Aphex Model 2020

FM Pro

Operating & Service Manual

Manufactured By
Aphex Systems Ltd.
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Sun Valley, California 91352
U. S. A.

The Model 2020 is protected under one or more of the following U. S. Patents:

5,930,374	5,612,612	5,424,488	5,115,471
5,898,395	5,485,077	5,422,602	4,939,471
5,896,458	5,483,600	5,359,665	4,843,626
5,848,167	5,463,695	5,334,947	4,633,501
5,737,432	5,450,034	5,155,769	4,578,648

Updates to this manual and other important information about the Model 2020 may be found at:
http://www.aphex.com

Notice

Following this guide will get you up and running but without much understanding of what you are doing. We highly recommend at least a minimal study of the manual. It is written for fast and easy reading.

Once the unit is installed, here's the procedure to get your FM Pro operating properly using a factory preset. You can later create your own presets if you wish.

Step 1, Understanding The Menus. The escape key backs you out of menus and commitments. It continuously takes you backwards towards the Main menu. The enter key selects menu items and makes commitments such as save and recall. The up and down cursers generally move the menu pointer while the left and right cursors generally flip through multiple menu pages. The Spin dial acts as an adjustment control to set menu values.

The Global Parameters

A. Processing Menu, Input/Output Menu pages 1,2,3
- Input reference level
- Peak output level
- Unit bypass on/off
- 20Hz High Pass Filter on/off
- 16.5kHz Low Pass Filter on/off
- Spectral Phase Refractor (SPR) on/off
- Stereo Insert on/off
- Input Source: A or D
- Output: A, D, or A&D

B. Processing Menu, Pre-emphasis Limiter Menu
- Pre-emphasis filter, 50 or 75 microsecond
- De-emphasis on/off

C. Processing Menu, Stereo Generator Menu pages 1,2
- Pilot Mix percentage
- Pilot on/off
- Input mode: Stereo, Mono (L, R or L+R)
- Mono Mode 90% or 100%

D. Processing Menu, AES Status
- Transmit rate: Slave, 32, 44.1, or 48kHz

All menus branch from the Main menu. Each menu may have one or more sub menus. The menu tree is so logical and easy to discover that within a few minutes you will be very comfortable with it.

Step, Setting Up Globals. Go to the processing menu, and, by navigating through the submenus, set up the "Global Parameters" to match your system's operating requirements. *The menus will state that a feature is unavailable if the associated option is not installed in the FM Pro.*

Important: Once the Globals are set, save to the Global preset, U01, by going to the "Save Preset" menu.

Step 3, Select A Preset. There are 8 factory presets designed for various programming formats. Chose one that approximately matches your station's format. If you are not sure, chose "Big Country" as a moderately aggressive starter. If you want to really risk it, chose "CHR" and get a very loud and intense air sound.

Step 4, Adjust The Output Level. While sending program audio to the FM Pro, adjust the analog line output (Input/Output menu) for the required peak output level. If you are using the PPDM multiplex option, adjust the multiplex output level (back panel trim) for 100% peak modulation of your transmitter.

That's It!

1.0 Table of Contents

2.0 Product Description

2.1 The FM Pro Story

Aphex entered the world of broadcast audio processing with the Type II Aural Exciter in 1981. Since then Aphex has continued developing leading broadcast audio products. The Compellor became and remains the world standard automatic level controller. The Dominator became the world standard multiband peak limiter. The Digicoder, the only patented stereo generator technology, took its place as the world's best stereo generator. Together the Compellor, Dominator, Aural Exciter, and Digicoder comprise the Aphex Audiophile Air Chain which has gained worldwide recognition as the premiere FM audio processing system for transparent, lifelike transmission of sound. Worldwide, many leading broadcast institutions have adopted these Aphex products as their technical standard and continue to employ them exclusively in high grade radio and television network systems.

As proud as we are of the Audiophile Air Chain and what it achieves in its sonic purity, we realize many broadcast applications demand something else. For example, a need exists to literally design the sound of a radio station, i.e., to generate a unique and competitive air sound intentionally modified in certain ways. Also, the processor may be required to operate directly within a digital audio path. Furthermore, there is a demand for programmability, remote control, and day-part automation. We developed the FM Pro to fulfill all these needs and more. Naturally, we borrowed upon many of the proven and exclusive Aphex patented circuits to achieve a technically excellent design, but while we were at it we invented at least six additional new and innovative audio processing concepts giving the FM Pro truly advanced capabilities and performance.

2.2 Description

The FM Pro is not just a boxed-up combination of prior Aphex products. It is a totally new and advanced audio processing system tailored especially to the demands of FM broadcasting. Competition ready, the FM Pro is completely adjustable from detailed and neutral to heavy and loud. Thanks to the numerous design innovations, even at extreme settings exceptional audio quality is maintained. Unlike the competing digital or analog audio processors which afford relatively little flexibility, the FM Pro is a virtually sound designer's palette allowing you to paint a truly unique and competitive on-air sound. Don't be misled, however, the FM Pro can just as well be set up to sound as neutral and transparent as desired.

Modular design makes the FM Pro space-saving and cost efficient compared to other processors. Plug-in options are easily field installed and can be shipped to specification. Sixteen recallable user presets are provided for manual selection or day-part automation. For convenience, eight factory presets covering a variety of formats are built-in to get you up and running quickly. Day-part automation is completely self-contained and driven by an accurate internal clock/calendar timer. Front panel operation is made easy and intuitive through the logical menu tree. An RS232 port permits remote control and monitoring of the FM Pro from almost any location through the use of a standard personal computer and a serial cable or modem. A simple but effective supervised password system can be engaged to maintain security over the FM Pro's operation. Fail-safe operation is assured by a power-off internal relay bypass of both the analog and digital audio I/O circuits. Power-up fast recovery gets the FM Pro back on line with all programmable parameters set correctly and operating in just a few tenths of a second. The front panel displays are more than token indicators. They indicate the actual operation of all major process functions in real time.

2.3 Applications

The flexibility of the Aphex FM Pro makes it suitable for nearly every FM broadcasting entity from government owned to purely commercial. Whether your audio system contains analog, digital or both formats, the FM Pro can be configured to fit in perfectly. Fully adjustable parameters allow you to optimize the FM Pro for any program format: classical music, talk, even heavy metal. Automated transformation of processing parameters by the day, hour, and minute allow radio stations with varying formats to tailor the FM Pro exactly as required for each day-part. Although the FM Pro is intended mainly for FM broadcasting, it will find many interesting applications in other fields where absolute audio control and flexible sound tailoring are desired. Digital and analog mastering, recording, satellite uplinks, and amplified sound are just a few examples.

2.4 FM Pro OPTIONS

	Analog Stereo Input	Analog Stereo Output	AES/EBU I/O	Pre-Emp Limiter	PPDM MPX Output
Basic	X	X			
Option 1	X	X	X		
Option 2	X	X		X	
Option 3	X	X		X	X

2.5 FUNCTIONS AND FEATURES

1. Analog and Digital Stereo Inputs
2. Input processing functions
   a. 20Hz Highpass Filter
   b. 16.5KHz lowpass filter
   c. SPR process
   d. Selectable external processing loop patch
3. **Frequency Discriminate Leveler**
   a. Improved parametric controls
   b. **New "sticky" leveling feature**
   c. **Selectable patented "DVG"**
   d. Selectable silence gate
   e. Adjustable silence gate threshold
   f. Adjustable AGC upper and lower control limits
4. Multiband compressor
   a. 4-bands
   b. adjustable crossover frequencies
   c. **Easyrider compression**
   d. **"Peak Accelerated Compression" (PAC) algorithm**
   e. Band-by-band stereo elastic coupling
   f. Band-to-band forward elastic coupling
   g. Adjustable compression drive
   h. Separate adjustable release time per band
   i. Selectable stereo hard coupling
   j. Compression drive control
   k. Output band mixing facility
   l. **New "post crossover" multiband technique**
   m. High or Low selectable ratio
5. Bass Processor
   a. **Distortion canceled bass clipper**
   b. "Warm bass" equalizer
   c. "Sub Bass" equalizer
   d. Total "Bass Mix" control
6. Peak Limiter
   a. **Bass interactive to reduce intermod distortion**
   b. Instant processing, i.e., no pumping
   c. Zero overshoot
   d. Master drive control configures loudness factors
7. Optional pre-emphasis processor
   a. **Special 50 or 75 microsecond pre-emphasis filter**
   b. Digicoder type pre-emphasis limiter
   c. Digicoder type non-overshoot final lowpass filters
   d. Output ready for any stereo generator
8. Optional digital I/O module
   a. AES/EBU format up to 20 bits
   b. Selectable output sample rate: 32K, 44.1K, 48K
   c. Auto "lock on" for input rates of 32K, 44.1K or 48K
   d. Input and output sample rates separately selectable
9. Optional Digicoder stereo generator module
   a. **Digicoder type PPDM stereo generator**
   b. Analog multiplex output
   c. stereo/mono mode switching
   d. Pilot on/off
   e. trimmable multiplex output level
10. Digital remote control
   a. RS232 digital interface
   b. Windows 3.1 or 95 virtual control panel software
   c. Complete operating capabilities
   d. Complete visual real time meter displays
   e. Password security options
11. Front panel user interface
   a. LCD graphics panel display
   b. Password security features
   c. Rotary encoder knob
   d. Up/Down, L/R cursor buttons
   e. Menu selection system
   f. Real time LED bargraph meters for:
      1. Leveling
      2. 4-band compression
      3. Limiting
      4. Stereo input VU
   g. Real time LED indicators for:
      1. 16.5KHz Lowpass
      2. 20Hz Highpass
      3. 50/75uS Pre-emphasis
      4. SPR
      5. Analog In selected
      6. Digital in selected
      7. Digital data presence
      8. DVG
      9. Sticky on/off
      10. Silence gate
      11. L><R hard couple
      12. L><R elastic couple
      13. 1><2 couple
      14. 2><3 couple
      15. 3><4 couple
      16. Analog output presence
      17. Digital output presence
      18. Bypass on/off
   h. Headphone monitor selector
      1. Monitor input signal & volume
      2. Monitor output signal & volume

3.0 The User Interface

Figure 3-1 illustrates the front panel features of the FM Pro. You may refer to that diagram for a quick summary of all available controls and indicators. The following description, taken in conjunction with the diagram, will give you a very good understanding of all the features and controls, their purpose and use.

3.1 Metering and Status

While other processors provide only drab, jittery, and inaccurate LCD meter displays and tell the status of structural functions only on buried menu pages, the FM Pro provides the user with colorful front panel metering that is true and easy to read. The major dynamic processes are metered by real-time LED bar graph displays while the on/off status of static functions are indicated by single LED's. The state of all major functions of the FM Pro can therefore be observed at any time with one one glance at the front panel.

Following is a more detailed description of all the panel indicators.

3.1.1 Input Meter

Two yellow 10-segment bargraph meters are provided to indicate the stereo audio input level relative to the current input reference setting. The scale indicates from -18dB to +9dB in 3dB steps.

Note: The input reference level is adjusted in the Processing I/O menu. Left and Right inputs are adjusted in tandem with one control in software. For analog inputs, set the input reference level to cause a program reference tone to read zero VU. The digital audio input automatically sets the 0VU reference to 10dB below digital maximum peak reference.

3.1.2 Leveling Meter

This red dot-mode 10-segment meter indicates the amount of automatic gain correction that is created by the leveler. The leveler gain control is applied equally to both audio channels. The scale indicates up to 15 dB of gain addition, and 7.5 dB of gain reduction. The amount of activity that is seen in this meter will vary widely depending on program material and the settings of the leveler/gate parameters.

Note: Control for the Leveling function is located in the Processing, Leveler/Gate section of the control menu.

3.1.3 Multiband Compressor

Four green 10-segment bargraph meters labeled “Low”, ”Mid 1”, “Mid 2”, and “High” indicate the amount of gain reduction taking place in each of the four compression frequency bands up to a maximum of 15dB. Since the multiband compressor is a two-channel process, 8 meters would ordinarily be required to display all bands. For convenience, however, the front panel meters merge the dual channel indications into one set of four displays. Each band meter displays the greater amount of gain reduction of the left and right channel at all times.

Note: All multiband compression parameters are controlled in the control menu under Processing, Multiband.

3.1.4 Limiter

The Limiting meter is a red dot-mode 10-segment display that indicates the amount of peak limiting occurring in the limiter/bass processor circuit after the multiband compression. This is separate from and <u>does not</u> display any pre-emphasis limiting that takes place when the pre-emphasis limiter option is used. The scale is from 1 dB of limiting to 10 dB of limiting in 1 dB steps.

Note: Limiter parameters are controlled in the users menu under Processing, Limiter/bass.

3.1.5 Status Indicators

Individual LED's indicate the status of principal processing functions other than functions indicated by the 8 bar graph meters. A more detailed description of these functions follows.

3.1.5.1 16.5 KHz Lowpass Filter

This is a member of the input pre-processing group of filters. You can invoke this filter to clean up unwanted high frequency noise which may have resulted from interference in the STL path, or other source. It is not related to the 15KHz lowpass filters associated with the stereo generator. This filter works on both the analog and digital audio inputs.

Note: Control of the 16.5KHz Lowpass Filter is located in Processing, Input/Output menu.

3.1.5.2 20Hz Highpass Filter

This is another member of the input pre-processing group of filters. You can invoke this filter to clean up unwanted rumble or subsonic noise which may be encountered. It is not related to the 15KHz lowpass filters associated with the stereo generator. This filter works on both the analog and digital audio inputs.

Note: Control of the 20Hz Highpass Filter is located in the Processing, Input/output menu.

3.1.5.3 75 uSec Indicator

This light tells if the pre-emphasis option is set to 75 microseconds. It remains dark if the pre-emphasis is set to 50 microseconds or if the pre-emphasis limiter option is not installed.

Note: This light responds to the pre-emphasis selection chosen in the Processing, Pre-emp Limiter menu.

3.1.5.4 SPR

Spectral Phase Refractor(SPR) is another member of the input pre-processing group of filters. The SPR filter contains a flat frequency response but introduces a frequency dependent phase rotation into the audio path for the purpose of increasing the subjective clarity of the audio. A further effect of SPR is to improve voice waveform symmetry as an aid to the limiter. This filter works on both the analog and digital audio inputs.

Note: Control of the SPR filter is located in the Processing, Input/Output menu.

3.1.5.5- HF Limiter

This is another name for the **Pre-emphasis Limiter**, which is an optional processing module. The light is continuously dark if the option is not installed. If the option is installed, then the LED indicates whether the Pre-emphasis Limiter is switched on or off.

Note: Control of the HF Limiter is located in the Processing, Pre-emp Limiter menu.

3.1.5.6 Analog In

This LED lights if the audio input is set to analog (versus digital).

Note: Selection of analog or digital input is located in the Processing, Input/Output menu.

3.1.5 Digital In (Optional)

Indicates if the audio input is set to digital (versus analog). Digital input is part of the digital audio option and is available only when the option is installed. If the digital audio option is not installed, this light stays dark and the Analog In light remains lit.

Note: Selection of the analog or digital input is located in the Processing, Input/Output menu.

Part of the digital audio option is the “Auto Source” feature that will automatically switch from the digital input to the analog input if excessive errors are detected at the digital audio input. In such a case, the Analog In and Digital In lights will indicate which input has been automatically selected.

Note: Enable/disable of the Auto Source feature is located in the Processing, AES Status menu.

3.1.5.8 Data Present

When the digital audio option is installed, this LED shows if there is AES/EBU data arriving at the digital audio input to the FM Pro. If there are errors present in the AES/EBU datastream, the Data Present light will flash, and as stated above, when the Auto Source feature is selected, the unit will automatically switch to the analog audio inputs. As the datastream error condition rectifies itself, the FM Pro will automatically switch back to the Digital input.

3.1.5.9 DVG

If the Dynamic Verification Gate (DVG) is turned on, this LED flashes in response to the DVG action. If the DVG is switched off, the LED remains dark. Dynamic Verification Gate is an Aphex patented invention that allows the leveler’s gain correction to move only during the brief periods when present peaks are above the historical average peak level. Based on the pattern of the audio waveform, the DVG will freeze the movement of gain during intervals when the ear is most cognizant of the gain movement, thus making the leveler more transparent sounding. The DVG also serves to freeze the leveler gain during brief program pauses of about 1.5 seconds or less duration, preventing immediate noise swelling. After about 1.5 seconds, the DVG releases control. The DVG works only with the leveler function of the FM Pro, and does not affect the compression system in any way.

Note: Enable/disable of the DVG feature is located in the Processing, Leveler menu.

3.1.5.10 Sil. Gate (Silence Gate)

The Silence Gate LED lights up whenever the Silence Gate activates. The Silence Gate is a delayed level detector which acts to freeze the leveler gain during extended pauses in program. Since the DVG serves this purpose for brief program pauses, the Silence Gate is delayed about 1 second to take over from the DVG for more lengthy periods. The delayed response prevents the Silence Gate from constantly interfering with the action of the Leveler, deferring gain gate control to the DVG between periods of silence.

Note: Enable/disable and Threshold of the Silence Gate feature is located in the Processing, Leveler menu.

3.1.5.11 Lvl. Stic. (Sticky Leveler)

The Lvl. Stic. LED lights up when the user selectable Sticky Leveler is turned on. The Sticky leveller is a new Aphex invention that keeps the leveler's gain frozen until the input signal amplitude changes by more than a certain amount. By holding constant gain until the audio level exceeds a user selectable window, the Sticky Leveler further improves the transparency of the leveler's action, especially at the faster rate settings.

Note: Controls for the Sticky leveler are located in the processing Leveler/Gate menu.

3.1.5.12 Insert

The Insert LED lights up when the processing "insert loop" is turned on. This control allows the user to insert any type of stereo audio processor between the Leveler and the Multiband Compressor. The insert inputs and outputs are supplied by rear panel unbalanced (pin 2 hot, 1&3 grounded) XLR connectors.

Note: On/Off control for the Insert Loop is located in the Processing, Input/Output menu.

3.1.6 Multiband Coupling

Five LED's are provided to show the current status of the Multiband Compressor coupling features. The L><R coupling features refer to the stereo coupling of the Multiband Compressor bands. The band><band features refer to coupling between adjacent bands of the same channel. When turned on, the band><band coupling is enacted in both channels at once. The band><band coupling does not produce any stereo coupling effect. Both band><band and L><R coupling may be enacted simultaneously.

Note: Multiband coupling features are controlled in the Processing, Multiband menu.

3.1.6.1 L><R Hard

Couples the compression gain of left and right compressors directly, i.e., whichever channel produces greater gain reduction controls the gain of both channels equally and at the same attack/release rates.

3.1.6.2 L><R Elast. (L><R Elastic)

Couples the compression gain of left and right compressors elastically, i.e., whichever channel produces greater gain reduction influences the the gain of the opposite channel through a slow averaging effect.

3.1.6.3 1><2

Couples the gain control of band 1 to the gain control of band 2 in such a way that whichever band produces the greater amount of gain reduction influences the the gain of the other band through a slow averaging effect.

3.1.6.4 2><3

Couples the gain control of band 2 to the gain control of band 3 in such a way that whichever band produces the greater amount of gain reduction influences the the gain of the other band through a slow averaging effect.

3.1.6.5 3><4

Couples the gain control of band 3 to the gain control of band 4 in such a way that whichever band produces the greater amount of gain reduction influences the the gain of the other band through a slow averaging effect.

3.1.7 Output Status LEDs

Three LED's are provided to show the processor output status, analog, digital, or bypass.

Note: The output controls are located in the Processing, Input/Output menu.

3.1.7.1 Analog

Indicates if the analog output is turned on. The analog output is permanently selected unless the digital audio option is installed in which case the user can select analog out, digital out, or both at once.

3.1.7.2 Digital

When the digital audio option is installed, this light indicates if the digital output is turned on.

3.1.7.3 Bypass

In bypass mode, the analog input jacks are relay connected directly to the analog output jacks, and the AES/EBU input jacks are relay connected directly to the AES/EBU output jacks. The bypass mode can be user selected and automatically occurs when the power is off.

3.1.8 LCD Display

All operating menus and messages will appear on this LCD display. Refer to section 4, “Navigating The Menus” for detailed information on using the menu system.

3.2 Front Panel Controls

The Fm Pro utilizes a sophisticated yet intuitive method of user interface. The right side of the front panel comprises the user controls. By means of LCD menus, the navigation/control keys, and the “Spin” dial, all processor controls are easily accessed. The right half of the panel also contains a high quality headphone monitor that provides a means for listening to the raw input signal in comparison to the fully processed output signal.

3.2.1 Spin Dial

The Spin dial is used to adjust the variable menu parameters which have been chosen by use of the navigation keys. Depending upon the situation, the Spin dial will act like a potentiometer or selector switch.

3.2.2 Navigation (Cursor) Buttons

The four lighted red buttons that are labeled with white arrows pointing up, down, right, and left are used to navigate the various pages and menus seen on the LCD display. Generally, the up and down cursors move the pointer up and down through selections on a menu page, while the left and right cursors switch between multiple pages of a multipage menu.

3.2.3 Esc (No), Enter (Yes)

These two lighted red buttons allow the user to commit to or escape from menu choices. Repeated escapes will back out of menus until the Main menu is reached. Enter (yes) is used to select a menu choice or verify a menu commitment.

3.2.4 Input, Output Monitor

The lighted red buttons above the headphone jack select their respective audio signals for monitoring. The selected signal appears at the stereo headphone jack only. The proper de-emphasis (if appropriate) is automatically inserted in the output monitor. The input monitor taps the unaltered input signal.

Note: The volume of the input and output signals can be set independently in the Processing, input/Output menu.

end

4.0 Navigating The Menus

The flow chart of Figure 4-1 illustrates the entire menu tree of the FM Pro. You may find referring to the chart helpful while reading this section of the manual.

4.1 Introduction To The Menu Display

The Liquid Crystal Display (LCD) menu system of the FM Pro is designed to be logical and easy to understand. Without any instruction at all you would probably be able to find your way around and control the unit through your own exploration.

Note: The various menus all branch from the Main Menu. You gain access to the Main Menu by first passing through the password security check, if it is activated. Refer to section 8, "Password Security System" for information about enabling and disabling the password requirement.

The LCD display has a number of features that give the user an indication of the present location within the menu, the time of day, the current on-air preset, and any parameter being adjusted. Following is a detailed description of the various menu features.

4.2 Menu Display Features

4.2.1 Header

The top line left side of the display shows the current menu name while the current page is indicated on the right. For example, the top line may read: [MAIN MENU P:1.2]. This indicates you are in the Main Menu, page 1 of 2 pages. Once past the Main Menu, the menu name always refers to the functional area you are controlling. For example, the Leveler menus are named, simply, "LEVELER".

4.2.2 Menu Selections

Displayed below the header are the messages or selection options as provided by various menus. If you are in a functional control menu such as the "PROCESSING" menu, then you will see up to four function options labeled F1, F2, F3, F4 per page. Each page of a multiple page menu will continue from the previous page such as F5, F6, F7, F8 etc.

If you are in other menus such as the PRESETS, and DAY PARTING menus, the logic of selection labeling is similar, except the selections may be labeled as U01, U02..., or S1, S2..., etc. The selections flow from page to page as necessary to accommodate all available selections in a given menu.

4.2.3 Footer

The bottom line reads out a short phrase indicating what is expected as input. It may be "Enter key", "Enter value", "Enter function", etc. The bottom right corner contains an alternating display that toggles between the time of day and the memory preset number (U01, P01, etc.) that is currently on-air.

4.3 Display Sleeping Mode

The menu system sleeps when not in use by a logged in user. During this time, the "Welcome to Aphex" logo screen is displayed. This screen displays the current time and software version. Pressing any key wakes up the menu system and clears the logo screen, bringing up the Password Security page.

How To Make Selections

1. To MOVE BETWEEN PAGES of multiple page menus, use the left and right arrow keys.
2. To MOVE THE POINTER, use the up and down arrow keys.
3. To SELECT THE FUNCTION, press the Enter (YES) button.
4. To ESCAPE FROM A SELECTION, press the Esc. (NO) key.

4.4 Password Security Page

This page is the pathway to the Main Menu if a password is required. Pressing any key when the logo screen is displayed brings you to this page. If no password is currently active in the security system, this page will be omitted and the screen will go to the LOG IN VERIFY page, telling you a password is not required. You can then press the enter key and pass directly to the Main Menu.

Note: For more information on the password and security system see section 8 of this manual.

4.5 Main Menu Page 1 of 2

The Main Menu consists of two pages containing 8 selections, F1 through F8, that lead you to all other menus. Selecting F1 through F8 opens the associated menu by pressing enter. The first page contains these selections which are described below:

[MAIN MENU P:1.2]
F1- Password
F2- Presets
F3- Processing
F4- Remote Link

4.5.1 Password Menu

This section contains all password security elements including logging in and out as a user, as well as password maintenance. The four menu selections available are as follows.

[PASSWORD MENU]
F1 - Edit password
F2 - View password
F3 - Log In
F4 - Log Out

4.5.1.1 Edit Password

This page allows you to overwrite the password with a new one or to delete the password all together. Do so by moving the cursor and selecting a character from the character table. Use the spin dial to select characters from the table, and press Enter to place the character at the cursor position. The cursor will automatically advance one more space. Repeat the process until your new password is all entered. You can use the space in the character table to put a blank space in the password. For example, you could have a password like "kitty cat". To delete the password, simply enter all blank spaces over the old password. When the password is ready to save, use the cursor key to highlight the word "Save" and press Enter.

Note: Do not use the Enter key to advance to the word "Save". You will inadvertently press Enter over "Clr" and erase your new password. Use only the left and right arrow keys to advance the cursor to "Save".

When you press Enter at the "Save" position, you will be passed to the View Password page and asked to verify that you want this new password to be entered into memory. Press Enter to take the new password or Esc. to revert back to the Edit Password page. Pressing Esc. again will revert you back to the Password Menu, and pressing Esc. once again will revert you all the way back to the Main Menu.

4.5.1.2 View Password

To observe (without the option of editing) the current password, select F2 "View Password" from the Password Menu. Press Esc. to return to the Password Menu.

4.5.1.3 Log in

In the software version provided at the time of this writing, this is a rather unnecessary menu option since you can't get to this option unless you are already logged in. It is being reserved for a future software version which may include and expanded password security system.

Selecting this option brings you to a page similar to the Edit Password page. In this case, you enter the required password and upon selecting "Save" you get a validation screen which tells you if your password matches or is in error.

4.5.1.4 Log out

From the Password Menu, select F4-Log Out and press Enter. The "Welcome to Aphex" logo screen will pop up. If a password is in effect, the FM pro is now in a secure mode safe from tampering by unauthorized personnel.

4.5.2 Presets

From the Main Menu, select F2-Presets and press Enter to go to the Presets Menu. From here you can recall processing presets, save current processing parameters to user presets, view the list of presets, or manage day parting functions. The selections available are:

[PRESETS MENU]
F1 - Recall Preset
F2 - Save Preset
F3 - View list
F4 - Day Parting

4.5.2.1 Recall Preset

Six pages of presets will be displayed. From here you can instantly put any available preset on the air. To do so, move to the page containing the desired preset and move the pointer to the preset you want. Press Enter and the preset will be transferred to the FM Pro's processing parameters. The transfer occurs softly so you won't hear any "pops" as the parameters change. There are two pages of factory presets labeled P01 through P08, and four pages of user presets labeled U01 through U16. Refer to section 7 of this manual for information about building and using presets.

4.5.2.2 Save Preset

From this menu you can save the processor settings currently running in the FM Pro to any user preset memory. All six pages of presets are made available just as in the Recall Preset menu. You will be informed that you can't save to a factory preset location if you attempt to do so. To save to a user preset, move to the page containing the preset memory you want, and move the pointer to the correct location on the page. Press Enter to send the current processing parameters to that preset. You will be asked to overwrite the old preset, even if it was not yet used. You press (YES) and the ENTER NAME page will appear. This page operates exactly as the Edit Password page. Select a character from the character table using the spin dial and press Enter to place the character at the cursor position. The cursor will automatically advance one more space. Repeat the process until your complete preset name is entered. You can use the space in the character table to put a blank space in the name. For example, you could have a preset name like "hot beat", or "test 1". When the preset name is ready to save, use the cursor key to highlight the word "Save" and press Enter.

Note: Do not use the Enter key to advance to the word "Save". You will inadvertently press Enter over "Clr" and erase your new preset name. Use only the left and right arrow keys to advance the cursor to "Save".

You will be shown the NEW PRESET page and asked to verify the new name. Press (YES) and the new preset is stored. Press (NO) and you will be jetted back to the SAVE PRESET screen from which you came.

4.5.2.3 View list

The View list function is provided to allow the user to see the name and date of origin of any of the presets in the Fm Pro. All six pages of presets are available to view, and can be accessed by pressing the left and right arrow buttons. Once a preset is selected, the VIEW PRESET DETAIL page will appear showing the preset name and date saved.

4.5.2.4 Day Parting

In the Parting Menu you can view and edit day parting schedules, and you can turn parting on or off. To turn day parting on or off, position the pointer to the "Parting status" line and turn the Spin dial to select on and off. Viewing and editing functions are performed as follows.

4.5.2.4.1 Edit Day Parting

The Edit Parting menu allows you to select "Daily Edit" or "Weekly Edit". If you chose "Daily Edit" you will be taken to the "Parting Sets" menu where you will be able to chose among 8 day-part sets to edit. If you chose "Weekly Edit", you will be taken to the weekday scheduler page. Please refer to section 7 of this manual for detailed information about setting up a day parting schedule.

4.5.2.4.2 View parting

The View Parting menu allows you to select "Daily Events" or "Weekly Events". If you chose "Daily Events" you will be taken to the "Parting Sets" menu from which you may select among 8 day-part sets to view. If you select "Weekly Events" you will be taken to the "View Weekly Set" menu to view the weekday schedule.

4.5.3 Processing

The "Processing" selection on page 1 of the Main Menu launches you to the two-page "Processing Menu" from which you can access all the audio processing controls. In all, submenus for 8 major processing functions, F1 through F8, are available from this menu. These submenus cover the entire processing structure of the FM Pro.

4.5.3.1 Processing Menu Page One

The first 4 of the 8 function submenus are accessed from page one. They are:

[PROCESSING P:1.2]
F1 - Input/Output
F2 - Leveler/Gate
F3 - Multiband Compressor
F4 - Limiter/Bass

Note: Refer to the appropriate sections of this manual for a detailed description of the processing parameters and how to set them up.

4.5.3.1.1 Input/Output Menu

This menu contains 3 pages. Page one gives control over the input and output levels, and the hardwire I/O bypass. Page two gives control over the three input processing filters and the stereo insert loop. Page three gives control over the input and output source selection and the headphone monitor volume. Use the up and down arrow keys to select a function and use the Spin dial to set the value.

4.5.3.1.2 Leveler/Gate Menu

This menu has three pages. Page one gives control over the leveling rate, the maximum gain and maximum attenuation settings. Page two gives control over the DVG and the Sticky Leveler function. Page three gives control over the Silence Gate function. Use the up and down arrow keys to select a function and use the Spin dial to set the value.

4.5.3.1.3 Multiband Compressor Menu

The Multiband Compressor menu has a total of five pages; the first of which allows the user to set the crossover frequencies for the 4 band compressor. To adjust crossovers, set the pointer to the desired crossover and use the Spin dial to set the frequency.

Page two lets you set the multiband compression drive. Simply use the Spin dial to set the value.

Page three gives control over the release times of each of the four bands of the compressor. To adjust release times, place the pointer adjacent to the band you want to adjust, then use the Spin dial to set the value.

Page four allows you to adjust the output mix of the multiband compressor. At this point you can make equalization adjustments to the audio based upon the band crossover frequencies. To make an adjustment, place the pointer next to the band you want to adjust, then use the Spin dial to set the value.

The final page gives control over stereo coupling and band-to-band linking. As with the other pages, place the pointer next to the selection and use the Spin dial to set the value.

4.5.3.1.4 Limiter/Bass Menu

This menu has two pages. Page one gives control over the master limiter drive, the bass drive, and the brightness enhancer. Page two gives control over the warm bass and sub bass equalizers. To adjust drive or equalization, set the pointer to the desired function and use the Spin dial to set the value.

4.5.3.2 Processing Menu Page Two

The second 4 of the 8 function submenus are accessed from page two. They are:

[PROCESSING P:2.2]
F5 - Preemp.-Limiter
F6 - Stereo Gen.
F7 - AES status
F8 - More or Less!

Note: Refer to the appropriate sections of this manual for a detailed description of the processing parameters and how to set them up.

4.5.3.2.1 Preemp-Limiter Menu

If the Pre-emphasis Limiter option is not installed in the FM Pro, then this menu will not be available. If the option is installed, a single page gives control over the limiter on/off, the limiter hardness, and the pre-emphasis/de-emphasis options. To adjust these parameters, place the pointer adjacent to the item then use the Spin dial to set the value.

4.5.3.2.2 Stereo Generator Menu

If the PPDM Stereo Generator option is not installed in the FM Pro, then this menu will not be available. If the option is installed, two pages give control over the stereo generator functions. Page one gives control over the pilot signal on/off and injection. Page two gives control over the mono/stereo modes and the mono modulation reference of 90% or 100%. To adjust these parameters, place the pointer adjacent to the item then use the Spin dial to set the value.

4.5.3.2.3 AES Status Menu

Digital input and outputs are available as an option in the FM Pro. If this option is not installed, the AES Status menu will not be available. If the option is installed, then a single page gives control over the output sample rate and the auto switch feature. The input data rate and error condition are also reported on this screen. To adjust parameters, place the pointer adjacent to the item then use the Spin dial to set the value.

4.5.3.2.4 More or Less! Menu

For a quick adjustment of the on-air loudness, enter the More or Less! menu and use the Spin dial to increase or decrease the FM Pro's processing density. This control simultaneously adjusts a number of parameters in the processor, and, when any processing limit is reached, the More or Less control stops any further adjustments.

4.5.4 Remote Link

The FM Pro has remote control capabilities via factory provided software. From the Remote Link menu, the user has the ability to turn the remote link on or off, specify the type of connection: either modem or RS-232 direct, and observe the status of the remote link.

4.6 Main Menu Page 2 of 2

As previously stated, the Main Menu consists of two pages containing 8 selections, F1 through F8, that lead you to all other menus. Selecting F1 through F8 opens the associated menu by pressing enter. The second page of the Main Menu contains these selections which are described below:

[MAIN MENU P:2.2]
F5 - Set time
F6 - Display mode
F7 - Unit options
F8 - Unit Info

4.6.1 Set Unit Time Menu

Upon entering this menu, the FM Pro's internal clock and calendar will be displayed. You can set both the clock and calendar from this page. To change settings, use all four arrow keys to place the pointer next to the item and use the Spin dial to set the value.

4.6.2 Display Mode Menu

This menu controls the Lock Out time and the Auto Save feature. To set these items, place the pointer next to the item and use the Spin dial to set the value.

4.6.3 Options Page

This page displays if the Stereo Generator, HF Limiter, and AES/EBU options are currently installed in the FM pro. There are no controls in this menu.

4.6.4 Unit Info Page

This page displays the version and date of the software installed in the FM Pro. The telephone number for Aphex Systems customer support is also displayed. Nice touch, right?

end

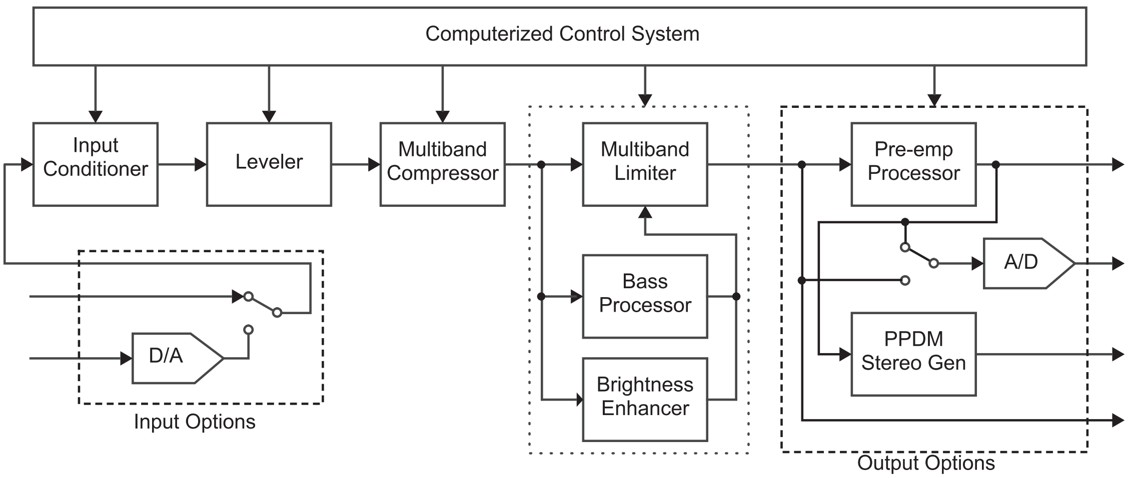
Blank Page

5.0 Detailed Audio Processing Description

5.1 Processing Overview

The Aphex FM Pro is a complete audio processor for FM broadcasting. Contained in an FM Pro are input conditioning circuits, an advanced automatic level controller, a unique and fully adjustable multiband compressor, a new technology multiband limiter integrated with a bass support subsystem, an FM pre-emphasis processor, a PPDM multiplex stereo generator, and a digital audio I/O subsystem. All parameters are programmable through built-in digital control and memory presets. While no additional audio processing is required for outstanding performance, provisions are made to insert outboard processing equipment, such as an Aphex Aural Exciter (tm), if desired. Figure 2-1 illustrates a simplified block diagram of the FM Pro. See figure 5-2 for a complete block diagram.

Figure 5-1 Simplified Block Diagram



5.2 Generalized Processing Structure

It can be seen from figure 5-1 that 6 main elements comprise the FM Pro. A 7th element, the digital audio interface, is not shown but will later be described. Even more detailed information about the digital audio interface is given in section 9, “Using Digital Audio”.

The input conditioner selects the analog or digital source and prepares it for further processing by inserting user selected filters. The leveler automatically and artistically rides gain over the program level. The multiband compressor provides fully adjustable program compression to improve the program consistency, compensate for low quality program material, and allow you to tailor the sound of your radio station. The multiband limiter and bass processor are interactively linked to absolutely peak limit the audio signal while allowing significant bass improvements to be obtained. The interactivity greatly reduces the occurrence of intermodulation distortion. The pre-emphasis processor adds pre-emphasis to the peak limited signal and constantly limits the pre-emphasized signal to fit within the 100% modulation limits. The PPDM stereo generator converts the fully processed audio signal into a stable and precise FM multiplex output.

The pre-emphasis processor and PPDM stereo generator are both optional features. In the case these are not installed, then the final output of the FM Pro would come directly from the multiband limiter.

The “insert loop”, which is not a structural element *per se*, adds yet another dimension to the FM Pro’s processing architecture. You can insert an external device, such as an equalizer or Aural Exciter (tm), at the optimal location: between the leveler and multiband compressor.

The following detailed descriptions will refer to the complete block diagram of figure 5-2.

5.3 Input Control And Conditioning

Several basic functions comprise the input conditioning structure of the FM pro. As shown in figure 5-2, these are:

1. Input Gain
2. 20Hz Highpass Filter
3. 16.5KHz Lowpass Filter
4. Pre-emphasis Assist Filter

5. Spectral Phase Refractor
6. Analog/Digital Input Select

5.3.1 Input Selection

The analog input signal is passed through a digitally controlled gain stage serving as an input reference normalizer. The output of this stage is nominally at 0dBu for a refence input level. A digitally controlled selector gathers either the normalized analog input signal or the output of the 20-bit D/A converter.

5.3.2 Filters

The analog/digital input selector feeds four filters which are each bypassable through the digital controls. These filters are described as follows.

5.3.2.1 20Hz Highpass Filter

This filter has a second order butterworth response. Its purpose is to cut off low frequency rumble or other noise which may be encountered in some systems.

5.3.2.2 16.5KHz Lowpass Filter

This has a fifth order butterworth response for the purpose of cleaning up ultrasonic noise which is sometimes encountered in an STL or other audio system. It does not perform the lowpass filtering for the stereo generator, and is a totally separate filter. This pre-processing filter is not overshoot compensated, but that is not of consequence since all processing including peak limiting occurs after this filter.

5.2.2.3 Spectral Phase Refractor (SPR)

The SPR filter is a fourth order allpass filter designed to enhance the clarity and detail of sound psychoacoustically. It spreads apart the phase of frequencies, particularly in the 100 to 400Hz frequency range in such a manner that it seems to advance the phase of certain bass frequencies. The ear responds by detecting more detail and apparently more bass energy. The ear also hears a less masked high end since the transient edges are slid somewhat apart in time and are thus more detectable as individual events.

A second advantage of the SPR filter is that it “scrambles” the phase of voice frequencies to make the voice waveforms more symmetrical on the average. This can help tame certain voices that tend to distort through processing.

5.3.2.4 Pre-emphasis Assist Equalizer *

This equalizer is a portion of the distributed pre-emphasis method used in the FM Pro. It is switched on for 75 microseconds and off for 50 microseconds. By placing this filter in front of the multiband processor, it helps relieve the final pre-emphasis limiter from the burden of a 75 microsecond equalization curve, allowing that limiter to remain active upon 50 microseconds at all times.

5.4 Frequency Discriminate Leveler (FDL) *

The FDL is an intelligent leveler circuit designed to remain as unnoticed as possible while effectively riding gain over the program level. The amount of automatic gain correction is user adjustable between 15dB of gain and 15dB of loss. There are 8 support systems which help the leveler operate with minimum effects upon the sound. They are:

1. Silence Gate
2. Dynamic Verification Gate (DVG)
3. Sticky Leveling
4. Jump Ahead
5. Attenuation Lock
6. Gain Lock
7. Leveling Rate Control
8. Frequency Discrimination

5.4.1 Silence Gate *

This is a gate which affects only the gain control system. The audio is not being gated. When the input signal falls below the user setable threshold, the leveler gain will be locked at its present value until sound returns at a level above threshold. The background noise will therefore be prevented from swelling up during pauses and silence. The silence gate threshold can be set from 10dB above to 38dB below the input reference level. A 1 second delay is applied to the action of the silence gate to prevent it from interfering with the action of the DVG. The silence gate can be defeated when not needed.

5.4.2 Dynamic Verification Gate DVG *

Like the silence gate, this affects only the gain control system, and audio is not being gated. During program material, the DVG inspects the instantaneous peaks and compares their amplitude with the recent historical average peak value. Only during the brief times when the instantaneous peak exceeds the peak average is the leveler gain allowed to move. In this manner, the slewing of leveling gain is converted to a series of small adjustments which take place at times in the audio envelope when the corrective gain changes will be less noticeable to the ear. The DVG naturally serves as a short term silence gate because brief pauses in program cause it to gate the leveling. However, after about 1 1/2 seconds of silence, the DVG releases control over gating. By that time the silence gate will have detected silence and the leveler gain will be locked by the silence gate. The DVG can be switched on and off as desired.

5.4.3 Sticky Leveling *

Imagine a leveler that just decides to stick at a fixed gain until the input level changes more than a certain amount and that's the FM Pro's sticky leveler. A sticky window in dB is set by the user and the leveler will not make corrections until the input level changes by more than that amount. The window size can be set from zero to 6dB. Usually it will be operated around 1 to 2dB. Small changes in signal level will no longer be tracked by the leveler. This has several benefits. First, the audio distortion caused by the leveler's control ripple will be eliminated for fairly steady signals. Second, The dynamics of the sound will be better preserved, especially with a fast leveling rate. A more open and punchy sound will result.

5.4.4 Jump Ahead

The attack and release of the leveler are intentionally quite slow. If the leveler tracked a fade and the gain became relatively high, for example +15dB, then a sudden loud signal might hit clipping while the leveler takes its time to correct the gain. A jump-ahead circuit detects if the leveler's output has jumped out of bounds and rapidly attacks the leveler gain down to avoid any overload. When the jump-ahead correction brings the gain within 3 to 6 dB of the corrected level, the jump-ahead detector drops away and the leveler continues correcting normally. The jump-ahead feature is not user controllable and is always present.

5.4.5 Attenuation Lock

There may be times when you want to stop the leveler from reducing the gain more than a certain amount The attenuation limiter, a user controllable feature, can be set to stop the leveler's attenuation anywhere between -3dB and -15dB.

5.4.6 Gain Lock

Often it will be desired to set the leveler's gain limit to something less than 15dB. Perhaps you don't want the leveler to correct for low level signals below -10dB, for example. This user controlled feature establishes limit of gain correction between 0 and +15dB.

5.4.7 Leveling Rate Control

The rate at which the leveler is allowed to make gain corrections is user adjustable by this control. The leveling rate is defined as the length of time it takes for the leveler to raise the gain by 10dB. Normally this control will be set between 7 and 2 seconds. The correct adjustment depends on the program format and the density of sound you want. A faster rate increases the density but can sometimes reduce the punch of bass and percussion. An optimum balance can be found between the leveling rate and multiband compression adjustments.

5.4.8 Frequency Discrimination *

The leveler, being a wideband process, does not change the timbre or tonal balance of the program signal. This is a great advantage, especially because it is supported by so many technical innovations to mitigate and eliminate sonic interactions and side effects of leveling. The frequency discrimination of the leveler adds a very significant improvement to the transparency of the leveler. Under frequency discrimination, the leveler attacks more slowly for very low frequencies than for the rest of the frequency range. Beginning at approximately 200Hz, the attack time gradually slows down as the frequency drops until it reaches half attack speed at 20Hz. The release time of the leveler remains normal at all frequencies. The effect of this discrimination is to alleviate the feeling that bass note attacks are being "pulled back" by the leveler. The full punch and power of dynamic bass in music is preserved.

5.5 Insert Loop

A relay controlled insert is supplied for external processing. The I/O interface is unbalanced and is intended for a short distance connection to other equipment. Connections are by XLR connectors, wired with pin 2 hot and pins 1&3 grounded. The output level at this point is nominally 0dBu. External equipment should be adjusted to return an approximate 0dBu signal level.

5.6 The Multiband Compressor

Numerous innovations have been incorporated in the FM Pro's multiband compressor section. It will be observed from figure 5-2 that the compressor bands are created after the VCA's rather than the conventional method of generating crossovers ahead of the VCA's. The compression sidechains are comprised of circuits incorporating wave dependent technology. Cross linking of bands and channels, output mixing, release times, and other parameters are adjustable. The main support features and innovations attached to the multiband compressor are:

1. Post crossover method
2. Wave dependent compression
3. Peak-accelerated compression
4. Tunable crossovers
5. Band mixing
6. Adjustable release timing
7. Band-to band coupling
8. Hard and elastic stereo linking
9. Proprietary VCA's

5.6.1 Post Crossover Method *

Conventional multiband compressors generate the frequency bands ahead of the VCA's and compression detectors. In the FM Pro, the band filters have been placed after the VCA's (hence the term "post crossover"). Although some technical obstacles needed to be overcome, certain advantages were gained by this technique. First, the total noise gained by adding together the sum of four VCA's would normally cause the output noise to rise by 6dB. With the post crossover method there is no noise increase because only the noise of each VCA that can pass through its own band filter is added to the sum. Second, since the lower three bands comprise lowpass or bandpass post VCA responses, much of the harmonic distortion in the VCA caused by control ripple will be filtered out.

5.6.2 Wave Dependent Compression (WDC)*

Aphex perfected it's "Easy Rider" (the commercial name we gave to the wave dependent compressor) compression algorithm and first applied it to studio compressors with great success. We found it also made a perfect multiband compressor in conjunction with the post crossover method. The WDC comprises a convoluting detector arrangement which acts as a peak responding fast compressor, a slow averaging compressor, and both at once. The action of the detector transfers control energy between the two modes seamlessly depending upon the audio waveform. The result is a compressor which will never "pump" and can easily reduce peaks while fattening up the average level. These attributes are just as desirable for multiband as for a single band compressors. The WDC feature is not user selectable, and is permanently active.

5.6.3 Peak Accelerated Compression (PAC)*

When a compressor is operated at a relatively low compression ratio there will be times when a very large and fast transient will pass through without sufficient amplitude reduction by the compressor. The PAC feature allows the compression ratio to increase for fast and transient signals which need more gain reduction above threshold for proper control. This feature marries very successfully with the WDC to help reduce the amount of peak clipping that will be required for adequate peak control after the multiband compressor. The PAC feature is not user selectable, and is permanently active.

5.6.4 Adjustable Crossovers (Band Filters)

All crossovers are first order filters with 6dB/octave slopes, and are user adjustable. Tuning the filters is accomplished by the use of multiplying DAC's and the digital DAC codes are sent by the microcontroller unit. The filters are derived from state variable sections using the DAC's as true attenuators in variable integrator circuits. Thus, the DAC's are not used as variable resistors and remain absolutely consistent providing extremely low distortion and noise.

5.6.5 Output Mixing

The band outputs are each sent through a multiplying DAC for level adjustment, then the DAC outputs are summed equally. At normal mix, all DAC's are attenuated 50% which represents 0dB on the multiband mix control. This allows up to 6dB relative boost of any band or up to infinite attenuation. The DAC's are used in linear feedback mode providing extremely low distortion and noise. No digital potentiometers or digital resistors are used for audio since these are all known to suffer from sonic degradation.

5.6.6 Adjustable Release Timing

The band compressors contain individual release timing adjustments controlled by the microcontroller unit. Only the slow averaging time constant of the convoluting WDC is adjustable. The peak responding portion remains unaltered at all times.

5.6.7 Band Coupling

The “longitudinal” (band-to-band, same channel) coupling links together the slow averaging part of the WDC detectors. This allows you to reduce the long therm equalization effects of the multiband compressor while retaining the frequency discrimination for peak control and transient compression. Any pair of adjacent bands can be linked in any combination, i.e., 1><2, 2><3, 3><4, or any combination of these are selectable.

5.6.8 Stereo Coupling

The left and right stereo channels can be compression linked in two modes, hard and elastic. In the hard mode the rule of “one controls both” is followed. This means that the channel of greater gain reduction controls both channels at any instant. Both channels will track each other quite perfectly at all times. In the elastic mode, the channel with the greater gain reduction at any instant drags upon the other channel, pulling it toward the greater amount of gain reduction. This tends to cause their average gain reductions to equalize while the fast changing compression of each channel remains independent.

5.6.9 The VCA Technology

One of the hallmarks of Aphex products is our extremely high quality proprietary VCA, the VCA1001. Having gone through exhaustive development it is somewhat costly, but it is simply the best audio VCA in the world. In terms of sound it is vastly superior to any other method of dynamically controlling the gain or level. Naturally we incorporated the VCA1001 into the FM Pro at every point where dynamic gain control is used, including the multiband compressor. This accounts in large measure for the very high quality of sound you can achieved with the FM Pro.

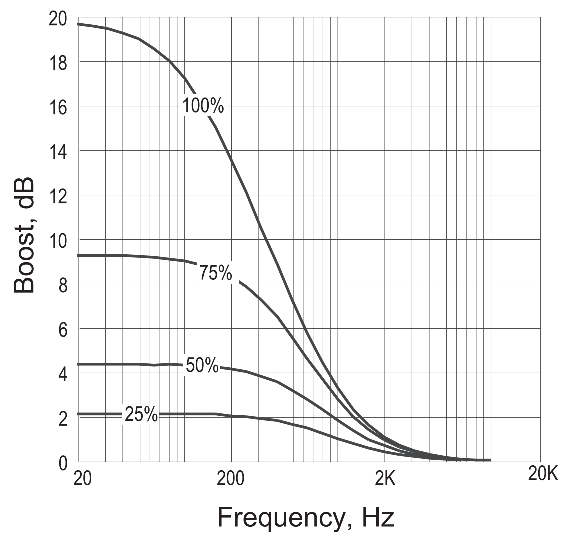
5.7 Split Band Bass Processor/Limiter *

FM broadcasting, when faced with difficult competition, demands rather extreme audio density to attain competitive on-air loudness within the 100% peak modulation constraint. This requires an aggressive peak limiter which can strip off peaks of the audio envelope without generating objectionable distortion. In addition, there is a great demand to create strong bass energy in music programs. The heavy compression and limiting needed to gather loudness tends to restrict the bass rather than expand it. The interactive bass processor and limiter of the FM Pro solve this problem both elegantly and effectively.

5.7.1 Warm Bass Equalizer

The audio signal form the multiband compressor passes through the Master Drive MDAC, under control of the microcontroller, serving as a limiter drive control. The Warm Bass equalizer is wrapped in a feedback loop involving the master drive signal. Up to 18dB of bass boost is available at this point having a boost curve as shown in figure 5-3 below.

Figure 5-3
Warm Bass Equalizer Curves



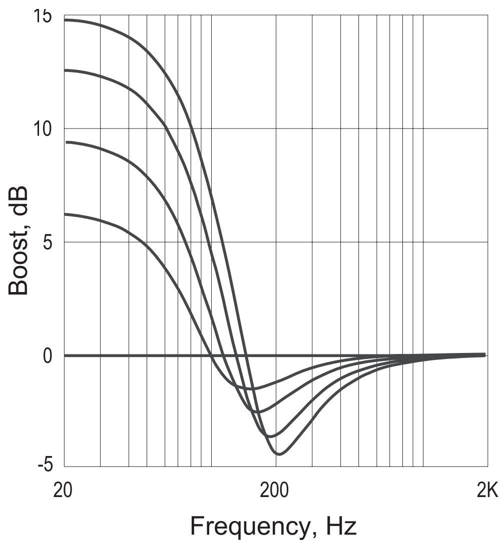
The Warm Bass Eq curve gives a musical and “warm” lift to the low end of audio spectrum without fully isolating the bass from the midrange. Excessive warm bass boost may cause the sound to get muddy. Just the right amount in combination with the proper multiband adjustments and sub bass eq will provide unusually satisfying bass response.

5.7.2 Sub Bass Equalizer

Prior to the split band clipper, but after the Warm Bass equalizer, the Sub Bass equalizer is inserted. This equalizer boosts the low bass frequencies according to a second order shelving response as shown in figure 5-4.

Principally bass frequencies alone are affected. Voices will not be thickened or muddled by its use. At high amounts of boost it can be seen that there is an actual cut occurring in the high bass frequency range. This depression in the net response is intentional and helps to emphasize

Figure 5-4
Sub Bass Equalizer Curves



the lower bass frequencies. With 15dB of boost available at 20Hz, the Sub Bass equalizer should be used carefully. Typically no more than 6dB of boost will be necessary to produce heavy pounding bass, especially when used in conjunction with the Warm Bass boost.

5.7.3 Phase Coherent Crossover *

Once the bass equalization is added to the audio signal, it is split into two bands at approximately 180Hz as shown in figure 5-5 below.

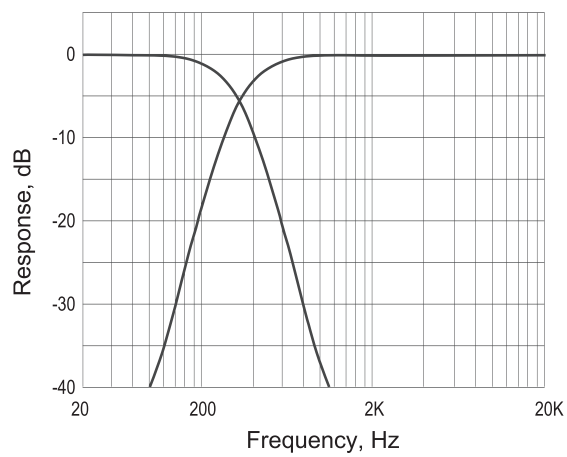
The crossover slope is 48dB/octave which creates a very sharp separation of bands. Phase coherency in a crossover is a new concept introduced by Aphex Systems. In such a crossover, all output frequencies remain in phase between the low and high bands. The time delay of the filters still exists, but the time delay of the high and low bands is always equal for any frequency. This is quite unlike any other “time corrected” or “phase compensated” crossover previously constructed. With a phase coherent crossover, new possibilities exist for audio processing. In the FM pro, we adapted this crossover to a very effective split band clipper inherently comprising distortion cancelling features as will be described.

5.7.4 Split Band Clipper (SBC) *

The basis of the FM Pro’s peak limiting is the split band clipper. Through the SBC, the high band and low band of the phase coherent crossover are processed in a unique and unusual fashion as will now be described.

The low band frequencies, which are for all intents and purposes the bass frequencies, are passed through a dynamic threshold distortion cancelled clipper controlled by an intermodulation detection circuit. The high band frequencies are passed to a summing amplifier and summed with the variably clipped bass frequencies. Following the variable bass clipper is a post-clip lowpass filter to clean up much of the clipping distortion caused by the extreme clipping which can be generated here. The total signal is finally passed through a fixed threshold clipper to exact a perfect peak limit on the output signal.

Figure 5-5
Phase Coherent Crossover



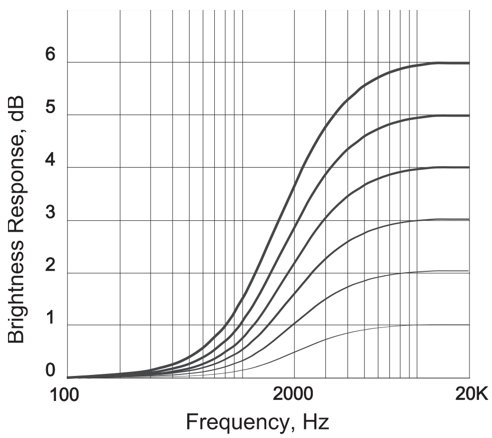
The variable bass clipper threshold is controlled by an intermodulation detector circuit of novel design which can either anticipate or directly detect middle and high frequency clipping that is caused by the presence of large bass waves. The bass clipper threshold is automaticaly rolled back (reduced) to accommodate the presence of other frequencies under the final clipper limit thus eliminating “pinch off” of smaller sound waves at the crests of bass waveforms. Bass waveforms are allowed nearly full amplitude in the final clipper output whenever possible, however. The assertion of bass in the total mix is thereby maximized while eliminating one of the grossest forms of distortion in broadcast processors, that of bass-caused intermodulation distortion.

If the Pre-emphasis limiter option is not installed in the FM Pro, the SBC output will be sent directly to the audio output control system. Otherwise, it will be sent to the Pre-emphasis limiter for further audio processing.

5.8 Brightness Processor

In FM processing, the ultimate need to dynamically limit the pre-emphasis boost takes its toll

Figure 5-6
Brightness Processor Curves



on the audio brightness. The 75 uSec pre-emphasis standard is considerably more troublesome than 50uSec, but both present a serious problem. Multiband compression helps greatly in retaining a bright FM signal, but adjusting the multiband crossovers and mix to achieve satisfactory brightness does not necessarily place the crossovers at optimum locations for compression. The brightness processor permits more optimal tuning of the multiband compressor by adding another layer of control over the presence and brightness of the sound in a manner that is easy to adjust and dynamically adaptive to the program source material.

The brightness processor gives a first order shelving boost beginning at about 2100Hz as depicted in figure 5-6. You have control over the boost from 0 to +6dB. The boost is dynamic by virtue of the brightness processor which measures the magnitude of the frequencies within the boosted shelf, and makes continual adjustments to subtly lift the brightness of dull and deficient material while inhibiting the excessive brightness boost of already bright material.

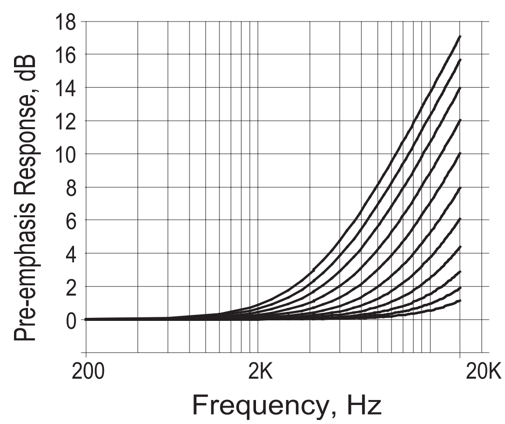
5.9 Pre-emphasis Limiter

The Pre-emphasis Limiter is an optional stereo module which generates and manages a dynamic 50 microsecond pre-emphasis curve, and provides the brickwall bandlimiting required for the stereo multiplex generator. For 75 microsecond operation, the pre-emphasis assist equalizer of the input control and conditioning section is activated as previously described.

Left and right channels are separate and discrete but identical in every way. Each channel of processing comprises a VCA controlled dynamic pre-emphasis generator, and a non-overshooting 15KHz lowpass filter/clipper. The dynamic pre-emphasis generator limits the pre-emphasis induced peak overshoot by sliding the pre-emphasis curve to the right on the frequency axis. Figure 5-7 illustrates the range of pre-emphasis values which are produced dynamically.

The Hardness control sets a range limit on the pre-emphasis slippage for the purpose of preserving more high frequency energy at the potential cost

Figure 5-7
Typical Range of Dynamic Pre-emphasis



of increased clipping distortion. Final peak control after pre-emphasis is by means of clippers embedded in the non-overshoot lowpass filters. The output of the non-overshoot lowpass filter constitutes FM Pro's the final processed output signal.

5.10 PPDM Stereo Generator

This is another optional module for the FM Pro. When installed, it receives the final processed audio signal from the Pre-emphasis Limiter and generates a stable and precise multiplex output.

The stereo and mono modes are selectable as is the pilot mix adjustment. Refer to section 9, “Using Multiplex” for detailed information about this option. Additionally, you might like to look at the appendicies for detailed information on multiplex generators and the Aphex proprietary PPDM technology.

5.11 Digital I/O Module

The AES/EBU interface module is another option available foir the FM Pro. Refer to section 8, “Using Digital Audio” for more information about the the FM Pro’s digital audio capabilities and features.

6.0 Setting Up Processing

The two major functions of FM audio processing are to absolutely control and limit the transmitter's modulation, and to generate a desired sonic character as perceived by the radio station's audience. The FM Pro integrates these functions in a manner that gives you unusual latitude in how you can make your radio station sound. In order to get the great on-air sound you are seeking, you should first become aware of all the features and capabilities the FM Pro has to offer. The actual controls and user interface details are discussed fully in sections 3, 4, and 5 of this manual. While many users will be satisfied with one of the factory supplied presets, many others will want to create their own sound design. This section focuses upon the "how to" part of designing your station's signature sound.

6.1 Steps In Setting Up

Taking a logical approach is helpful in getting the sound you like. The following steps should be taken as you design your sound.

1. Target the processing goals
2. Create a Global preset
3. Select a factory preset to start from
4. Adjust processing
5. Save your preset

6.2 Targeting Processing Goals

You should begin with an idea of what you need the FM Pro to provide. Do you want an open sound, or a wall of sound? Do you want a stressed, intense effect, or a laid back effect? Do you want it as transparent as possible? Are you willing to accept distortion in order to squeeze out an ounce more loudness? You may change your mind as you tune up your processing and listen to the competing stations. You may find that what you thought you wanted is not what you like now. That's all right. You just need a place to start.

One good way to pick a processing goal is to target a successful radio station and make yours sound better. A fact often overlooked when considering competitive FM processing is that how you sound compared to other stations in your market is far more important than how you sound absolutely. For example, if your radio market is filled with trashy sounding stations but they are all pushing the envelope on loudness, you don't have to be perfectly clean, just cleaner and equally loud, to beat the competition.

In your quest to create the perfect on-air sound, you will be faced with the question of how "loudness" factors in. How loudly your station jumps from the dial compared to other stations in your area can easily become a paranoia. We urge you to remain rational. It has been demonstrated in many markets that sounding better, rather than louder, wins the audience. The perception of loudness is also variable. Many programmers and engineers get trapped into gathering fatness and grunge for loudness when actually a station will be perceived as louder and "bigger" by the audience when it sounds more open and dynamic.

6.3 Create the Global Preset

Refer to section 7 of this manual and create a Global preset if you have not already done so. This will match the FM Pro to your system and allow the factory presets to function correctly.

6.4 Select A Factory Preset

One of the eight factory presets (P01 through P08) will probably come close to your processing goal. We evaluated eight widely used radio formats and gathered information about how they typically process their sound. We then built our presets to be competitive against other audio processors for those formats. There are certainly many more than eight identifiable radio formats in existence, and new ones continue to appear. If your FM format is not among those provided for, simply pick one that seems most relevant. You can listen to all of them, if necessary, and then pick the one that comes closest to your needs.

6.5 Adjusting the Processing

This is where the FM Pro gets fun. So as not to fill these pages with redundant material, it shall be assumed that you have previously read sections 2 through 5 of this manual. You also need to skip ahead to section 7 and learn how to create your Global preset if not already saved. You should now understand the menus and processing characteristics of the FM Pro and be ready to get started designing your air sound.

The initial sequence of processing adjustments follows a logical order: the order of signal flow through the processor. It is always ok to jump ahead or back and readjust a processing function, however, and you will do that after you become more expert. For now, we will step through things sequentially. The steps of adjustments are as follows.

1. Leveling
2. Multiband Compression
3. Limiting
4. Bass Enhancement
5. Brightness Enhancement
6. Pre-emphasis Limiting

6.6 Leveling

The leveler provides automatic gain control over your program level. In pondering how to set the various adjustments, consider what you want the leveler to do. It can virtually “jock-proof” your radio station by compensating for widely varying levels or it can merely smooth out variations in an otherwise well produced program stream.

6.6.1 Control Locks

Set the “Max gain” lock according to the amount of correction you need for low program levels. If you want really tenacious control, set it to +15dB. This makes the leveler bring everything down to 15dB below reference level up to full level. A consequence of this tenacity is that it will kill program fades. On the other hand, if you want program fades to be allowed, you should set the Max gain to +6dB. This will make the leveler hold up a falling level only until the signal falls below -6dB. From that point signal will be allowed to continue fading naturally.

The “Max atten” lock can usually be left at -15dB. This allows excessive levels to be fully compensated by the leveler before the multiband compressor gets the signal. In a few rare cases you may want excessive levels above a certain point to go uncompensated by the leveler and ram the multiband compressor to create a dramatic effect. An example of this might be a classical music station wanting to emphasize the orchestral crescendos.

6.6.2 DVG

Generally, the DVG should be turned on at all times except when you are striving for the fattest and loudest, most stressed sound. When on, the DVG allows the leveler to control level in a less obvious manner. If the DVG is off, the leveler can press harder upon the signal envelope and create slightly more loudness at the expense of potential pumping. When on, the DVG reduces the pumping effect and makes the audibility of leveling much less perceptible.

6.6.3 Sticky Leveler

Generally the Sticky Leveler should be turned on at all times, especially for a leveling rate faster than 10 seconds. The sticky window is usually best around 1 to 2dB. You may not always hear the effect of the sticky leveler, but it becomes quite evident with certain program material. You can turn off the sticky to get the fattest, most compressed audio quality. The sticky can dramatically improve very fast leveling (2-3 seconds), especially with voice material and open, punchy music such as found on Jazz and AOR formats.

6.6.4 Silence Gate

You’ll almost always want the silence gate turned on to eliminate background noise swelling during program pauses. The silence gate on/off control is mainly for testing. You should set the silence gate threshold below the lowest expected program signal and above the highest expected background noise. This will usually be between -17 and -38dB. In practice this threshold is not critical since the silence gate is delayed and will not modify the leveler’s action other than to lock the gain during silence. You cannot adjust the silence gate to “flicker” and modulate the leveler’s release rate as on some other FM processors. Typically, a threshold of -38dB (the lowest setting) is appropriate. Test the threshold by stopping the program source and finding whether the silence gate comes on after about 1 second. Don’t do this test by unplugging the audio lines from the FM Pro. You need to see the natural noise floor of the program feed.

6.6.5 Leveling rate

The attack of the leveler is approximately two times faster than the indicated rate. In other words, if the leveler rate is set to 10, it will take about 5 seconds to attack and reduce the gain by 10dB and it will take 10 seconds to release back and increase the gain by 10dB. In the context of leveling, a rate of 2 seconds (the fastest rate of the FM Pro) should be considered quite fast. If your program tends to change levels very quickly, as typically occurs between the microphones of live interview shows, you may want to use the fastest leveling rate of 2 seconds. In such a case you can turn on the Sticky leveler with a 1 to 2dB window to help reduce the perception of gain riding. If you are processing mixed programming and want to leave a sense of texture in the mix, then you should use a moderate rate in the range of 7 to 14 seconds.

Note: The leveling rates associated with the factory presets are pretty close to ideal for their formats. We suggest you use the presets as a basis to your own leveler adjustments.

6.7 Multiband Compression

This is where you will do the most to design your sound. To be sure, this element of the FM Pro is the most difficult to teach because the optimum parameters tend to be interdependent. You will have to experiment with the adjustments to get your best settings. Because the adjustments are grouped and arranged logically, you will quickly develop a friendly relationship with this multiband compressor, however. We offer you the following advice to get you started.

6.7.1 Crossovers

Set the crossover frequencies to enhance the sound as you make other adjustments such as release time, and the mix ratio. The crossovers associated with the factory presets are good choices in most cases.

To keep all 4 bands fully formed, crossover tuning should follow the 4X rule, i.e., a crossover frequency should be at least 4 times higher than the next lower crossover. For example, if F1 is 120Hz, then F2 should be at least 480Hz. Typical ranges for crossovers are as follows. The ranges are typical, not the law.

F1 - 80 to 280Hz
F2 - 330 to 1700Hz
F3 - 3000 to 6000Hz

If crossovers are tuned too close together then one or more of the bands may become tuned out of existence leaving you with a 1, 2, or 3 band compressor. This may be done intentionally by some users who want the characteristics of less than 4 bands. For example, you can tune F1 to 10Hz, essentially taking away band 1. You can also tune F2 to 25Hz, essentially leaving only bands 3 and 4. It is impossible to visualize all the effects of irrational crossover tuning unless you are using the remote control Windows software which graphs the actual shape of the bands for you. Nevertheless, it is completely acceptable to tune the crossovers in any way that works for you. This means that it is not important to keep the bands properly shaped using the 4X rule if the resulting sound is what you like. Do not be afraid to experiment in this area.

6.7.2 Multiband Drive

This sets the depth of compression, i.e., increased drive pushes more compression. The texture and density of the sound can be controlled by the depth of drive and other settings. More compression makes the audio louder and more homogeneous. Light compression makes the sound more punchy and open. Compression depth is indicated on the multiband gain reduction meters. Running more than 12dB gain reduction on peaks would be considered heavy compression. Light compression is under 6dB of gain reduction.

The FM Pro allocates sufficient dynamic range to the Multiband Compressor to allow 20dB of gain reduction (well off the scale) before any distortion creeps in. Do not be afraid to push the drive if you want very heavy processing. On the other hand, the Multiband Compressor has a low enough noise floor to allow light compression without an appreciable noise penalty. With Aphex you have it all.

6.7.3 Band Release

Faster release results in more density and loudness over all, or in a given band.

One strategy for setting the band releases is to tune all four bands at once, starting them at the slowest end of the range and gradually speeding up the time until you reach a “sweet spot”. When all bands are at the same time setting you get a more cohesive sound. All the bands will tend to recover at the same rate which reduces the effect of obvious dynamic equalization.

Another strategy for setting the band releases is to set all the bands to a slow or moderate time and speed up the bands you want boosted or emphasized. For example, you may want to brighten up the mix, so speed up B3 and B4, making B4 the fastest. You may want to peak up the midrange, so speed up B3 only, etc. Varying the band releases in conjunction with varying the band mix will bring you to an optimal compromise between a desired overall tonal rebalance and the desired dynamic program enhancement.

6.7.4 Band Mix

This is where you can set the overall tonal balance of the program. You should start with all bands at 0dB and then boost or cut bands as desired. It is not correct to assume that 0dB in all bands contribute to a flat frequency response. With a multiband compressor, the above-threshold frequency response is always changing depending on the gain reduction of each band. In addition, the bands may have different band-center gain if the crossovers are not tuned sufficiently apart in frequency. It is therefore incumbent upon you to set the band mixes according to the ear, and not by estimation or assumption.

When tuning the band mix, you may end up with none of the bands remaining at 0dB. It would then be a good idea to normalize your mix by going to “update all” and raising or lowering all the mix levels simultaneously until one of the middle two bands lands on 0dB. This will keep your relative mix intact while adjusting the overall mix to a normal level.

6.7.5 Band Coupling

Hard stereo coupling should be chosen when you want the absolute stereo imaging preserved. This option is excellent for Classical music, or a highly definitive Jazz station. Elastic coupling is an FM Pro unique coupling method that allows global coupling, i.e., the baseline compression will tend to track between channels while the faster “peak” compression will remain independent. Elastic coupling is excellent for nearly every format since it maintains the perceptual stereo balance and imaging while preventing a transient event in one channel from poking a gain hole into the other channel.

Band-to-band coupling can be used to reduce the dynamic equalization that occurs with multiband compression. Usually we want the dynamic equalization to remain since it constitutes enhancement of the sound and helps get the unique on-air sound we are after. However, there are times when we may want to keep a flatter frequency response, at least between two of the bands. Band-to-band coupling is elastic like the elastic stereo coupling. This allows coupled bands to remain independent for compression of transients and thus preserves many of the benefits of multiband compression.

6.8 Peak Limiting

The limiter’s master drive control may be considered to be a master gain control operating after the multiband compressor’s 4 band output mix controls. It sets the amount of mixed multiband compressor signal which will be pushed into the FM Pro’s split band clipper. The clip threshold is set to match your transmitter’s 100 percent modulation level. The multiband compression threshold is coupled to the split band clipper in such a manner as to cause the compression threshold to vary as a function of clipping. This allows you to get the most limiting possible within minimum distortion constraints. As you increase the master drive level, you cause a greater depth of multiband compression because the split band clipper is progressively instructing the compression threshold to decrease, thus avoiding excess clipping, causing more gain reduction. The greater you increase the master drive, the louder your signal will become. At some point, as you advance the master drive, you must begin to accept more noticeable distortion to permit the loudness you demand. This point occurs at approximately +4.8dB.

A good method for setting the Master Drive is to start at about 0dB and try to increase it if the loudness is insufficient. There is an optimum balance between limiter drive and compression for gathering loudness. It is sometimes better to highly compress and moderately limit. This brings the most density and that “wall of sound” effect. However, for a more open effect that is still loud, you should try lighter or slower compression and more limiting.

6.9 Bass Enhancement

Consistent, strong bass is an essential part of a competitive air sound. A good method for using the FM Pro’s bass processing is to start with the Master Bass at 0dB and the Warm Bass and Sub Bass boosts at 0%. Mix the multiband compression output as you like, but do not use the B1 mix to get the full bass boost, rather use it moderately like up to about 2.5dB only. Go to the Warm Bass and increase it slowly until the overall bass takes on a slightly boosted feel. Next, go to the Sub Bass and increase it until you hear the lower bass come up noticeably. This will probably occur at around 30% boost. Next, go to the Master Bass and drop it slightly, like about 1 to 2 dB. Finally, go back to the Warm and Sub Bass and adjust them for the overall bass punch and resonance you are after. Dropping the Master Bass has the effect of shelving down all frequencies below 200Hz, and reducing the drive into the bass portion of the split band clipper. This is beneficial because it allows you to equalize the Warm and Sub bass for a very good spectral balance while reducing the overall bass energy drive to the bass interactive clipper. By optimally driving the bass interactive clipper, you can optimize the loudness-presence-bass equation.

6.10 Brightness Enhancement

This process allows you pick up the presence, brightness, and loudness of your air sound without overtaxing the multiband compressor. You will usually find that about 2dB of brightness boost will be sufficient. We recommend adjusting the multiband mix with only moderate boost not over +3dB in bands 3 and/or 4 then using the brightness enhancement to obtain the additional brightness you desire. In some cases, running a brightness boost above +3dB may cause you to observe an increased noise floor. This is due to picking up the high frequencies of the program material. In most cases, the increased noise is greatly offset by the benefits of the bright and sparkling sound quality.

6.11 Pre-emphasis Limiting

There are only two parameters to adjust here. One parameter was set up when you created your Global preset, the pre-emphasis/de-emphasis characteristic. What is left is to set the “Hardness” setting. In the FM Pro, the pre-emphasis is created dynamically and automatically readjusted to prevent high frequency overmodulation caused by the pre-emphasis boost. An embedded distributed clipper is associated with the final non-overshooting 15KHz low-pass filter of each channel. This clipper catches all peak overshoot remaining after the dynamic pre-emphasis limiting. Setting the Hardness to 0% leaves little for the clipper, the work being done dynamically, but the sound tends to become duller due to the sweeping pre-emphasis curve. At 100% Hardness, the dynamics are locked and the clipper does all peak control. This provides a brighter sound but causes sibilance distortion. Somewhere between 0 and 100% Hardness you will find a good setting with no appreciable distortion and good brightness. Generally that will be at 50%, but it depends on your Multiband Compressor mix, limiter Master Drive, and other factors. You should go for the highest Hardness setting that does not cause sibilance distortion.

There is an optimum balance to be found between the Hardness setting and the Multiband Compressor mix. You may have attempted too high a boost in the B4 mix if you have to drop the Hardness much below 50%.

6.12 Save Your Preset

After you have reiterated your adjustments and are at a point you want to save what you have created, follow the steps outlined section 7 of this manual to save your user preset. By saving several variations to presets, and recalling them, you can compare your processing ideas and make appropriate further adjustments. Gaining on a competitor may be as easy as a first time shot, or might require incremental adjustments.

6.13 Tutorial: A “HOW TO” Experience

The following is a slightly different slant on setting up processing, being somewhat anecdotal, and it may shed further insights for many readers.

The first question to consider is format. The type of music will dictate the overall processing amount, equalization, and “personality”. An FM processing system is designed to control modulation primarily, but as competition in the marketplace has increased, so has the need to aggressively process the audio signal to increase apparent loudness. It goes without saying that a radio station that can sound different and even better than the competition can be at some sort of an advantage in the ratings/revenue game.

At Aphex, we believe that high quality and the ability to be competitive in audio processing can go hand in hand.

Once the format type is determined, choose a similar factory preset that will give you a starting place in getting the sound desired. Copy the factory preset to a User preset location and name it. Then recall that preset and begin the fun of making adjustments.

Concentrate immediately on the mechanics. Is the Input level correct? Is the modulation level appropriate? Is the pilot level correct? Make the necessary adjustments to bring the above parameters to the desired levels, then save those settings to the Globals preset. Now go back and insure that you are currently using the preset that was selected and named above. The processor is now able to be customized to fit the stations needs.

Provided modulation level and density are close to expectations, the equalization of the compressor and the bass enhancements sections are the next areas to deal with. Examine the crossover frequency settings , and the gain reduction readings on the multiband compressor. If the compressor seems to be working harder on one band more than the others, and the program material is wideband, then consider moving the crossover frequencies to better suit your needs. Remember that the Multiband equalizer will be affected by the crossover changes, so be careful with any extreme equalization boosts or cuts while changing crossover frequencies. As tuning continues, and the high frequency information is meeting expectations, move to the Limiter/Bass menu. Examine the amount of master drive that the limiter is currently set at, and how much limiting is being indicated by the meter. Provided there are not any strange gain settings, the limiter should be indicating anywhere from 3-6 dB of limiting during normal program material. If the low frequency content is not meeting expectations, either adjust the Bass drive, or move to the next page and make adjustments to the bass enhancement circuits. Additional adjustments may be made to the mix and release times in the Multiband compressor in order to continue to bring the overall tonal balance to a place that works.

Loudness in FM broadcasting is a factor of modulation density. This is generally referred to as peak to average ratio. The Fm Pro is capable of creating a very dense audio signal with very acceptable levels of degradation. Remember, the louder or more dense the signal becomes, the more potential for distortion. If loudness is desired, there are two ways to get it: limit and clip, or compress hard. Combining the two in this processor allows the user the ability to keep the audio as clean as possible, yet increase density. To further avoid artifacts that will be most noticeable on simple voice material when processing aggressively, place the multiband crossover points at frequencies that allow the voices to fall in to one band of the compressor almost exclusively. These suggestions will allow the user to increase loudness by driving the compressor and limiter harder with fewer apparent artifacts.

The less distortion and apparent loudness, the more open and clean a station will appear to sound. No matter the system, this processing tradeoff is always present. The FM Pro has a great many tools that allow the user to increase loudness and density, yet retain brightness in the high frequencies and dimension in the low end. Unlike other processors, the processing systems in the FM Pro are very interactive. The input signal from one section is very dependent upon the output from the preceding section. One of the keys to running the processor aggressively with success lies in understanding what the various sections of the device are doing and how they are interacting with each other as the audio is passed through.

Like any Fm Processing chain, time, patience and awareness will net a great deal of information and direction. Be sure to take the time to be consistent with the tuning of the Fm Pro. Attempt one change at a time, log your changes, and update presets frequently. Listen in many different environments, and develop a language that will allow effective expression of what your ears are experiencing.

end

7.0 All About Presets

7.1 What Are Presets?

Presets are data arrays kept in memory to mass program the FM Pro on demand. Each preset contains all data needed to digitally control every programmable element of the processor.

A total of 24 preset memories are provided. These comprise 8 “factory” presets and 16 “user” presets. The factory presets were designed to serve a variety of programming formats and are stored in read-only memory locations designated P01 through P08. The user presets are stored in flash memory locations designated U01 through U16 and can be repeatedly overwritten as desired.

7.2 What You Can Do With Presets

You can save, recall, and view presets as well as set up day-parting to make the FM Pro change processing characteristics according to a day and time schedule. You can also upload or download your presets as data files through the remote control software. This feature facilitates the distribution of presets in radio groups or between other interested parties.

7.2.1 Recalling Presets

The recall menu allows you to page through all 24 memory locations and selectively recall any preset. When a preset is recalled, the FM Pro immediately transforms all processing functions to the new operating parameters. The transformation is smoothed out by ramping each individual control node to its new value rather than suddenly jumping. This greatly reduces the transition artifacts which could otherwise be generated by switching between widely different presets.

Note: Recalling any preset establishes that preset as the new default for current on-air processing, power-up rebooting and other purposes.

7.2.2 Saving Presets

The save menu allows you to save the operating parameters currently running on the FM Pro to any user memory U01 through U16. The logged-in user name and current time are automatically stored with the selected preset. For protection, you cannot save to any factory preset location. You can create a new name for the preset you’re saving or use the existing name when overwriting an older preset.

Note: Saving a preset automatically makes it the new default preset. The Recall Menu pointer will be moved to the newly saved preset location.

7.2.3 Viewing Presets

The view menu allows you to inspect the list of presets. You can see the preset name, user name, and time the preset was saved, but you cannot modify anything.

7.2.4 Day-Part Scheduling

The day-part menu takes you into the world of automated preset recalls. The day-parting system is composed of 8 day-sets and a rotating 7-day scheduler. The scheduler assigns any one of the 8 day-sets to each day of the week. Each day-set allows you to elect up to 4 events (processing changes) in a 24 hour period. Each event comprises a designated preset and take-over time defined by the hour and minute. In this manner, you can have completely different day-parting for every day of the week, if desired. You can just as easily part every day by the same day-set or part one or more selected days. Any combination of day-sets and weekdays is possible.

Both the scheduler and the day-sets are linked to an accurate self-contained clock/calendar module which can be set through the front panel menu. The high accuracy of the internal clock should make the necessity of correcting it very infrequent, however the clock/calendar can be accessed through the remote control interface permitting machine control using custom software.

Note: Day-parting rotates on a 7-day repeating schedule. Weekly, monthly, and annual scheduling are not supported.

7.3 The Auto Save Feature

If you are working in the Processing menus, and you walk away from the unit without saving your work, the Auto Save feature will save your new settings to user location U16, naming it “Auto Save”. This will occur at the time when the FM Pro reaches the time-out and automatically logs you out. If you log out manually without saving, the Auto Save will also occur. In this way, you will not lose your work even if you neglect to manually save it.

The Auto Save feature can be turned on and off through the “Display Mode” Menu (F6 in the Main

Menu). We highly recommend you always keep it "on" unless you absolutely need the 16th preset memory for a running preset, in which case you wouldn't want Auto Save to overwrite it. If Auto Save is turned "off" then you will lose all your work if you don't manually save it before you're logged out.

Note: Unlike manual save, when the Auto Save updates U16, it does not make U16 the new default preset. It's simply there to save your butt.

7.4 Do I Really Need To Create Presets?

The answer is yes, at least one. You will need to set up the global parameters and save to user preset U01. This procedure is described in later sections of this chapter.

Note: One of the joys in life is building and using FM Pro presets. Likely as not you'll get into creating your own sound designs right from the start.

7.5 Classes Of Presets

There are three classes of presets in the FM Pro: factory presets, user presets, and the global preset. Every preset has a memory designation and a name. Memories are designated by P01 through P08 for factory presets and U01 through U16 for user presets. The global preset occupies U01 and is a special case user preset as will later be explained.

7.5.1 Factory Presets

There are 8 factory presets comprising the memory locations P01 through P08. These are designed to complement a variety of generalized program formats, and are named accordingly. Perhaps not everyone will love the factory presets, but they were developed in consultation with experienced broadcasters to assure reasonable suitability. Whether you like them or not, you will find them an excellent starting place to build your own presets.

Certain key parameters are not implanted in the factory presets and must be inherited from the global preset U01. This method allows the factory presets to generate their intended effects while adapting readily to the constants of your specific system. The global parameters are discussed more fully in section 7.4.3. of this chapter.

7.5.2 User Presets

There are 16 user presets comprising the memory locations U01 through U16. These presets are re-writable by users having read-write log in privileges. Unlike the factory presets P01 through P08, user presets are independent of the global parameters of U01, i.e., presets U02 through U16 inherit nothing at all from preset U01. This allows you to re-globalize your user presets if you wish. For example, you would need to create a user preset with different global settings if you wanted to switch inputs (analog or digital) as a day-parting function.

Note: You may conveniently use preset U01 as a completely normal user preset even though it serves double duty parenting globals to the 8 factory presets P01 through P08.

Note: Out of the box, all user memories U01 through U16 come loaded with the same parameters as factory preset P01 with an arbitrary set of globals.

7.5.3 Global Preset

User memory U01 is an otherwise normal user preset doing the double duty of parenting certain global parameters ("globals") to the factory presets P01 through P08. The globals are routine parameters (listed below) kept individually in all the user presets but purposely not implanted in the factory presets. This is done simply to allow adaptation of the factory presets to every user's operating environment.

Although preset U01 is initially named "Globals", it can be renamed by a user just as any other user memory without changing it's global function.

Note: When initially starting the FM Pro, you should set up the unit's parameters to match your requirements then save to U01. This will subsequently allow full and free use of the factory presets.

The following table summarizes the "global" parameters that are linked to the factory presets from U01:

The Global Parameters

A. Processing Menu, Input/Output Menu pages 1,2,3
- Input reference level
- Peak output level
- Unit bypass on/off
- 20Hz High Pass Filter on/off
- 16.5kHz Low Pass Filter on/off
- Spectral Phase Refractor (SPR) on/off
- Stereo Insert on/off
- Input Source: A or D
- Output: A, D, or A&D

B. Processing Menu, Pre-emphasis Limiter Menu
- Pre-emphasis filter, 50 or 75 microsecond
- De-emphasis on/off

C. Processing Menu, Stereo Generator Menu pages 1,2
- Pilot Mix percentage
- Pilot on/off
- Input mode: Stereo, Mono (L, R or L+R)
- Mono Mode 90% or 100%

D. Processing Menu, AES Status
- Transmit rate: Slave, 32, 44.1, or 48kHz

7.6 Tutorial--Building And Using Presets

Perhaps the best way to teach is by example. This tutorial walks you through the steps of recalling a preset, building the global preset U01, and then setting up a day-part schedule.

7.6.1 Menu Navigation

Figure 7-1 shows the menu tree as it will be referenced in this tutorial. Three branches are shown leading to: recalling presets, saving presets, and making a day-part schedule. You can get an easy grasp of the logical menu paths simply by glimpsing at this diagram.

Once you are through the log in procedure, the screen will show the Main Menu. If you somehow advanced past the main menu, simply repeat pressing the ESC key until the Main Menu appears. Among the Main Menu selections is F2, the selection for Presets. Select F2 and press enter.

You are now in the Preset Menu. From here you may select Recall Preset (F1), Save Preset (F2), View List (F3), or Day Part (F4). Selecting these items will open other menu pages providing further choices. To avoid confusion, the individual presets are always referenced the same way and contain the same data wherever they are listed in the various menus.

7.6.2 Recalling A Preset

Action: Select (F1) from the Presets Menu

When F1 is selected the screen will show the page containing the last recalled preset and the cursor will point to that particular preset. (Note: There are six pages in Recall Presets tree. The page you are on is indicated at the upper right hand of the screen.) To scroll to another preset on the page use the up/down cursor keys. To turn to another page use the left/right cursor keys.

To recall a preset, set the cursor on the desired preset and press the Enter key. The chosen preset will appear in the lower right corner of every screen. To return to the main menu press the Escape key.

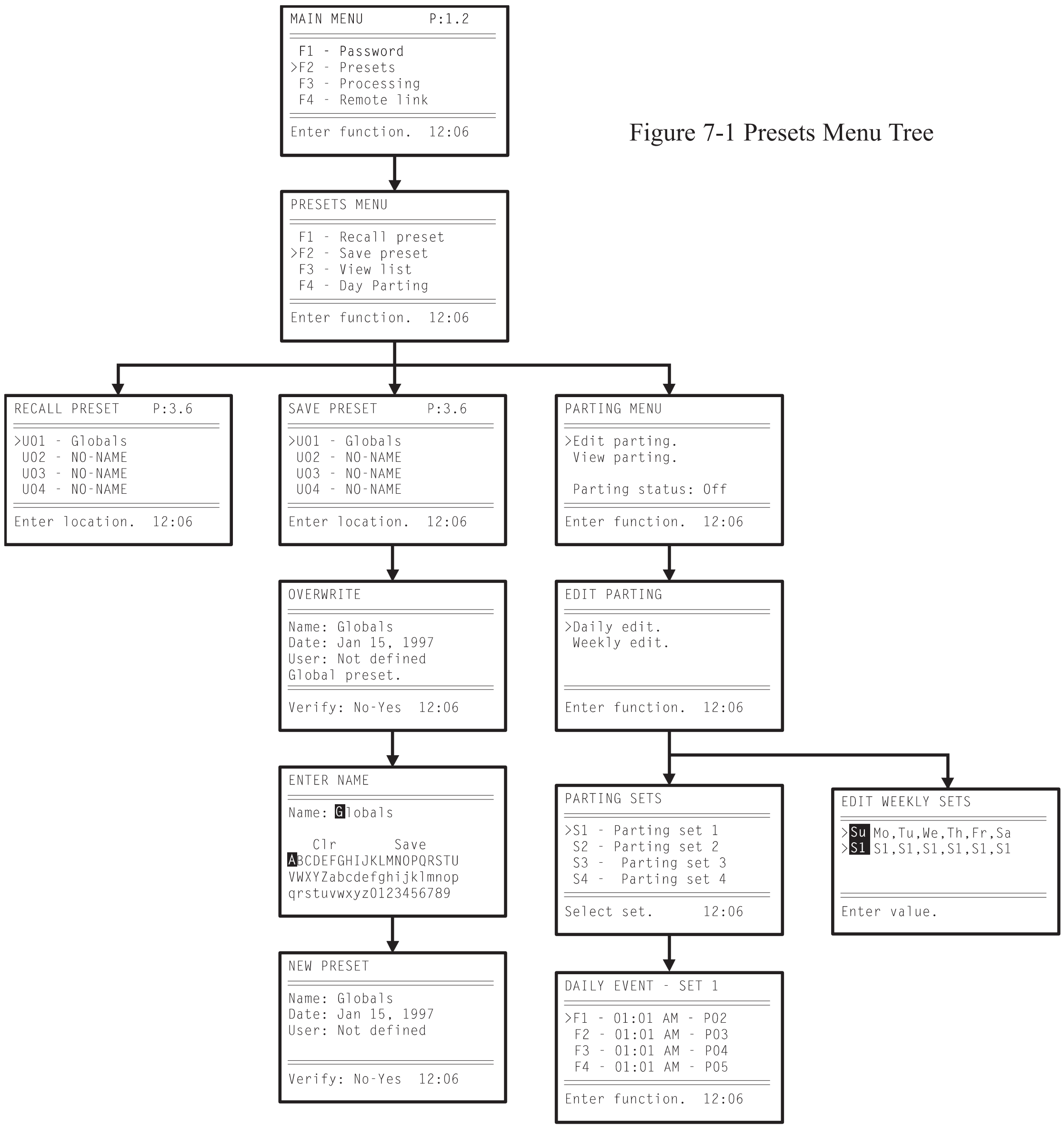


Figure 7-1 Presets Menu Tree

Note: When the unit wakes up for the first time, Factory Preset #1 (P01) is loaded into all the user presets U01 through U16.

7.6.3 Saving A Preset (and making the global preset)

In order to kill two birds with one stone, we'll show how to make and save the global preset. This will help you get your FM Pro initialized while at the same time teaching you the procedures used to save any other preset.

Action: Recall a starting preset

Once you set up the global parameters you are ready to start building your own presets. Choose any factory preset which you think may be close to the sound you want to achieve. Modify that preset through the Processing Menu if you wish, although right now that is not necessary.

Action: Set up the global parameters

Go through the Processing Menus and reset all parameters listed in 7.4.3 above. Some of the parameters may be blocked out of the menu depending on which options are installed in your FM

Pro. You needn't be concerned about any blocked out parameters since the unit deals with them automatically. You may have already become familiar with setting up the processing, but If you don't know what to do with a feature (for example, "insert loop", or "SPR"), just turn the feature off. Make sure that at least your audio I/O levels are set approximately correctly and you have the correct input selected (analog or digital). Also be sure the 'Insert Loop" is turned off unless you have an external device connected to the insert loop.

Action: Save preset
When you want to save the work you have done go to the Preset Menu and select Save Preset (F2). The first screen will come up with the particular page containing the last recalled preset. On the bottom, the screen will ask you to 'Enter Location'. Scroll through the pages, and, this time, choose U01, "Globals". At other times you may choose other memory locations. Be careful not to chose a location containing a preset you don't want to lose, since it will be overwritten by your new preset.

Note: You simply cannot save to a factory preset.

Once the user preset is selected, press Enter. The next screen is labeled "Overwrite". If you didn't want to save the changes, you could choose 'No' by pressing the Escape key and you will be brought back to Save Preset Menu. Since you do want to save the changes, press the Enter key.

Once you have chosen to save the changes, the next screen is 'Enter Name'. If you want to keep the name already on the preset, use the curser keys to move the cursor past the existing name over to 'Save' and press the 'Enter' key.

If you want to create a new name for the preset, pick the desired characters by scrolling through the alphanumerics using the Spin Dial. Press Enter to select each character as you build the name. After 'Enter' is pressed the cursor will automatically move ahead one space. You can also move the cursor manually using the left/right cursor keys. For a blank space use the position after the number 9 and press 'Enter'. After you have created the name, scroll to 'Save' and press the 'Enter' key.

Once you have entered 'save' in the 'Enter Name' menu, the "New Preset" screen appears to ask for verification. It will show the name for the preset you've entered and will update the date to the current date. It will also show the name of the user who is currently logged in. It will ask you to verify Yes of No. You still have a chance to back out and recover the old preset you are attempting to overwrite. If no, press 'Escape' and you will be brought back to the Save Preset Menu. If yes, press the 'Enter' key and your preset will be saved along with the new name. If you do not change the name, only the date on which you have made changes will be updated.

7.6.4 Day Parting (F4)

From the Parting Menu you can choose 'Edit Parting' or 'View Parting' as well as 'Parting Status' on/off. In order to build 'Weekly' parting, 'Daily' parting must be built first.

Note: At this time you might as well check and set the system clock/calendar. This is done from the Main Menu, F5, "Real Time Clock". You'll avoid a lot of confusion by having the correct time and date in the system.

Action: From the Parting menu, turn parting off.
For the present, until your day parting schedule is ready, turn off the day parting. Cursor to the "Parting Status" position and use the Spin Dial to turn the parting off.

Action: Choose "Daily Edit"
After choosing 'Daily edit' the next screen will be 'Parting Sets'. There are eight separate parting sets (S1 through S8) listed in two pages.

Action: Choose a parting set to edit
Select one of the sets, i.e., S1, and press 'Enter'. That will take you to the next screen which is labeled 'Daily Events' for the set you have chosen.

Action: Edit the parting set
Each set contains up to four changes (events) for a 24 hour period. Select an event (E1 thru E4) by using the up/down cursor keys. Move the cursor using the left/right keys to time and adjust the time using the Spin Dial. Move the cursor to AM/PM and select by using the Spin Dial.

Note: It is not necessary to program the events in order of time, although it would be easier for you to review the schedule if they are in chronological order.

Note: If you set the same time on two or more events, the lowest numbered event is enacted, while the others are ignored by the scheduler.

Move the cursor to the right hand side of the screen and using the Spin Dial choose a preset (P01 thru P08 or U01 thru U16). If you spin past U16 the screen will show ‘Off’. This means that event is cancelled.

Note: You should set all unused events to ‘Off’.

Note: Whenever you change anything on a Daily Event (Set) Menu, the changes are taken to memory automatically. You do not have to save the settings.

Action: Chose “Edit Weekly”
This opens up the “Edit Weekly Sets” menu which controls the rotating weekly scheduler.

Action: Schedule the weekdays
Select days using left/right cursor. Select parting sets to associate with each day using the Spin Dial. If you want to use the same set for all days move the cursor to ‘Set all’ and, using the Spin Dial, select the desired set which will be attached to all weekdays at once.

Note: Whenever you change anything on the “Edit Weekly Sets” menu, the changes are automatically taken to memory. You do not have to save the settings.

Action: Turn day parting back on.
From the Parting Menu curser down to “Parting Status” and use the Spin Dial to turn it on.

Action: You’re done already!
The day parting is now set and operating, the globals are set, and you know how to wander through the menus and recall or save presets.

7.7 Summary

You have now learned the flexible yet simple preset system of the FM Pro and no doubt you are feeling all warm and fuzzy about that. We hope you will now take advantage of the many possibilities offered by the FM Pro, live long, and prosper.

end

8.0 Password Security System

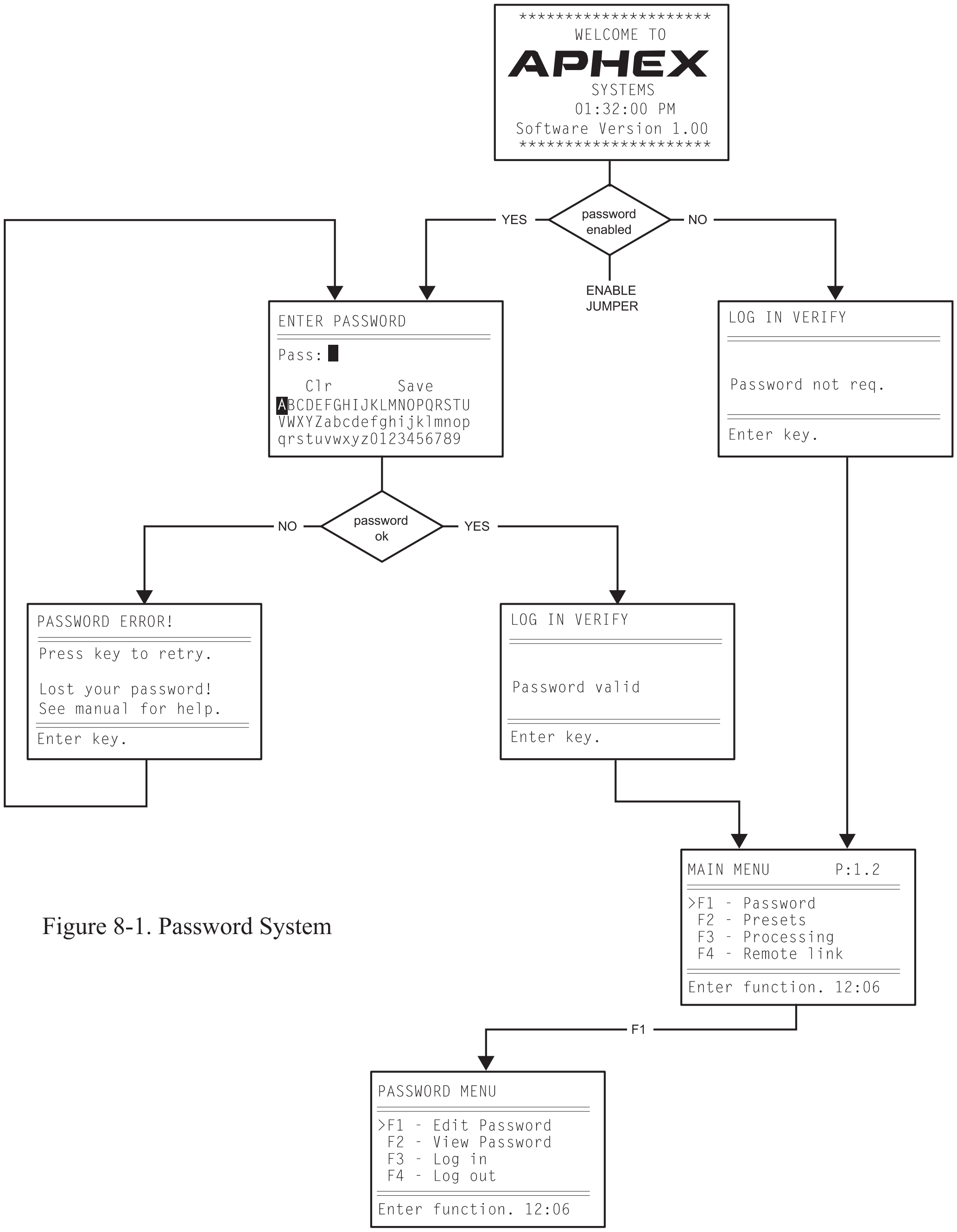


Figure 8-1. Password System

8.1 Why A Security System?

In most modern broadcast installations, it is important to prevent any unauthorized viewing or changing of the processing. This protection is achieved in the Model 2020 when the password security system is enabled. In applications which do not require this password security feature, the password requirement can be disabled.

8.2 Log-in Access

To pass from the logo screen to the main menu you must log in. Pressing the enter or escape key at the logo screen will either prompt you to enter the password or tell you no password is required depending upon whether any password has been set up.

Note: If no password is required, then you will

be notified by the screen and allowed to enter the main menu by again pressing the enter key. If a password has been set, then the password entry screen will be displayed and you will need to successfully enter the password in order to gain entry to the main menu.

8.3 Logging Out

8.3.1 Manual Log Out

After you finish adjusting the FM Pro, you should log out to protect the FM Pro from tampering. This is a simple procedure accessed through the main menu "Password" function. Select "Log Out" and press enter. The logo screen will appear indicating you're now logged out.

8.3.2 Auto Lock Out

The FM Pro will automatically log you out after a period of inactivity. The period will be either 1 or 5 hours, depending on how it is set in the Main Menu "Display Mode" page. Any front panel operation resets the timer back to zero, so the lock out will not occur unless you remain logged in but leave the unit alone for a long enough period. This is a very handy way of making sure the unit will return to a tamper proof mode should you walk away and forget to manually log out.

Note: We suggest using the 1 hour Auto Lock Out if you are likely to receive frequent work interruptions. During extended uninterrupted sessions, use the 5 hour setting.

8.4 Auto Save

As a convenience to you, the auto lock out feature will also save your current operating parameters to the Auto Save memory U16 when "Auto Save" is enabled. This feature is activated through the same Main Menu "Display Mode" page where you find the Auto Lock Out time.

Note: If the Auto Save feature is set OFF, then the at the instant the Auto Lock Out function logs you out, the FM Pro will revert back to the preset which was in effect at the time you logged in, or the last preset you saved while working. You will lose any unsaved work you did while you were logged in.

If the Auto Save feature is set ON, then at the instant the Auto Lock Out function logs you out, your new settings will be saved to memory U16, automatically named "Auto Save", and the FM Pro will revert back to the preset which was in effect at the time you logged in, or the last preset you saved while working. This saves you from losing your work should you forget to save it to your own user preset. You can later come back and retrieve your work from the Auto Save memory.

8.5 Password Enabling

There are two password enable methods in the FM Pro, hardware and software. The hardware enable consists of a movable jumper located inside the FM Pro's chassis. This jumper is only accessible by removing the top chassis cover. As shipped from the factory, this jumper is set to <u>enable</u>. When the jumper is set to enable, the software enable is active. When the jumper is moved to disable, the software enable is defeated at all times.

8.5.1 Software Password Enabling

Assuming the hardware is enabled, (hardware jumper set to <u>enable</u>) the software enable exists implicitly by setting up a password or deleting the password. If the password is null, i.e., the word is blank, then the password system is software <u>disabled</u>. If a password is set up, i.e., at least one character is entered and saved as a password, the password security system is software enabled and you will thereafter have to enter the password to gain log in access.

8.5.2 Hardware Password Enabling

The hardware password jumper serves two functions. First, it permanently enables or disables the password capability depending on whether it is moved to the <u>enable</u> or <u>disable</u> position. Second, it can be used to reset the password in software to null (blank) by first booting up the unit with the jumper in the disable position and then rebooting with the jumper in the enable position. This jumper is located inside the unit on the front panel computer board assembly.

Note: As shipped from the factory, the hardware password jumper is set to <u>enable</u>, and the password in software is blank.

8.6 Access With Lost Password

If, for any reason, you lose the password and are locked out of your FM Pro, you can reset it to null and start over with a new password. Doing this requires somewhat drastic measures, discouraging unauthorized persons from gaining access to your audio processing. The only way to overcome an unknown password is to clear the password memory using the hardware password jumper as follows.

8.6.1 Clearing The Password Memory

To reset the password to null: remove ac power, take out the the 14 cover screws, and remove the cover. Locate the password jumper on the front panel computer board and move it to the disable position. Apply power and boot up the unit. You should now be able to access the Main Menu without a password. Power down and move the jumper back to the enable position. Replace the cover and restore the unit to operation. You can now enter a new password or operate without a password.

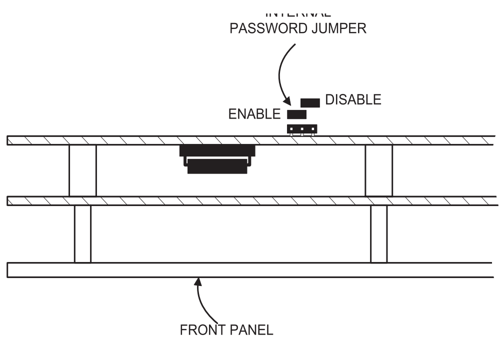


Figure 8-2. Hardware Password Jumper

8.7 Setting The Password

Once logged in, you can create a password or change the existing password. Enter the “Password” page from the Main Menu and select “Edit Password”. The password edit screen will appear. If a password exists it will appear in the edit screen. Simply overwrite it and save. If there was no previous password, the edit screen will be blank. Simply enter a new password and save. From now on the new password will be required to log in.

8.8 Clearing The Password

Should you want to clear the password and run without password security, follow the above procedure but overwrite the existing password with blank spaces. Once the blank password is saved, the password system becomes “software disabled” and you will not be asked for a password for subsequent log-ins.

end

Blank Page

9.1 Connecting AES/EBU Lines

The FM Pro digital audio interface operates with professional AES/EBU standard specifications. If your digital audio cables are somewhat lengthy, your cabling should use only 110 ohm twisted-pair shielded wire designed specifically for digital audio use. For very short runs of a few feet, standard twisted pair cable is usually acceptable. Keep cables as short and neat as possible to avoid noise pickup. Remember, digital audio signals are equivalent to radio frequency transmission and can suffer similar propagation anomalies. To avoid generating or receiving interference, pay attention to proper assembly of the XLR cable connectors.

9.1.1 Cable Pinout

The following is the correct wiring pinout for AES/EBU cables: Pin 1 shield; Pin 2 positive signal; Pin 3 negative signal

9.2 AES/EBU Synchronization

The AES/EBU receiver will lock up to incoming sample rates between 25 and 55 kilosamples per second (KS/s). Once lockup occurs, the FM Pro's AES/EBU status screen will display the standard incoming rates of 32K, 44.1K, and 48KS/s. The received digital audio bitstream is subsequently fed to the analog-to-digital converter.

```
AES STATUS
=========================
 Transmit rate: SLAVE
>Auto Source: OFF
 Rcv. rate: 44.1
 Rcv. error:
=========================
Enter value.
```

Figure 9-1
Processing, AES Status Menu

9.3 Digital-To-Analog Conversion

The FM pro utilizes a no compromise 20-bit digital-to-analog converter to receive the AES/EBU digital input signal and generate high quality analog for audio processing. We realize that, at the present time, there is virtually no chance that full 20-bit digital audio will be available to the FM Pro. However, developments presently in progress will soon open the door to vast improvements in broadcast digital audio. The FM pro will be ready for all such improvements. Furthermore, use of a 20-bit a/d converter assures there will be no appreciable noise or distortion generated by the converter itself, leaving the question of audio quality entirely up to the digital audio source.

9.4 Auto Source

If Auto Source is turned on, the FM pro will automatically switch over from the digital to the analog input when digital audio data errors are detected. Error levels detected, in order of severity, are are as follows :

Validity bit high
Confidence flag
Slipped sample
CRC error
Parity error
Bi-phase coding error
No lock

You can set which error level is the lowest level that will trigger the Auto Source. The Auto Source will trigger after 10 error hits are detected within a 1 second interval. The Auto Source will therefore not switch over for the minor data errors which are normal in many digital STL's. After an Auto Source trigger, if no more errors are detected for an additional 4 seconds, the Auto Source switches back to accept the digital input.

If you are using the Auto Source mode, then you need to have a suitable analog input available to continue the broadcast program. Many users will maintain an analog STL or land line feed to the FM Pro as a backup to their digital audio link. If no analog input is supplied to the FM Pro, then when Auto Source switches to the analog input, your FM modulation will go silent.

Note: To select the lowest error level you want to trigger the Auto Source, move the curser pointer to Auto Source in the AES/EBU menu and turn the Spin dial to scan through the error list, stopping at the desired indication. To turn the Auto Source off, turn the Spin dial until "OFF" is indicated in the data error list.

9.5 Digital Output

The FM Pro's digital audio output is generated by a no compromise, drift stabilized, 20-bit analog-to-digital converter. You can select asynchronous output sample rates of 32, 44.1, and 48KS/s, or you may slave synchronize the digital output to the digital input.

Many users will see the benefit of asynchronous digital output. For example, if your digital STL runs at 32KS/s, 16-bits, you can come into the FM Pro at that rather inferior rate and resolution, process the sound in analog thereby elegantly reconstructing the audio waveform, and resample out at 48KS/s, 20-bits. Your digital FM exciter will happily accept the higher sample rate and resolution making your FM transmission all the better with much lower distortion and spurious content.

Note: Thanks to an Aphex patent pending design, the digital output of the FM Pro is free of all dc drift and bounce. All thermal offset drift of the a/d converter has been eliminated. Furthermore, the dc offsets of the analog audio stages are fully regulated by dc servo control. There will be no long or short term frequency drift or bounce of the FM exciter caused by the FM Pro.

9.6 Facts About Digital Audio, Bit Rate Reduction and Dynamics Processing

Digital audio is a sampled and quantized approximation of the original analog sound. The higher the number of bits used to digitize an audio signal, the more closely it's reconstructed analog signal will resemble the original. The number of quantization levels available in the digital domain is a measure of the digital resolution of the audio signal. Too few quantization levels results in harsh distortion known as "grunge" or "splatter". There can never be too many quantization levels.

16-bit quantization has become a standard resolution in most of today's digital audio systems. Although this moderate digital precision has been found generally satisfactory for mass consumption, many people find that the presently available 18 and 20-bit digital audio is vastly superior in terms of realism and the reduction of noticeable digital audio anomalies. The reason is that, even if large amplitude waves can be quantized adequately in the 16-bit domain, small signals reflecting significant sonic details may be grossly underquantized or even lost.

The sample rate of digital audio directly affects the precision of the reconstructed sound as well. A low sample rate, i.e., 32KS/s, results in a higher noise floor, a more spurious noise characteristic, and a narrower audio bandwidth. Higher sample rates improve not only those factors, but also permits better signal processing within the digital domain. It is desirable to have the highest sample rate possible and the largest number of bits possible.

The transmission and storage of digital audio requires a data bandwidth directly proportional to sample rate and geometrically proportional to the data size (number of bits). System economies forbid extravagant quantizing and sampling of digital audio, and compromises must be reached. Unfortunately, the compromises needed to make digital audio practical for most purposes are severe. For this reason, not only is the digital audio found in broadcasting usually of low sample rate and quantizing (32KS/s, 16-bits), but there may also be lossy data compression applied.

Lossy data compression, for example Musicam or Apt-X, will bring down the required data bandwidth, but causes further degradation of the sound. Whether the degradation is audible depends on the listener's criticality, of course, and upon many other factors. Multiple compression passes may be encountered in a complex distribution path and each compression stage may build upon the last causing severe damage to the sound. For this reason it is advisable to always avoid using compressed data pathways.

As stated before, the sound quality of digital audio improves as resolution is increased. It is therefore good practice to maximize the analog signal input amplitude to the analog-to-digital converter in order to obtain the maximum available digital resolution. However, the potential danger of driving too close to maximum input level is digital overload, a most unpleasant sound. To create sufficient headroom, standard practice is to establish the 0VU reference level of digital audio at 18 dB below peak clip. With each bit comprising 6dB of the available dynamic range below clipping, 0VU signals thereby receive only 13dB quantization. Small but important signals 20-30dB down from reference become only 8-bit audio. It is not hard to see why much digital audio can sound harsh and gritty. Furthermore, once a signal is poorly digitized there is no way to increase its resolution.

The quality of digital audio can be noticeably improved by raising the 0VU reference level closer to clipping, and by using a high quality analog brick wall peak limiter such as the Aphex Dominator to prevent audio peaks from ever exceeding the a/d clip level. Of course, the Dominator could also be used to raise the density of the sound, getting a very loud and highly resolved digital conversion, if desired. When 18 or 20-bit digital audio is converted to 16 bits, the lowest bits are either truncated or dithered to noise. With either method, it would be best to maintain as much of the audio in the higher bits as possible to maximize resolution. Therefore, an 18 or 20 bit analog-to-digital conversion system also benefits from the Dominator.

Linear digital audio requires wide spectrum for transmission and large space for storage. For example, one channel of 16 bit sampled at 44.1kHz translates to more than 700,000 bits per second. The requirement for ever larger drives and wider transmission systems has been answered in part by various methods of bit rate reduction. Although sometimes called “data compression” they are all ‘lossy’ systems, which means that resolution is reduced, thus the noise floor and distortion are increased while subtle (sometimes not so subtle) details of the audio are lost.

One of the prime methods of all the systems to reduce data is the use of the psychoacoustic principle of masking. Essentially, the theory states that a higher level signal will mask lower level signals within a certain frequency range surrounding the higher level signal frequency, depending on the relative levels of the signals. The higher the high level signal is in comparison to the low level signals, the wider the masking frequency range. The frequency range of the masking effect is called the “critical band”. The bit rate reduction system discards the supposedly masked signals in the critical band and retains mainly the masking signals.

While there are differing opinions regarding the audibility of the artifacts of the various bit rate reduction schemes, the controlled listening tests upon which the proponents have obtained acceptable results have employed reproduction systems which were as flat as possible. There were no dynamics processors such as audio compression and equalization placed in the reproducing system. This means that data compression techniques are not designed with the anticipation that any dynamics processing will be used. In broadcasting this can be a problem.

Dynamics processing, by definition, will change the level relationships between high level and low level signals. Since the most common form of broadcast audio processing is dynamic range compression, the high level signals will be reduced in level in comparison to the low level signals. Multiband audio processing will continuously change the masking relationships of the sound. This explains why data-compressed digital audio subjected to broadcast audio processing will often become exceptionally dirty and grungy because the masking algorithm presumed for data compression becomes greatly violated.

It would seem that if one must use a data compressed digital audio link, it would be better to place the link after the FM Pro, rather than before. In that way the FM pro could receive uncompressed digital audio, process it, then send the processed audio out to the digital link whereby the digital data compression would not be subjected to further audio processing. The problem with this proposal is that the highly processed audio from the FM Pro is too dense to fit the supposed masking parameters for inaudible digital data reduction. The data reduction processor has to make the data fit the available bandwidth one way or another, so, having no reasonable masking opportunities, it makes whatever severe and disastrous alterations to the digital audio that are necessary. It is, in fact, better to place the data compressed digital audio link ahead of the FM Pro. This arrangement, although still undesirable, usually yields better sound than data compressing the FM Pro’s output.

If a data compressed digital audio link absolutely must be used ahead of the FM Pro, then maximizing the quantization level by use of the Dominator to control audio peaks and a Compellor to ride average gain in the analog domain ahead of a/d conversion can mitigate some of the audio degradation of the data reduction system, and give the FM Pro a better quality digital audio input.

One final note about using data compressed digital audio. The digital algorithms of data reduction used in digital STL’s will sufficiently modify the audio input waveform to cause considerable peak overshoot. This will not be a problem when the digital link is placed ahead of the FM Pro since the FM Pro will re-limit the peaks prior to transmission.

However, if the STL is placed after the FM Pro, the STL digital audio output will not be suitable for direct connection to a digital FM Exciter. For these reasons and reasons already given, we recommend placing the FM Pro at the transmitter site in all cases involving a data reduced digital STL.

Note: If a linear (uncompressed) digital STL is used, then it theoretically would be possible to maintain the FM pro at the studio and send the AES/EBU output up the STL directly to a digital FM exciter. At the time of this writing such an STL is available only on a T-1 wideband link. However, some T-1 equipment does rate conversion or other signal processing. You need to test your link for digital overshoot prior to making a commitment to this operating method.

end

10.0 Using Multiplex

10.1 Stereo Generator Option

Although the stereo generator is a field installable option, we recommend ordering it factory installed. This allows us to perform a thorough calibration of the unit for you. However, should you insist on doing it yourself, you can rest assured it will perform well as long as you correctly follow the installation procedure. Field installed stereo generator options are shipped with installation instructions which you should later insert in this manual, preferably at the end of the present section.

The FM Pro stereo generator is similar to the PPDM circuit used in the world renown Aphex Digicoder. PPDM (parallel path digital modulation) technology is an Aphex exclusive patented method of generating a perfect stereo multiplex signal. You may refer to Appendix 1, “Stereo Generator Theory With PPDM Explained” for detailed information on PPDM and general stereo generator information.

10.2 The Stereo Generator Menus

From the Main menu, go to the Processing menu. On page 2 of the Processing menu select F6, “Stereo Gen.”. There are two stereo generator menu pages. Page 1 gives you the pilot settings and page 2 gives you the mode options. These two pages are shown below.

```
STEREO GEN
==========================
 Pilot mix:         9.00%
>Pilot On/Off:         ON

 Pi:
=====MIN=============MAX=
Enter value.        10:16
```

```
STEREO GEN
==========================
 Input mode:  Stereo
>Mono mode:    100%


==========================
Enter value.        10:16
```

Figure 10-1
Stereo Generator Menus

10.3 Presets Memory

The mode and pilot injection level is saved in the user preset memory whenever a preset is saved.

10.4 Mode Settings

Stereo generator modes are saved in the user presets. You can therefore save presets for stereo and mono operation, and run them as day parting events if you wish. When starting the FM Pro for the first time, you should set the stereo generator and save to the global preset U01 so the factory presets may inherit the mode and thus work properly in your system. Refer to section 7.

10.4.1 Mono Modes

Great flexibility is afforded the user by providing three mono modes: Mono-L, Mono-R, and Mono-L+R. You can also decide if the mono mode should remain at 90% peak modulation as for the stereo mode, or be expanded to use the full 100% peak modulation limits. Normally you will want the Mono-100% mode, and indeed most other stereo generators provide only this mono mode.

Occasionally we encounter broadcasters who switch from stereo to mono simply for the purpose of inserting an additional subcarrier signal for a certain period during the day. In this case it is necessary to limit the mono modulation to 90%. In effect, the 9% pilot injection is being replaced by the additional subcarrier. For those broadcasters who need it, we provide the Mono-90% mode of operation. Just another thoughtful service of Aphex technology and design.

10.4.2 Stereo Mode

In the stereo mode, the peak modulation of left and right channels hits 90% modulation while the pilot is customarily adjusted by the user to equal 9% modulation. This condition will be restored whenever stereo mode is selected regardless of which mono mode may have been in use.

10.5 Pilot Function

The pilot signal is automatically switched off when any mono mode is selected, and on when the stereo mode is selected. Once you are in the stereo or mono mode you can switch the pilot on and off for testing or other purposes. For example, you can switch on the pilot while in a mono mode, and switch off the pilot while in the stereo mode. Regardless of the state of the pilot in any mode, it will automatically be returned to the normal state when another mode is selected. The user may adjust the pilot injection from approximately 7%

to approximately 11% modulation. The typical setting is 9% modulation. Though probably not good practice, you can create different pilot injection levels in different user presets if you wish.

10.6 Multiplex Output

The stereo multiplex output level is adjusted by a rear panel multiturn precision trimmer. The level is adjustable from zero to approximately 7 volts peak (14Vp-p) at 100% modulation. The output impedance of the BNC jack is approximately 10 ohms sourced by a high current video output driver circuit to maintain extremely fast slew rate and a strong drive capability. This will reliably drive up to 100 feet of low capacitance coaxial cable, either terminated or unterminated.

10.7 Multiplex Cable Connections

Many stereo generators suffer from the effects of output cable capacitance, and their manufacturers tell you to keep the output cable under 6 feet long. The FM pro can drive much longer cables without a problem either of distortion or stereo separation. You can use either 50 or 75 ohm coax line without any noticeable difference in performance. We have found no reason to terminate the line in practice since the bandwidth of the multiplex signal does not reach a high enough frequency to excite the transmission line characteristics of a typical cable. You may, however, use a line terminator if you wish. It is best to use low capacitance coaxial cable, especially for runs over 20 feet long.

10.8 Modulation Overshoot Problems

Once you have adjusted the multiplex output level to satisfy proper total peak modulation limits of your transmitter, the pilot injection indicated on your modulation monitor may be significantly lower than the indication on the Stereo Generator menu. If this occurs, then you are experiencing a transmission problem between the FM Pro and your FM exciter, or with the FM exciter itself.

10.8.1 STL Problems

Any filters or amplifiers in the path from the FM Pro's stereo multiplex output to the FM Exciter input will probably introduce modulation overshoot. A composite STL typically introduces such overshoot, although newer types are available which almost eliminate the problem. It cannot be over-emphasized how important the STL can be to your on-air sound and loudness.

One way you can test your STL is with a squarewave generator and oscilloscope. Feed the tone into the STL transmitter at around 20% modulation and observe the receiver's output on the scope. Of course, this can most easily be done with the transmitter and receiver on the same workbench, but you can also check an installation by having workers at each end of the link. First use a squarewave frequency of 50Hz. The output wave should be nearly rectangular with less than 3% trapezoidal tilt. This test reveals the typical limitation of most STL's which is an inadequate phase lock loop filter in the FM modulator. Next, sweep the squarewave frequency upwards through 20KHz. There should be no tilting or ringing visible on the output. Unfortunately, there is little you can do if your STL does not pass muster except to purchase a better set of STL equipment.

10.8.2 FM Exciter Problems

Older FM exciters (and many newer types, especially units used in Europe and the far east) produce modulation overshoot as a result of inadequate low frequency response. Highly processed audio containing heavy bass frequencies will cause the exciter to overmodulate.

You can investigate your FM exciter using the squarewave technique. For this you need an FM demodulator with d.c. response. Typical FM monitors are not adequate in that regard. Our experience has shown you cannot trust even the very expensive European FM modulation analyzers for this test. One fairly reliable method is to use a common old fashioned FM receiver with the AFC defeated. You can look at the FM detector's output with a d.c. coupled scope probe (prior to the stereo decoder) to see a good demodulated replica of the FM. An FM exciter should be capable of modulating a 50Hz squarewave with less than 3% trapezoidal tilt. Unfortunately there is little you can do besides purchase a better FM exciter should your exciter not pass the test.

10.8.3 Composite Clippers

If your system has overshoot caused by the STL, but not the FM exciter, then you may be able to mitigate the problem by using a composite clipper on the STL composite output. We highly recommend avoiding composite clippers whenever possible, but to compensate for a bad STL we would relent. The use of composite clipping will introduce additional distortion to the sound and generate spurious frequencies which will fall into the subcarrier frequency band. The result can be to create a synthetic multipath distortion effect which will reduce your effective broadcast coverage area. Please do not use a composite clipper beyond the minimum clipping needed to compensate for STL overshoot. The FM Pro is designed to create high on-air loudness without composite clipping.

10.9 Using RDS Encoders

The FM Pro does not directly support an RDS encoder interface. If you intend to use an RDS encoder, you can insert it as an inline device or you can let it lock up to the multiplex pilot and inject its RDS subcarrier into one of your FM exciter auxiliary inputs.

end

The FM Pro can be remotely operated by a suitable personal computer running Microsoft Windows 3.1x or Windows 95. Remote control software is normally shipped with the FM Pro but is also available from any Aphex dealer and it can be downloaded from the Internet web site at http://www.aphexsys.com. The computer may be connected to the FM Pro directly by an RS232 cable or by a telephone modem.

11.1 Hardware Requirements

1. 386-33 or higher class personal computer.

2. Windows 3.1, 3.11, 9x, 2000 operating system

3. Minimum 4MB RAM

4. 2.5MB Hard Disk Space for program files

5. 3.5"-1.44MB Floppy Drive

6. One available serial port (com1, 2, 3, or 4)

7. 800x600 or better Color or Monochrome Display

8. Pointing device (mouse, trackball, etc.)

9. Two modems-- one for the P.C. and one for the FM Pro (no modems needed for port-direct connection).

11.2 Software Installation

We constantly strive to improve our software. Please take into account any changes that may be implemented in newer software versions and alter your interpretation of these instructions accordingly. Check the 2020 web site at www.aphex.com for new release information.

The FM Pro comes ready for remote operation, so you will not be required to install software into the unit itself. You need only to install the remote control software onto your own computer.

To begin installation, insert Install diskette 1 and, from the run menu, enter a:\setup then click OK. If your floppy drive letter is other than a:\ then simply substitute the correct drive letter, i.e., b:\setup.

The setup program will automatically install all required components into your system and create a directory named C:\FMPRO to contain the program files. You will be instructed to insert additional diskettes as needed.

11.3 Communications Cable Hook-Ups

You can connect your computer to the FM Pro through a direct RS232 cable or a pair of modems. The serial cable will differ depending upon the type of connection

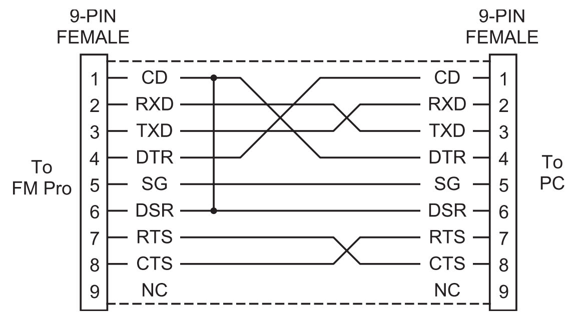


Figure 11-1 9-Pin To 9-Pin Null Modem Cable

11.3.1 Direct Connection

The direct connection uses a "null modem" serial communications cable to connect your PC to the FM pro. Commercially available null modem cables come with various configurations of male and female connectors. The FM Pro wants a 9-pin female cable end while your PC wants either a 9-pin or 25-pin female cable end. Simply obtain or construct the proper cable and use gender changers, if necessary. If you make your own cable, then follow the pinouts of figure 11-1 or 11-2.

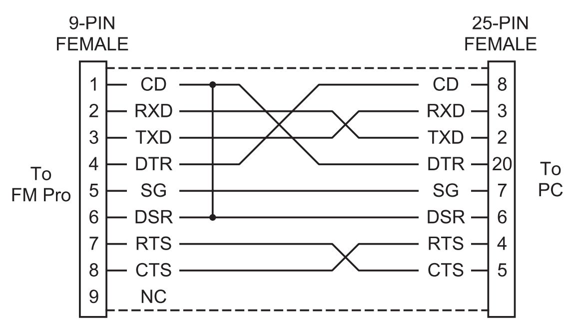


Figure 11-2 9-Pin To 25-Pin Null Modem Cable

11.3.2 Modem Connections

11.3.2.1 Modem To FM Pro

For remote control through telephone lines, an external modem must be connected to the serial port of the FM Pro. Most external modems contain a 25-pin female serial port connector. Therefore you will need to obtain a standard modem cable, not a “null modem” cable as described above, with the proper connectors attached. If off-the-shelf cables are unavailable, you may may construct the cable according to the wiring diagram of figure 11-3.

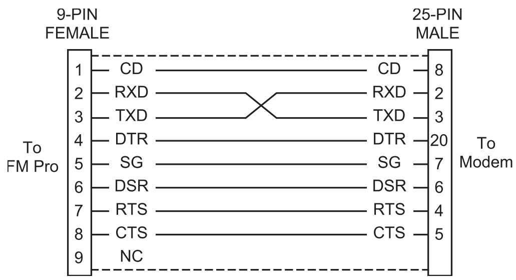


Figure 11-3 Modem to FM Pro Serial Cable

11.3.2.2 PC To Modem

You may use either an internal or an external modem with your PC. If using an external modem, you’ll need a standard serial cable from the PC serial port to the modem. You must not use a null modem cable. Your PC can have either a 25-pin or 9-pin serial port, therefore you need to get the appropriate cable and possibly an adapter. For a 25-pin serial port the proper cable contains a 25-pin male and a 25-pin female connector. For a 9-pin serial port the proper cable contains a 25-pin male and a 9-pin female connector. If off-the-shelf cables are not available, you can make the cable according to figure 11-4.

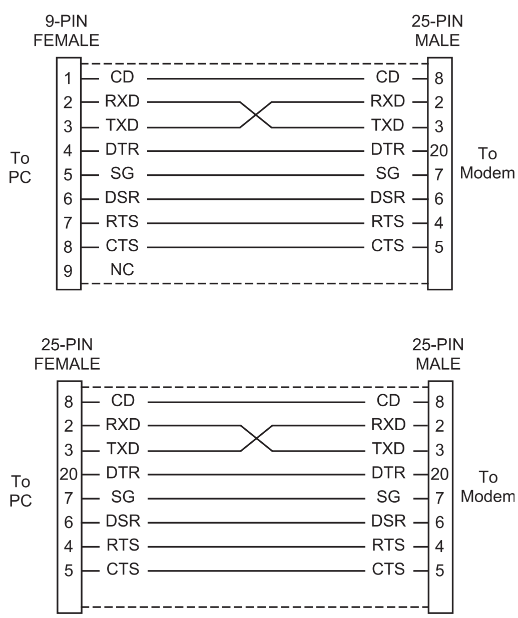


Figure 11-4 Modem to PC Cables

11.4 Operation By Modem

11.4.1 Usable Modem Types

Generally, any Hayes-compatible modem capable of 9600 baud or greater should work. Newer V32.bis modems will connect at the highest speed the line quality supports, usually 14,400 to 33,600 baud, while older fixed rate modems will just lock up at their specified baud. The serial communications of the FM Pro and the PC are set to 9600 and need only to see support for 9600 baud from the modems. The newer, faster modems usually offer better performance because of improved error handling.

11.4.2 Modem In The Windows Operating System

Even though the Aphex remote software runs under Windows 3.1x or Windows 95, you do not have to configure the modem for the Windows environment. We take care of all modem control through the software. If your PC modem is already configured with windows drivers that is not a problem. You only need to check which com port it is attached to. If the modem is internal, usually it will be configured to Com4 but you should double check your modem configuration so you will select the correct port in the remote program.

11.4.3 Modem Installation Procedure

When setting up the FM Pro for a modem connection, please follow these steps to properly initialize and activate the modem.

1. Power down the FM Pro.
2. Connect the FM Pro to the modem by a standard (not a ‘null modem’) cable.
3. Power up the FM Pro and the modem.
4. In the remote link menu, set the remote link “OFF” then set the link type to “modem”. You can only set the link type when the remote link is off. Lastly, set the remote link to “ON”. The link status should first show “wait” and then “M-OK” indicating that the modem is initialized. The FM pro will force the correct modem settings regardless of how the dip switches may be set on the modem.

Note: Do Not set the “Remote Link” ON if the modem is not connected! This will cause the FM pro to lock up it’s front panel for about 20 seconds.

The FM pro will continuously check if the modem is still connected. In case of power interruptions, the FM Pro will automatically recover itself and re-initialize the modem.

11.4.4 Troubleshooting The Modem

If you have problems connecting to the modem, here is an easy way to check if the units are communicating properly.

1. Power-off the modem.
2. Turn off the modem’s auto answer feature using the modem’s dip switches or jumpers. If using a US Robotics modem, flip switch #5 “DOWN” to set auto answer off.
3. Set remote link “OFF” and link type to “modem”.
4. Power-on the modem. The AA (auto answer) light must be off.
5. Set remote link “ON” and watch the modem’s AA light. When you turn the remote link on, the AA light must come on. If the light comes on, that proves the FM Pro is commanding the modem and the unit should work properly. If the modem’s AA light does not come on, then either the modem or the cable is defective.

11.4.5 Compatibility Issues

We have experienced problems supporting some manufacturer’s modems. Our lab has verified that the 33.6K baud modems by US Robotics, Best Data, and Supra will work with the FM Pro. If you have another brand of modem and you find it works well with the FM Pro, we would like you tell us so we may add it our list of known modems. As time goes by, we will post additional modem information on the Internet at our web site **www.aphex.com**.

11.5 PC Remote Control Main Form

When you first start the program you will see the Main Form of figure 11-5. This form links you to all the program functions such as the control panels, presets, and communications. The various indicators on this form show you how well the data is being passed between the FM Pro and your PC.

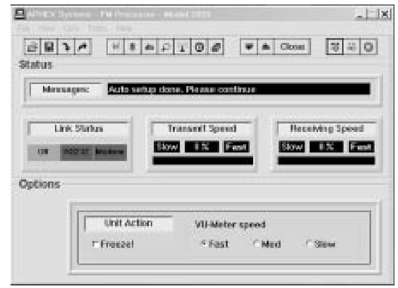


Figure 11-5 Main Form

11.5.1 Link Status

This indicator shows you if your link status is working, and whether by direct line or modem. The operating condition is indicated in green.

Note: The FM Pro must have the Remote Interface feature turned on and set to the correct mode (modem or direct) or you cannot establish a remote link.

11.5.2 Transmit Speed

This shows how efficiently the data is being carried from your PC to the FM Pro. Data is not always being transmitted, so this indicator may be blank or at a low value until you begin moving settings.

11.5.3 Receive Speed

Your PC normally receives a continuing stream of meter data. This indicator allows you to see how efficiently the receive data is streaming through the link. If your modem is operating on a bad phone line, you may see this value fall low and the meters on the panels may become erratic.

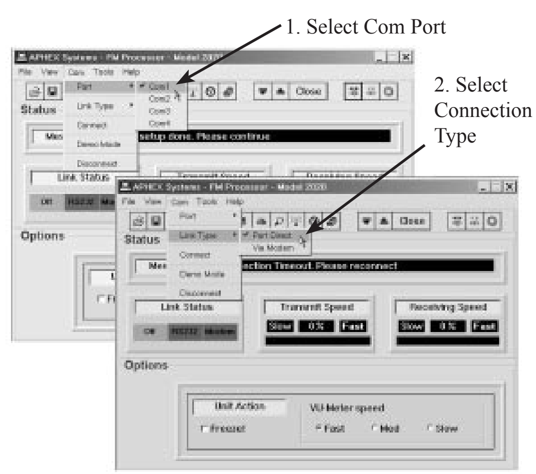


Figure 11-6 Establishing a Connection

11.6 Port Direct Linking

To begin direct communications, open the "Com" menu and select the com port you will be using. Next, select "Port Direct" out of the "Connect" choices. If the FM pro is properly connected to your PC, then you will immediately receive a series of messages indicating the success of the connection.

Note: You must set the remote link status on the FM Pro to "DIRECT" (not "modem") to allow direct linking.

If there is a password in the FM Pro, you will be shown a dialog box requiring the password to be entered. Once the password is correctly entered you'll be able to continue.

11.6.1 A Note On Day Parting

If day parting by the FM Pro is currently in effect at the time you are logging into the remote link, you will get the dialog box of figure 11-7. This box allows you to decide if you want to turn off the day parting, leave it on while you work, or put it at rest. When at rest, the day parting is temporarily shut off while you are connected, but automatically turns back on when you disconnect the remote link.

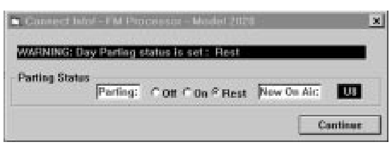


Figure 11-7 Day Parting Dialog Box

11.7 Modem Linking

Follow the same procedures as 11.6 above, but select "Modem" instead of "Port Direct". A dialog box will open up indicating the modem is being initialized. When the modem initializes, you will be then be given an input box to enter the telephone number which must be dialed. While the number is being dialed, you can abort by pressing "cancel" on the dialing message box. If a password is in effect, you will be asked to enter the log-in password.

Note: You must set the remote link status on the FM Pro to "MODEM" (not "direct") to allow modem linking. To avoid trouble, please read the information about modem hook-ups included later in this section.

11.8 Using The Remote Control Panel

From the Tools menu, select Controls. This opens up the tabbed control of figure 11-8, giving you access to all the FM Pro programmable functions. Each major processing group can be reached by clicking the appropriate choice tab. To adjust any variable control, simply put the pointer on the handle, hold down the left mouse button, and drag the mouse pointer.

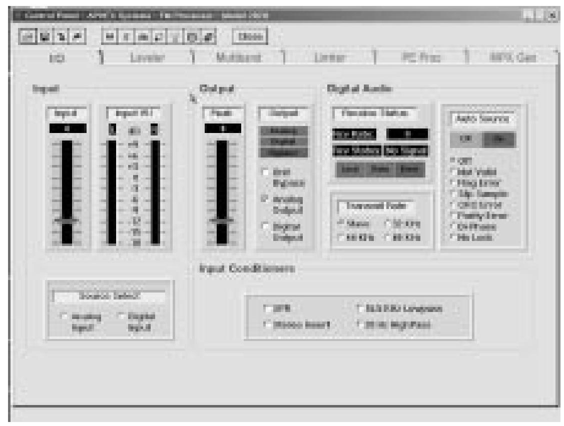


Figure 11-8 Remote Control Panel

11.9 Dealing With Presets

There are several ways you can deal with presets in the remote link. You can import and export presets to and from data files and you can recall or save presets in the FM Pro memory. Recalling and saving are similar to what you can do on the FM Pro front panel itself. Import and export are additional capabilities given to you by the remote software program. You can upload one or more presets to a file on your PC or floppy disk for transport to other FM Pro units, and you can download presets from such a file. Export files contain the extension *.fmd.

11.9.1 Recall Preset

From the Tools menu, select Recall Preset. This opens the recall form which proceeds to download all the presets currently in the FM Pro's memory including the factory and user presets. Simply select the preset you want from the list and click recall.

11.9.2 Save Preset

From the Tools menu, select Save Preset. A panel will open showing you all the user presets currently in the FM Pro and allowing you to chose one location for your new preset. Once you chose the location, click Save. The preset editor opens and allows you to name the preset, and add other information. When you again click Save the information will be uploaded to the FM Pro's presets memory.

11.10 The View Menu

This menu gives you several useful choices such as the current on-air preset, day parting status, and unit info from the FM Pro. You can readily find out what software version is running on the FM Pro through this menu.

end

12.0 Product Specifications

12.1 Getting Meaningful Numbers

Once an audio processor exceeds the simplest single-band structure, traditional performance specifications tend to become irrelevant. The usual test tone measurements will almost never reflect the performance of the processor under normal operating conditions. Specifications for dynamic conditions, if possible to measure, would require specially built measuring instruments unavailable to the user. Therefore, the user would be incapable of verifying the specifications.

The above facts notwithstanding, there is a demand from various entities for a set of standard specifications. The following list constitutes a rational set of specifications based upon typical or normal operation of the unit. The user should be able to verify these specifications by direct measurement using measuring instruments equivalent to the following types:

1. Audio Precision System One
2. Belar Laboratories FMSA-1 Digital FM Stereo Monitor
3. General purpose 100MHz oscilloscope

12.2 FM Pro Setup

The specifications will be given under conditions of the following FM Pro setup. The setup parameters, as given, establish the net gain and operating level approximately equal to normal operating conditions. The leveler is locked to zero dB gain while the multiband compressor is fully released. Any operating parameter not shown may be considered inconsequential to the specifications.

Unless otherwise specified, the analog measurements are taken from the FM Pro stereo multiplex output jack and decoded through the Belar FMSA-1 stereo monitor. Measurements are taken from the FMSA-1 left and right test outputs and analyzed by the System One, or measured directly by the FMSA-1 as indicated in the specifications list.

The FM Pro Setup for Specifications

Input/Output Menu

Input Reference:	+4dBu
Output Level:	+12dBp
Input :	Analog
Pre-process Filters:	All Off

Limiter

Master Drive:	+3dB
Bass Drive:	0dB
Warm Bass:	0%
Sub Bass:	0%

Leveler

Rate:	2 Sec
Gain limit:	0dB
Atten Limit:	3dB
DVG:	Off
Sticky:	Off
Silence Gate:	Off

Pre-emphasis Limiter

Limiter:	On
Pre-emphasis:	75uS & de-emphasis
Hardness:	50%

Multiband Compressor

Xovers:	200, 2000, 10000
Drive:	-20dB
Release:	All bands = 2 Sec
Mix:	Adj. for flattest response (typical +.8,0,-1.2,+1
Coupling:	All Off

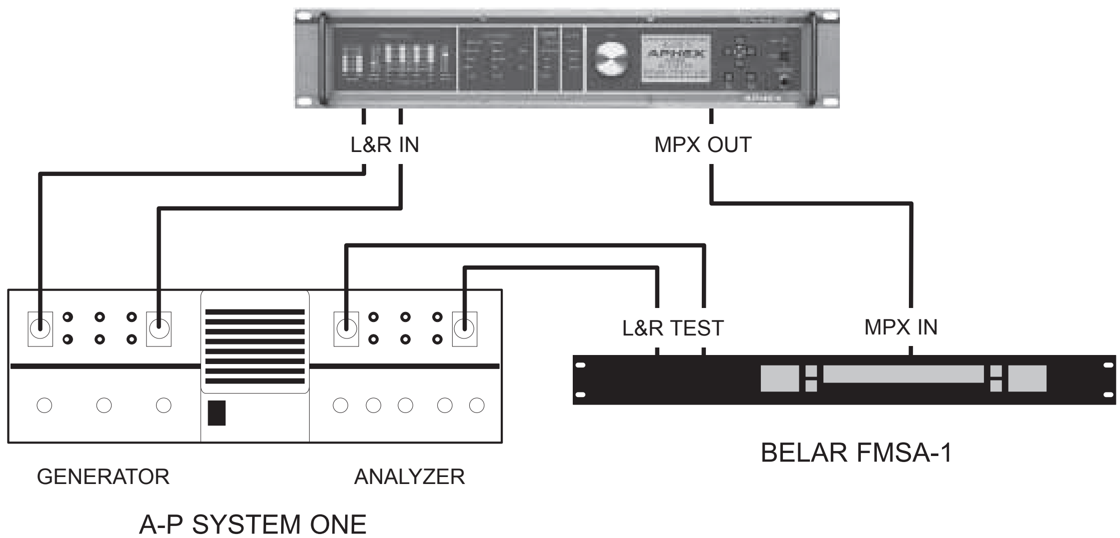
Stereo Generator

Mode:	Stereo
Pilot:	On
Pilot Level:	9%

12.3 Test Equipment Connections

Figure 12-1 shows how equipment was arranged for development of the specifications. It is not the intention here to instruct you specifically how to measure the specifications, however. Please refer to the section on test and calibration for detailed measurement instructions.

Figure 12-1 Test setup for the specifications relating to analog I/O.



12.4 The Specifications

12.4.1 General Analog

Internal Frequency Response
1Hz to 70KHz +/- 0.1dB

Basic Pre-emphasis Accuracy
+/- 0.1dB 20Hz to 15KHz disregarding any audio processing alterations

Basic THD (for all signals below clip threshold)
Demodulated MPX output, 0dB 1KHz tone input, THD <0.05%

Active Process Distortion (typical worst case)
Demodulated MPX output, 0dB 1KHz tone input, CHR factory preset, THD <0.5%
Note: This typifies peaks only, and not signals below clip threshold.

Stereo Output Noise (left or right channel)
CHR Factory preset, 20Hz-20KHz measurement bandwidth re 100% modulation = -70dB

Mono Output Noise
CHR Factory preset, 20Hz-20KHz measurement bandwidth re 100% modulation = -71dB

Processing Peak Overshoot
Less than 1.5% above 100% modulation

System Stereo Separation
Greater than 65dB 20Hz to 15KHz

.

12.4.2 Analog Input

Configuration
Left and right

Input Impedance
10K Ohms

Common Mode Rejection
>70dB 50Hz - 20KHz

Sensitivity
-24dBu to +10dBu for nominal input level

Maximum Input Level
+27dBu

Connector Type
XLR 3-Pin Female EMI Suppressed. Pin 1 chassis ground, Pins 2 & 3 electronically balanced, floating and symmetrical. Pin 2 in in phase with multiplex and digital output

12.4.3 Analog Line Outputs

Configuration
Left and right. Flat or pre-emphasized

Source Impedance
62 Ohms electronically servo balanced

Load Impedance
600 Ohms or greater balanced or unbalanced. Termination not required.

Maximum Output Level
+24dBu onto 600 ohms balanced, +27dBu unloaded

Connector
XLR 3-Pin male, EMI Suppressed. Pin 1 chassis ground, Pins 2 & 3 electronically balanced, floating and symmetrical. Pin 2 in in phase with multiplex and digital output.

12.4.4 Digital Input

Configuration
Two-channel AES/EBU standard. Pre-emphasized or non pre-emphasized.

Sampling Rate
32, 44.1, and 48KHz auto detect and lock

Connector
XLR 3-Pin male EMI Suppressed. Pin 1 chassis ground, Pins 2 & 3 transformer balanced and floating

Input Data Size
20-bits

Input Frequency Response
1 to 20KHz +/- 0.1dB

12.4.5 Digital Output

Configuration
Two-channel AES/EBU standard

Sampling Rate
Input clock-slaved or independently selected at 32, 44.1, or 48KHz.

Connector
XLR 3-Pin Female EMI Suppressed. Pin 1 chassis ground, Pins 2 & 3 transformer balanced and floating

Output Data Size
20-bits

Output Frequency Response
1Hz to 20KHz +/- 0.1dB

12.4.6 Composite Baseband Output

Configuration
Single output with output level control

Source Impedance
10 ohms suitable to drive long or short coaxial cable

Load Impedance
50 ohms or greater

Output Level
Adjustable from 0 to 13V peak-to-peak with multiturn output control

Connector
BNC, EMI Suppressed.

Maximum recommended cable length
100ft RG58A/U or equivalent. Use low capacitance coax for best results.

Pilot Level
Adjustable 7% to 11% mix

Frequency Stability
Pilot and subcarrier +/- 10ppm -50 to +80 deg C ambient

Stereo Generator Technology
Aphex patented Parallel Path Digital Modulation (PPDM)

Pilot Phase Error
0 degrees guaranteed by PPDM design

Spurious Output
better than -72dB above 55KHz, typical <-85dB

Subcarrier Rejection
better than -60dB

Harmonic Distortion
Better than 0.003% within stereo generator

Intermodulation Distortion
Better than 0.003% within stereo generator

Frequency Response
1Hz to 15KHz +/1 0.1dB, -3dB at 0.159Hz and 15.5KHz

12.4.7 Remote Control Interface

Configuration
RS232 standard serial port

Connector
DB-9 male

Control capability
All FM Pro functions and parameters

Connectability
Null modem cable to P.C. or modem cable to modem

Modem Compatibility: U.S. Robotics Sportster. For others check with factory.

12.4.8 Power

Line Voltage Input
77 to 266VAC 50-1000Hz automatic (no selection required)

Power Requirements
50 Watts

Safety Standards
CE, UL, CSA, VDE
.

12.4.9 Physical

Front Panel Size
Standard 2-RU (3.5” vertical) 19-inch panel

Chassis Depth
13.125 in. not including rear connectors

Weight
14 pounds net, 22 pounds shipping

12.4.10 Environmental

Operating Temperature Range
32-122 deg. F (0-50 deg C)

Humidity
0 to 95% RH, non-condensing

end

13.1 Introduction

A reasonable confidence check can be made using conventional test equipment by following the procedure given below. This will determine if the audio processing systems are operating in a nominal fashion but will not permit a full and detailed analysis.

Presently, we regret that we are unable to supply a comprehensive test and calibration procedure that could be performed in the field. A full and proper procedure requires numerous specially constructed test fixtures and other precision laboratory equipment. We are forced to advise you that, should you run into any doubt about the proper performance of your Model 2020, you must return the unit to the factory or one of our overseas factory authorized service centers.

We continuously put new technical information on the worldwide web at www.aphex.com/2020. Be sure to check it occasionally to discover if any pertinent information is available concerning the problems you may be experiencing.

13.2 Quick Confidence Check

13.2.1 Purpose

Some customers may wish to test their FM Pro to see if everything is working correctly. This is a fast and easy test to verify nominal operation of nearly all parameters. Each unit undergoes this confidence test at the factory just prior to shipping.

13.2.2 Test Procedure

You will need a pink noise generator equivalent to what is supplied in the Audio Precision System One analyzer. You do not need an analyzer, only the signal generator.

1. Set the pink noise output to +6dBu and feed only one channel of the model 2020.

2. Set up the 2020 as follows (unspecified parameters are not important):

Input: No filters, Input ref = +4dBu

Leveler: 2Sec, +15, -15, all other options OFF

Multiband: 300, 1200, 3400; Drive = 0dB; All releases at 10; All mixes at 0dB; all coupling OFF

Limiter: Master Drive = +6; everything else at 0

Pre-emp Limiter: 75uSec/Pre-De

3. The panel meters should now indicate as follows:

VU Meter = -9dB

Leveling = 0dB

Multiband GR: all = -9dB (can jump a step plus or minus)

Limiting: Flickering between 3 and 5dB

That's it. If you cannot accurately set your pink noise generator to +6dBu, then simply adjust its output level to get 0dB on the leveling meter and the other meters should all line up correctly. You can also adjust the model 2020's input ref level to move the leveling meter to 0dB if the output adjustment of your generator is insufficient. Switch input channels of the 2020 to verify both channels.

end

14.0 Installation Instructions

14.1 Unpacking

Your FM Pro was packed carefully at the factory in a container designed to protect the unit during shipment. Nevertheless, Aphex recommends making a careful inspection of the shipping carton and the contents for any signs of physical damage.

> If damage is evident, do not discard the container or packing material. Contact your carrier immediately to file a claim for damages. Customarily, the carrier requires you, the consignee, to make all damage claims. It will be helpful to retain the shipping documents and the waybill number.

14.2 Damage & Claims

14.3 Mains Voltage And Fuses

The FM Pro is built with a custom designed universal off-line switch mode power supply. There are no fuse changes or voltage taps to change for the primary input voltage. The power supply accepts any primary input voltage between 85 and 265VAC at 50 to 1000 Hertz. Fuses inside the chassis will normally fail only from a catastrophic failure of the power supply. Therefore, need for fuse replacement suggests a malfunctioning power supply requiring component level repairs. Should fuses fail, please proceed cautiously while investigating the failure. Extremely hazardous voltages appear on the pc board including the heat sinks. Observe all the printed cautions and refer servicing only to qualified personnel.

14.4 Power Cord

The Fm pro uses a standard IEC power cord set. The appropriate mains plug for each country is normally shipped with each unit. However, if you must install or replace the plug, use the correct wiring code as follows:

Power Cord Color Codes	
USA Color Code	IEC/Continental Color Code
Black = Hot (live)	Brown = Hot (live)
White = Neutral	Blue = Neutral
Green = Ground	Yellow/Green = Ground

14.5 Mounting In A Rack

The FM Pro occupies two standard 19 in. x 1 3/4 in. rack spaces (2RU). Chassis depth is 13.25 inches not including connectors. Allow at least 3.5 inches additional space in back for wiring and connectors. The chassis is designed to be fully supported by front panel mounting alone. To avoid cosmetic damage to the panel, use the cushioned rack screws provided in the shipping kit or other cushioned rack screws.

14.6 Proper Ventilation

The FM pro uses one cooling fan located on the right-hand side of the chassis. Unit ventilation passes through the chassis side walls, so no cooling space is required above or below the chassis. Please be sure there is adequate clearance at both sides of the chassis. This is normally not a problem since the Model 2020 was designed to be mounted in a typical rack which usually allows 2 inches or more of clearance on each side of the chassis. The fan has an attached filter cover which should be checked periodically for blockage. The filter element is a cellular elastomer which can simply be washed out in water and towel dried. The filter can be removed and replaced while the unit is operating but be careful not to poke objects into the fan blade.

14.7 Safety Considerations

To minimize the risk of shock or fire, do not expose the unit to moisture. Allow adequate ventilation for cooling. Do not open the chassis cover: there are no user serviceable parts inside.

Installation should be performed only by qualified individuals. It is the installer's responsibility to insure his personal safety and the safety of others in the work area. It is never a good idea to work alone in the vicinity of high power electrical and radio frequency equipment.

14.8 Analog Balanced I/O

14.8.1 Standard Wiring

The analog inputs and outputs are RFI protected and utilize industry standard 3-pin female XLR jacks. Connections are by the industry standard pinout as follows:

14.8.2 Main Input Wiring

For maximum RFI suppression, pin-1 is connected directly to chassis ground. To eliminate input ground loop hum, the balanced input stage ground references are coupled to pin 1 of the input jacks. For maximum RFI suppression and hum loop rejection using balanced lines, you should connect the shield only at the input connector of the FM Pro and let it float at the sending end of the line. Refer to Appendix A for detailed information.

For unbalanced use, tie pin-3 to pin 1 for the ground and connect pin-2 as "hot". Connect the shield to ground at both ends of the cable. Interfacing with unbalanced sources can sometimes be improved with a pseudo-balanced connection. For a complete tutorial on balanced and unbalanced interfacing to other equipment, please refer to Appendix A of this manual.

Whether using balanced or unbalanced wiring, be sure to follow the same pin connection scheme for both channels of the input wiring to avoid audio phasing problems.

14.8.3 Main Output Wiring

The balanced output impedance of 65 ohms is optimized for driving long cables and consequently an FM Pro can drive just about any kind of line, balanced or unbalanced, of any length. Unique servo balanced output circuitry automatically maintains the proper gain and level into a balanced or unbalanced output line.

For best ground loop rejection and RFI suppression using balanced lines, do not connect the shield to the output ground, rather connect it only to the ground (pin-1) of the receiving end. For unbalanced use, tie pin-3 to pin 1 for the ground and connect pin-2 as "hot". Connect the shield to ground at both ends of the cable.

Just as with the input wiring, unbalanced outputs can sometimes be improved using a pseudo-balanced connection. For a complete tutorial on balanced and unbalanced interfacing to other equipment, please refer to Appendix A of this manual.

Main Analog I/O Connections	
Female (Input) Pin-1 = GND Pin-2 = Positive Pin-3 = Negative	Male (Output) Pin-1 = GND Pin-2 = Positive Pin-3 = Negative

14.8.4 Insert Loop I/O Connections

The insert loop utilizes 3-pin XLR jacks. It is expected that equipment connected to this loop will be located within a very short distance of the FM Pro, probably mounted only a few rack spaces away, so the insert loop I/O operates in the unbalanced mode.

Insert Loop I/O Connectors	
Female (Input) Pin-1 = GND Pin-2 = Hot Pin-3 = GND	Male (Output) Pin-1 = GND Pin-2 = Hot Pin-3 = GND

14.9 Digital Audio I/O

The digital audio I/O utilizes standard 3-pin XLR connectors as specified by the professional AES/EBU standards. Complete details about connecting and using the digital audio interface are available in section 9, “Using Digital Audio”, of this manual.

AES/EBU Connector Wiring	
Female (Input) Pin-1 = GND Pin-2 = Positive Pin-3 = Negative	Male (Output) Pin-1 = GND Pin-2 = Positive Pin-3 = Negative

14.10 Multiplex Output

The BNC output connector is chassis isolated to reduce the possibility of ground loop output hum. The output ground does have a d.c. ground path to the chassis, but it flows by way of the multiplex output amplifier’s ground reference point to eliminate any hum loops that may intercede from the chassis or rack frame. The BNC connector is directly RF-coupled to the chassis to facilitate effective RFI suppression of RF signals which may enter the FM Pro through the BNC jack and cable. For best results, use only top quality BNC cables and connectors. Loose or intermittent connectors may cause noise and instability of your FM Exciter.

14.11 RS-232 Connector

This DB9 connector is for use with remote control as described in section 11 on Remote Operation. The pins of the connector are RFI filtered, but it is best to use shielded cables to reduce the possibility of interference entering the FM Pro if operating in a strong RF field.

14.9 Summary

You should have no trouble installing the FM Pro. If any difficulties are experienced, other information contained in this manual will probably supply adequate assistance. Please study this manual before contacting the factory for assistance.

end.

15.0 Schematics

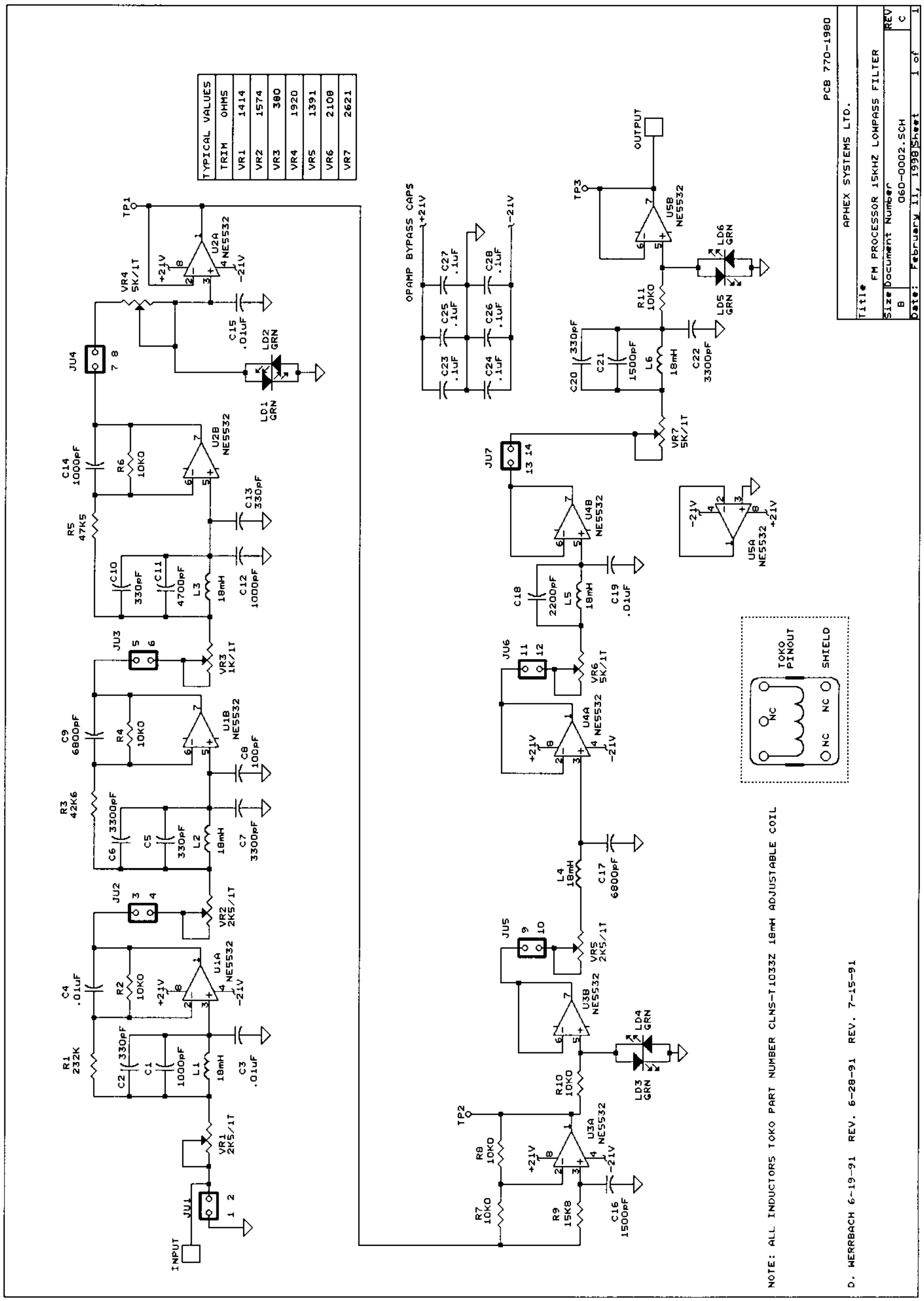
15.1 Advisory

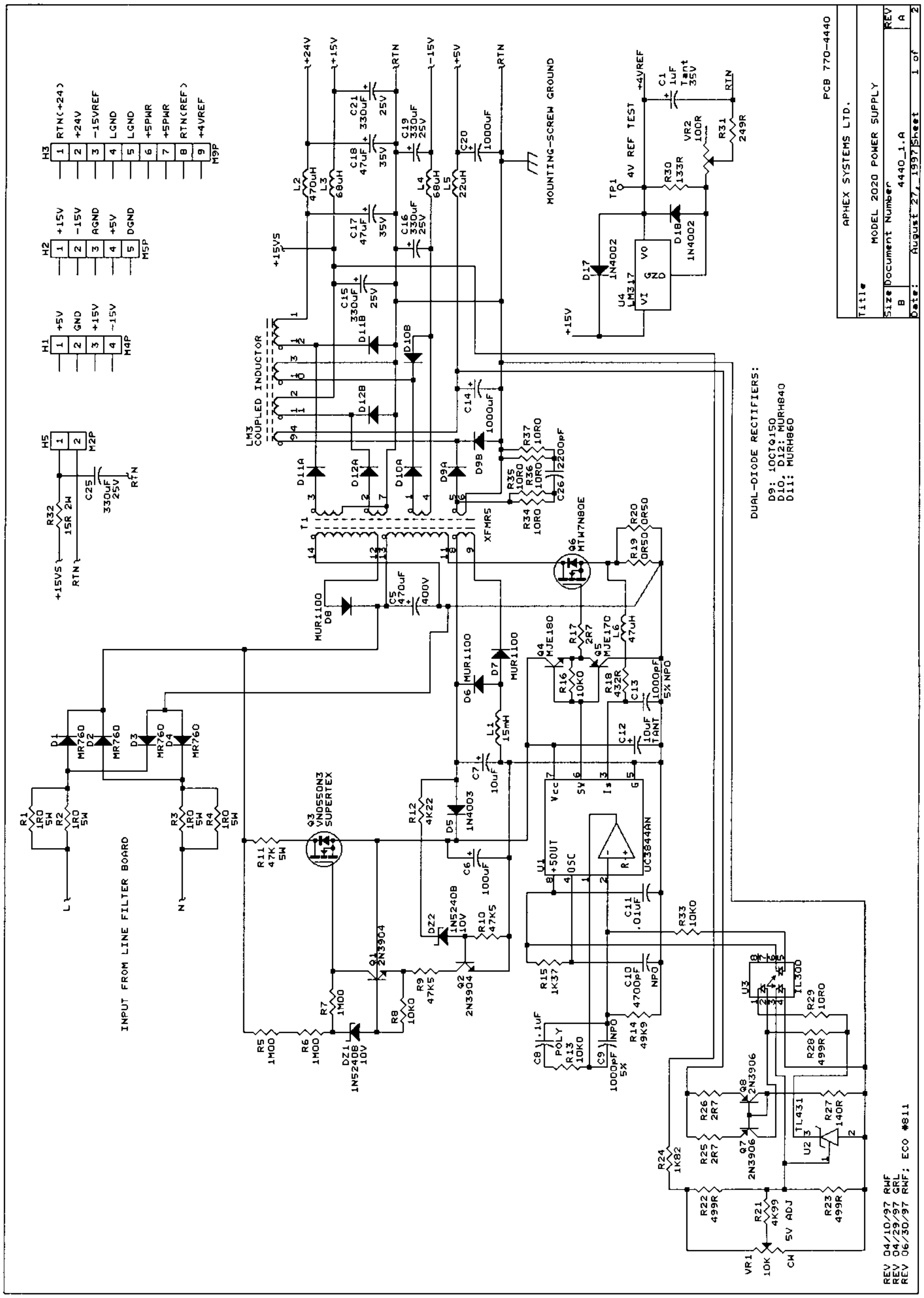
The diagrams included in this manual are intended principally as a resource for competent maintenance personnel. Many of the circuits are patented and are therefore protected under patent laws. Aphex Systems reserves all patent rights. It shall not be construed that publishing of schematics in this manual grants or implies permission for use in any way.

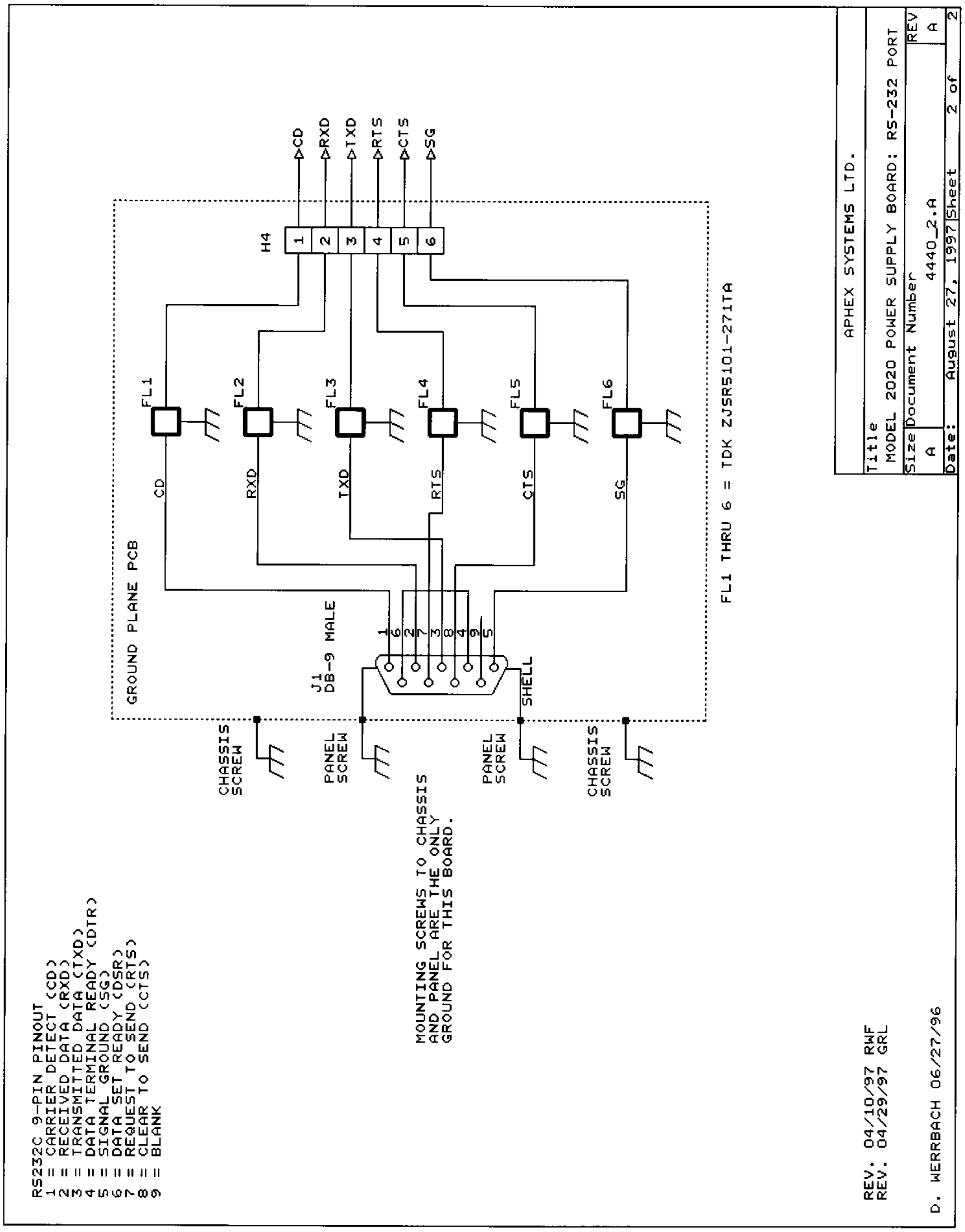
All schematic diagrams included in this manual are believed to be essentially accurate, although they may not reflect the running upgrades and modifications that have occurred since they were drawn. The Model 2020 unit shipped with this manual may contain updates not shown by schematics of the manual. It is our belief that any such variances will be obvious to trained maintenance personnel upon examination, and that such variances will cause little difficulty in the successful repair of the unit.

It is not our intention to provide assistance to entrepreneurs who may want to experiment with or modify the product in any way. The schematics are not warranted for any such use. All 1ability rests with the entrepreneur in such as case.

15.2 The Schematics ⟶


TYPICAL VALUES
TRIM OHMS
VR1 1414
VR2 1574
VR3 380
VR4 1920
VR5 1391
VR6 2108
VR7 2621
OPAMP BYPASS CAPS
INPUT
OUTPUT
TP1
TP2
TP3
TOKO PINOUT
SHIELD
PCB 770-1980
APHEX SYSTEMS LTD.
Title FM PROCESSOR 15KHZ LOWPASS FILTER
Size B Document Number 060-0002.SCH REV C
Date: February 11, 1998 Sheet 1 of 1
NOTE: ALL INDUCTORS TOKO PART NUMBER CLNS-T1033Z 18mH ADJUSTABLE COIL
D. WERRBACH 6-19-91 REV. 6-28-91 REV. 7-15-91


APHEX SYSTEMS LTD.
PCB 770-4440
Title
MODEL 2020 POWER SUPPLY
Size B
Document Number 4440_1.A
REV A
Date: August 27, 1997
Sheet 1 of 2
DUAL-DIODE RECTIFIERS:
D9: 10CTQ150
D10, D12: MURH840
D11: MURH860
INPUT FROM LINE FILTER BOARD
MOUNTING-SCREW GROUND
COUPLED INDUCTOR
4V REF TEST
5V ADJ
REV 04/10/97 RWF
REV 04/29/97 GRL
REV 06/30/97 RWF; ECO #811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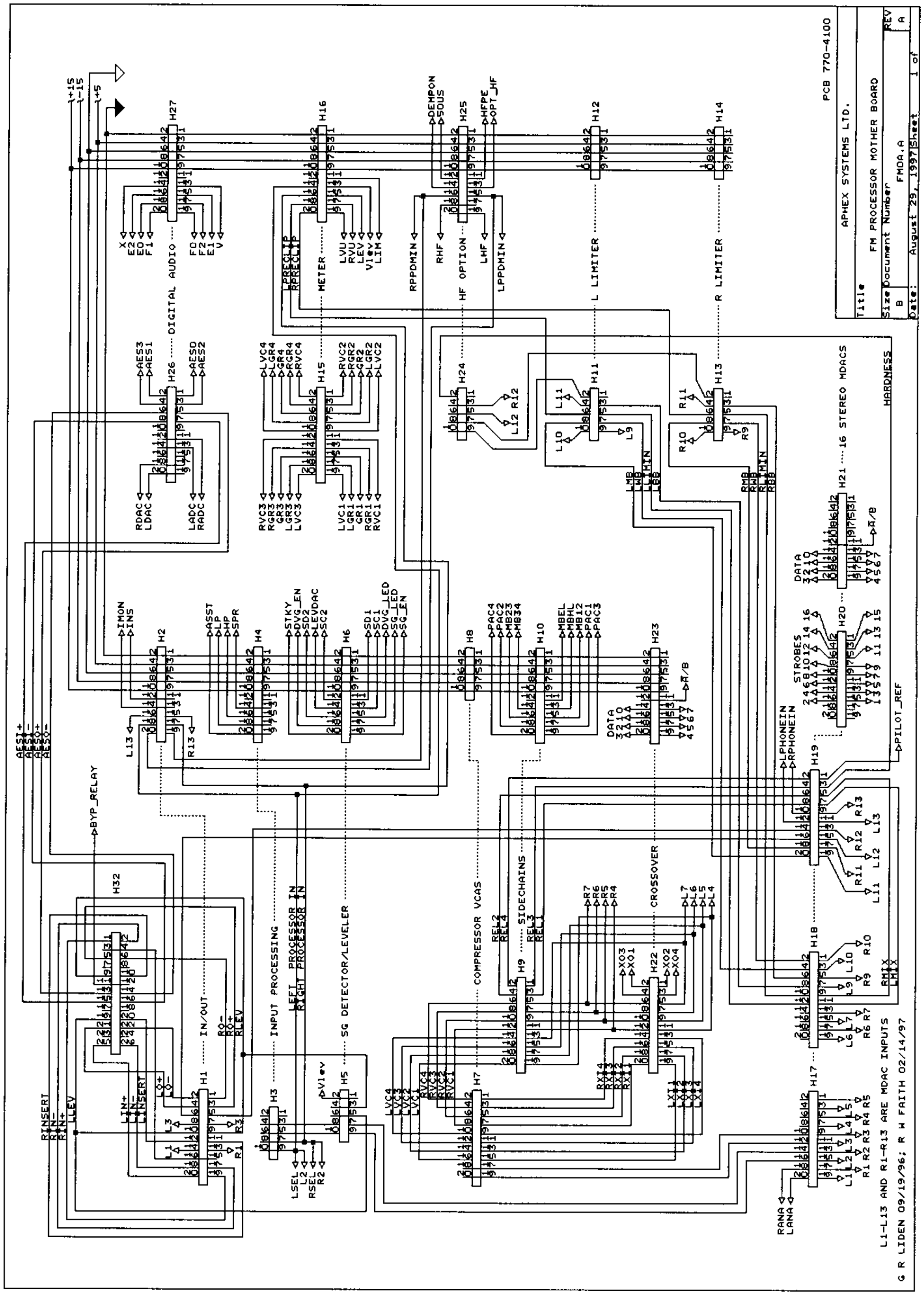
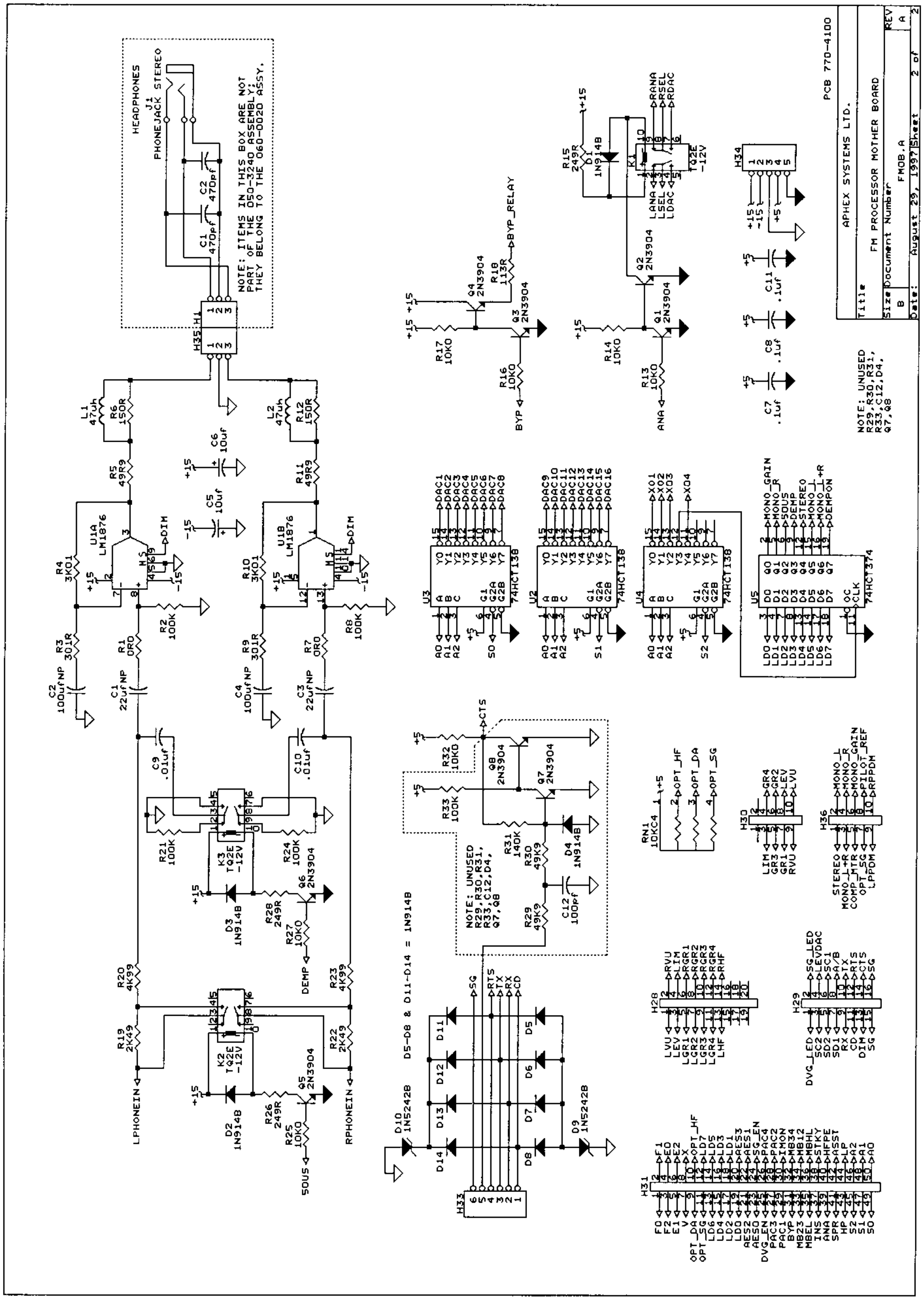
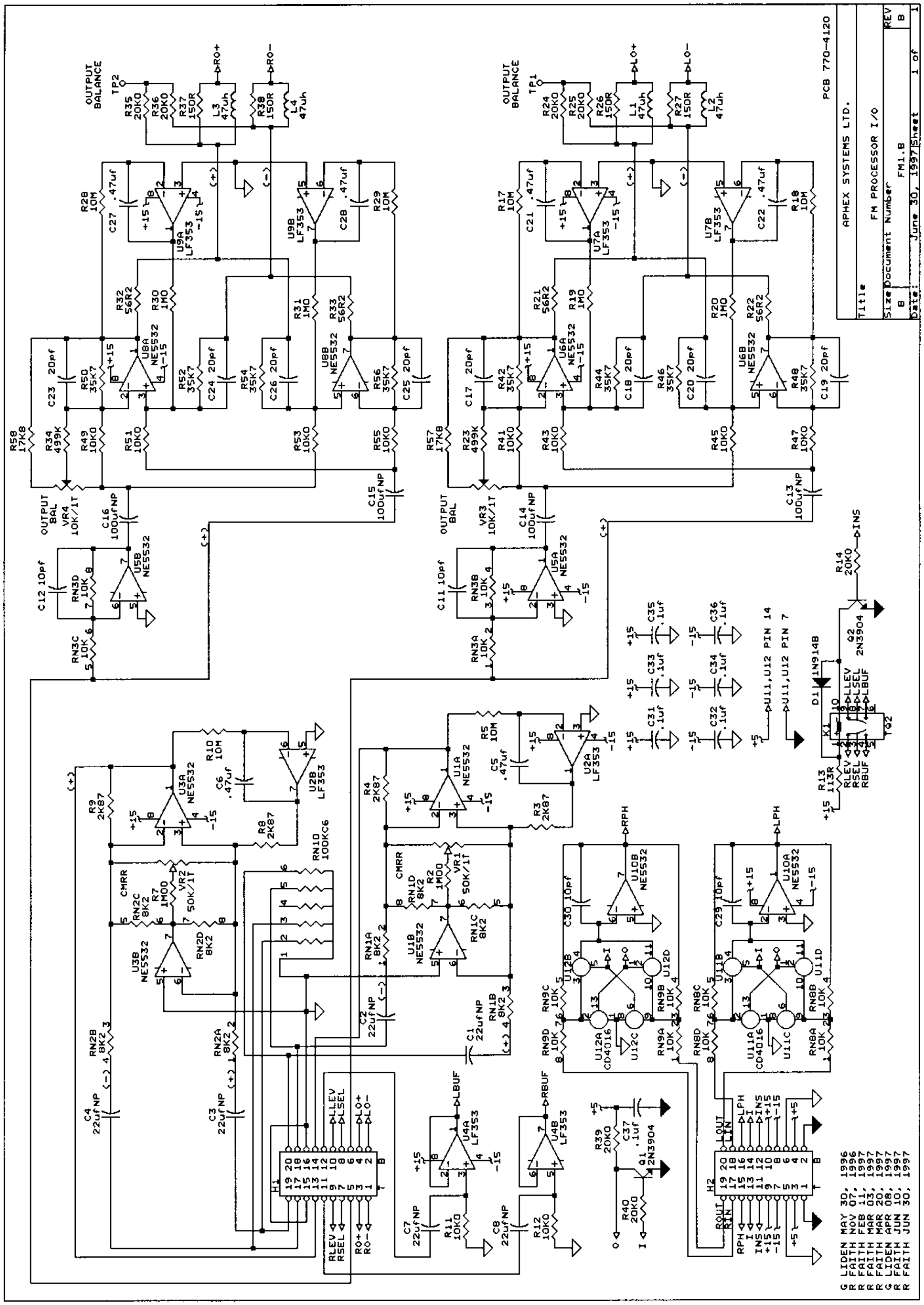
Title
MODEL 2020 POWER SUPPLY BOARD: RS-232 PORT
Size
A
Document Number
4440_2.A
REV
A
Date:
August 27, 1997
Sheet
2 of 2
CD
RXD
TXD
RTS
CTS
SG
H4
FL1
FL2
FL3
FL4
FL5
FL6
FL1 THRU 6 = TDK ZJSR5101-271TA
GROUND PLANE PCB
J1
DB-9 MALE
SHELL
CHASSIS SCREW
PANEL SCREW
PANEL SCREW
CHASSIS SCREW
MOUNTING SCREWS TO CHASSIS AND PANEL ARE THE ONLY GROUND FOR THIS BOARD.
RS232C 9-PIN PINOUT
1 = CARRIER DETECT (CD)
2 = RECEIVED DATA (RXD)
3 = TRANSMITTED DATA (TXD)
4 = DATA TERMINAL READY (DTR)
5 = SIGNAL GROUND (SG)
6 = DATA SET READY (DSR)
7 = REQUEST TO SEND (RTS)
8 = CLEAR TO SEND (CTS)
9 = BLANK
REV. 04/10/97 RWF
REV. 04/29/97 GRL
D. WERRBACH 06/27/96


PCB 770-4100
APHEX SYSTEMS LTD.
FM PROCESSOR MOTHER BOARD
FMOA.A
August 29, 1997
L1-L13 AND R1-R13 ARE MDAC INPUTS
G R LIDEN 09/19/96; R W FAITH 02/14/97


APHEX SYSTEMS LTD.
FM PROCESSOR MOTHER BOARD
PCB 770-4100
FMOB.A
August 29, 1997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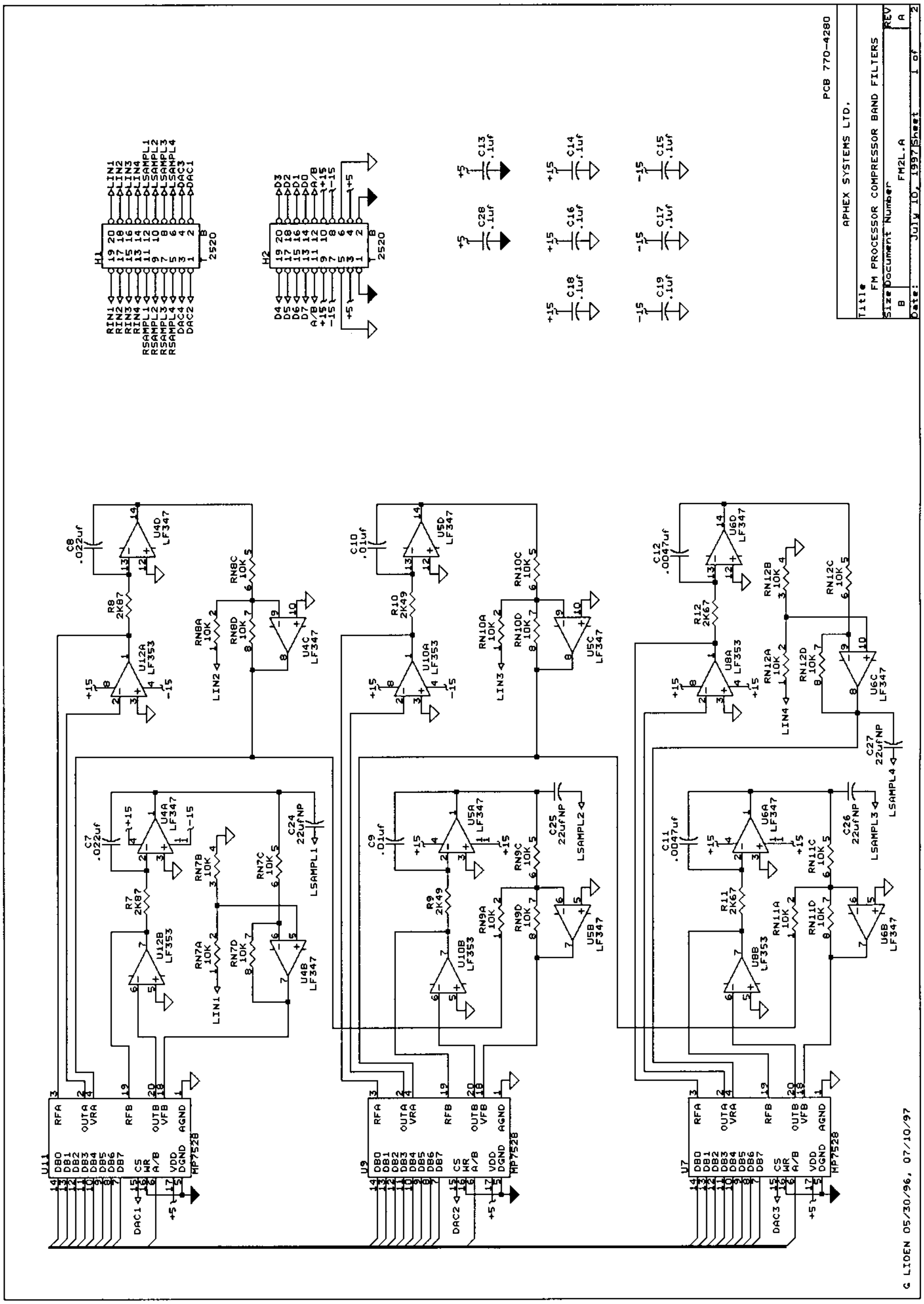
APHEX SYSTEMS LTD.
FM PROCESSOR I/O
FM1.B
June 30, 1997


PCB 770-4280
APHEX SYSTEMS LTD.
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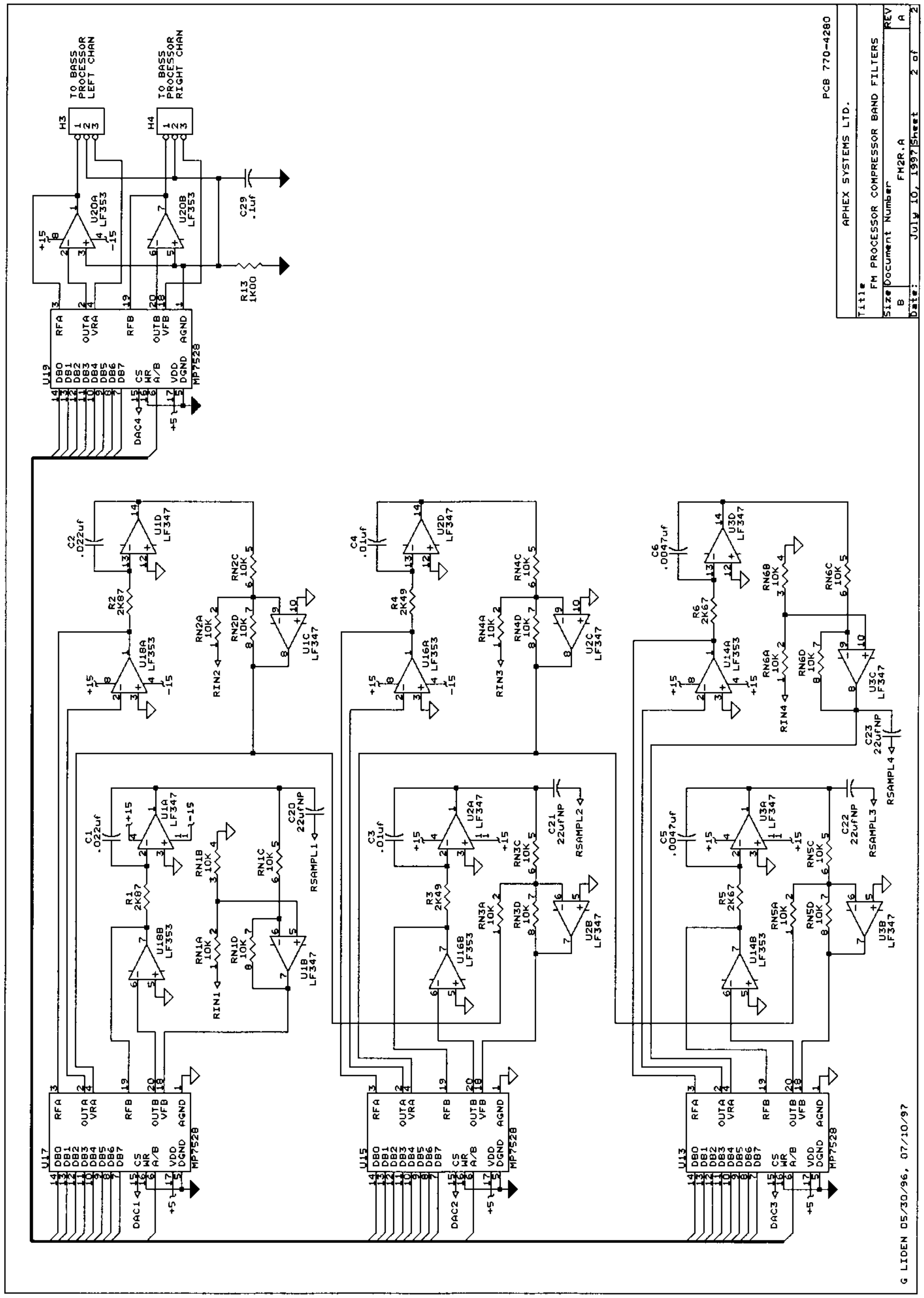
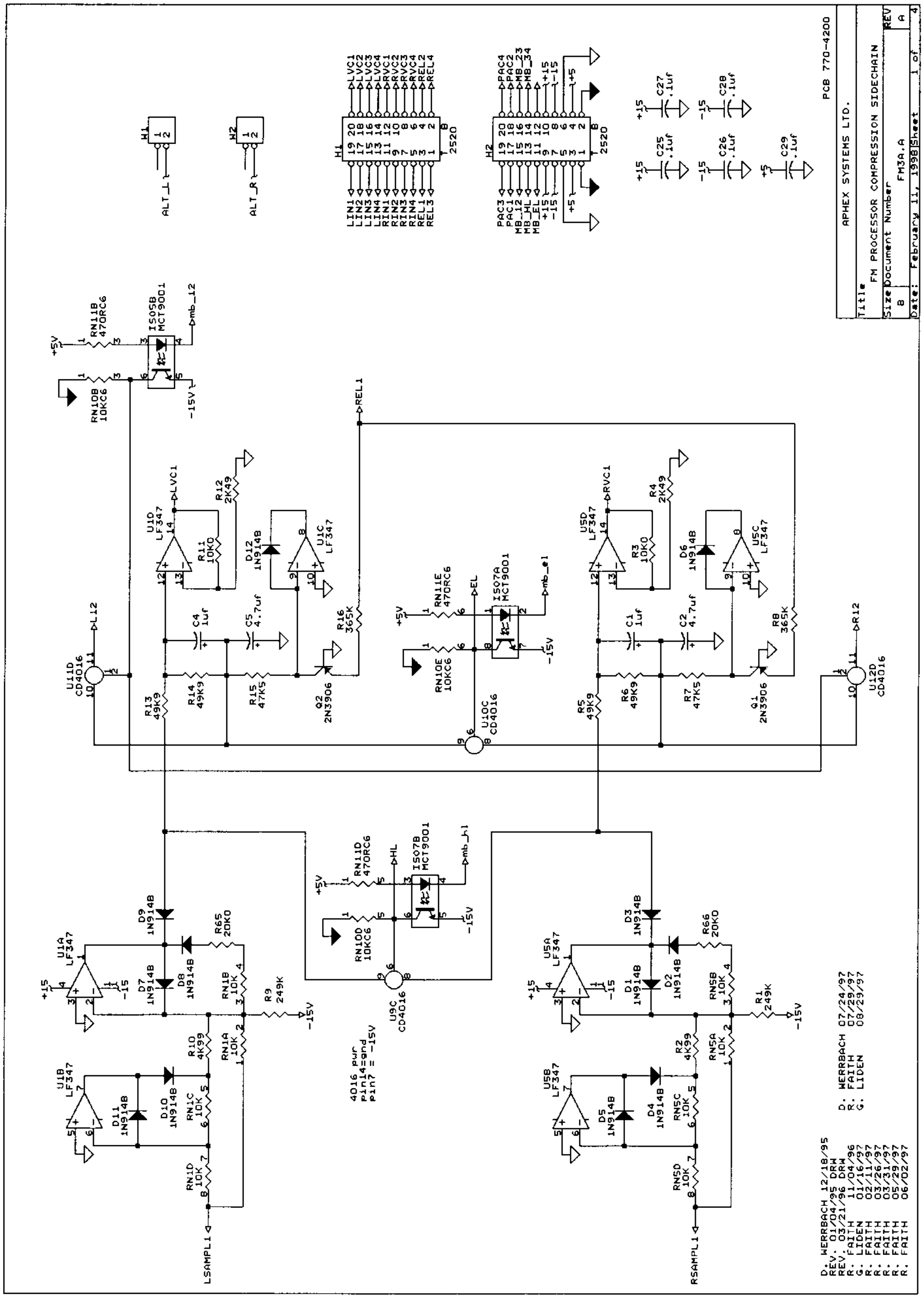
Size Document Number REV
B FM2L.A A
Date: July 10, 1997 Sheet 1 of 2
G LIDEN 05/30/96, 07/10/97


APHEX SYSTEMS LTD.
PCB 770-4280
Title
FM PROCESSOR COMPRESSOR BAND FILTERS
Size Document Number REV
B FM2R.A A
Date: July 10, 1997 Sheet 2 of 2
G LIDEN 05/30/96, 07/10/97
TO BASS PROCESSOR LEFT CHAN
TO BASS PROCESSOR RIGHT CHAN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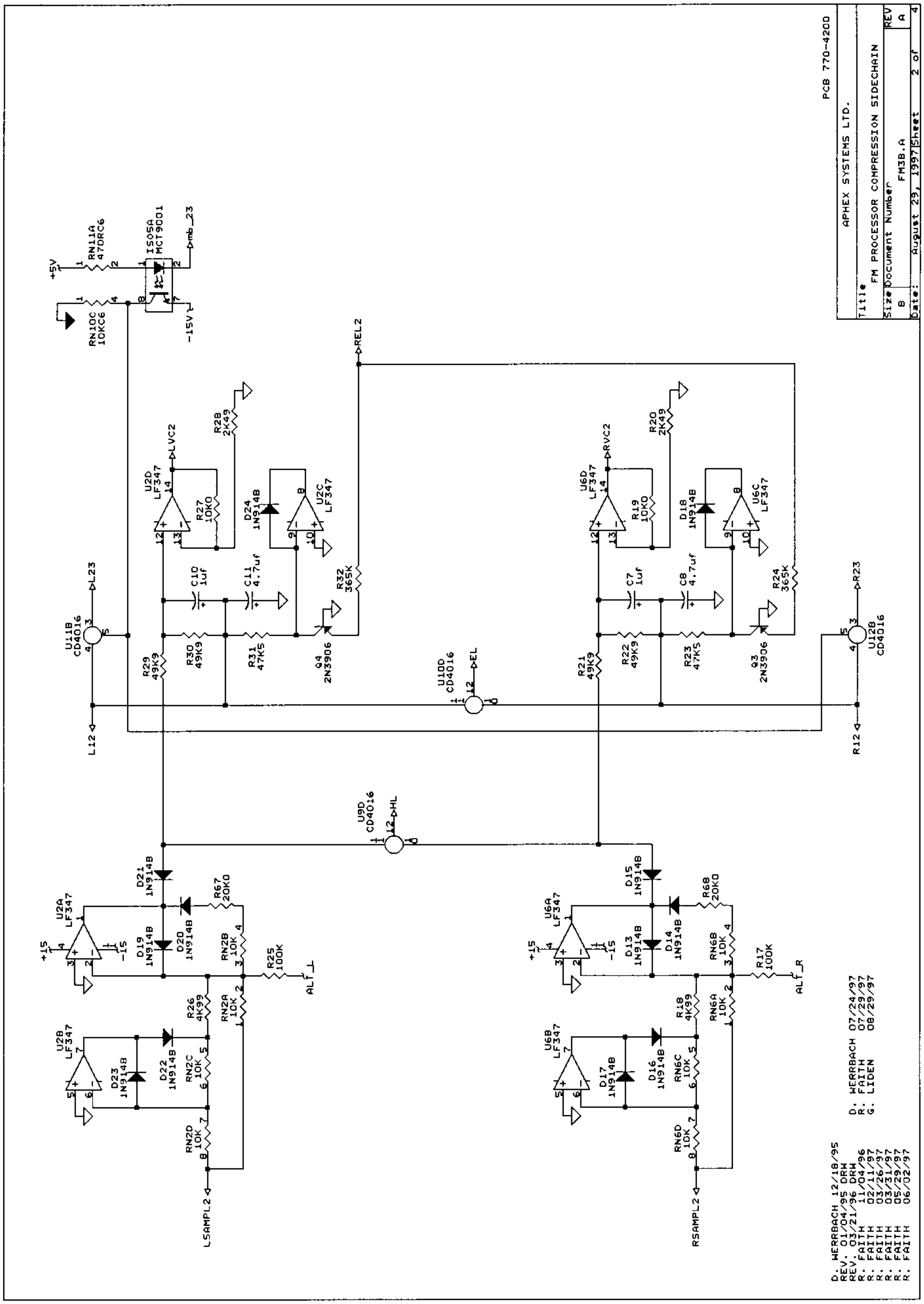
PCB 770-4200
Title
FM PROCESSOR COMPRESSION SIDECHAIN
Size Document Number
B
FM3A.A
REV
A
Date: February 11, 1998 Sheet 1 of 4



PCB 770-4200
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Title
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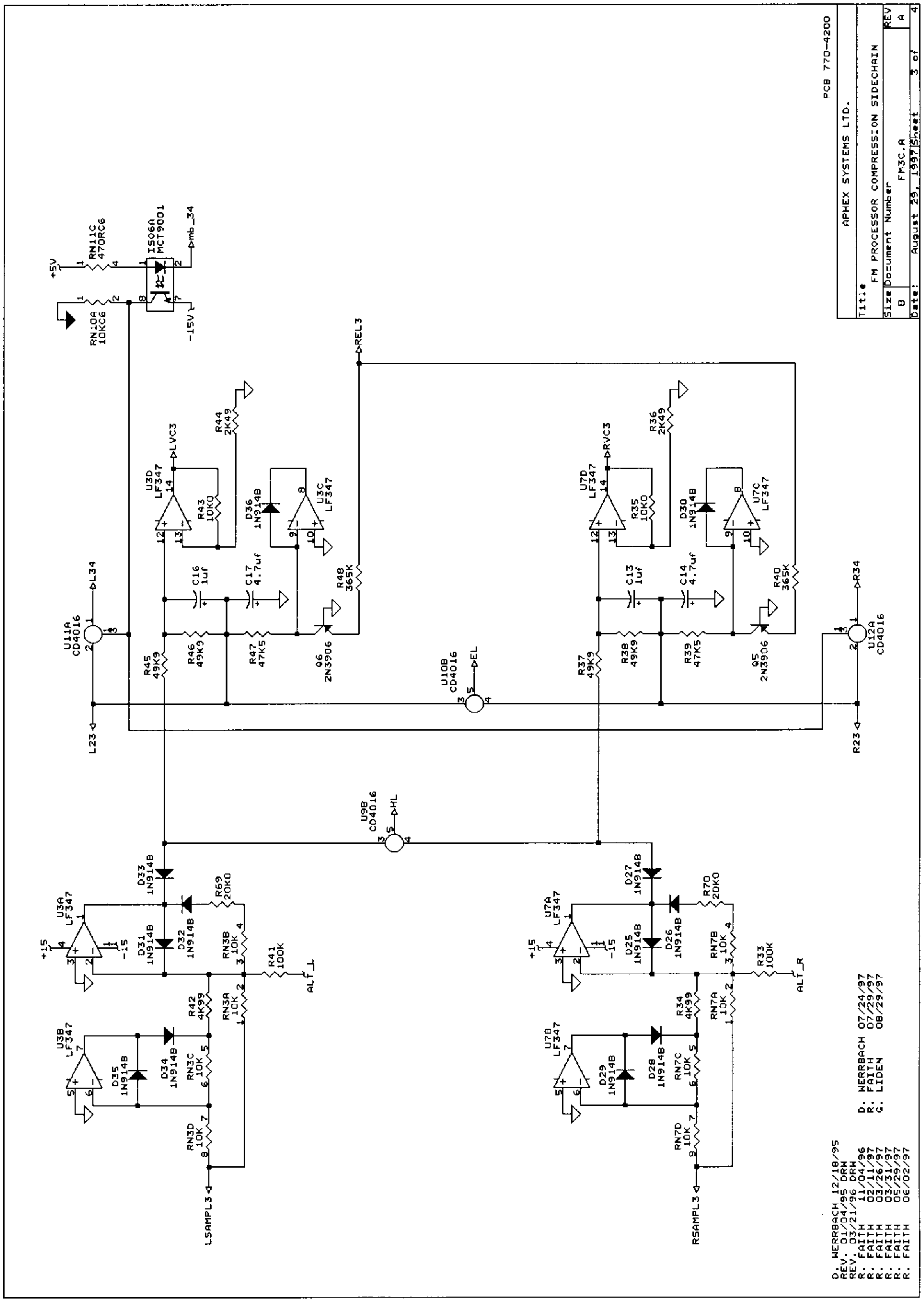
Size B
Document Number FM3B.A
REV A
Date: August 29, 1997
Sheet 2 of 4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FM PROCESSOR COMPRESSION SIDECHAIN
Size Document Number REV
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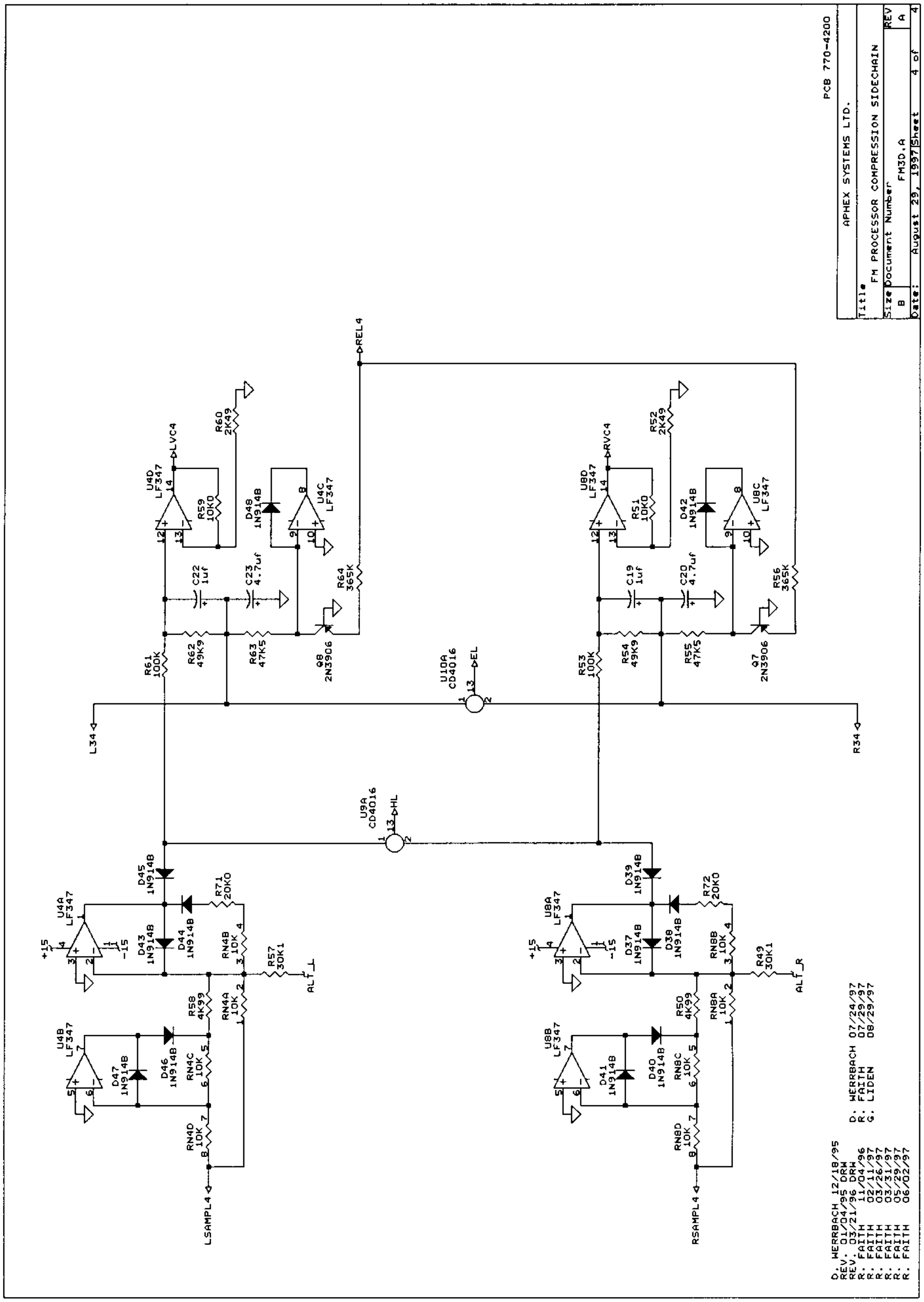
D. WERRBACH 12/18/95
D. WERRBACH 07/24/97
R. FAITH 07/29/97
G. LIDEN 08/29/97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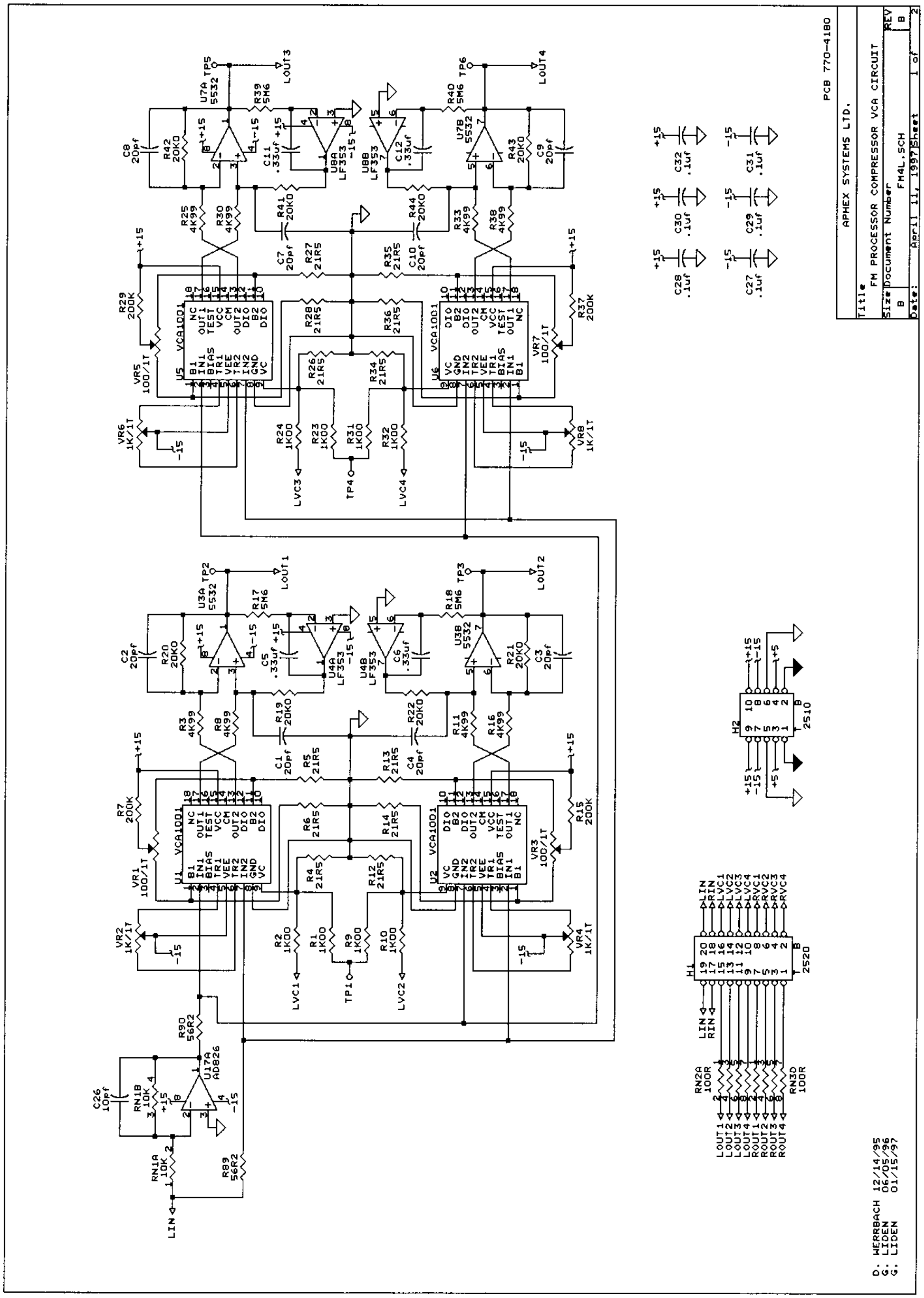
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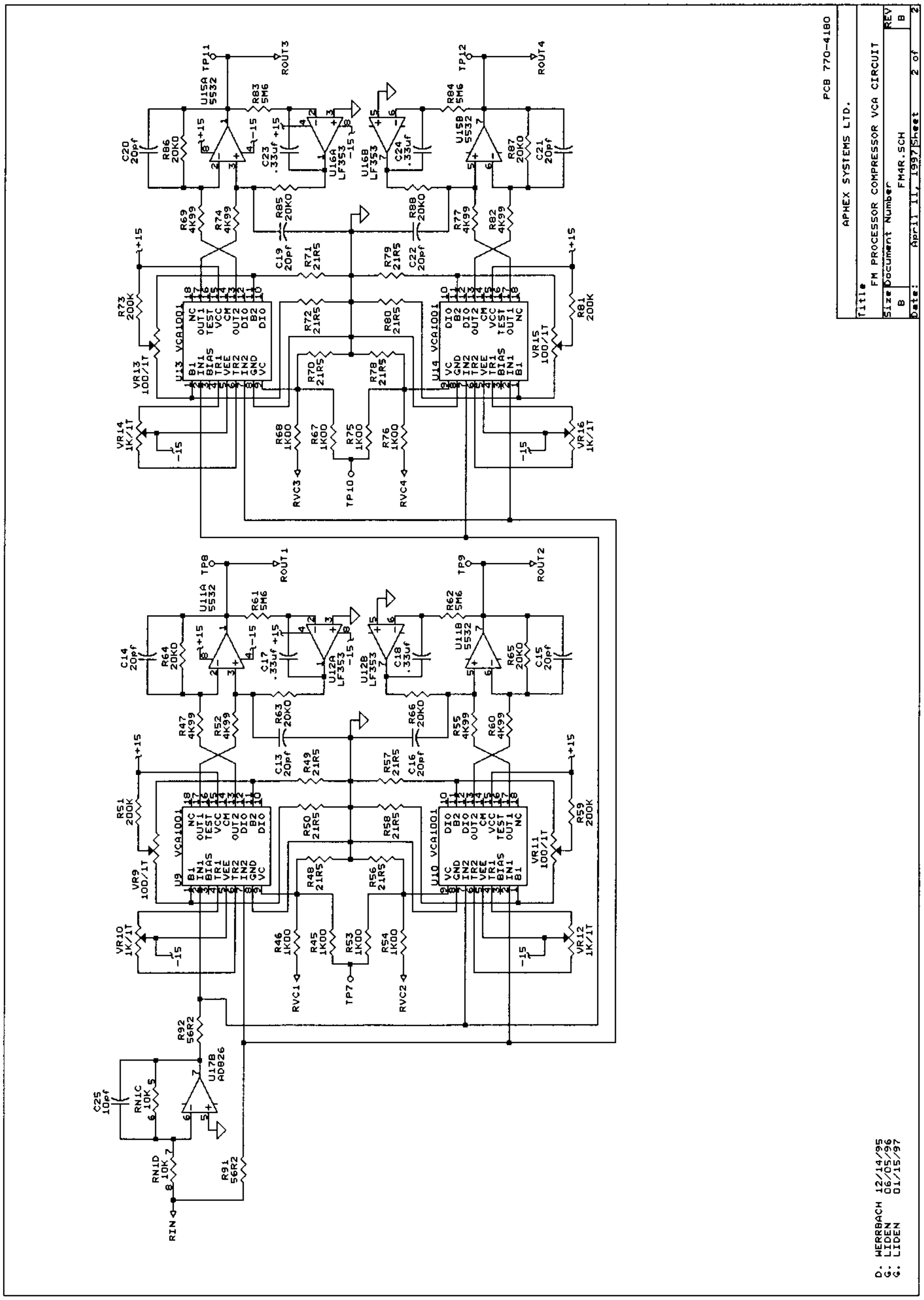
D. WERRBACH 12/14/95
G. LIDEN 06/05/96
G. LIDEN 01/15/97



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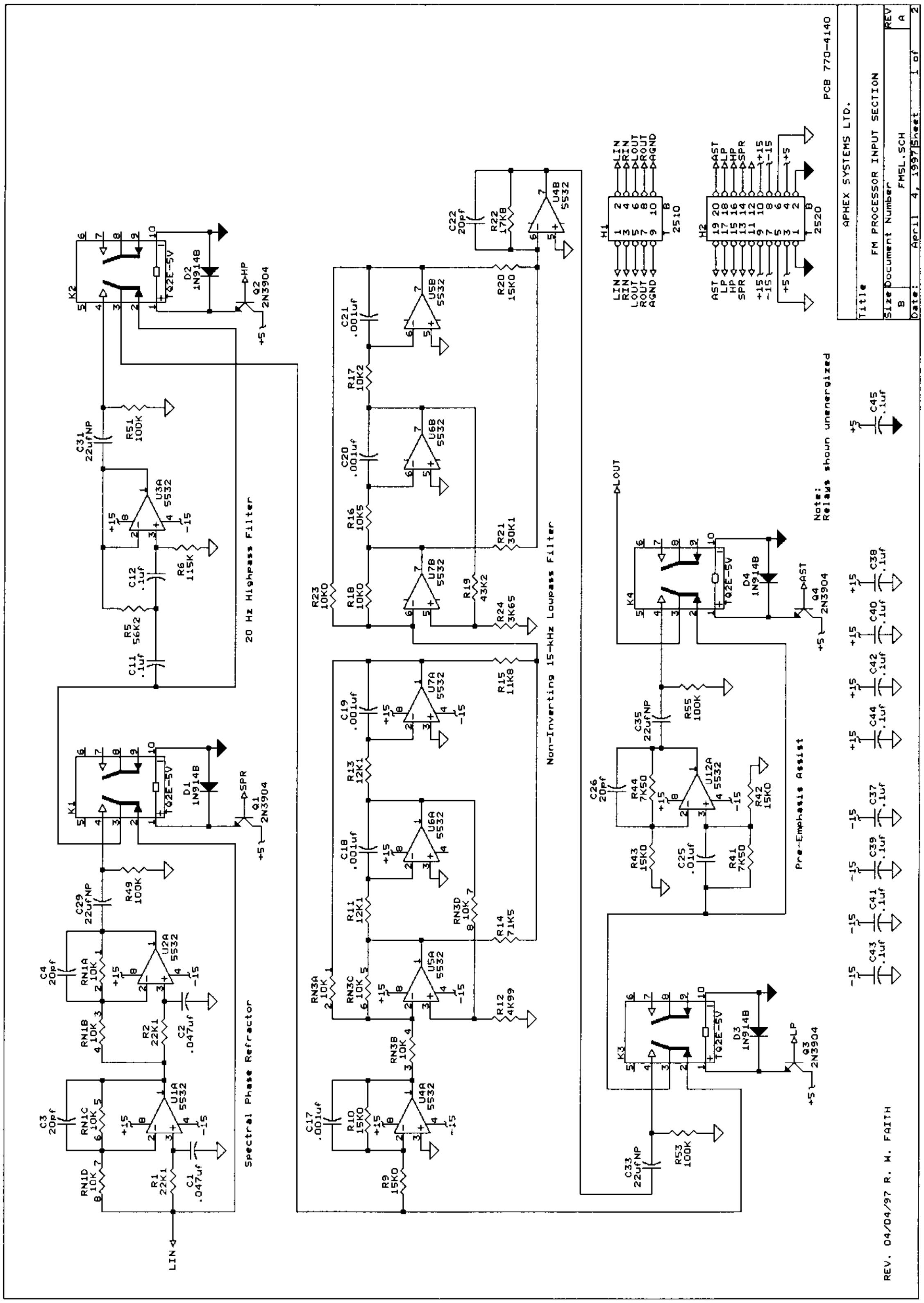
C: LIDEN 06/05/96
C: LIDEN 01/15/97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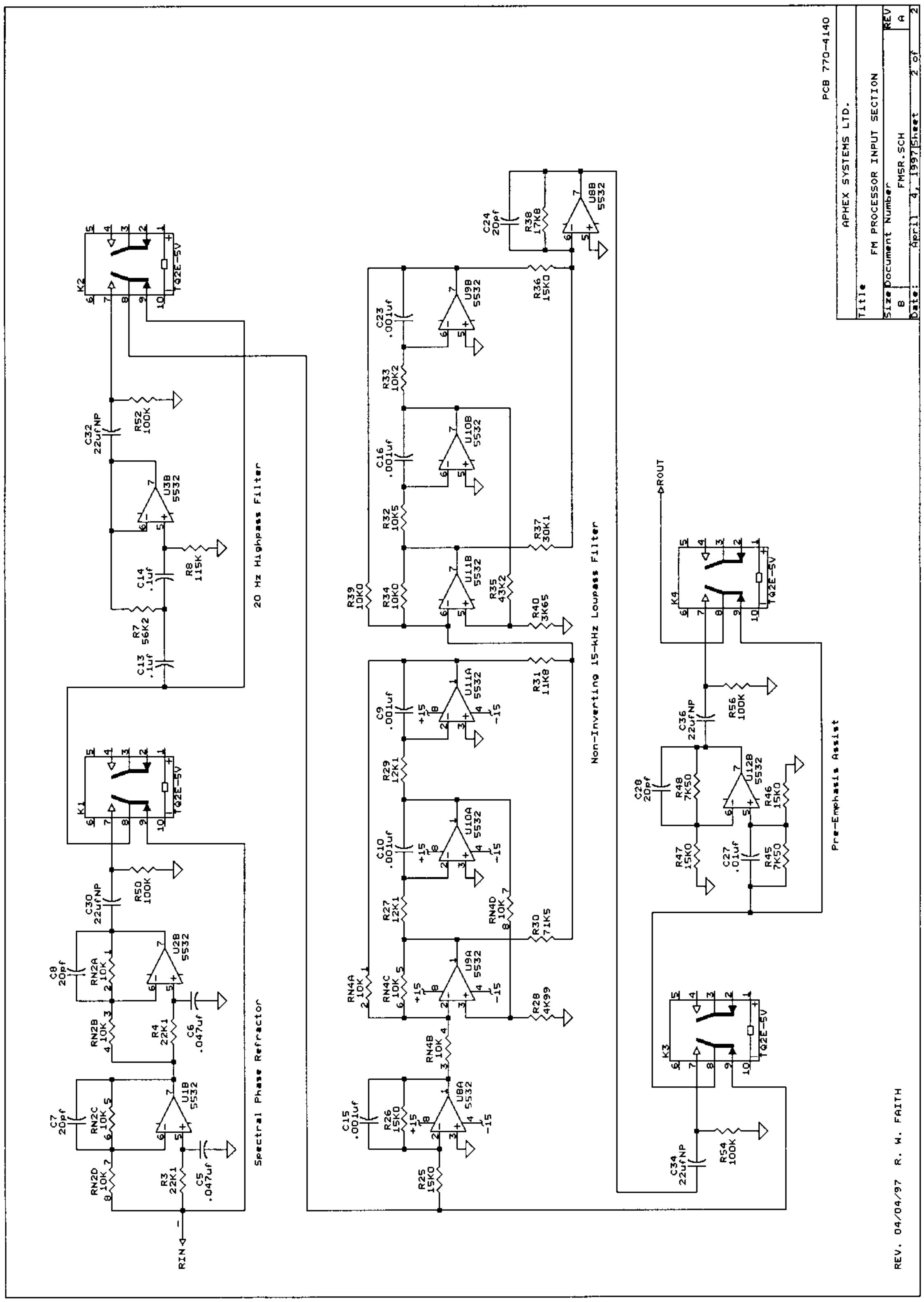
FM PROCESSOR INPUT SECTION
FMSL.SCH
April 4, 1997
Spectral Phase Refractor
20 Hz Highpass Filter
Non-Inverting 15-kHz Lowpass Filter
Pre-Emphasis Assist
Note:
Relays shown unenergized
REV. 04/04/97 R. W. FAITH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Title
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Size Document Number
B FMSR.SCH
REV A
Date: April 4, 1997 Sheet 2 of 2
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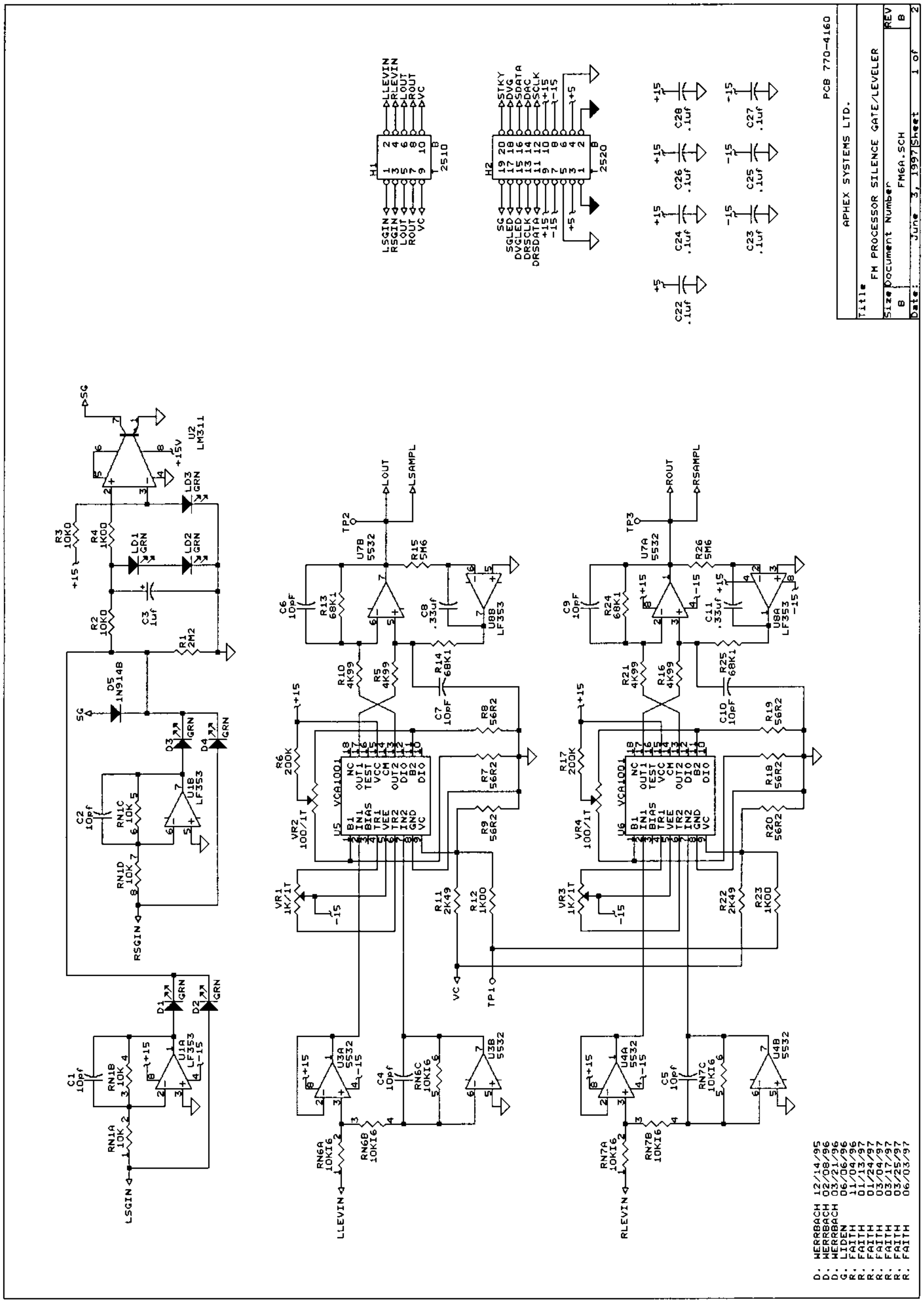
20 Hz Highpass Filter
Non-Inverting 15-kHz Lowpass Filter
Pre-Emphasis Assist
RIN
ROUT
REV. 04/04/97 R. W. FAITH


PCB 770-4160
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Title
FM PROCESSOR SILENCE GATE/LEVELER
Size Document Number REV
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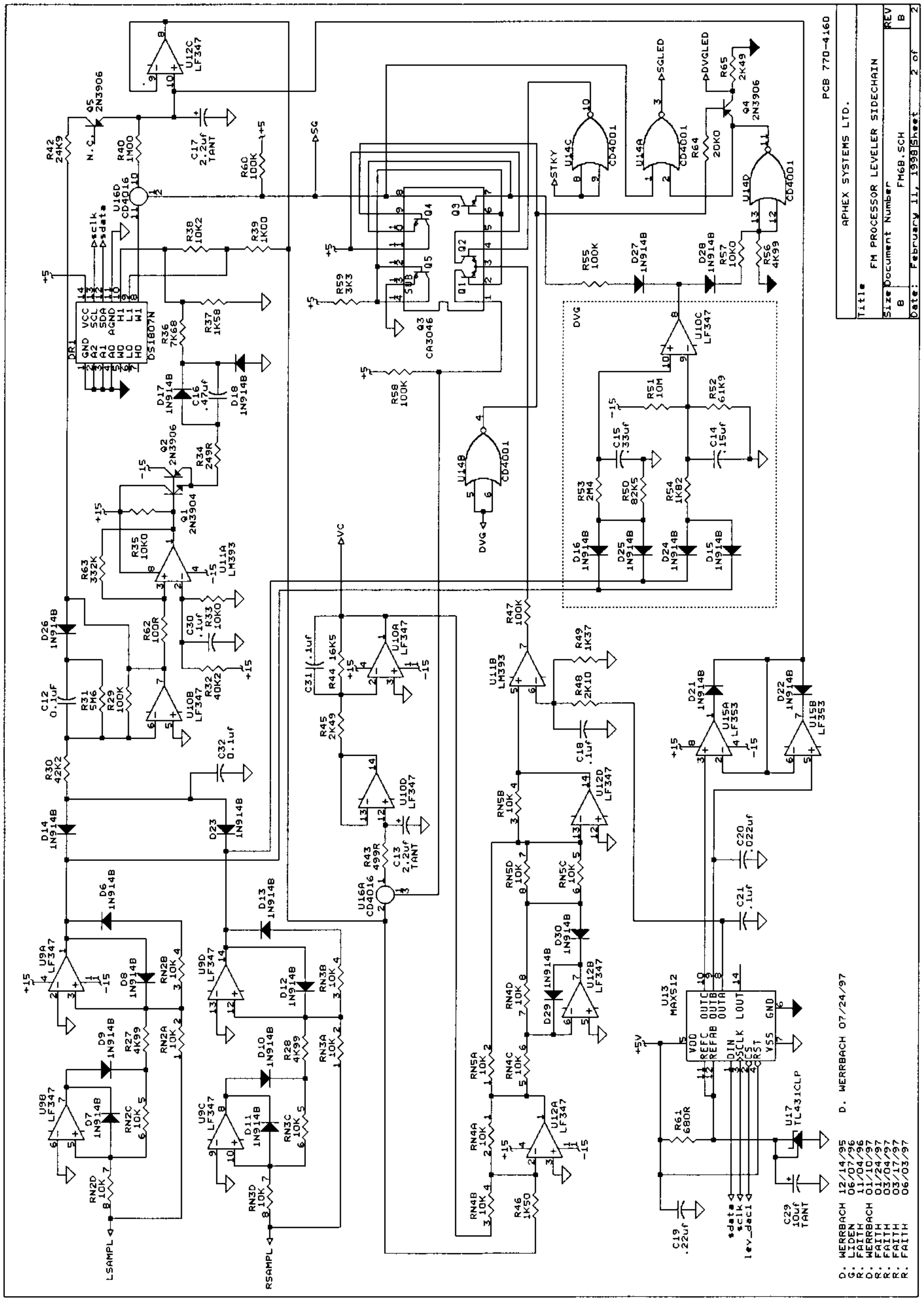

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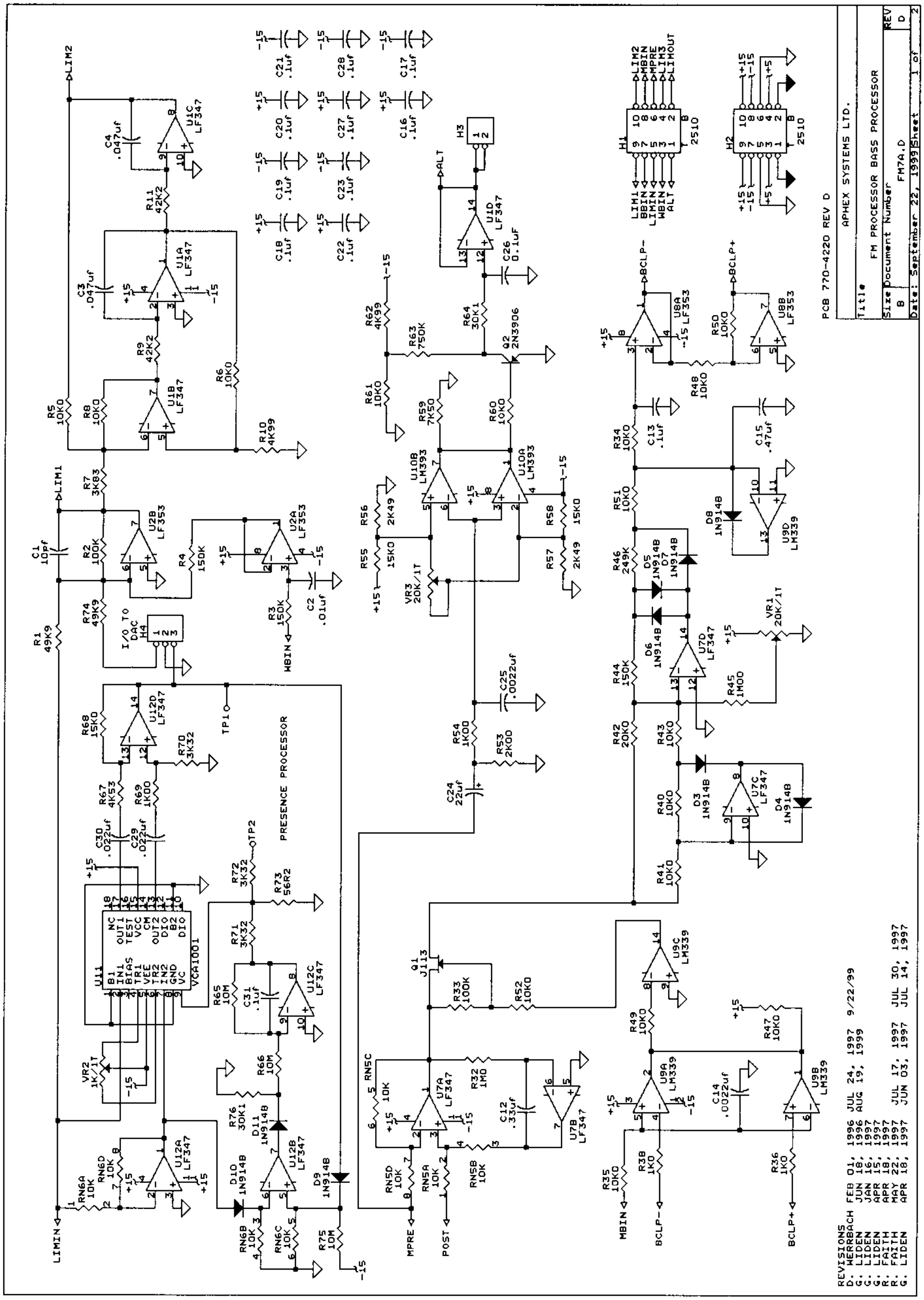
FM PROCESSOR LEVELER SIDECHAIN
PCB 770-4160
FM6B.SCH
February 11, 1998
D. WERRBACH 07/24/97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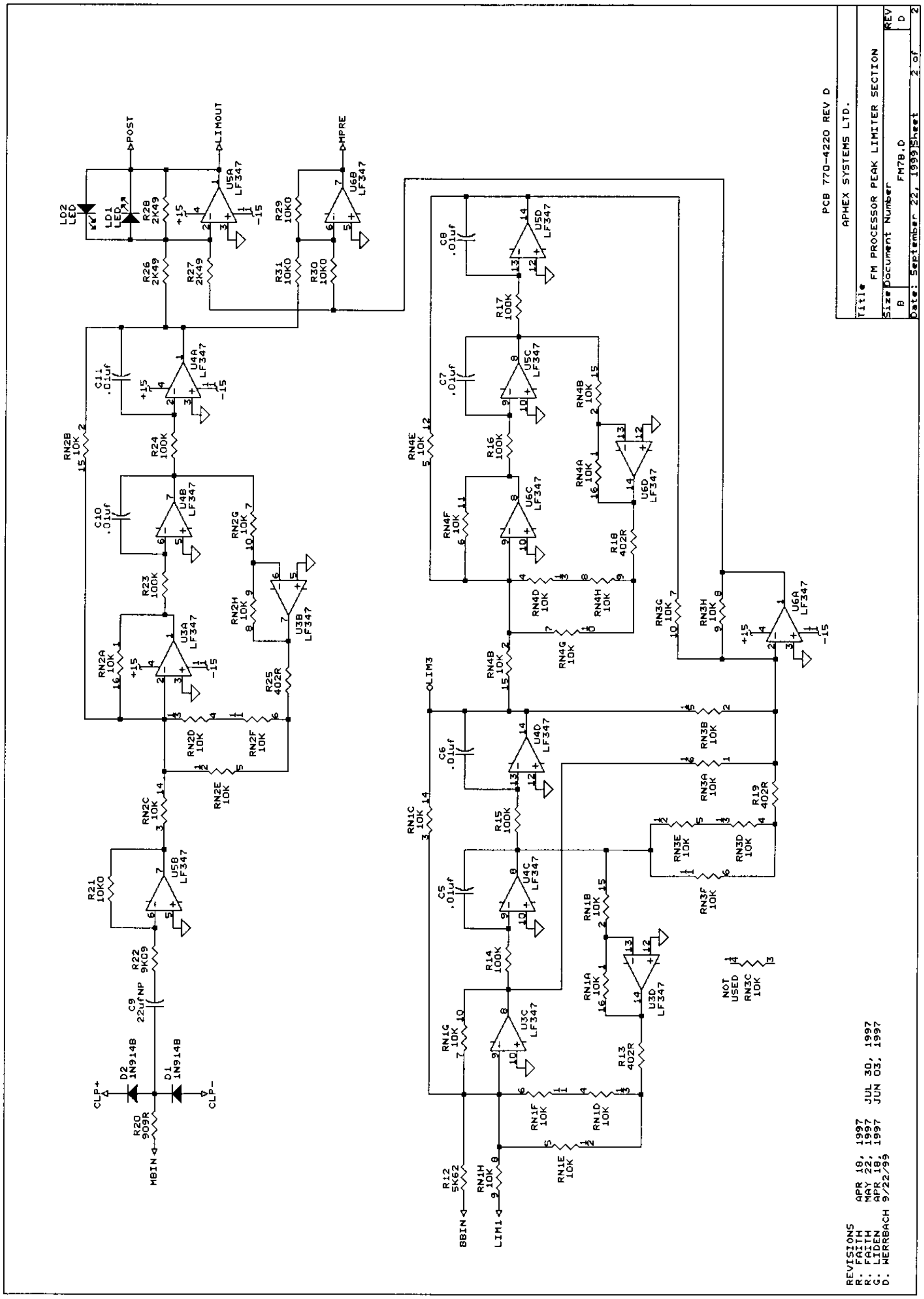
FM PROCESSOR BASS PROCESSOR
FM74.D
PRESENCE PROCESSOR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Size B
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REV D
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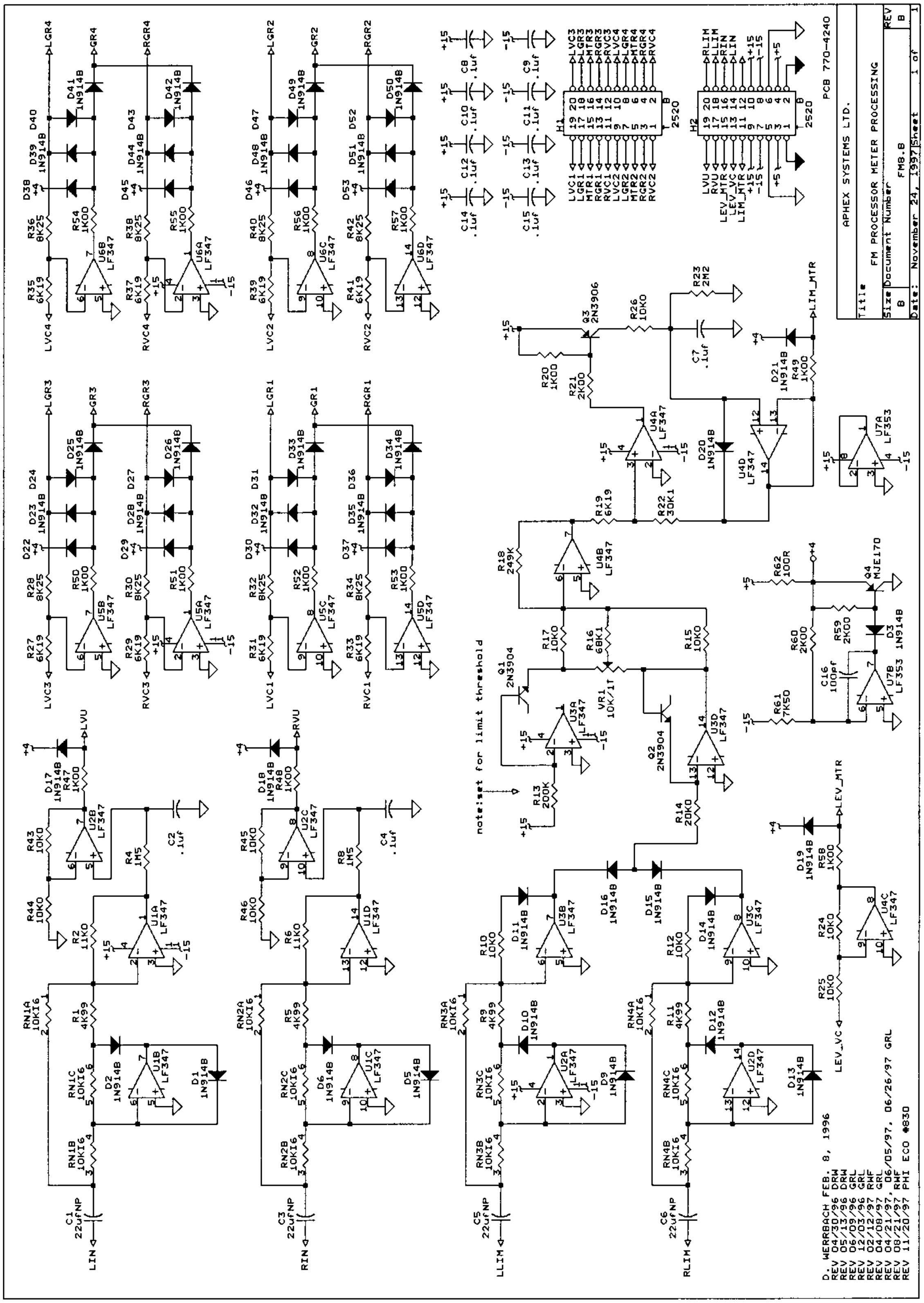
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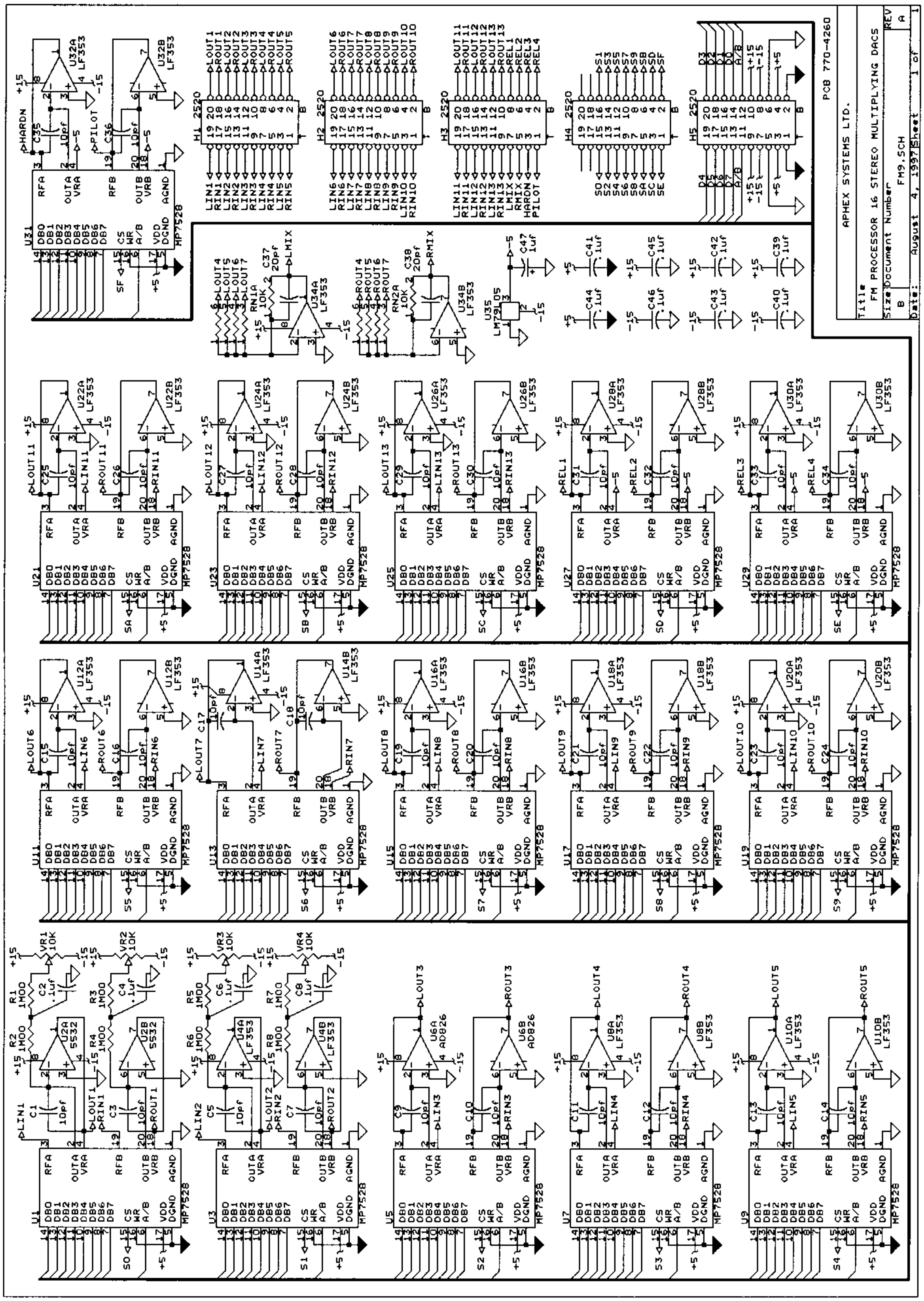
FM8.B
November 24, 1997
PCB 770-4240
note:set for limit threshold
D. WERRBACH FEB. 8, 1996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REV
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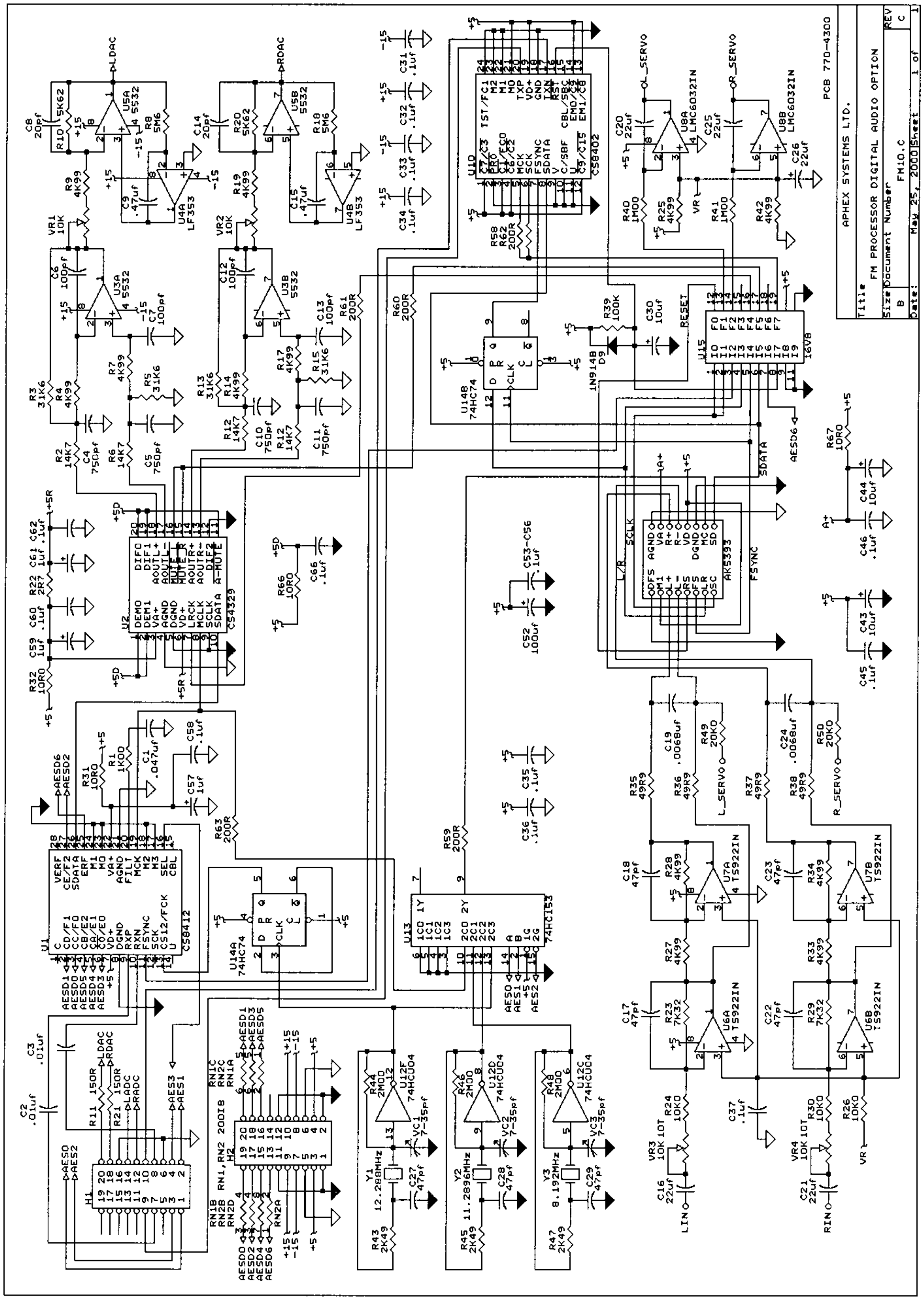
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PCB 770-4300
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Size
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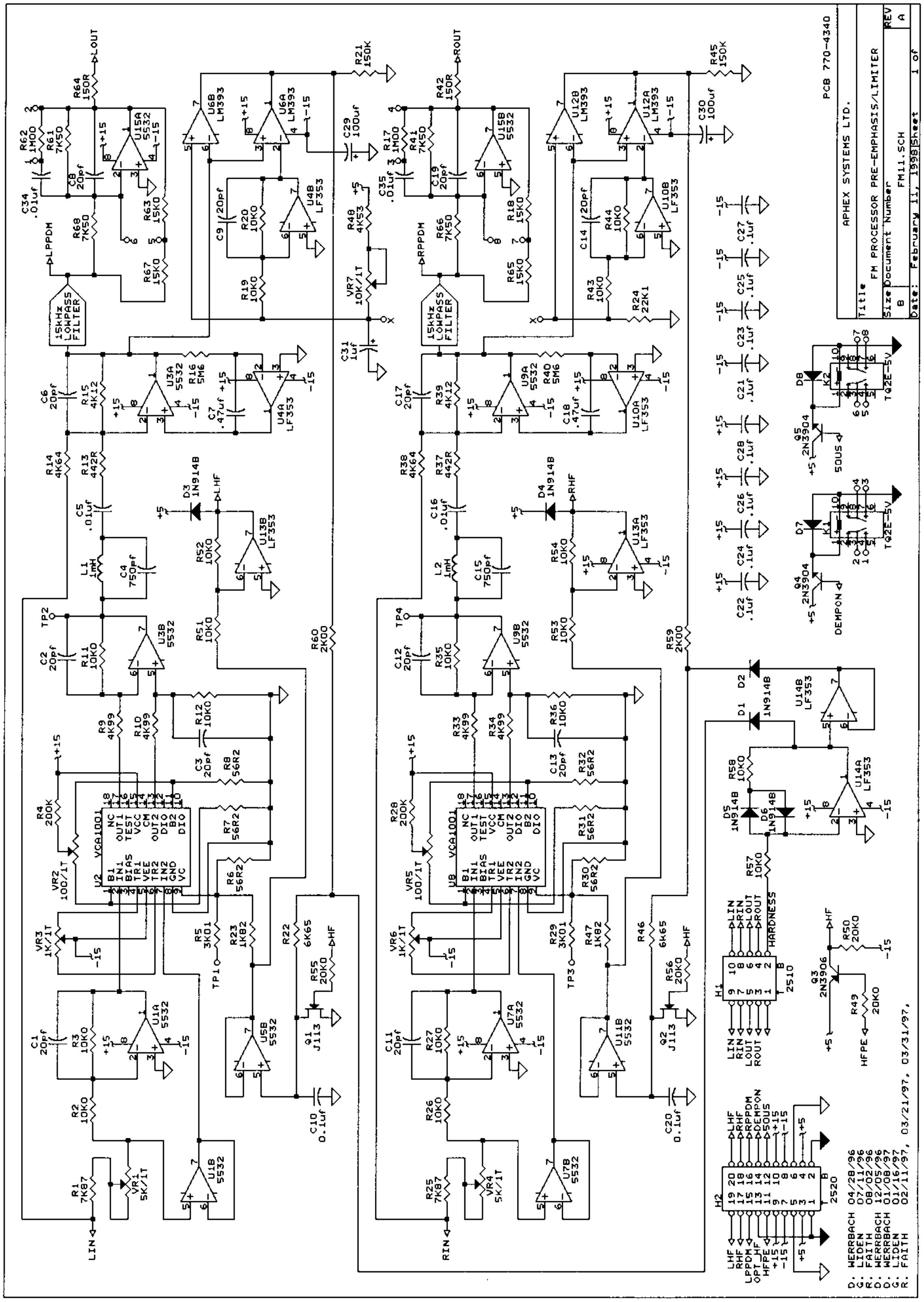
Document Number
FM10.C
REV
C
Date:
May 25, 2000
Sheet
1 of 1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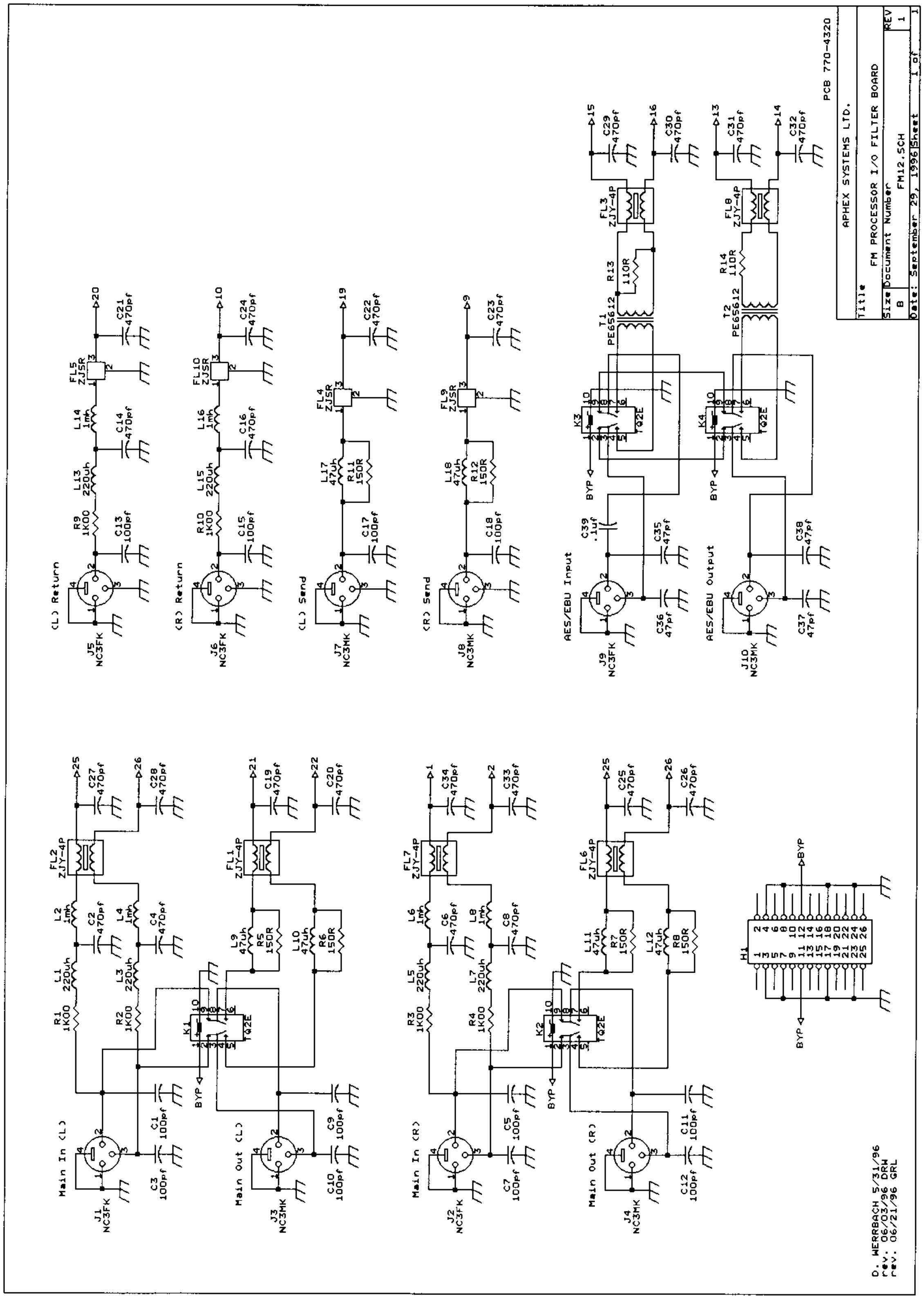
APHEX SYSTEMS LTD.
FM PROCESSOR PRE-EMPHASIS/LIMITER
FM11.SCH
February 11, 1998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Size Document Number
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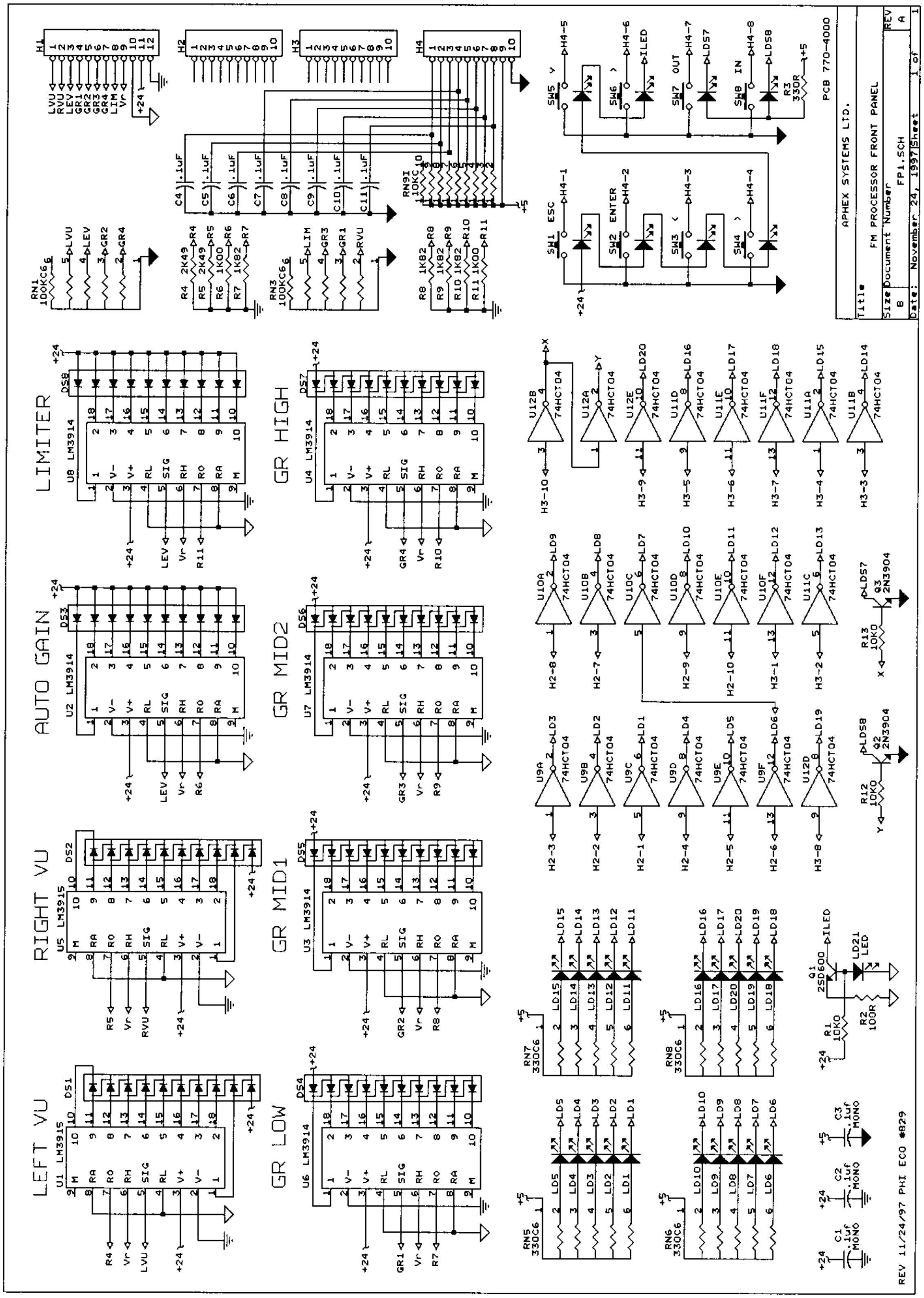
REV
1
Date: September 29, 1996 Sheet 1 of 1
D. WERRBACH 5/31/96
rev: 06/03/96 DRW
rev: 06/21/96 GRL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Size B
Document Number FP1.SCH
REV A
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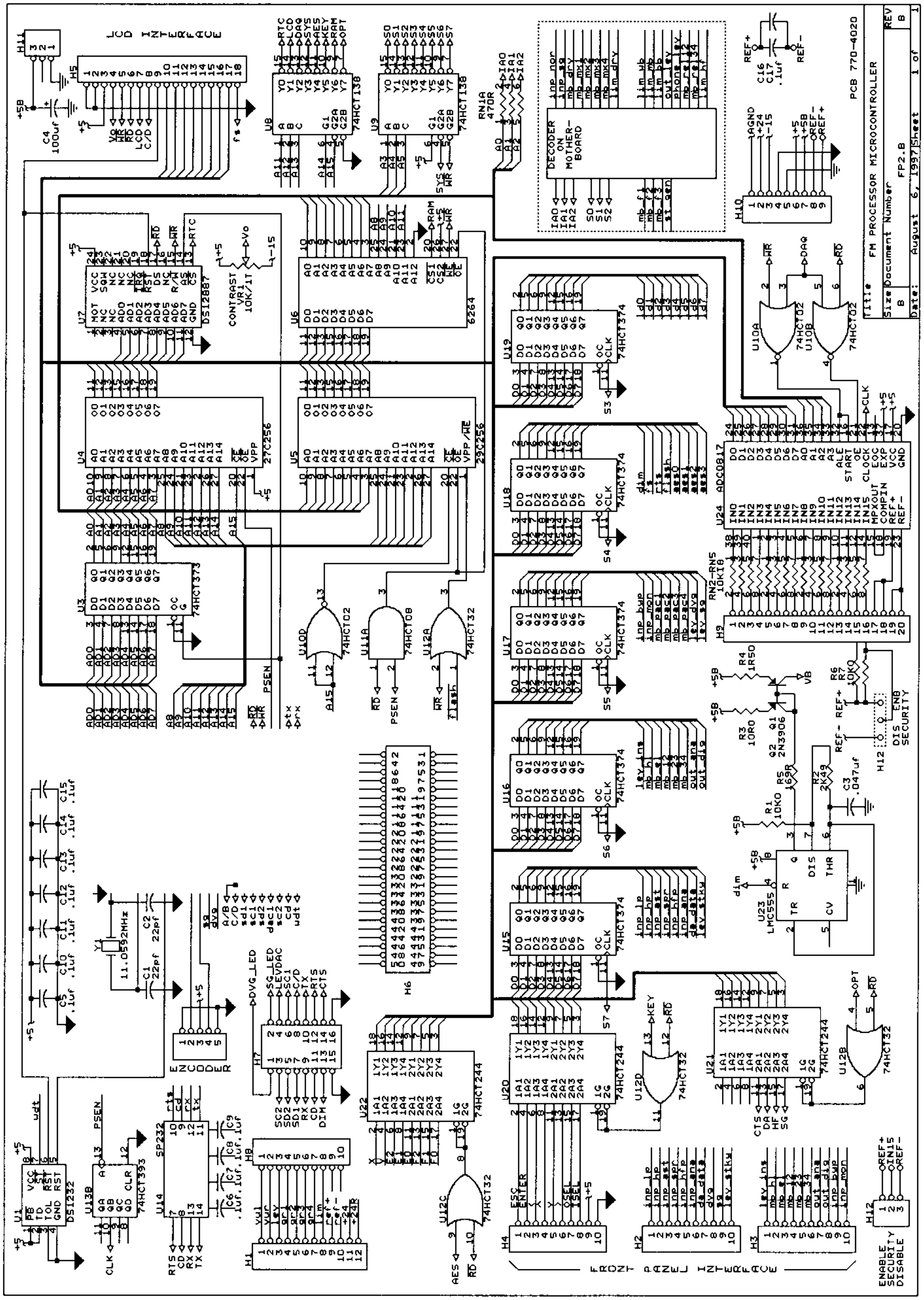
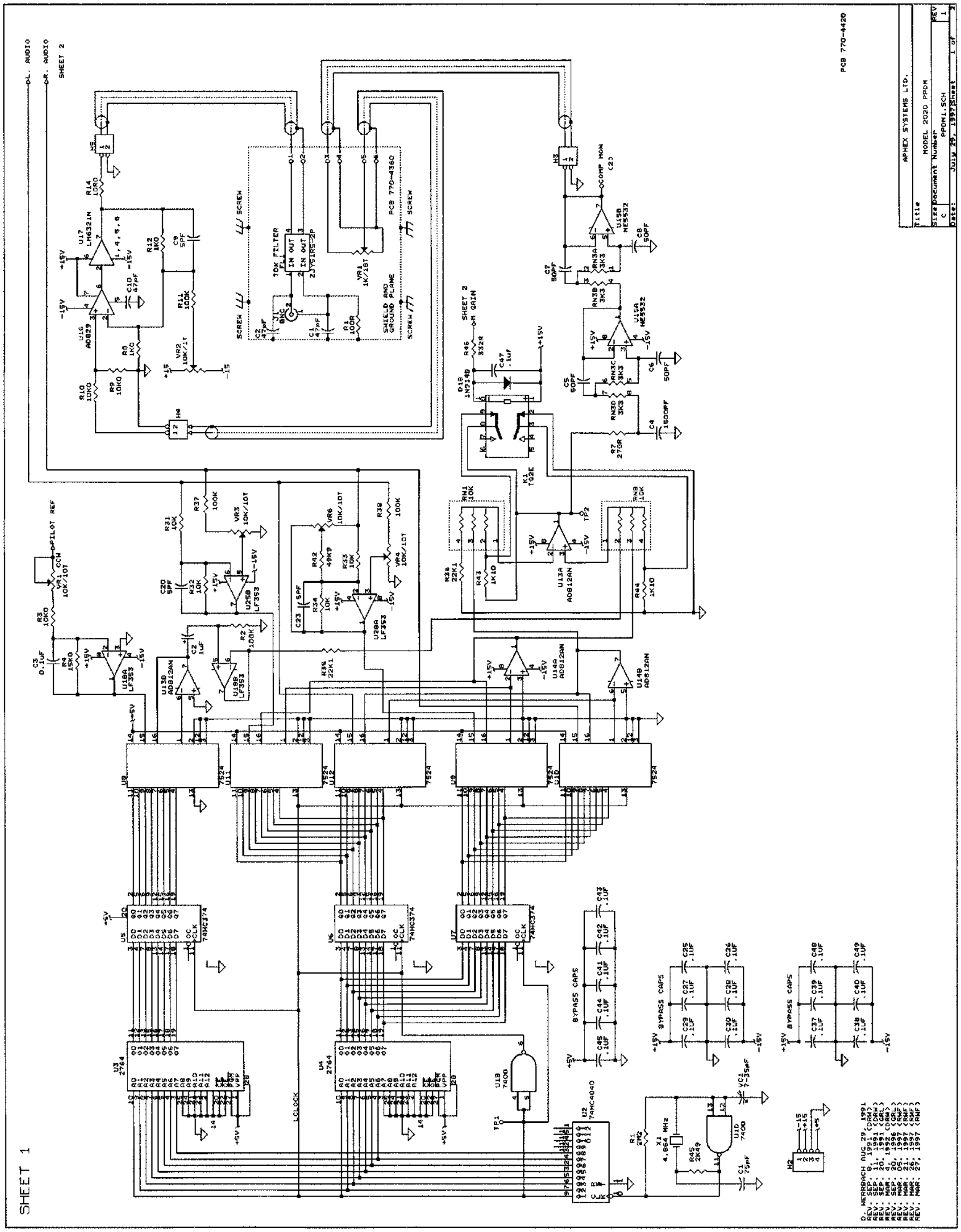
PCB 770-4000
REV 11/24/97 PHI ECO #829


FM PROCESSOR MICROCONTROLLER
PCB 770-402D
FP2.B
August 6, 1997
LCD INTERFACE
FRONT PANEL INTERFACE



SHEET 1
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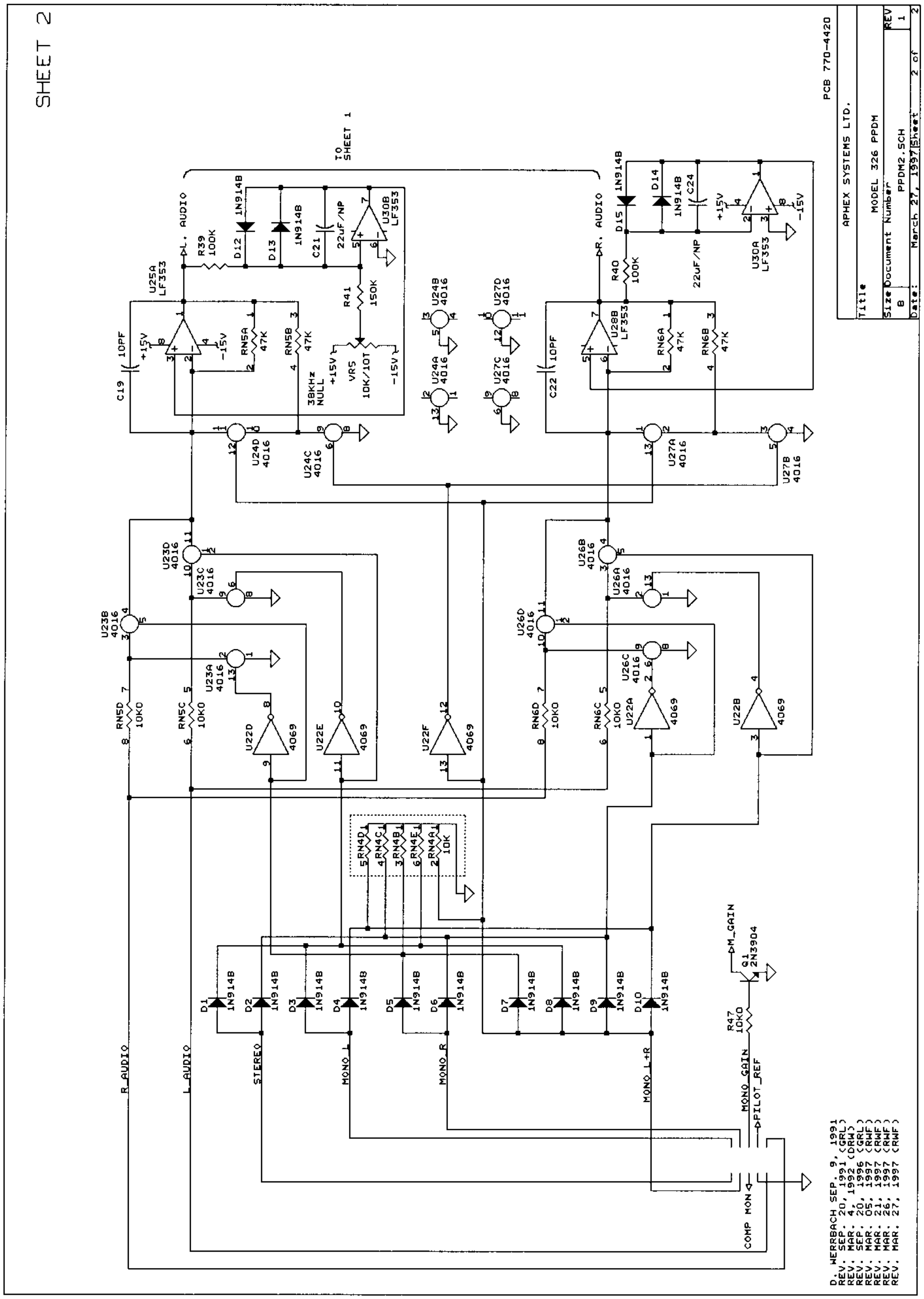
MODEL 2020 PPDM
PPDM1.SCH
PCB 770-4420
BYPASS CAPS


SHEET 2
PCB 770-4420
APHEX SYSTEMS LTD.
MODEL 326 PPDM
PPDM2.SCH
March 27, 1997
D. WERRBACH SEP. 9, 1991
COMP MON
MONO_GAIN
PILOT_REF
R_AUDIO
L_AUDIO
STEREO
MONO L
MONO R
MONO L+R
TO SHEET 1

16.1 Purpose

The parts list of this manual is supplied for the purpose of repair and maintenance of the FM Pro by qualified technicians only. Parts may be ordered for replacement purposes by the Aphex part number and part description. Always replace damaged parts with original parts supplied by Aphex if the exact replacement is not available from local resources. Certain parts are made exclusively by or for Aphex and must be ordered directly from us or through our distributors.

16.2 Disclaimer

Publishing this parts list does not imply that Aphex grants license to modify the product in any way or that Aphex shall supply parts for any purpose other than servicing a defective unit. While the present partlist has been compiled from actual manufacturing bills of materials and is believed to be substantially accurate, publishing discrepancies may occur due to running production changes, design changes, typographical errors, or other reasons. When ordering repair parts, be sure to acertain the part number and revision level of the faulty assembly, the unit manufacturing date or date purchased.

16.3 The Parts List

Mother Board

Assembly: 050-3240

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	2	C1,C3	Cap, Elect., NP, Music	22mF	+/-20%	225-0020
2	2	C2,C4	100ufNP	100mF	+/-20%	225-0040
3	2	C5,C6	10uf TANT	10mF	+/-10%	220-0280
4	2	C7, C8	Cap, Monolithic Ceramic, Z5U	.1mF	+/-20%	235-0020
5	2	C9, C10	Cap, polypropylene	.01mF	+/-2.5%	190-1380
6	3	D1, D2, D3	Diode, 1N4003			470-0120
7	19	H1,H2,H4,H6,H7,H9,H10, H15,H16,H17,H18,H19,H20, H21,H22,H23,H25,H26,H27	Female 8500			290-0480
8	8	H3,H5,H8,H11,H12,H13,H14, H24	10-Pin Female 8500			290-0500
9	1	H28	20-Pin Male 2500 Box			290-0580
10	1	H29	16-Pin Male 2500 Box			290-0940
11	2	H30,H36	10-Pin Male 2500 Box			290-0560
12	1	H31	50-Pin Male 2500 Box			290-0960
13	1	H32	Ribbon-Cable Assy (26 Pin)			030-4140
14	1	H33	5-Pin Molex header: .100” cen.			310-1080
15	1	H34	5-Pin Molex header: .156” cen.			310-2940
16	1	H35	3-Pin Molex header: .100” cen.			310-0780
17	3	K1, K2, K3	Relay, Aromat TQ2E12V*			630-0100
18	2	L1,L2	Inductor	47mH	+/-10%	360-0100
19	6	Q1,Q2,Q3,Q4,Q5,Q6	Transistor, NPN SI, 2N3904			500-0260
20	2	R1,R7	Resistor, Zero Ohms (Jumper)			840-1160
21	4	R2,R8,R21,R24	Resistor, Tin-Oxide Film	100K	+/-1%	120-1006
22	2	R4, R10	Resistor, Tin-Oxide Film	3K01	+/-1%	120-3014
23	2	R3,R9	Resistor, Tin-Oxide Film	301R	+/-1%	120-3013
24	2	R5,R11	Resistor, Tin-Oxide Film	49R9	+/-1%	120-4992
25	2	R6,R12	Resistor, Tin-Oxide Film	150R	+/-1%	120-1503

26	1	R18	Resistor, Tin-Oxide Film	113R	+/-1%	120-1133
27	6	R13,R14,R16,R1,R25,R27	Resistor, Tin-Oxide Film	10K0	+/-1%	120-1005
28	2	R19, R22	Resistor, Tin-Oxide Film	2K49	+/-1%	120-2494
29	3	R15, R26, R28	Resistor, Tin-Oxide Film	249R	+/-1%	120-2493
30	2	R20, R23	Resistor, Tin-Oxide Film	4K99	+/-1%	120-4994
31	1	RN1	Resistor Network	10KC4	+/-2%	140-1400
32	1	U5	IC, 74HCT374			480-1040
33	1	U1	IC, Pwr Amp, NSC LM1876TF			480-1900
34	3	U2, U3, U4	IC, CMOS, 74HCT138*			480-0980
35	1	None	PC Board, 100% Tested			770-4100
36	3	None	IC Socket, 16-pin			310-0060
37	1	None	Heat sink (for U1)			700-0100

I/O Board
Assembly: 050-2940

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	6	C1,C2,C3,C4,C7,C8	Cap, Elect., NP, Music	22mF	+/-20%	225-0020
2	6	C5,C6,C21,C22,C27,C28	Cap, Polyester, .2” LS	.47mF	+/-5%	190-1360
3	4	C11,C12,C29,C30	Cap, MICA	10pF	+/-5%	160-0020
4	4	C13,C14,C15,C16	Cap, Elect., NP, Music	100mF	+/-20%	225-0040
5	8	C17,C18,C19,C20,C23,C24, C25,C26	Cap, MICA	20pF	+/-5%	160-0060
6	7	C31,C32,C33,C34,C35,C36, C37	Cap, Monolithic Ceramic, Z5U	.1mF	+/-20%	235-0020
7	1	D1	Diode, Silicon Signal, 1N914B			470-0040
8	2	H1,H2	Connector, male 2520RA*			290-0600
9	1	K1	Relay, DPDT, 12V, TQ2E-12V			630-0100
10	4	L1,L2,L3,L4	Inductor	47mH		360-0100
11	2	Q1,Q2	Transistor, NPN SI, 2N3904			500-0260
12	2	R2,R7	Resistor, Tin-Oxide Film	1M00	+/-1%	120-1007
13	2	R11,R12	Resistor, Tin-Oxide Film	10K0	+/-1%	120-1005
14	4	R3,R4,R8,R9	Resistor, Tin-Oxide Film	2K87	+/-1%	120-2874
15	6	R5,R10,R17,R18,R28,R29	Resistor, Tin-Oxide Film	10M	+/-5%	070-1008
16	1	R13	Resistor, Tin-Oxide Film	113R	+/-1%	120-1133
17	4	R19,R20,R30,R31	Resistor, Tin-Oxide Film	1M00	+/-1%	120-1007
18	4	R21,R22,R32,R33	Resistor, Tin-Oxide Film	56R2	+/-1%	120-5622
19	2	R23,R34	Resistor, Tin-Oxide Film	2M21	+/-1%	120-2217
20	7	R14,R24,R25,R35,R36,R39, R40	Resistor, Tin-Oxide Film	20K0	+/-1%	120-2005
21	4	R26,R27,R37,R38	Resistor, Tin-Oxide Film	150R	+/-1%	120-1503
22	2	RN1,RN2	Resistor Network, 8-Pin, Isolated	8K2	+/-2%	140-0300
23	3	RN3,RN8,RN9	Resistor Network, 8-Pin, Isolated	10K	+/-2%	140-0120
24	8	RN4A,RN4D,RN5A,RN5D, RN6A,RN6D,RN7A,RN7D	Resistor, Tin-Oxide Film	10K0	+/-1%	120-1005
25	8	RN4B,RN4C,RN5B,RN5C, RN6B,RN6C,RN7B,RN7C	Resistor, Tin-Oxide Film	35K7	+/-1%	120-3575
26	1	RN10	R Network, 100KC6	100K	+/-2%	140-1460
27	6	U1,U3,U5,U6,U8,U10	IC, Op Amp, NE5532			490-0300
28	4	U2,U4,U7,U9	IC, Op Amp, LF353			490-0140
29	2	U11,U12	IC, Analog Switch, CD4016			460-0060
30	2	VR1,VR2	Trimpot, PT10H	50K		420-0100
31	2	VR3,VR4	Trimpot, PT10H	10K		420-0240
32	1	-	PC Board, 100% Tested			770-4120
33	10	-	IC Socket, 8 Pin			310-0020
34	2	-	IC Socket, 14 Pin			310-0040
35	2	TP1, TP2	Test Point			840-0720

Crossover Board
Assembly: 050-2960

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	4	C1,C2,C7,C8	Cap, Polypropylene, 63V	.022mF	+/-2.5%	190-1570
2	4	C3,C4,C9,C10	Cap, Polypropylene, 63V	.01mF	+/-2.5%	190-1380
3	4	C5,C6,C11,C12	Cap, Polypropylene, 63V	.0047mF	+/-2.5%	190-1520
4	7	C13,C14,C15,C16,C17,C18, C19	Cap, Monolithic Ceramic, Z5U	0.1mF	+/-20%	235-0020
5	2	H1,H2	Connector, male 2520RA*			290-0600

6	4	R1,R2,R7,R8	Resistor, Tin-Oxide Film	2K87	+/-1%	120-2874
7	4	R3,R4,R9,R10	Resistor, Tin-Oxide Film	2K49	+/-1%	120-2494
8	4	R5,R6,R11,R12	Resistor, Tin-Oxide Film	2K67	+/-1%	120-2674
9	12	RN1,RN2,RN3,RN4,RN5,RN6, RN7,RN8,RN9,RN10,RN11, RN12	Resistor Network, 8 Pin Isolated	10K	+/-2%	140-0120
10	6	U1,U2,U3,U4,U5,U6	IC, Op Amp, Quad, JFET, LF347			490-0460
11	6	U7,U9,U11,U13,U15,U17	IC, Dac, Mult, Dual, MP7528JN			490-0500
12	6	U8,U10,U12,U14,U16,U18	IC, Op Amp, Dual, JFET, LF353			490-0140
13	1	None	PC Board, 100% Tested			770-4280
14	6	None	IC Socket, 20 Pin			310-0220
15	6	None	IC Socket, 14 Pin			310-0040
16	6	None	IC Socket, 8 Pin			310-0020
17	8	C20, C21, C22, C23, C24, C25, C26, C27	Cap, Elec., NP	22mF	+/-20%	225-0020

Octal Sidechain Board

Assembly: 050-2980

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	16	C1,C3,C4,C6,C7,C9,C10,C12, C13,C15,C16,C18,C19,C21, C22,C24	Cap, Tant	1.0mF	+/-20%	220-0020
2	8	C2,C5,C8,C11,C14,C17,C20, C23	Cap, Tant	4.7mF	+/-20%	220-0220
3	5	C25,C26,C27,C28,C29	Cap, Monolithic Ceramic, Z5U	0.1mF	+/-20%	235-0020
4	48	D1,D2,D3,D4,D5,D6,D7,D8, D9,D10,D11,D12 D13,D14,D15,D16,D17,D18, D19,D20,D21,D22,D23,D24 D25,D26,D27,D28,D29,D30, D31,D32,D33,D34,D35,D36 D37,D38,D39,D40,D41,D42, D43,D44,D45,D46,D47,D48	Diode, Silicon Signal, 1N914B			470-0040
5	2	H1,H2	Connector, male 2520RA*			290-0600
6	7	ISO1,ISO2,ISO3,ISO4,ISO5, ISO6,ISO7	Optocoupler, Dual, Quality Tech P/N MCT9001			490-0550
7	8	Q1,Q2,Q3,Q4,Q5,Q6,Q7,Q8	Transistor, SI PNP, 2N3906			500-0180
8	8	R1,R9,R17,R25,R33,R41,R49, R57	Resistor, Tin-Oxide Film	499K	+/-1%	120-4996
9	8	R2,R10,R18,R26,R34,R42, R50,R58	Resistor, Tin-Oxide Film	4K99	+/-1%	120-4994
10	8	R3,R11,R19,R27,R35,R43, R51,R59	Resistor, Tin-Oxide Film	10K0	+/-1%	120-1005
11	8	R4,R12,R20,R28,R36,R44, R52,R60	Resistor, Tin-Oxide Film	2K49	+/-1%	120-2494
12	2	R5,R13	Resistor, Tin-Oxide Film	42K2	+/-1%	120-4225
13	22	R6,R14,R21,R22,R29,R30, R37,R38,R45,R46,R53,R54, R61,R62,R65,R66,R67,R68 R69,R70,R71,R72	Resistor, Tin-Oxide Film	20K0	+/-1%	120-2005
14	8	R7,R15,R23,R31,R39,R47, R55,R63	Resistor, Tin-Oxide Film	47K5	+/-1%	120-4755
15	8	R8,R16,R24,R32,R40,R48, R56,R64	Resistor, Tin-Oxide Film	365K	+/-1%	120-3656
16	8	RN1,RN5,RN2,RN6,RN3,RN7, RN4,RN8	Resistor Network, 8-Pin Isolated	10K	+/-2%	140-0120
17	2	RN9, RN11	Resistor Network	470RC6	+/-2%	140-1420
18	1	RN10	Resistor Network	10KC6	+/-2%	140-0600
19	8	U1,U5,U2,U6,U3,U7,U4,U8	IC, Quad Op Amp, JFET, LF347			490-0460
20	4	U9,U10,U11,U12	IC,CMOS,Quad Ana. Sw,CD4016			460-0060
21	1	None	PC Board, 100% Tested			770-4200
22	7	None	IC Socket, 8-Pin			310-0020
23	12	None	IC Socket, 14-Pin			310-0040

Octal VCA Board
Assembly: 050-3000

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	16	C1,C2,C3,C4,C7,C8,C9,C10 C13,C14,C15,C16,C19,C20, C21,C22	Cap, Mica	20pF	+/-5%	160-0060
2	8	C5,C6,C11,C12 C17,C18,C23,C24	Cap, Polyester	.33mF	+/-10%	190-0760
3	2	C25,C26	Cap, Mica	10pF	+/-5%	160-0020
4	6	C27,C28,C29,C30,C31,C32	Cap, Monolithic Ceramic, Z5U	.1mF	+/-20%	235-0020
5	1	H1	Connector, male 2520RA*			290-0600
6	1	H2	Connector, male 2510RA*			290-0620
7	16	R1,R2,R9,R10,R23,R24,R31, R32,R45,R46,R53,R54,R67, R68,R75,R76	Resistor, Tin-Oxide Film	1K00	+/-1%	120-1004
8	16	R3,R8,R11,R16,R25,R30, R33,R38,R47,R52,R55,R60, R69,R74,R77,R82	Resistor, Tin-Oxide Film	4K99	+/-1%	120-4994
9	24	R4,R5,R6,R12,R13,R14, R26,R27,R28,R34,R35,R36 R48,R49,R50,R56,R57,R58, R70,R71,R72,R78,R79,R80	Resistor, Tin-Oxide Film	21R5	+/-1%	120-2152
10	8	R7,R15,R29,R37 R51,R59,R73,R81	Resistor, Tin-Oxide Film	200K	+/-1%	120-2006
11	8	R17,R18,R39,R40 R61,R62,R83,R84	Resistor, Carbon Film	5M6	+/-5%	070-5607
12	16	R19,R20,R21,R22,R41,R42, R43,R44,R63,R64,R65,R66, R85,R86,R87,R88	Resistor, Tin-Oxide Film	20K0	+/-1%	120-2005
13	4	R89,R90,R91,R92	Resistor, Tin-Oxide Film	56R2	+/-1%	120-5622
14	1	RN1	Resistor Network, 8-Pin Isolated	10K	+/-2%	140-0120
15	2	RN2,RN3	Resistor Network, 8-Pin Isolated	100R	+/-2%	140-1440
16	8	U1,U2,U5,U6,U9,U10,U13,U14	IC, VC Attenuator, VCA1001			480-0900
17	4	U4,U8,U12,U16	IC, Op Amp, Dual, JFET, LF353			490-0140
18	1	U17	IC, Op Amp, Dual, Bipolar, AD826			490-0760
19	4	U3,U7,U11,U15	IC, Op Amp, Dual, Bipolar, 5532			490-0300
20	8	VR1,VR3,VR5,VR7 VR9,VR11,VR13,VR15	Trimpot, PT10H	100/1T		420-0160
21	8	VR2,VR4,VR6,VR8 VR10,VR12,VR14,VR16	Trimpot, PT10H	1K/1T		420-0200
22	1	None	PC Board, 100% Tested			770-4180
23	8	None	IC Socket, 18-Pin			310-0080
24	9	None	IC Socket, 8-Pin			310-0020
25	8	None	Heat Spreader			040-0040
26	16	None	TP1, TP2, TP3, TP4, TP5, TP6, TP7, TP8, TP9, TP10, TP11, TP12			840-0720

Input Conditioner Board
Assembly: 050-3020

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	4	C1,C2,C5,C6	Cap, Polyester	0.047mF	+/-5%	190-0340
2	8	C3,C4,C7,C8,C22,C26,C24, C28	Cap, Mica	20pF	+/-5%	160-0060
3	10	C9,C10,C15,C16,C17, C18,C19,C20,C21,C23	Cap, Polypropylene	.001mF	+/-2.5%	190-1540
4	4	C11,C12,C13,C14	Cap, Polyester	0.1mF	+/-5%	190-0840
5	9	C37,C38,C39,C40,C41,C42 C43,C44,C45	Cap, Monolithic Ceramic, Z5U	0.1mF	+/-20%	235-0020
6	2	C25,C27	Cap, Polypropylene	.01mF	+/-2.5%	190-1380
7	8	C29,C30,C31,C32,C33,C34, C35,C36	Cap, Elect., NP, Music	22mF	+/-20%	225-0020
8	4	D1,D2,D3,D4	Diode, Silicon Signal, 1N914B			470-0040
9	1	H1	3M P/N 2510-5002-UG*			290-0620
10	1	H2	3M P/N 2520-5002-UG*			290-0600
11	4	K1,K2,K3,K4	Relay, DPDT, 5V, TQ2E-5V			630-0140

12	4	Q1,Q2,Q3,Q4	Transistor, NPN SI, 2N3904			500-0260
13	4	R1,R2,R3,R4	Resistor, Tin-Oxide Film	22K1	+/-1%	120-2215
14	2	R5,R7	Resistor, Tin-Oxide Film	56K2	+/-1%	120-5625
15	2	R6,R8	Resistor, Tin-Oxide Film	115K	+/-1%	120-1156
16	10	R9,R10,R20,R25,R26,R36, R42,R43,R46,R47	Resistor, Tin-Oxide Film	15K0	+/-1%	120-1505
17	4	R11,R13,R27,R29	Resistor, Tin-Oxide Film	12K1	+/-1%	120-1215
18	2	R12,R28	Resistor, Tin-Oxide Film	4K99	+/-1%	120-4994
19	2	R14,R30	Resistor, Tin-Oxide Film	71K5	+/-1%	120-7155
20	2	R15,R31	Resistor, Tin-Oxide Film	11K8	+/-1%	120-1185
21	2	R16,R32	Resistor, Tin-Oxide Film	10K5	+/-1%	120-1055
22	2	R17,R33	Resistor, Tin-Oxide Film	10K2	+/-1%	120-1025
23	4	R18,R23,R34,R39	Resistor, Tin-Oxide Film	10K0	+/-1%	120-1005
24	2	R19,R35	Resistor, Tin-Oxide Film	43K2	+/-1%	120-4325
25	2	R21,R37	Resistor, Tin-Oxide Film	30K1	+/-1%	120-3015
26	2	R22,R38	Resistor, Tin-Oxide Film	17K8	+/-1%	120-1785
27	2	R24,R40	Resistor, Tin-Oxide Film	3K65	+/-1%	120-3654
28	4	R41,R44,R45,R48	Resistor, Tin-Oxide Film	7K50	+/-1%	120-7504
29	8	R49,R50,R51,R52,R53,R54, R55,R56	Resistor, Tin-Oxide Film	100K	+/-1%	120-1006
30	4	RN1,RN2,RN3,RN4	Resistor Network, 8-Pin Isolated	10K	+/-2%	140-0120
31	8	U1,U2,U3,U4,U5,U6, U7,U8,U9,U10,U11,U12	IC, Op Amp, Bipolar, Dual, 5532			490-0300
32	1	None	PC Board, 100% Tested			770-4140
33	8	None	IC Socket, 8-Pin			310-0020

Silence Gate/Leveler Board

Assembly: 050-3040

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	8	C1,C2,C4,C5,C6,C7,C9,C10	Cap, Mica	10pF	+/-5%	160-0020
2	1	C3	Cap, Tant	1mF	+/-20%	220-0020
3	3	C8,C11,C15	Cap, Polyester	0.33mF	+/-10%	190-0760
4	2	C12,C16	Cap, Polyester	0.47mF	+/-5%	190-1360
5	2	C13,C17	Cap, Tant	2.2mF	+/-10%	220-0140
6	1	C14	Cap, Polyester	0.15mF	+/-5%	190-0380
7	2	C18,C21	Cap, Polyester	0.10mF	+/-5%	190-0840
8	1	C19	Cap, Polyester	0.22mF	+/-5%	190-1780
9	1	C20	Cap, Polyester	.022mF	+/-5%	190-0260
10	9	C22,C23,C24,C25,C26,C27, C28,C30,C31	Cap, Monolithic Ceramic, Z5U	0.10mF	+/-20%	235-0020
11	1	C29	Cap, Tant	10mF	+/-20%	220-0280
12	29	D1,D2,D3,D4,D5,D6 D7,D8,D9,D10,D11,D12, D13,D14,D15,D16,D17,D18, D19,D21,D22,D23,D24,D25 D26,D27,D28,D29,D30	Diode, Silicon Signal, 1N914B			470-0040
13	1	DR1	IC, Digital Resistor, DS1807			490-0650
14	1	H1	3M P/N 2510-5002-UG*			290-0620
15	1	H2	3M P/N 2520-5002-UG*			290-0600
16	3	LD1,LD2,LD3	LED, Green LTL			530-0260
17	1	Q1	Transistor, NPN SI, 2N3904			500-0260
18	3	Q2,Q4,Q5	Transistor, PNP SI, 2N3906			500-0180
19	1	Q3	Transistor Array, NPN SI, CA3046			480-0660
20	1	R1	Resistor, Carbon Film	2M2	+/-5%	070-2207
21	5	R2, R3, R33, R35, R57	Resistor, Tin-Oxide Film	10K0	+/-1%	120-1005
22	4	R4, R12, R23, R39	Resistor, Tin-Oxide Film	1K00	+/-1%	120-1004
22	7	R5,R10,R16,R21,R27,R28, R56	Resistor, Tin-Oxide Film	4K99	+/-1%	120-4994
23	2	R6,R17	Resistor, Tin-Oxide Film	200K	+/-1%	120-2006
24	6	R7,R8,R9,R18,R19,R20	Resistor, Tin-Oxide Film	56R2	+/-1%	120-5622
25	4	R11,R22,R45,R65	Resistor, Tin Oxide	2K49	+/-1%	120-2494
26	3	R15,R26,R31	Resistor, Carbon Film	5M6	+/-5%	070-5607
27	4	R13, R14, R24, R25	Resistor, Tin-Oxide Film	68K1	+/-1%	120-6815
28	5	R29,R47, R55, R58, R60	Resistor, Tin-Oxide Film	100K	+/-1%	120-1006
29	1	R30	Resistor, Tin-Oxide Film	42K2	+/-1%	120-4225
30	1	R32	Resistor, Tin Oxide Film	40K2	+/-1%	120-4025
31	1	R34	Resistor, Tin-Oxide Film	249R	+/-1%	120-2493

32	1	R36	Resistor, Tin-Oxide Film	7K68	+/-1%	120-7684
33	1	R37	Resistor, Tin-Oxide Film	1K58	+/-1%	120-1584
34	1	R38	Resistor, Tin-Oxide Film	10K2	+/-1%	120-1025
35	1	R40	Resistor, Tin-Oxide Film	1M00	+/-1%	120-1007
36	1	R42	Resistor, Tin-Oxide Film	24K9	+/-1%	120-2495
37	1	R43	Resistor, Tin-Oxide Film	499R	+/-1%	120-4993
38	1	R44	Resistor, Tin-Oxide Film	16K5	+/-1%	120-1655
39	1	R46	Resistor, Tin Oxide Film	1K50	+/-1%	120-1504
40	1	R48	Resistor, Tin-Oxide Film	2K10	+/-1%	120-2104
41	1	R49	Resistor, Tin-Oxide Film	1K37	+/-1%	120-1374
42	1	R50	Resistor, Tin-Oxide Film	82K5	+/-1%	120-8255
43	1	R51	Resistor, Tin-Oxide Film	10M	+/-5%	070-1008
44	1	R52	Resistor, Tin-Oxide Film	61K9	+/-1%	120-6195
45	1	R53	Resistor, Tin-Oxide Film	2M4	+/-5%	070-2407
46	1	R54	Resistor, Tin-Oxide Film	1K82	+/-1%	120-1824
47	1	R59	Resistor, Tin-Oxide Film	3K3	+/-5%	070-3304
48	1	R61	Resistor, Carbon Film	680R	+/-5%	070-6803
49	1	R62	Resistor, Tin Oxide Film	100R	+/-1%	120-1003
50	1	R63	Resistor, Tin Oxide Film	332K	+/-1%	120-3326
51	1	R64	Resistor, Tin Oxide Film	20K0	+/-1%	120-2005
52	5	RN1,RN2,RN3,RN4,RN5	Resistor Network, 8-Pin Isolated	10K	+/-2%	140-0120
53	2	RN6,RN7	Resistor Network, 6-Pin Isolated	10KI6	+/-2%	140-1350
54	3	U1,U8,U15	IC, Op Amp, Dual, JFET, LF353			490-0140
55	1	U2	LM311			480-0040
56	3	U3,U4,U7	IC, Op Amp, Dual, Bipolar, 5532			490-0300
57	2	U5,U6	IC, VC Attenuator, VCA1001			480-0900
58	3	U9,U10,U12	IC, Op Amp, Quad, JFET, LF347			490-0460
59	1	U11	IC, Comparator, Dual, LM393			480-0240
60	1	U13	DAC, Multiplying, Triple, MAX512			490-0600
61	1	U14	IC, NOR, Quad, CMOS, CD4001			460-0100
62	1	U16	CD4016			460-0060
63	1	U17	IC, Voltage Reference, TL431CLP			480-1620
64	2	VR1,VR3	Trimpot, PT10H	1K/1T		420-0200
65	2	VR2,VR4	Trimpot, PT10H	100/1T		420-0160
66	1	None	PC Board, 100% Tested			770-4160
67	8	None	IC Socket, 8-Pin			310-0020
68	6	None	IC Socket, 14-Pin			310-0040
69	2	None	IC Socket, 18-Pin			310-0080
70	2	None	Heat Spreader			040-0040
71	3	None	Test Point			840-0720

Limiter/Bass Proc Board
Assembly: 050-2900

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	1	C1	Cap, Mica	10pf	+/-5%	160-0020
2	7	C2,C5,C6,C7,C8,C10,C11	Cap, Polypropylene	.01uf	+/-2.5%	190-1380
3	2	C3,C4	Cap, Polyester	.047uf	+/-5%	190-0340
4	1	C9	Cap, Elec, Non-Polar, Music	22uf	+/-20%	225-0020
5	1	C12	Cap, Polyester	.33uf	+/-10%	190-0760
6	2	C13,C31	Cap, Polyester	.1uf	+/-5%	190-0840
7	2	C14,C25	Cap, Polyester	.0022uf	+/-2.5%	190-1560
8	1	C15	Cap, Polyester	.47uf	+/-5%	190-1360
9	10	C16,C17,C18,C19,C20,C21, C22,C23,C27,C28	Cap, Monolithic Ceramic, Z5U	.1uf	+/-20%	235-0020
10	1	C24	Cap, Electrolytic	22uf	+/-20%	200-0040
11	1	C26	Cap, Tant	1uf		220-0020
12	2	C29,C30	Cap, Polyester, 0.2" L.S.	.022uf		190-1720
13	11	D1,D2,D3,D4,D5,D6,D7, D8,D9,D10,D11	Diode, SI Signal, 1N914B			470-0040
14	2	H1,H2	3M P/N 2510-5002-UG			290-0620
15	1	H3	Header, 2-Pin R/A on 0.1" Centers			310-0920
16	1	H4	Header, 3-Pin R/A on 0.1" Centers			310-0800
17	2	LD1,LD2	LED, Green LTL			530-0260
18	1	Q1	JFET, N-Channel, J113			500-0160
19	1	Q2	Transistor, PNP SI, 2N3906			500-0180
20	4	R1,R3,R64,R74		49K9	+/-1%	120-4995
21	8	R2,R14,R15,R16,R17,R23,		100K	+/-1%	120-1006

		R24,R33				
22	6	R4,R36,R38,R55,R58,R68		15K0	+/-1%	120-1505
23	20	R5,R6,R8,R21,R29,R30, R31,R34,R35,R40,R41,R43, R47,R48,R49,R50,R51,R52, R60,R61	Resistor, Tin-Oxide Film	10K0	+/-1%	120-1005
24	1	R7	Resistor, Tin-Oxide Film	3K83	+/-1%	120-3834
25	2	R9,R11	Resistor, Tin-Oxide Film	42K2	+/-1%	120-4225
26	2	R10,R62	Resistor, Tin-Oxide Film	4K99	+/-1%	120-4994
27	1	R12	Resistor, Tin-Oxide Film	5K62	+/-1%	120-5624
28	4	R13,R18,R19,R25	Resistor, Tin-Oxide Film	402R	+/-1%	120-4023
29	1	R20	Resistor, Tin-Oxide Film	909R	+/-1%	120-9093
30	1	R22	Resistor, Tin-Oxide Film	9K09	+/-1%	120-9094
31	3	R26,R27,R28	Resistor, Tin-Oxide Film	2K49	+/-1%	120-2494
32	1	R32,R45	Resistor, Tin-Oxide Film	1M00	+/-1%	120-1007
33	4	R37,R39,R54,R69	Resistor, Tin-Oxide Film	1K00	+/-1%	120-1004
34	1	R42	Resistor, Tin-Oxide Film	20K0	+/-1%	120-2005
35	1	R44	Resistor, Tin-Oxide Film	150K	+/-1%	120-1506
36	1	R46	Resistor, Tin-Oxide Film	249K	+/-1%	120-2496
37	1	R53	Resistor, Tin-Oxide Film	2K00	+/-1%	120-2004
38	2	R56,R57	Resistor, Tin-Oxide Film	21R5	+/-1%	120-2152
39	1	R59	Resistor, Tin-Oxide Film	7K50	+/-1%	120-7504
40	1	R63	Resistor, Tin-Oxide Film	750K	+/-1%	120-7506
41	3	R65,R66,R75	Resistor, Tin-Oxide Film	10M0	+/-1%	120-1008
42	1	R67	Resistor, Tin-Oxide Film	4K53	+/-1%	120-4534
43	3	R70,R71,R72	Resistor, Tin-Oxide Film	3K32	+/-1%	120-3324
22	1	R73		56R2	+/-1%	120-5622
23	1	R76		30K1	+/-1%	120-3015
24	4	RN1,RN2,RN3,RN4	Resistor network, 16-pin Iso DIP	10KI16	+/-1/4%	140-1500
25	2	RN5,RN6	Resistor Network, 8-Pin Isolated	10K	+/-2%	140-0120
26	7	U1,U3,U4,U5,U6,U7,U12	IC, Op Amp, JFET, Quad, LF347			490-0460
27	2	U2,U8	IC, Op Amp, JFET, Dual, LF353			490-0140
28	1	U9	IC, Comparator, Quad, LM339			480-0080
29	1	U10	IC, Comparator, Dual, LM393	LM393		480-0240
30	1	U11	IC, VC Attenuator, VCA1001			480-0900
31	1	VR1	Trimpot, PT10H	20K/1T		420-0960
32	1	VR2	Trimpot, PT10H	1K/1T		420-0200
33	1	None	PC Board, 100% Tested, Rev B			770-4220
34	1	None	Heat Spreader for VCA			040-0040
35	3	None	IC Socket, 8-Pin			310-0020
36	8	None	IC Socket, 14-Pin			310-0040
37	4	None	IC Socket, 16-Pin DIP			310-0060
38	1	None	IC Socket, 18-Pin DIP			310-0080

Meter Board

Assembly: 050-3060

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	4	C1,C3,C5,C6	Cap, Elect., NP, Music	22mF	+/-20%	225-0020
2	3	C2,C4,C7	Cap, Polyester, .3" LS	0.1mF	+/-5%	190-0840
3	6	C8,C9,C10,C11,C12,C13	Cap, Monolithic Ceramic, Z5U	0.1mF	+/-20%	235-0020
4	53	D1,D2,D3,D4,D5,D6,D7,D8, D9,D10,D11,D12,D13,D14, D15,D16,D17,D18,D19,D20, D21,D22,D23,D24,D25,D26, D27,D28,D29,D30,D31,D32, D33,D34,D35,D36,D37,D38, D39,D40,D41,D42,D43,D44, D45,D46,D47,D48,D49,D50, D51,D52,D53	Diode, SI Signal, 1N914B			470-0040
5	2	H1,H2	3M P/N 2520-5002-UG*			290-0600
6	2	Q1, Q2	Transistor, NPN SI, 2N3904			500-0260
7	1	Q3	Transistor, SI PNP, 2N3906			500-0180
8	5	R1,R5,R9,R11, R19	Resistor, Tin-Oxide Film	4K99	+/-1%	120-4994
9	3	R2,R6,R16	Resistor, Tin-Oxide Film	68K1	+/-1%	120-6815
10	3	R3,R7,R20	Resistor, Tin-Oxide Film	1M00	+/-1%	120-1007
11	2	R4,R8	Resistor, Tin-Oxide Film	1M50	+/-1%	120-1507
12	1	R13	Resistor, Tin-Oxide Film	200K	+/-1%	120-2006

13	1	R14	Resistor, Tin-Oxide Film	20K0	+/-1%	120-2005
14	14	R10,R12,R15,R17,R25,R26, R28,R30,R32,R34,R26,R38, R40,R42	Resistor, Tin-Oxide Film	10K0	+/-1%	120-1005
15	1	R18	Resistor, Tin-Oxide Film	249K	+/-1%	120-2496
16	1	R21	Resistor, Tin-Oxide Film	1K00	+/-1%	120-1004
17	1	R22	Resistor, Tin-Oxide Film	30K1	+/-1%	120-3015
17	1	R23	Resistor, Carbon Film	2M2	+/-5%	070-2207
18	1	R24	Resistor, Tin-Oxide Film	12K4	+/-1%	120-1245
19	8	R27, R29, R31, R33, R35, R37, R39, R41	Resistor, Tin-Oxide Film	6K19	+/-1%	120-6194
20	2	R43,R44	Resistor, Tin-Oxide Film	42K2	+/-1%	120-4225
21	4	RN1,RN2,RN3,RN4	Resistor Network, 6-Pin Isolated	10KI6	+/-2%	140-1350
22	6	U1,U2,U3,U4,U5,U6	IC, Op Amp, JFET, Quad, LF347			490-0460
23	1	VR1	Trimpot, PT10H	10K		420-0240
24	1	None	PC Board, 100% Tested			770-4240
25	6	None	IC Socket, 14-Pin			310-0040

DAC Board

Assembly: 050-3080

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	4	C1,C3,C5,C7	Cap, Mica	10pF	+/-5%	160-0020
2	4	C2,C4,C6,C8	Cap, Polyester, .1" L.S. MKS02	.1mF	+/-10%	190-1820
3	28	C9,C10,C11,C12,C13,C14, C15,C16,C17,C18,C19,C20, C21,C22,C23,C24,C25,C26, C27,C28,C29,C30,C31,C32, C33,C34,C35,C36	Cap, Mono Ceramic, NP0 LS=0.1"	10pF	+/-5%	235-0120
4	8	C39,C40,C41,C42,C43,C44, C45,C46	Cap, Monolithic Ceramic, Z5U	0.1mF	+/-20%	235-0020
5	2	C37,C38	Cap, Mica	20pF	+/-5%	160-0060
6	1	C47	Cap, Tant	1mF	+/-20%	220-0020
7	5	H1,H2,H3,H4,H5	3M P/N 2520-5002-UG			290-0600
8	8	R1,R2,R3,R4,R5,R6,R7,R8	Resistor, Tin-oxide film	1M00	+/-1%	120-1007
9	2	RN1,RN2	Resistor Network, 6-Pin Common	10K	+/-2%	140-0600
10	16	U1,U3,U5,U7,U9,U11,U13, U15,U17,U19,U21,U23,U25, U27,U29,U31	IC, Dac, Mult, Dual, MP7528JN			490-0500
11	1	U2	IC, Op Amp, Bipolar, Dual, 5532			490-0300
12	15	U4,U8,U10,U12,U14,U16, U18,U20,U22,U24,U26,U28, U30,U32,U34	IC, Op Amp, JFET, Dual, LF353			490-0140
13	1	U6	IC, Op Amp, Bipolar, Dual, AD826AN			490-0760
14	1	U35	IC, Volt Reg, -5V, LM79L05			465-0220
15	4	VR1,VR2,VR3,VR4	Trimpot, 10K25T,R/A Adj.			420-0980
16	1	None	PC Board, 100% Tested (Rev. "A")			770-4260
17	16	None	IC Socket, 20 Pin			310-0220
18	17	None	IC Socket, 8 Pin			310-0020

Digital Audio Board

Assembly 050-3300

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	1	C1	Cap, Polyester, 0.2 L.S.	.047mF	+/-5%	190-1640
2	2	C2,C3	Cap, Polyester, 0.2 L.S.	.01mF	+/-5%	190-1700
3	4	C4,C5,C10,C11	Cap, Mica	750pF	+/-5%	160-0550
4	4	C6,C7,C12,C13	Cap, Mica	100pF	+/-5%	160-0160
5	2	C8,C14	Cap, Mica	20pF	+/-5%	160-0060
6	2	C9,C15	Cap, Polyester, 0.2" L.S.	.47mF	+/-5%	190-1360
7	4	C16,C20,C21,C25	Cap, Electrolytic, non-polar	22mF	+/-5%	225-0020
8	1	C26	Cap, Electrolytic	22mF N/P		200-0040
9	4	C17,C18,C22,C23	Cap, Mica	47pF	+/-5%	160-0100
10	3	C27,C28,C29	Cap, Monolithic Ceramic, NP0	47pF	+/-5%	235-0160
11	2	C19,C24	Cap, Polypropylene	.0068mF	+/-2.5%	190-1500
12	1	C30	Cap, Tant	10mF		230-0280
13	26	C31,C32,C33,C34,C35,C36, C37,C38,C39,C40,C45,C46, C47,C48,C49,C51,C53,C54,	Cap, Monolithic Ceramic, Z5U	.1mF	+/-20%	235-0020

		C55,C56,C58,C60,C62,C64,				
		C65,C66				
14	2	C41,C42	Cap, Polyester, 0.2" L.S.	.1mF	+/-5%	190-1680
15	6	C43,C44,C57,C59,C61,C63	Cap, Tant	1mF		230-0020
16	1	C50	Cap, Electrolytic, 35V	100mF	+/-20%	200-0240
17	1	C52	Cap, Electrolytic, 25V	100mF	+/-20%	200-1300
18	9	D1,D2,D3,D4,D5,D6,D7,D8,	Diode, Philips BAT85			470-0780
		D9				
19	2	H1,H2	3M P/N 2520-5002-UG*			290-0600
20	3	L1,L2,L3	Inductor	220mH		360-0080
21	1	Q1	Transistor, MJE180 (Motorola)			500-0560
22	3	Q2,Q7,Q8	2N3906			500-0180
23	1	Q3	MJE170 (Motorola)			500-0580
24	3	Q4,Q5,Q6	2N3904			500-0260
25	1	R1	Resistor, Tin-Oxide Film	1K00	+/-1%	120-1004
26	4	R2,R6,R12,R16	Resistor, Tin-Oxide Film	14K7	+/-1%	120-1475
27	4	R3,R5,R13,R15	Resistor, Tin-Oxide Film	31K6	+/-1%	120-3165
28	10	R4,R7,R9,R14,R17,R19,R27,	Resistor, Tin-Oxide Film	4K99	+/-1%	120-4994
		R28,R33,R34				
29	2	R8,R18	Resistor, Carbon Film	6M8	+/-5%	070-6807
30	2	R10,R20	Resistor, Tin-Oxide Film	17K4	+/-1%	120-1745
31	2	R11,R21	Resistor, Tin-Oxide Film	150R	+/-1%	120-1503
32	1	R22	Resistor, Tin-Oxide Film	2R7	+/-5%	070-2701
33	2	R23,R29	Resistor, Tin-Oxide Film	24K9	+/-1%	120-2495
34	3	R24,R30,R66	Resistor, Tin-Oxide Film	10K0	+/-1%	120-1005
35	2	R25,R31	Resistor, Tin-Oxide Film	2K00	+/-1%	120-2004
36	2	R26,R32	Resistor, Tin-Oxide Film	21R5	+/-1%	120-2152
37	4	R35,R36,R37,R38	Resistor, Tin-Oxide Film	39R2	+/-1%	120-3922
38	1	R39	Resistor, Tin-Oxide Film	100K	+/-1%	120-1006
39	2	R40,R41	Resistor, Tin-Oxide Film	1M00	+/-1%	120-1007
40	1	R42	Resistor, Tin-Oxide Film	499K	+/-1%	120-4996
41	6	R43,R45,R47,R51,R54,R55	Resistor, Tin-Oxide Film	2K49	+/-1%	120-2494
42	3	R44,R46,R48	Resistor, Tin-Oxide Film	2M00	+/-5%	070-2007
43	2	R49,R50	Resistor, carbon film, 1/2 W	51R	+/-5%	100-5102
43	3	R52,R64,R65	Resistor, Tin-Oxide Film	7K50		120-7504
44	2	R53,R56	Resistor, Tin-Oxide Film	100R		120-1003
45	1	R57	Resistor, Tin-Oxide Film	49R9		120-4992
46	6	R58,R59,R60,R61,R62,R63	Resistor, Tin-Oxide Film	200R		120-2003
47	2	R67,R68	Resistor, Carbon Film	4R7		070-4701
48	2	RN1,RN2	Resistor Network, 8-Pin Isolated	200R		140-0840
49	1	U1	IC, AES/EBU receiver, CS8412-CP*			490-0740
50	1	U2	IC, D/A converter, dual, CS4329-KP*			490-0780
51	5	U3,U5,U6,U7,U11	IC, op amp, bipolar, dual, NE5532			490-0300
52	2	U4,U8	IC, op amp, bi-fet, dual, LF353			490-0140
53	1	U9	IC, A/D converter, dual, CS5390-KP*			490-0800
54	1	U10	IC, AES/EBU transmit, CS8402-ACP			490-0720
55	1	U12	IC, CMOS, inverter, hex, 74HCU04			480-2000
56	1	U13	IC, CMOS, MUX, dual, 74HC153			480-1960
57	1	U14	IC, CMOS, flip-flop, dual, 74HC74			480-1980
58	1	U15	IC, PAL, AMD PALCE16V8H-15**			480-2020
59	3	VC1,VC2,VC3	Variable Capacitor, trimmer	7-35pF		170-0040
60	4	VR1,VR2,VR3,VR4	Trimpot, PT10H	10K		420-0240
61	1	Y1	Crystal, 12.288 MHz, +/-20 ppm 0~+			440-0140
62	1	Y2	Crystal, 11.2896 MHz, +/-20 ppm 0~			440-0160
63	1	Y3	Crystal, 8.192 MHz, +/-20 ppm 0~+7			440-0180
64	1	None	PC Board (4-Layer)			770-4300
65	1	None	IC Socket, 24-Pin X 0.3" *			310-3220
66	7	None	Socket, IC, 8-pin			310-0020
67	2	None	Socket, IC, 14-pin			310-0040
68	1	None	Socket, IC, 16-pin			310-0060
69	2	None	Socket, IC, 20-pin, 0.3" wide			310-0220
70	2	None	Socket, IC, 28-pin, 0.6" wide			310-1480

Pre-emphasis Limiter

Assembly 050-3180

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	12	C1,C2,C3,C6,C8,C9,C11,	20pf MICA	20pF	+/-5%	160-0060

		C12,C13,C14,C17,C19				
2	2	C4,C15	750pf MICA	750pF	+/-5%	160-0550
3	4	C5,C16,C34,C35	Cap, Polypropylene	.01mF	+/-2.5%	190-1380
4	2	C7,C18	Cap, Polyester, box, .2" LS	.47mF	+/-10%	190-1360
5	2	C10,C20	Cap, Polyester	.10mF	+/-5%	190-0840
6	8	C21,C22,C23,C24,C25,C26, C27,C28	Cap, Monolithic Ceramic, Z5U	.1mF	+/-20%	235-0020
7	2	C29,C30	Cap, Electrolytic	100mF	+/-20%	200-1300
8	1	C31	Capacitor	1mF	+/-20%	220-0020
9	8	D1,D2,D3,D4,D5,D6,D7,D8	Diode, Silicon Signal, 1N914B			470-0040
10	2	FILTER1,FILTER2	15KHz Lowpass Filter Assy			050-3280
11	1	H1	3M P/N 2510-5002-UG*			290-0620
12	1	H2	3M P/N 2520-5002-UG*			290-0600
13	2	K1,K2	Relay, DPDT, 5V, TQ2E-5V			630-0140
14	2	L1,L2	Inductor	1mH		360-0060
15	2	Q1,Q2	Transistor, JFET, N-Chan., J113			500-0160
16	1	Q3	Transistor, SI, PNP, 2N3906			500-0180
17	2	Q4,Q5	Transistor, SI, NPN, 2N3904			500-0260
18	2	R1, R25	Resistor, Tin-Oxide Film	7K87	+/-1%	120-7874
19	18	R2,R3,R11,R12,R19,R20, R26,R27,R35,R36,R43,R44, R51,R52,R53,R54,R57,R58	Resistor, Tin-Oxide Film	10K0	+/-1%	120-1005
20	2	R4,R28	Resistor, Tin-Oxide Film	200K	+/-1%	120-2006
21	2	R5,R29	Resistor, Tin-Oxide Film	3K01	+/-1%	120-3014
22	6	R6,R7,R8,R30,R31,R32	Resistor, Tin-Oxide Film	56R2	+/-1%	120-2522
23	4	R9,R10,R33,R34	Resistor, Tin-Oxide Film	4K99	+/-1%	120-4994
24	2	R13,R37	Resistor, Tin-Oxide Film	442R	+/-1%	120-4423
24	2	R14,R38	Resistor, Tin-Oxide Film	4K64	+/-1%	120-4644
25	2	R15,R39	Resistor, Tin-Oxide Film	4K12	+/-1%	120-4124
26	2	R16,R40	Resistor, Tin-Oxide Film	5M6	+/-5%	070-5607
27	2	R17,R62	Resistor, Tin-Oxide Film	1M00	+/-1%	120-1007
28	4	R18,R63,R65,R67	Resistor, Tin-Oxide Film	15K0	+/-1%	120-1505
29	2	R21,R45	Resistor, Tin-Oxide Film	150K	+/-1%	120-1506
30	2	R22,R46	Resistor, Tin-Oxide Film	6K65	+/-1%	120-6654
31	2	R23,R47	Resistor, Tin-Oxide Film	1K82	+/-1%	120-1824
32	1	R24	Resistor, Tin-Oxide Film	22K1	+/-1%	120-2215
33	4	R41,R61,R66,R68	Resistor, Tin-Oxide Film	7K50	+/-1%	120-7504
34	2	R42,R64	Resistor, Tin-Oxide Film	150R	+/-1%	120-1503
35	1	R48	Resistor, Tin-Oxide Film	4K53	+/-1%	120-4534
36	4	R49,R50,R55,R56	Resistor, Tin-Oxide Film	20K0	+/-1%	120-2005
37	2	R59,R60	Resistor, Tin-Oxide Film	2K00	+/-1%	120-6815
38	7	U1,U3,U5,U7,U9,U11,U15	IC, Op Amp, Bipolar, Dual, 5532			490-0300
39	2	U2,U8	Voltage-Controlled Atten.,VCA1001			480-0900
40	4	U4,U10,U13,U14	IC, Op Amp, JFET, Dual, LF353			490-0140
41	2	U6,U12	Comparator, Bipolar, Dual, LM393			480-0240
42	2	VR1,VR4	Trimpot, PT10H	5K/1T		420-0450
43	2	VR2,VR5	Trimpot, PT10H	100R/1T		420-0160
44	2	VR3,VR6	Trimpot, PT10H	1K/1T		420-0200
45	1	VR7	Trimpot, PT10H	10K/1T		420-0240
46	1	None	PC Board, 100% Tested			770-4340
47	2	None	Heat Spreader for U4 & U8			040-0040
48	13	None	IC Socket, 8-Pin			310-0020
49	2	None	IC Socket, 18-Pin			310-0080
50	4	TP1, TP2, TP3, TP4	Test Point			840-0720

I/O Filter Board

Assembly 050-3100

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	12	C1,C3,C5,C7,C9,C10,C11, C12,C13,C15,C17,C18	Cap, Ceramic Disk, NP0	100pF	+/-5%	230-0320
2	22	C2,C4,C6,C8,C14,C16,C19, C20,C21,C22,C23,C24,C25, C26,C27,C28,C29,C30,C31, C32,C33,C34	Cap, Monolithic Ceramic, NP0	470pF	+/-5%	235-0140
3	4	C35,C36,C37,C38	Cap, Ceramic Disk, NP0	47pF	+/-5%	230-0040
4	1	C39	Cap, Ceramic Disk, Z5U	0.10mF	+/-20%	230-0140
5	1	D1	Diode, 1N4003			470-0120

6	6	FL1,FL2,FL3,FL6,FL7,FL8	RFI Filter, TDK ZY51R5-2P			360-0120
7	4	FL4,FL5,FL9,FL10	RFI Filter, TDK ZJSR 5101-271TA			360-0150
8	1	H1	Male 2500 (3M P/N 2526-6002-UG)			290-0800
9	5	J1,J2,J5,J6,J9	Conn., XLR, Neutrik NC3FK-V			310-2920
10	5	J3,J4,J7,J8,J10	Conn., XLR, Neutrik NC3MK-V			310-2900
11	4	K1,K2,K3,K4	Relay, DPDT, 12V, TQ2E-12V			630-0100
12	6	L1,L3,L5,L7,L13,L15	Inductor	220mH	+/-10%	360-0080
13	6	L2,L4,L6,L8,L14,L16	Inductor	1.0mH	+/-10%	360-0060
14	6	L9,L10,L11,L12,L17,L18	Inductor	47mH	+/-10%	360-0100
15	6	R1,R2,R3,R4,R9,R10	Resistor, Tin-Oxide Film	1K00	+/-1%	120-1004
16	6	R5,R6,R7,R8,R11,R12	Resistor, Tin-Oxide Film	150R	+/-1%	120-1503
17	2	R13,R14	Resistor, Tin-Oxide Film	110R	+/-1%	120-1103
18	2	T1,T2	Xfmr., Pulse Eng. P/N PE65612			380-0750
19	1	None	PC Board, 100% Tested			770-4320

15KHz Lowpass Filter
Assembly 050-3280

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	3	C1, C12, C14	Capacitor Poly	1000pF	+/-2.5%	190-1540
2	5	C2, C5, C10, C13, C20	Capacitor Poly	330pF	+/-2.5%	190-1440
3	4	C3, C4, C15, C19	Capacitor Poly	.01uF	+/-2.5%	190-1380
4	3	C6, C7, C22	Capacitor Poly	3300pF	+/-2.5%	190-1460
5	1	C8	Capacitor Poly	100pF	+/-2.5%	190-1480
6	2	C9, C17	Capacitor Poly	6800pF	+/-2.5%	190-1500
7	1	C11	Capacitor Poly	4700pF	+/-2.5%	190-1520
8	2	C16, C21	Capacitor Poly	1500pF	+/-2.5%	190-1420
9	1	C18	Capacitor Poly	2200pF	+/-2.5%	190-1560
10	6	C23, C24, C25, C26, C27, C28	Capacitor Mono	0.1uF	+/-20%	235-0020
11	6	L1, L2, L3, L4, L5, L6	Inductor, CLNS-T033Z, 10PA Type	18mH		360-0200
12	1	R1	ResistorTin-Oxide Film	232K	+/-1%	120-2326
13	6	R2, R4, R6, R7, R8, R10, R11	ResistorTin-Oxide Film	10K0	+/-1%	120-1005
14	1	R3	ResistorTin-Oxide Film	42K2	+/-1%	120-4725
15	1	R5	ResistorTin-Oxide Film	47K5	+/-1%	120-4755
16	1	R9	ResistorTin-Oxide Film	15K8	+/-1%	120-1585
17	6	LD1, LD2, LD3, LD4, LD5, LD6	LED, LTL1234N GRN			530-0260
18	3	VR1, VR2, VR5	Potentiometer, PT6KV, 1T, Piher	2K5		420-0540
19	1	VR3	Potentiometer, PT6KV, 1T, Piher	1K0		420-0520
20	3	VR4, VR6, VR7	Potentiometer, PT6KV, 1T, Piher	5K0		420-0420
21	1		PCB, 15 kHZ Low Pass Filter, 100%			770-1980
22	5	U1, U2, U3, U4, U5	IC, Op Amp, 5532			490-0300
23	5		Socket, 8 Pin DIP, LTO-308-TA			310-0020
24	3		Test Point, TP-101-20			840-0720
25	7	JU1, JU2,JU3,JU4,JU5,JU6,JU7	Shunt, Test, TT-02			840-1080
26	14		Header, 36P, M740-240-400			310-0360
27	1		Schematic, FM PROC 15K Filter			060-0002

Front Panel Display Board
Assembly 050-3200

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	3	C1,C2,C3	Cap, Monolithic Ceramic, Z5U	0.1mF	+/-20%	235-0020
2	8	C4,C5,C6,C7,C8,C9, C10,C11	Cap, Ceramic Disk	0.1mF	+/-20%	230-0140
3	2	DS1,DS2	LED Bar, 10-segment isol. Yel			530-0550
4	2	DS3,DS8	LED Bar, 10-segment isol. Red			530-0600
5	4	DS4,DS5,DS6,DS7	LED Bar, 10-segment isol. Grn			530-0500
6	1	H1	Header 12 (Make from CA-F36-23B-5			310-3140
7	3	H2,H3,H4	Header 10 (Make from CA-F36-23B-5			310-3140
8	21	LD1,LD2,LD3,LD4,LD5,LD6, LD7,LD8,LD9,LD10,LD11, LD12,LD13,LD14,LD15,LD16, LD17,LD18,LD19,LD20,LD21	LED, Red			530-0280
9	1	Q1	Transistor, NPN SI, 25D600			500-0360
10	2	Q2, Q3	Transistor, NPN SI, 2N3904			500-0260
11	3	R1, R12, R13	Resistor, Tin-Oxide Film	10K0		120-1005
12	1	R2	Resistor, Tin-Oxide Film	100R		120-1003
13	1	R3	Resistor, Tin-Oxide Film	330R		120-3303
14	2	R4, R5	Resistor, Tin-Oxide Film	2K49		120-2494

15	4	R6, R7, R8, R9	Resistor, Tin-Oxide Film	1K82		120-1824
16	2	R10, R11	Resistor, Tin-Oxide Film	1K0		120-1004
17	2	RN1,RN3	Resistor Network, SIP	10KC6		140-0600
18	4	RN5,RN6,RN7,RN8	Resistor Network, SIP	330C6		140-0140
19	4	RN9	Resistor Network, SIP	10KC10		140-0260
20	8	SW1,SW2,SW3,SW4,SW5, SW6,SW7,SW8	ITW			250-0520
21	2	U1,U5	LM3915			480-0600
22	6	U2,U3,U4,U6,U7,U8	LM3914			480-0360
23	4	U9,U10,U11,U12	74HCT04			480-0940
24	1	None	PC Board, 100% Tested			770-4000
25	8	None	Socket, 20-pin X 0.3”			310-3200
26	8	None	Socket, 18-pin			310-0080
27	4	None	Socket, 14-pin			310-0040

Computer Board
Assembly 050-3160

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	2	C1,C2	Cap, Mono. Ceramic, NP0, .1” LS	22pF	+/-5%	230-0540
2	1	C3	Cap, Polyester	0.047mF	+/-5%	190-1640
3	1	C4	100uf	100mF	+/-20%	200-0260
4	13	C5,C6,C7,C8,C9,C10,C11, C12,C13,C14,C15,C16,C17	Cap, Monolithic Ceramic, Z5U	0.1mF	+/-20%	235-0020
5	1	ENCODER	HP P/N HRPG-AD32 #14R			250-0700
6	1	H1	Socket 12 (Make from 310-0400)			310-3100
7	3	H2,H3,H4	Socket 10 (Make from 310-0400)			310-3100
8	1	H5	Socket 18 (Make from 310-0400)			310-3100
9	1	H6	“RIBBON 50” Assembly***			030-4200
10	1	H7	“RIBBON 16” Assembly***			030-4180
11	1	H8	“RIBBON10” Assembly			030-4100
12	1	H9	“RIBBON20” Assembly			030-4120
13	1	H10	MOLEX9M			310-0680
14	1	H11	Socket 3 (Make from 310-0400)			310-0400
15	2	Q1,Q2	Transistor, PNP Silicon, 2N3906			500-0180
16	1	R1	Resistor, Tin-Oxide Film	10K0	+/-1%	120-1005
17	1	R2	Resistor, Tin-Oxide Film	2K49	+/-1%	120-2494
18	1	R3	Resistor, Tin-Oxide Film	15R0	+/-1%	120-1502
19	1	R4	Resistor, Tin-Oxide Film	1R50	+/-1%	120-1501
20	1	R5	Resistor, Tin-Oxide Film	249R	+/-1%	120-2493
21	1	RN1	Resistor network	470RI6	+/-2%	140-1480
22	1	U1	IC, Watchdog, Dallas DS1232			480-1760
23	1	U2	IC, mP, Philips P80C31EBPN			480-1820
24	1	U3	IC, CMOS, 74HCT373			480-1020
25	1	U4	IC, EPROM, 27C256			480-1840
26	1	U5	IC, EPROM, FLASH, 29C256			480-1860
27	1	U6 (Note speed = 150 nS)	IC, CMOS, SRAM, 6264			480-0440
28	1	U7	IC, RTC, Dallas Semi DS12887			480-1780
29	2	U8,U9	IC, CMOS, 74HCT138			480-0980
30	1	U10	IC, CMOS, 74HCT02			480-0920
31	1	U11	IC, CMOS, 74HCT08			480-0950
32	1	U12	IC, CMOS, 74HCT32			480-1080
33	1	U13	IC, CMOS, 74HCT393			480-1050
34	1	U14	IC, Bipolar, SP232CP (Sipex)			480-1800
35	5	U15,U16,U17,U18,U19	IC, CMOS, 74HCT374			480-1040
36	3	U20,U21, U22	IC, CMOS, 74HCT244			480-1070
37	1	U24	IC, A/D, National ADC0817CCN			480-1880
38	1	U23	IC, Bipolar, Timer, LM555			480-0200
39	1	VR1	Trimpot, PT1H	10K/1T		420-0240
40	1	Y1	Crystal, HC49S, 11.0592MHz		**	440-0120
41	1	None	PC Board, 100% Tested			770-4020
42	2	None	IC Socket, 8-pin			310-0020
43	4	None	IC Socket, 14-pin			310-0040
44	3	None	IC Socket, 16-pin			310-0060
45	9	None	IC Socket, 20-pin			310-0220
46	1	None	IC Socket, 28-pin X .3” spacing			310-3060
47	2	None	IC Socket, 28-pin X .6” spacing			310-1480
48	2	None	IC Socket, 40-pin			310-1300

49	1	H12	Header, 3-pin RA***			310-0440
50	1	None	Jumper, shorting			840-1080

BNC Filter
Assembly 050-3220

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	2	C1,C2	Capacitor, Ceramic Disk, NP0	47pF	+/-5%	230-0040
2	1	FL1	Filter, TDK P/N ZJY51R5-2P			360-0120
3	1	J1	BNC Jack			310-1000
4	1	R1	Resistor, Tin-Oxide Film	100R	+/-1%	120-1003
5	1	VR1	Trim 70Y (See Note 1)	1K, 18T		420-0930
6	1	None	PC Board, 100% Tested			770-4360

PPDM Stereo Generator Board
Assembly 050-3340

Item	Quan	Reference Designators	Description	Value	Tol.	Aphex P/N
1	1	C1	75pF	75pF	+/-5%	160-0200
2	1	C10	47pF	47pF	+/-5%	160-0100
3	1	C2	Cap, Tant, 35V	1uF		220-0020
4	1	C3	Cap, Polyester, 0.3” L.S.	0.1uf		190-0840
5	1	C4	Cap, Polyester, 0.3” L.S.	1500pF	+/-5%	190-0080
6	4	C5,C6,C7,C8	Cap, Mica	50pF	+/-5%	160-0420
7	3	C9,C20,C23	Cap, Mica	5pF		160-0320
8	2	C19,C22	Cap, Mica	10pF	+/-5%	160-0020
9	2	C21,C24	Cap, Electrolytic, Non-polar, 35V	22uF		225-0020
10	22	C25,C26,C27,C28,C29,C30, C37,C38,C39,C40,C41,C42, C43,C44,C45,C47,C48,C49	Cap, Ceramic, Z5U, Monolithic	0.1uF	+/-20%	235-0020
11	15	D1,D2,D3,D4,D5,D6,D7,D8, D9,D10,D12,D13,D14,D15, D18	Diode, 1N914B			470-0040
12	1	H1	Cable Assembly, 4”			030-4100
13	1	H2	Header, 4-pin, .1” spacing			310-1040
14	3	H3,H4,H5	Header, 2-pin, 0.1” spacing			310-0760
15	1	K1	Relay, Aromat TQ2E-12V			630-0100
16	1	Q1	Transistor, SI NPN, 2N3904			500-0260
17	1	R1	Resistor, Tin-Oxide Film	2M2	+/-1%	070-2207
18	6	R2,R11,R37,R38,R39,R40	Resistor, Tin-Oxide Film	100K	+/-1%	120-1006
19	1	R3	Resistor, Tin-Oxide Film	10K0	+/-1%	120-1005
20	1	R4	Resistor, Tin-Oxide Film	15K0	+/-1%	120-1505
21	1	R7	Resistor, Tin-Oxide Film	270R	+/-1%	070-2703
22	2	R8,R12	Resistor, Tin-Oxide Film	1K00	+/-1%	120-1004
23	4	RN5C,RN5D,RN6C,RN6D	Resistor, Tin-Oxide Film (Stand up on RNET pattern)	10K0	+/-1%	120-1005
24	1	R14	Resistor, Tin-Oxide Film	10R0	+/-1%	120-1002
25	7	R9,R10,R31,R32,R33,R34, R47	Resistor, Tin-Oxide Film	10K0	+/-1%	120-1005
26	2	R35,R36	Resistor, Tin-Oxide Film	22K1	+/-1%	120-2215
27	1	R41	Resistor, Tin-Oxide Film	150K	+/-1%	120-1506
28	1	R42	Resistor, Tin-Oxide Film	49K9	+/-1%	120-4995
29	2	R43,R44	Resistor, Tin-Oxide Film	1K10	+/-1%	120-1104
30	1	R45	Resistor, Tin-Oxide Film	2K49	+/-1%	120-2494
31	1	R46	Resistor, Tin-Oxide Film	332R	+/-1%	120-3323
32	2	RN1,RN8	Resistor network, 4-pin common	10K	+/-2%	140-1400
33	1	RN3	Resistor network, 8-pin isolated	3K3	+/-2%	140-1040
34	1	RN4	Resistor network, 6-pin common	10K	+/-2%	140-0600
35	4	RN5A,RN5B,RN6A,RN6B	Resistor, Tin-Oxide Film (Stand up on RNET pattern)	47K5	+/-1%	120-4755
36	1	U1	IC, CMOS, Quad And, 74HCT00			480-1120
37	1	U2	IC, CMOS, Ripple Counter, 74HC404			480-1160
38	1	U3	2764 EPROM w/ pilot software			480-0160B
39	1	U4	2764 EPROM w/ PPDM software			480-0160A
40	3	U5,U6,U7	74HCT374			480-1040
41	5	U8,U9,U10,U11,U12	IC, Maxim AD7524JN			480-0520
42	2	U13,U14	IC, op amp, bipolar, dual, AD812AN			560-0090
43	1	U15	IC, op amp, bipolar, dual, NE5532			490-0300
44	1	U16	IC, op amp, bipolar, AD829JN			490-0080

45	1	U17	IC, Analog Buffer, LM6321N		480-1260
46	4	U18,U25,U28,U30	IC, op amp, bi-fet, dual, LF353		490-0140
47	1	U22	IC, inverter, hex, CMOS, 4069		460-0120
48	4	U23,U24,U26,U27	IC, analog switch, quad, 4016		460-0060
49	1	VC1	Variable Capacitor, trimmer	7-35pF	170-0040
50	1	VR1	Trimpot, R/A Adj.	10K	420-0980
51	1	VR2	Trimpot, PT10V	10K/1T	420-0020
52	4	VR3,VR4,VR5,VR6	Trimpot, Multi-Turn, Top Adj.	10K/10T	420-0040
53	1	X1	Crystal, quartz	4.864 MHz	440-0080
54	1	None	PC Board, 100% tested		770-4420
55	2	None	Socket, IC, 28-pin, 0.6” wide		310-1480
56	9	None	Socket, IC, 8-pin		310-0020
57	3	None	Socket, IC, 20-pin, 0.3” wide		310-0220
58	6	None	Socket, IC, 16-pin		310-0060
59	6	None	Socket, IC, 14-pin		310-0040

Master Assembly
Assembly 020-3000

Item	Quan	Ref. Des.	Description	Aphex P/N
1	1	None	Front Panel Assembly	060-0020
2	1	None	Bracket Assembly, Card Guide	060-0080
3	1	None	Chassis Assembly	060-0060
4	1	None	Fan	600-0020
5	4	None	Screw, #6-32 X 5/8” Ph F/H, Black	795-1660
6	4	None	Nut, #6-32 KEPS	780-0120
7	1	None	Rear Panel Assembly	060-0040
8	1	None	BNC Filter Assembly	060-0100
9	2	None	Screw, #4-40 X 3/8” Ph Pan	795-1600
10	1	None	Cover, Metal, anodize and chem-film	675-1850
11	1	None	Processing: paint, Cover	040-3800
12	1	None	Foam Rubber UL94V-1 card retainer strip	
13	1	None	Adhesive	
14	2	None	Screw, #4-40 X 1/4” F/H, S/S, 100 Deg.	795-1340
15	6	None	Screw, #6-32 X 1/4” Ph Pan, Black	795-1040
16	3	None	Screw, #6-32 X 3/8” Ph Oval [1]	795-0600
17	10	None	Screw, #4-40 X 1/4” Ph Pan (050-3360 mtg.)	795-0260
18	5	None	Standoff, #4-40 X 7/16” X 1/4” Hex (050-3360 mtg.)	790-1400
19	2	None	Handle	840-2350
20	4	None	Screw, #6-32 X 1/2” Ph Pan	795-0920
21	4	None	Washer, split ring, #6	800-0600
21	6	None	Screw, #6-32 X 3/8” Ph Oval 1	795-0600
22	1	None	Screw, #8-32 X 3/4” Ph Pan 2	795-1380
23	1	None	Nut, #8-32 KEPS 2	780-0260
24	1	None	Nut, #8-32 hex 2	780-0060
25	2	None	Washer, #8 star, external tooth 2	800-0220
26	2	None	Washer, flat, #8 2	800-0460
27	5	None	Screw, #4-40 X 1/2” Ph Pan, zinc 3	795-0060
28	14	None	Screw, #4-40 X 1/4” Ph Pan, black 4	795-0260
29	*4*	*None*	*Screw, #6-32 X 1/4” Ph F/H, 100 deg. 5*	*795-1320*
30	6	None	Standoff, #4-40 X 7/16” X 1/4” Hex 6	790-1400
31	12	None	Screw, #4-40 X 1/4” Ph Pan, black 6	795-0260

end

17.0 Warranty & Service

17.1 Obtaining Service For The FM Pro

Aphex Systems supports its customers with spare parts and technical assistance. You may contact us by phone, fax, and the Internet. Out-of-warranty repair work should be performed only by qualified service personnel. We highly recommend using the factory or other authorized service agencies to obtain all repair work.

Units may not be shipped to Aphex for service without first obtaining an RMA (returned material authorization). Equipment received without an RMA may be refused for delivery and returned to the sender. Contact Aphex customer support for an RMA. The RMA number must be placed on the outside of the shipping carton to identify the unit. Please also include within the container a brief letter describing the defect or the problem's symptoms, your name and return shipping address, and contact telephone number for you or someone else who is familiar with the equipment problem.

You may contact Aphex customer support through:

Telephone 1-818-767-2929
Fax: 1-818-767-2641
Internet: techsup@aphex.com

Outside the USA, contact your local authorized Aphex distributor or dealer for service. You can find the appropriate world-wide service agencies by contacting Aphex Systems by phone, fax, or on the Internet.

17.2 Warranty Claims

All warranty claims must be presented to the Aphex factory customer support department or to an authorized dealer, distribtor, or agency for processing. Aphex does not honor unauthorized repairs under warranty claims. Unauthorized repairs and modifications to the unit may void the warranty at the sole discretion of Aphex Systems.

Warranty claims will be validated by the unit serial number and the purchase date. Generally, an owner registration form mailed to Aphex shortly after the date of purchase satisfies the requirements for warranty validation. In any case of doubt or the absence of a valid owner registration on file at Aphex, you may be required to furnish proof of the purchase date or proof of ownership to obtain warranty service. Units obtained through fraudulent means such as units known or suspected to be stolen goods will not be honored under warranty claims.

Factory supplied field upgrade kits installed by the customer will be honored under warranty if all installation procedures were properly followed using adequate care and workmanship. Damage caused by careless or unskilled workmanship or accident is the full responsibility of the owner and may void the warranty which is solely at the discretion of Aphex Systems.

17.3 What Is Covered

Refer to the warranty certificate for further details.

Aphex Systems Ltd.

Limited Warranty

PERIOD
One year from date of purchase

SCOPE
All defects in workmanship and materials. The following are not covered:
a. Voltage conversions
b. Units on which the serial number has been defaced, modified,or removed
c. Damage or deterioration:
1. Resulting from installation and/or removal of the unit.
2. Resulting from accident, misuse, abuse, neglect, unauthorized product modification or failure to follow instructions contained in the User's Manual.
3. Resulting from repair or attempted repair by anyone not authorized by Aphex Systems.
4. Occurring from shipping (claims must be presented to shipper).

WHO IS PROTECTED
This warranty will be enforceable by the original purchaser and by any subsequent owner(s) during the warranty period, so long as a copy of the original Bill of Sale is submitted whenever warranty service is required.

WHAT WE WILL PAY FOR
We will pay for all labor and material expenses for covered items. We will pay return shipping charges if the repairs are covered by the warranty.

LIMITATION OF WARRANTY
No warranty is made, either expressed or implied, as to the merchantability and fitness for any particular purpose. Any and all warranties are limited to the duration of the warranty stated above.

EXCLUSION OF CERTAIN DAMAGES
Aphex Systems' liability for any defective unit is limited to the repair or replacement of said unit, at our option, and shall not include damages of any other kind, whether incidental, consequential, or otherwise.

Some States do not allow limitations on how long an implied warranty lasts and/or do not allow the exclusion or limitation of incidental or consequential damages, so the above limitations and exclusions may not apply to you.

This warranty gives you specific legal rights, and you may also have other rights which vary from State to State.

18.0 Web Broadcasting Applications

18.1 Why process the Audio?

Listenability

· Consistent Level
· Consistent Tonal Balance
· Less Distortion
· Improved Intelligibility
· Less Listener Fatigue
· Longer Listener Retention

Designed Sound

· Adjustable Peak and Average Program Density
· Adjustable Dynamic Equalization
· Adaptive Audio Processing
· Distinctive Dial Presence
· Professional Quality Finish
· Competitive Impressiveness
· Faster Audience Capture
· Higher Audience Loyalty

The Competitive Edge

· Faster Audience Capture
· Longer Listener Retention
· Higher Audience Loyalty
· Higher Hit Ratings
· Stronger Market Position
· Greater Sales Volume
· Longer Client Retention
· Greater Profitability
and…
· A Greater Assurance of "Mission Accomplished"

18.2 Basics of Processing

18.2.1 Leveling

Audio programs contain a wide range of levels over time. It is usually desirable to pull all the program elements, voices, music, commercials, etc., together toward a consistent sound level. This makes listening to the program much more enjoyable without the need to constantly readjust the volume control. Bringing together the sound levels of a program is called "Leveling".

18.2.2 Peak Limiting

The peaks of a program don't necessarily relate directly to the sound level as our ears detect it. If the leveling was done to satisfy the ear, then there remains a great deal of peak variability in the audio stream. This can cause problems in trying to get a full level of performance from the streaming audio encoder. It is therefore desirable, in addition to leveling the program, to cause the peak audio levels to be limited to a maximum value, and preferably at relatively lower values than they originally occur. This is because the analog to digital conversion and the subsequent digital data compression algorithms all relate to the peak level of the audio stream. If the audio peaks are processed to be consistent and predictable in level, then the streaming audio encoder can be driven to full scale encoding without ever having an overload occur. There will be no need to allow excessive headroom for occasional high peak levels. This results in better sound encoding at all bit rates.

18.2.3 Designing the Sound

Beyond the technical reasons for using audio processing, there is the notion of aesthetics. Some may relate this to "competitiveness". The worldwide web has an ever-growing abundance of streaming audio sources to choose from. With such a great amount of competition between webcasters, it can be important to "stand out on the dial". You may want to design your sound to be something different from all the others, and hopefully sound impressive to listeners. Unfortunately, CD's and talk played unprocessed and raw may sound great at the time they are recorded, but usually sound low in level and unprofessional when broadcast over the internet media. With good audio processing you can obtain a professional finish that listeners will appreciate, leading to greater audience loyalty.

18.3 Getting the Best Processing

You could piece together a combination of various compressors, limiters and equalizers in an effort to build a complete audio processing system. This is seldom satisfactory, since the equipment that is designed for general studio applications is not specialized for broadcast processing. You will not be able to keep peaks under control without creating a great deal of sonic artifacts such as pumping and hole punching. The Model 2020 melds many specially designed audio processing techniques, including at least 11 that are patented exclusively by Aphex, into one unit. The entire complex processing chain works interactively and harmoniously, specifically avoiding all the sonic aberrations caused by conventional audio processors.

18.4 Digital Versus Analog Processing

The output media of web broadcasting is of a digital nature. Because of that, some people think it makes the most sense to process the sound only in the digital domain. Why, then, did we build the Model 2020 in the analog domain? The fact is that digital audio processing is very limited in it's capabilities, especially sound quality. This is because digital audio is time sampled and quantized. Digital audio processors have to contend with many, many approximations within their mathematical algorithms and inevitably generate digital distortions from truncation, rounding, and aliasing. That is why digitally processed sound is edgy and dry. It may have an impressive zippy or splashy quality at first listen, but that effect quickly becomes irritating. Analog processing, especially with the advanced circuits of the Model 2020, is inherently free of these aberrations. To provide the means to interface with a digital audio system, we included an optional 20-Bit AES/EBU input/output module. The digital input is converted to analog, processed, then converted back to digital. Any small quality loss that may be experienced through the converters is overshadowed by the greatly improved sound quality of the analog processing.

18.5 How to Use the Model 2020

Generally, you will interface the Model 2020 between the audio source and the streaming media encoder. When the encoder is an external hardware box, that becomes relatively easy because you can gain direct access to the encoder's analog or digital audio input. However, if the encoder is in software, then you will need to interface with the computer's audio board input cabling. The idea is to get as close as possible to the encoder's input point.

18.5.1 Set the Input Level

Once the audio source is fed into the Model 2020, you need to normalize the input gain. Set the Input Gain so the input level meters are peaking at 0 for a nominal program level. You operate the input gain through the Processing Input/Output menu page.

18.5.2 Set the Output Level

If you are using the digital output, then you don't need to worry about this. The Model 2020 automatically normalizes the output peaks to 1dB under digital full scale. If you are using the analog outputs, then you need to set the output level to properly drive your encoder's input. You operate the output gain also through the Processing Input/Output menu page. Your encoder may have an input meter to indicate the proper drive level, but if not, set the output to the highest level that still gives undistorted sound when listening to the stream through your reference media player.

18.5.3 Set Up the Processing

Now is the time to go to the chapters on setting up processing. First you will set up all the Globals and save them. Then, you will experiment with the presets. Finally, you will fine tune your own sound. This process may take anywhere from 10 minutes to a couple of hours depending upon your dedication and level of satisfaction.

18.6 Bit Rate Effects

Web audio media varies through a wide range of bit rates and consequential audio quality. Very low bit rate encoders often benefit from limiting the audio bandwidth of the input signal. In many cases, you cannot anticipate what bit rate will be served for any given stream. Nevertheless, you have only one audio processor and it needs to sound great on the highest bit rate or why bother. Therefore, it is not possible to specifically optimize the audio processing for low bit rate and high bit rate at the same time. We have found, however, that the Model 2020, if adjusted to sound great on a high rate stream such as 22KB, it also sounds quite good at low rates like 3KB. If you have separate high and low rate converter banks, then you can aid the low rate converter by adding a lowpass filter such as a parametric equalizer shelved off at 5-6KHz, to the converter's audio input. This may improve any splatter distortion you may be experiencing. It should be emphasized that splatter is not a problem caused by the Model 2020, rather it is a characteristic generally inherent in all low bit rate converters.

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