DANMIKE

DSP-NIR

Digital Signal Processing
Noise and Interference Reduction Unit

USER'S MANUAL

09.95

TABLE OF CONTENTS

BRIEF DESCRIPTION OF THE INSTRUMENT

The DSP-NIR unit performs LF signal processing which improves the reception quality of MF-HF radios. Reception conditions for both professional users and ham operators can be substantially improved by using this unit. It uses the latest 16 bit DSP (Digital Signal Processing) technology, which makes it possible to reprocess signals with even a high noise content. The DSP-NIR instrument (**N**oise and **I**nterference **R**eduction) reduces atmospheric noise and signal interferences in a manner that makes the received signal understandable. Furthermore, the unit is fitted with a range of FIR (**F**inite **I**mpulse **R**esponse) filters, each one of which is optimised for specific communications objectives.

The controls are easy to use with the help of multi-setting MODE switch on the frontpanel, which always clearly indicates which MODE is currently in use. When the AGC function is activated, the LF signal is boosted so that the dynamic range is expanded. The BYPASS switch makes it possible for the user to quickly switch between the original and the reprocessed LF signal.

The DSP-NIR instrument is installed between the radio and an external speaker or headset. The in-built LF amplifier is able to drive a 2 – 8 Ω speaker. In addition to the speaker/headset output, the instrument is provided with an output line for data communications.

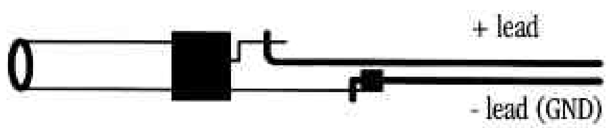
Pages 8 and 9 show how the DSP-NIR instrument can be connected to external units. The required cables can be made by the user with the plugs supplied. A more detailed description of how to connect the instrument is given below.

POWER Connection:

The DSP-NIR should be supplied with a constant DC voltage of between 11 and 15 V. The instrument can be damaged if a DC voltage of over 15 VDC is applied. The power supply should deliver a minimum of 0.5 Amps, however the unit will use up to 1 Amp at full volume. *A power supply that can deliver a constant 12 VDC / 1 Amp is therefore recommended.*

The enclosed 5.5 mm DC-plug is wired as shown in Figure 1. The short terminal is connected to the + lead and the long terminal is connected to the – lead. If the plug is correctly wired, the POWER LED will light up when the unit is turned on (by turning the volume knob from the OFF position to 1/2 max.). If the plug is incorrectly polarised, the instrument will not be harmed. When the unit is turned on, it will not work and the POWER LED will not light up.

Figure 1.



AUDIO INPUT:

Connect the Speaker Output from your receiver or transceiver to AUDIO INPUT on the backpanel of the DSP-NIR. An RCA phono jack is supplied for this purpose.

SPEAKER OUTPUT:

Connect a 2 – 8 Ω speaker to SPEAKER OUTPUT with the RCA phono jack supplied.

LINE OUTPUT:

Can possibly be connected to Packetmodem, RTTY printer, SSTV decoder, etc. The line's output signal is a constant 700 mV (RMS) at a normal input signal level. The line output can be loaded up to 600 Ω without distortion.

PHONES:

The output is connected to a headset which has a 6.35 mm stereo Phone Jack. If the headset is fitted with a mono jack, a converter jack (mono to stereo) should be used, or a stereo jack should be wired up as per Figure 4. If the headset is used, the speaker output will be interrupted.

Figure 2.

DSP-NIR backpanel

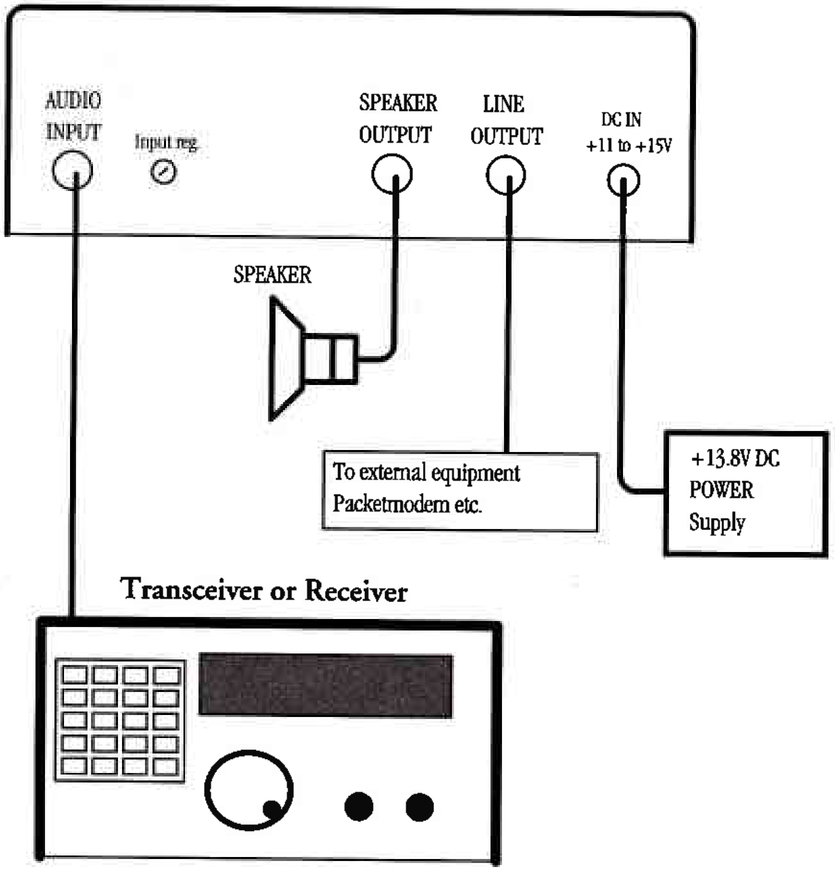


Figure 3.

DSP-NIR frontpanel

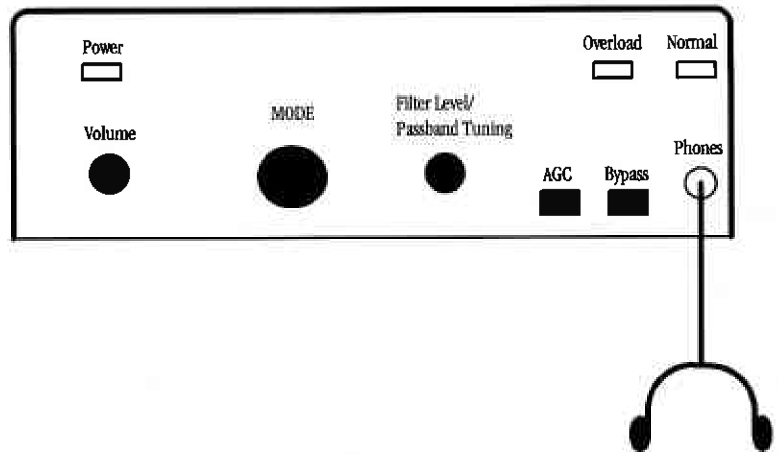
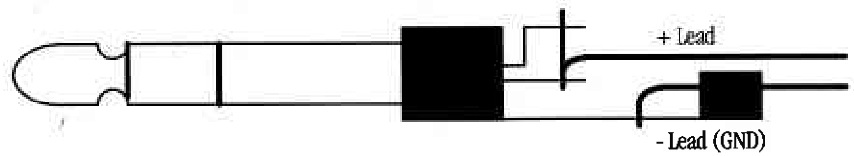


Figure 4.

Connection of 6.35 mm stereo phone jack to a 2-lead cable



CONTROLS

This section describes the controls on the DSP-NIR.

Volume & Power On/Off

The volume knob (and associated circuit breaker) is used partly to turn on/off the unit and partly to regulate the LF signal's output level. When the knob is turned anticlockwise to its full extent, the unit is turned off and the POWER diode is off. When the knob is turned clockwise, the unit is turned on and the POWER diode emits a yellow light. When the knob is turned clockwise the output signal is amplified. The unit should be turned on via the volume knob and not by the power supply switch, otherwise the unit might possibly not start up correctly.

MODE Switch (Multiple Settings)

This switch sets the instrument to the desired filter. There are 12 different filter modes in all. It is possible to choose the following modes: CW Narrow, SSB Narrow, SSB Wide, NOTCH, Peak, NT+Peak, Peak adj., PBT Wide adj., PBT Narrow adj., Packet, SSTV, and RTTY. A detailed description of each filter mode can be found in the section entitled "Operation" on page 12.

Filter Level/Passband Tuning:

This rotating knob has two functions. It is partly used to change the Peak filtering in Peak adj. mode and is partly used to adjust the centre frequency in PBT Wide and PBT Narrow mode.

AGC:

When the push-button is in, the AGC function is active and boosts the LF signal so that the dynamic range is increased. If the AGC push-button is out, the DSP-NIR will not have an AGC function.

BYPASS:

When the push-button is in, the instrument is bypassed, in other words the signal runs through the instrument without being processed. The signal is processed when the push-button is out. If the instrument is turned off, the BYPASS button needs to be pushed in for the signal to run straight through the unit.

Normal and Overload LEDs:

These two LEDs indicate whether the input signal to the DSP-NIR is adjusted to an appropriate level. A detailed description of how the input signal is properly adjusted can be found on page 12.

Internal Settings (Jumpers):

The DSP-NIR can be changed from 22 Ω to 6 KΩ input impedance by an internal jumper. Removing the jumper (JMP1) makes the audio input high impedance (see Figure 5 on page 40). The instrument is set to 22 Ω, as most receivers/transceivers give the best performance with this load. In CW Narrow mode, it is possible to choose between 400 Hz, 600 Hz and 750 Hz centre frequencies. The desired centre frequency is chosen by an internal jumper (see Figure 5 on page 40). The instrument is factory-set to 750 Hz.

OPERATION

This section describes how the level of the LF input signal is correctly adjusted and a description is given of each filter mode.

Adjustment of the Audio Input Signal

The DSP-NIR is connected as described in the section "Installation", that is to say the receiver's SPEAKER OUT is connected to AUDIO INPUT on the backpanel of the DSP-NIR. To be able to use the DSP-NIR's large dynamic range without overloading, an appropriate LF input signal is required from the output of the receiver/transceiver. By holding the input signal to the DSP-NIR just under the OVERLOAD level, the optimal S/N ratio is obtained with minimum distortion.

The following procedure can be used to adjust the Audio Input signal. First place the receiver/transceiver's volume knob in an appropriate mid-range setting. The radio is tuned to a powerful signal and then the Input reg. potentiometer on the backpanel of the DSP-NIR is adjusted so that the OVERLOAD LED lights up every once in a while. The NORMAL LED is continually lit.

Modern receivers/transceivers with in-built automatic gain control (AGC) in most situations will keep the input signal to the DSP-NIR constant, however it can sometimes be necessary to adjust the receiver/-transceiver's volume knob. The AGC function can be used to advantage in many filter modes. Particularly in NOTCH and CW mode, the incre-

ased dynamics will give a better readability of the signal if the DSP-NIR is connected to a receiver/transceiver that has sub-optimal AGC regulation.

If the input signal is too low, it will lead to a worse S/N ratio because the quantisation noise will become audible. An input signal that is too large will overload the AD converter and create an audible distortion of the signal.

With the Input Reg. potentiometer it is possible to raise the input signal -6 dB to 20 dB. This makes it possible to supply the DSP-NIR normally with input signals from 0.35 Vpp to 7 Vpp.

CW MODE

In CW Narrow mode, the user has the possibility to choose between 3 different CW filters with centre frequencies of 400 Hz, 600 Hz, and 750 Hz respectively with a bandwidth of 200 Hz. The desired centre frequency is chosen via the internal jumper JMP2 (see Figure 5 on page 40). The DSP-NIR is set at the factory to 750 Hz.

The CW Narrow filter can be used in conjunction with the receiver/transceiver's in-built CW filter in order to avoid channel interference. In CW operation, both the CW Narrow and SSB Narrow modes can be advantageously used. First set the DSP-NIR in SSB Narrow mode, and then tune in to the desired station. Then choose CW Narrow mode to obtain the largest possible selectivity.

The CW Narrow filters are designed as equi-ripple filters and make use of the decimation/interpolation technique. Because of this, the filters have very sharp edges, which makes it possible to selectively choose the desired station. The filters are linear phase, which gives a filtering with minimum "ringing".

DSP-NIR also has in PBT Narrow a variable passband filter that can also be used in CW operation (see page 25 for a more detailed description).

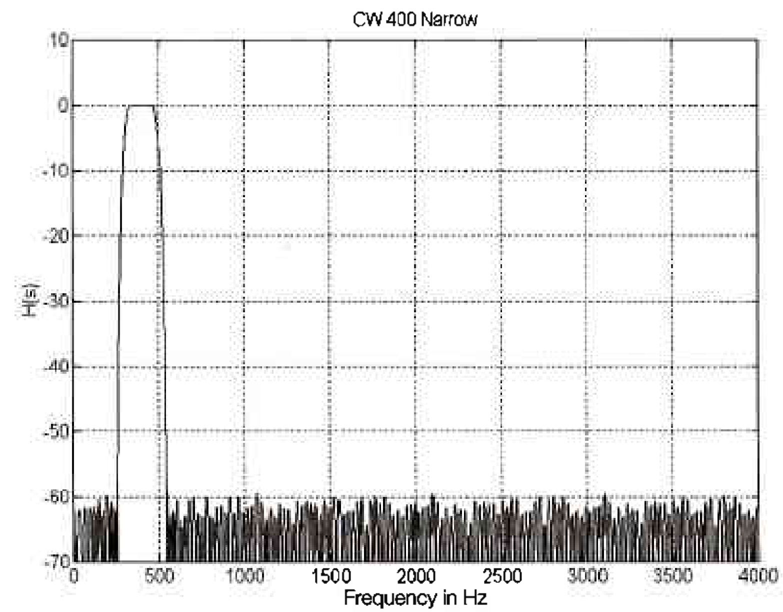
The CW 400 Narrow Filter

The passband is the -6 dB points and the stopband is the -60 dB points.

The CW 400 filter:	0	- 250 Hz	stopband
	300	- 500 Hz	passband
	545	- 4 KHz	stopband

Shape factor: 1.48

Delay: 12.8 mS

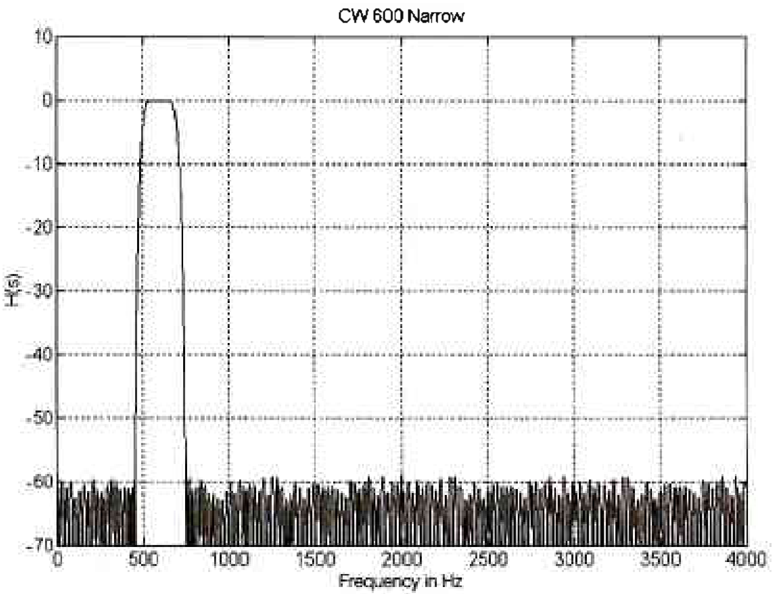


The CW 600 Narrow Filter

The CW 600 filter:	0 – 460 Hz	stopband
	500 – 750 Hz	passband
	750 – 4 KHz	stopband

Shape factor: 1.45

Delay: 12.8 mS

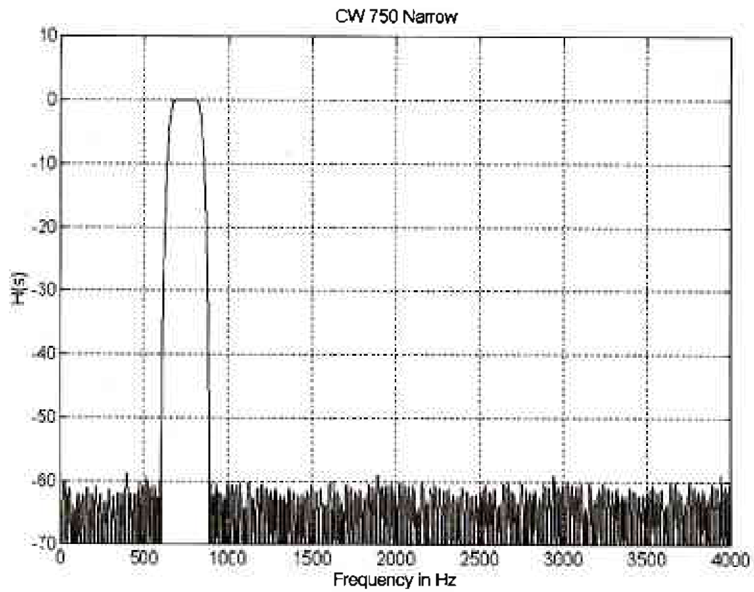


The CW 750 Narrow Filter

The CW 750 filter:	0 – 605 Hz	stopband
	650 – 850 Hz	passband
	890 – 4 KHz	stopband

Shape factor: 1.43

Delay: 12.8 mS



SSB Mode

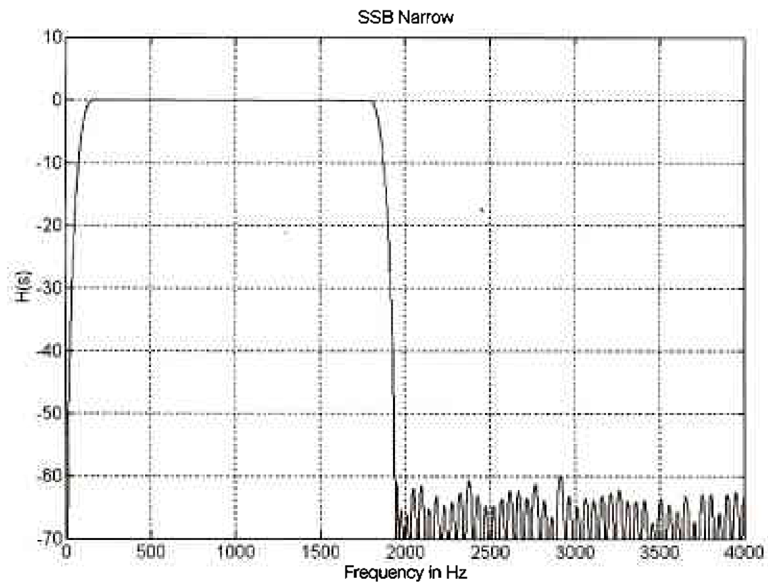
The intelligibility of a broadband SSB LF signal can be poor due to the S/N ratio. If the lowest and highest frequencies are removed with the help of a passband filter, the readability of the signal is increased. Both the SSB Wide and SSB Narrow digital filters have very sharp edges and low ripple. The filters can be used most advantageously if the receiver/transceiver being used does not have enough IF filtering.

The filters have linear phase, so group delay distortion is avoided. The filters are designed as equi-ripple filters with a lower cut-off frequency of 150 Hz, since a large part of a voice signal's intelligibility lies in the lower portion of the frequency spectrum.

The DSP-NIR also has, in the PBT Wide, a variable passband filter that can also be used in SSB voice operation (see page 25 for a more detailed description). If the user wants noise and tone reduction, he can use the NOTCH and Peak modes (see pages 21 and 23).

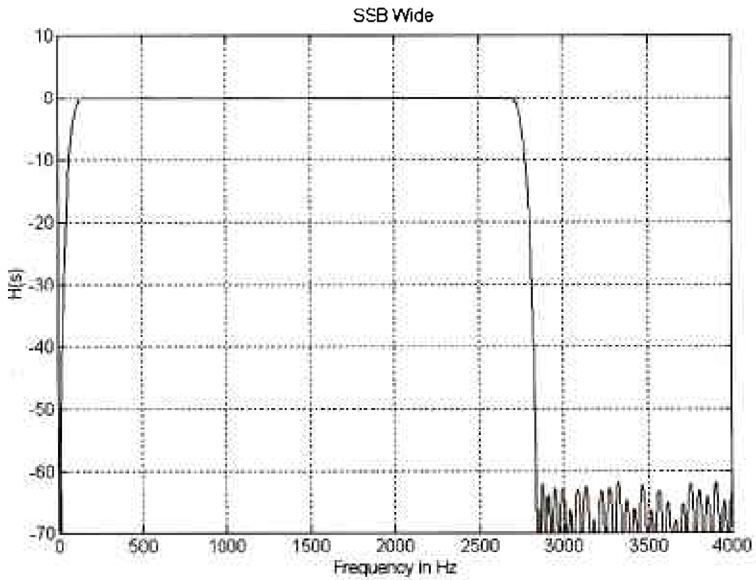
The SSB Narrow Filter:

The passband filter has a bandwidth of 1650 Hz.
The pass area is from 150 to 1800 Hz (cut-off frequency).
The stop area is from 1951 Hz to 4000 Hz (cut-off frequency).
Ripple in pass area: < 0.1 dB
Attenuation in stop area: > 60 dB
Shape Factor: 1.1 : 1
Delay: 10 mS



The SSB Wide Filter:

The passband filter has a bandwidth of 2550 Hz.
The pass area is from 150 to 2700 Hz (cut-off frequency).
The stop area is from 2843 Hz to 4000 Hz (cut-off frequency).
Ripple in pass area: < 0.1 dB
Attenuation in stop area: > 60 dB
Shape Factor: 1.06 : 1
Delay: 10 mS



NOTCH Mode:

NOTCH mode is used to remove spurious and interfering tones from the received LF signal. The notch filter attenuates a single tone up to 50 dB. The filter effectively removes up to 3-4 tones at the same time without degrading the voice signal's intelligibility.

Conventional filters use the frequency as the filtration criterion, as opposed to the NOTCH filter which uses the input signal's correlation as the filtration criterion. Correlation is a statistical value which indicates how often a given frequency component appears in the frequency spectrum. When a fixed pure sound is completely correlated and speech is less correlated, it is possible to distinguish the pure sounds from the speech signal. For the NOTCH filter to be effective against spurious and interfering tones, these tones should not oscillate too much in frequency, as the NOTCH filter will treat them as a voice signal. With multipatch such a condition could appear, and the NOTCH filter will not be effective.

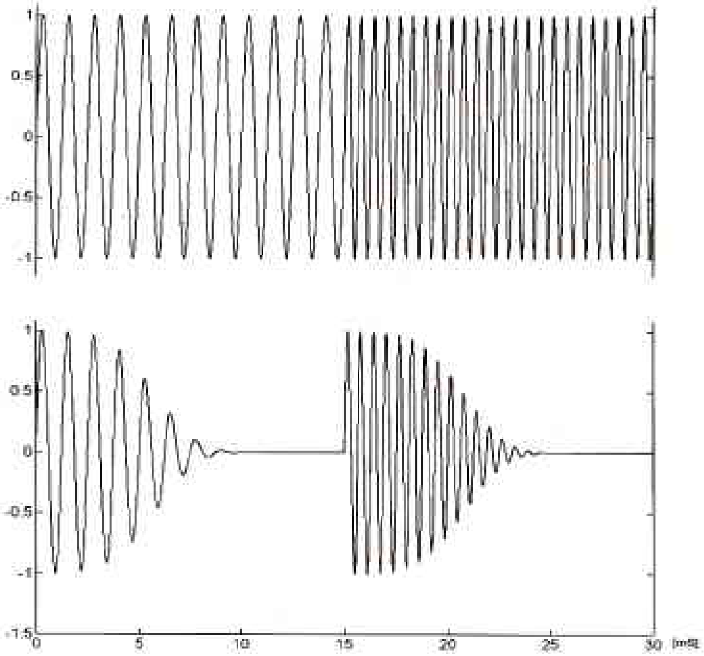
The input signal to the NOTCH filter is filtered by a conventional digital passband filter. The filter has a pass area from 150 Hz to 2700 Hz.

The NOTCH filter can either be used by itself, or be combined with the PEAK filter. The NOTCH filter cannot be used for CW, RTTY, SSTV and Packet operation.

The NOTCH filter:

A suddenly appearing tone is removed automatically without the user having to manually adjust the filter. The lowest graph below shows how effectively the notch filter filters out an 800 Hz and a 1600 Hz pure sound.

NOTCH reaction time: < 10 mS
Attenuation: Up to 50 dB depending on the tone's correlation.
Delay: 10 mS



PEAK Mode:

In PEAK mode atmospheric (white/pink) noise is attenuated by the filter dynamically forming passband filters around the voice signal. Ranges which do not contain any voice signal are effectively attenuated. In PEAK mode random noise is attenuated up to 20 dB.

The PEAK filter uses, just like the NOTCH filter, correlation as its filtration criterion. As random noise is completely uncorrelated and speech is more correlated, it is possible to distinguish random noise from the voice signal.

In PEAK adj. mode it is possible with the Filter Level button to vary the level of filtration. With Filter Level set to High, the LF output signal is slightly attenuated in a manner that makes it necessary to adjust the DSP-NIR's volume knob.

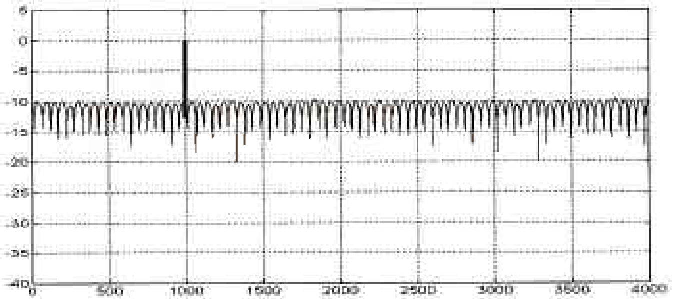
The input signal to the PEAK filter is filtered by a conventional digital passband filter. The filter has a pass area from 150 Hz to 2700 Hz.

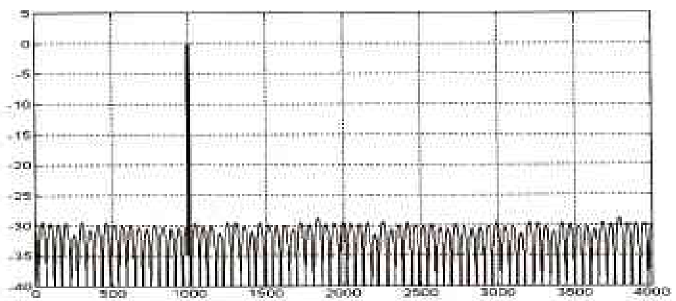
The PEAK filter can either be applied alone, or combined with the NOTCH filter. The PEAK filter can also be used to advantage in CW operation.

The PEAK Filter:

The graph shows a tone with random noise before and after peak filtering. It can be seen that the PEAK filter makes a passband filter around the correlated signal (the tone), while the rest of the frequency spectrum is attenuated.

PEAK reaction time: < 10 mS
Attenuation: Up to 20 dB
Delay: 10 mS





PBT Mode:

PBT stands for PassBand Tuning, which is to say that it is possible for the user himself to vary the passband filter's centre frequency in the area from 300 Hz to 3200 Hz. With the multi-setting mode switch one can choose between PBT Wide and PBT Narrow. The centre frequency can be varied in increments of 62.5 Hz with use of the Passband Tuning knob.

PBT Wide is mainly used with SSB reception, but can also be applied to diverse data communications objectives where a large bandwidth is required. When the Passband Tuning knob is turned to the Low position, the PBT filter will have an upper limiting frequency of 1350 Hz and a lower limiting frequency of zero Hz – where the filter will work as a low passband filter. With the Passband Tuning knob set to the High position, the PBT Wide filter will have an upper limiting frequency of 3650 Hz.

PBT Narrow is used with CW and data communication.

The filters in PBT mode are FIR filters with linear phase, where group delay distortion is avoided.

When the Passband Tuning knob is turned to change the filter's centre frequency, some short-lived noise will appear. This noise appears because the new filter has a certain rise time.

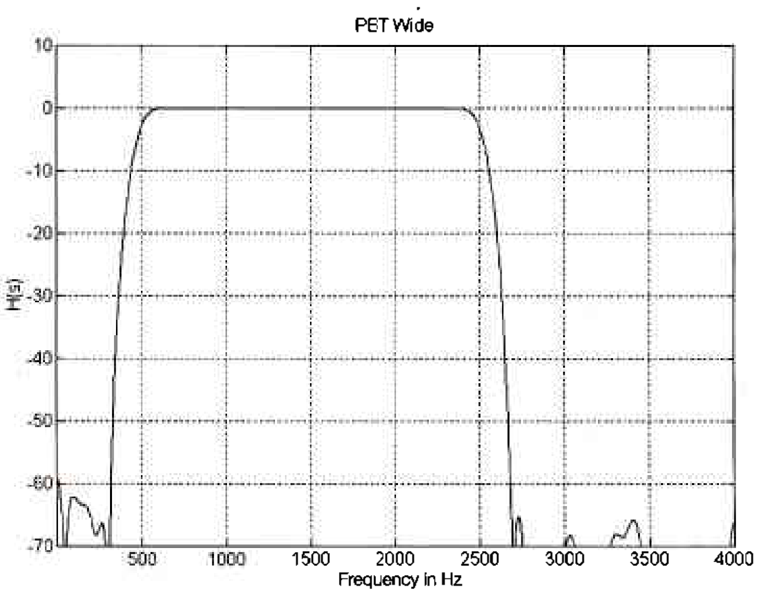
The PBT Wide Filter:

The filter has a bandwidth of 2100 Hz (-6dB point).

The centre frequency can be varied from 300 to 3200 Hz, however the DSP-NIR unit has an upper limiting frequency of 3650 Hz.

Attenuation in the stop area: > 60 dB
Delay: 8 mS

The PBT Wide filter with PBT=2000 Hz is shown below:

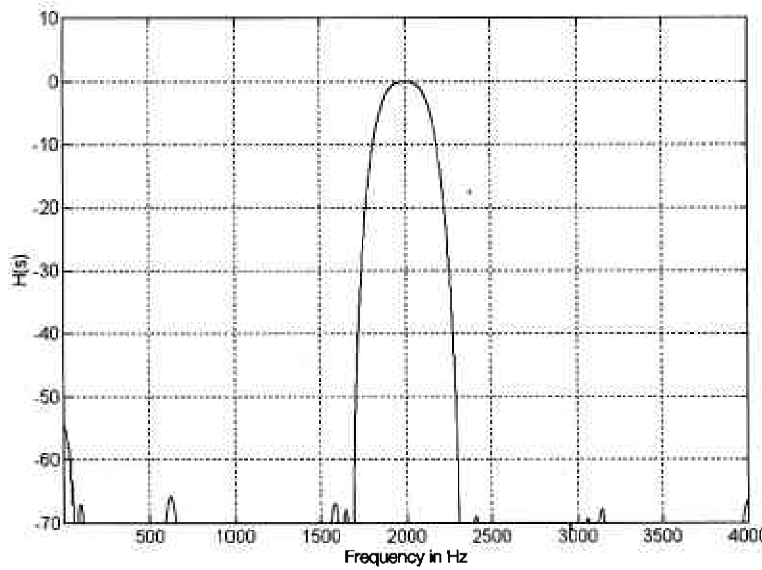


The PBT Narrow Filter:

The filter has a bandwidth of 300 Hz (-6 dB point).
The centre frequency can be varied from 300 to 3200 Hz.

Attenuation in stop area: > 60 dB
Delay: 8mS

The PBT Narrow filter with PBT=2000 Hz is shown below.

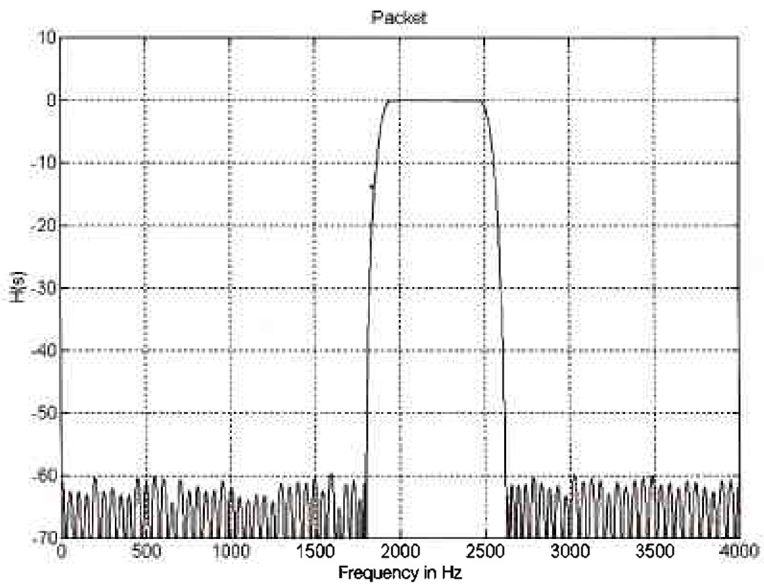


The Packet Filter:

The PACKET filter is used for HF packet communication. This filter has very sharp edges and low ripple in the pass area.

The filter has a BW of 540 Hz and a centre frequency of 2210 Hz. The pass area is from 1940 to 2480 Hz (cut-off frequency).
The stop areas are from 0 – 1811 Hz and from 2629 Hz to 4000 Hz.

Ripple in pass area:	< 0.1 dB
Attenuation in stop area:	> 60 dB
Shape Factor:	1.24 : 1
Delay:	10 mS



SSTV Mode:

The filter in SSTV mode supports HF slow-scan television (SSTV) format and Vertical Interval Signal (VIS) code format. The filter consists of a double passband filter for selective differentiation of picture information and the synchronisation pulses. The SSTV filter effectively attenuates interference tones outside the SSTV format's passband, and also reduces the random noise due to the reduced bandwidth.

The SSTV Filter:

The first passband filter has the following bands:

0	- 975 Hz	stopband (-60 dB point)
1050	- 1350 Hz	passband (-6 dB point)
1400	-	stopband (-40 dB point)

Shape Factor #1: 1.45 : 1

The second passband filter has the following bands:

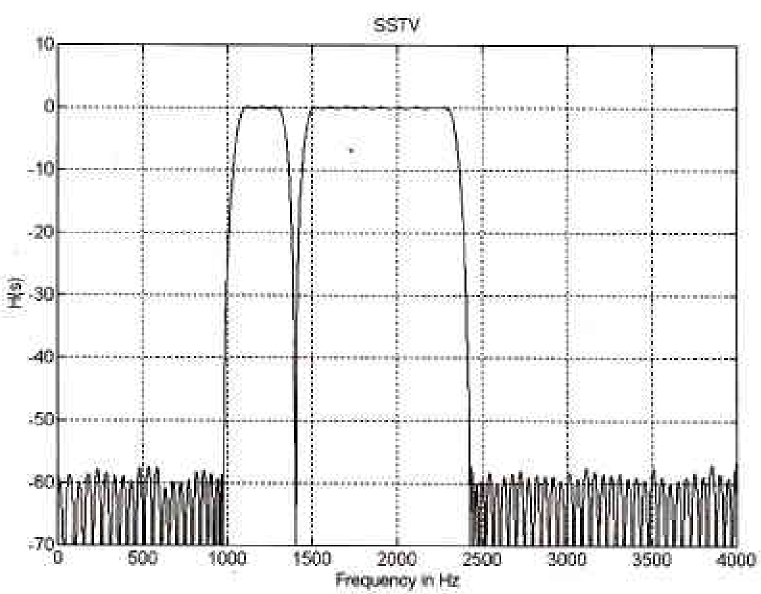
	- 1410 Hz	stopband (-40 dB point)
1460	- 2350 Hz	passband (-6 dB point)
2450	- 4000 Hz	stopband (-60 dB point)

Shape Factor #2: 1.17:1

Ripple in pass area: < 0.1 dB

Attenuation in stop area: > 55 dB

Delay: 10 mS



The RTTY Filter

The RTTY filter works in accordance with the standardised RTTY Mark and Space tone specifications.

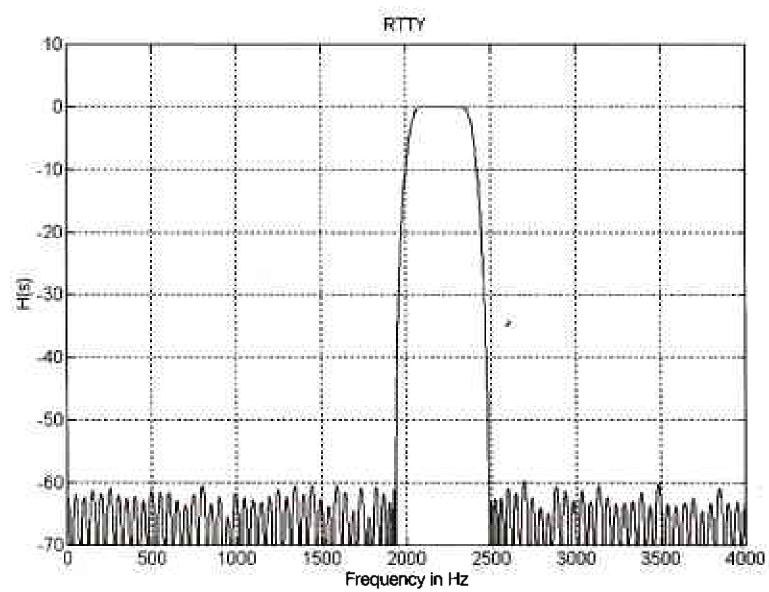
The filter's centre frequency is 2210 Hz and the bandwidth is 270 Hz. The passband filter has a bandwidth of 270 Hz - from 2075 to 2345 Hz (cut-off frequency). The stop area is from 0 – 1936 Hz and from 2492 Hz to 4000 Hz (cut-off frequency).

Ripple in pass area: < 0.1 dB

Attenuation in stop area: > 60 dB

Shape Factor: 1.43 : 1

Delay: 10 mS



SPECIFICATIONS

LF Input

Input Level	0.35 Vpp – 7 Vpp adjustable via potentiometer.
Input Impedance	22 Ω or 6 K Ω selectable via internal jumper.

LF Output

Outputs	Headset output, Line output and Loudspeaker output
LF Amplifier	1.8 Watt with 8 Ω /3.2 Ω with 4 Ω speaker impedance
Distortion	< 1% at 1 kHz.

Filters

CW Filter Bandwidth	200 Hz
CW Filter Centre Frequency	400 Hz, 600 Hz or 750 Hz selectable by internal jumper.
CW Attenuation in Stop Area	> 60 dB attenuation.
CW Filter Type	FIR Linear Phase, Passband filter.
SSB Filter Passband	150 Hz – 1800 Hz (SSB N) 150 Hz – 2700 Hz (SSB W)
SSB Shape Factor	SSB N: 1.1 : 1; SSB W: 1.1 : 1 > 60 dB attenuation
SSB Filter Type	FIR Linear Phase, Passband Filter
Packet Filter Frequency Response	540 Hz Bandwidth, 2210 Hz centre freq.
Packet Shape Factor	1.24 : 1, > 60 dB attenuation
Packet Filter Type	FIR Linear Phase, Passband Filter
SSTV Filter Passband	1050 Hz – 1350 Hz and 1460 Hz – 2350 Hz
SSTV Shape Factor	BP1: 1.45 : 1; BP2: 1.17 : 1, > 55 dB attenuation
SSTV Filter Type	FIR Linear Phase, Double Passband Filter
RTTY Filter Frequency Area	270 Hz bandwidth, 2210 centre frequency
RTTY Shape Factor	1.43 : 1, > 60 dB attenuation
RTTY Filter Type	FIR Linear Phase, Passband Filter

NOTCH Filter Response	150 Hz – 2700 Hz
NOTCH Attenuation of 1 KHz tone	Up to 50 dB – depends on the characteristics of the input signal.
NOTCH Reactions Time	< 10 mS
NOTCH Filter Type	Adaptive
PEAK Filter Area	150 Hz – 2700 Hz
PEAK Attenuation of Random Noise	10 dB – 20 dB, the attenuation can be varied in PEAK adj. mode.
PEAK Reaction Time	< 10 mSec
PEAK Filter Type	Adaptive
PBT Filter Bandwidth	300 Hz (PBT N) or 2100 Hz (PBT W), selectable
PBT Filter Centre Frequency	Variable in the area 300 to 3200 Hz in 62.5 Hz increments.
PBT Attenuation	> 60 dB attenuation
Combinations	NOTCH + PEAK

Frontpanel	Volume/Power On/Off, 12 position rotary switch, Filter Level Potentiometer, Bypass Switch, AGC On/Off, Headset Output 6.35 mm Phone Jack -- Stereo, Power LED (yellow), Normal LED (green), Overflow LED (red).
Rearpanel	Audio Input Phone Jack, Audio Output Phone Jack, Line Output Phone Jack, Input reg. (potentiometer), DC IN 5.5 mm DC Jack.
Size	60 mm x 193 mm x 155 mm (HxWxD), Weight 1.4 kg.
Power	11 – 15 Volt DC / 500 mA

Fault Finding Help:
Short descriptions of possible faults and fault symptoms are given in this section:

Fault: **"POWER" LED does not light when the instrument is turned on.**
Possible Cause: Check the power connection to the DSP-NIR, the DC plug should be properly polarised.

Fault: **"POWER" LED lights up but the instrument does not work.**
Possible Cause: The DC power voltage is too low, check that the voltage from the power supply is correct. Check that the power supply can at a minimum deliver 500 mA. Try to turn the instrument off and after waiting at least 10 seconds, turn it on again via the DSP-NIR's volume knob.

Fault: **"Normal" LED and "Overload" LED are constantly lit.**
Possible Cause: Reduce the audio input level either with Input Reg. or with the receiver's volume knob.

Fault: **"Normal" LED and "Overload" LED do not light up.**
Possible Cause: Use BYPASS to check if a signal is coming from the receiver. Turn up the audio input either with the Input Reg. or with the receiver's volume knob.

DANMIKE

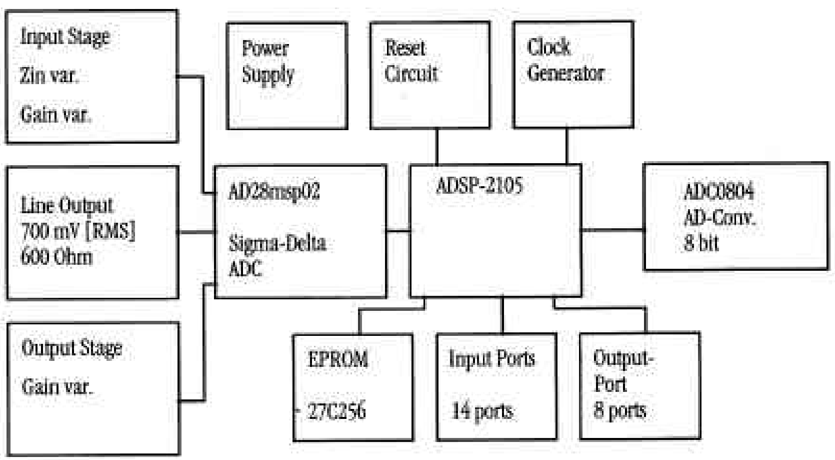
Fault: **Sound from speaker output, but not from headset output.**
Possible Cause: Check the headset and the connection to the phono port. Check if the phone jack is a stereo jack.

Fault: **Sound from the headset output, but not from the speaker output.**
Possible Cause: Check the speaker and the connection to the speaker port. Check that the phono jack is out when the speaker port is used. Check that the DSP-NIR's volume knob is turned up.

Fault: **The Line Output does not work.**
Possible Cause: Check the line connection. Use the BYPASS switch to check the equipment being interfaced. Check that the equipment being interfaced does not load the Line Output with less than 600 Ω.

On page 39 a complete diagram of the DSP-NIR is shown. A short description will be given here of that part of the system which a ham operator would be able to troubleshoot on his own. Descriptions of the input stage, the line output, the output stage and the Power Supply are given.

The DSP-NIR System



The DSP-NIR mainly consists of a conventional digital signal processing system. By changing the software in an interchangeable EPROM (U2), the system can be updated/improved so that the newest types of digital filters can always be used in the DSP-NIR unit.

Input Stage:
The Audio Input signal is connected to the J2 connector, which is the input to a low-pass filter. The filter consists of an active filter constructed from the op-amp U8A. The filter also ensures that the input signal is DC-shifted by +2.5VDC. This DC reference voltage is delivered from the ADC converter to U8A pin 3. The amplification of the op-amp (U8A) can be varied by P1. The filter's output furnishes the input voltage to the analogue to digital converter (U3).

Output Stage:
After the signal has been processed by the DSP and converted by the digital to analogue converter (U3), the output signal is low-pass filtered. The low-pass filter is constructed around the op-amp U8B where R12 and C24 determine the cut-off frequency. The output from the low-pass filter is connected to the input of the LF amplifier via C23. The LF amplifier consists of a TDA2003 (U9) and some passive components. The output signal from the LF amplifier goes through the BYPASS switch (S3) to the speaker output and the phone output.

Line Output:
From the output of the low-pass filter, U8B pin 7, the signal is taken through a coupling capacitor to the input to a buffer (U15A). From the output of this buffer, the line output signal runs through the BYPASS switch (S3) before it goes to the line output jack (J5).

Power Supply:
The power supply has separate +5 VDC analogue and +5 VDC digital voltages. The voltages are regulated by U10 (7805CT) and U11(7805LN). The regulators are stabilised by a 2200 uF electrolytic that sits on the input to the regulators. C1, L1 and C2 form a filter that impedes HF signal break-through. The diode D1 protects against incorrect polarisation of the power plug and R1 prevents the regulators from receiving too large a voltage.

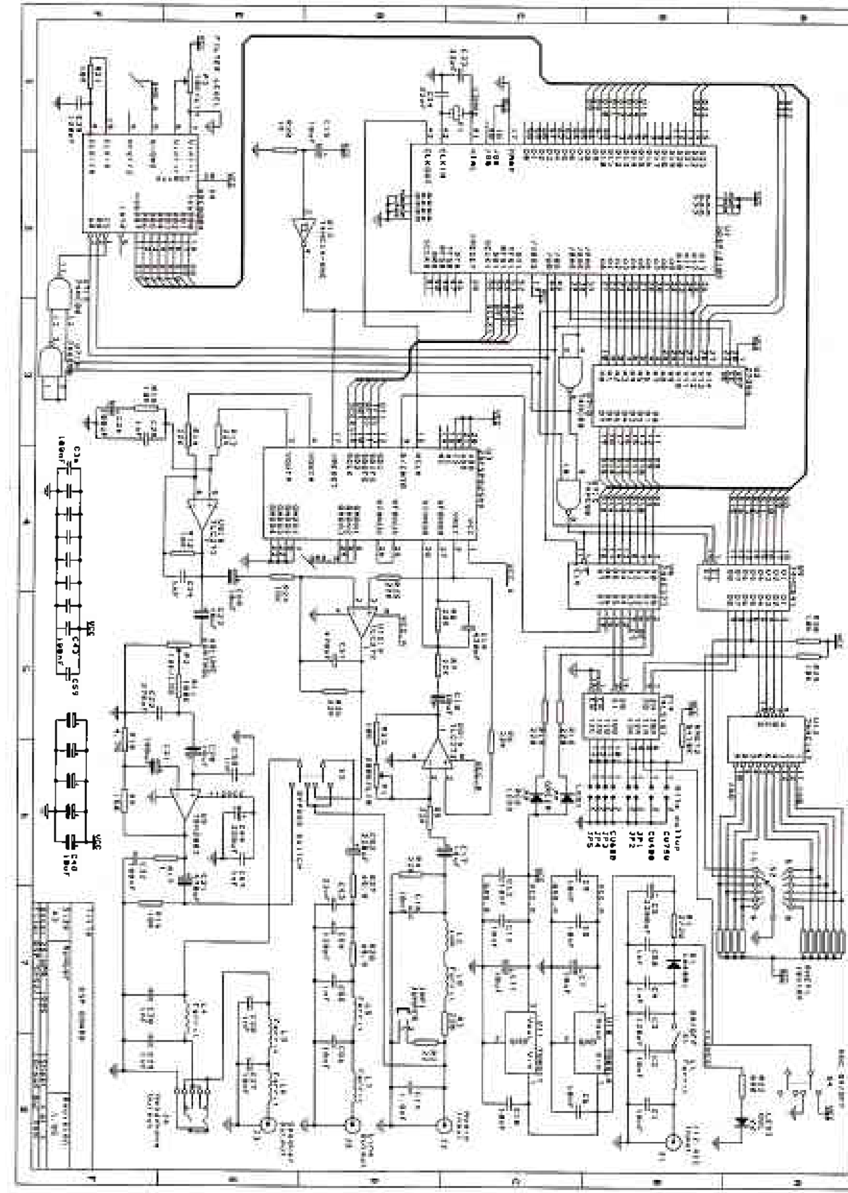
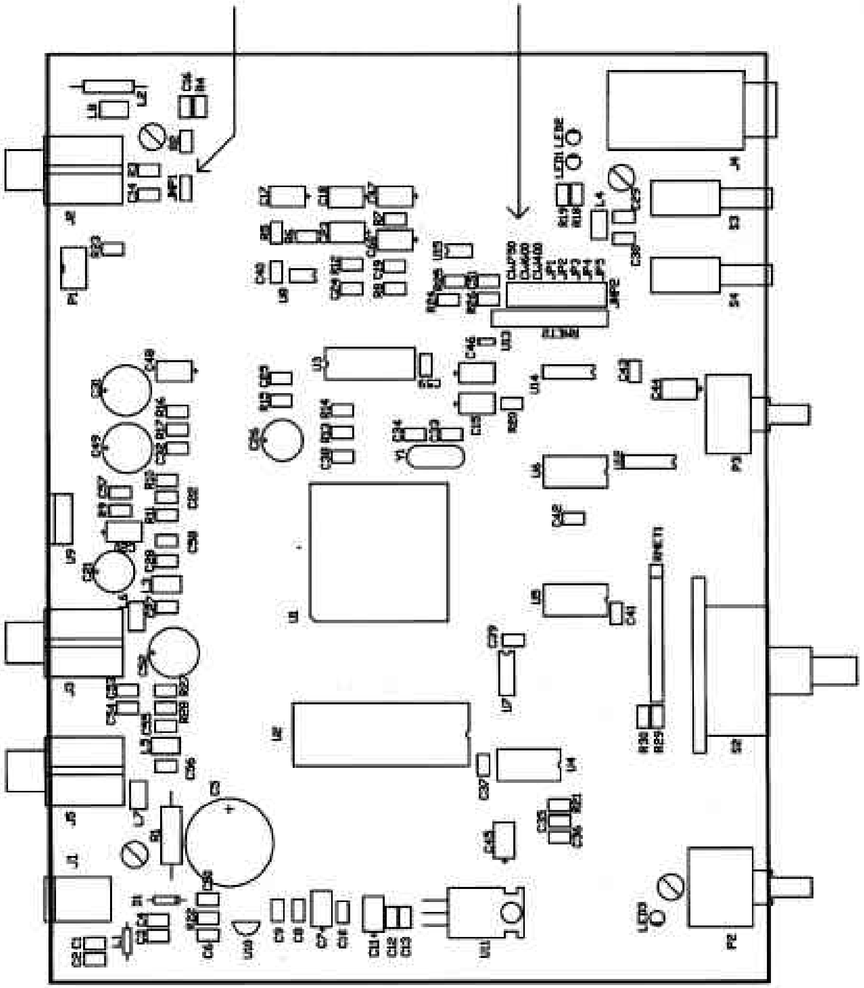


Figure 5.

Use JMP1 to choose between 22 Ω (closed) and 6 KΩ (open) input impedances.
JMP2 is set either CW750, CW600 eller CW400.



DANMIKE DSP NIR

* **Digital Signal Processing Noise and Interference Reduction Unit**
* **Microprocessor Controlled (Digital Signal Processor)**
* **Latest 16 bit DSP technology**
* **SSB, CW, Packet, SSTV, RTTY, Notch, PBT, and Peak Filters**
* **14 Different Filter Functions**
* **Automatic multi-tone notch**
* **Digital linear phase filters with up to 60 dB attenuation**
* **AGC function (Automatic Gain Control)* Passband Tuning (PBT) with 300Hz or 2100 Hz bandwidth**
* **Variable setting of peak filter level**
* **Integrated AF amplifier with 3.2 watts @ 4 ohms**
* **Bypass Function**
* **Easy connection**
* **Improves reception considerably and removes noise**
* **Extends your active working time**
* **13.8VDC operation**

DSP-NIR SPECIFICATIONS:

INPUT INPUT LEVEL: 0.35 Vpp - 7 Vpp adjustable via potentiometer
INPUT IMPEDANCE: 22Ohms or 6 K selectable via internal jumper

A.F. OUTPUT OUTPUTS: Headphone, Line, and Speaker outputs
A.F. AMPLIFIER: 1.8 watts with 8 ohm / 3.2 watts with 4 ohms imp.
DISTORTION: <1% at 1kHz

FILTERS CW BANDWIDTH: 200 Hz CW FILTER CENTER FREQ. 400 Hz, 600 Hz or 750 Hz selectable by internal jumper.
CW ATTEN. IN STOP AREA: > 60dB attenuation.
CW FILTER TYPE: FIR Linear Phase, Passband filter

SSB FILTER PASSBAND: 150 Hz - 1800 Hz.(SSB N) ... 150 Hz - 2700 Hz (SSB W)
SSB SHAPE FACTOR: SSB N: 1.1 : 1, SSB W: 1.1 : 1 > 60 dB attenuation
SSB FILTER TYPE: FIR Linear Phase, Passband Filter

PACKET FILTER RESPONSE: 540 Hz Bandwidth, 2210 Hz center freq.
PACKET SHAPE FACTOR: 1.24 : 1, > 60dB attenuation
PACKET FILTER TYPE: FIR Linear Phase Passband Filter

SSTV FILTER PASSBAND: 1050 Hz. - 1350 Hz and 1460Hz - 2350 Hz
SSTV SHAPE FACTOR BP1: 1.45 : 1, BP2: 1.17 : 1, > 55 dB attenuation
SSTV FILTER TYPE FIR: Linear Phase, Double passband Filter

RTTY FILTER FREQ. AREA: 270 Hz bw, 2210 Hz center frequency
RTTY SHAPE FACTOR: 1.43 : 1, > 60dB attenuation
RTTY FILTER TYPE: FIR Linear Phase, Passband Filter

NOTCH FILTER RESPONSE: 150 Hz - 2700 Hz.
NOTCH ATTEN. 1 KHz TONE: Up to 50 dB - depends on the variable characteristics of the input signal
NOTCH REACTIONS TIME: < 10 mS
NOTCH FILTER TYPE: Adaptive

PBT FILTER BANDWIDTH: 300 Hz (BT N) or 2100 Hz (PBT W) Selectable
PBT FILTER CENTER FREQ.: Variable 300 to 3200 Hz in 62.5 Hz increments
PBT ATTENUATION: > 60 dB

COMBINATIONS: Notch + Peak

FRONT PANEL: Volume/Power On/Off, 12 position rotary switch, Filter Level Potentiometer, Bypass Switch, AGC On/Off, Headphone Output - Stereo, Power LED, Normal LED, Overflow LED
REAR PANEL: Audio Input Phone Jack, Audio Output Phone Jack, Line Output, Input Level regulator potentiometer, PTT, DC IN
SIZE 60 mm x 193mm x 155 mm (HxWxD) Weight 1.4 kg.
POWER 11 - 15 VDC / 500 mA

DSP-NIR « DANMIKE » DE PROCOM

PROCOM est une société danoise qui fabrique du matériel, essentiellement des antennes et accessoires de radio. Récemment, elle a mis sur le marché un filtre DSP utilisable par les radioamateurs.

Denis BONOMO, F6GKQ

Le filtre DSP de PROCOM n'est pas sans rappeler certains modèles déjà présentés dans *MEGAHERTZ MAGAZINE*. Solidement enfermé dans un lourd boîtier métallique peint en noir, d'une robustesse exemplaire, la produit m'a rappelé ceux fabriqués aux USA par JPS. Ce parallèle étant établi, voyons à quoi ressemble le «DANMIKE» de PROCOM.

UNE BATTERIE DE FILTRES

En deux mots, rappelons que la technique DSP (Digital Signal Processing) permet de traiter numériquement un signal. Plus la fréquence du signal est élevée (vidéo, FI, etc), plus le traitement sera délicat puisqu'il faudra utiliser un échantillonnage à cadence rapide. Avec les signaux audio, le problème est un peu moins ardu ce qui explique la mise sur le marché, depuis quelques années, de ces filtres DSP. Leur avantage par rapport aux filtres purement analogiques, est qu'ils nécessitent une faible mise au point, que l'on peut adapter leur courbe de réponse à tous les cas de figures (ou presque). En effet, tout le secret du fonctionnement réside dans le logiciel qui les anime. Dans le cas du DSP DANMIKE, les flancs des filtres obtenus sont raides (facteur de forme assez édifiant !) et l'ondulation résiduelle quasi inexistante. Les lettres NIR qui suivent DSP, sont là pour indiquer qu'il s'agit d'un réducteur de bruit et d'interférences (Noise and Interference Reduction).

Au cœur des DSP se trouve un circuit spécialisé. PROCOM a adopté un processeur de signal de chez Analog Devices. Le logiciel tient dans une EPROM qui lui est associée. Le reste de la circuiterie est quasiment «banal» avec, entre autres, des conversions analogiques digitales et des adaptations de niveaux. Les composants sont essentiellement des CMS. Malgré cela, remarquera

Dans un boîtier robuste, le DSP-NIR de Procom.

que PROCOM n'a pas recherché à miniaturiser à l'extrême...

Il faut souligner que, dans le cas du DANMIKE, PROCOM a choisi d'ajouter un amplificateur audio délivrant une puissance BF voisine de 3 W (sous 4 Ω). Le filtre DSP vient donc naturellement s'insérer dans la chaîne BF, en sortie du récepteur (ou du transceiver). Comme tous les filtres BF, il a un défaut : il ne pourra pas grand chose contre les signaux qui perturbent fortement les circuits HF du récepteur et provoquent son «blocage».

LA MISE DANS LE CIRCUIT

Le DANMIKE est livré avec les fiches BF mais sans les cordons. Vous devrez donc commencer par confectionner les câbles de liaison pour l'alimentation en 12 V, l'entrée du signal audio (dont le niveau peut être sélectionné par un cavalier, dans le DSP et ajusté par l'extérieur avec un petit potentiomètre), la sortie vers le haut-parleur supplémentaire. La sortie ligne pourra être reliée à votre TNC ou interface de décodage, améliorant sensiblement les performances (surtout en SSTV), comme on le verra plus loin.

Le simple examen des branchements montre quels sont les avantages de ce DSP par rapport à certains de ses concurrents : il permet de maîtriser le niveau BF. Soit on l'utilise à partir d'une sortie bas niveau, soit à partir d'une sortie HP. Mais dans tous les cas, on peut ajuster facilement le niveau de signal injecté dans le DSP, et c'est indispensable si l'on veut profiter de l'efficacité du filtre. Quant au contrôle de volume sur le DSP, il sera fort apprécié par l'utilisateur. La touche BYPASS permet de passer ou non par le DSP. Là encore, l'équilibre des niveaux BF est un atout. Si on ne prend pas le temps d'ajuster le petit réglage placé à l'arrière du DSP, on risque «d'en prendre plein les oreilles». Cette abondance de détails pour montrer les avantages du DANMIKE. Quant aux connecteurs, ils sont tous au format «RCA» ou «CINCH». Vos liaisons BF seront impérativement en câble blindé... Fallait-il le préciser ?

Le panneau avant du DSP supporte les commandes suivantes :

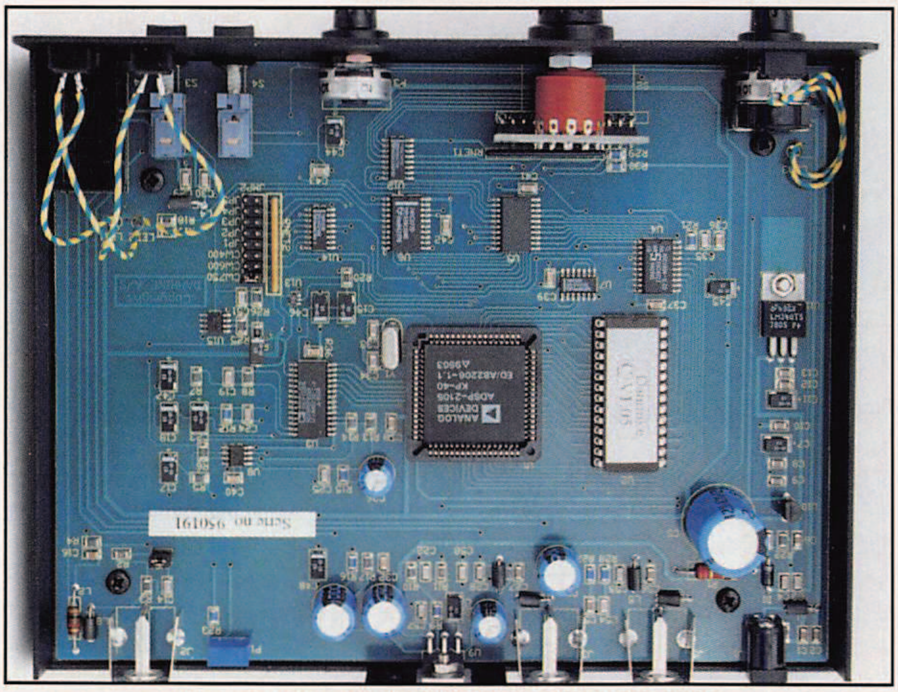
- potentiomètre de réglage du volume (avec inter marche-arrêt). La mise sous tension est indiquée par une LED jaune.
- commutateur rotatif à 12 positions pour le choix du mode.
- potentiomètre pour ajuster le passband tuning ou le niveau du filtre.
- touche de mise en/hors service du CAG du filtre.
- touche bypass, pour «court-circuiter» le filtre.
- prise 6,5 mm pour un casque.
- deux LED, une rouge, une verte, pour optimiser le niveau d'entrée injecté au DSP.

PREMIERES ECOUTES

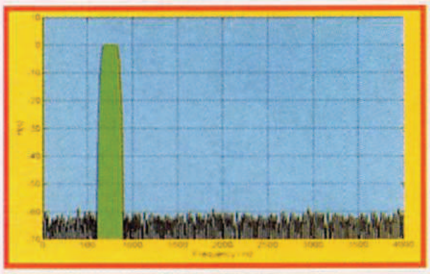
On n'insistera jamais assez sur ce point : l'utilité d'un DSP est inversement proportionnelle à la qualité du récepteur. C'est donc avec un transceiver d'entrée de gamme que le DSP sera le plus utile. En effet, si votre appareil est doté, d'origine, de tous les raffinements de la technique, le DSP ne sera probablement pas indispensable. Pour essayer le DSP, j'ai donc utilisé mon matériel «comme s'il était nu», sans mettre en œuvre les filtrages et dispositifs anti-interférences dont il dispose.

Le réglage du niveau injecté au DSP est important. Il est grandement facilité par la présence des LED «NORMAL» et «OVERLOAD». La LED verte «NORMAL» devra s'allumer au rythme de la modulation. La LED rouge «OVERLOAD» ne devra le faire que très occasionnellement (sur des pointes). Les premiers essais ont été faits en BLU (SSB). Le DANMIKE propose plusieurs solutions dans ce mode : SSB W (large), SSB N (étroite) qui agissent un peu comme si vous commutiez un filtre FI... sans toutefois présenter les avantages de ce dernier. Déjà, la position étroite élimine les «moustaches» des stations un peu trop proches. Autre arme fournie par le DSP, la position PBT W (Pass Band Tuning) où l'on peut ajuster la fréquence centrale du filtre dans une plage de 300 à 3200 Hz, par pas (oui, c'est du numérique, alors on entend un petit clic quand on tourne le potentiomètre du PBT, à chaque incrément du filtre).

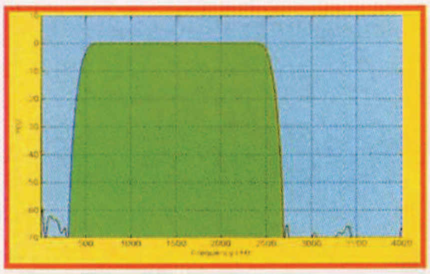
Si vous devez lutter contre une porteuse agressive, rien de tel que le filtre NOTCH. Dans cette position, le DSP élimine automatiquement, en quelques millisecondes, l'excité craignos qui «tune» sur antenne.



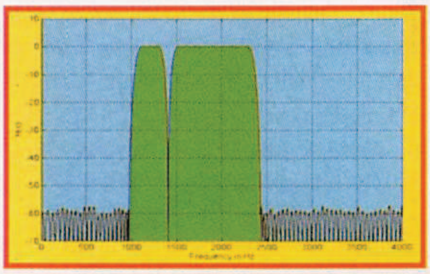
Des composants CMS et un montage très aéré.



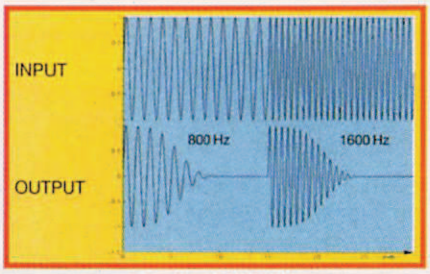
Mode CW N.



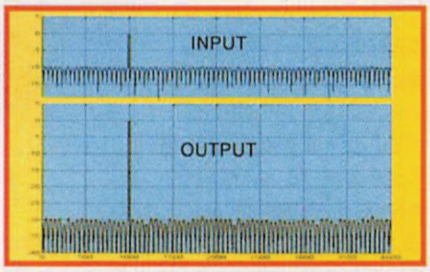
Mode PBT.



Mode SSTV.



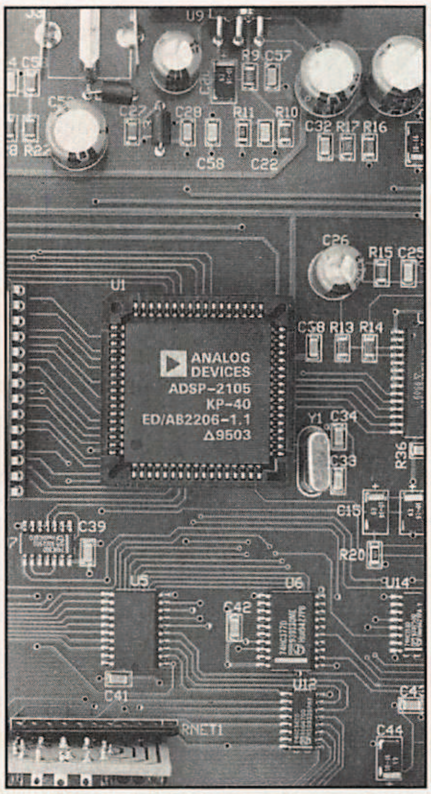
Mode NOTCH.



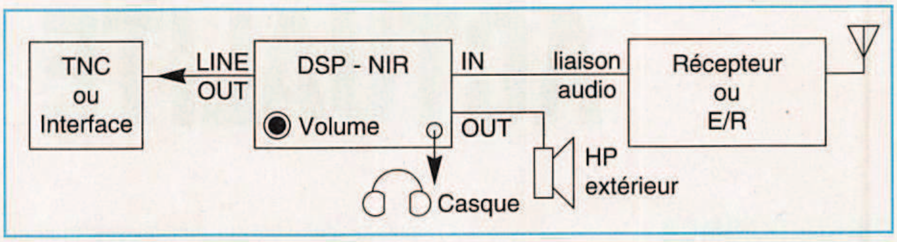
Mode PEAK.

Et le DANMIKE sait courir plusieurs lièvres à la fois, puisqu'il peut ainsi éliminer jusqu'à 4 porteuses simultanément, entre 150 et 2700 Hz...

La position «PEAK» est remarquable pour éliminer les bruits de fond gênants : bruit



Au cœur du DSP, le processeur « Analog Devices » et son EPROM.



Les branchements du DSP.

atmosphérique (qu'il soit blanc ou rose nous dit la doc) ou même, bruit généré par des lignes électriques. L'effet obtenu est spectaculaire, et l'on pardonnera la modification de signal qui en résulte (on a l'impression d'écouter de la BLU avec une petite rotation de phase due à la propagation). En mode PEAK ADJ, il est possible de régler (avec le potentiomètre voisin), le seuil du filtre. Ce mode PEAK peut être combiné au filtre NOTCH dans la position NT + PEAK.

L'opérateur CW disposera, lui aussi, de possibilités de filtrage. CW N est le filtre élémentaire, qui sans remplacer un bon filtre à quartz placé dans la FI, pallie son absence. La fréquence centrale est réglable en déplaçant un petit cavalier à l'intérieur du boîtier (400, 600 ou 750 Hz). La bande passante est de 200 Hz avec un effet de «cloche» pratiquement absent. Mais on peut aussi, en CW, utiliser les services du PBT (position PBT N) ou du filtre de crête (PEAK).

Au passage, il convient de dire un mot sur la présence de la touche AGC. Quand elle est enfoncée, la commande automatique de gain (CAG) est mise en service. Il en résulte un renforcement de la BF et une plus grande dynamique des signaux. En contrepartie, on observe une légère remontée du bruit.

Le DANMIKE sera aussi très apprécié par les amateurs de SSTV (ou FAX), packet et RTTY puisqu'une position est prévue pour chacun de ces modes. Nous avons conduit quelques essais en SSTV : il faut reconnaître que les résultats sont assez surprenants. Le DSP aide à lutter contre l'indiscipline qui est née autour de la fréquence unique de 14.230 MHz, où beaucoup se plaisent à démarrer en émission sans se soucier des liaisons déjà établies. Grâce au filtre, on peut éviter de voir une belle image dégradée par les éclaboussures d'une station «phone» voisine. La bande passante est taillée pour la SSTV (voir courbe) et ne démarre qu'à 1000 Hz. En packet, la fréquence centrale du filtre est ajustée sur 2210 Hz, avec une bande passante de 500 Hz. En RTTY, le DSP est réglé pour les tonalités «hautes» (fréquence centrale 2210 Hz, BP 270 Hz) ce qui pourra poser quelques problèmes aux stations européennes si l'on veut, en même temps, profiter du filtre étroit du transceiver...

Globalement, après quelques jours d'utilisation dans les différents modes, il est indéniable que le DANMIKE apporte un sérieux coup de pouce à la réception, même si l'on dispose d'un récepteur déjà bien équipé. Toutefois, cela ne se fait pas sans quelques défauts mineurs qu'une oreille exercée pourra déceler : des petits «clics» sur les parasites, un certain bruit de fond (apporté par l'horloge ?) dans l'ampli BF et les claquements déjà mentionnés plus haut, quand on tourne le potentiomètre PBT. Attention, si le niveau d'entrée est mal ajusté, on perd le signal, dans certains cas, en bougeant le commutateur de mode, quand la touche AGC est sortie. Une fonction Noise Blanker (NB) aurait été la bienvenue. Néanmoins, le DANMIKE a été conçu sans oublier les exigences de l'utilisateur ni son confort (réglages des niveaux en entrée ET sortie), ce qui le place avantageusement devant tous ceux que j'ai pu tester à ce jour. L'appareil devrait être proposé par les revendeurs PROCOM à un prix situé entre 2500 et 3000 FF.

Vous pourrez le découvrir lors du salon d'Auxerre.

CARACTERISTIQUES CONSTRUCTEUR

Niveau d'entrée	: 0.35 à 7 V c/c
Puissance de sortie	: 1,8 (8 Ω) ou 3,2 (4 Ω) watts
Dimensions / poids	: 60 x 193 x 155 mm, 1.4 kg
Alimentation	: 11 - 15 V, 500 mA
Filtre CW	: 200 Hz (sur 400, 600 ou 750 Hz)
Filtre SSB-N	: 150 à 1800 Hz - FF 1.1:1
Filtre SSB-W	: 150 à 2700 Hz - FF 1.06 :1
PBT - N et W	: 300 Hz et 2100 Hz dans plage 300 à 3200 Hz
NOTCH	: 150 à 2700 Hz, 50 dB en <10 mS
SSTV	: 1050-1350 Hz et 1460-2350 Hz, avec FF 1.45:1 et 1.17:1
RTTY	: centré 2210 Hz, BP 270 Hz - FF 1.43:1
Packet	: centré 2210 Hz, BP 540 Hz - FF 1.24:1

(FF : facteur de forme ; BP : bande passante)