

GBPPR 'Zine



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"We need to do this every day of the week, and just really brainwash people into thinking about guns in a vastly different way."

--- 1995 quote from the now current Attorney General Eric Holder discussing his plans to decrease the violence in Washington D.C.

Note that Eric Holder is proponent of Derrick Bell's "Critical Race Theory," so instead of just banning "sons of Obama" from carrying firearms (or sending them all back to Africa), he wants to brainwash YOU against owning an inanimate object. Change!

(www.theblaze.com/stories/holder-in-1995-we-must-brainwash-people-against-guns)

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Privacy/Data Products Interface – AUTOPLEX System 100

AT&T PRACTICE
Standard

AT&T 231-290-621
Issue 1, November 1985

PRIVACY/DATA PRODUCTS INTERFACE FEATURE DOCUMENT 1A ESS™ SWITCH AUTOPLEX™ SYSTEM 100

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1.06 Cellular system providers may offer mobile customers privacy/data service for the radio segment of all calls on a subscription basis. A separate DN (directory number) is required.

1.07 Customers may make private-voice/data service calls between CTS 1620 privacy/data accessories without involving the switching office feature.

INTERACTIONS

1.08 Customers involved in a private-voice/data service call should consider the following features:

(a) **Three-Way Calling:** A customer involved in a private-voice/data service call should not add a third party.

(b) **Call Waiting:** While not prohibited, subscribers should be advised not to also subscribe to the Call Waiting feature. Call waiting tones can produce audible noise during private-voice service calls. During data service calls, call waiting tones interrupt carrier resulting in a loss of data.

Note: Both of these features operate normally in the clear-voice mode.

1.09 Roamer II service uses a temporary local DN for roaming that allows customers to subscribe to private-voice/data service in a foreign area.

2. PRODUCT DESCRIPTION

2.01 The product line consists of the following types of equipment:

- CTS 1620 Privacy/Data Accessory (1AMAS)
- SRE (Switch Resident Equipment) consisting of SCUs (SRE Channel Units) and associated equipment.

The CTS 1620s and SCUs communicate in pairs, like data sets, over a voiceband transmission path using a proprietary protocol. All units can provide both private-voice and data service.

2.02 The product line also consists of two modules, an FKM (fixed key module) and a CKM (configuration and key module).

MOBILE RESIDENT UNIT

2.03 The CTS 1620 resides in the cellular customer's vehicle, either in the trunk or in the passenger compartment. It is connected to the cellular telephone at the interface between the control unit and the transceiver unit. For data service, DTE (data terminal equipment) is connected to the CTS 1620 via an EIA (Electronic Industries Association) RS-232C cable.

KEY MODULES

2.04 Encryption keys are required to start private-voice and data service calls. Each CTS 1620 can store up to 64 encryption key records in an EEROM (electrically erasable read only memory) which retains data when power is off. Each encryption key record consists of a master key and a KIN (key identification number).

2.05 At the beginning of a private-voice or data call, the units exchange KINs and determine which, if any, prestored encryption key to use. If the CTS 1620 does not have a common prestored encryption key, they use the default encryption key known to all units. Although this results in reduced security, the channel is nevertheless protected from casual listening by a third party.

A. Configuration and Key Module

2.06 A CKM is supplied with each CTS 1620. It consists of a mode switch, 16 configuration switches, and an EEROM chip containing one encryption key record. It has two modes, as follows, controlled by the mode switch.

(a) The configuration mode is used to feed the default configuration selections into the CTS 1620.

(b) The key mode is used to assign (write) a random encryption key record into the CKM, then feed it into other CTS 1620s for end-to-end calls.

2.07 The KIN in a CKM consists of a 1 in the first bit followed by 31 random bits.

B. Fixed Key Module

2.08 An FKM is used for service provided by cellular system providers for the CTS 1620-to-SCU

portion of the call and not for end-to-end calls. The FKM contains a PROM (programmable read only memory) chip containing one encryption key record. This record is programmed at the factory and cannot be changed by customers.

2.09 Mobile subscribers use FKMs to feed encryption key records to their CTS 1620s.

2.10 A 10-digit KIN is printed on the surface of the FKM and its sealed shipping container. Mobile subscribers and cellular service providers do not normally know the content of FKM encryption keys.

2.11 The FKM KIN consists of a 0 first digit, a 12-bit system index, and a 19-bit master key index. The system index is associated with a specific cellular system. It is assigned by the FKM supplier and is different from the cellular SID (system identification). The master key index specifies the address location of a master key in SRE key processors.

SWITCH RESIDENT EQUIPMENT

2.12 The SRE consists of the following equipment:

- Common control processor having two operations processors and two key processors
- A variable number of SCUs and data sets.

Duplicated operations and key processors provide high reliability.

A. SRE Operations Processors

2.13 The SRE operations processors consist of terminals, processors, and optional disk drives. They provide the following functions:

- (a) **Security Dial Back:** This capability allows remote access to the processors.
- (b) **Equipment Test:** Upon request, processors perform operational tests on the SCUs and report results.
- (c) **SCU Software Down Load:** The processors can download software to each SCU for the purposes of rebooting, changing software, or updating the generic program.

(d) **Record Keeping:** The SRE operations processors keep the following records.

(1) **Class of Service:** The class of service record for each KIN indicates: private-voice service, data service, combined private-voice and data service, or no service. The SRE limits service to that indicated by the service record. The records are changeable by cellular system providers.

(2) **Data Collection:** Monitors SCU usage and produces records detailing privacy/data call histories.

B. SRE Key Processors

2.14 The SRE key processors provide encryption key records to all SCUs. Each processor has a ROM (read only memory) circuit pack containing up to one-half million prerecorded encryption words. At the beginning of each private-voice or data service call, the key processor uses the KIN (received from the CTS 1620) to retrieve the proper encryption key.

C. SRE Channel Unit

2.15 The SCU is connected in the MTSO to either network interface trunks or loop-around trunks. It is compatible with 4-wire E&M type II trunks. The SCUs respond to request (signaling tones) from CTS 1620s for private-voice and data services.

2.16 For data service, SCUs provide modem functions that are compatible with 212AR and 2224 data sets, thus enabling CTS 1620s to communicate with computer modems.

2.17 Mobile-to-mobile private-voice and data service calls are provided by installing two SCUs back to back on loop-around trunks. This arrangement allows the individual CTS 1620s to use different encryption keys.

2.18 The SCUs are mounted in a channel unit cabinet (J-41657B) which is supplied with a minimum configuration of 16 units. The SCUs can be grown in increments of 8 units (channels) for a maximum of 48 channels per cabinet.

2.19 When a second channel cabinet is added, a common control cabinet (J-41657A) must also

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be provided. This cabinet can control up to 12 channel cabinets. (This 576-channel maximum configuration can accommodate approximately 14,000 subscribers.)

D. Data Sets

2.20 One 2224 data set (modem) is required in the MTSO for each SCU. When signaling tones from the CTS 1620 indicate a request for data service, the SCU connects the data set. The data sets are housed in a modem cabinet (J-41657C) which has a minimum configuration of 16 data sets. Growth is in increments of 8; the cabinet has a maximum capacity of 96 data sets.

E. Trunking Modifications

2.21 A 4-wire path has been provided between the CTS 1620 and SCU. This 4-wire path eliminates the need for echo cancelers and the resulting 2-second gap following handoff. Since the physical path through the 1A ESS switch is 2-wire, equivalent 4-wire is accomplished by frequency multiplexing the regular voiceband with a 20-kHz frequency band. A 20-kHz modulator circuit is contained in each SRU and a 20-kHz demodulator circuit is added to each SD-1A236-05 cell-site trunk circuit.

2.22 Voice signals toward a mobile customer use the 20-kHz frequency band; voice signals from a mobile customer use the baseband. Zone-office trunks and loop-around trunks are not equipped with demodulators.

3. USER PERSPECTIVE

SERVICE PROVISION

3.01 To obtain private-voice and data services, customers need to purchase or lease CTS 1620s and FKMs from cellular service providers, their resale agents, or other retailers. There are two distinct types of services: service provided by cellular service providers and end-to-end use. A customer may obtain both types of service.

3.02 The service provided by the cellular service provider allows the mobile user to make/receive private-voice calls. The portion of the call between the mobile user and the MTSO is encrypted.

3.03 End-to-end calls can be made between CTS 1620s without subscribing to the cellular ser-

vice provider's service. However, encryption does not apply to any other calls.

TYPES OF CALLS

3.04 In the following discussions, mobile customers who subscribe to the private-voice/data services provided by cellular system providers are called "subscribers". All other customers (mobile or land) are termed "nonsubscribers".

Mobile-to-Land Calls

3.05 *Private-Voice Calls From a Subscriber to a Land Customer:* These calls involve a CTS 1620 and an SCU. Calls are encrypted only over the radio link.

3.06 *Data Calls From a Subscriber to a Land Customer:* These calls involve a CTS 1620 and an SCU (including a modem) where the land customer has a data set. Calls are encrypted only over the radio link.

Land-to-Mobile Calls

3.07 *Private-Voice Calls From a Land Customer to a Subscriber:* These calls involve a CTS 1620 and an SCU. Calls are encrypted only over the radio link.

3.08 *Data Calls From a Land Customer to a Subscriber:* These calls involve a CTS 1620 and an SCU (including a modem) where the land customer has a data set. Calls are encrypted only over the radio link.

Mobile-to-Mobile Calls

3.09 *Calls Involving Two Subscribers:* These private-voice and data calls involve two CTS 1620-SCU pairs. Separate encryption keys are used.

3.10 *Calls Involving One Subscriber:* If the non-subscriber has a CTS 1620, one CTS 1620-SCU pair is involved. The default encryption key is used for the nonsubscriber segment. If the nonsubscriber has no CTS 1620, privacy is protected only over the subscriber CTS 1620-SCU segment of the call.

3.11 *Calls Involving Nonsubscribers:* These end-to-end private-voice and data calls involve a pair of CTS 1620s. No SCUs are involved.

USER OPERATION

3.12 For user operations required to use the CTS 1620s, refer to the CTS 1620 Privacy/Data Accessory User's Manual.

4. CELLULAR SYSTEM PROVIDERS

MTSO ENGINEERING

4.01 Routing of private-voice and data service calls is accomplished by using a special group of cellular DNs and defining the proper rate-center routing in the MTSO.

4.02 System providers need to coordinate the establishment of SCU-equipped trunk group(s) and associated translation data with the zone office(s) for proper routing of all incoming private-voice and data service calls to mobile subscribers. Trunk group traffic measurements, collected at the MTSO, and long-term forecasts should be used to size and reconfigure the SCU-equipped trunk groups.

ADMINISTRATIVE FUNCTIONS

4.03 Cellular system providers should update and maintain each subscriber's class-of-service record.

4.04 The ROM circuit packs in the SRE key processors should be periodically replaced with new ones. Encryption key usage records should be provided to the supplier who can determine which KINs should be changed. The supplier should maintain the uniqueness of the FKM encryption key records and administer the encryption keys in the ROM circuit packs.

MAINTENANCE

4.05 The SCUs provide status indicators that identify failed units. When a SCU has an internal failure, it disconnects the E&M leads and applies an off-hook signal to both the MTSO and network interface end. This causes the MTSO to report the trouble automatically and prevents the zone office from seizing the trunk. When a trunk failure is indicated, cellular system providers should check the SCU status indicators before diagnosing the trunk.

4.06 A 20-kHz modulator circuit has been added to the ROTL (remote office test line) to enable it

to be used to test the demodulators in the cell-site trunks. When the ROTL tests a cell-site trunk, it places the modulator in series with the demodulator of the cell-site trunk. The modulator-demodulator operation appears transparent to the ROTL tests, if operating properly.

4.07 The modulator is located in the 52A responder. The ROTL program turns it on at the beginning of a cell-site trunk test and turns it off at the beginning of a zone-office or loop-around trunk test. This is accomplished using signal distributors or central pulse distributors.

5. SUPPLEMENTARY INFORMATION

ABBREVIATIONS

5.01 The following abbreviations are used in this practice:

CKM—Configuration and Key Module

DES—Data Encryption Standard

DTE—Data Terminal Equipment

EEROM—Electrically Erasable Read Only Memory

EIA—Electronic Industries Association

FKM—Fixed Key Module

KIN—Key Identification Number

LRU—Land Resident Unit

MRU—Mobile Resident Unit

MTSO—Mobile Telephone Switching Office

PROM—Programmable Read Only Memory

ROM—Read Only Memory

SCU—SRE Channel Unit

SID—System Identification

SRE—Switch Resident Equipment.

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REFERENCES

5.02 The following documents are applicable to this feature:

- (1) CTS 1620 Privacy/Data Accessory User's Manual
- (2) AT&T 231-200-005—Mobile Telephone Switching Office, Cell Site, and Subscriber's Unit—System Description
- (3) AT&T 231-090-219—Remote Office Test Line—Feature Document
- (4) AT&T 231-218-301—Recent Change Formats and Implementation—Description and Procedures

(5) AT&T 231-290-609—Three-Way Calling—Feature Document

(6) AT&T 231-290-610—Call Waiting—Feature Document

(7) AT&T 231-290-616—Roamer II—Feature Document

(8) AT&T 231-390-212—Cellular Mobile Radio Office—Feature Document.

6. COMMENT FORM

6.01 A comment form is located at the back of this practice to provide a communications channel from the user to the writer.

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DYNAMIC CHANNEL ALLOCATION FEATURE DOCUMENT 1A ESS™ SWITCH AUTOPLEX™ SYSTEM 100 CELLULAR TELECOMMUNICATIONS SYSTEM

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Dynamic Channel Allocation – AUTOPLEX System 100

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BACKGROUND

1.02 At the MTSO (mobile telephone switching office), a voice radio channel is associated with a cell site voice trunk. The cell site voice trunk is identified by its TNN (trunk network number). The channel selection process uses the cell and the antenna face to determine a RI (route index) from which a TG (trunk group) is obtained. The TG is then accessed for an idle member which is identified by a TNN. A list of interfering TNNs is associated with a TNN using DCA. The selected TNN is used only if all interfering TNNs are idle.

AVAILABILITY

1.03 The DCA feature is available with the 1AE9.05 and later generic programs.

FEATURE GROUPS

1.04 The DCA feature is an optional custom feature and is available in the AMPSCP (System 100 call processing) feature package.

FEATURE ASSIGNMENT

1.05 The DCA feature is assigned on a per MTSO basis.

2. USER PERSPECTIVE

FEATURE DESCRIPTION

2.01 The assignment of a DCA to a voice radio channel is through its TNN to TGN auxiliary block. The TNN to TGN auxiliary block includes a DCA indicator and TNNs of the interfering voice radio channels. When a cell site trunk (voice radio channel) is selected that uses DCA, the PMT (path memory for trunk) state of each interfering TNN is examined. If any interfering voice channel is in use, the selected voice radio channel is idled and another channel is selected. This process can be repeated up to eight times before the selection routine indicates that no cell site trunks are available.

2.02 Dynamic channel allocation can be used with 1-way or 2-way trunks. If 1-way trunks are selected, they are chosen on a "longest idle" basis. There is no preference for selecting DCA versus non-DCA trunks in a 1-way trunk group.

2.03 If 2-way trunks are selected, the algorithm returns the lowest number idle member. If the 2-way trunk group is arranged with both non-DCA low members and DCA high members, a preference to non-DCA exists. Two-way trunk groups allow nonadjoining members. Therefore, a "gap" may exist between non-DCA and DCA members to permit growth of DCA and non-DCA subgroups independently.

INTERACTION

2.04 Interfering cell sites are limited to those served by a single AUTOPLEX System 100 MTSO. Cellular networking is not affected by the DCA feature.

3. ENGINEERING

CELL

3.01 It is not intended for DCA to replace standard cell engineering methods. Neither is it expected that complex dependencies between many cells will be established by DCA. Rather, DCA should be used to overcome the few instances where interference exists when standard cell engineering methods are applied. It may also be useful for areas that have shifting load peaks (e.g., morning commuter traffic moving into a city). In any event, the number of DCA trunks in a trunk group and the number of interfering TNNs per channel should be minimized. The system call capacity should be monitored to assure that it is not degraded.

SOFTWARE

A. Base Generic Program

3.02 Dynamic channel allocation adds approximately 1000 words to the generic program.

B. Parameter/Call Store Areas

3.03 Two 2-word call store parameter blocks, QG2DCACOLL and QG2DCAHOLD, provide the address and size of arrays to collect and hold DCA traffic counts. Each array contains two words per cell. The first word contains the count of successful DCA channel selections. The second word contains the count of failed selection attempts due to DCA conflict.

Dynamic Channel Allocation – AUTOPLEX System 100

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C. Translations

3.04 The TNN-TGN for *auxiliary block cell site trunks* is modified to include a list of other TNNs that identify interfering channels. (See Fig. 1.)

(a) *Word 1:* The DCA (Bit 23) indicator is set to 1 when this TNN represents a voice radio channel that uses DCA.

(b) *Words 3 through N, Type B:* Bits 22 through 19 when set to 0101 indicate that the word contains the TNN (Bits 14 through 0) of an interfering voice radio channel.

3.05 For example, consider cells A, B, and C where channel 1 interferes on cells A and B and channel 2 interferes on cells B and C.

- The TNN for cell B channel 1 is included as an interfering TNN in the TNN-TGN auxiliary block for cell A channel 1 TNN.
- The TNN for cell A channel 1 is included as an interfering TNN in the TNN-TGN auxiliary block for cell B channel 1 TNN.
- The TNN for cell C channel 2 is included as an interfering TNN in the TNN-TGN auxiliary block for cell B channel 2 TNN.
- The TNN for cell B channel 2 is included as an interfering TNN in the TNN-TGN auxiliary block for cell C channel 2 TNN.

For additional information regarding the TNN-TGN auxiliary block for cell site trunks, refer to Part 6 B(5).

	23.	22.	21.	20.	19.	18.	17.	16.	15.	14.	13.	12.	11.	10.	9.	8.	7.	6.	5.	4.	3.	2.	1.	0.
WORD 0	0	WRDN				0	0	0	0	0	0	0	0	0	MEMN (2-WAY) OR ALL ZEROS									
WORD 1	*	0	0	0	0	TCC						TGN												
WORD 2 TYPE A	0	WRDFN=0			0	0	FRAME	FG	VR	CHN														
WORD 2 TYPE B	0	WRDFN=0			0	0	VRG		VR	CHN														
WORD 3 THRU N TYPE A	0	WRDFN			AKMN						UTYMN													
WORD 3 THRU N TYPE B	0	WRDFN			0	0	0	0	TNN															

N<30

Note 1: Words 3 through N can either indicate carrier group alarm, trunk make-busy, or trunk network number and is repeated as necessary. All unused words are built all zeros.

Note 2: Required location LUCS (lower unduplicated call store).

• Dynamic Channel Allocation

Fig. 1—Network Number to Trunk Group Number for Auxiliary Block Cell Site Trunks

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4. IMPLEMENTATION

SET CARDS

4.01 Set card FF035 is assigned to the DCA custom feature. Set card calculation is as follows:

- If FF035 equals 1, the DCA custom feature is turned on.
- If FF035 equals 0, the DCA custom feature is not turned on.

RECENT CHANGE MESSAGE

4.02 A new recent change message, RC:DCHA, is applicable to the DCA feature. This message builds information for the DCA capability using keyword DCATNN (dynamic channel allocation trunk network number) and either ATNN (add trunk network number) or DTNN (delete trunk network number). Refer to Part 6 for detailed RC procedures.

VERIFICATION

4.03 Comprehensive information concerning TTY input and output messages are found in Part 6.

4.04 The VF:TNNSVY input message is modified to allow the newly defined keyword DCATNN to survey for those TNNs that have DCA feature capabilities.

4.05 The TR14 output message is modified to indicate if the specified TNN uses DCA.

Note: The TR14 output message currently provides the TNN-TGN auxiliary block address. This can be dumped to identify interfering TNNs.

5. ADMINISTRATION

MEASUREMENTS

5.01 New per-cell peg counts are required to show usage and blockage for DCA. These counts will be added to existing cell data on the AMPSTRAF (System 100 Traffic) printouts. No new EGOs are required. Refer to Part 6 for detailed traffic information.

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6. SUPPLEMENTARY INFORMATION

REFERENCES

6.01 The following documentation contains information related to or affected by the DCA feature.

A. AT&T Practices

- (1) 231-200-005 Mobile Telephone Switching Office, Cell Site, and Subscriber Unit System Description—AUTOPLEX System 100
- (2) 231-218-301 Recent Change Formats and Implementation Description Procedures
- (3) 231-318-334 CAMA, CCIS, CFTRK, DCHA, SCGA, TG, TGBVT, TGMEM, TKCONV, TMBCGA, and TRK—Trunk Translation Recent Change Formats
- (4) 231-290-600 Mobile Telephone Switching Office Feature—AUTOPLEX System 100
- (5) 231-290-604 Traffic Measurements Feature—AUTOPLEX System 100.

B. Other Documentation

- (1) Input Message Manual IM-6A001
- (2) Output Message Manual OM-6A001
- (3) Office Parameter Specification PA-6A001
- (4) Translation Guide TG-1A
- (5) Translation Output Configuration PA-6A002
- (6) Parameter Guide PG-1A.

7. COMMENT FORM

7.01 A comment form is located at the back of this practice to provide a communications channel from the user to the writer.

8. ISSUING ORGANIZATION

Published by
The AT&T Documentation Management Organization

Portable Low-Power FM Broadcast Transmitter

Overview

This is quick overview of a portable low-power FM broadcast station using pre-built modules from Broadcast Warehouse.

The modules consist of a Broadcast Warehouse PLL+ 1 Watt FM Exciter, a Broadcast Warehouse Limiter PLUS, and a Broadcast Warehouse DIGILOG Stereo Encoder. These modules were originally sold as hobbyist kits and were quite popular in the late 1990s to early 2000s with the low-power FM (i.e. "pirate radio") crowd.

Broadcast Warehouse's newest transmitter modules are "all-in-one," eliminating the need for building separate modules in order to get a high-quality FM radio station on the air.

Pictures & Construction Notes



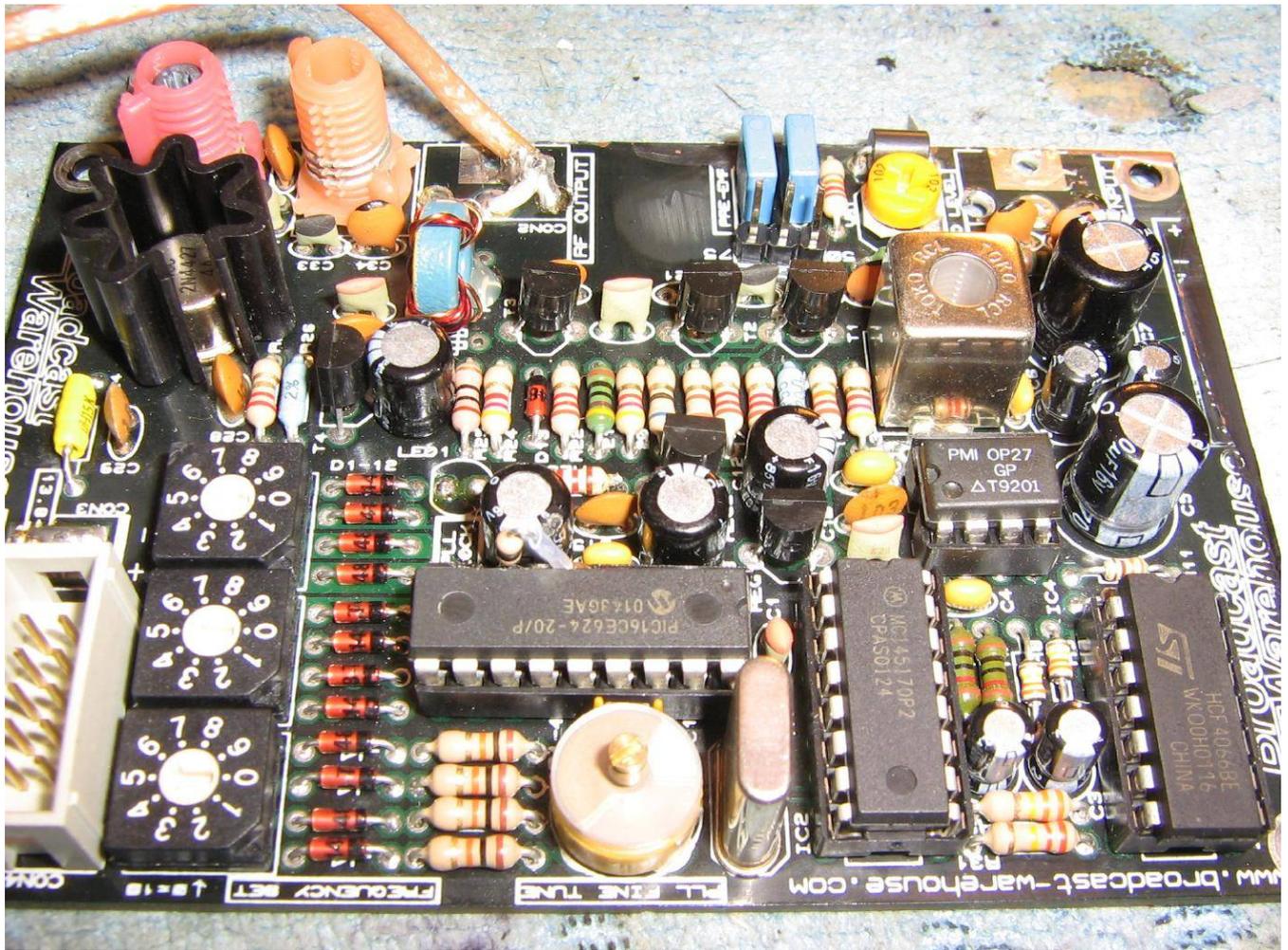
Overview of the Broadcast Warehouse PLL+ 1 Watt FM Exciter module.

Broadcast Warehouse's designs have surprisingly high-quality for being hobby kits. They incorporate a number of features found only on high-end broadcast exciters, including excellent audio response and rejection of spurious RF emissions which tend to be found on other designs.

The modulation input is via the solder pads on the lower-left. There was a RCA jack there originally but I removed it. The exciter has a jumper setting for 75 μ S, 50 μ S, or no pre-emphasis. No audio pre-emphasis will be used here as the input limiter discussed later will take care of that.

Next to the modulation input is the modulation adjust potentiometer VR1. This will need to be adjusted to give a 100% modulation level, which is a maximum deviation of \pm 75 kHz.

The RF output is via the bottom-center solder pads. There was also a RCA jack there originally, but I replaced with a direct coax connection going to a panel-mounted SMA jack.



Alternate view.

The exciter module requires +12 to +16 VDC at around 300 mA. The RF output power is around 1 watt (+30 dBm) over the entire FM broadcast band (87.5 – 108 MHz in 100 kHz steps).

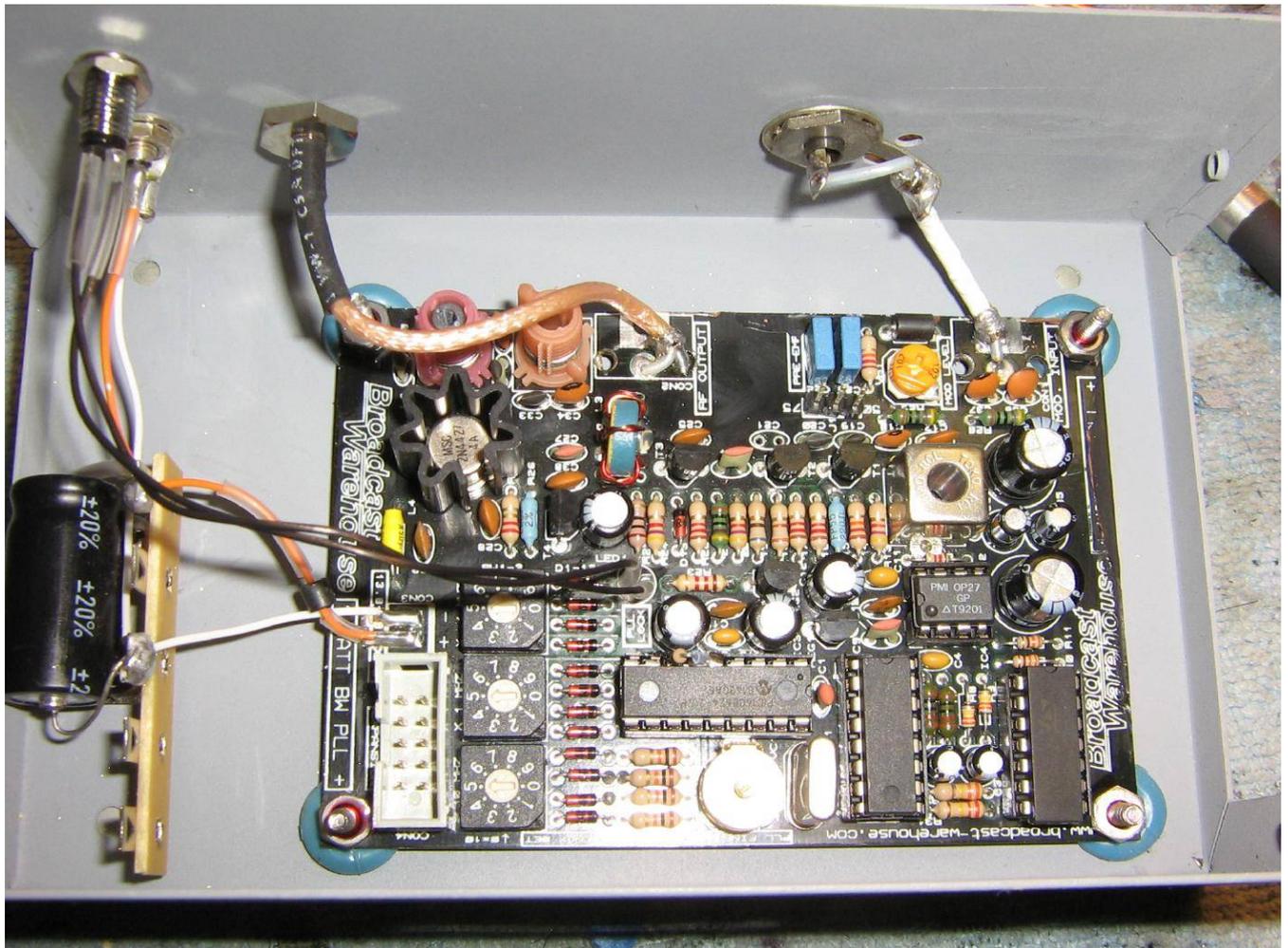
The yellow trimmer capacitor (bottom-center) can be used to tweak the final output frequency.

There is also a LED which lights when the PLL is locked. This PLL lock/unlock indicator LED should be panel-mounted.

I also replaced the stock LF351 PLL loop filter op-amp with a lower-noise OP27.

A dab of hot glue should be used to prevent RFT1 (the blue/yellow ferrite toroid) from flopping around.

The output frequency is selected by three DIP switches or by an external panel with a LCD display. The LCD control panel option will be covered here. You'll need to set the DIP switches to "555" if you are using the external LCD panel frequency control.



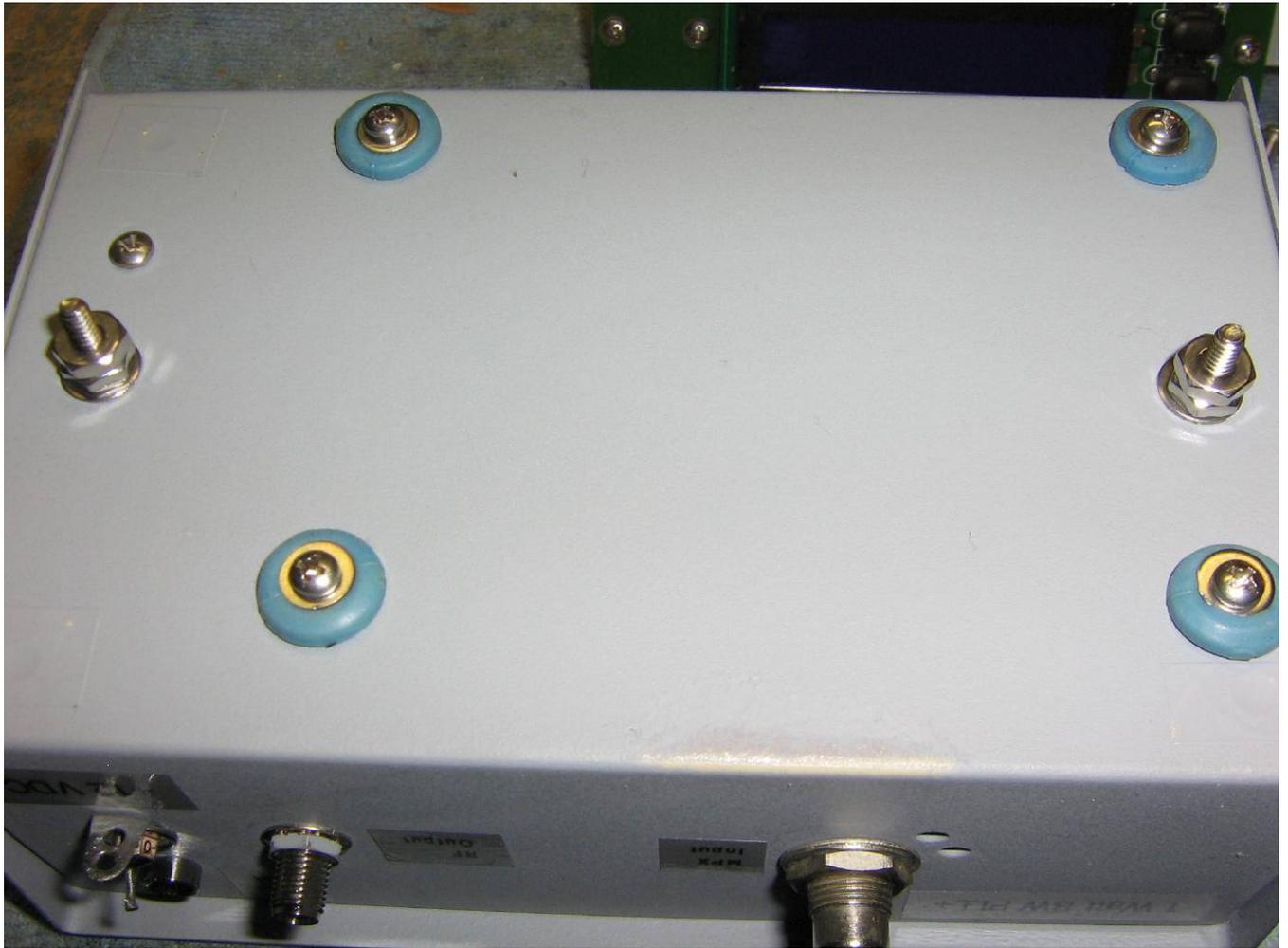
Mounting the exciter inside an old printer switch case.

DC power input is via the orange/white wires on the left. They first go to a terminal strip where a 470 μF capacitor helps to condition the input DC power.

Above the DC power input is the panel-mounted PLL lock LED.

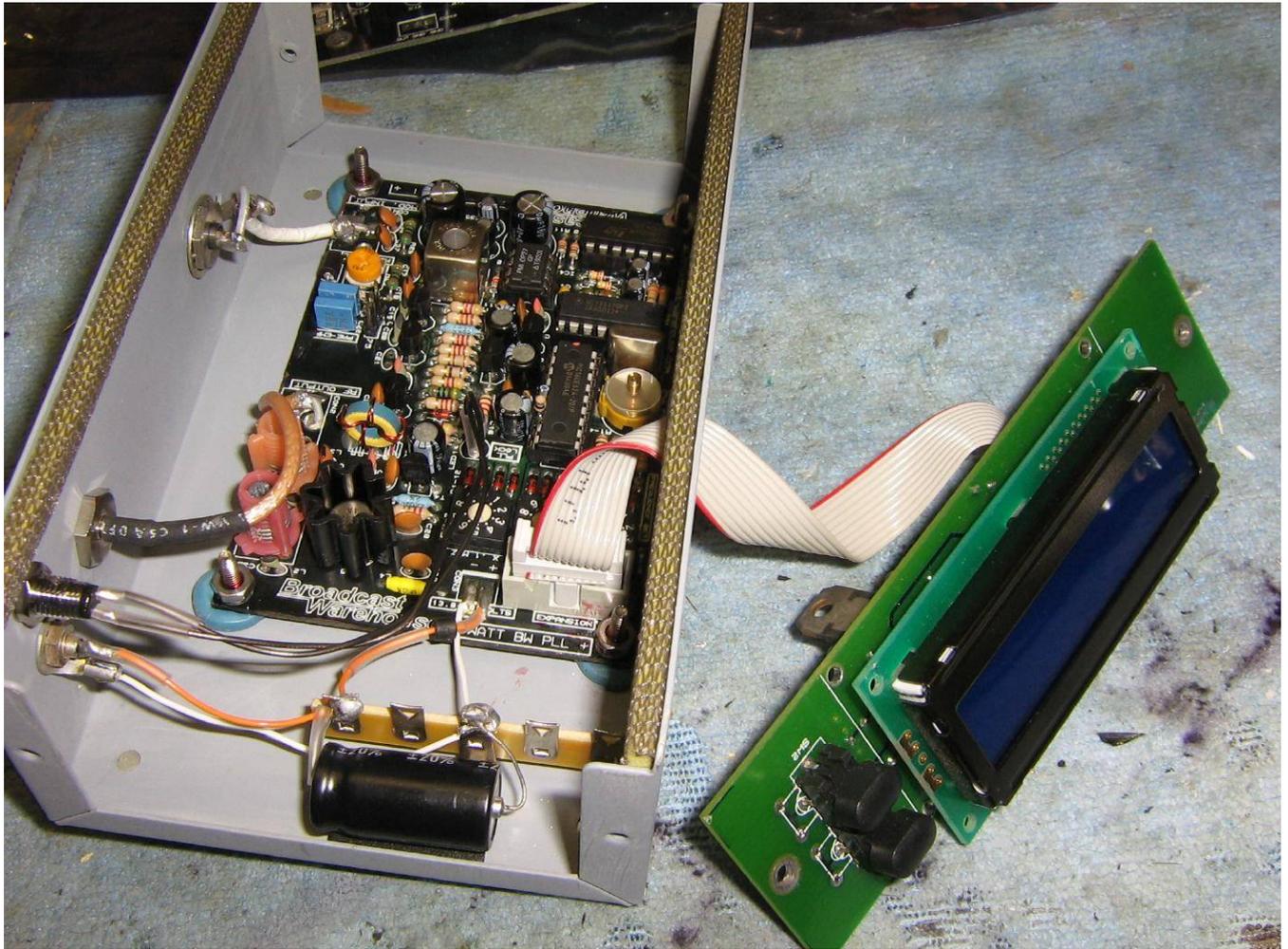
Next to that is a panel-mounted SMA jack for the RF output.

On the right is a panel-mounted RCA jack for the modulation input, which will be coming from the stereo encoder/limiter in this case. It's possible to run "line level" audio directly into the exciter module, but the use of a pre-processing limiter/compressor is *HIGHLY* recommended.



Because the exciter is vulnerable to microphonics induced by vibrations, the board itself will be mounted on four rubber grommets sandwiched using #4 stainless steel hardware.

The two #8 screws coming out of the bottom of the printer switch box will hold the case to the side of the ammo can it will be mounted in.



Connecting the external frequency control and LCD display panel to the exciter module.

The LCD display board also comes as a kit (or pre-built module) and with a short ribbon cable.

The two buttons toggle the frequency up or down between 87.5 MHz and 108 MHz.

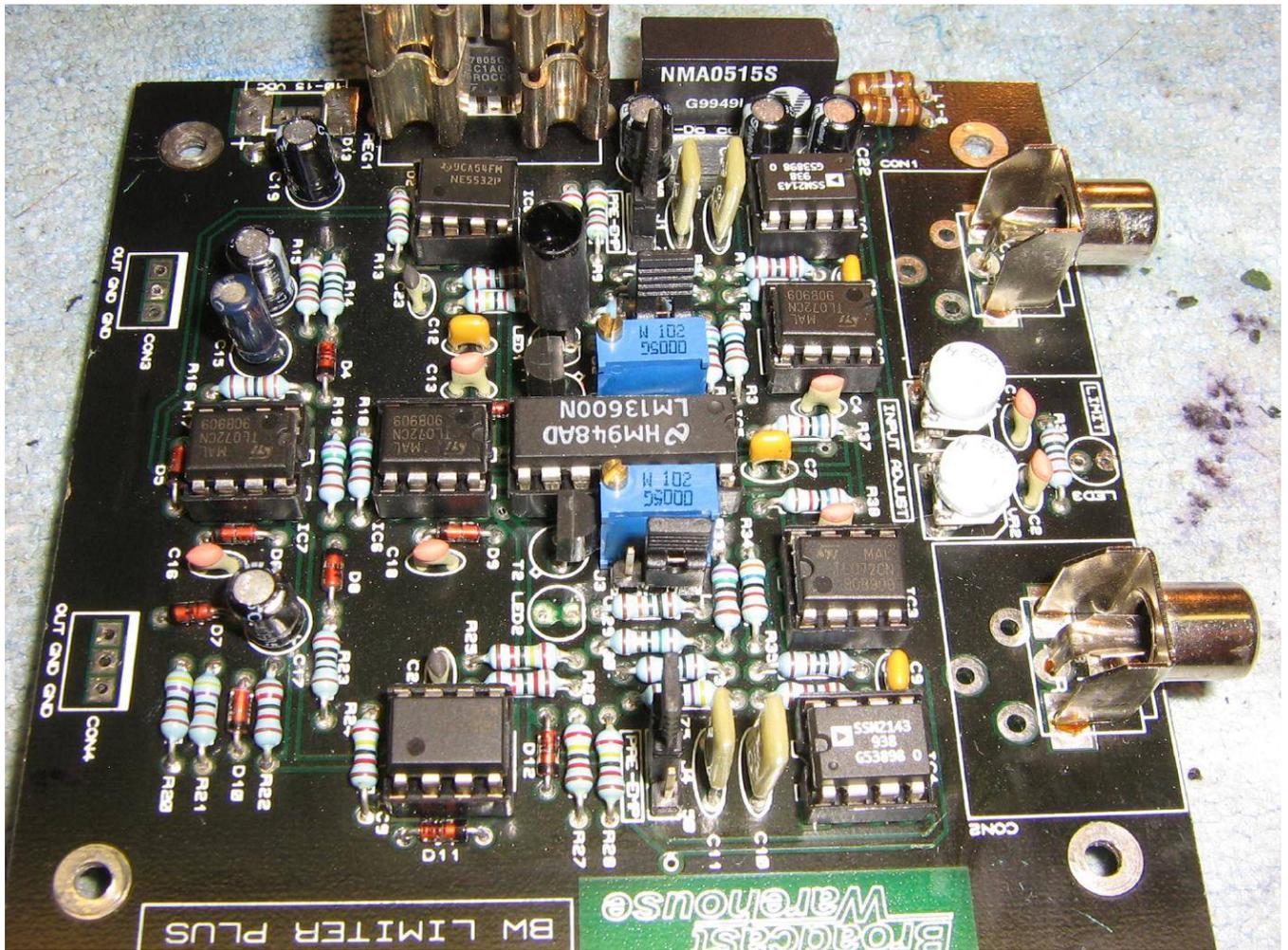
The LCD displays the final chosen frequency and the words <UNLOCKED> when the PLL is unlocked and <LOCKED> when the PLL is locked.

The exciter powers down its final RF stage when the PLL is unlocked to prevent spurious RF emissions.



Finished exciter case external overview.

A hole was drilled in the top cover to allow access to the VR1 modulation adjustment potentiometer.



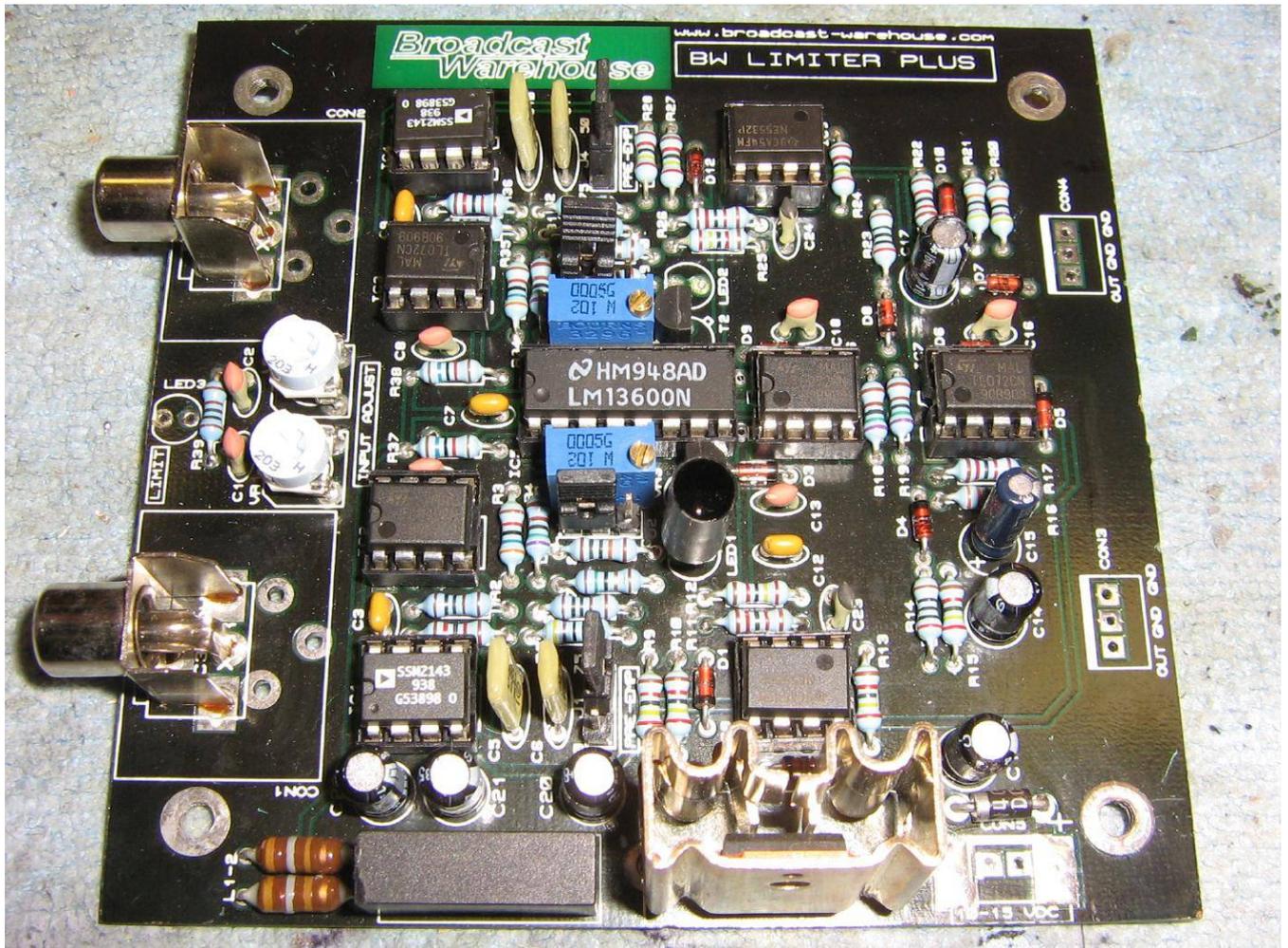
Overview of the Broadcast Warehouse Limiter PLUS.

The limiter requires +12 to +16 VDC with a minimal current draw.

The Broadcast Warehouse Limiter PLUS takes any audio input between -10 dBu and +18 dBu and either increases or decreases its level so it won't overmodulate the exciter.

The limiter is stereo, so it has two audio inputs: left & right. It will also work in mono, if so desired.

There is also an onboard selectable "clipper" circuit. This can be enabled to artificially increase the "loudness" of your audio by clipping the peaks. This is what commercial radio stations do, but it tends to fatigue the ear of the listener.



Alternate view

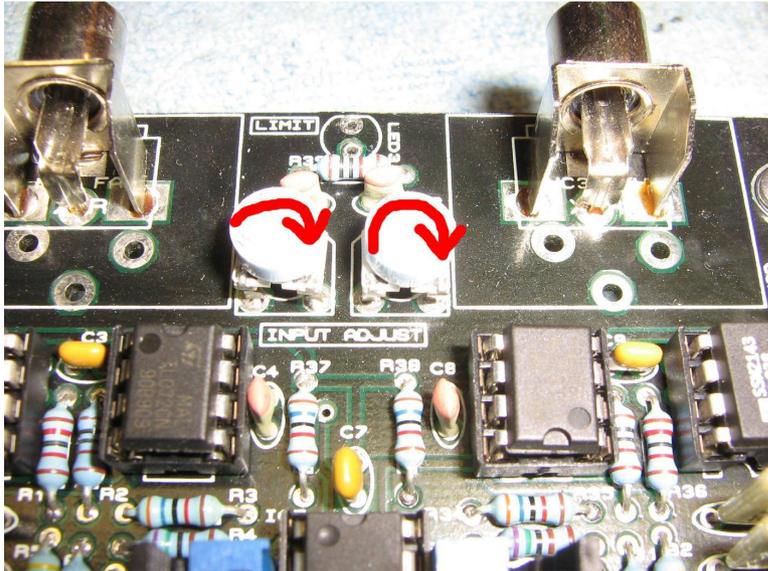
The limiter has a jumper setting for 75 μ S, 50 μ S, or no pre-emphasis. The 75 μ S pre-emphasis setting will be used here. The pre-emphasis settings tend to vary around the world.

The limiter has either unbalanced or balanced input/output connections. The unbalanced settings will be used here. Most "pro-level" audio gear uses balanced audio connections, while consumer-grade equipment is unbalanced.

Capacitors C5/C10 (6800 pF) and C6/C11 (4700 pF) help set the pre-emphasis time constant. These should be replaced with their polystyrene equivalent.

The "Limit" LED (labeled LED3) should be panel-mounted. Set the limiter's input audio level level so this LED flickers a bit.

The two other LEDs (labeled LED1 & LED2) should be painted black, or at least well shielded. These LEDs act as "hard limiters" in the clipping circuit and stray light modulation striking these LEDs can be induced onto the passing audio.



The two limiter audio input adjustment potentiometers (labeled VR1 & VR2) should be set full clockwise.

The Broadcast Warehouse Limiter PLUS will handle just about anything, so let it do its job. Use an external audio mixer to control the audio levels into the limiter, if so desired.

The first step in setting up the limiter after it has been built is to trim any voltage offsets in the op-amps. This will reduce the overall distortion in the rest of the system. Apply DC power (+12 VDC) to the limiter without any audio and follow these steps:

Connect a multimeter to the output of the left channel and set the meter onto a millivolt DC range. Now adjust the multiturn potentiometer VR4 for the *minimum* output voltage on the meter. Aim for a reading of a few millivolts or less. Repeat the procedure for the right channel with VR3.

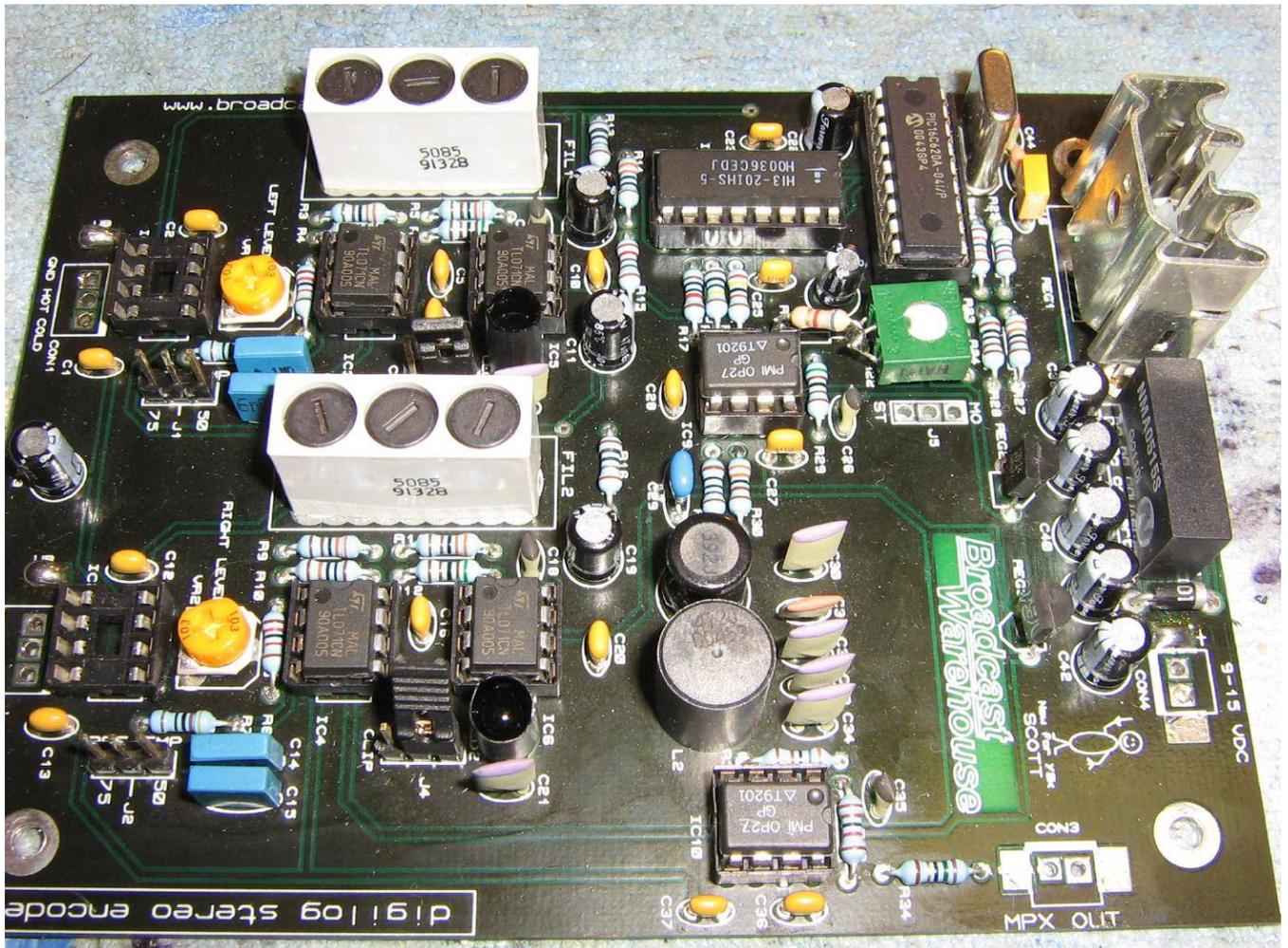
Decide on the pre-emphasis settings for your country / region. 75 μ S for the Americas and Japan and 50 μ S for the rest of the world, usually. Set the pre-emphasis with the jumpers.

Make sure to remove the pre-emphasis on the stereo encoder and exciter.

Decide on "Clarity" or "Loud" modes. Clarity will produce a closer to the original sound while loud will give you a more processed commercial sound. Set the jumpers to your chosen mode.

Apply audio to the inputs of the limiter and adjust the the input gain controls to maximum (clockwise). You should have the LED limit indicator flashing with the peaks of the audio, if not, then you need to apply more audio level from your audio source. When the limiter is limiting (LED is flickering) then you can adjust your transmitter's modulation control for a peak deviation of +/- 75 kHz.

You may wish to readjust the input gain controls on the limiter so that the limiter starts to limit at your desired input level, or you can leave the input level controls at maximum to get the limiter to act more as a compressor and increase your average volume.

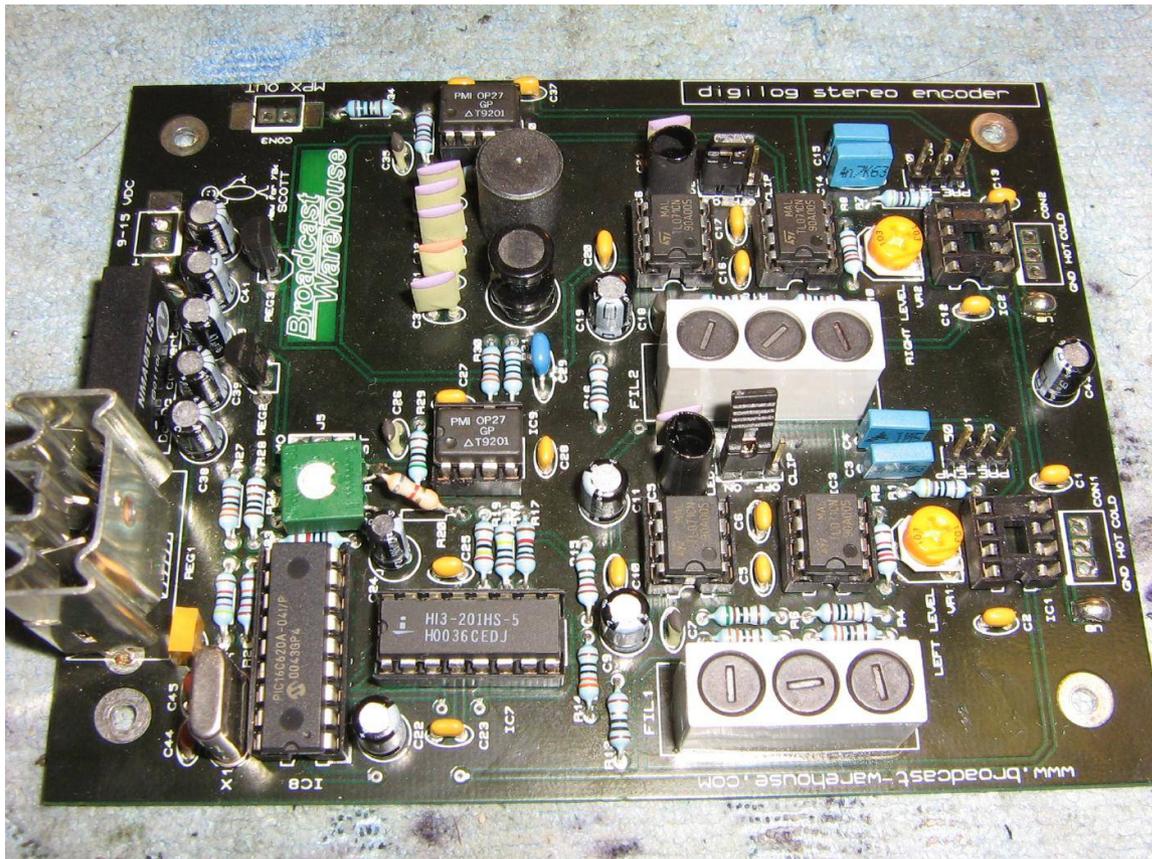


Overview of the Broadcast Warehouse DIGILOG Stereo Encoder.

The stereo encoder requires +12 to +16 VDC with a minimal current draw.

FM broadcast stereo encoders are designed take the two left and right channels and combine them into "Left + Right" and "Left - Right" components. The "Left + Right" spectrum occupies everything below 15 kHz to provide for monaural receivers. The "Left - Right" component is converted into a double-sideband suppressed signal at 38 kHz. This modulation creates a upper- and lower-sideband centered around a 38 kHz (suppressed) carrier.

The stereo encoder also generate a 19 kHz pilot tone signal. This pilot tone is used to indicate a stereo signal is present at the receiver and the receiver also doubles this 19 kHz signal to help demodulate the "Left - Right" sideband. The receiver then reverses the process, converting the "Left + Right" and "Left - Right" signals back into the individual left and right audio channels.



Jumper J5 selects either "Stereo" or "Mono" mode. You should change this out to a panel-mounted switch. It's handy to quickly switch between stereo or mono for certain low-power FM applications. Mono tends to work better in the fringes, which is just about everywhere when dealing with low-power FM.

The stereo encoder should also have its pre-emphasis settings disabled.

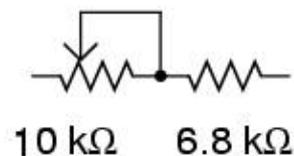
The "clip" jumpers should be set. This is to clip anything which may have gotten past the limiter.

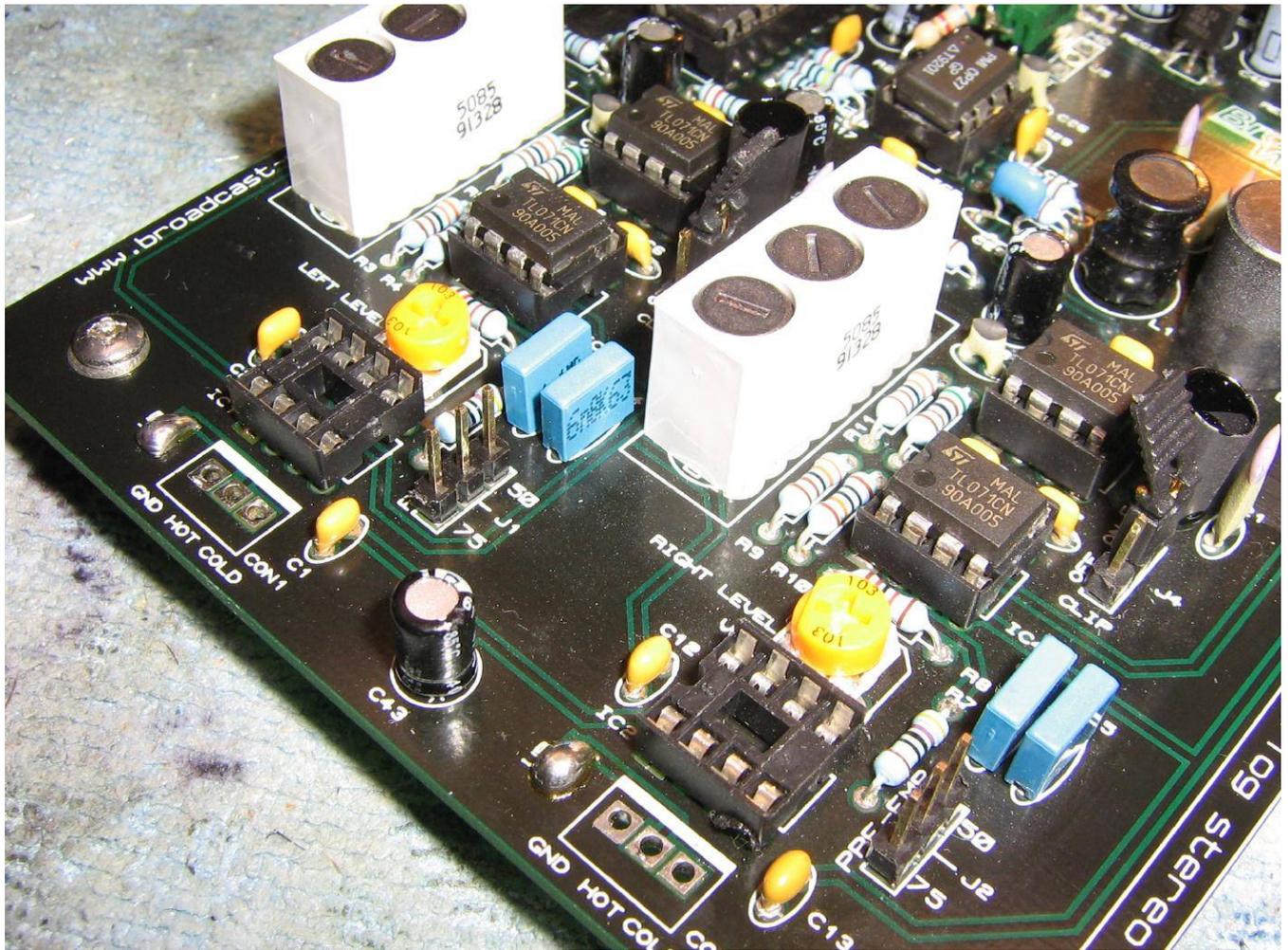
The clipping LEDs should also be painted black or shielded, just like the limiter.

IC9 and IC10 were replaced with OP27 low-noise op-amps.

The stereo encoder has a fixed 9% modulation for the 19 kHz pilot tone, but you can tweak it a bit by replacing resistor R20 (13 kohm) with a 10 kohm potentiometer in series with a 6.8 kohm resistor on its wiper.

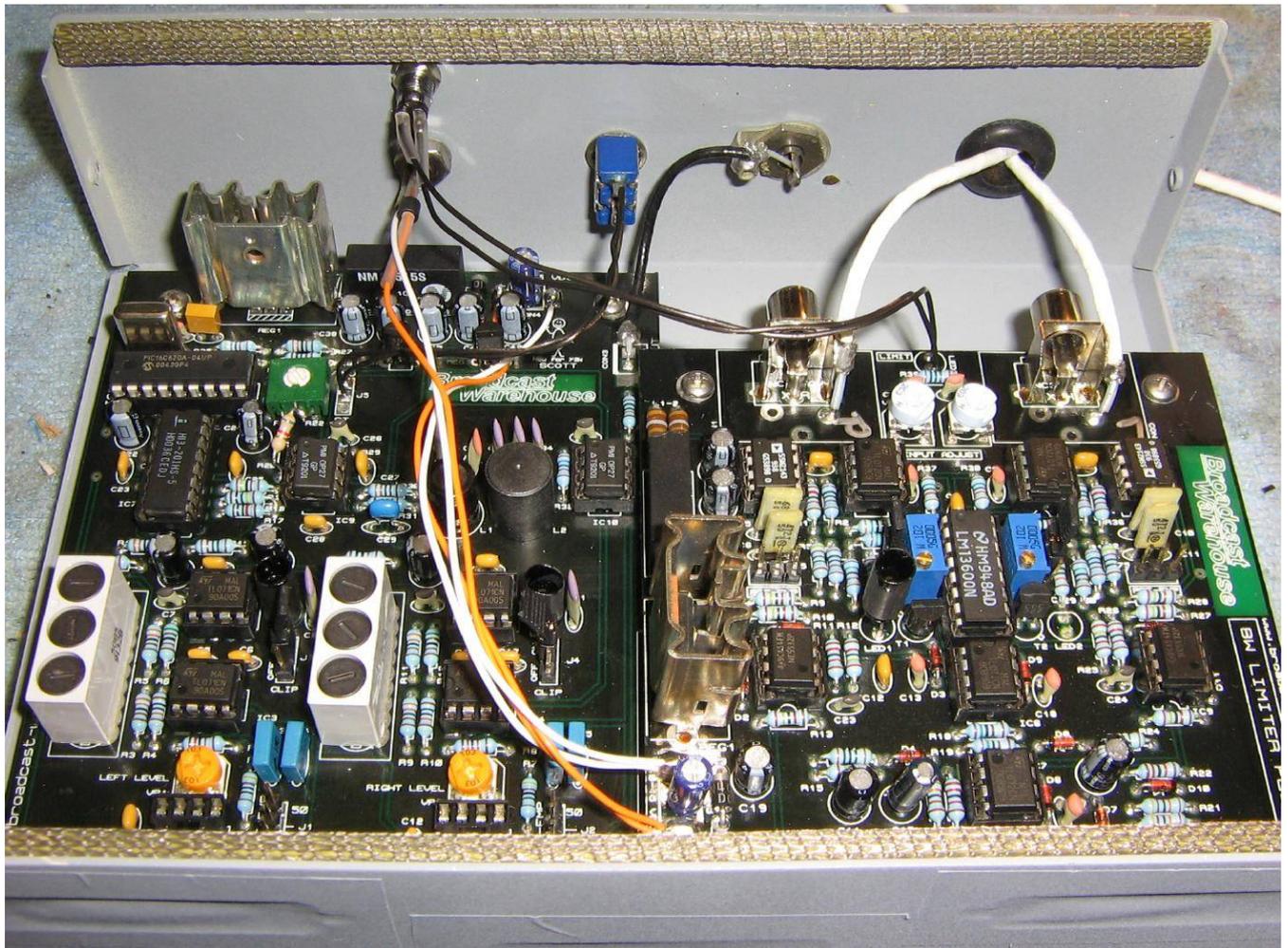
Replace R20 with this:





For unbalanced audio inputs to the stereo encoder, apply the audio via the "COLD" and "GND" solder pads. You should also apply a solder jumper to the exposed "UB" solder pad next to the audio inputs.

Potentiometers VR1 and VR2 will be used for setting the main left and right audio levels.



Mounting the Broadcast Warehouse Limiter PLUS and Broadcast Warehouse DIGILOG Stereo Encoder in a large printer switch case.

On the right-side, the left and right audio inputs are via two pieces of white coaxial cable.

To the left of those coax cables is a panel-mounted RCA jack for the output signal from the stereo encoder. This connects to the exciter's modulation input.

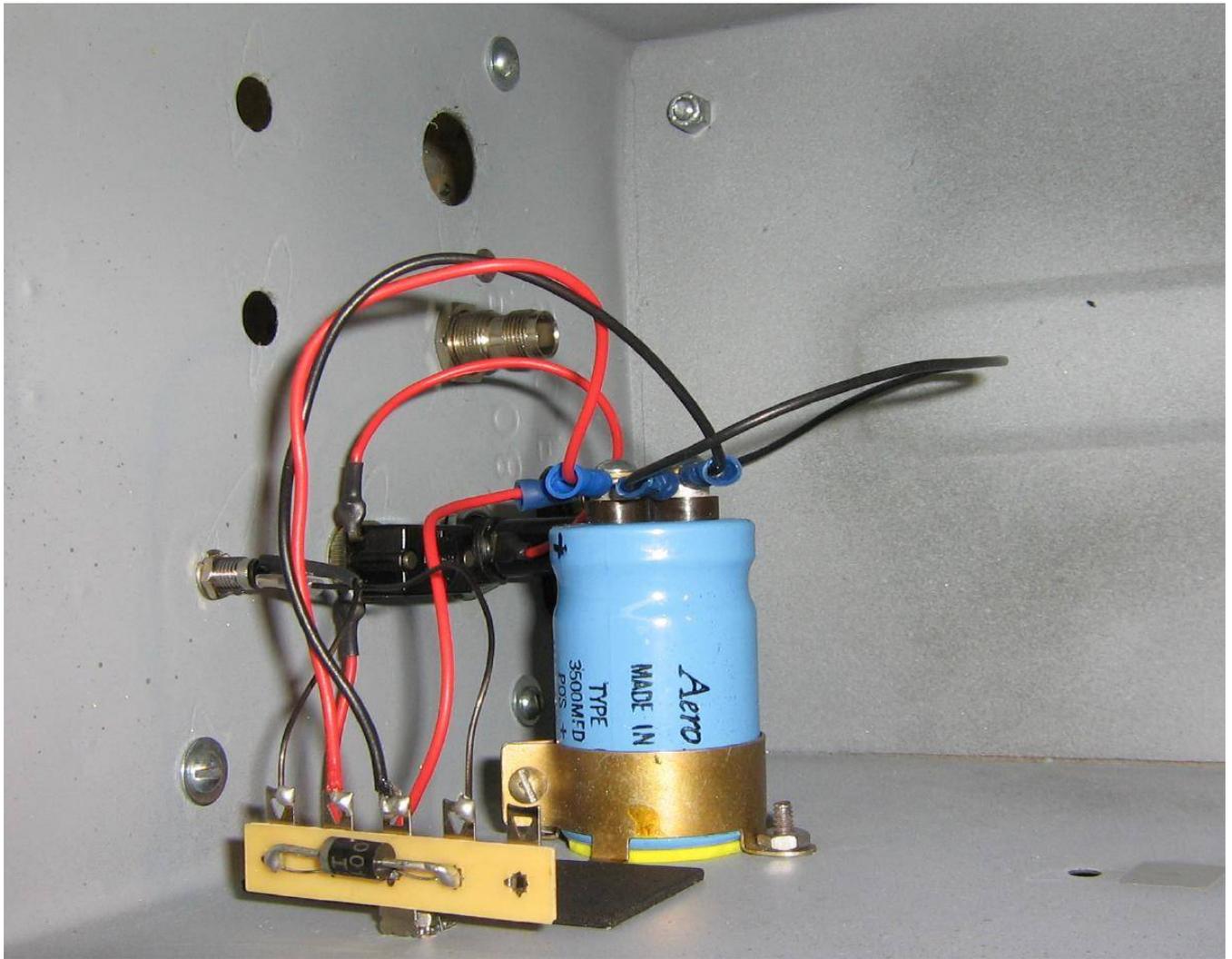
To the left of that RCA jack is a panel-mounted switch to select between stereo and mono. Just solder wires from the J5 solder pads to directly to the switch.

A panel-mounted LED for the limiting indicator and a feed-through capacitor for the DC input power are on the left-side.



Finished limiter / stereo encoder case overview.

The two holes in the cover are for access to the audio level input controls on the stereo encoder.



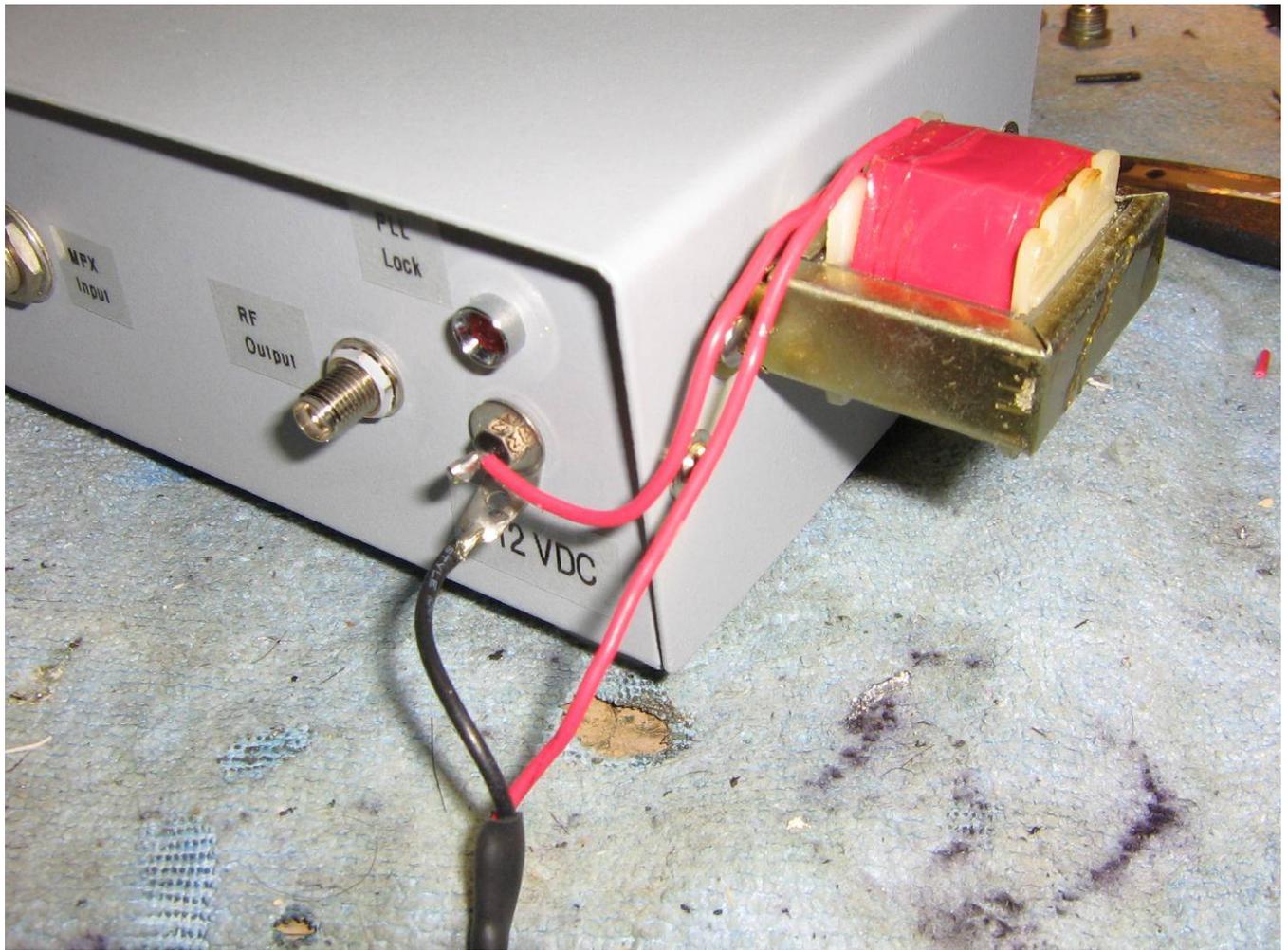
The modules will be housed inside an old ammo can.

Regulated +12 VDC input is via banana jacks along the bottom.

A SPST switch controls the main DC power.

A 2 amp fuse and 1N5401 shunt diode protect against voltage polarity reversal.

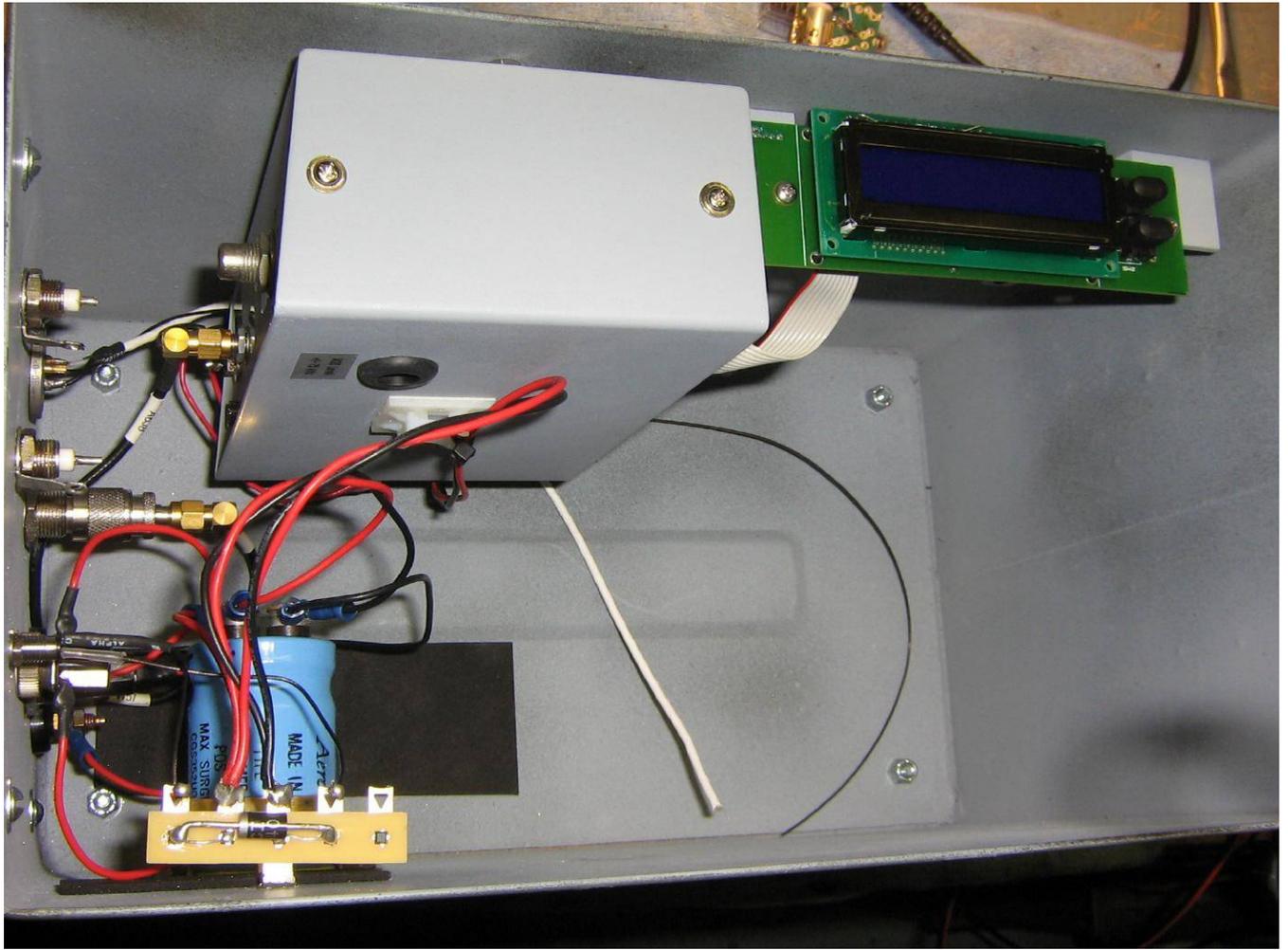
An optional 10,000 μ F capacitor was added to filter any residual ripple on the DC input.



Another option is adding a 1 henry inductor in series with the DC power lines. These inductors are very useful for cleaning up any "hash" when powering the transmitter from a vehicle's DC power system.

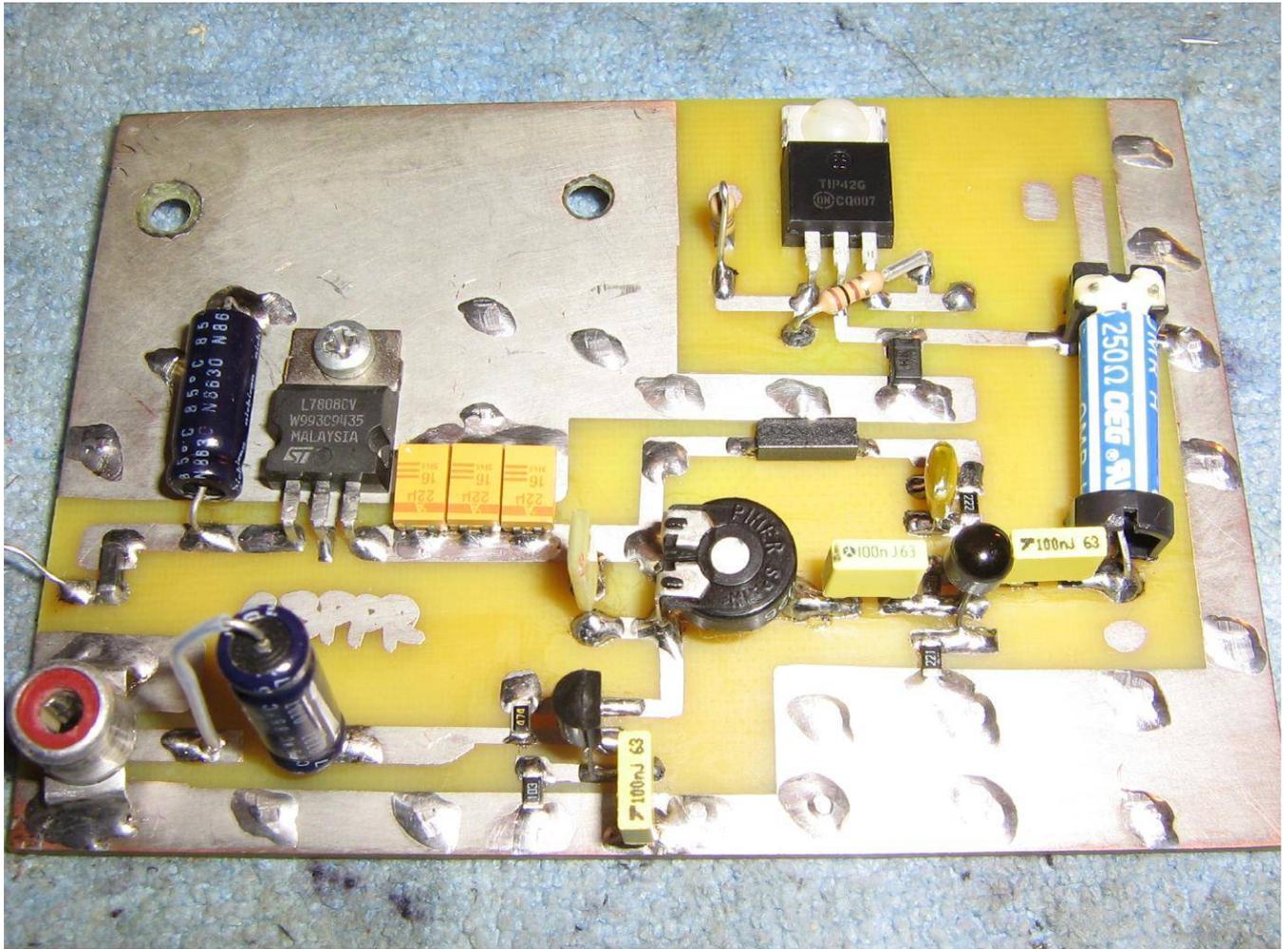
Ideally, a single large filter inductor should have been mounted in series with the main DC input, right before the ripple capacitor.

You can find these large inductors in surplus mobile-mount CB radios.



Mounting the exciter case and the LCD display.

The LCD display is mounted on two little aluminum L-brackets.



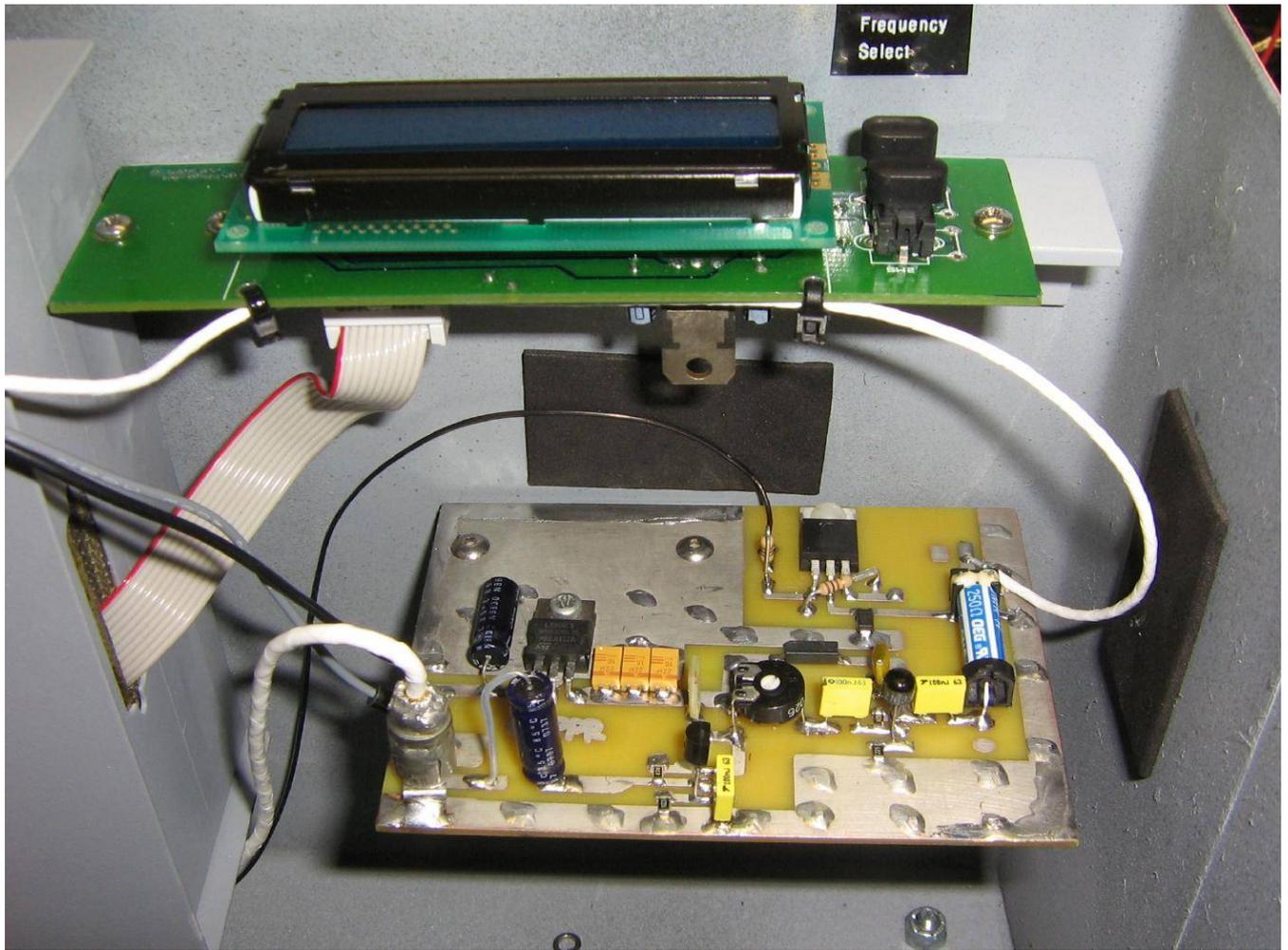
This is an optional microphone amplifier and PTT control circuit board.

It's designed for use with a surplus Sonetronics H-250/U noise cancelling military handset.

The handset's PTT switch controls a relay which applies the microphone audio to one of the channels on the limiter's audio input.

This is useful for emergency situations where you may need to notify a large number of people quickly via radio, or for calling in bomb strikes.

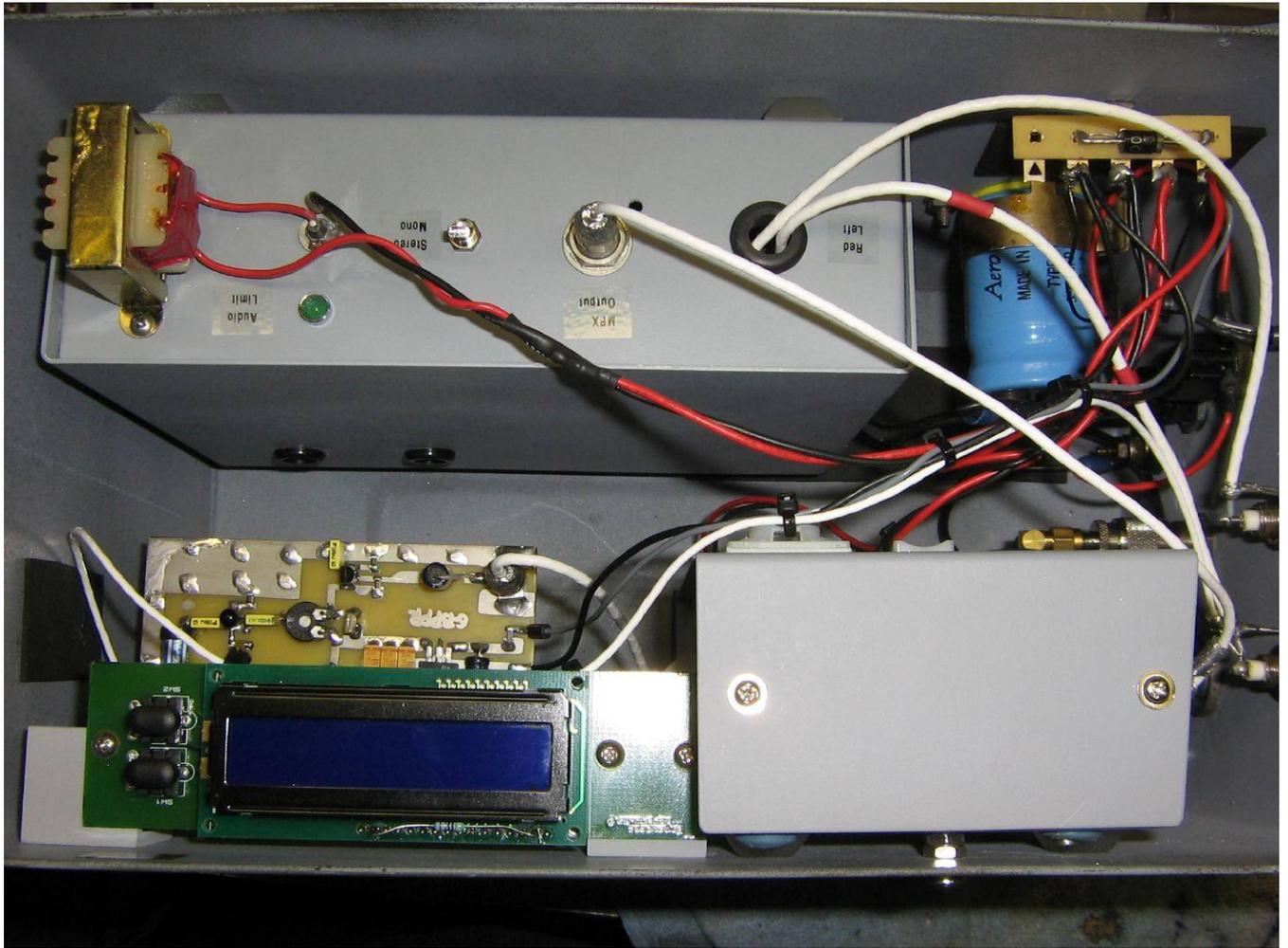
It's also useful for when you tune all the FM radios at Best Buy to the same frequency and then transmit rude comments over the air...



Mounting the microphone amplifier and control circuit board just below the LCD display.

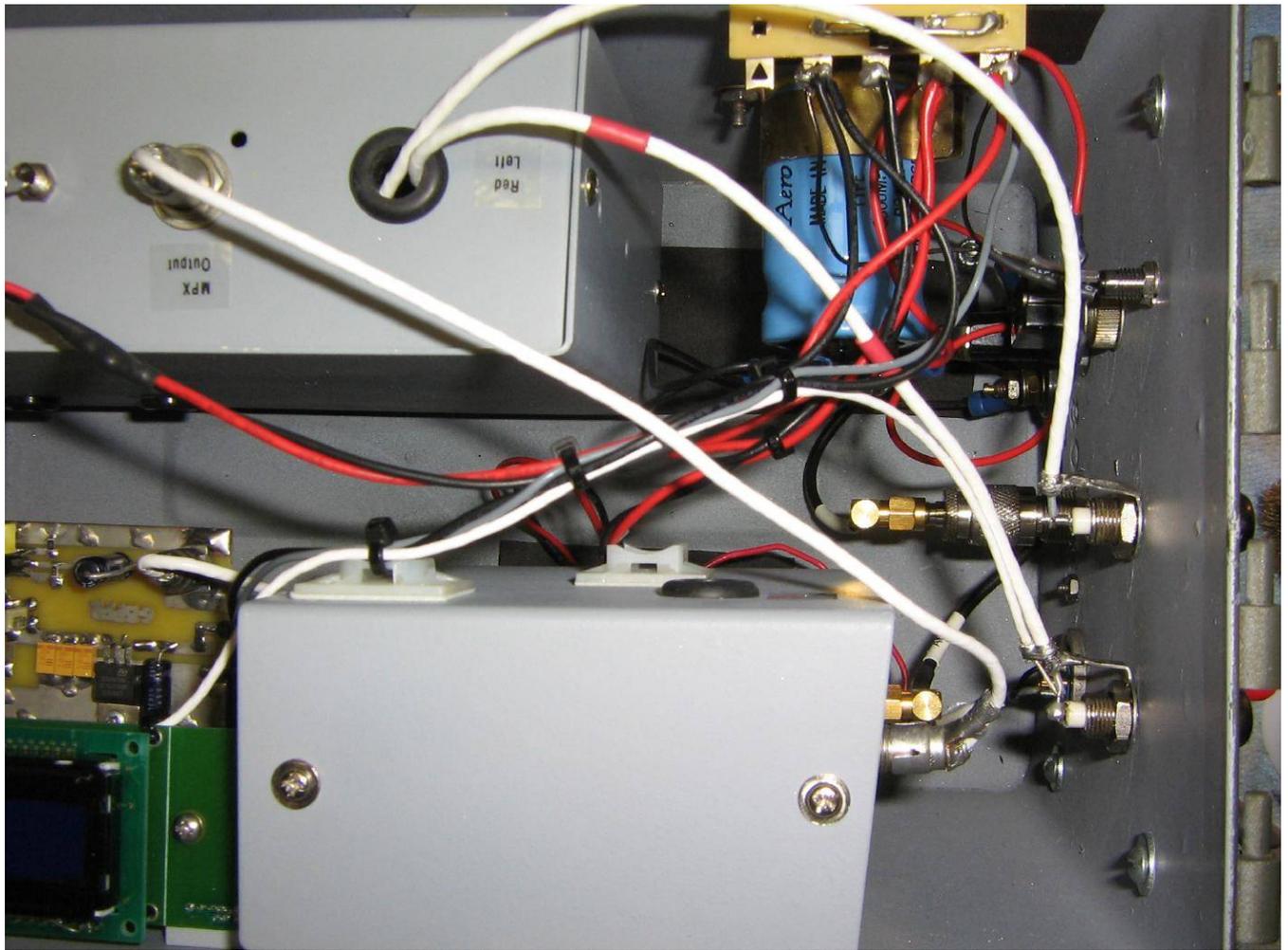
The microphone audio input from the H-250/U handset is via the RCA jack along the lower-left.

The amplified microphone audio is then sent to the "Left" channel on the limiter via the coax cable after the relay on the right.



Mounting the audio limiter / stereo encoder case.

Be sure to use shielded wire or coax on all the audio connections.



Behind the front-panel overview of the finished low-power FM broadcast transmitter.

Audio inputs (left and right) are via the BNC jacks on the top.

RF output is via a SMA-to-TNC adapter to a panel-mounted TNC jack.



Front-panel overview of the finished low-power FM broadcast transmitter.

The U-229 connector for the H-250/U handset is on the bottom-left. On the bottom-center is the TNC jack for the RF output.

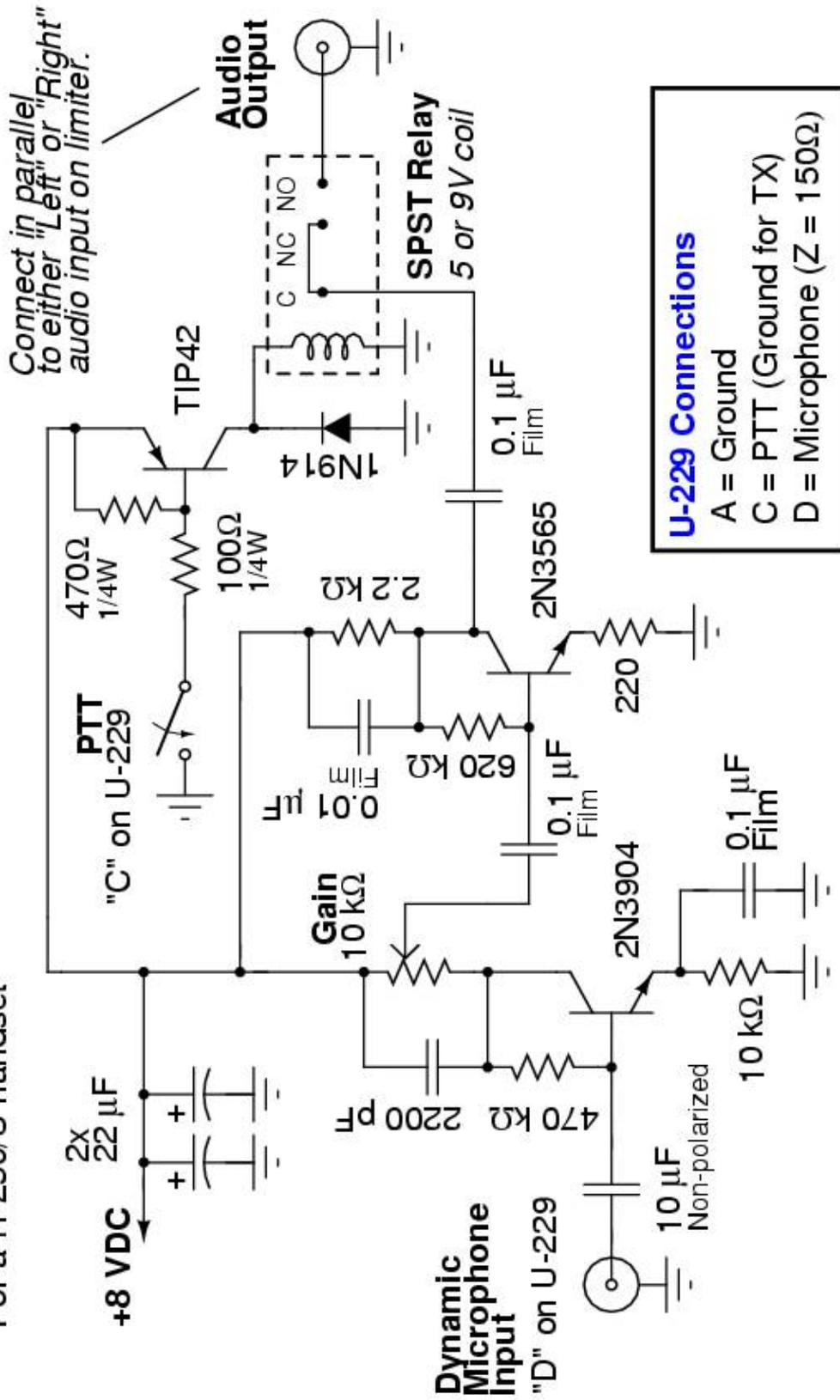
The two BNC jacks along the top are for the audio input.

Main DC input is via the banana jacks on the lower-right.

Avoid mounting the LCD display externally to keep the transmitter unit fairly weather resistant.

Dynamic Microphone Amplifier

For a H-250/U handset



Broadcast Warehouse PLL+ 1 Watt Exciter Manual

The Broadcast Warehouse PLL+ 1 Watt Exciter is a compact FM broadcast exciter with specifications that put many commercial exciters to shame. The modern innovative design allows audio and RF performance never before seen in kit or module exciters. The 'virtual VFO' dual-loop system allows perfect audio flatness to below 10 Hz. AFC bounce and modulator overshoot are a thing of the past. You can now pass that low bass without distortion and get that perfect stereo separation that you have been demanding from your exciter. Broadband 'no tune operation' allows for ease of use. The only adjustment required is of the direct-reading decimal dial switches for frequency selection. RF power is muted during PLL out-of-lock conditions and the built-in harmonic filter keeps your signal clean. The expansion connector allows for external modules to be connected to the board, such as the Broadcast Warehouse PLL+ LCD.

Features

- ◆ Phase Locked Loop System
- ◆ Dual-Speed PLL
- ◆ Low-Noise Oscillator
- ◆ Broadband Design
- ◆ No-Tune Operation
- ◆ Direct-Read Switches
- ◆ Very-Low Distortion
- ◆ Switchable Pre-Emphasis
- ◆ 1 Watt RF Output
- ◆ Harmonic Filter
- ◆ Expansion Connector
- ◆ Compact Size
- ◆ Black Oxide High-Grade PCB

Specifications

RF Power Output	1,000 mW (+/- 100 mW), 50 ohms
DC Power Requirements	13.3 – 16 VDC, 300 mA max.
Harmonic Output	-60 dBc
Spurious Output	-85 dBc
Frequency Steps	100 kHz steps
Out-of-Lock Power Down	-50 dBc
Frequency Stability	+/- 200 Hz
Audio Input Level	adjustable
Audio Frequency Response	10 Hz – 100 kHz
S.N.R	>80 dB
Distortion	<0.05%
Pre-Emphasis	None, 50 μ s, or 75 μ s (switchable)

Principles of Phase Locked Loop Systems

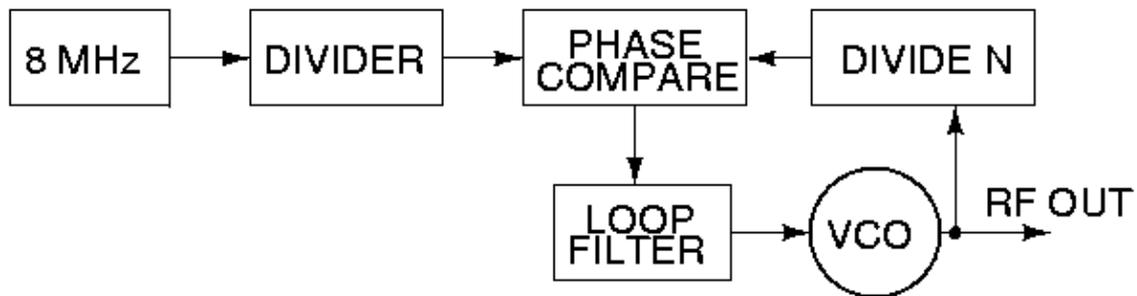
The Voltage Controlled Oscillator (VCO) feeds a portion of its RF into one side of a Phase Locked Loop (PLL) chip. The other side of the PLL chip is fed with a reference frequency, usually derived from a quartz crystal, which is very stable. The phase locked loop chip outputs a *high* or *low* voltage. High or low is subject to whether the reference frequency input is *lagging* in phase or *leading* in phase compared to the RF input from the VCO. In other words, *high* if the reference frequency is higher in frequency than the VCO frequency and *low* if the reference is lower. The reference frequency is usually in the range of 10 kHz to 100 kHz and also forms the step size of the VCO. A reference frequency of 100 kHz can not have a lower step size than 100 kHz. Crystals are physically very large at these frequencies so we tend to use a higher frequency crystal and divide it

down to the reference frequency. The 100 MHz signal from our VCO needs to also be divided down to the reference frequency and to do this we need a 'divide-by-N' counter. "N" is any number which can divide our frequency to the reference.

The phase locked loop system will comprise of:

1. The Voltage Controlled Oscillator (VCO).
2. The 'divide-by-N' (100 MHz to reference frequency).
3. A stable crystal for the reference.
4. A fixed divider (to divide the crystal to the reference frequency).
5. A phase comparator.
6. The loop filter (voltage smoother).

In the example below we will use a 8 MHz crystal, a reference of 100 kHz and the RF frequency we will lock to is 99.9 MHz. The reference divider is 80 and the RF divider is 999.



The 8 MHz crystal is divided by 80 down to 100 kHz. This stable signal is fed into one of the inputs of the PLL chip. The RF signal from the VCO is fed into the 'divide-by-N' counter. This counter will need to have "N" set to 999 to achieve a divide-down from 99.9 MHz to 100 kHz. When the VCO has a frequency of 99.9 MHz, both the inputs to the phase locked loop chip will have the same frequency and phase. The output pulses from the phase locked loop chip are fed into a loop filter circuit. This low-pass filter circuit smoothes and averages the phase locked loop pulses and produces a DC voltage which is applied to the frequency determining element of the VCO, which is usually a varicap diode. This slightly changes the frequency of the VCO, and the process is repeated. This is why the name 'loop' is used. The frequency is checked against the reference, the voltage is changed in respect of any frequency error, the voltage is applied to the oscillator, the frequency moves. This process is happening continually within the PLL chip. Adjusting the VCO until it is on frequency and will keep readjusting to keep it there. If we changed the 'divide-by-N' number to 997 then the PLL would adjust the VCO until both inputs to the phase comparator were equal in phase and frequency. This would force the VCO to now have an output of 99.7 MHz.

The Broadcast Warehouse phase locked loop system employs a modern chip that contains an oscillator for a quartz crystal, a divider for the reference, a 'divide-by-N' counter and a phase locked loop section (phase comparator). All of these sections are configurable by serial control. This control is fed from a Broadcast Warehouse software program contained in a microcontroller. The loop filter is built around a standard op-amp. Some exciter still use many logic chips for the various dividers and associated functions but the Broadcast Warehouse system uses only two, if we do not count the loop filter section.

The Problems of Phase Locked Loop Systems

The loop filter is the most crucial part of the phase locked loop system and plays the biggest part in achieving a high-quality exciter. The design goal is to have the PLL system get the VCO to the correct frequency fast and to appear transparent. When we FM modulate the VCO, we are moving

the frequency of the VCO in proportion to the audio signal we apply. The PLL circuit's job is to *correct* any frequency errors. *Hmmm...* Audio introduces frequency shifts and the PLL is trying to correct it. You can see that the two do not go hand-in-hand. If we design the loop filter too well, the quick response will strip the audio and not allow any deviation and hence no or minimum audio. If we relax the requirement to allow better audio to pass uncorrected, then we introduce other problems, such as long PLL lock time (the time it takes the PLL to correct any frequency or get the VCO to frequency). The ideal PLL system would allow us to get to frequency fast and then somehow relax itself and change the loop filter characteristics to improve the audio. We need the PLL circuit to not correct the audio (modulation) as much as when the VCO is genuinely off frequency.

Multispeed Loop Systems

Multispeed loop systems can be designed in many ways. We have seen and tested systems from complex to the very complex. We have chosen a system that has a minimum component count and still retains excellent performance. We have managed to keep the component count down by putting the intelligence of the system into software. The dual speed loop system we use is only one extra component above our standard single loop system. This component is an analog switch which has two of its switches placed across two of the resistors in the loop filter. When out-of-lock, the switch shorts out the resistors enabling more current to be dumped into the capacitors of the loop filter and hence, quicker charge time and faster lock up. When on-channel, the switches are opened. The hard part is knowing when to switch. Some other exciter's use the lock detect signals from the phase comparator chip to determine when the VCO is in lock. We have found this to be far from perfect as high level, low frequency content in the audio (heavy bass) can make the lock signal from the phase comparator read wrong. This could cause the transmitter to switch to fast lock when heavy bass is applied and then we would be back to square one, distortion.

Broadcast Warehouse has taken these lock detect signals from the phase comparator and connected them to a microcontroller where they are analyzed by a propriety software routine to determine whether the VCO is really *on* frequency or off frequency. The software can detect that the VCO is still on frequency even if we deviate the carrier with audio by 1 MHz. This enables us to obtain very, very low bass response with very, very low distortion figures and still have an accurate lock detect system and fast lockup time.

Circuit Description

The frequency determining element is formed by coil L1 and varicap VD1 together with capacitors C17 - C20. These components are used as part of a cascade oscillator whose output is then buffered by transistor T3. The RF output from T3 is impedance matched to the base of transistor T5 by RFT1, a 4-to-1 matching transformer. The high-power output from T5 is impedance matched by coils L2 and L3 and associated capacitors to the 50 ohm output socket CON2. These components also provide harmonic filtering.

The PLL circuit is primarily IC2 which is a serially-programmed PLL chip. The microcontroller IC3 reads the dial switches at power on and outputs a serial code to the PLL chip in a format that determines the output frequency that the PLL will try to lock the transmitter to. The PLL chip outputs control pulses to the loop filter built around op-amp IC4. The loop filter takes the sharp pulses from the PLL chip and converts them into a 'smoothed' signal ready to apply to the frequency determining component, varicap diode VD1. IC1 is an analog switch that shorts out two of the resistors in the loop filter which enables the transmitter to get on frequency faster. When the oscillator is on frequency, the analog switch switches out, which greatly improves the audio response of the transmitter. The microcontroller IC3 determines when to switch the analog switch

in and out by reading the lock detect signals from the PLL chip. The microcontroller can also use this information to switch off transistor T3 with open collector configured T4. This mutes the RF output when the transmitter is in an 'out-of-lock' condition. LED1 provides visual indication of the PLL locked condition.

Audio is fed into the modulation input connector CON1. It is passed through a high-frequency low-pass filter formed by C37, C38, and a ferrite bead to keep any RF from feeding back into the modulation circuitry. From here the signal passes to variable resistor VR1 where modulation levels can be set. From the output of the variable resistor the audio signal passes through resistor R30 and jumper J1. This jumper allows either capacitor C1 or C2 to be put in parallel with R30 forming a pre-emphasis filter. 0, 50, or 75 microseconds are selectable depending on jumper selection. From here, audio is fed via a resistive potential divider to the varicap diode VD1. The audio imposed onto VD1 causes the frequency of the transmitter to shift and modulation is achieved.

There is an expansion connector on the board to allow connection of other Broadcast Warehouse products, such as a LCD frequency selector. Connection details are provided with the relevant expansion product.

Assembly Instructions

This kit is not really a first-time kit builders project. If you have not soldered before, we recommend you get some soldering experience from a simpler project or get this kit assembled by someone who has previous experience in electronic construction and soldering.

1. Empty the contents of the kit and proceed to check all the components off against the component list. It is a good idea to tick off each component as you go through. When you have double checked all the parts, proceed.
2. We always start with the lowest height components first, which are the resistors. Insert each resistor and solder one at a time taking care to make a good solder joint and not to short across other pads/holes. Double check the component is the correct one before soldering.
3. Now insert and solder diodes D1 - D13 observing the polarity (SEE DIAGRAM). Do the same for varicap VD1 and inductor L4. Ferrite bead (marked FB) is next.
4. Next, it is time to insert the ceramic capacitors C1, C4, C8, C9, C10, C11, C14, C15, C18, C19, C20, C24, C25, C27, C28, C29, C30, C31, C32, C33, C34, C35, C37 and C38. These are non-polarized and can be inserted and soldered either way around.
5. Switches 1 to 3 should be next and these can be followed by the chip holders for IC1 - IC4. Make sure you line the notch on the chip holder with the notch on the ident on the printed circuit board (PCB). This will help you in making sure you insert the chip the correct way around in the socket. (See Diagram)
6. Variable resistor VR1 should be put in next followed by voltage regulators REG1 and REG2, and then transistors T1 - T4. LED LED1 should be next, marking sure the flat on the LED aligns with the flat on the silkscreen ident on the PCB. Transistor T5 can be inserted and soldered next. Leave the heatsink for T5 off for now.
7. Now insert the polarized electrolytic capacitors C2, C3, C6, C10, C13, C16 and C26 **MAKING 100% SURE** they are soldered in correctly. (See Diagram). The board has a positive symbol next to the positive hole of each polarized capacitor. Insert the negative stripe side away from the positive (+) marking. Now insert ceramic capacitor C17.
8. Insert and solder jumper J1. You may, if you wish, put the jumper tab on, but we recommend you wait until the end when we will configure the settings of the board. The pre-emphasis capacitors C22 and C23 can be put in next. Connectors 1 to 4 can be soldered in if you wish to use them. Variable capacitor VC1 is next.
9. Inductors L1 (metal can) and plastic type L2 and L3 can be inserted next, followed by crystal X1. The push on heatsink for T5 can now be pushed on, taking care to avoid twisting and damage to the transistor.
10. Make the RF transformer from the toroid core (blue/yellow ring) and twisted enameled wire as shown in the diagram.
11. Oh! You can now insert all the chips into their correct chip holders.

It is advisable that you check your work and all the components are where they should be and that there are no solder splashes or shorts underneath the circuit board. It is better to spend five minutes double checking everything, rather than risk damage at switch on due to a mistake during assembly.

If you are sure everything is OK, you can proceed to the setup and testing section.

Setup and Testing

Make sure you have the Broadcast Warehouse PLL+ 1 Watt Exciter assembled before proceeding, consult the assembly instructions for more info. Once constructed the PLL+ Exciter should not need any adjustments.

Power Supply

OK! Now that the unit is assembled, and you have double-checked for construction errors, we can get ready to switch on the unit. For correct full-band operation, you will need a regulated power supply that is capable of giving out between 13.8 and 15 volts. 13.4 volts is the minimum needed to allow the PLL to cover the full 87.5 to 108 MHz. 15 volts is a safe maximum voltage. Any more and the components may run too hot. If you do not supply a minimum of 13.4 volts then we cannot guarantee that the PLL will work correctly at the top of the band. 12 volts may only allow the unit to lock to 105 MHz or so. With the correct supply, connect a 50 ohm load to the PLL's RF output connector. A dummy load is preferred over an antenna.

Frequency Selection

Before you turn the power on, you must select your frequency.

The *FIRST* switch represents units of 10 MHz, where "8" would mean 80 MHz. (0 = 10 = 100 MHz)

The *SECOND* switch represents units of 1 MHz, where "9" would mean 9 MHz.

The *THIRD* switch represents units of 0.1 MHz (100 kHz), where "7" would mean 700 kHz.

Taking the above as an example; if we set SWITCH1 to **8**, SWITCH2 to **7**, and SWITCH3 to **9**, we would set the PLL to a frequency of 87.9 MHz. Example: $[(8 \times 10) + (9 \times 1) + (7 \times 0.1)] = 87.9 \text{ MHz}$

If you select an invalid frequency then the PLL lock LED will flash repeatably and no RF output will occur on any frequency.

To reset a new frequency, you must turn power to the unit off then back on again.

If you have a frequency meter you can also fine tune the frequency by the adjustment of VC1. For example, 99.200001 instead of 99.201341 MHz. Disconnect the audio before trying to adjust VC1. You will obviously need the unit on and powered up first before this adjustment can be made. If you don't have a frequency meter, don't worry. The unit will still be in spec.

Audio Input and Pre-Emphasis

Audio is fed in via the RCA/PHONO connector CON1. If you have an external stereo encoder, then remove the jumper J1. If you have an audio limiter with pre-emphasis capability then also remove the jumper J1. Otherwise, if no stereo encoder or limiter with pre-emphasis is in-line with the PLL you should configure jumper J1 to suit the pre-emphasis requirement for your region. 75 microseconds is used for the USA and Japan and 50 microseconds for the rest of the world. With your audio applied at the desired level to the PLL+, adjust variable resistor VR1 for 100% modulation (which is a maximum peak deviation of +/- 75 kHz).

RF Output

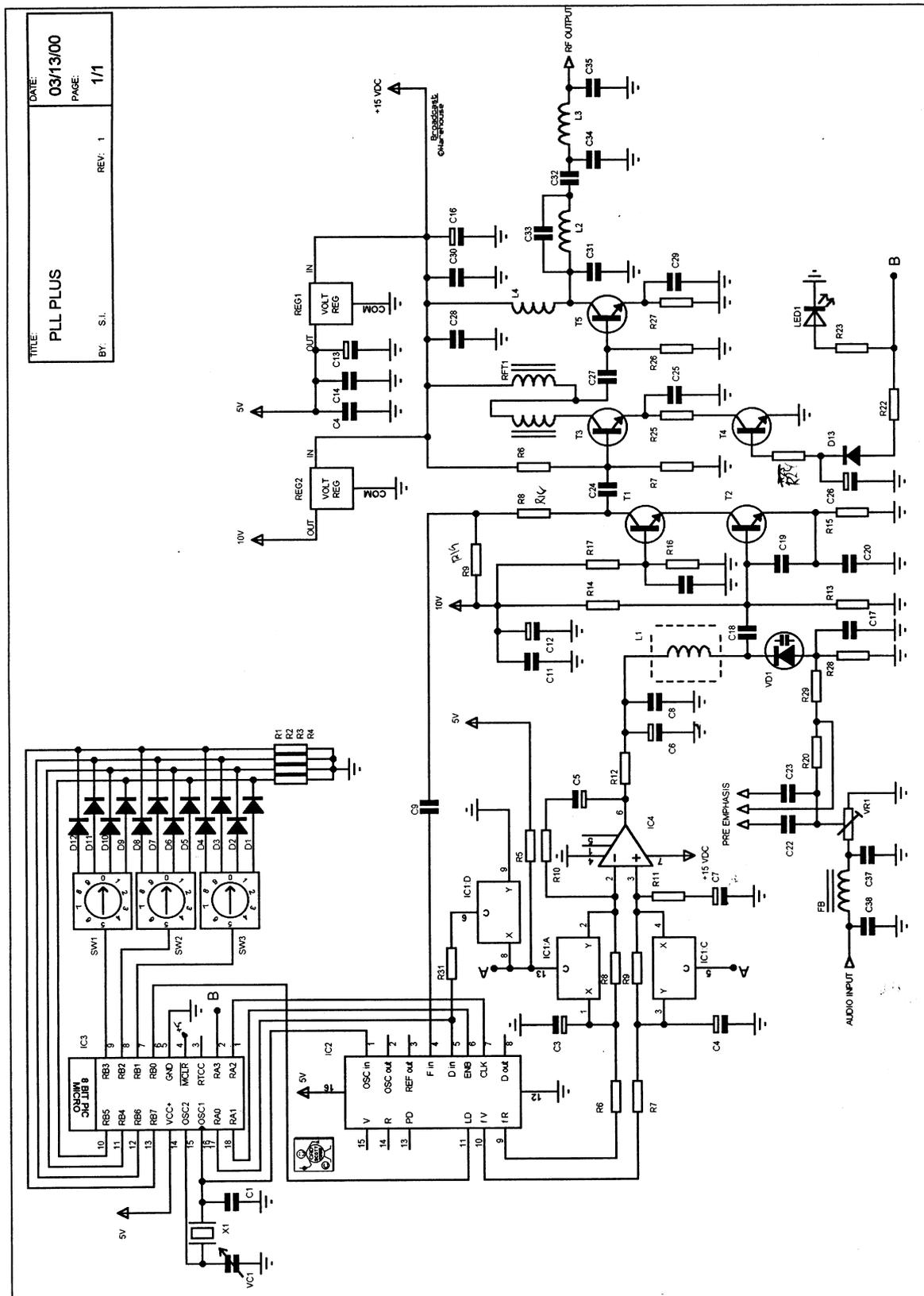
The RF output can be connected to CON2 or you can solder to the pads on the top or bottom of the board. The RF output power from the PLL+ is fixed at about 1 watt and can not be adjusted. Please do not alter the coils L2 and L3. They form part of the harmonic filtering and *should not* be adjusted. If you require less RF output power, then use a resistive attenuator formed from three resistors. Details are in any good radio handbook, such as the *ARRL Handbook for Radio Amateurs*. Always connect a good 50 ohm load on the RF output to avoid damage to T5.

Component List

Component	Value	Marking / Identification
R1, R2, R3, R4, R6, R7	10k	BROWN, BLACK, ORANGE, GOLD
R5, R8, R9, R31	330k	ORANGE, ORANGE, YELLOW, GOLD
R10, R11	330	ORANGE, ORANGE, BROWN, GOLD
R12, R16, R17, R22, R23	1.2k	BROWN, RED, RED, GOLD
R13, R20, R24	4.7k	YELLOW, PURPLE, RED, GOLD
R14, R30	12k	BROWN, RED, ORANGE, GOLD
R15, R26	220	RED, RED, BROWN, GOLD
R18	180	BROWN, GREY, BROWN, GOLD
R19	68	BLUE, GREY, BLACK, GOLD
R21, R28, R29	470	YELLOW, PURPLE, BROWN, GOLD
R25	10	BROWN, BLACK, BLACK, GOLD
R27	2.2	RED, RED, GOLD, GOLD
VR1	1k potentiometer	Small, yellow pot marked "102"
C1	39 pF	39 pF
C2, C3, C6, C35	2.2 μ F	2.2 μ F
C4, C8, C11, C14	100 nF	104 or 100N
C5, C7	470 μ F	470 μ F
C9, C24, C27	82 pF	82 pF
C10, C15, C29	10 nF	103 or 10N
C12, C13, C16, C26	100 μ F	100 μ F
C17	220 pF	220 pF
C18	4.7 pF	4P7 or 4.7 pF
C19, C30	27 pF	27 pF
C20, C33	56 pF	56 pF
C21	Not Used	
C23	4.7 nF	4700
C22	6.8 nF	6800
C25, C28, C32, C38	1 nF	102 or 1N
C31	12 pF	12 pF
C34, C36, C37	33 pF	33 pF
IC1	4066	4066

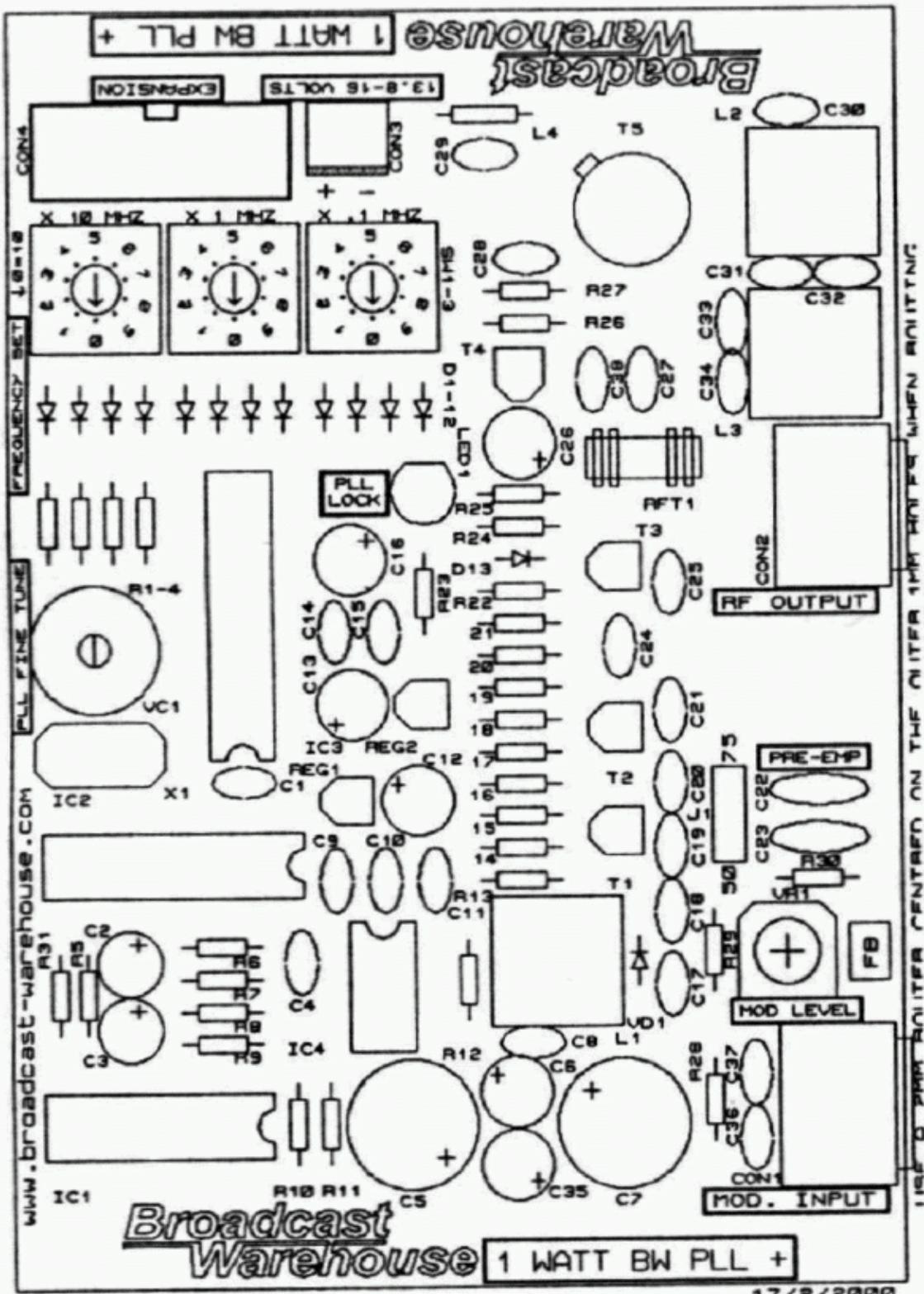
IC2	MC145170	MC145170
IC3	PIC16CXX	PIC16CXX
IC4	LF351	LF351
T1, T2, T3, T4	MPSH10	MPSH10
T5	2N4427	2N4427
L1	5–1/2 MC120	Metal can 00754
L2	2–1/2 S18	Red coil
L3	3–1/2 S18	Orange coil
L4	0.15 μ H inductor	Yellow axial – μ H15
LED1	Red LED	Red LED
REG1	78L05	78L05
REG2	78L10	78L10
X1	8 MHz crystal	8.000
VC1	5 – 65 pF trimmer	Yellow adjustable trimmer
SWITCH1, SWITCH2, SWITCH3	Decimal rotary switch	Black switch marked 0 – 9 in circle
D1 – 13	1N4148 diode	1N4148
VD1	BB909A varicap	Black axial with yellow stripe
CON1, CON2	RCA/PHONO connector	RCA/PHONO connector
CON3	2–pin Molex socket	2–pin Molex socket connector
CON4	10–way IDC connector	10–way ribbon socket
J1	3–pin jumper header	3–pin header
HEATSINK	Clip on heatsink	Black finned heatsink
RFT1	Toroid and wire	Blue/yellow ring with enameled wire
8–pin IC socket	8–pin IC socket	8–pin IC socket
14–pin IC socket	14–pin IC socket	14–pin IC socket
16–pin IC socket	16–pin IC socket	16–pin IC socket
18–pin IC socket	18–pin IC socket	18–pin IC socket
PCB	Black board	You are joking!

PLL PLUS 1 WATT EXCITER



TITLE: PLL PLUS
 BY: S.I.
 REV: 1

DATE: 03/13/00
 PAGE: 1/1

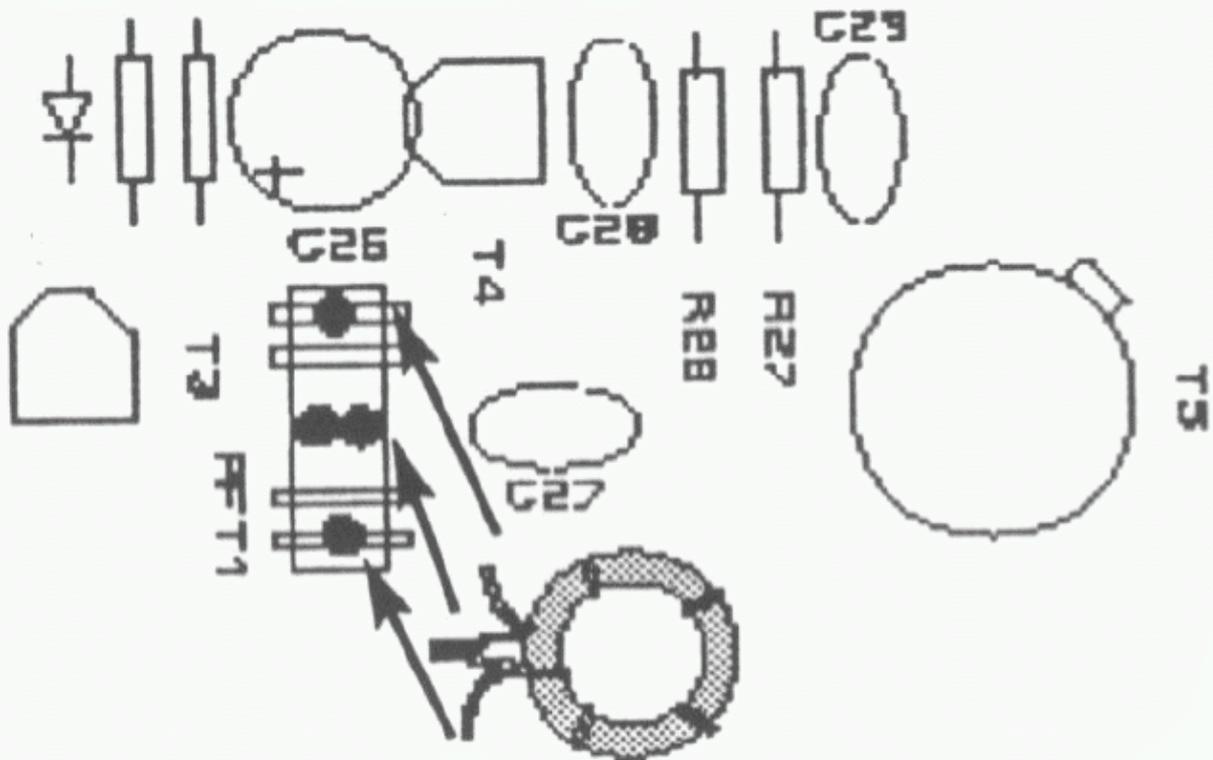


PCB Layout

4 to 1 TRANSFORMER CONSTRUCTION

Take the supplied twisted enamel wires and pass FOUR turns through the supplied ferrite ring (see diagram below). Take a color from one end of the twisted pair and join with the opposite color from the other end of the twisted pair. These two wires solder to the two pads as shown below. The other wires of the twisted pair connect to the other pads as shown (which wire to which pad is not significant).

N:B: Please make sure you melt the enamel off the ends of the wire prior to soldering to the pads, this ensures a REAL connection. Keep the wires as short as possible after coming off of the ferrite ring.



RF Transformer Construction

GBPPR Speech Jammer

Introduction

"When even one American – who has done nothing wrong – is forced by fear to shut his mind and close his mouth, then all Americans are in peril."

--- Quote from Harry S. Truman, 33rd President of the United States of America.

Freedom of speech is under attack. Anyone who is awake has noticed a sharp increase in the number of people silenced for exposing corruption and extremism in the Obama administration. From Helen Thomas to Pat Buchanan, Glenn Beck to Andrew Napolitano, Mel Gibson to Kirk Cameron – the message is clear. You can only speak your mind if your political views follow those of a small group of oligarchs who wish to [rule the world](#). So much for that "diversity" or "tolerance" liberals are always harping about... Conservative/Gentile/Christians who have the same views as our Founding Fathers are routinely attacked and silenced, while loony left-wing Marxists are paraded around and maintain control of all the major news or media outlets.

Sometimes the attacks on speech are subtle. For example, as I type this, the price of gas is almost \$4 a gallon. The reason, of course, is that oil is priced in U.S. dollars and as the Federal Reserve (which is neither "federal" nor has any reserves) creates interest-bearing loans (i.e. never-ending debt we call "money") out of thin air, the purchase power of the U.S. dollar will continue to fall. Just don't count on any politicians (except Ron Paul) or media outlets questioning the purpose, or even the constitutionality, of this "Federal Reserve" scheme.

Another example of suppressed speech (and press) involves the out-of-control union corruption going on in Wisconsin right now. I'm sure you've heard all the stories about the massive corruption and drug/steroid use by the Green Bay Police Department, the racketeering and general incompetence within the Green Bay Public School District, or the shady insider real estate dealings with land around Lambeau Field by Green Bay Packers' employees. *Oh wait...* That's right... Not a peep in the mainstream media! Thankfully, I hate the Packers, I hated my teachers, and I hate the cops:

```
FROM: 211A.I DON'T LIKE THIS CALL.I DO LIKE THE.BLOND  
      IN THIS CAR UP HERE,SHOULD I MAKE.A TRAFFIC STOP?
```

Reclaim Wisconsin! Don't let it turn into [Illinois](#) or [California](#)! But how can ***YOU***, the legal citizen and tax payer, fight back? Simple, study the enemy's methods and use them yourself!

In U.S. Patent 6,052,336 "[Apparatus and Method of Broadcasting Audible Sound Using Ultrasonic Sound as a Carrier](#)" by Austin Lowrey, he describes several methods to impede communication between a speaker and a crowd.

Lowrey's patent states: *"One of the most interesting techniques includes playing back to a speaker his/her own voice with a slight delay (less than a second). The speaker stutters and trips on his/her words unless he/she slows down his/her rate of delivery a great deal."* How rude!

More recently, Kazutaka Kurihara at the [National Institute of Advanced Industrial Science and Technology](#) and Koji Tsukada at [Ochanomizu University](#) (Japan) have taken this concept further by constructing a portable "speech jammer" using a conventional Sony directional microphone, an electronic audio delay line, and a directional parametric speaker for the rebroadcast.

The student's paper "SpeechJammer: A System Utilizing Artificial Speech Disturbance with Delayed Auditory Feedback" delves into the psychological factors of this effect in a little more detail. That paper should be read before hand, as those details won't be covered here.

The GBPPR Speech Jammer covered in this article will use what I've labeled the "Obstruction By Acoustic Meddling Action (OBAMA) effect." This is similar to the stuttering and stammering effect you get when your teleprompter breaks down and you're actually forced to think on your own.

I first noticed the OBAMA effect back in 2009 when O'bummer came to Green Bay to give a speech and I was monitoring all the media's wireless microphones for security holes. I discovered that some of the people Obama "chose" at random during the town hall question portion of his speech were pre-interviewed a day before. When these people were "randomly selected" by Obama to ask their question, his teleprompter gave him a quick and clear answer for him to recite. But not all the questioners were planted... There were a few truly random people selected to ask Obama a question. When this happened, Obama would suddenly start to stutter and stammer as his tiny, dog-eating, low-I.Q. Kenyan brain searched for an answer to suit his billionaire banker handlers. Change!

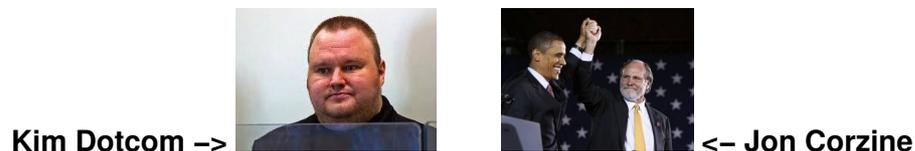
"And here's how we're going to work it. There are no prearranged questions here. You just raise your hand. I haven't pre-selected anybody..."

"Well, my name is Jean Marsch. I am the president of the Green Bay School Board, and I'm also a registered nurse and I work at St. Vincent Hospital..."

--- Quotes from the June 11, 2009 [townhall meeting](#) with Barack Hussein Obama at [Southwest High School](#) in Green Bay, Wisconsin. The "permission slip" skit was also prearranged.

Now, do you really think Obongo called on the fucking Green Bay Public School Board president at "random?" *Nope!* Change!

Here's some other examples of suppressed speech or media stories:



The man on the left ran a website for people too stupid to understand FTP or torrents. He's probably going to prison. The Democrat on the right "lost" \$1.2 billion while at MF Global. Corzine is helping run Obama's reelection campaign. He'll never go to prison...



The man on the left does the job the people of Arizona elected him to do. He's now being attacked for "racial profiling." Eric Holder also "racial profiled" when he told Congress showing an ID to vote hurts minorities – and by "minorities" he meant non-Whites, when the *REAL* minorities are hard-working, law-abiding, tax-paying White people! The father of the Jewish supremacist on the right helped blow up the King David Hotel in 1946. Rahm was Obama's Chief-of-Staff and is now the mayor of Chicago. The DOJ will never go after Emanuel family for terrorism...

Overview

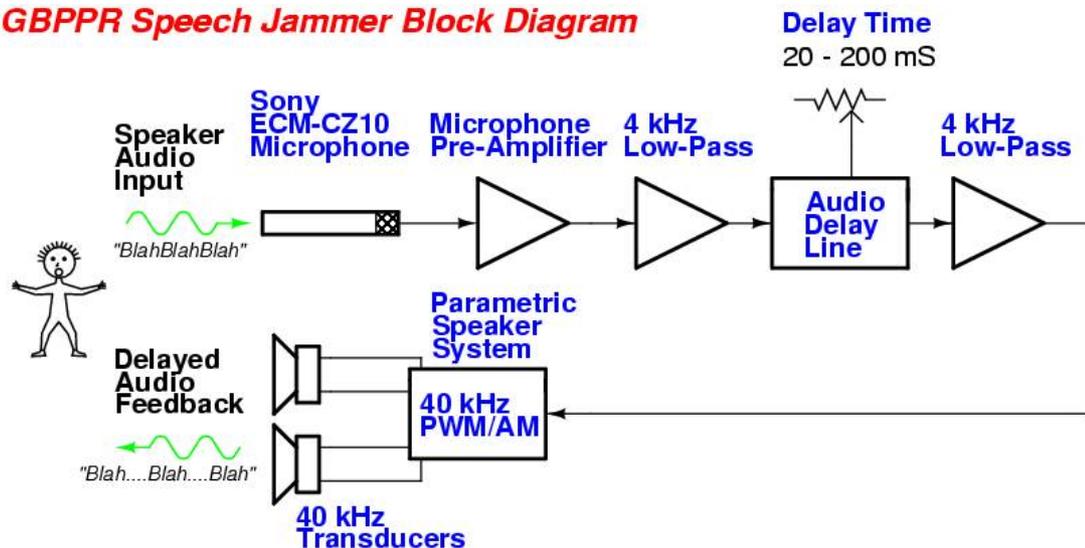
The GBPPR Speech Jammer works by picking up the audio from the target and rebroadcasting it back to them with a slight delay. This interferes (or overpowers) the natural auditory/bone vibrations they received when delivering normal speech. The effect causes slight confusion within the target, forcing them to slow or completely stop speaking in order to regain their composure.

The audio is picked up via a Sony ECM-CZ10 directional electret condenser microphone feeding a low-noise microphone pre-amplifier circuit. The microphone audio is filtered around the speech band (approximately 300 – 4000 Hz) to remove any out-of-band frequency components and to maintain intelligibility.

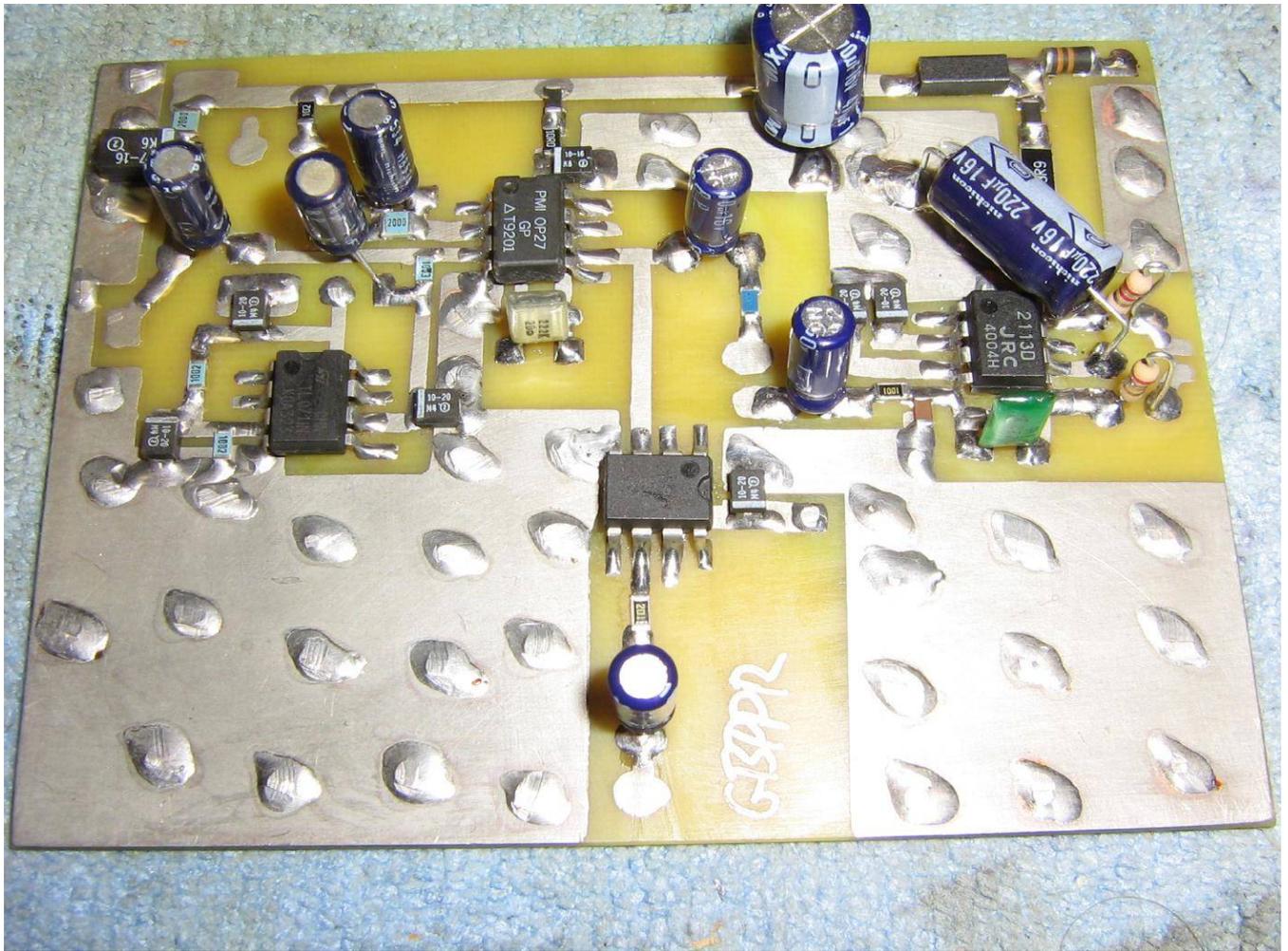
The filtered audio is then sent to a Matsushita/Panasonic MN3005 4096-stage "bucket-brigade" audio delay line. The delay time setting is controlled by a Matsushita/Panasonic MN3101 clock generator/driver specifically designed for use with the MN3000-series chips. A panel-mounted 250 kohm potentiometer varies the overall audio delay time from 20 to 200 milliseconds. The output of the MN3005 is then low-pass filtered again to remove any "noise" introduced during the delay stage. The MN3005 and MN3101 are no longer in production, but are available on eBay from time-to-time. They were widely used by the music/guitar effects community, so you may want to search your local swapfests for old guitar effects pedals. The MN3008 (and others in this series) may also be used, but it only has a maximum delay time of around 100 milliseconds. It is possible to series several chips together to increase the overall delay time. You may want to make the first stage with a fixed delay feeding a variable delay final stage.

The delayed audio is rebroadcast back to the target using a "parametric" directional speaker system. This is a method for producing extremely directional audio by modulating it onto an ultrasonic (40 kHz) carrier wave. Because air is non-linear (air compresses faster than it can uncompress), this action acts like a giant diode from the parametric speaker to your ear. By pulse-width or amplitude modulating the 40 kHz carrier with your audio, only the person within the narrow beamwidth of the parametric speaker will hear the demodulated audio. The problem is this method also highly distorts the recovered audio. Commercial parametric speakers "pre-distort" the audio using DSP processing, which is where the money really is. The experimental parametric speaker described here won't do this, so the final audio quality isn't great. It still sorta works and is useful for remotely inducing tones into people's heads. Commercial parametric speaker systems are available from [Holosonics](#) with their "Audio Spotlight" series. These speakers are still quite expensive, but the audio they produce is crisp and clear.

GBPPR Speech Jammer Block Diagram



Pictures & Construction Notes



Overview of the low-noise microphone pre-amplifier circuit.

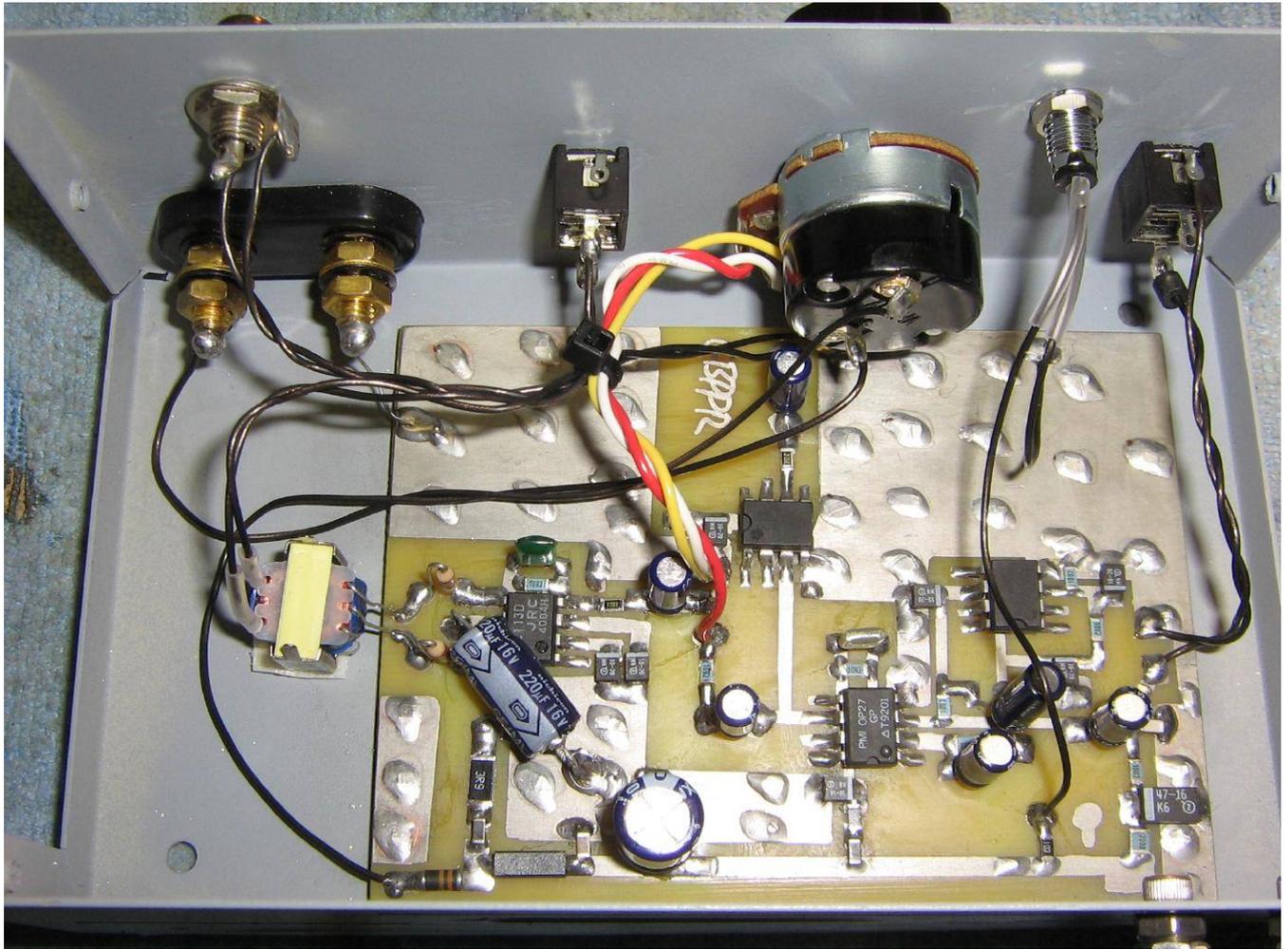
The **Microphone Input** is on the left-side, the optional **Low-Z Output** (headphone) is on the right-side, and the **Line Level** audio output is along the bottom.

The pre-amplifier is nothing really fancy, just an OP27 op-amp with an active split-rail bias for added stability. An active bias can both source and sink current, while a resistive divider can not. This also eliminates the need for a negative voltage supply.

The OP27's feedback network is configured for around 60 dB of gain in the "speech band." It rolls off anything above 7 kHz and below 100 Hz. The amplified microphone audio output from the OP27 is split into two legs.

One leg feeds a TL071 op-amp to act as a buffered **Line Level** output and the leg other feeds a JRC NJM2113 (or Motorola MC34119) low-noise audio power amplifier for driving standard low-impedance (8/16/32 ohm) headphones or a speaker.

The following pictures of the circuits may vary from the schematics due to tweaking, but the schematics are correct. TL071 op-amps were used to ease debugging, but they can be replaced with TL072 or TL074 op-amps to reduce the component count.



Mounting the low-noise microphone pre-amplifier circuit inside an old printer switch case.

Banana jacks on the left supply the circuit's regulated +12 VDC input. There isn't an onboard voltage regulator as the pre-amplifier will actually work over a large range of voltages. Just be sure the DC power is clean.

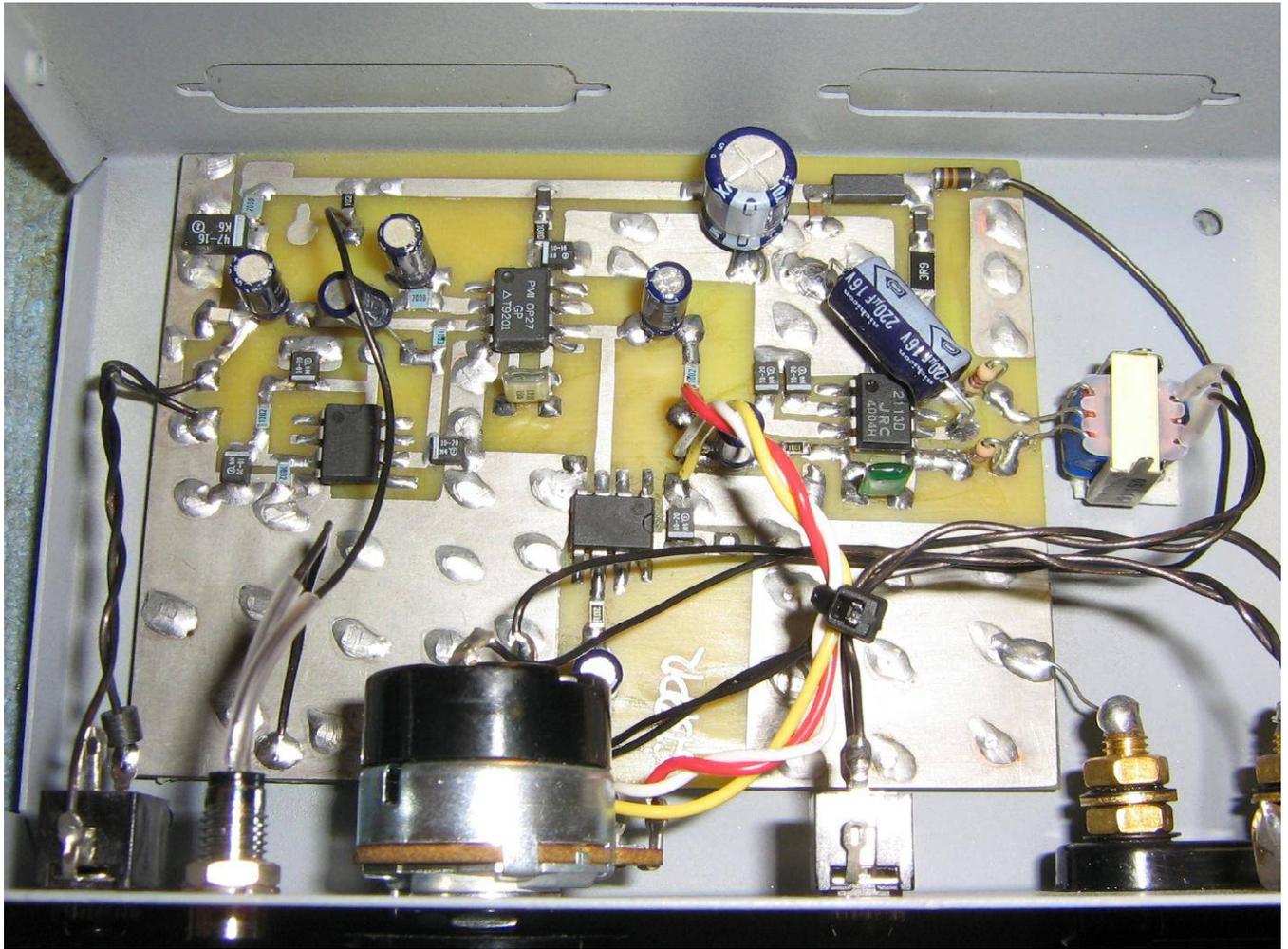
Above the banana jacks is a RCA jack for the **Line Level** output signal.

To the right of the banana jacks is a panel-mounted 1/8-inch stereo headphone jack. This jack's ground tab *MUST* be isolated from the metal case when using the NJM2113. There are special little plastic washers made for this purpose, but they may be hard to find. An audio isolation transformer will also work.

The panel-mounted 10 kohm potentiometer controls the headphone volume from the NJM2113 audio amplifier. It doesn't effect the level of the **Line Level** output signal. The potentiometer has a built-in power switch.

On the right is another 1/8-inch jack for the input from the Sony ECM-CZ10 directional microphone and a panel-mounted LED as a power indicator.

1% metal-film resistor should be used and the audio coupling capacitors should be non-polarized and polyfilm-based to avoid microphonics.



Alternate view.

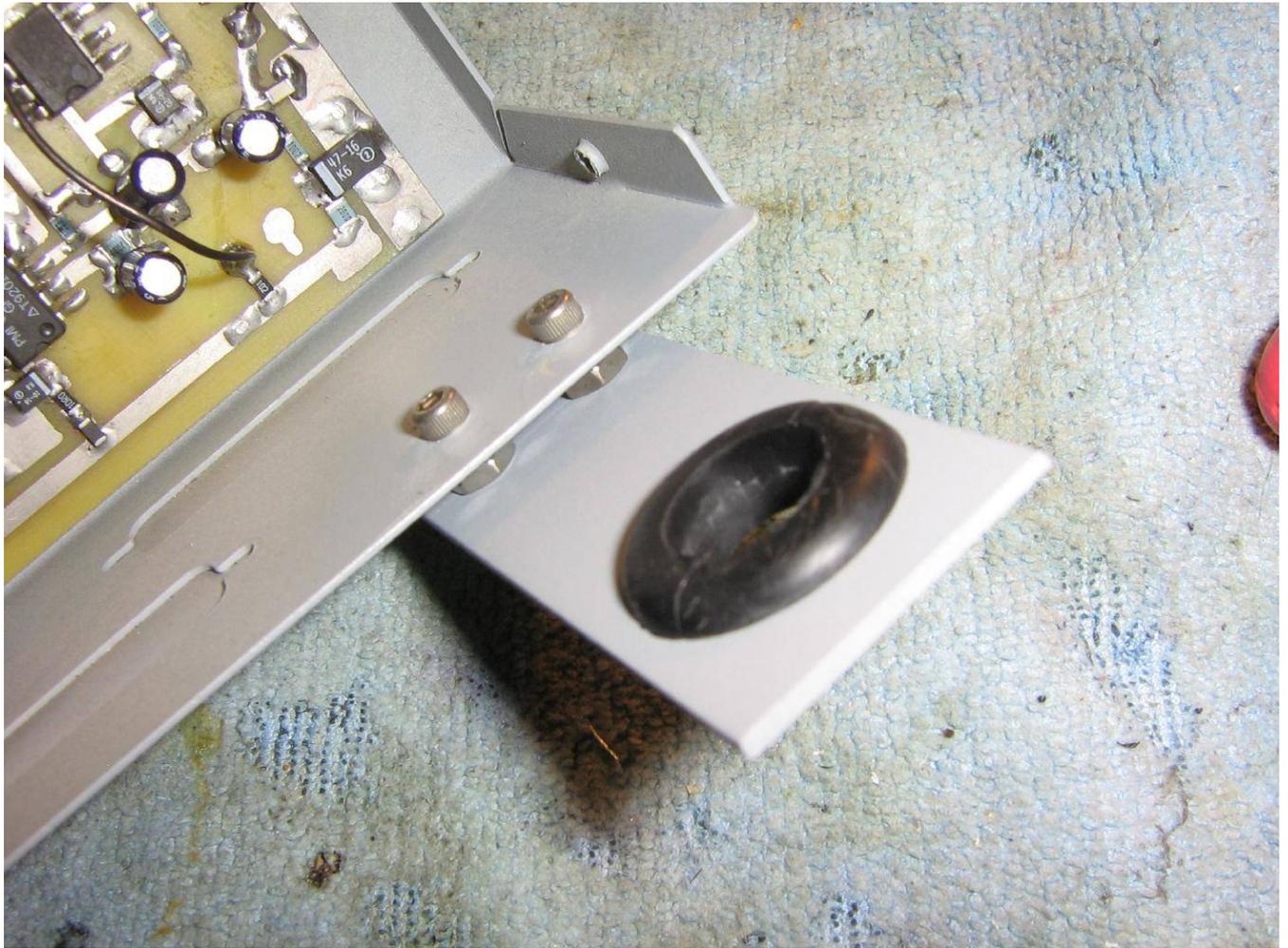
Note the addition of an optional isolation transformer (1000 ohm to 8 ohm, center-tapped at 500 ohm) on the output from the NJM2113.

If you don't have the proper plastic isolation washer for the 1/8" stereo headphone jack and you are mounting the circuit in a metal case, you *MUST* use an isolation transformer.

The isolation transformer shown above is similar to Radio Shack 273-1380. It's probably not ideal, but it works and it also acts as an additional bandpass filter for the final audio output signal.

When using the Radio Shack audio isolation transformer, connect the black/green wires to the output of the NJM2113 and the red/white wires to the headphone jack.

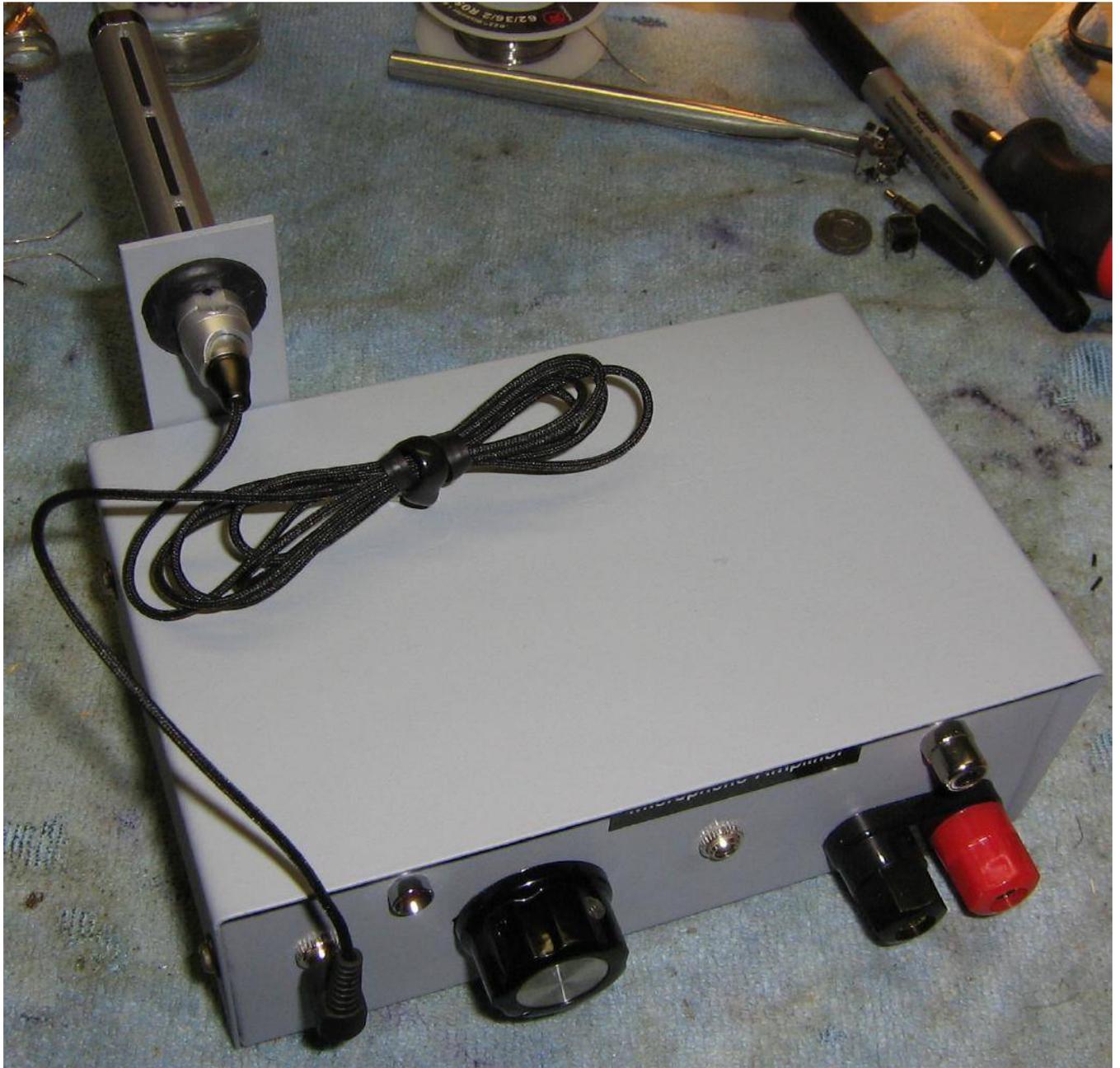
All the wires connecting the audio jacks and volume potentiometer should be as short as possible and twisted together to prevent oscillation.



Optional ECM-CZ10 microphone mount.

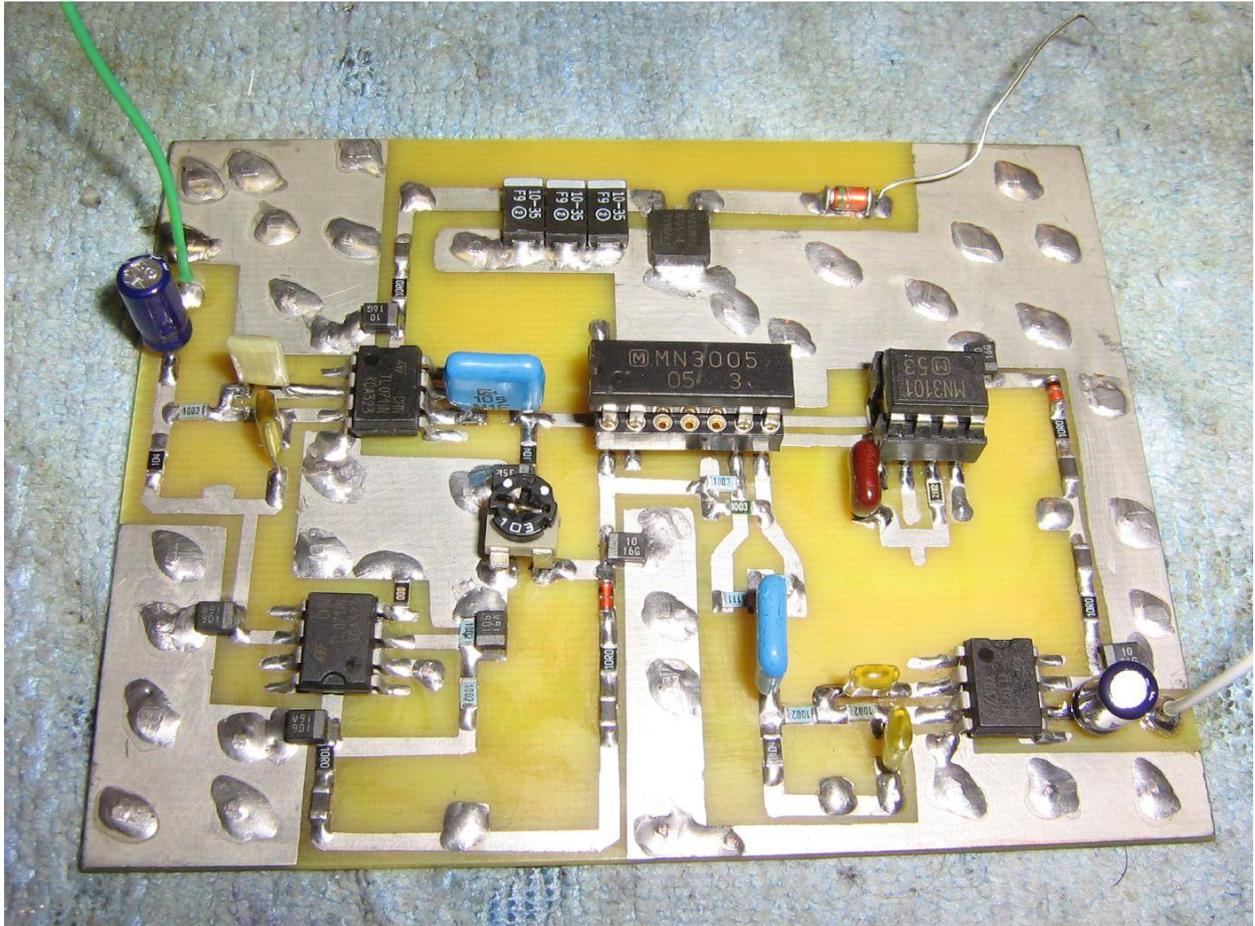
It's just a small piece of aluminum plate with a hole drilled in it which was then fitted with a 1/2-inch rubber grommet.

Attach it to the rear of the microphone pre-amplifier's case.



Completed low-noise microphone pre-amplifier with an attached Sony ECM-CZ10 microphone.

Be sure the slots along the microphone's phase tube are not covered up.



Overview of the variable audio delay circuit.

The **Audio Input** is on the left-side and the **Delayed Audio Output** is on the right-side.

The MN3005 and MN3101 normally requires operation at -15 VDC, but we are going to cheat a bit here by reversing the V_{dd} and GND connections via isolation diodes. This will allow the chips to run at positive voltage.

Be sure to take this into account when reviewing the datasheets for the MN3005/MN3101 as the connections will look "backwards." The polarity of any polarized capacitors in the circuit should also be double-checked.

The delay time of the MN3000-series chips is set by an external MN3101 clock generator:

$$DT = 0.5 * N / CLK$$

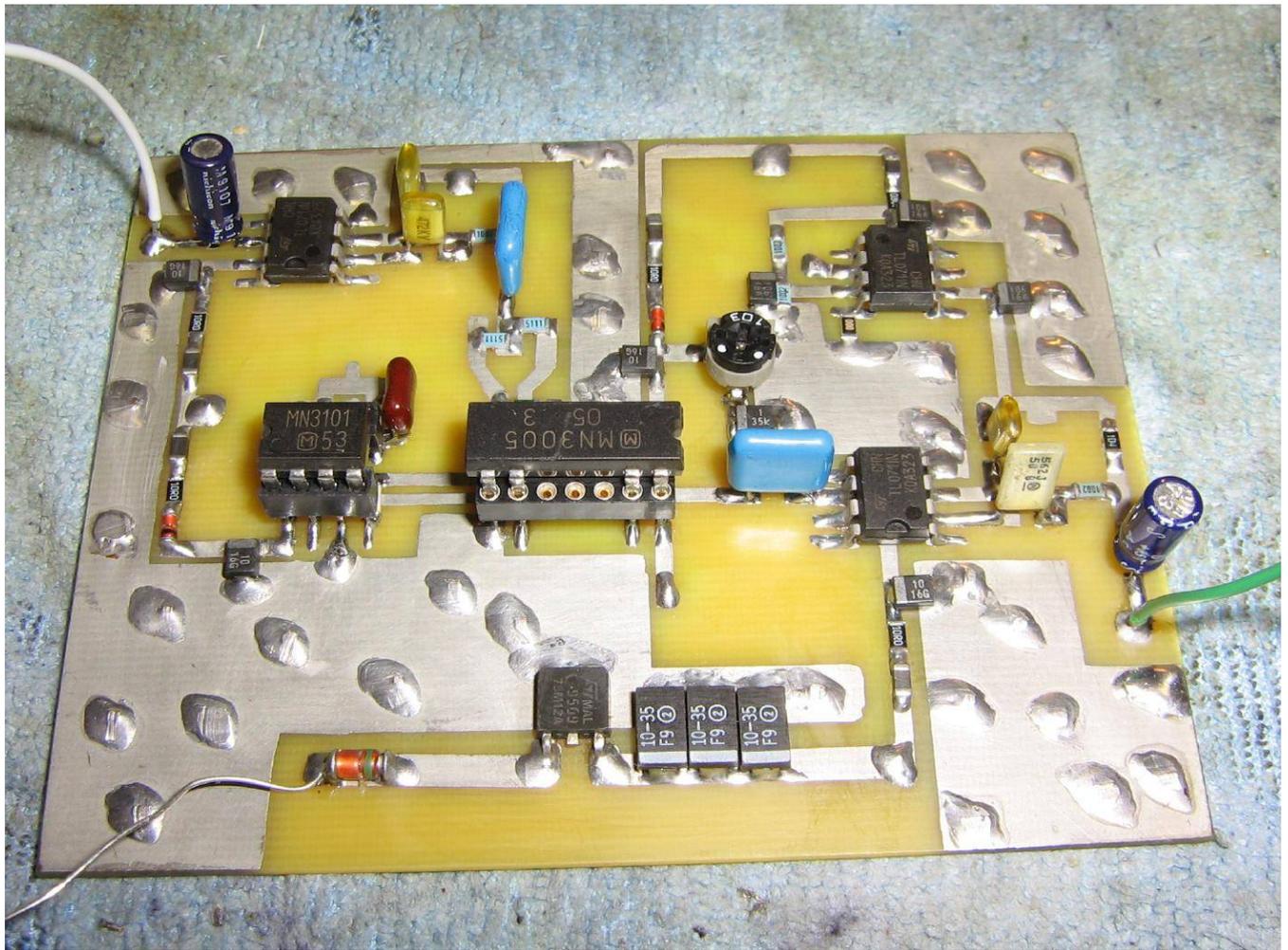
DT = Delay time in seconds.

N = Number of on-chip delay stages. (MN3005 = 4096, MN3008 = 2048, MN3007 = 1024)

CLK = Clock pulse frequency in Hertz.

The MN3005 has 4096 stages and requires a (dual) clock pulse frequency between 10 – 100 kHz. This works out to a variable delay time between 20 – 200 milliseconds. A 10 kHz clock pulse on the MN3005 gives the maximum delay time of 200 milliseconds.

For speech jamming, longer delay times seem to work better. Randomizing the delay times works the best, but this adds complexity to the circuit.



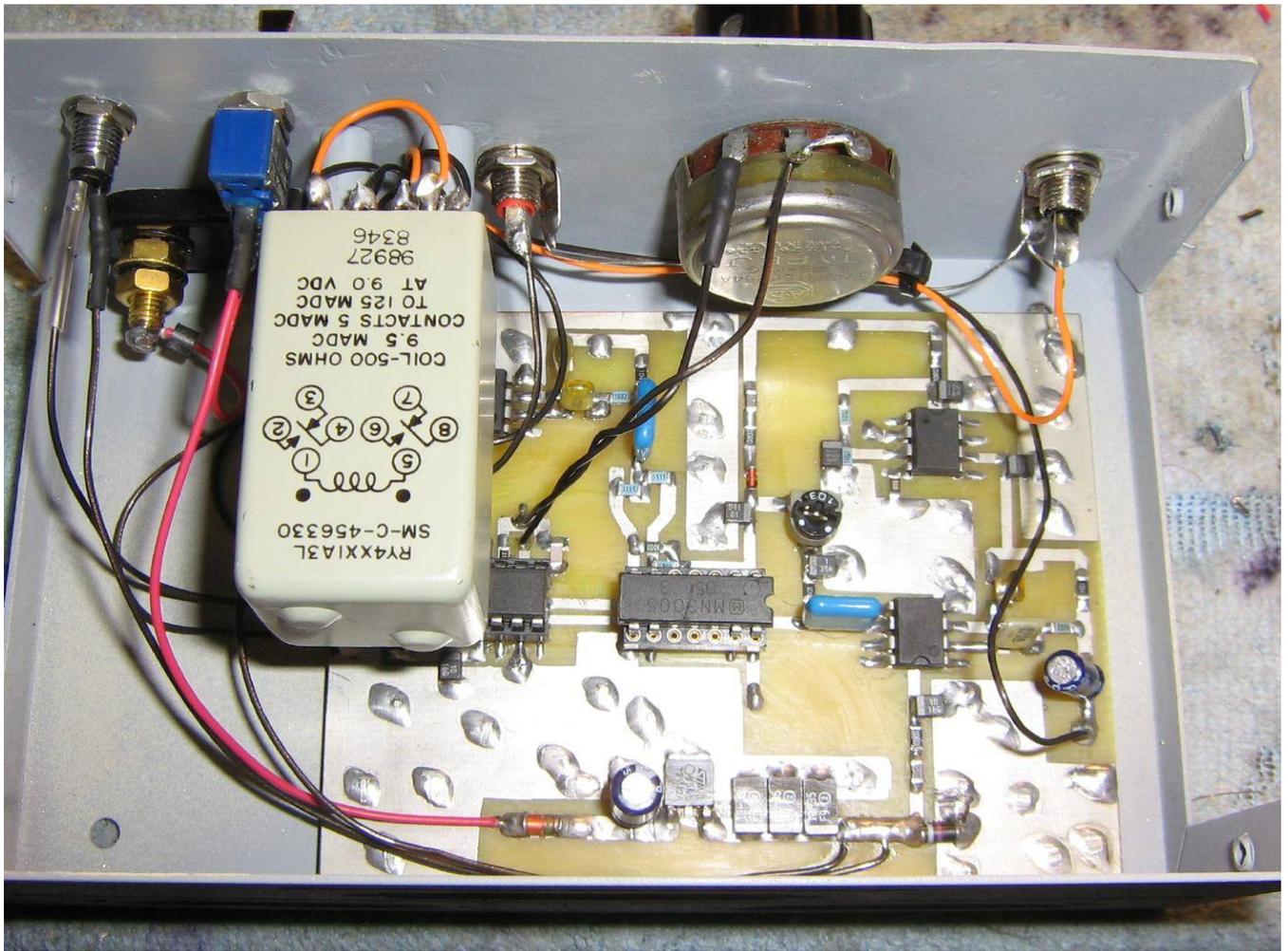
Alternate view.

The MN3005 and MN3101 should be run at 15 volts, but they will work at 12 volts. This is regulated with a 78M12 voltage regulator on the main DC input. This helps to keep the MN3101 clock oscillator from drifting in frequency. The circuit should be fed from +15 VDC.

The 10 kohm bias adjust potentiometer should be adjusted for minimum distortion of the output delayed audio signal. Normally, then will be at or just slightly below 1/2 the V_{dd} voltage, which is 6 volts in this case.

The low-pass filter on the delayed audio output is used to eliminate any clock pulse feed-through.

Note that several software audio editing and recording packages have a feature to introduce a delay in real-time audio playback. This would eliminate the need for a hardware-based audio delay line, but where's the fun in that?



Mounting the audio delay line circuit inside an old printer switch case.

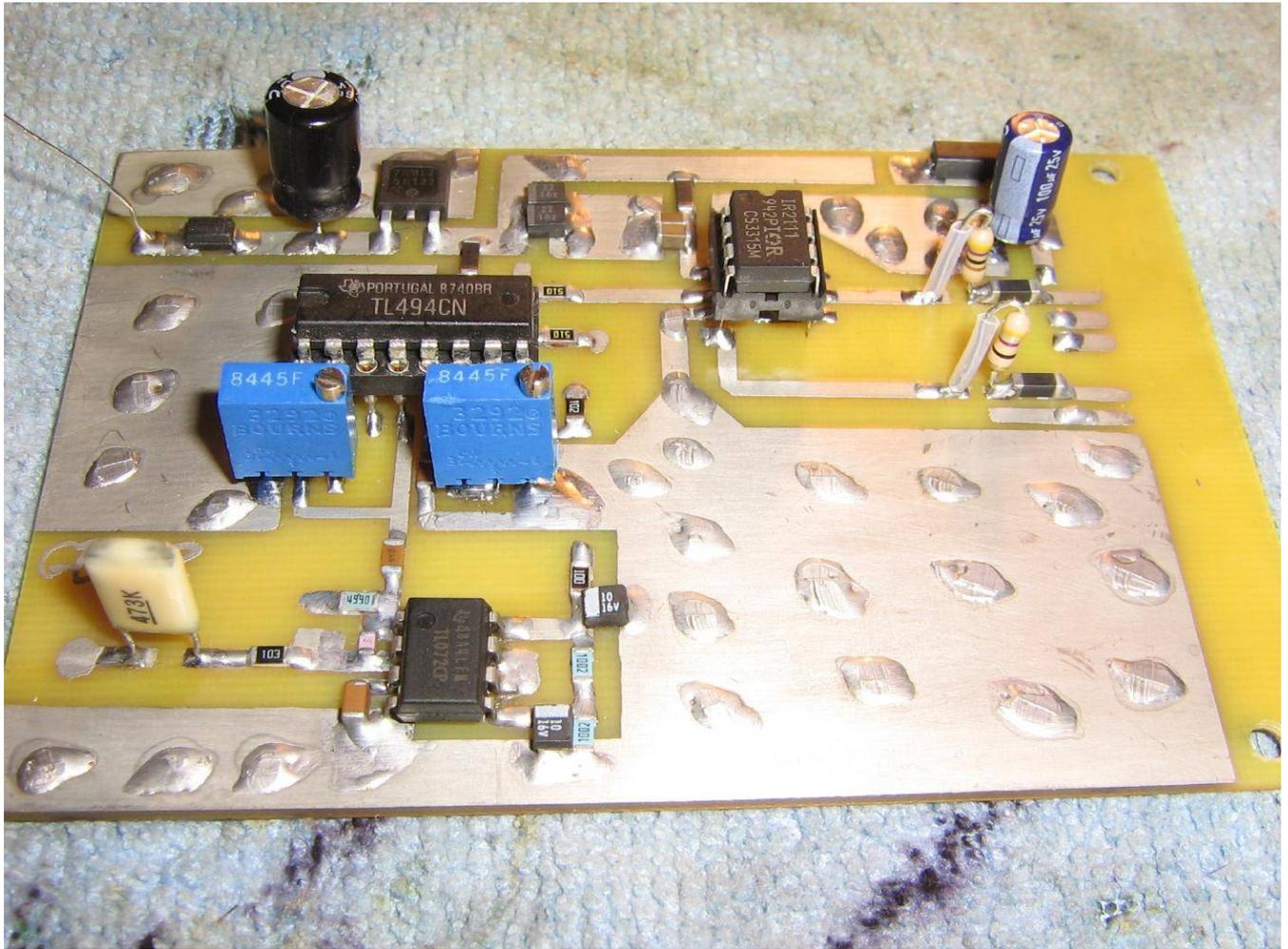
Banana jacks on the left supply the circuit's main +15 VDC input and a 78M12 voltage regulator will handle the rest.

RCA jacks are used for the audio input/output.

I added an optional DPDT relay to toggle the delay line circuit in-and-out of the audio path. When no power is applied to the circuit the audio passes through unaffected.

The panel-mounted 250 kohm potentiometer tunes the MN3101 clock generator from approximately 20 to 200 kHz. The MN3101 has an onboard "divide-by-2" stage which provides the dual 10 – 100 kHz clocks for the MN3005.

The audio delay line circuit is fed from the **Line Level** output on the microphone amplifier.



Overview of the experimental 40 kHz Pulse-Width Modulator (PWM) circuit.

The **Audio Input** is on the lower-left and the output **40 kHz PWM Signal** is on the right-side.

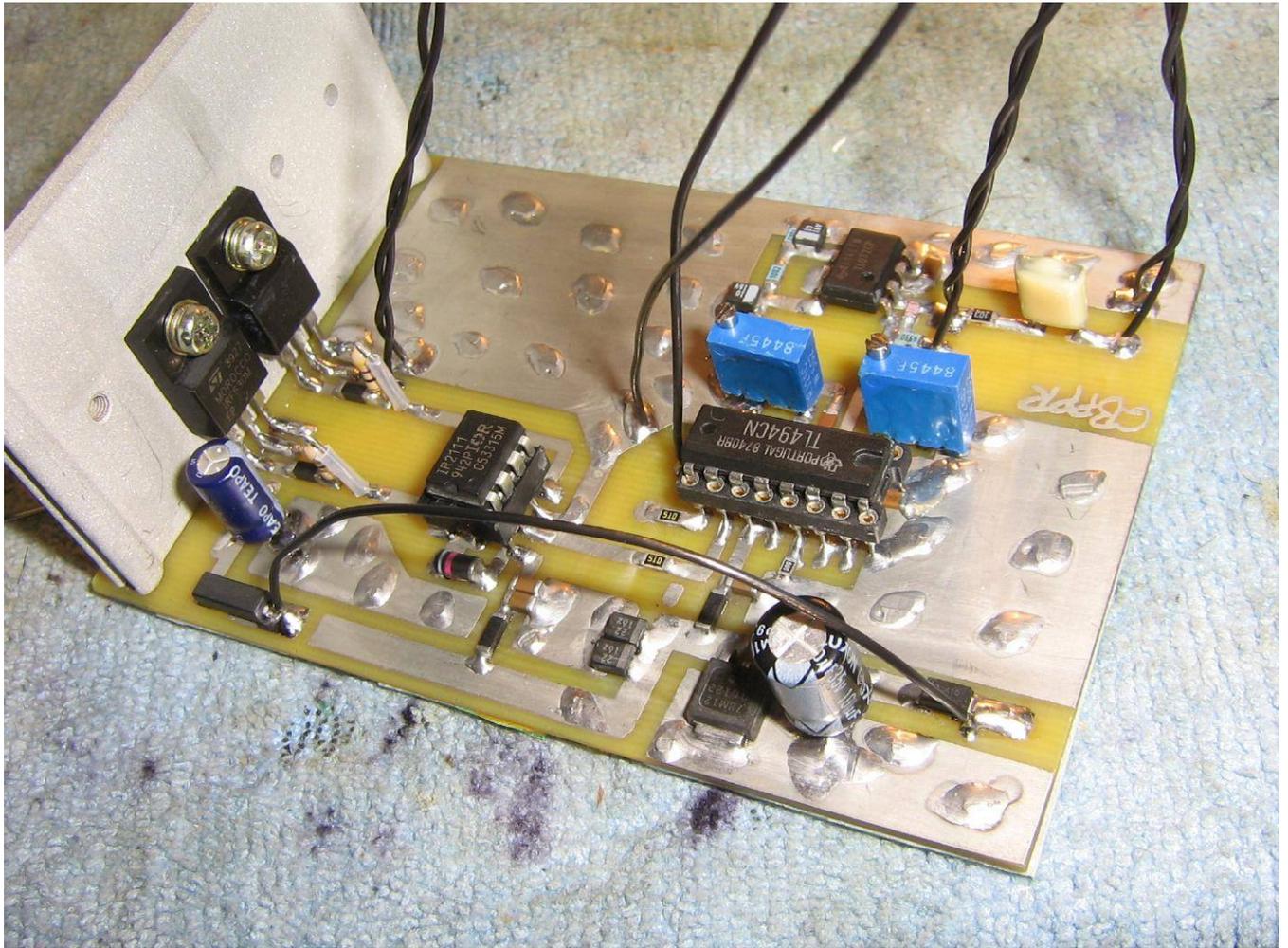
This circuit takes an audio signal and pulse-width modulates it onto a 40 kHz carrier wave. This modulated carrier is then output via two MOSFETS in a half-bridge arrangement, similar to a class-D audio amplifier.

The modulator is based around a standard Texas Instruments TL494 PWM control circuit and an International Rectifier IR2111 half-bridge driver controlling two IRF630MFP N-channel MOSFETS.

Two multiturn 5 kohm potentiometers control the width and frequency of the TL494's pulse output. Since the transducers we'll be using for the speaker system are designed to work at 40 kHz, the TL494 will be tuned to oscillate at around 40 kHz.

A simple TL072 op-amp gain stage drives the TL494 from the remote audio source. A panel-mounted 100 kohm potentiometer controls the gain (1-10) of this op-amp.

The TL494 likes to see higher voltages (around 2 to 3 V_{p-p}) when "audio modulated" like this, so additional amplification may be required on the input signal.

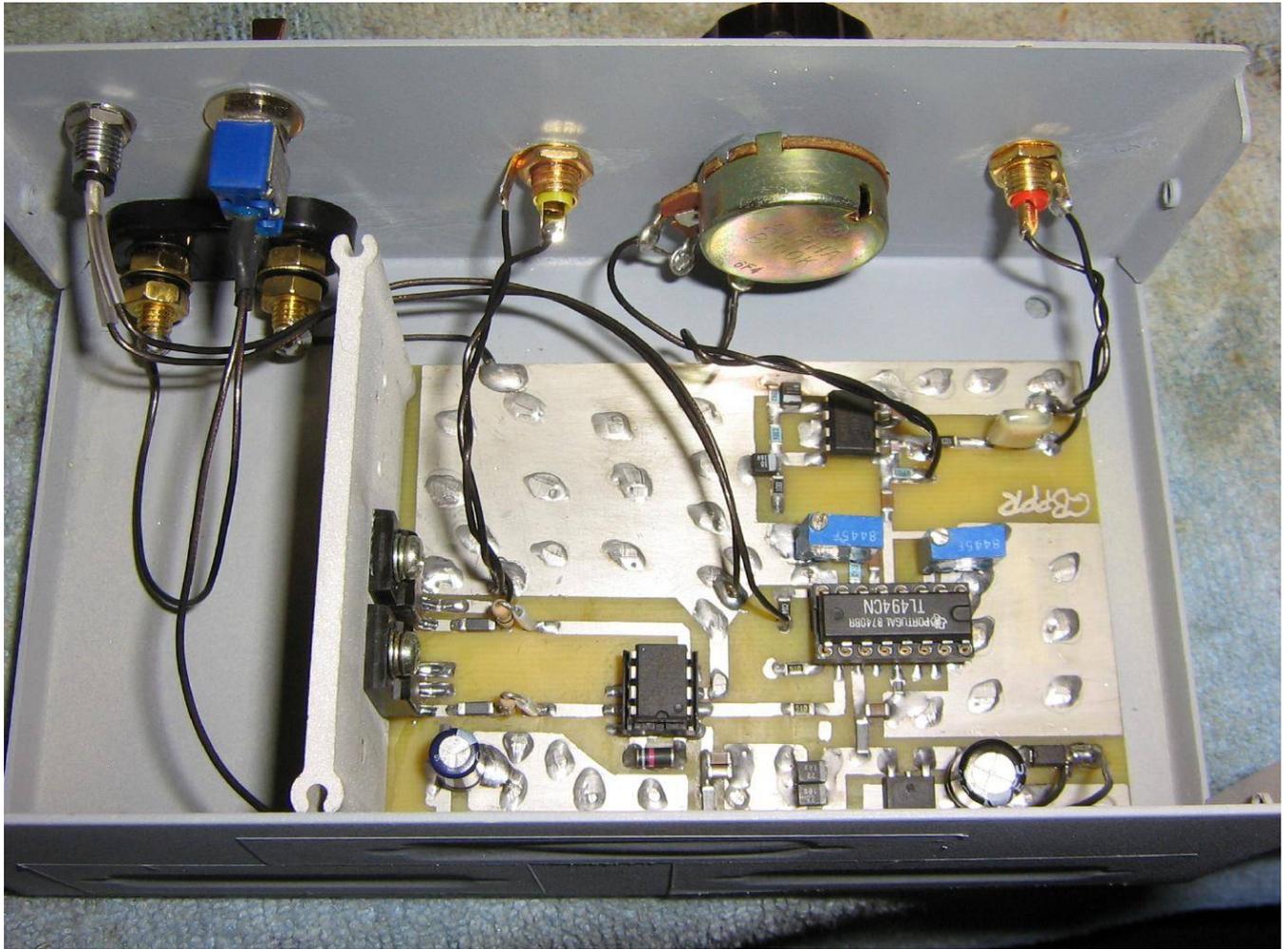


Alternate view with the IRF630MFP MOSFETs installed.

The heatsink is probably optional, but makes a handy mounting point for the MOSFETs. The IRF630MFP MOSFETs have an isolated tab, so mounting them is quite easy and doesn't require special isolation hardware.

You can sometimes salvage IRF630MFP MOSFETs from old computer monitors.

This modulator design is based on the article "Ultrasonic Directive Speaker" by Kazunori Miura in *Elektor*, March 2011.



Mounting the 40 kHz PWM circuit inside an old printer switch case.

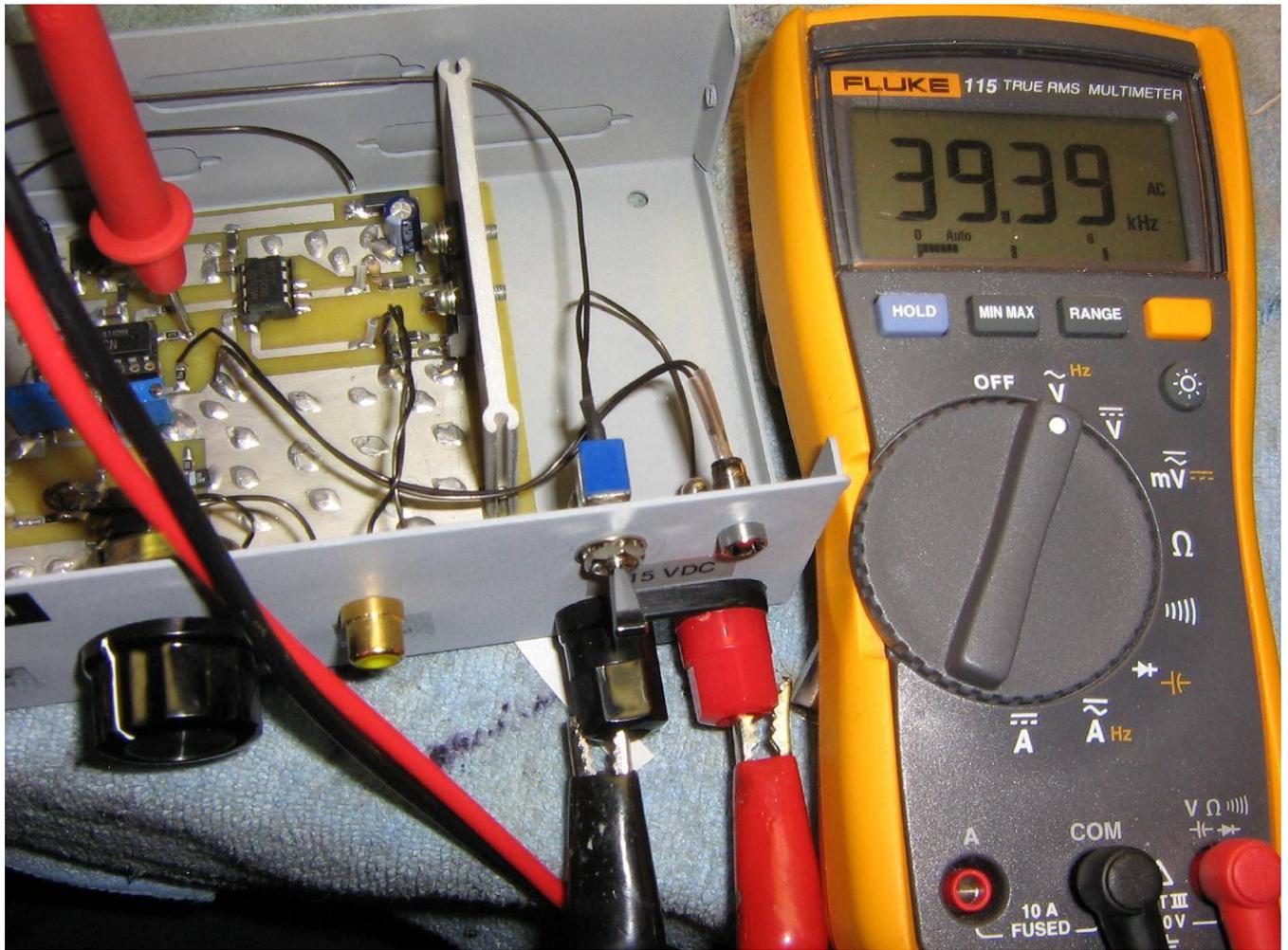
Banana jacks on the left supply the circuit's main +15 VDC input and a 78M12 voltage regulator will handle the rest.

The IRF630MFP MOSFETs should *NOT* be run through the voltage regulator as they have a high peak current draw, and this will also give you the option for running them at a higher voltage.

RCA jacks are used for the audio input and 40 kHz PWM output.

The panel-mounted 100 kohm potentiometer controls the gain of the TL072 op-amp input stage.

The pulse-width modulator circuit is fed from the **Delayed Audio Output** on the delay line circuit.

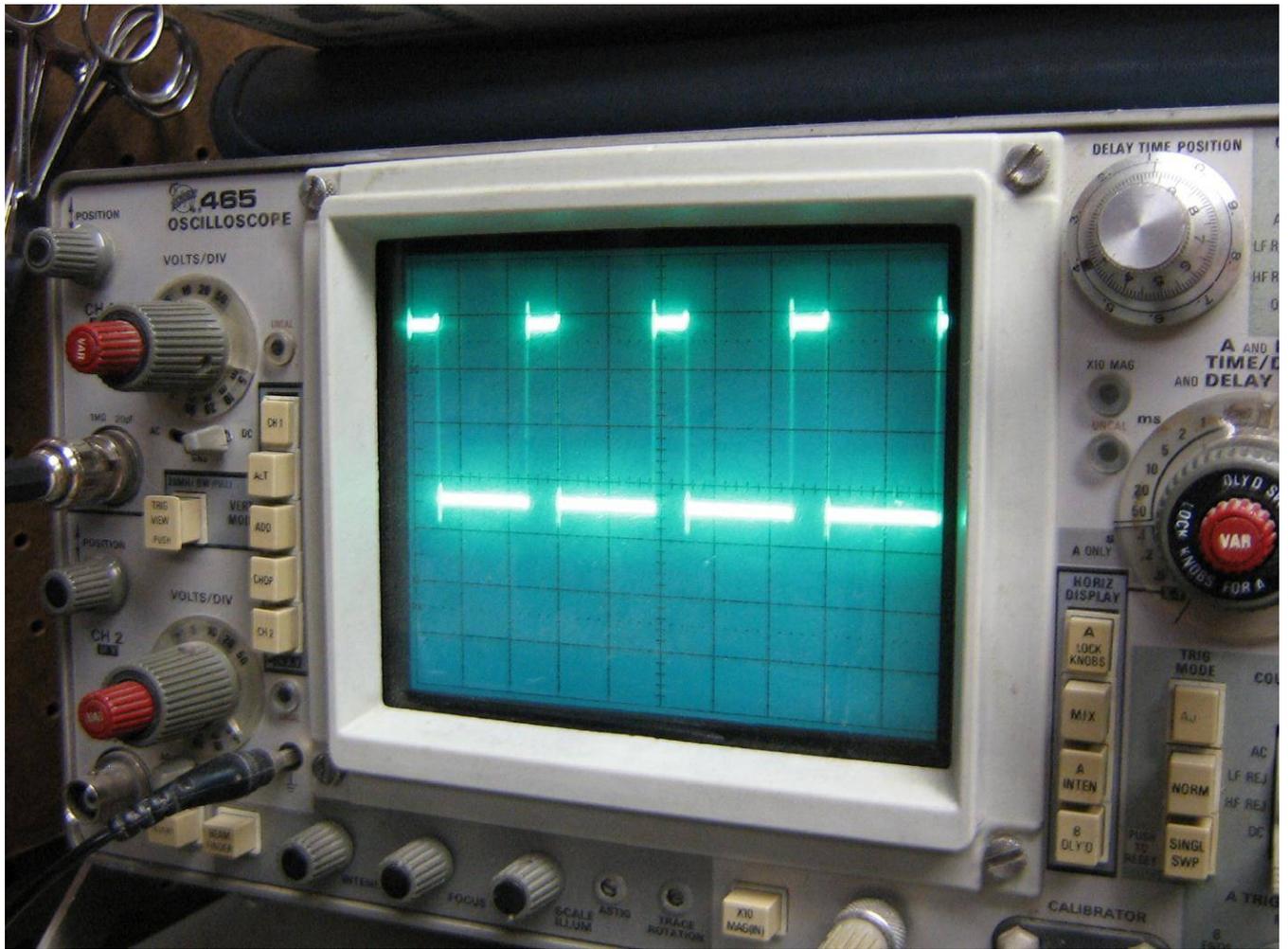


Setting the oscillation frequency of the TL494, 39.4 kHz in this case.

Measure the TL494's oscillator frequency at pin 9 to avoid loading the circuit down.

The transducers seem to work best when operated slightly above or below their resonant frequency of 40 kHz. This can be a problem when using lots of transducers in parallel as they all seem to resonant at a slightly different frequency, creating even more distortion.

You'll need to fine tune the TL494's pulse width and frequency potentiometers for minimal distortion on the final projected audio signal.



Overview of the unmodulated 40 kHz PWM output signal.

5 volts/division, 20 μ S/division.

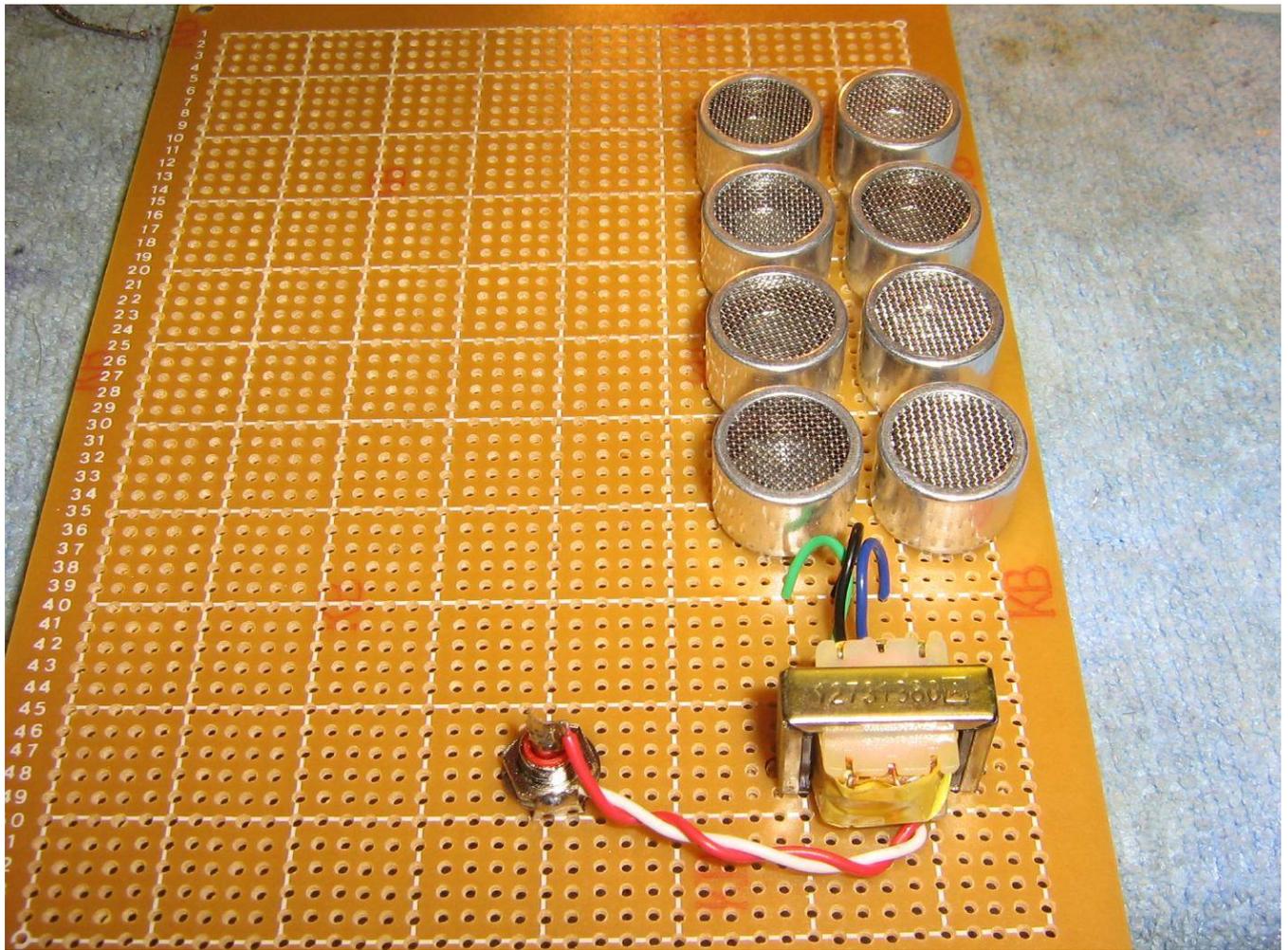


Overview of the Kobitone 400ST16-ROX 40 kHz ultrasonic transducers.

The plastic ring around one of the leads will be used for the "+".

The transducers are not technically polarized, but they'll all need to be in phase when fed in parallel.

Each transducer can handle around 60 volts peak-to-peak and has a capacitance of around 2400 pF.

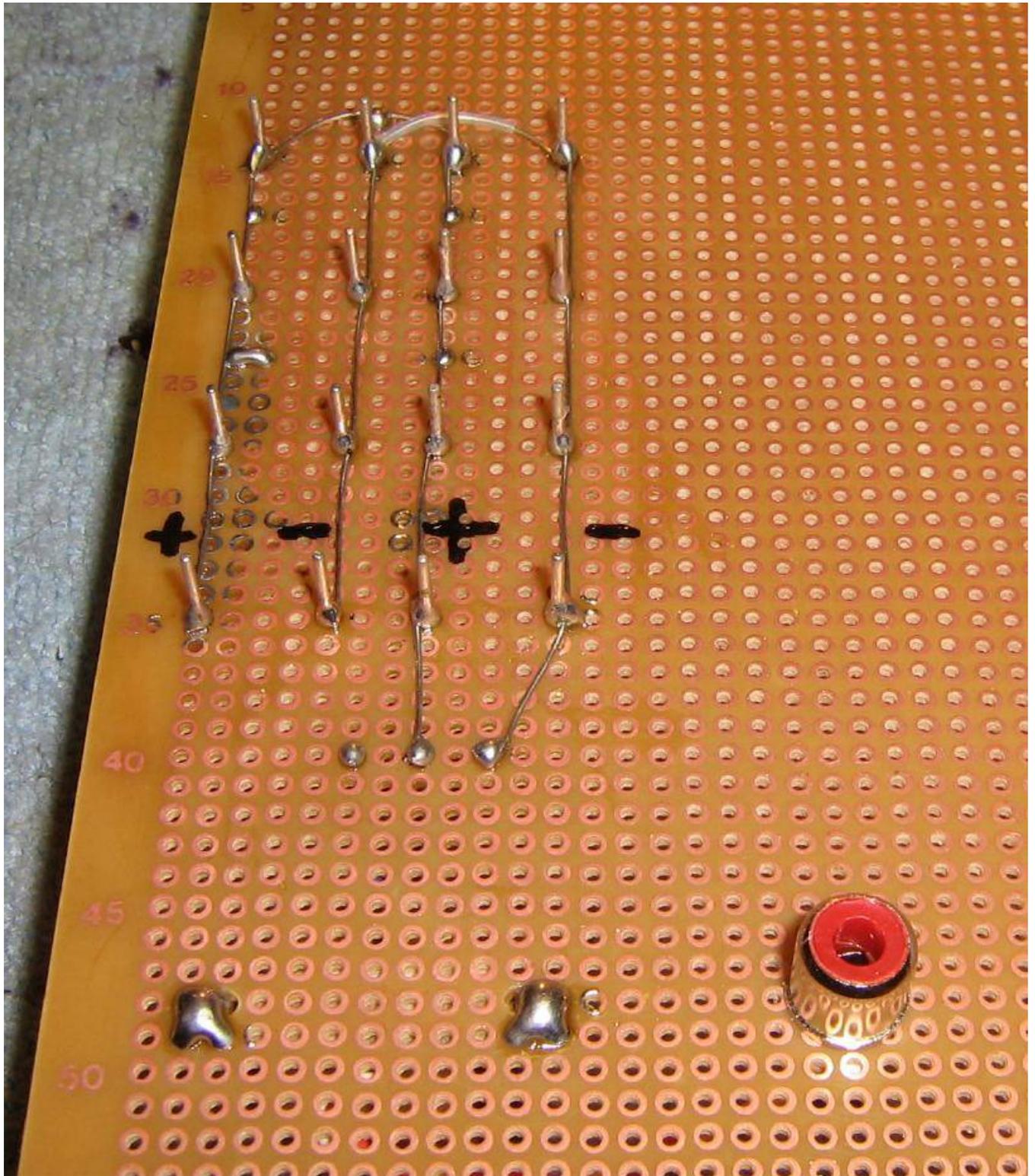


Experimental speaker array.

The Kobitone transducers (Mouser 255-400ST16-ROX) are fairly expensive, so I only have eight of them for now. This severely limits the range (and effectiveness) of the parametric speaker, but makes for a good starting point.

You'll need a minimum of around 50 transducers for an effective parametric speaker array, and at least 100 of them if you want any sort of range outdoors. 2,000 of them will reach out to a kilometer...

The transformer (Radio Shack 273-1380) is for an experimental method to step-up the voltage to the transducers. It increased the 15 volt PWM signal to nearly 40 volts peak-to-peak in this application. The transformer did get fairly warm after awhile, so you may want to look for a step-up transformer with a heavier gauge wire on its windings.



Transducer wiring overview.

They are all wired in parallel.

The eight transducers looks like a $0.02 \mu\text{F}$ capacitor. It's possible to add a series $800 \mu\text{H}$ inductor inline with the 40 kHz PWM signal to make a resonant circuit. This is a simple way to step-up the voltage to the transducers without the need for a transformer.

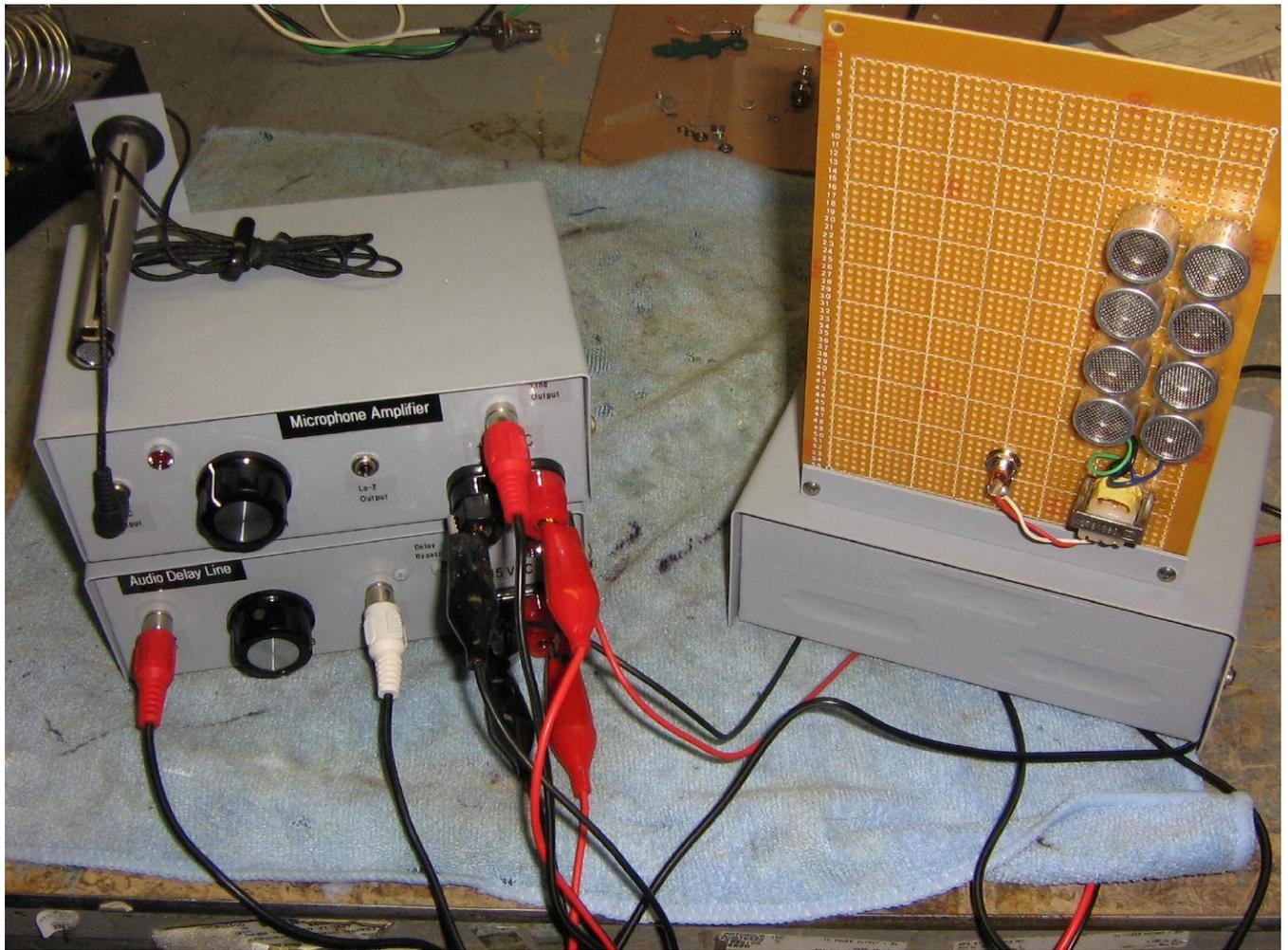


Overview of the experimental parametric directional speaker array.

A directional speaker isn't really needed for speech jamming rebroadcast application.

A standard audio amplifier and speaker, like the Sony SS-TS502 on the left, will work. Cheap piezo horn tweeters (center) will also work and most can be run at 20+ kHz, eliminating the need for expensive ultrasonic transducers.

When using conventional speakers, you may lose the directional nature of the rebroadcast audio which is required for maximum effectiveness when speech jamming.



Completed GBPPR Speech Jammer.

Using modules for each of the main sections allows you to experiment with other configurations.

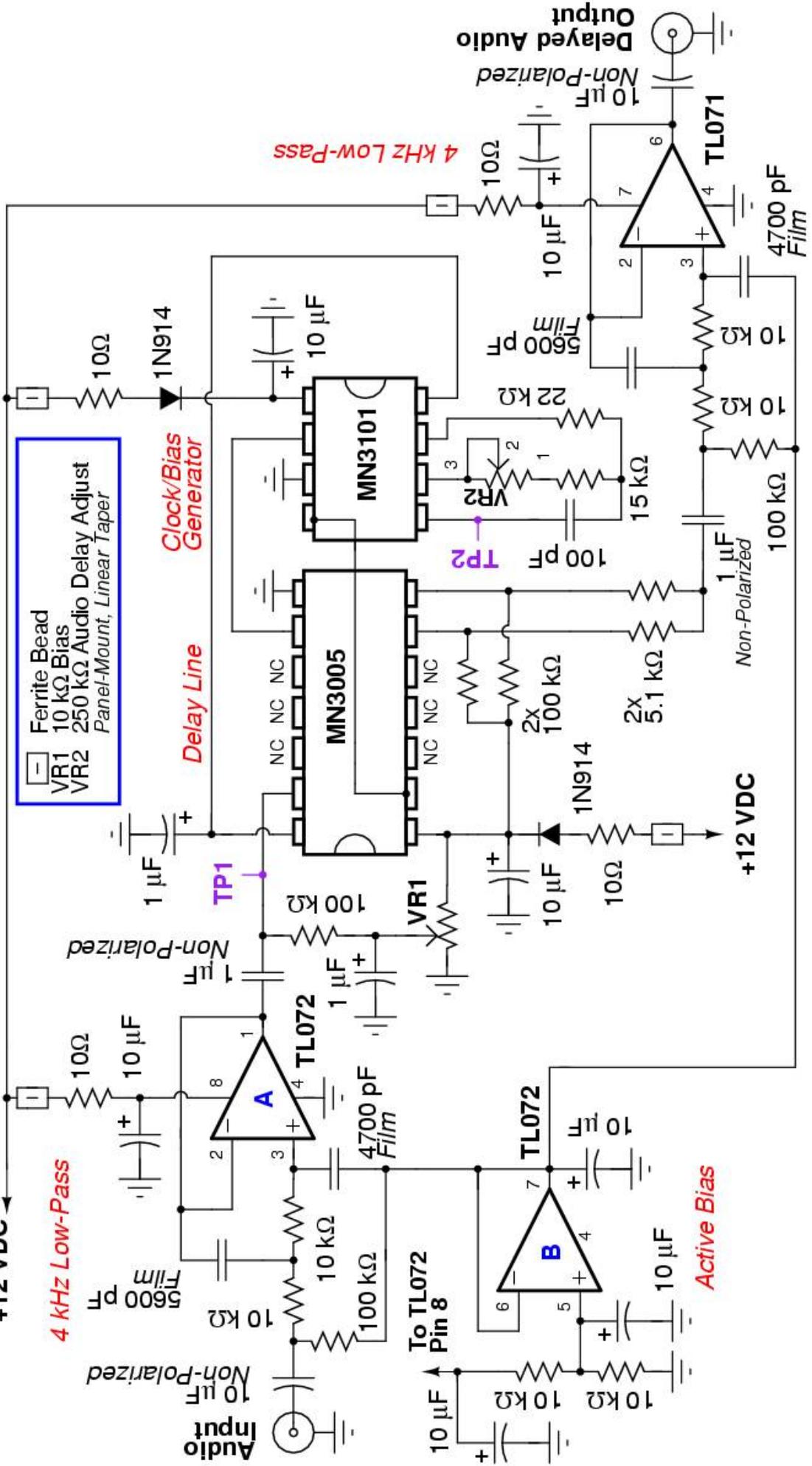
Some countermeasures to defeat this type of speech jamming include:

1. Speak softly. If the microphone can't pick up your audio, it won't work.
2. Concentrate on what you're saying. You can "talk through" the jammer if you know what you want to say ahead of time.
3. Pad out your speech with a series of timed pauses or "Ahhs..." to allow you to regain your composure.

Combine the speech jammer with the "GBPPR MIL-SPEC Laser Dazzler" project in *GBPPR 'Zine* Issue #89 for increased annoyance.

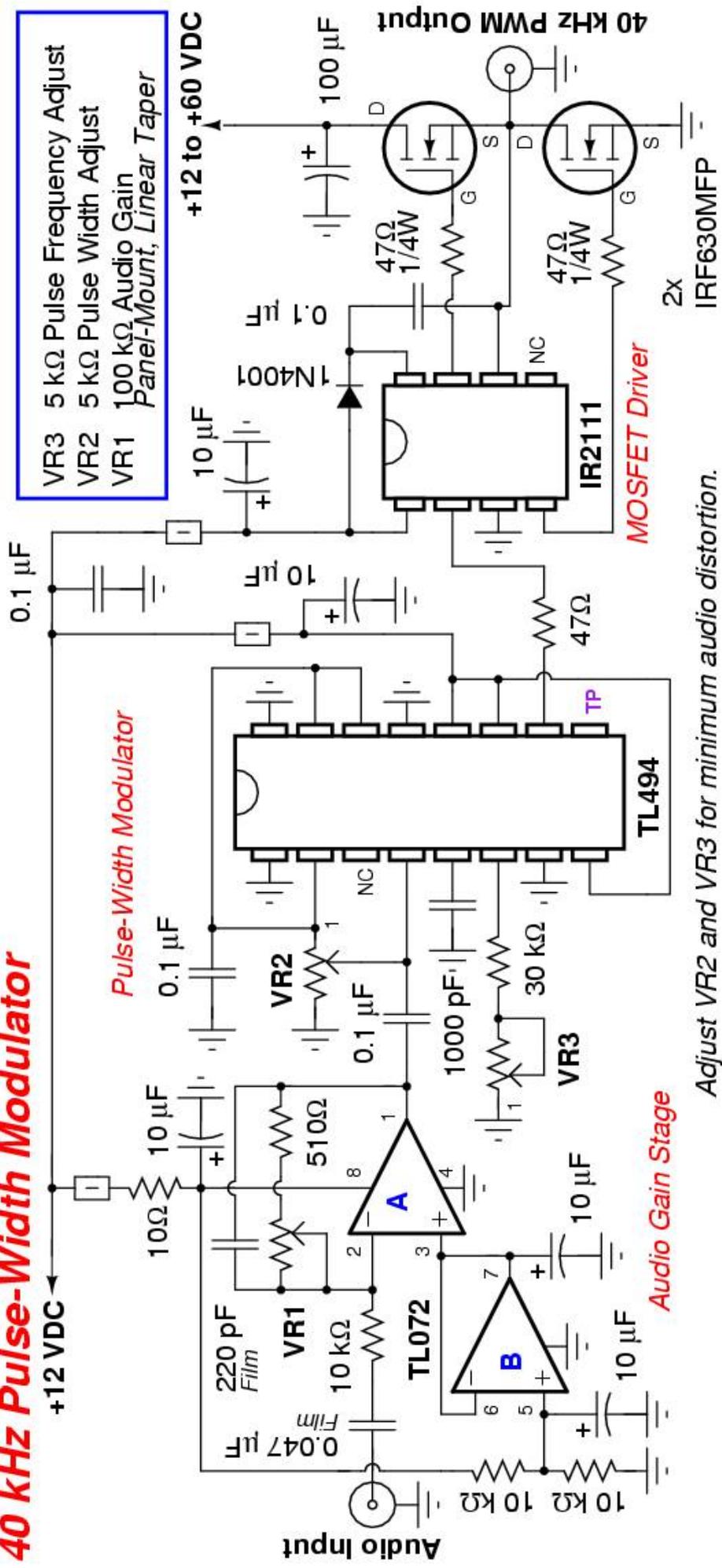
Audio Delay Line

TP1 Adjust for a 6 volt bias on the MN300x input.
 TP2 VR2 should tune from 20 to 200 kHz.

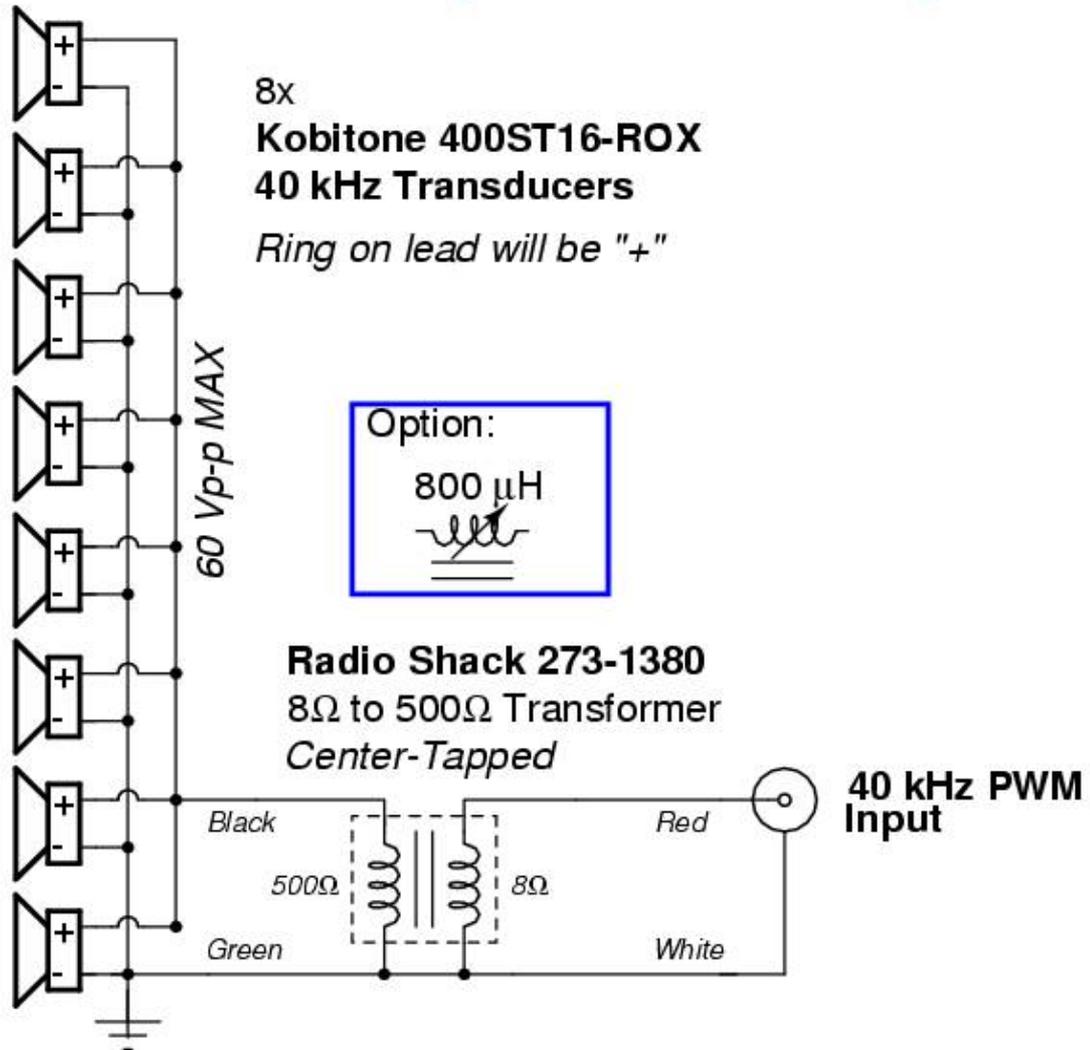


- Ferrite Bead
- VR1 10 kΩ Bias
- VR2 250 kΩ Audio Delay Adjust
- Panel-Mount, Linear Taper

40 kHz Pulse-Width Modulator



40 kHz Ultrasonic Speaker Assembly



Bonus



End of Issue #96



Any Questions?

Editorial and Rants

More "change" in Obama's Chicago! And they want to build a high-speed rail system between Chicago and Wisconsin. Umm... No!

Chicago Reaches 100 Homicides in 2012

March 21, 2012 – From: redeyechicago.com

by Tracy Swartz

Chicago on Wednesday reached 100 homicides for the year – the fastest the city has hit this mark in at least seven years, RedEye and police data show.

Seventeen homicides have been recorded in the last week – including four each on Saturday and Sunday, a RedEye analysis of preliminary police information found.

Chicago has not reached 100 homicides in March since 2004, when the city logged 106 homicides before April 1, according to police data. The city reached 100 homicides last year on April 26, RedEye data shows.

Citywide, gunshot homicides were recorded in the last week in Auburn Gresham, Chicago Lawn, Englewood, Gage Park, Greater Grand Crossing, Humboldt Park, the Near West Side, New City, Washington Heights, West Englewood, West Lawn, West Ridge and Woodlawn, data shows.

In the last week, South Lawndale recorded three homicides, police data shows.

On Wednesday, a man was fatally shot in the 2800 block of South Kildare Avenue, officials said. On Saturday, a 6-year-old girl was shot to death in the 3100 block of South Springfield Avenue, police said. Three days earlier, a 19-year-old man was fatally shot in the 3000 block of South St. Louis Avenue, officials said.

Meanwhile, a 58-year-old man was beaten to death Sunday in the 3500 block of West Lawrence Avenue in Albany Park, police said.

Thirty-three homicides have been recorded so far in March. Police logged 22 homicides in March last year, RedEye found.



Oh lordy... Now the liberal/Jew media is going ape-shit over the shooting of this useless nigger.

Let's take a closer look...

MIAMI (AP) – Wearing a hoodie. Listening to music and talking on his cellphone. Picking up Skittles for his soon-to-be stepbrother. Friends say that's how they would have imagined 17-year-old [Trayvon Martin](#) on a Sunday afternoon.

[More news, photos about Trayvon Martin](#)



Starting a fight? Possibly high on drugs and up to no good? No, friends say that description of Martin from the neighborhood crime-watch volunteer who shot and killed the unarmed black teenager doesn't match the young man they knew.

"There's no way I can believe that, because he's not a confrontational kid," said Jerome Horton, who was one of Martin's former football coaches and knew him since he was about 5. "It just wouldn't happen. That's just not that kid."

Trayvon Martin, 17, was slain in the town of Sanford, Fla., on Feb. 26 in a shooting that has set off a nationwide furor over race and justice.

AP

STORY: Thousands in Philly join

Storie
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"Raw
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(www.usatoday.com/news/nation/story/2012-03-24/trayvon-martin-friends/53744670/1)

That is a screen capture from an [USA Today](#) article on the Trayvon Martin shooting. Note that the picture's caption states that he is 17-years-old, but the picture is clearly of a younger Trayvon Martin. Mostly likely from when he was in middle school (11- to 13-years-old). Also in that article is this quote:

"... [Trayvon] Martin's parents kept a close eye on him, but they didn't have to be too strict, since he stayed out of trouble, [Fred] Collins said. However, he had recently been suspended from school for five days for tardiness, his English teacher, [Michelle Kypriss](#), told the Orlando Sentinel. School officials did not respond to a request for comment."

Hmm... Anyone with a brain (i.e. not a public school teacher or an AP reporter) knows that it's **ILLEGAL** for a public school to suspend a student for "tardiness."

In fact, [Florida Statute 1006.09](#) states that "No student shall be suspended [out-of-school] for unexcused tardies, lateness, absences, or truancy." In order for Trayvon Martin to get that long of a suspension, he would have to do something pretty serious (i.e., alcohol, drugs, violence, sexual assault, general TNB, etc.).

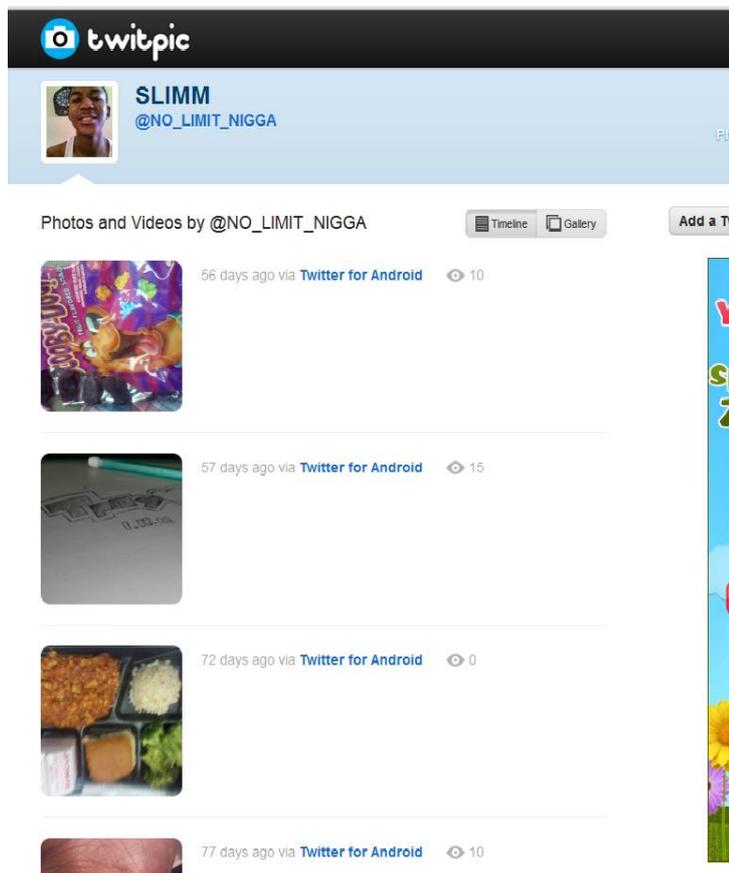
Someone is lying!

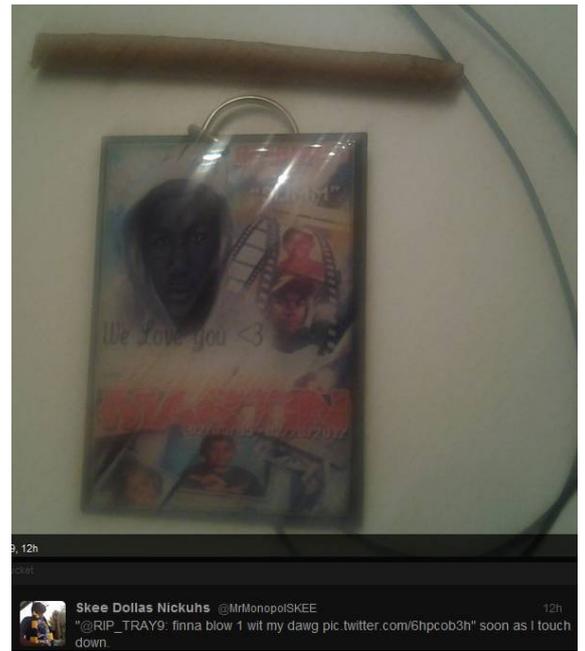
Remember, this fine bit of "investigative journalism" is from the same tribe who is currently pushing for World War 3 with Iran...



Above is a much more recent picture of Trayvon Martin from his Twitter account "NO_LIMIT_NIGGA." I guess his limit was really only 9 mm or so...

A quick pursual of Trayvon's Twitter postings showed no interesting schematics or Linux kernel patches. Note his gold "teefs" in the above picture. Niggers can't pay for their mortgages or Skittles, but seem to have plenty of money for pointless shit. I'll bet we lost a real academic powerhouse here...



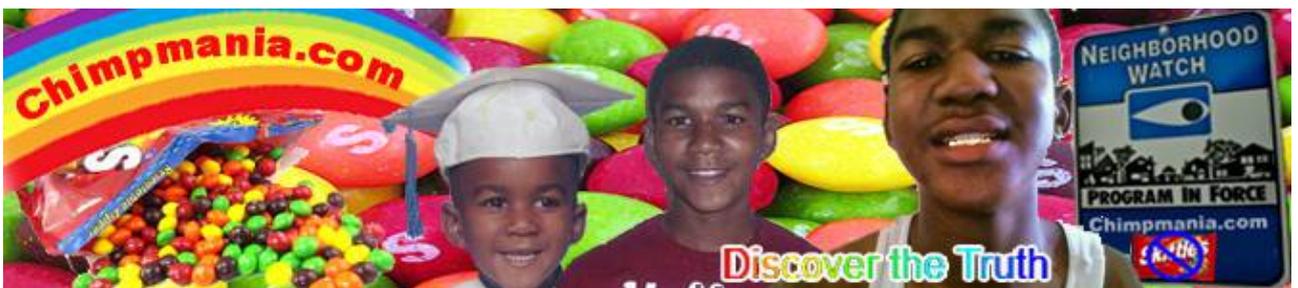


- 
TRAY'S BIG BRUH @RIP_TRAY9 21 Feb
 @NO_LIMIT_NIGGA <<<< yu ain't tell me yu swung on a bus driver
 In reply to SLIMM
- 
TRAY'S BIG BRUH @RIP_TRAY9 21 Feb
 @SelfMade_Kutta #team4dat
 In reply to K.Mitch
- 
TRAY'S BIG BRUH @RIP_TRAY9 21 Feb
 "@JetMan_Twin DON'T GET SLICK WIT ME U ONLY GON HURT YO SELF"real shit kause I ka always say sumn much worst
- 
TRAY'S BIG BRUH @RIP_TRAY9 21 Feb
 #team4dat #team4dat #team4dat #team4dat #team4dat #team4dat
 all da way 2 da grave

Trayvon Martin's brother has some *really* interesting Tweets and pictures linked from his Twitter account.

Gang signs (or is he having trouble with a calculus equation?), a marijuana joint, and a reference to "swinging on a bus driver."

Hmm... Don't hold your breath seeing these in the liberal/Jew media!





Notice how Zimmerman's mugshot appears to become lighter every day!

And then there was the shooter, George Zimmerman.

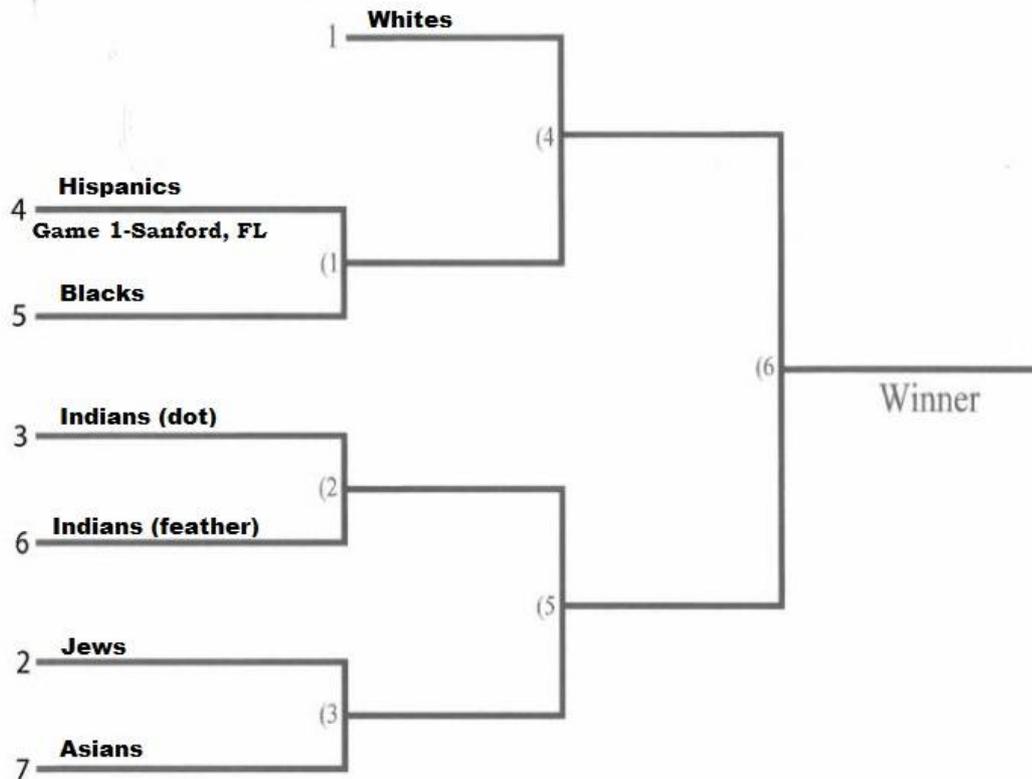
George Zimmerman has a Mestizo mother, but the media is going all out to label him "White."

Reuters even called him a "White Hispanic" (there is no such thing) in one of their articles and took to censoring any comments pointing this blatantly false quote out. It's funny, because if Hispanics are now Whites, then how can Sheriff Arpaio be charged for "racial profiling" Hispanics?

Derp... Change!

It's almost like the liberal/Jew media is pushing for another race war, just like in the 1960s with their staged "civil rights" movement...

2012 March Raceness Opening Round



While the Trayvon nonsense was going on, this happened. Let me guess... You didn't hear about this? Nancy Strait has since died and the nigger (Tyrone Woodfork) who attacked them was thankfully captured.

There will be no memorials or outrage from the mainstream media.

Elderly Couple Found Badly Beaten in North Tulsa Home

March 14, 2012 – From: newson6.com

Tulsa Police are looking for the attackers who broke into an elderly couple's home and beat both of them.

It happened in the 3300 block of East Virgin Street in Tulsa.

Family members say 90-year-old Bob Strait and his 85-year-old wife, Nancy, were fine Tuesday evening. A family member found the front door kicked in and the two victims badly beaten Wednesday afternoon.

EMSA says it took both victims to the hospital in serious condition.

Tulsa police spent the late afternoon Wednesday interviewing neighbors for clues, hoping for a lead to whomever would beat up an old couple and leave them for dead.

"It's terrible – there is no other word to describe it. We're doing everything that we can think of doing to try to catch the person that did this, first to find out who it was, and then try to get them in custody as soon as we can," said Tulsa Police Captain Dave Roberts.

The victims are well liked in the neighborhood where they lived for decades.

"They were very good people. You couldn't meet a nicer person. Have you heard that old statement: man amongst men? That's what he was," said neighbor Roosevelt Russell.

Police believe the attack was mid-day Wednesday, meaning the couple may have been laying there all afternoon before their family found them.

"I can't think of nicer people this could have happened to. They were just as nice as can be. You couldn't ask for nicer neighbors," said neighbor Ronald Hinnen.

Police say the victims were so badly hurt they were unable to speak to officers. At last check, the male victim was conscious. The current condition of the female victim is not known.

Tulsa Police are looking for two vehicles, one driven by the attackers and one belonging to the victims. Police say the attackers also stole items from inside the home.

The attackers' vehicle is a late-model maroon SUV, missing a hubcap on its right rear wheel. The victim's vehicle is a 2001 Dodge Neon, reddish orange in color and has an Oklahoma license tag of 402GCV.

Police have not released a description of the attackers yet.

If you have any information that would lead to arrest of the attackers, call 918-596-COPS.



Bob & Nancy Strait and Tyrone Woodfork

Oh yeah... This White college student, John Sanderson, was also killed at Mississippi State University by a group of three niggers. I'll bet you didn't hear about this one either?

Supporters carried bags of Skittles at the "Justice for Trayvon Martin" marches. We should all carry physics textbooks at the "Justice for John Sanderson" marches. Oh, that's right... There won't be any...



(www.reuters.com/article/2012/03/25/us-mississippi-shooting-idUSBRE82O03N20120325)



If they had fathers, he'd look like Obama!

(www.wyff4.com/news/30787874/detail.html)

And in Seneca, South Carolina, a group of six niggers attacked a single White man for no apparent reason at an Applebee's on March 17, 2012.

Hello? Jesse Jackson? Al Sharpton? ACLU? CNN? MSNBC? The Daily Show? See the Jew...

