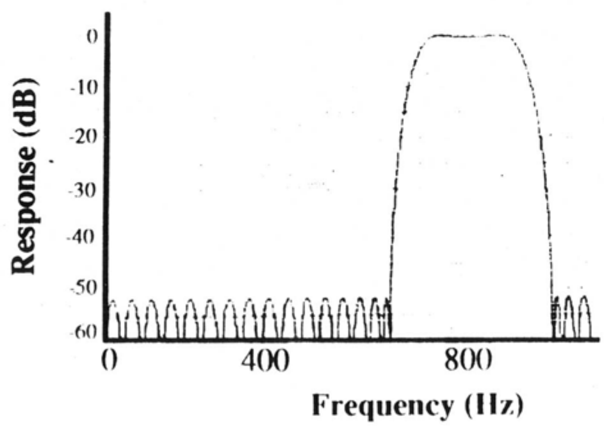
Bandpass Filters

Bandpass filters allow signals of a certain frequency to pass from the input to the output while attenuating signals of other frequencies. The use of digital signal processing allows the construction of filters which would not be practical using discrete components. The passband can be tailored to present very little distortion to signals of the desired frequency, while attenuating to more than 40 dB signals outside of the passband.

The use of linear phase Finite Impulse Response (FIR) filters eliminates time delay distortion in the filter output. In CW mode, this means far less ringing for any given bandwidth. When using the filter for FSK reception, there is less inter-symbol interference, and on SSTV video distortion is reduced.

Center Frequency: 700 Hz
Bandwidth: 200 Hz

A good filter for use while browsing the band, the 200 Hz wide bandpass filter centered on 800 Hz will effectively increase the signal to noise ratio yet allow you to tune the band at a reasonable speed without missing weak stations. Once you begin a QSO, you will probably want to decrease the bandwidth, especially if there is QRM from an adjacent station.



Center Frequency: 700 Hz
Bandwidth: 100 Hz

The 100 Hz wide filter centered on 800 Hz is a good filter for contest and DX operation when the band is crowded, or the signal is very weak. At this bandwidth, the signal to noise ratio is improved significantly over wider filters. With a DSP filter, there is no ringing, even with an extremely narrow filter such as this one.

Center Frequency: 700 Hz
Bandwidth: 30 Hz

The 30 Hz super narrow bandpass filter has been included for those operators working with very weak signals. This filter offers a substantial perfomance increase for satellite, meteor scatter and EME operations.

Center Frequency: 400 Hz
Bandwidth: 100 Hz

This filter is designed for those operators who prefer to listen to a lower pitched CW signal. The passband is 100 Hz wide and centered on 400 Hz.

A natural effect of converting an analog signal to a digital integer value and then back to an analog signal is a certain amount of quantization noise. This occurs because there are only a certain number of possible discrete values allowed in the digital representation of the signal. A technique called *dithering* is used to reduce the audible effect of quantization on the desired signal. However, a side effect of dithering is a background noise which is audible when no signal is present. This background hiss will not be a problem when a signal is present, but you may notice it when there is no input signal. It is normal.

HF Packet Bandpass Filter

The HF Packet filter is a 120th order linear phase FIR filter with a passband of 1550-1850 Hz and a very steep rolloff above and below these frequencies. Passband ripple is less than 0.3 dB. Signals outside the passband are attenuated 40 dB or more.

RTTY Bandpass Filter

The RTTY filter is a 120th order linear phase FIR filter with a passband of 2075-2345 Hz and a very steep rolloff above and below these frequencies. Passband ripple is less than 0.3 dB. Signals outside the passband are attenuated 40 dB or more.

The RTTY bandpass filter is equally effective for improving reception of **Pactor** and **Clover** FSK modes.

SSTV Bandpass Filter

The Slow Scan TV filter is a 120th order linear phase FIR filter with a passband of 1150-2350 Hz with a very steep rolloff above and below these frequencies. Distortion within the passband is less than 0.3 dB. Signals outside the passband are attenuated 40 dB or more.

Why does the volume decrease when a big carrier comes on?
Because the notches introduced by the W9GR DSP II filter are extremely narrow, you should not be able to detect a change in the sound of the speech signals. However, a strong carrier causes your receiver's automatic gain control (AGC) to reduce its sensitivity, thus reducing the volume output. The amount of reduction is shown on the S-Meter which is monitoring the AGC signal. For best reception of weak SSB signals in the presence of very strong carriers, switch to manual AGC and adjust your RF gain control.

When I turn on my linear, I can hear myself. What can I do?
You have a problem with RFI. The W9GR DSP II filter is an audio device. We have designed it to be relatively insensitive to RF interference. However, if you are running high power and your antenna is mismatched, or very close to the shack, it is possible that some RF will get into the filter and appear as "talkback" in your headphones. There are many handbooks dedicated to the subject of eliminating RFI. All of the suggestions you find there will be applicable. We recommend the following: 1. Use only shielded cable for all connections to the filter, and make sure these cables are as short as possible. 2. Ensure that your station is adequately grounded for RF at the frequency you are using. Connect the metal case of the filter to the common ground point. 3. Make sure your SWR is as low as possible. 4. Wrap cables entering the filter around a ferrite core.

Is the W9GR DSP Filter effective against Sideband Splatter?
The denoiser parameters are set to pass signals which have the same moment to moment correlation as speech, and to filter out signals which are less correlated. The automatic notch parameters are set to filter out signals which are more correlated than human speech. Because splatter is actually speech distorted by a change in frequency, the correlation will be the same as the signals you want to hear. Because of this, the W9GR DSP II filter does little to reduce splatter interference.

Where can I learn more about how DSP filters work.
Dave Hershberger W9GR and Dr. Steven Reyer WA9VNJ wrote an excellent article in the September 1992 issue of QEX on the use of the LMS Algorithm for QRM and QRN Reduction. The original W9GR DSP Filter was described in QST, September 1992. Back issues of both of these publications are available from the ARRL. In addition: B. Widrow, et. al, "Adaptive Noise Cancelling: Principles and Applications," Proceedings of the IEEE, Vol 63, No. 12, Dec 1975, pp 1692-1716. R. Chassaing and D. Horning, *Digital Signal Processing with the TMS320C25* (New York: Wiley, 1990). *First Generation TMS320 User's Guide*, Texas Instruments, Inc, 1988.


CW 400 Hz Bandpass 100 Hz wide

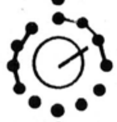

CW 700 Hz Bandpass 200 Hz wide

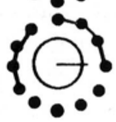

CW 700 Hz Bandpass 100 Hz wide


CW 700 Hz Bandpass 30 Hz wide


SSB Combination denoiser and notch


SSB Optimized denoiser


SSB Optimized automatic notch


SSB Weak signal automatic notch

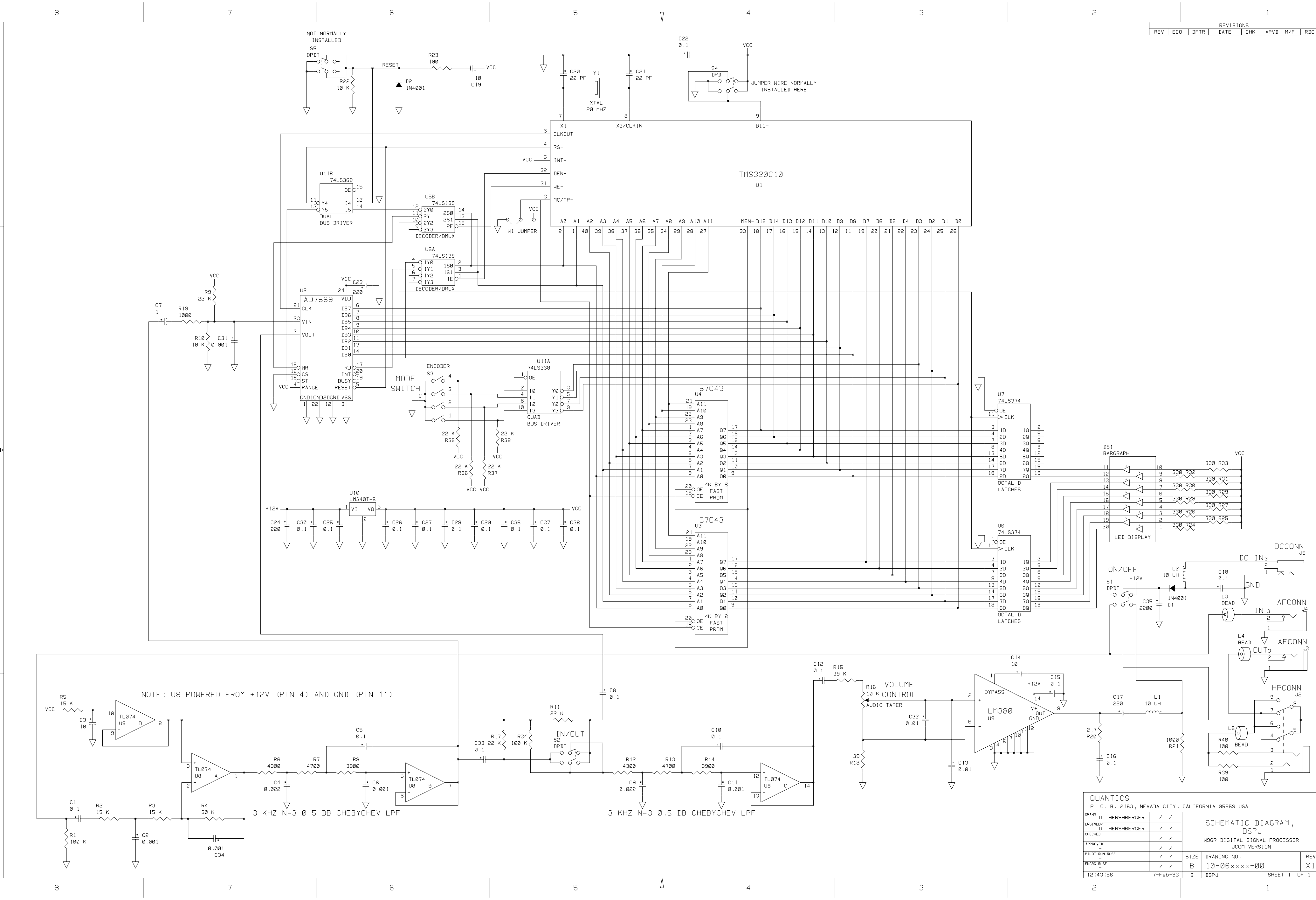

HF Packet Filter


RTTY Filter


SSTV Filter

W9GR DSP II Specifications

Notch rejection	> 40 dB
Bandstop rejection	> 40 dB
Passband ripple	< 0.3 dB
Power Requirements	12 VDC @ 250 mA
Audio Output	2 watts into 8 ohms
Audio Connectors	0.125" mono phone
DC connector	0.220" coaxial
Size	1.5" x 5.5" x 6.4"


NOT NORMALLY INSTALLED
S5 DPDT
R22 10 K
RESET
R23 100
D2 IN4001
10 C19
VCC
C22 0.1
C20 22 PF
Y1 XTAL 20 MHZ
C21 22 PF
S4 DPDT
JUMPER WIRE NORMALLY INSTALLED HERE
X1
X2/CLKIN
BIO-
CLKOUT
RS-
INT-
DEN-
WE-
MC/MP-
TMS320C10
U1
A0 A1 A2 A3 A4 A5 A6 A7 A8 A9 A10 A11
MEN- D15 D14 D13 D12 D11 D10 D9 D8 D7 D6 D5 D4 D3 D2 D1 D0
U11B 74LS368
DUAL BUS DRIVER
U5B 74LS139
DECODER/DMUX
W1 JUMPER
U5A 74LS139
DECODER/DMUX
C23 220
U2 AD7569
CLK VIN VOUT
DB7 DB6 DB5 DB4 DB3 DB2 DB1 DB0
WR CS ST RANGE
RD INT BUSY RESET
GND1 GND2 DGND VSS
R9 22 K
C7 1
R19 1000
R10 10 K
C31 0.001
ENCODER
MODE SWITCH
S3
U11A 74LS368
QUAD BUS DRIVER
22 K R35
22 K R38
22 K R36
22 K R37
57C43
U4
4K BY 8 FAST PROM
U3
U7 74LS374
OCTAL D LATCHES
U6 74LS374
DS1 BARGRAPH
LED DISPLAY
330 R33
330 R32
330 R31
330 R30
330 R29
330 R28
330 R27
330 R26
330 R25
330 R24
U10 LM340T-5
+12V
C24 220
C30 0.1
C25 0.1
C26 0.1
C27 0.1
C28 0.1
C29 0.1
C36 0.1
C37 0.1
C38 0.1
DCCONN J5
DC IN
GND
ON/OFF
S1 DPDT
L2 10 UH
C18 0.1
IN4001 D1
C35 2200
L3 BEAD
AFCONN J4
IN
L4 BEAD
AFCONN J3
OUT
HPCONN J2
L5 BEAD
R40 100
R39 100
NOTE: U8 POWERED FROM +12V (PIN 4) AND GND (PIN 11)
R5 15 K
C3 10
TL074 U8
C8 0.1
R11 22 K
IN/OUT
S2 DPDT
C33 0.1
R17 22 K
R34 100 K
R6 4300
R7 4700
R8 3900
C5 0.1
C4 0.022
C6 0.001
C1 0.1
R2 15 K
R3 15 K
R4 30 K
R1 100 K
C2 0.001
0.001 C34
3 KHZ N=3 0.5 DB CHEBYCHEV LPF
R12 4300
R13 4700
R14 3900
C10 0.1
C9 0.022
C11 0.001
C12 0.1
R15 39 K
R16 10 K
VOLUME CONTROL
AUDIO TAPER
C32 0.01
39 R18
C13 0.01
C14 10
C15 0.1
BYPASS
LM380 U9
C17 220
L1 10 UH
2.7 R20
C16 0.1
1000 R21
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ENGRG RLSE
SCHEMATIC DIAGRAM, DSPJ
W9GR DIGITAL SIGNAL PROCESSOR
JCOM VERSION
SIZE B
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REVISIONS
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