WARNINGS

DON'T EXPOSE THE UNIT TO RAIN OR MOISTURE OF ANY KIND.

DON'T USE ANY POWER CABLE OTHER THAN A STANDARD IEC LINE CORD SUCH AS THE ONE SUPPLIED WITH YOUR SP2016.

DON'T PLUG YOUR UNIT INTO ANYTHING BUT A PROPERLY-GROUNDED THREE-PRONG OUTLET.

DON'T RELY SOLELY ON THE FRONT SUPPORT SCREWS WHEN RACK-MOUNTING; SUPPORT THE BACK
END OF THE BOX, TOO.

DON'T BLOCK THE TOP OR REAR VENTILATION; LEAVE AT LEAST AN INCH OF CLEAR SPACE ON SIDES
TO MAKE SURE THERE'S NO CHANCE OF OVERHEATING.

WHAT IT IS

The SP2016 Effects Processor/Reverb is something special in the way of studio gear. If you've never worked with one before, your ears are in for some very pleasant surprises.

What makes the SP2016 so unusual is that (unlike most products in this specialized age) it isn't just one thing. Not simply a reverb, or a digital delay, or an equalizer . . . but all of these things, and more. We call it a digital EFFECTS PROCESSOR/REVERB because that's precisely what it is: a black box with a whopping huge capacity for "number-crunching" built into it. The SP2016 takes your analog audio signal, converts the signal into digital information, processes it, and finally sends the data back out as a (considerably) enhanced audio signal. As for just what that computing does . . . Well! That's up to you. We've programmed in a wide variety of useful functions—delays, reverbs, digital filtering and EQ, gain control, chorusing, flanging, signal analysis, even self-diagnostic tests and synthesis—and left plenty of memory space for you to create and store your own variations, for instant recall when needed.

Our goal in designing the SP2016 was to provide you with the most powerful, most flexible, most expandable nineteen inches of audio magic you've ever packed into your rack.

PLEASE NOTE: *This manual was designed to teach you how to operate the SP2016 to its fullest. Please take it in order, and don't skip over any of the training examples. Otherwise you'll miss out on some of the fun!*

TABLE OF CONTENTS

part five: THE FACTORY PRESETS

part six: EXPANDING THE SYSTEM

appendices

part one: SETUP

INSTALLING THE SP2016

This is easy to do; only six brief steps are involved.

1) Check the unit's voltage setting to make certain it matches your local power. (We'll show you how to do this immediately after running through the list.)

2) Connect the power cord supplied with the unit to the power jack on the rear panel.

3) Make certain the six INPUT and OUTPUT section sliders on the front panel are all the way down.

4) Connect the correct XLR cables (from your mixing board, patchbay, or signal source) to the INPUT and OUTPUT jacks on the rear panel. Always use quality cables—otherwise you won't be hearing everything the SP2016 is giving you.

5) Put the unit in place. If you rackmount it, make sure you A) support the back end, too, and B) leave at least an inch of clear ventilation space above and behind.

6) Plug the power cord into a properly grounded three-prong outlet.

Voila! Setup complete. Next you will check the installation, turn the unit on, and adjust your audio levels. But first...

SETTING VOLTAGE

IMPORTANT!

Next to where you plugged the power cord into the unit is a plastic window. (While the cord is plugged in, the window closes and cannot be opened until the power cord is removed.) Inside that window is a power fuse and a small plug-in circuit card. Should the fuse ever blow, you can replace it easily, using the built-in "*fuse pull*" lever, but that isn't what's important here. *What's important is the number you see on the circuit card beneath the fuse.*

* If you are operating your SP2016 with voltage that is 110 to 120 VAC, and 50/60 Hz, that number must read "120."

* If your voltage is 220 to 240 VAC, and 50/60 Hz, then the number must read "220."

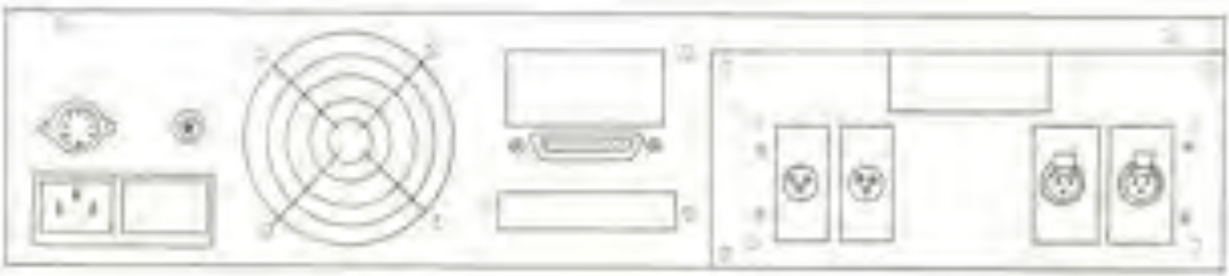
THIS IS CRITICAL. Operating the unit with the wrong setting could do it Extreme and Serious Harm. This you do *not* want. If you don't know what your voltage is, FIND OUT ... and make certain card and current match up before you turn the unit on. Protect your investment.

If the wrong number is visible, carefully pry out the card (using a pencil or small screwdriver placed through the hole in the card). Then reverse it so the proper number becomes visible and put the card back in place.

THE REAR PANEL

Before going on, you should take a quick look at the back of the unit. From left to right, you'll see the following connections:

- a REMOTE CONTROL plug and a CONTROL VOLTAGE/TRIGGER jack (both of which are covered in detail in part six);
- two male XLR OUTPUT jacks;
- two female XLR INPUT jacks;
- an optional IEEE-488 GENERAL PURPOSE INTERFACE, for connecting the SP2016 to a computer (also covered in part six).



Hooking your INPUTS and OUTPUTS up properly is a must. Here are the specs you should go by:

INPUT: the input will handle −10 dbm to +24 dbm. You should have no trouble connecting these jacks directly to line level signals. But they do have an impedance of 10K Ohms, so you'll have to use a preamp, direct box, or buffer if you want to run direct input from a guitar (or any similar high impedance, low level signal source).

OUTPUT: maximum output is +12 dbm, electronically balanced, from 150 Ohms. It's fine for driving 600 Ohm (or even higher) line-level load impedances.

AVOIDING HUM AND INTERFERENCE

Simple: Use high quality audio cables and stick to standard grounding procedures.

If you do both these things but still hear line noise when the unit is on, check to make certain you haven't got your unit, cables, or direct boxes (if you are using any) too near a power supply. Some power supplies aren't as well-shielded as others, and their transformers can induce hum.

TESTING THE INSTALLATION

When the SP2016 isn't on, the input signal bypasses the signal processor electronics entirely. In other words, what goes in goes straight through to the OUTPUTs, untouched.

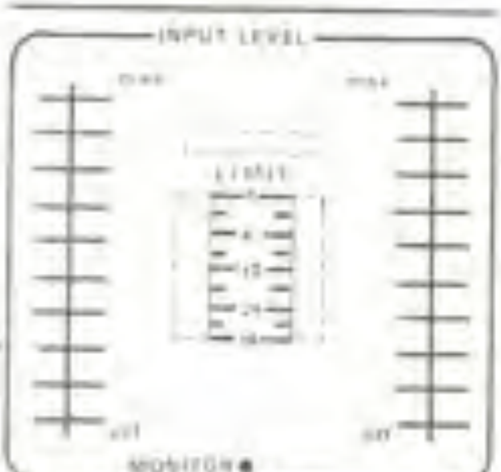
This allows you to quickly confirm that the unit has been installed properly.

With the SP2016 *off*, send a signal into INPUT 1. Check to make sure you are getting the same signal from OUTPUT 1, at the level and location (in your patch bay, mixing board, recorder, or amplifier) where you expect it. This will, of course, vary depending on how you have things wired up. Then do the same for INPUT 2 and OUTPUT 2.

If everything is okay, you're ready to move on. If not, check your connections and cables to make sure they aren't crossed or defective. If those are okay but the problems persist, odds are the source is something in the chain other than the SP2016. Because of the signal bypass, it's just an innocent bystander until its juice starts to flow.

SETTING LEVELS

Setting the audio levels is even simpler than installing the unit. This time there are only five steps:



1) Turn on the unit. (If the SELFTEST ROM is installed, wait for the 2016 to finish going through its automatic diagnostic routines. This should take about 10 seconds.)

2) Start your INPUT signal. Send it into INPUT 1, INPUT 2, or both.

3) Slowly raise the appropriate front panel INPUT sliders (1, 2, or both) and watch the red bargraphs in that part of the panel. These will flash up and down along with the intensity of the signal.

4) When the bargraph limit LED starts to flash intermittently, stop and go no higher. You'll have attained maximum signal-to-noise ratio without clipping.

(Watch for the appearance of a bright red OVERFLOW indicator above the bargraphs. This goes on when the signal is too hot for the SP2016 to successfully process. When it's on, you've got digital distortion. Drop your input levels accordingly.)

5) Once INPUT is set, adjust the appropriate OUTPUT sliders (e.g., OUTPUT LEVEL 1 for INPUT 1) so that the output signal is at the proper level for your PA or recording gear. This should be done with the mix faders set to DRY.

part two:
WHAT'S IN THE BOX

BASICS

DEFINITION OF AN SP2016: a digital signal processor with stereo inputs and outputs, tremendous flexibility, self-testing procedures, room for 11 PROGRAM ROMs (each ROM holding from 1 to 5 programs, depending on complexity), and room in battery-backed memory for up to 65 presets (program variations that you create for yourself).

That kind of power you only get by going digital.

But going digital has some side-effects, including one you should pay attention to: The interface. That is, the way the unit operates ... what controls there are, how they work, what the displays tell you. There isn't a separate knob or button for every parameter. There isn't room! So engineers get the most out of the least by creating standard ways of doing things, called OPERATING SYSTEMS, and then using only as many separate controls as they must to run the system.

A real world example: your car. It's a tremendously complicated machine, doing thousands of different things simultaneously, but you can drive anywhere by just keeping track of the operating system—a wheel, a gas pedal, a brake, and a dashboard display.

Another real world example: the SP2016. The PROCESSOR CONTROL section on the front panel has only six buttons, a slider, and a display. But when you know their operating system, you can whip through programs and parameters like a breeze.

Take the time to learn the system. If you follow things step-by-step you won't find it hard.

The rest of ***part two*** covers the basics.

Part three teaches the operating system in detail by stepping you through practical applications.

Part four shows how to create and store your own presets.

Part five covers the factory programs in depth. By that time you'll be an SP2016 expert, and ready to explore them at your own speed.

Finally, in ***part six***, we'll discuss how to expand your unit and make it even more powerful.

THE FACTORY PROGRAMS

Turn on your SP2016. When it finishes its self-test, you'll notice a bright red name glowing in the front panel display. That's one of the unit's built-in PROGRAMS. (Further proof: the red light above the PROGRAM key is glowing. When this light is steady and unblinking, the program you see in the display is the "active" one—i.e., the one you will hear.)

There are lots more programs. To see what they are, you can either:

1) Repeatedly press the PROGRAM key, which will step through the entire list, or

2) Hold down the PROGRAM key, which after a moment will automatically step through the list until you let go, or

3) Move the ADJUST/SELECT slider back and forth, which does the same (but a lot faster).

WHAT'S IN THERE, ANYWAY?

Which programs you've got will vary, depending on the SP2016 configuration you purchased. But the display will likely show you most of the following, in pretty much this order:

STEREO ROOM
ROOM REVERB
LOOP EDIT
FLANGER
ENVELOPE FLANGER
CHORUS
PLATE REVERB
MULTITAP DELAY
GENERIC REVERB
RMX SIMULATION +
HI DENSITY PLATE
TIMESCRAMBLE
BAND DELAY
MUSICAL COMBS
DUAL ROBOTS
LONG DELAY
DUAL DELAY
DUAL DIGIPLEX
LONG DIGIPLEX
LOSSLESS ROOM

When you add your own presets (there's room for 65), or new program ROMS, the list will get longer.

THE PROGRAM KEY INDICATOR LIGHT

As you moved through the list, the light above the PROGRAM key started flashing. This tells you that the program you see in the display isn't the one that's active.

HOW TO CHANGE FROM ONE PROGRAM TO ANOTHER

1) Press PROGRAM, or move the ADJUST/SELECT slider, until the display shows the program you want.

2) Press the *EXECUTE key. In about a second the red light above* PROGRAM *will stop flashing; your new choice is now active.*

PLEASE NOTE: *because of the digital nature of the controls, there is often some slight noise associated with their movement: some slight clicking, a "zipper" effect as a setting is scaled up or down ... this is normal for controls of this type and is not a defect in your SP2016.*

In typical usage you won't adjust any controls while a take or a mix is actually going down, so these noises should be no problem. Set the controls where you want, then leave them there. Routing signals to and from the SP2016 through your board's EFFECTS or ECHO SEND lets you handle levels, mix, and necessary mutes noiselessly.

THE FRONT PANEL

There are four blocks of controls on the front panel, outlined in little white round-cornered boxes: INPUT LEVEL, OUTPUT, STATUS, and PROCESSOR CONTROL. Each one of these has an important role to play in operating the SP2016.

INPUT controls the stereo signal coming into the unit.

OUTPUT controls both the volume of the stereo output and the stereo mix of dry signal and effect.

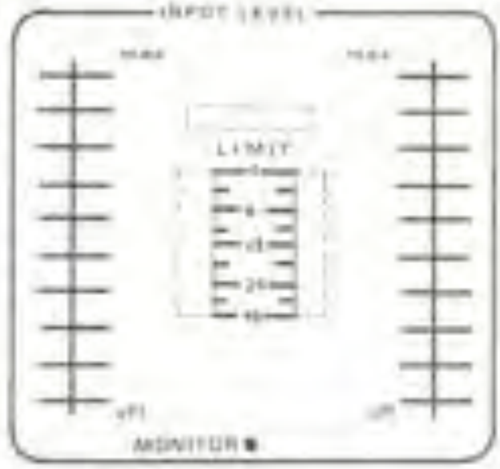
STATUS tells you the input mode and bandwidth of the active program (as well as giving valuable troubleshooting help with its problem indicators—you can't see them, but they'll light up if needed).

PROCESSOR CONTROL is the gateway to the real heart of the SP2016. It's the part we'll be spending the most time on in the rest of this section.

INPUTS

There are two input jacks, marked 1 and 2, on the rear panel. The control sliders for these inputs, also marked 1 and 2, are on the front panel. You can adjust each channel input as needed, either separately (when running two different signal sources) or together (for a matched stereo input).

The higher the slider, the hotter the input signal.



A NOTE ON THE BARGRAPHS

These two displays, one for each channel, flash along with the intensity of the input. But they aren't precision instruments. They don't have enough segments for that. Just use them as a general guide to what's going on. (NOTE: *as input volume goes down, so does bargraph accuracy.*)

THE IDEAL LEVEL SETTING

The ideal input level, for maximum signal-to-noise ratio and the best effects output, is one where the LIMIT level of the bargraph flashes occasionally and the OVERFLOW indicator just above it does not flash at all. Please note: *for optimum SP2016 performance we recommend that you set your input level as high as possible (below the level of distortion). A signal which is too low will not use the SP2016's full dynamic range.*

OVERFLOW

Above the bargraphs, where you can't see it until it flashes, is the OVERFLOW indicator. It only comes on when your input is hotter than the SP2016 can handle.

Don't worry about hurting the unit. Too high an input doesn't physically damage anything—it just makes the signal processor try to do calculations that it can't handle. The result is distortion in the output signal.

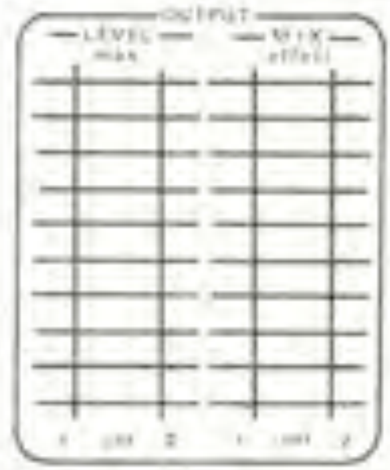
So don't OVERFLOW. It isn't like the red part of a VU meter, where tape headroom lets you get away with crossing the line. When the OVERFLOW light goes on, you've got distortion. Period.

THE MONITOR BUTTON

When you press the MONITOR button beneath the bargraphs, they switch over and show the OUTPUT levels instead of the INPUT levels. This is helpful in level matching.

After 10 seconds, the bargraphs automatically switch back. If you don't want to wait that long, press MONITOR a second time.

OUTPUTS



There are two stereo sets of sliders in the OUTPUT section. Set them to suit your mixing board or PA system's input needs and then leave them alone except for whatever minor variations are demanded as you change programs.

OUTPUT LEVEL 1 AND 2

The higher these sliders, the greater the volume of their respective outputs.

OUTPUT MIX 1 AND 2

The higher these sliders, the greater the proportion of effect signal to dry signal. All the way up gives you only the effect. All the way down gives you only the original.

If your SP2016 is connected to a mixing board, you will probably want to keep these all the way up and control the effects mix at the board itself, blending this "wet" track with a separate "dry" track fed directly from the signal source.

STATUS

INPUT MODE

This indicator shows either MONO or STEREO, depending on what program is active.

STEREO means both input signals are being accepted and processed. You have three choices—use the inputs for separate signals, for the same signal (doubled), or for two stereo halves.

MONO means the SP2016 is only processing the signal coming in over INPUT CHANNEL 1.

> PLEASE NOTE: *this indicator shows only input information, and tells you nothing about what's happening with the output. Most of the MONO INPUT programs on the SP2016, for example, actually send outputs over both channels.*

BANDWIDTH

This indicator shows either 16 kHz or 8 kHz, depending on what program is active. Bandwidth indicates how wide a range of frequencies there can be in the SP2016's output signal.

> 16 kHz means the ceiling on high frequencies is 16,000 cycles per second. This is a high-fidelity signal. Nearly all the programs operate at this bandwidth.
>
> 8 kHz means the ceiling is 8000 cycles per second. This is good but not great (though certainly suitable for vocals), and is of audibly lower quality.

OTHER INDICATORS

There are other indicators that you can't see in the STATUS section (such as LOW BATTERY) — and as long as your unit functions properly, you never will. They are all involved in identifying malfunctions and breakdowns. Consult the *Troubleshooting* appendix if you ever need more detail.

THE PROGRAM KEY

Pressing PROGRAM repeatedly, or holding it down, cycles you among the available programs.

If the PROGRAM indicator light is STEADY, it means that the program in the display is active. When it FLASHES, it means the program in the display isn't active.

THE PARAMETER KEY

Most programs have a set of PARAMETERS, sometimes many, sometimes as few as one. You reach these parameters and cycle through them by pressing the PARAMETER key.

An example:

First, make sure that STEREO ROOM is your active program. Then press the PARAMETER key. The light above that key will come on—letting you know the display is now showing you parameters, not programs—and DECAY TIME will appear above.

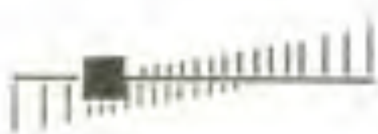
Press PARAMETER again. You'll see an F, an *, and an R (this is a FRONT/REAR display, used for controlling apparent position within an ambient "room").

Press it again and get PRE-DELAY; again for LOW FACTOR; again for LOW ROLLOFF; again for HIGH FACTOR; ...and so on, cycling through all the available parameters until you get back to DECAY TIME and start all over again.

Once you've called up a parameter, the next step is knowing how to change it.

THE ADJUST/SELECT SLIDER

Pick any program and any parameter: it doesn't matter which. Move the ADJUST/SELECT slider back and forth and watch what happens.



As before, when you used it to change programs, moving the slider alters the display. Only now we are dealing with parameters, not programs, and the slider is altering the chosen parameter's setting, from one end of that setting's range to the other.

The ADJUST/SELECT slider is the major editing tool on the SP2016.

Experiment. Fool around with changing different parameters. Call them up with the PARAMETER key and then use the ADJUST/SELECT slider to alter their settings. (Don't worry about keeping track. You aren't storing any of these changes in memory, so you can always get the original settings back by calling up the program again.)

PLEASE NOTE: *on different types of parameters, the slider works differently. This will be covered in more detail in part three.*

ALSO NOTE: *If the slider isn't having any effect on the display, move it the other way until the slider's ←|→ indicators light up and you've "locked in" to the parameter, after which it will work fine.*

THE DEFINE KEY

The DEFINE key provides steady (if low-level) help. Pressing it makes the SP2016 show you a definition or further explanation of whatever is in the display. This "help message" stays visible until you let go of the key, at which point the display returns to what it was.

Make certain the active program is STEREO ROOM and then press DEFINE. Two things happen: a bright, red triangle lights up next to DEFINE (showing showing that the SP2016 is in DEFINE mode) and the display changes to read PROGRAM NAME F08.

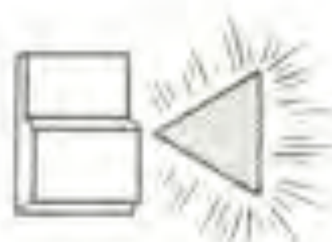
What does this tell you? First, that STEREO ROOM is a program name, instead of a parameter or some-such. Second, that its code number is F08. (There will be more about these codes and how to use them in *part four*).

DEFINE doesn't work on just the program level. Try calling up three or four different parameters, and press DEFINE for each one. It also works with COMMAND KEY and SOFTKEY displays.

PLEASE NOTE: *DEFINE is useful in general practice, but no substitute for using this manual to thoroughly learn the programs and how they work. A 16-character alphanumeric display just can't explain things clearly enough. Sometimes you might even find it confusing, or when a DEFINE message makes reference to something you don't know about. In these cases, fall back on the detailed program information in part five.*

ALSO NOTE: Pressing DEFINE and PROGRAM allows you to scroll through the programs in reverse order.

THE SOFTKEY



Nearly every program has one or more "instant features" which you can trigger at any time by pressing the SOFTKEY. For example, if STEREO ROOM is your active program and you press SOFTKEY, the inputs to the processor shut off. Press SOFTKEY again and the input comes back on.

Of course, what happens and what seems to happen can be considerably different, until you know what's going on.

If STEREO ROOM is active, here's what seems to happen on pressing SOFTKEY the first time:

1) The COMMAND indicator light comes on.

2) The words ENABLE INPUT appear in the display.

3) A red triangle next to SOFTKEY starts flashing.

What's going on? If the inputs went off—and they did—why does it say ENABLE INPUT in the display?

PLEASE NOTE: when the SOFTKEY triangle is flashing, it means exactly the opposite of what a flashing PROGRAM indicator means—i.e., the thing you see in the display is now available and will be happen the next time you press the SOFTKEY.

The SOFTKEY default is active, but invisible, as soon as you call up a program. Many of these commands are paired in an on/off format, so that using one of them automatically calls up its opposite number. Thus when you DISABLE the inputs by pressing SOFTKEY you get ENABLE INPUT in the display (plus the flashing light), because that's what's available on SOFTKEY now. Press it again and the reverse effect occurs: the inputs come back on and DISABLE INPUT pops up in the display.

WHEN THERE'S MORE THAN MEETS THE EYE...

Sometimes there are more SOFTKEY commands in a program than one, or even one automatically toggling pair. To call those other commands up, do the following:

1) Press COMMAND until the SOFTKEY indicator comes on, and a SOFTKEY command name appears in the display. Please note that this command is available, but hasn't been triggered yet, because you haven't pressed SOFTKEY.

2) Move the ADJUST/SELECT slider back and forth to display the other available SOFTKEY commands.

   (The SOFTKEY indicator light will stop flashing and become steady, indicating that the displayed command isn't yet active. This is the opposite of the way the PROGRAM indicator light works.)

3) When the command you want is visible in the display, press EXECUTE. The SOFTKEY light will flash, telling you the new command is ready to trigger when you choose.

CLEARING THE DISPLAY

To return the display to either the currently active program or parameter, press the PROGRAM or PARAMETER keys. To move on to the next system command, press the COMMAND key. (COMMAND key functions will be described shortly.)

THE EXECUTE KEY

In the last few pages you encountered two of the EXECUTE key's three uses, i.e.:

1) activating a new program;

2) changing the available SOFTKEY command.

But there is a third use for EXECUTE, which is to activate the various SYSTEM COMMANDS. You'll learn how this works next.

THE COMMAND KEY

As well as calling up SOFTKEY *commands*, the front panel's COMMAND KEY gives you access to the SP2016 SYSTEM COMMANDS. Pressing COMMAND repeatedly, or holding it down, cycles you through the list:

—the currently active SOFTKEY command—
? SYSTEM HIT EXEC (SP2016 help message)
PROGRAM HELP (for some programs)
SAVE USER PRESET
KILL PRESET
LINE IN/LINE OUT
BUS ADDRESS
SHORT SELFTEST
CONTINUOUS TEST
CONFIGURATION

PLEASE NOTE: *the KILL PRESET command only pops up after you've saved at least one preset. Before that, you won't see it.*

To activate any of these, just place the command in the display and press EXECUTE.

Here is additional information about each command. Some are described in more detail later; doing that here would be putting the cart considerably before the horse.

(CURRENTLY READY SOFTKEY)

As demonstrated earlier, when a program has more than one SOFTKEY command you use the COMMAND key, the ADJUST/SELECT slider, and the EXECUTE key together to change which one is actively available.

? SYSTEM HIT EXEC

This is a scrolling HELP message that briefly describes the SP2016 and its operation. Moving the ADJUST/SELECT slider varies the speed and direction the message scrolls. To stop it, center the slider.

This message is a useful memory jogger, but should not be considered a substitute for studying this manual.

To leave the HELP message, press either PROGRAM, PARAMETER, or COMMAND.

SAVE USER PRESET

This is the command you'll call on when saving and naming your own preset programs. How to use it is described in detail in part *four*.

KILL PRESET

Once you've saved a preset, this command becomes available. Use it to get rid of a preset you no longer want to keep in the SP2016's memory. For details, see *part four*.

LINE IN

Need to bypass the processor so you can compare signal levels and frequencies, like you did during installation check, but without turning off the SP2016? Just put the LINE IN command up in the display and move the ADJUST/SELECT slider so it changes to read LINE OUT. (You don't need to press EXECUTE with this command.)

To kick the processor back in, return the slider to LINE IN. REMEMBER TO DO THIS, otherwise you might mistakenly think your SP2016 is broken.

BUS ADDRESS

It is possible to control your SP2016 with a personal computer, using the optional IEEE-488 bus on the back panel. Part of the "bus protocol" (the rules of the electrical road) is that every device on the bus route has an "address" number; commands prefaced with that number won't be accepted by anything else on the bus.

This command sets the number that the SP2016 recognizes as its address. Unless you are a programmer, it is unlikely you will ever need to use it.

Of course, if you are a programmer, be sure and check out part six for more details.

SHORT SELFTEST

This command essentially makes the SP2016 repeat its startup diagnostic tests. For more information, see the *Troubleshooting appendix*.

CONTINUOUS TEST

This command initiates an elaborate and wide-ranging series of tests. As long as your SP2016 is working well, you will never want to bother with them. On the other hand, if you are experiencing problems, these tests can help pinpoint the difficulty and make repairs a lot easier. For complete information on how to run them, see the *Troubleshooting appendix*.

CONFIGURATION

Pressing EXECUTE when CONFIGURATION is visible in the display makes the SP2016 list its pedigree for you: rev number and date, serial number, birthday, a list of the installed program ROMs, and lastly the number of programs and presets (your own variations on the programs) that are available.

To leave the configuration message, press either COMMAND, PROGRAM, or PARAMETER.

part three:
TRICKS, TECHNIQUES, AND TIPS (A HANDS-ON TUTORIAL)

REAL-TIME EDITING

REALTIME EDITING can be summed up in four steps:

- Use PROGRAM (or the ADJUST/SELECT slider) to choose a program or preset, and EXECUTE to activate it.
- Use PARAMETER to step through the available parameters.
- Use the ADJUST/SELECT slider to alter the settings as you wish.
- Listen to the results. When you like them, stop editing.

CHOOSING PROGRAMS: THE BACK-TO-SQUARE-ONE EFFECT

When using PROGRAM to select programs, you'll notice an interesting twist. For a clear demonstration of the effect, do this:

1) Go through the programs in your unit one by one, and write them down in order.

2) Activate a program from the middle of the list.

3) Press PROGRAM again and watch the display. Logically, you would expect that the next program that would appear would be the next one on your list ...

... but no. Instead the SP2016 jumps back to the beginning of its internal list of the programs (which may not be the same as your written list in all cases, but that's not important here).

What's going on?

PLEASE NOTE: *using PROGRAM to select a new program always sends you back to the beginning. You'll have to step through all the programs in between to get back to where you were.*

ALSO NOTE: *the ADJUST/SELECT slider doesn't work that way. Nudge it to the right a little bit and you step up the list from the active program. Nudge it left, and you step down. And, as an added goodie and convenience, if you go all the way to the left with the slider you land right back in the active program you started from.*

GOING BACKWARDS

To step backwards through any of the display listings—of programs, parameters or system commands— just hold down DEFINE and press the appropriate key at the same time.

For example, to step backwards through the available programs, just hold down DEFINE and keep pressing PROGRAM. To step backwards through the parameters of the active program, press DEFINE and PARAMETER. And so on.

GETTING THE HANG OF THE SOFTKEY

There are six things to remember:

1) SOFTKEY commands vary from program to program.

2) The available SOFTKEY name can be displayed by pressing COMMAND.

3) Whatever SOFTKEY command is set to go can be triggered at any time, no matter what's visible in the display.

4) When the SOFTKEY indicator is flashing, the command in the display is the one that will be triggered the next time SOFTKEY is pressed.

5) If the SOFTKEY light is solid, the command in the display is not the one that will be triggered the next time SOFTKEY is pressed.

6) Lastly, if a program has more than one SOFTKEY command (or pair of same), you select the one you want by pressing COMMAND until the SOFTKEY indicator comes on, calling up the command you want with the ADJUST/SELECT slider, and pressing EXECUTE.

GETTING THE HANG OF THE ADJUST/SELECT SLIDER

Most of what the ADJUST/SELECT slider does is extremely straightforward: moving it changes the display with great precision and a quick response.

But not quite always.

Case in point: parameters with wide numerical ranges. With these the slider works a little differently. To see this fact for yourself:

1) Call up the STEREO ROOM program.

2) Go to the PRE-DELAY parameter.

3) Quickly shove the ADJUST/SELECT slider from hard left to hard right, and watch the display. Notice how the numbers take a moment to catch up!

4) Now shove the lever hard left, but watch the slider indicators, instead of the display. You will see a minus sign light up as the SP2016 counts down. Push the lever to the right and you'll see the opposite, a plus sign, as the unit counts up.

Now for something a little different:

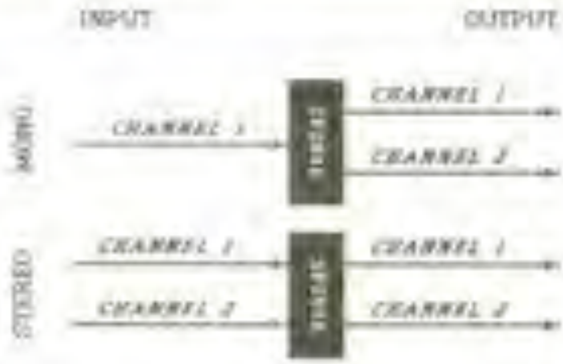
5) Move the slider either way, but stop when one of the two indicators lights up. If you look up at the display, you will see the numbers changing—but slowly.

This is what we want you to see. When moving over large ranges of numbers, the ADJUST/SELECT slider doesn't work on a one-for-one basis (its range from left to right just isn't great enough). Instead, it sets the *rate* at which the the setting steps up or down.

The middle third of the slider is a "dead band"—nothing changes while the slider is in this area. Then, as you move either right (+) or left (−) out of the dead band the appropriate indicator comes on and the unit starts counting. The farther away from the center, the faster the count.

In practice this means you'll have some slider-jockeying to do in order to make precise settings, or recover old ones. (It's a little like trying not to overshoot while setting a digital alarm clock.)

STEREO VS. MONO



A reminder: DUAL PROGRAMS accept inputs over both channels, and send them out the same way. I.e., a signal that comes in over CHANNEL 1 goes out over CHANNEL 1.

Not so with MONO. There the only accepted input is from CHANNEL 1—although you'll still usually have two outputs (and sometimes pretty different outputs, at that). The SP2016 does this on virtually all the MONO programs, to give you flexibility in processing and routing. Synthesized stereo from single sources is a snap.

part four:
YOUR OWN PRESET EFFECTS

CREATING PRESETS

To create your own presets (i.e., variations on the factory programs) you should first go crazy with those programs. Muck about with them, change their settings, experiment like mad, and listen very carefully to the results. You've got memory space for up to 65 presets, depending on how complex they get, so be inventive. When you get something you like enough to think you'll need it again, and want to save it, follow the instructions in the next section.

SAVING PRESETS

To place a preset in the SP2016's memory, you have to give it a number and a name. Naming, especially, can be a great convenience and will help you remember later on what a given effect is supposed to do.

Once you've created a program variation you like, and want to save it:

1) Press COMMAND until SAVE USER PRESET is in the display.

2) Move the ADJUST/SELECT slider back and forth. You'll see numbers 1 through 65 in the display; pick the number you want for your preset. (While there are 65 numbers, you may run out of memory space before you actually save that many; some presets will take more room than others.)

   Each preset gets a different number. If you do not assign a number, the SP2016 will do it for you. After a preset is saved, its number stops showing up as an available preset number.

3) Press EXECUTE. The display will change to read NAME, followed by eight periods (........). The SP2016 is now asking you to give your preset a name up to eight characters and/or numbers long.

4) Move the ADJUST/SELECT slider back and forth and watch as the first period cycles through the alphabet and 0-9.

5) When you see the character you want, press EXECUTE to lock it into place.

6) Now the slider will alter the second period. Repeat steps 4 and 5 until you've finished the name.

7) When finished and ready to actually save the preset, press EXECUTE again.

PLEASE NOTE: *If you want to use the same character or number twice in a row (like "FF" or "77") you must go through the standard selection process each time. There are no short cuts. Do NOT make the mistake of thinking you can make a letter or number repeat just by pressing EXECUTE twice in a row—that's the command to make the SP2016 save your preset. If you make this mistake your preset will be saved, but with an incomplete name. (To fix this situation you would have to delete the misspelled version, and then save the preset again, with the proper name.)*

If you want to check for yourself that your preset has been saved, press PROGRAM and then hunt through the list. When you see the name you chose, you know the save worked.

If you decide to quit in the middle of a save, without actually saving anything, just press either COMMAND, PROGRAM, or PARAMETER.

PLEASE NOTE: unlike the factory programs, your preset comes up in the display with two additional pieces of information.

The first extra is the letter U and a number. The U stands for "User-created." The number is the one you chose at the beginning of the save.

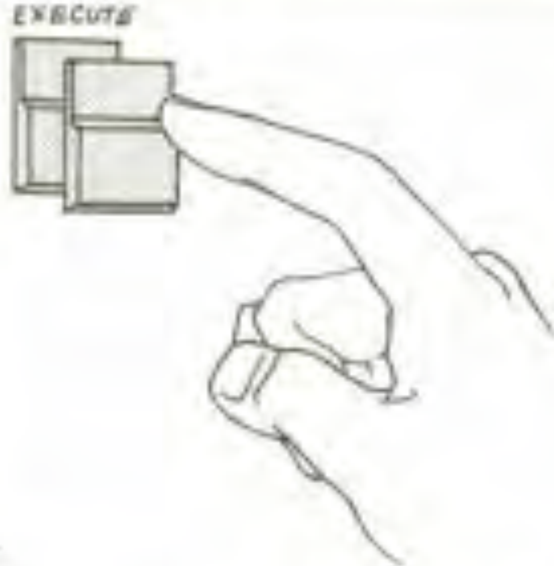
The second extra is the code for the program your preset is a variation of. If you wish to see the name of that factory program instead of its code, press DEFINE while your preset is showing in the display.

CALLING PRESETS BACK UP

To call up a preset, step through the available programs and presets with PROGRAM or the ADJUST/SELECT slider. This should be standard operating procedure to you, by now. When you see the one you want to activate, press EXECUTE.

GETTING RID OF PRESETS

Here's where those preset numbers come in handy.



Once you've stored a preset into memory, a new SYSTEM COMMAND becomes available: KILL USER PRESET. To get rid of a preset you no longer want in memory, just:

1) Press COMMAND until you see KILL USER PRESET.

2) Move the ADJUST/SELECT slider back and forth until you see the number of the preset you wish to erase. The only numbers that show are those belonging to presets, so they may seem to skip around a lot.

3) Press EXECUTE twice, quickly. (This protects you from accidentally wiping out a preset you don't want to lose. If you hesitate, the SP2016 display changes from HIT KEY TO KILL to NONE DELETED, and you'll have to start over by pressing EXECUTE again.)

4) When the preset is gone, the display reads PRESET DELETED.

To QUIT at any time without erasing, and return to normal operation, press either COMMAND, PROGRAM, or PARAMETER before hitting EXECUTE.

part five:
THE FACTORY PROGRAMS

STEREO ROOM

BASIC DESCRIPTION:

STEREO ROOM recreates the ambience of a large concert hall, with very clear, natural reverberation. The program gives you control over the apparent size, wall texture, and number of people in the room (through alteration of basic reverb characteristics), as well as your relative position near or far from the "stage."

FACTORY PROGRAM NAME: F08

INPUT MODE: STEREO (switchable to MONO)

BANDWIDTH: 16 kHz

PARAMETERS:	RANGE:	DEFAULT:
DECAY TIME	0.2 to 30.0 sec.	3.0 sec.
FRONT/REAR (room position)	14 steps	step 10
PRE-DELAY	1 to 250 ms.	25 ms.
LOW FACTOR (lo freq scaling)	−8 to +4	−4
LOW ROLLOFF (rolloff point)	50 to 500 Hz (in 50 Hz steps)	250 Hz
HIGH FACTOR (hi freq scaling)	−8 to 0	−4
HIGH ROLLOFF (rolloff point)	1.0 to 8.0 kHz (in 500 Hz steps)	5.0 kHz
DIFFUSION	low, medium, high	high
INPUT	stereo, mono (ch. 1)	stereo

WHAT'S GOING ON:

SOFTKEY FUNCTIONS:	DEFAULT:
DISABLE INPUT alternating with ENABLE INPUT	disable
CLEAR REVERB alternating with RESUME REVERB	

DECAY TIME

If you press DEFINE for this parameter you will see that decay time is defined as RT60 IN SECONDS. That's how long it takes a mid-band frequency of 1kHz to fall 60 dB in amplitude. DECAY TIME's default for this is 3 seconds. (Higher frequencies will tend to decay more quickly.) Use the ADJUST/SELECT slider to change the decay time from .2 to 30 seconds.

PLEASE NOTE: *How decay time is perceived by the ear will vary with the sounds you are processing. As in a real room, different sounds decay in different ways, depending on their spectral characteristics.*

ROOM POSITION

This parameter appears in the display as the letters F, R, and an asterisk.

The asterisk represents the relative position of the listener in the "room." F is for FRONT, or nearest the stage; R is for REAR, the back of the hall. Use the ADJUST/SELECT slider to move the asterisk between F and R. The nearer to F, the "closer" you'll feel to the sound's apparent source; the nearer to R, the "farther away," as later reflections blur the sound.

PRE-DELAY

PRE-DELAY is the length of time that passes before reverb begins. It ranges from 1 to 250 milliseconds. Effective use of some pre-delay can help you keep even heavily reverberant sounds clear, especially during relatively fast runs of notes (and long pre-delays can become a special echo effect in their own right).

LOW FACTOR

This parameter increases or decreases low frequency decay time relative to the decay time of the mid-band. With long decay times it should usually be set below 0, otherwise the sound will "boom."

LOW ROLLOFF

This sets the highest frequency that LOW FACTOR will affect. It creates a ceiling, below which the decay time will be cut or boosted according to LOW FACTOR, and above which everything will be untouched. The range is from 50 Hz to 500 Hz, moving in 50 Hz steps.

Example: if you wanted to moderately decrease low frequency decay time below 300 Hz, you would set the LOW FACTOR to about −4 and LOW ROLLOFF to 300 Hz.

HIGH FACTOR

Same basic idea as LOW FACTOR, but the other end of the audio spectrum. With this parameter you decrease the high frequency decay time relative to the mid-band decay. The range is −8 to 0. Values from −2 to 0 will create bright, "unnatural" rooms: in a real space the air, furniture, walls, and people tend to absorb high frequencies.

HIGH ROLLOFF

Same as LOW ROLLOFF, except that it sets a frequency floor, not a ceiling, above which HIGH FACTOR does its thing (and below which HIGH FACTOR has no effect).

DIFFUSION

Subtle stuff. This control simulates the effect of having different types of wall surfaces in the "room." LOW DIFFUSION is like having smooth, polished walls: the sound reflections will be more discrete and the resulting reverb effect more coherent. HIGH DIFFUSION is like having walls with many angles or a really rough texture, which breaks up the sound into more and smaller reflections. MEDIUM DIFFUSION, obviously, is somewhere in between.

The DIFFUSION setting doesn't change the decay time, but it does have an effect on the evident nature of the decay; it "thickens" and "thins" the reverb. (You may feel like fine tuning DECAY TIME slightly after setting DIFFUSION.)

The shorter the DECAY TIME, the more you'll be able to notice the impact of the DIFFUSION setting.

INPUT MODE

Two choices here: STEREO (inputs from both channel 1 and channel 2) or MONO (input strictly from channel 1, with any signal coming into channel 2 completely ignored).

In either case your output will be in stereo, because in the STEREO ROOM program some early reflections from one channel appear in the opposite channel. That's why STEREO ROOM, unlike other "dual" programs, does not have discrete right and left channels. It blends them.

If you have only one signal source, choose MONO and send it in over channel 1 (or split it in your mixer, if possible, and send the same signal in over both channels with the SP2016 in STEREO). Don't stay in STEREO and then send in a signal over just channel 1—you won't get any blending of early reflections with the other channel, and the effect won't sound as nice.

SOFTKEY:

DISABLE INPUT/ENABLE INPUT

DISABLE INPUT comes up first in line for triggering. Press SOFTKEY to instantly cut off the reverb effect for new signals coming into the SP2016 (reverberations already in progress are not interrupted; they finish out their natural lives).

This command automatically toggles over to ENABLE INPUT, so the next time you press SOFTKEY the reverb effect will start up again.

CLEAR REVERB/RESUME REVERB

An alternative set of SOFTKEY commands: call them up by pressing COMMAND, moving the ADJUST/SELECT slider to display the one you want, and pressing EXECUTE.

Pressing SOFTKEY to trigger CLEAR REVERB cuts off all current reverberations, period. Nothing hangs over (as it does with DISABLE INPUTS). The next time you press SOFTKEY it triggers RESUME REVERB and the STEREO ROOM starts up fresh.

To return to the other pair of SOFTKEY commands, repeat the call-up procedure described above.

ROOM REVERB

BASIC DESCRIPTION:

ROOM REVERB is your basic no-nonsense, no-frills ambient "room." It's a MONO effect with STEREO outputs. ("Pseudo-stereo" anyway, since the two outputs aren't identical; think of it as putting two microphones in an echo chamber.) The extreme PRE-DELAY range makes it possible to use this effect for reverberant doubling and single echoes.

FACTORY PROGRAM NAME: #10

INPUT MODE: MONO

BANDWIDTH: 16 kHz

PARAMETERS:	RANGE:	DEFAULT:
DECAY TIME	0.2 to 10.0 sec. (in .4, then .5 second steps)	3.0 sec.
PRE-DELAY	0.0 to 999.9 ms.	0.0 ms.
FRONT/REAR (room position)	7 steps	step 5

SOFTKEY FUNCTIONS:	DEFAULT:
DISABLE INPUT alternating with ENABLE INPUT	disable

WHAT'S GOING ON:

DECAY TIME

Exactly as just described for STEREO ROOM, except that the maximum decay is 10 seconds (instead of 30).

PRE-DELAY

As described for STEREO ROOM, except with four times the range: maximum delay is 1 second. This allows you to use this program as a doubler or single-echo generator, in which the doubled or echoed signal is always reverberant.

ROOM POSITION

As described for STEREO ROOM, except that there are fewer discrete steps between the FRONT and REAR of the room.

SOFTKEY:

DISABLE INPUT/ENABLE INPUT

Standard DISABLE/ENABLE INPUT, as described for STEREO ROOM.

LOOP EDIT

BASIC DESCRIPTION:

LOOP EDIT is a MONO program that lets you digitally (sample) an audio signal up to 1.6 seconds long (or 3.2 seconds at half-bandwidth), edit it, and then either A) loop it for endless playback or B) trigger it for single-shot playback from either the SOFTKEY or the envelope of a signal coming in over channel 2.

The thing to remember about LOOP EDIT is that the loop is always recording—and always erasing itself to make room for the new stuff being recorded. It's how you interrupt and manipulate that process that gives this program its power.

FACTORY PROGRAM NAME: F14

INPUT MODE: MONO

BANDWIDTH: 16 kHz

PARAMETERS:	RANGE:	DEFAULT:
LENGTH	20 to 1636 ms. (in 4 ms. steps, at 16 kHz bandwidth)	1636 ms
	40 to 3272 ms. (at 8 kHz bandwidth)	———
PLAY (loop playback mode)	continuous man'l shot one shot	continuous
RECORD (loop recording mode)	punch-in, punch-out	punch-in
LOOP LENGTH/BANDWIDTH	normal (16 kHz) or extended (8 kHz)	normal
ENVELOPE TRIGGER	off, on	off

SOFTKEY FUNCTIONS:	DEFAULT:
PUNCH IN/OUT *alternating with* RELEASE LOOP PLAYBACK TRIGGER	punch in/out

WHAT'S GOING ON:

Digital sampling time . . . with a few twists.

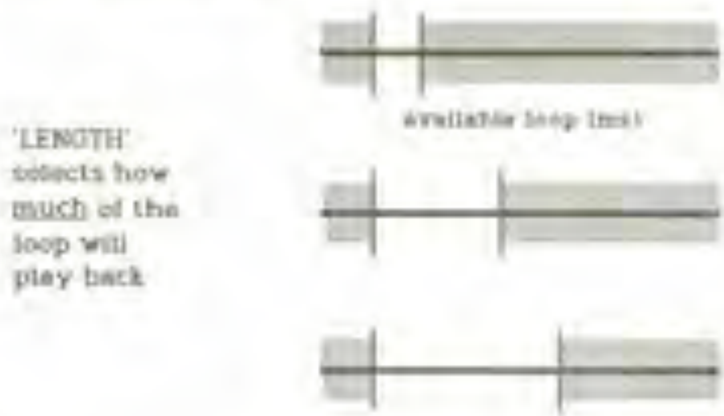
Simple demo, first. Don't change any defaults: just send a signal into the SP2016 over channel 1, and press the SOFTKEY. The display will say RECORDING . . . for a bit less than 2 seconds, and then—bang!—you'll hear those 1636 milliseconds (to be precise) playing back again and again and again in a loop.

You'll notice the display now says RELEASE LOOP. To shut off the loop, press the SOFTKEY again.

Got the basic idea? Good. But that's not all LOOP EDIT does, not by a long shot, as a quick walk through the parameters will make clear.

LENGTH

Your maximum sample length is either 1636 or 3272 milliseconds, depending on the choice you make later concerning LENGTH BANDWIDTH. But just because you've got all that time available doesn't mean you actually want to use all of it. In fact, most of the time you'd be better off with something exactly the length you prefer, be it 2 milliseconds or 2000. This parameter sets this.



Start a loop running, as described above, and then play with this parameter. As you reduce the LENGTH setting you'll notice that playback gets shorter and shorter.

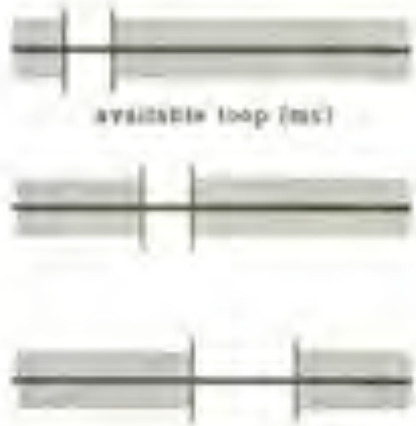
LENGTH also sets how long a sample you'll record when you press SOFTKEY. Just as with playback, it can be made as short or long as you prefer, within the limits of available memory. This is useful when matching segments to an established rhythm: do it once, and you're set.

If the end of your loop has annoying splice noise, vary the loop's length until you can't hear the noise anymore.

POSITION

To understand POSITION, you should think of your sample as a tape loop. It doesn't really have a beginning or an end—it just keeps on rolling forever (if you let it). LENGTH, as just described, sets how much of the loop will play back or record . . . but you have to use POSITION to determine where that portion will start.

'POSITION' selects where on the loop recording will start (default: total available loop less current length setting)



Unless you choose otherwise by adjusting this setting yourself, POSITION always tracks the LENGTH setting. It does not stay at 0. Here's what happens: if your available memory space is 1636 milliseconds, and your LENGTH setting is 1000 ms, then POSITION will automatically set itself to 636 ms. The equation is POSITION = LOOP SIZE minus LENGTH.

PLEASE NOTE: *using LENGTH and POSITION you can turn a single loop into a patchwork of different, shorter samples, and then mix and match their playbacks so that you get the last part of one and the first part of another, or three in a row and the leading edge of the fourth, or ... well, whatever "window" on the loop you want.*

RECORD MODE

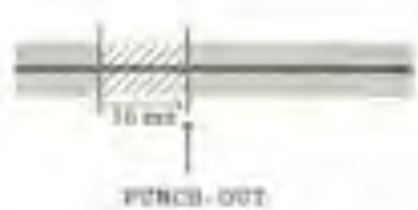
This parameter only appears if you do not have a loop recorded into memory. It offers you a simple choice: PUNCH IN or PUNCH OUT.

PUNCH IN you know from the example given—you press SOFTKEY and the loop records for an amount of time set by the LENGTH parameter. The alternative, PUNCH OUT, works in exactly the opposite fashion. Choose it, and when you press SOFTKEY you store into the loop everything that was playing *before* you pressed the button, for a time equal to the LENGTH setting. Example: with LENGTH set to 1000 ms, pressing PUNCH IN will record the next 1 second of sound into the loop. Pressing PUNCH OUT will send into the loop everything you'd played for the previous second.

If your length setting is 10 ms, PUNCH-IN records the next 10 ms of signal

But since the free space of the loop is always recording...

If your length setting is 10 ms, pressing PUNCH-OUT makes the SP2016 keep the previous 10 ms of signal

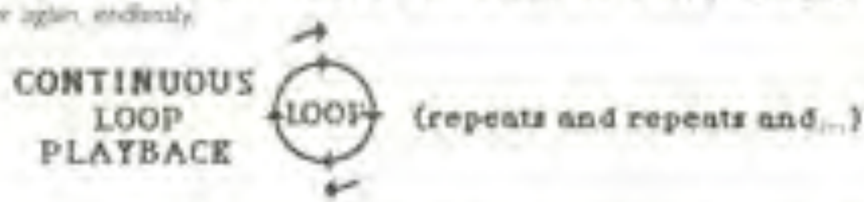


If something is already recorded, you don't see the RECORD MODE parameter at all. This is because the only useful tool for dropping new fragments into existing loops is PUNCH IN.

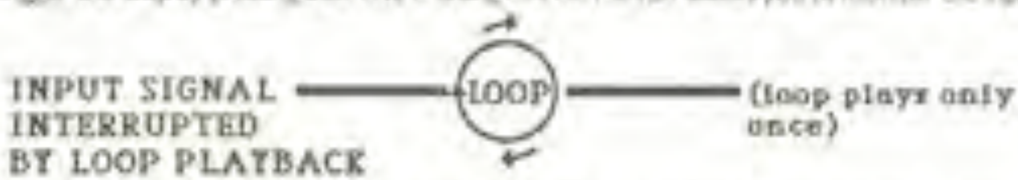
PLAYBACK MODE

This parameter only appears when there is a loop in memory.

Three settings here. One, CONTINUOUS, you've already met. It simply means your recorded loop will play over and over again, endlessly.

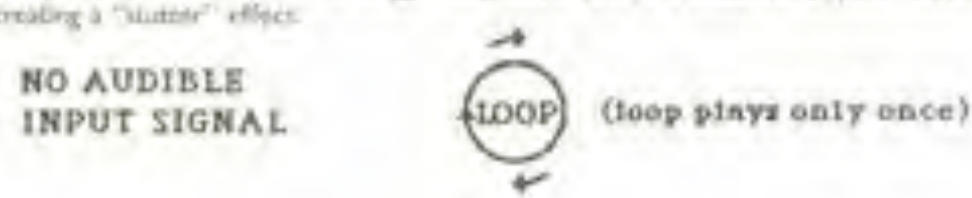


MNTR/1 SHOT is different in that it only plays the loop back a single time, on cue. That cue comes from the signal level of channel 2 (and it can be turned on and off with the next parameter, ENVELOPE TRIGGER). You can also trigger the loop by pressing SOFTKEY. Using MNTR/1 SHOT allows you to monitor the input signal.



PLEASE NOTE: *when the signal triggers playback of the loop, the unit temporarily shuts out any incoming signal; in effect, it sounds like the loop itself barged in and cut off the other sound, which only returns when the loop is over.*

PLAY ONE SHOT works the same way as MNTR/1 SHOT—except you cannot monitor your input. PLAY ONE SHOT is triggered by the signal level of channel 2, or by pressing SOFTKEY. Be aware that you don't have to wait for the loop to play out before you can trigger it again; but you do truncate the playback that is in progress if you do so, creating a "stutter" effect.



ENVELOPE TRIGGER

This parameter affects both the MNTR/1 SHOT and CONTINUOUS settings just discussed. It has two settings—ON and OFF. In both cases, OFF means there will be no external trigger effect. As for ON...

ON, with MNTR/1 SHOT or PLAY ONE SHOT: the signal coming in over channel 2 can be used to trigger a single playback of the loop.

ON, with CONTINUOUS: the signal coming in over channel 2 can be used to reset the loop to its starting point.

PLEASE NOTE: *it is channel 2's volume that does the triggering, so you can adjust trigger sensitivity by lowering and raising the channel 2 input volume slider.*

LENGTH/BANDWIDTH

Another simple choice, this time between the default, 16 kHz, and 8 kHz. 16 kHz has greater bandwidth, and therefore better sound, but only half the available loop length. Make the choice that suits your music.

As a special effect, you can slow down or speed up loops by recording them at one bandwidth and then playing them back at the other. Going from 16 kHz to 8 kHz will drop pitch an octave; going the other way will raise it an octave.

SOFTKEY:

PUNCH IN/PUNCH OUT

The default. With this active, pressing SOFTKEY will record either the next X milliseconds of signal onto the loop, or the previous X milliseconds, depending on whether you have chosen PUNCH IN or PUNCH OUT as your RECORD MODE.

After use, this automatically toggles to whichever of the two other SOFTKEY COMMANDS is active.

RELEASE LOOP

RELEASE LOOP does just what it says—it dumps whatever sound is in the loop and leaves it empty, and toggles automatically back to PUNCH IN/PUNCH OUT. This is only available automatically if your PLAYBACK MODE choice is CONTINUOUS. Otherwise, you'll have to use the COMMAND/SLIDER/EXECUTE combination to call it up.

You do not have to release a loop before punching in again if you don't want to. Just sidestep this command and call PUNCH IN/PUNCH OUT back up. Doing this allows you to create loops that are made up of many little segments.

PLAYBACK TRIGGER

When PLAYBACK MODE is set to either MNTR/1 SHOT or ONE SHOT, then pressing SOFTKEY activates this command and the loop plays back a single time. Unlike RELEASE LOOP, PLAYBACK TRIGGER doesn't toggle with PUNCH IN/PUNCH OUT. If you want to get PUNCH IN/PUNCH OUT back, you'll have to specifically call it up and reactivate it.

FLANGER

BASIC DESCRIPTION:

Flanging is a classic effect caused by phase cancellations between between two signals with varying relative delays. It was originally achieved by literally pressing against the flange of a tape reel in order to slow down one tape recorder in a set; but the SP2016 flanges by using digital circuitry to combine a short fixed delay with a swept delay (one which varies over time). The rate and sweep limits of this second delay are variable.

FACTORY PROGRAM NAME: F18

INPUT MODE: STEREO (switchable to MONO)

BANDWIDTH: 16 kHz

PARAMETERS:	RANGE:	DEFAULT:
SWEEP SPEED (slow to fast)	1 (min) to 50 (max)	25
<LIMITS (ch 1, in milliseconds)	0.0 to 3.0 ms. (.1, then .2 steps)	0.0 ms.
LIMITS> (ch 2, in milliseconds)	0.0 to 10.0 ms. (.1, then .2, then .5 steps)	3.0 ms.
INTENSITY (cancellation amount)	−10 to +10	+10
FEEDBACK (by percentage)	−99% to +99% of	+0% total feedback in 3% steps)
DIFFUSION	1 (min) to 50 (max)	1
<PRE-DELAY (ch 1, in milliseconds)	0 to 400 ms. (in 5 ms. steps)	45 ms.
PRE-DELAY> (ch 2, in milliseconds)	0 to 400 ms. (in 5 ms. steps)	45 ms.
INPUT (input mode)	stereo, mono (ch1)	stereo

SOFTKEY FUNCTIONS:

FREEZE FLANGING alternating with RESUME FLANGING

DEFAULT:

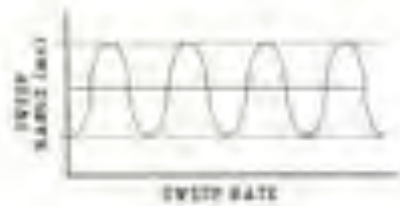
freeze

WHAT'S GOING ON:

SWEEP SPEED

SWEEP SPEED sets the rate of the flanging effect. The default is 25, exactly halfway between the slowest (1) and fastest (50) settings. Just how long the flanging takes to go through a complete cycle depends only partially on this setting. This only sets the speed. SWEEP LIMIT is also important.

SLOW SWEEP

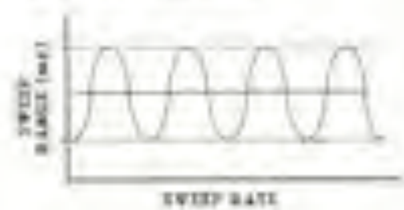


FAST SWEEP
(same range as above)

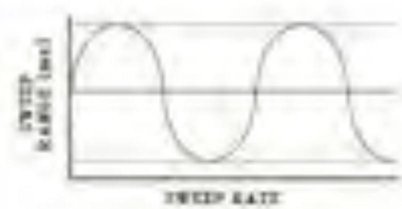
LIMITS

Because it sets the distance to be traveled at the chosen rate. You get to set separate upper and lower limits. Both are displayed at the same time, although you change them one after the other, starting with the lower limit. (Which is active is indicated by the < and > arrows in the display.) The range varies from 0.0 to 10.5 *milliseconds*, and the only thing you have to *remember* is that the LOW LIMIT setting can go no higher than the HIGH LIMIT shown in the display, and vice-versa.

LOW LIMITS



HIGH LIMITS



You could not, for example, have a HIGH LIMIT of 1.0 and a LOW LIMIT of 2.0. The SP2016 won't let you do it.

PLEASE NOTE: *the perceived sweep speed will depend on this setting. A given rate will sound faster with tight limits than it will with wide limits, an interesting and useful psychoacoustic trick.*

Remember that you can return to adjusting the LOWER LIMIT after working on the UPPER LIMIT by just pressing DEFINE and PARAMETER at the same time.

INTENSITY

With this parameter you adjust the amount and type of phase cancellation, by mixing more or less of the fixed delay with the sweeping delay. It's on an arbitrary scale of −10 to +10, with 0 meaning there is no fixed delay mixed in at all. (At 0 INTENSITY, really high SWEEP SPEED settings cause a Doppler pitch shift.)

As the setting rises from 0 towards +10, increasing levels of fixed delay are mixed "in phase" with the sweeping delay. This is called positive flanging, and has a warm sound. As the setting sinks from 0 towards −10, the fixed delay is mixed in "out of phase". This is negative flanging, and will sound hollower, and at the extreme very metallic.

FEEDBACK

You can route some of the flanged signal back in on itself, using the FEEDBACK parameter. This can result in markedly increased cancellations and resonances and some very interesting effects.

The range is from −99% to 99%. The absolute number is the amount of the feedback. In other words, at −50% and 50% the same amount of recirculation—half the signal—is happening. What's different is this: a positive (+) FEEDBACK setting means that the feedback is added to the input; a negative (−) FEEDBACK setting means that the feedback is subtracted from the input. The +/− choice will make less audible difference at high settings, and a greater difference at low setting (i.e., +99 and −99 will sound less different from one another than +10 and −10 will).

CAUTION! High FEEDBACK settings will cause extreme resonance peaks . . . which, in turn, cause sudden and drastic increases in volume. Be careful not to damage your monitors.

DIFFUSION

This is a psychoacoustic setting, determining how widely "spread" in space the sound appears to be. The range is from 0 to 50; the higher the number, the wider the spread.

Since this effect is achieved by adding bits of signal from multiple delay elements to each of the channels, it will be most noticeable in MONO input mode. In STEREO input elements from both channels blend and weaken the illusion.

PRE-DELAY

With STEREO input, each channel has an independent PRE-DELAY setting. This sets the time that will pass before the flanging effect kicks in on that channel. MONO inputs also have two PRE-DELAYS; the signal is split ahead of the flanging circuitry.

INPUT MODE

Standard STEREO and MONO choices, as described elsewhere. Your choice here does affect some of the other flanging parameters. Check them for details.

SOFTKEY:

FREEZE FLANGING/RESUME FLANGING

FREEZE FLANGING instantly stops the sweeping delay (although you will still hear a difference in the signal, caused by the fixed delay). RESUME FLANGING starts it again at precisely the point it stopped. There is no noise created by this transition, so flicking back and forth between the two is neat and effective.

For an even more interesting feeling, set the SWEEP SPEED to MAX and then start toggling this pair of commands.

ENVELOPE FLANGER

BASIC DESCRIPTION:

A different kind of flanger, similar to the last but with one essential difference—the flanging is not constant; rather, it is controlled by one of three things: the input volume, the ADJUST/SELECT slider, or the SOFTKEY.

FACTORY PROGRAM NAME: F19

INPUT MODE: STEREO (switchable to MONO)

BANDWIDTH: 16 kHz

PARAMETERS:	RANGE:	DEFAULT:
<LIMITS (ch 1, by milliseconds)	0.0 to 3.0 ms. (.1, then .2 steps)	0.0 ms.
LIMITS> (ch 2, by milliseconds)	0.0 to 10.0 ms. (.1, then .2, then .5 steps)	3.0 ms.
INTENSITY (cancellation amount)	−10 to +10	+10
FEEDBACK (by percentage)	−99% to +99% of total feedback (in 3% steps)	+0%
DIFFUSION	1 (min) to 50 (max)	1
INPUT	stereo, mono (ch 1)	stereo
PROGRAM MODE	manual, follow, or trigger	follow
ENVELOPE GAIN*	1 (min) to 50 (max)	20
ENVELOPE SPEED* (slow to fast)	1 to 20	10
FLANGE DELAY*	limit dependent	0.0 ms.
DECAY RATE* (slow to fast)	1 to 30	16

*PLEASE NOTE: *ENVELOPE GAIN and ENVELOPE SPEED appear only if PROGRAM MODE is set to FOLLOW.*

FLANGE DELAY appears only if PROGRAM MODE is on MANUAL.

DECAY RATE appears only if PROGRAM MODE is on TRIGGER.

SOFTKEY FUNCTIONS:	DEFAULT:
FLANGE TRIGGER	———

WHAT'S GOING ON:

LIMITS, INTENSITY, FEEDBACK, DIFFUSION, and INPUT MODE

As described for FLANGER.

PROGRAM MODE

Something different. There are three settings: MANUAL MODE, FOLLOW MODE, and TRIGGER MODE. The default is FOLLOW.

FOLLOW means that the flanging is determined by the input signal's intensity. Two further parameters are exclusively involved--ENVELOPE GAIN and ENVELOPE SPEED.

MANUAL lets you set the SP2016 so you can sweep the flange effect by hand, using the ADJUST/SELECT slider and the FLANGE DELAY parameter that pops up when MANUAL is in effect.

TRIGGER sets things up so that the SOFTKEY turns flanging on and off exclusively. It also has a parameter of its own, DECAY RATE, that is not available until TRIGGER has been chosen.

ENVELOPE GAIN and ENVELOPE SPEED

These two parameters appear only if you've selected FOLLOW mode.

ENVELOPE GAIN uses channel 2's signal to sweep the flange over the range circumscribed by the LIMITS setting. The general idea is this: at a proper ENVELOPE GAIN setting, if you play soft you'll get little flanging; play louder and you'll get a lot.

ENVELOPE SPEED runs in a range from slow (1) to fast (30). The faster the setting, the more tightly the envelope will track the input signal. Choosing this setting is a judgement call. Too high, and the envelope will track undesired transients and noise and signal variations, changing when you don't want it to. Too low, and the flanging will be "sluggish" and unresponsive. Only experimentation will find an inbetween that pleases you.

FLANGE DELAY

This parameter appears only if you select MANUAL mode. When it says FLANGE DELAY in the display, you can sweep the flange with the ADJUST/SELECT slider. The available range of sweep is set with the LIMITS commands.

DECAY RATE

If TRIGGER mode is selected, the display will offer you the DECAY RATE parameter. The range is slow (1) through 30 (fast), and sets how long the flange effect will last after it is triggered by pressing SOFTKEY.

In between triggers, the signal stays at the upper LIMIT setting.

SOFTKEY:

FREEZE FLANGING/RESUME FLANGING

As described for FLANGER.

CHORUS

BASIC DESCRIPTION:

CHORUS adds pairs of voices to the original signal, "thickening" it. The relative psychoacoustic placement of these add-on voices can be varied. Ask most musicians, and they'll tell you that chorusing is, in effect, the next step past flanging.

FACTORY PROGRAM NAME: FIC

INPUT MODE: STEREO (switchable to MONO)

BANDWIDTH: 16 kHz

PARAMETERS:	RANGE:	DEFAULT:
x VOICE CHORUS (number of voices)	2, 4, 6, or 8	4
INTENSITY	1 to 10	3
FEEDBACK	0% to 99% (in 3% steps)	0%
DIFFUSION	1 (min) to 50 (max)	1
< PRE-DELAY (ch 1, in milliseconds)	0 to 400 ms. (in 5 ms. steps)	15 ms.
PRE-DELAY > (ch 2, in milliseconds)	0 to 400 ms. (in 5 ms. steps)	15 ms.
INPUT	stereo, mono (ch 1)	stereo

SOFTKEY FUNCTIONS:	DEFAULT:
NONE	——

WHAT'S GOING ON:

VOICES

This parameter sets the number of voices in your digitally simulated "chorus"—2, 4, 6, or 8

INTENSITY

The scale of the CHORUS INTENSITY is from low (1) to high (10). As it goes up, increasing amounts of delay and pitch shift are being applied to the pairs of voices. The best setting is highly dependent on the signal coming through— broad sounds with changing spectrums can be highly chorused with no problem. Some others, particularly those with sustained high-frequencies, take on a "burbling" quality you may not want. If that's the case, lower the INTENSITY.

FEEDBACK

This recirculates some of the CHORUS effect back into itself; just how much you can tell by the percentage in the display. Low settings give a subtle and liquid quality to each voice. High settings create resonant and sweeping filter effects.

FEEDBACK works better with fewer voices.

DIFFUSION

Standard psychoacoustic "spreading" effect. Works as described for this parameter in the FLANGER program.

PRE-DELAY

Standard PRE-DELAY, determining how long passes between initial signal and effect, with separate settings for both channels. Works as described elsewhere.

INPUT MODE

Same as described elsewhere.

PLEASE NOTE: *there are no SOFTKEY commands in CHORUS.*

PLATE REVERB

BASIC DESCRIPTION:

This program simulates a plate-type reverb unit with dual pickups. Although it is a MONO input, it has two differing outputs, creating a pseudo-stereo effect.

FACTORY PROGRAM NAME: F20

INPUT MODE: MONO

BANDWIDTH: 16 kHz

PARAMETERS:	RANGE:	DEFAULT:
DECAY TIME	.2, .6, 1, 1.5, 2, 2.5, 3, 3.5, 4, 5, 7, and 10 seconds	3.0 sec
PRE-DELAY	0 to 999.9 ms	0.0 ms

SOFTKEY FUNCTIONS:	DEFAULT:
DISABLE INPUT alternating with ENABLE INPUT	disable

WHAT'S GOING ON:

DECAY TIME

As described in STEREO ROOM, except for range.

PRE-DELAY

As described in STEREO ROOM, except for range.

SOFTKEY:

DISABLE INPUT/ENABLE INPUT

As described in STEREO ROOM.

MULTITAP DELAY

BASIC DESCRIPTION:

Think of MULTITAP DELAY as the mother of all Echoplexes and you won't be too far wrong. That's basically how it operates, with a lovely flexibility added by the fact that you can choose the number of "tape heads", their positions, their volumes, and even their feedback characteristics.

To give you a sense of this program's versatility, you can use it to create everything from an effective "backwards reverb" to a massed church choir singing Gregorian Chants in Notre Dame Cathedral (from one source voice, yet!). There'll be more detail at the end of the parameter descriptions.

WARNING!

BE VERY CAREFUL WITH THE FEEDBACK PARAMETER IN THIS PROGRAM. BECAUSE YOU HAVE SO MANY TAPS AVAILABLE, IT'S EASY TO CREATE FEEDBACK LOOPS. THOSE IN TURN CAUSE OSCILLATION TO SET IN AND BLOW YOUR SPEAKERS OUT IF THINGS GET OUT OF CONTROL.

THE SOFTKEY IN THIS PROGRAM IS YOUR SAFETY SWITCH. IF FEEDBACK STARTS TO RUN WILD, HIT THE SOFTKEY TO RESET THINGS!

TO PREVENT OSCILLATION WHEN YOU CHANGE CERTAIN PARAMETERS, THE SP2016 AUTOMATICALLY RESETS THE FEEDBACK TO ZERO.

FACTORY PROGRAM NAME: F24

INPUT MODE: MONO (switchable to stereo)

BANDWIDTH: 16 kHz

PARAMETERS:	RANGE:	DEFAULT:
DELAY TAPS (how many per channel)	1 to 50	30
LAST TAP	10 to 1300 ms. (in 5 ms. steps)	1000 ms.
ENVELOPE SHAPE	exponential decrease linear decrease flat linear increase exponential increase triangular	exp increase
SPACING (type of)	decreasing (5-1) constant increasing (1-5)	increasing 3

OUTPUT MIX	normal or mixed	mixed
CH 1 PRE-DELAY	0 to 400 ms (in 5 ms steps)	0 ms
CH 2 PRE-DELAY (only in stereo)	0 to 400 ms (in 5 ms steps)	0 ms
FEEDBACK	-99.9% to +99.9%	0.0%
FEEDBACK TAP	10 to 1600 ms./mono (10-800 ms./stereo)	500 ms " "
FEEDBACK TYPE	single or multitap in mono (single, multitap, or crossfed in stereo input mode)	single
INPUT	mono or stereo	mono

SOFTKEY FUNCTIONS: DEFAULT:

ZERO FEEDBACK ——

WHAT'S GOING ON:

DELAY TAPS

Right. So you have this digital "tape" path, and along it you have a bunch of "playback heads." Those are the taps. You set how many you want using this parameter. The more taps, the more little echoes there will be in your sound, making it bigger and denser.

In MONO input mode, you can have up to 50 taps on the loop. In STEREO mode, you have two shorter, completely independent, loops with up to 25 taps each.

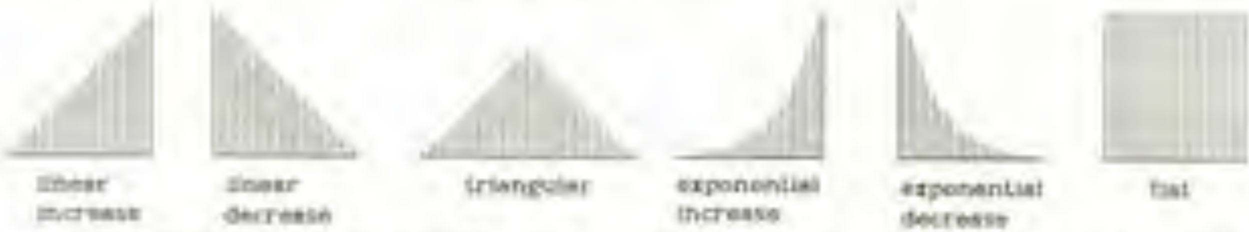
LAST TAP

The "tape heads" are set an equal distance apart from one another. You set that distance by picking the length of the loop, in milliseconds, with the LAST TAP parameter. For example, if you picked 1200 milliseconds and 12 taps, then each tap would be 100 milliseconds apart. If you picked 24 taps they'd be 50 milliseconds apart, and so on.

If MULTITAP DELAY is in STEREO input, there is a 600 ms limit on the value of the LAST TAP delay. In MONO it is 1200 ms.

ENVELOPE SHAPE

How successive taps will vary (or not) in volume depends on the ENVELOPE SHAPE you pick. There are six; you can pretty much tell their effect just by looking at them.



Each line in these diagrams represents a single, equally-spaced tap. (To visualize how these envelopes would look if the taps were spaced further apart, just imagine for a moment that they are Slinkies, and you are pulling them from both ends.)

TYPE OF SPACING

Up until now we've said the taps were equally spaced. But they don't have to be — and as a matter of fact, the default for this program is that they aren't. As you will see if you look at this parameter, they are set to INCREASING SPACING.

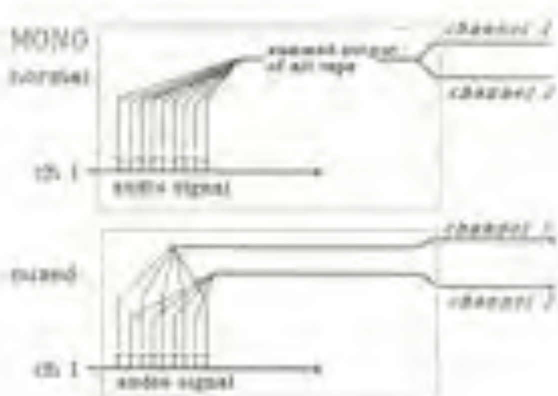
That's one of three options available: CONSTANT SPACING, where the taps are spaced equally; INCREASING SPACING, where the taps are closer together in the beginning and spread out as they go; and DECREASING SPACING, which is the opposite—far apart at first, with shrinking spaces as you move down the line.

In addition, both INCREASING and DECREASING have numerical settings from 1 to 5 which control how extreme the change is from one end to the other. Choosing 1 means there is very little change; it's almost like having the CONSTANT SPACING setting instead. 5, on the other hand, is quite extreme indeed.

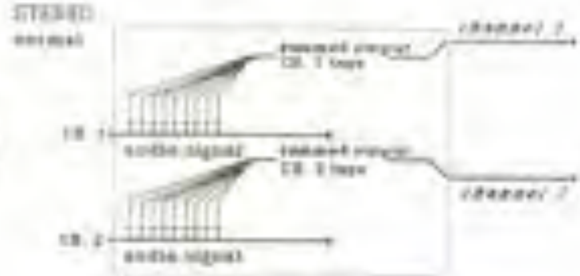
OUTPUT MODE

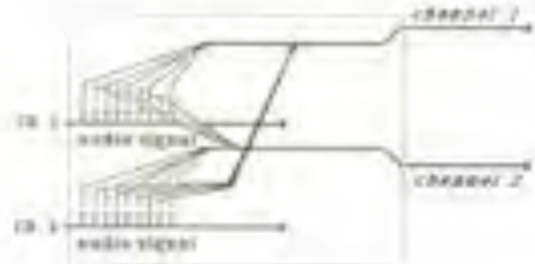
Your choice is NORMAL or MIXED. What you get with either of these depends on whether your input is mono or stereo.

If your input is MONO, NORMAL means that the output of all the taps will be summed together and sent to both output jacks. MIXED will send some tap outputs to channel 1 and some to channel 2.



If your input is STEREO, NORMAL will send the signals from channel 1's taps to OUTPUT 1 and channel 2's taps to OUTPUT 2. MIXED will take some of channel 1's taps and blend them into channel 2's output, and vice-versa.





DIFFERENTIAL DELAY
(STEREO mode only)

This parameter appears only if you are using STEREO for input. It has two settings, TRACKED or SKEWED. TRACKED makes all the same-numbered taps in the two different loops line up in time, so that tap 3 of channel 1 and tap 3 of channel 2 sound together, etc. (assuming you haven't given the two channels different PRE-DELAY settings). SKEWED deliberately offsets the taps on channel 1 against those of channel 2, so they don't quite match up. It thickens the sound.

TRACKED

SKEWED

CH 1 PRE-DELAY

A pre-delay is pretty much the same, the world over, but here there is a twist based on INPUT mode. In MONO, the available range goes up to 400 milliseconds. In STEREO, it only goes to 200 ms.

CH 2 PRE-DELAY

Since this only appears in the display if you have a STEREO input going, this PRE-DELAY has an upper limit of 200 ms.

FEEDBACK

IF YOU HAVEN'T READ THE FEEDBACK WARNING EARLIER IN THIS PROGRAM DESCRIPTION, GO BACK AND DO IT NOW!

FEEDBACK is the percentage of original signal that is sent back to the beginning of the taps, after having passed through some or all of them. If it is passed back in phase with the original signal, the feedback is positive; out of phase and the feedback is negative.

This parameter controls the amount of feedback. The type and position are controlled by the next two parameters.

FEEDBACK TAP

This parameter only appears if you have chosen SINGLE TAP or CROSSFED modes when picking a feedback type. And actually, it is out of order in the display, because it shows up one parameter before you get to the place where you make that choice!

Ah well. Please skip ahead to the next parameter, FEEDBACK TYPE, read it, and then come back for an explanation of FEEDBACK TAP.

Done that? Good.

If you picked SINGLE TAP or CROSSFED modes, you've now got to tell the SP2016 where this special feedback tap should go on the "tape path." This is done in milliseconds, with a range from 10 to 1600 ms in MONO input and 10 to 800 ms in STEREO.

FEEDBACK TYPE

You have three choices of FEEDBACK TYPE: MULTITAP, SINGLE TAP, or CROSSFED.

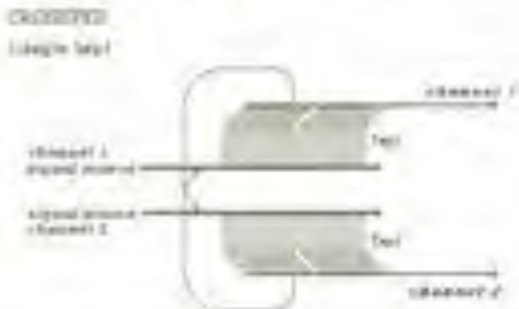
Choose MULTITAP, and all of the taps for each channel are summed together and used to send a feedback signal (the intensity of which was set by the original FEEDBACK parameter) to the beginning of the path.

SINGLE

Choose SINGLE TAP and an additional tap is slotted in at the location of your choice (set with the FEEDBACK TAP parameter).

MULTITAP

Choose CROSSFED—which you can only do if in STEREO input mode—and what you get is a SINGLE TAP in each channel, feeding its signal back to the beginning of the opposite channel. (This result is even more dramatic if the channels are SKEWED.)



PLEASE NOTE: *MULTITAP and SINGLE TAP both work fine in STEREO input, but there is no crossing of feedback from one channel to another.*

INPUT MODE

Standard choice, here, same as always, except that the decision you make has massive impact on what the other parameters will do (and which ones will be available).

SOFTKEY:

ZERO FEEDBACK

Your Panic Button. If feedback starts to get out of control, hit this immediately in order to protect your equipment.

THE MASSIVE ONE-VOICE CHOIR

(as promised)

Set up a microphone, bring MULTITAP DELAY around to the following settings, and then hum or sing long, sustained notes. A scale will do for a start, and you can wing it from there.

DELAY TAPS:	15/stereo or 30/mono
LAST TAP:	600 ms/stereo, 1200 ms/mono
ENVELOPE SHAPE:	flat
TYPE OF SPACING:	increasing 2
OUTPUT MODE:	mixed
DIFFERENTIAL DELAY:	skewed
PRE-DELAYs:	0 and 0
FEEDBACK:	–50%
FEEDBACK TAP:	695/stereo or 1300/mono
FEEDBACK TYPE:	single tap
INPUT MODE:	split mono source, feed to STEREO INPUT (otherwise signal over channel 1 in MONO)

GENERIC REVERB

BASIC DESCRIPTION:

Many less-sophisticated reverbs than the SP2016 use a little piece of mathematics called the "Schroeder Algorithm" to calculate their reverberation simulation. The SP2016 doesn't—normally—because we think the Schroeder Algorithm is too simple to sound convincing.

However, there are lots of you out there who have used it to good purpose, and will want to use it again, so we offer it here, to please you and to widen the SP2016's available palette of reverb types.

The parameters of GENERIC REVERB are identical in all ways with STEREO ROOM, except for the addition of a ROOM SIZE command. The range of this setting is 1 through 10, with larger numbers representing progressively larger rooms.

FACTORY PROGRAM NAME: F26

INPUT MODE: STEREO

BANDWIDTH: 16 kHz

PARAMETERS:	RANGE:	DEFAULT:
DECAY TIME	.1 to 10 seconds (.1, .2, .5, and one second steps)	2.0 sec
FRONT/REAR	14 steps	6 steps
ROOM SIZE (small to big)	1 to 10	5
PRE-DELAY	0 to 100 ms. (in 2 ms. steps)	0 ms
LOW FACTOR (lo freq scaling)	−8 to 2	−4
LOW ROLLOFF (rolloff point)	50 to 500 Hz (in 50 Hz steps)	300 Hz
HIGH FACTOR (hi freq scaling)	−8 to 0 −6	
HIGH ROLLOFF (rolloff point)	1.0 to 8 kHz (in 500 Hz steps)	8.0 kHz

SOFTKEY FUNCTIONS:	DEFAULT:
DISABLE INPUT alternating with ENABLE INPUT	disable

RMX SIMULATION +

BASIC DESCRIPTION:

The RMX SIMULATION + program simulates two popular AMS RMX16 effects programs: REVERSE REVERB and NONLIN REVERB. RMX SIMULATION + is a dual channel program which includes both of these effects and adds a third effect, NATURAL REVERB. This program can be operated in stereo mode, with two separate reverb channels and independant control over decay time, diffusion and predelay, or it may be operated in mono, like an AMS.

PARAMETERS:	RANGE:	DEFAULT:
CH 1 DECAY TIME*	50-375 ms.	200
CH 1 PRE-DELAY*	0-400 ms.	0
CH 1 DIFFUSION*	0-98	78
CH 1 PROG*	REVERSE/NONLIN/NATURAL	REVERSE
CH 2 DECAY TIME*	50-365 ms.	200
CH 2 PRE-DELAY*	0-400 ms.	0
CH 2 DIFFUSION*	0-98	78
CH 2 PROG*	REVERSE/NONLIN/NATURAL	REVERSE
DECAY TIME**	50-750 ms.	200
PRE-DELAY**	0-800 ms.	0
DIFFUSION**	0-98	78
PROGRAM**	REVERSE/NONLIN/NATURAL	REVERSE
MODE	16 MONO/STEREO	MONO

* Active in stereo mode
** Active in mono mode

WHAT'S GOING ON:

MODE

MODE selects the input/output mode of the program. It can be either mono or stereo. When in stereo mode, channel 1 and channel 2 are completely independant. The decay time, pre-delay and diffusion can be controlled separately for each channel, as well as the program selection. For example, channel 1 can be running REVERSE REVERB while channel 2 is running NONLIN. When in mono mode, only the channel 1 input is active. Channel 1 output is the effect without diffusion, channel 2 output is the effect with diffusion.

It should be noted that while in mono mode the parameters are in effect the channel 1 parameters. If parameters are changed while in mono mode, you will find the channel 1 parameters changed on returning to stereo mode. Conversely, if you change the channel 1 parameters while in stereo mode, these same changes will appear when returning to mono mode.

DECAY TIME

This parameter selects the decay time, with 750 milliseconds maximum decay in mono mode and 375 milliseconds maximum decay in stereo mode.

PRE-DELAY

PRE-DELAY selects the amount of delay before the effect, with 800 milliseconds maximum delay in mono mode and 400 milliseconds maximum delay per channel in stereo mode.

DIFFUSION

This parameter determines how diffuse the output will sound, with 0 as no diffusion and 98 as maximum diffusion. DIFFUSION has the effect of 'smoothing out' the decay of the reverb.

PROGRAM

This selects which decay pattern is in effect: REVERSE, NONLIN, or NATURAL. REVERSE REVERB gives a decay pattern which increases with time, opposite to that of a natural reverb. NONLIN has a decay pattern which is random, maintaining a more or less constant intensity during the decay time. NATURAL sets a decay pattern that decreases with time, hence the name of natural. All of these decay patterns have the characteristic of a sharp cutoff of the decay period. This gives these reverbs a 'gated' sound.

HI DENSITY PLATE

BASIC DESCRIPTION:

Similar to PLATE REVERB, but simulating the effect of a bigger, heavier plate. As with PLATE REVERB there are two slightly different outputs being driven by one input, creating a rich pseudo-stereo.

The parameters and operation of the two programs are identical; only their sound varies.

FACTORY PROGRAM NAME: F29

INPUT MODE: MONO

BANDWIDTH: 16 kHz

PARAMETERS:	RANGE:	DEFAULT:
DECAY TIME	0.2 to 100 seconds (exponential steps)	2.5 sec
PRE-DELAY	0.0 to 999.9 ms	0.0 ms

SOFTKEY FUNCTIONS:	DEFAULT:
DISABLE INPUT alternating with ENABLE INPUT	disable

TIMESCRAMBLE

BASIC DESCRIPTION:

This effect slices your incoming MONO signal into small pieces and reassembles them as you direct.

FACTORY PROGRAM NAME: F2C

INPUT MODE: MONO

BANDWIDTH: 16 kHz

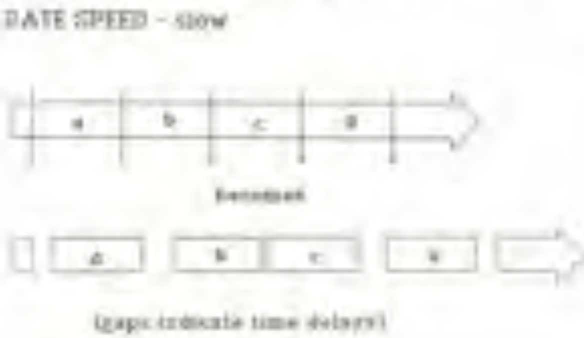
PARAMETERS:	RANGE:	DEFAULT:
SPLICE RATE (slow to fast)	1 to 50	20
SPLICING MODE	fixed or randomized	randomized
RATE TYPE (splicing mode)	randomized or fixed	randomized
LENGTH (segment size)	random or fixed	random
VOICE (number of output voices)	two, three, or five	two

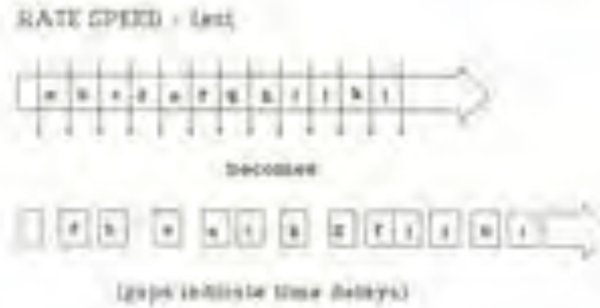
SOFTKEY FUNCTIONS:	DEFAULT:
REPEAT ON alternating with REPEAT OFF	repeat on

WHAT'S GOING ON:

SPLICE RATE

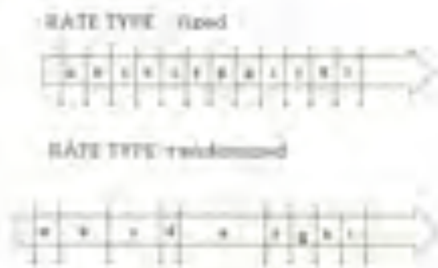
Step one is to chop up the incoming signal. That's where SPLICING comes in. The higher the rate (maximum is 50) the smaller the pieces this audio Cuisinart will spit out at you.





RATE TYPE

FIXED or RANDOMIZED! That is, do you want your knives to cut at one speed or lots of different ones? Pick FIXED and the speed you set in SPLICE RATE is what you get. Pick RANDOMIZED and it will bounce around that figure in its own unpredictable way.



LENGTH

Again, FIXED or RANDOMIZED! Do you want the pieces to be of equal length, or realio trulio tattered!

VOICE

Your choice of VOICE determines how "layered" the scrambled sound is going to be. Each voice is subject to random delays on output, utterly random, so the more voices you add the more jumbled the effect.

SOFTKEY:

REPEAT ON/REPEAT OFF

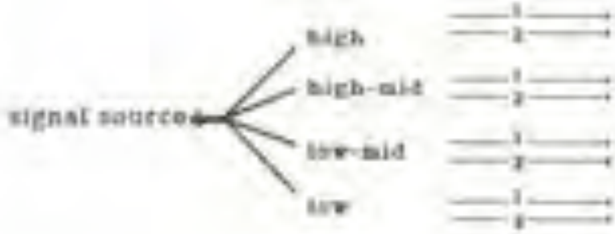
Triggering REPEAT ON will make the current scramble keep on randomizing itself forever—or until you toggle it off with the SOFTKEY, anyway.

BAND DELAY

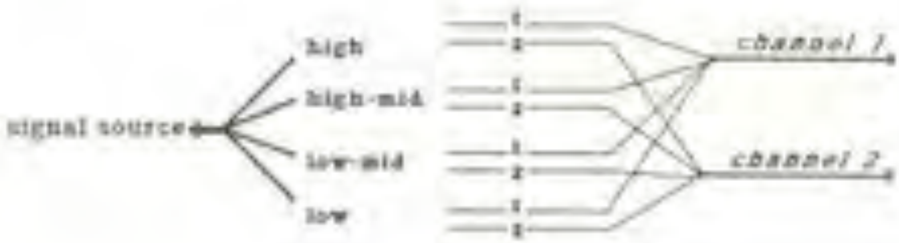
BASIC DESCRIPTION:

BAND DELAY splits an incoming MONO signal into four different frequency bands.

Each band is sent out over two separate delay lines (for a total of eight lines altogether).



The delayed band outputs are then combined and played back over two separate output channels as indicated below.



FACTORY PROGRAM NAME: F30

INPUT MODE: MONO

BANDWIDTH: 18 kHz

PARAMETERS:	RANGE:	DEFAULT:
CH 1 OUTPUT SEQUENCE (1/LO, 2/MD, 3/MD, 4/HI)	1234 TO 4321 (23 varying orders)	1234
CH 1 OUTPUT SPACING	0.0 to 150.0 ms.	50.0 ms.
CH 1 OUTPUT PRE-DELAY	0.0 to 150.0 ms.	50.0 ms.
CH 2 OUTPUT SEQUENCE (1/LO, 2/MD, 3/MD, 4/HI)	1234 to 4321 (23 varying orders)	4321

CH 2 OUTPUT SPACING	0.0 to 150.0 ms	83.3 ms
CH 2 OUTPUT PRE-DELAY	0.0 to 150.0 ms	83.3 ms

SOFTKEY FUNCTIONS: DEFAULT:

COPY CH 1 TO CH 2 ———

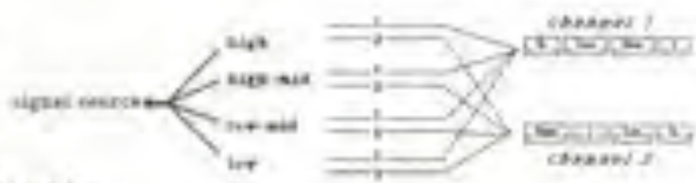
COPY CH 2 TO CH 1

CH 2 TRACKING ON alternates with CH 2 TRACKING OFF

WHAT'S GOING ON:

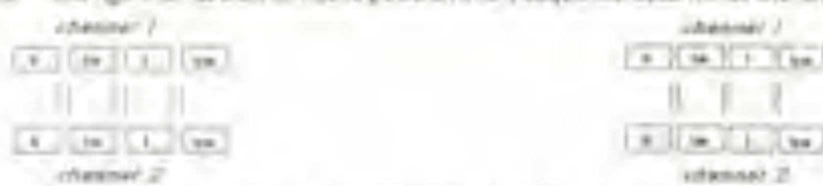
CH 1 OUTPUT SEQUENCE

The numbers in the display—1,2,3, and 4—represent the four different audio bandwidths, from low to high. By moving the ADJUST/SELECT slider here you alter the order in which they will be played back. An order of 4-3-2-1, for example, will play back high/high-mid/low-mid/low, as opposed to one of 3-2-4-1, which will play back high-mid/low-mid/high/low.



CH 1 OUTPUT SPACING

This parameter sets the time interval separating the playback of the split-off bandwidths from each other. It's actually a multiplier—this figure times a bandwidth's position in the sequence determines the length of playback delay.



CH 1 OUTPUT PRE-DELAY

Our old friend PRE-DELAY again; this sets the amount of time between the beginning of the original signal and the beginning of the effect.

CH 2 OUTPUT SEQUENCE, SPACING and PRE-DELAY

Exactly as described for channel 1, above, except that they send an independently sequenced, spaced, and delayed version of the signal out over channel 2, instead.

SOFTKEY:

COPY CH 1 TO CH 2

This SOFTKEY duplicates all channel 1 settings on to channel 2.

COPY CH 2 TO CH 1

This SOFTKEY command does precisely the opposite of the above.

CH 2 TRACKING ON/CH 2 TRACKING OFF

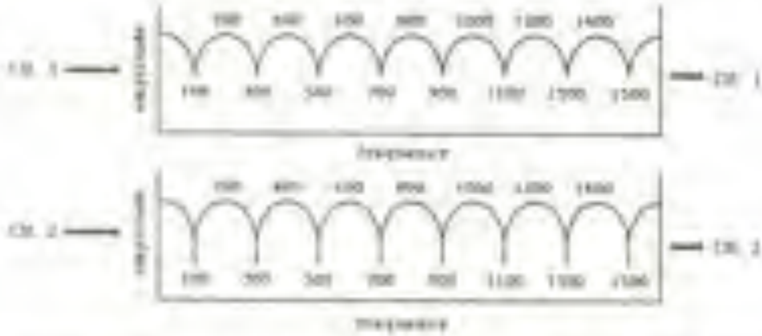
And this one, finally, allows you to toggle channel 2 back and forth between its own setting and a duplicate of channel 1's.

MUSICAL COMBS

BASIC DESCRIPTION:

Meet two comb filters in one package. What's a comb filter? It's what you get when two or more sets of delay line taps, each with a fixed spacing, are put together. Vary the spacing of the taps, and you change the filter's resonant frequencies.

It's called a "comb" because testing its frequency response reveals regularly spaced peaks and valleys, like the teeth on a comb. Simple. A typical comb filter might have peaks at 400, 800, 1200, and 1600 Hz, and nulls at 200, 600, 1000, and 1400 Hz, continuing on out to the cutoff frequency of the processor.



MUSICAL COMBS can be used to "tune" non-musical material such as noise and other rich sound sources, so that they have a working pitch. It does this by emphasizing harmonically related frequencies at the peaks of the comb filters.

FACTORY PROGRAM NAME: F32

INPUT MODE: MONO

BANDWIDTH: 16 kHz

PARAMETERS:	RANGE:	DEFAULT:
UNISON (musical interval)	-12 to + 12 (in semitones)	unison
INTENSITY (number of taps)	1 to 30	
TUNING (center freq of scale)	0.500 to 2.000 (1.000 = 220 Hz)	1.000
CH 1 PRE-DELAY	0.1 to 200.0 ms	25.1 ms
CH 2 PRE-DELAY	0.1 to 200.0 ms	40.1 ms

SCALE TYPE	12 note scale major scale harmonic minor scale natural minor scale full tone scale	12 tone

SOFTKEY FUNCTIONS: DEFAULT:

CH 2 RECIPROCAL	———
CH 2 SET TO UNITY	
CH 2 TRACKS CH 1	

WHAT'S GOING ON:

UNISON PLUS/MINUS

Moving the ADJUST/SELECT slider at this parameter sets the musical interval, in semitones, that will be created between the original signal and the filtered sound. At UNISON there is no pitch difference.

INTENSITY

INTENSITY sets the number of delay taps in each comb filter. The larger the number (maximum is 20), the sharper the "teeth" of the comb, and the greater the tonality of the outputs. Lowering this setting will reduce the sense of fixed pitch.

TUNING

You are creating a scale with your comb filters; that's how you can use the UNISON command to establish intervals. But another command, TUNING, is required to tell the SP2016 just where you want that scale located in the musical scheme of things. The number you put in here will establish the center frequency of your chosen scale type. It defaults to approximately 220 Hz, which is A below middle A.

PRE DELAY

There are two PRE DELAYs, one for each output channel. You can set each one independently from 0 to 200 milliseconds. This will aid you in creating and coloring pitch offsets and textures.

SCALE CHOICE

If you want to tune to intervals in different scales than the standard 12 note variety, you change that default setting here: the other available types are the major, harmonic minor, natural minor and full tone scales.

SOFTKEY:

CH2 RECIPROCAL

Normally the tuning of channel 2 tracks that of channel 1, and their only differences are in PRE-DELAY. But pressing this SOFTKEY command causes channel 2 to to go down one note for each note up that channel 1 is tuned, in mirror fashion.

CH2 SET TO UNITY

This SOFTKEY command sets channel 2 to UNISON.

CH2 TRACKS CH1

DUAL ROBOTS

BASIC DESCRIPTION:

Time for your classic science fiction special effect. DUAL ROBOTS gives you two different independent "robotlike" sounds. Music played through these will sound harsh and steely. Voices like . . . well, like robots. Metallic, monotonous, but pretty much intelligible. (Remember the Cylons on *Battlestar Galactica*?)

Each channel provides separate tuning, intensity, and variable frequency comb filters. PRE-DELAY is also adjustable, allowing for a gap between original and processed signals that aids considerably in maintaining clarity.

DUAL ROBOTS differs from MUSICAL COMBS in that it has two inputs and tap spacing that is "continuously" variable, instead of being locked into musical intervals.

FACTORY PROGRAM NAME: F33

INPUT MODE: STEREO

BANDWIDTH: 16 kHz

PARAMETERS:	RANGE:	DEFAULT:
CH 1 PRE-DELAY	0.1 to 200.0 ms	0.1 ms
CH 1 TAPS	1 to 20	20
CH 1 SPACING	2.0 to 20.0 ms	5.0 ms
CH 2 PRE-DELAY	0.1 to 200.0 ms	0.1
CH-2 TAPS	1 to 20	20
CH-2 SPACING	2.0 to 20.0 ms	5.0 ms

SOFTKEY FUNCTIONS:	DEFAULT:
COPY CH 1 TO CH 2	———
COPY CH 2 TO CH 1	

WHAT'S GOING ON:

CH 1 PRE-DELAY

PRE-DELAY sets the length of time between when the initial signal comes in over channel 1 and the start of the "robotic" digital delay effect.

CH 1 TAPS

Changing the number of signal taps in the comb filter will change the intensity of the effect, the more, as they say, the scarier.

CH 1 SPACING

How far apart you choose to space the taps will change the tuning of the the effect.

CH 2 PRE-DELAY, TAPS, and SPACING

All settings function exactly as described for channel 1, above, except that they independently affect signals coming in over channel 2.

SOFTKEY:

COPY CH 1 to CH 2

This SOFTKEY command makes all the settings of channel 2 the same as those for channel 1.

COPY CH 2 TO CH 1

This SOFTKEY command does precisely the opposite of the above.

LONG DELAY

BASIC DESCRIPTION:

LONG DELAY is a MONO input delay line that uses up the full memory of the processor to provide two separate outputs, each with over three seconds of delay, to give you what is essentially a three-pulse echo with extreme variance in time. A SOFTKEY repeat command lets you make a snatch of recorded signal repeat continuously.

FACTORY PROGRAM NAME: F40

INPUT MODE: MONO

BANDWIDTH: 8 kHz

PARAMETERS:	RANGE:	DEFAULT:
CH 1 OUTPUT DELAY	0.0 to 3276.5 ms	200.0 ms.
CH 2 OUTPUT DELAY	0.0 to 3276.5 ms.	300.0 ms.

SOFTKEY FUNCTIONS:	DEFAULT:
REPEAT ON alternating with REPEAT OFF	repeat on

WHAT'S GOING ON:

CH 1 AND 2 OUTPUT DELAYS

The signal from INPUT 1 can be delayed from 0 to slightly over three seconds with this parameter, separately, for each of these outputs.

SOFTKEY:

REPEAT ON/REPEAT OFF

Pressing SOFTKEY latches onto the delayed signal and repeats it constantly, like a simple loop. Switching noise may make this difficult to use at first, until practice teaches you how to use it most effectively.

DUAL DELAY

BASIC DESCRIPTION:

DUAL DELAY is a stereo version of LONG DELAY. The only significant differences are that it has double the bandwidth (shortening each channel's delay range to a max of 800 ms.), two inputs, and a wider range of SOFT-KEY continuous repeat functions—in this version you can choose whether you want channel 1 to loop and repeat, or channel 2 ... or both.

FACTORY PROGRAM NAME: F41

INPUT MODE: STEREO

BANDWIDTH: 10 kHz

PARAMETERS:	RANGE:	DEFAULT:
CH 1 DELAY	0.0 to 800.0 ms.	50.0 ms.
CH 2 DELAY	0.0 to 800 ms.	75.0 ms.

SOFTKEY FUNCTIONS:	DEFAULT:
CH 1 REPEAT ON alternating with CH 1 REPEAT OFF	repeat on
CH 2 REPEAT ON alternating with CH 2 REPEAT OFF	
CHANNEL 1 + 2 REPEAT ON alternating with CHANNEL 1 + 2 REPEAT OFF	

DUAL DIGIPLEX

BASIC DESCRIPTION:

DUAL DIGIPLEX is two separate digital delay lines operating in stereo, each with two different taps. The result is a complex mix of echoes that can give the illusion of spatial movement.

FACTORY PROGRAM NAME: F42

INPUT MODE: STEREO

BANDWIDTH: 16 kHz

PARAMETERS:	RANGE:	DEFAULT:
CH 1 PRE-DELAY	0.0 to 800.0 ms.	50.0 ms.
CH 1 LOOP DELAY	0.0 to 800.0 ms.	100.0 ms.
CH 1 FEEDBACK	–99.9 to +99.9% (of total feedback)	–50.0%
CH 2 PRE-DELAY	0.0 to 800.00 ms.	200.0 ms.
CH 2 LOOP DELAY	0.0 to 800.00 ms.	250 ms.
CH 2 FEEDBACK	–99.9 to +99.9% (of total feedback)	49.9%

SOFTKEY FUNCTIONS:	DEFAULT:
ZERO FEEDBACK	———

WHAT'S GOING ON:

CH 1 PRE-DELAY, LOOP DELAY and FEEDBACK

The signal coming in over channel 1 can be delayed up to 800 milliseconds by the PRE-DELAY setting before it makes it out of the processor and through the OUTPUT 1 jack. And if LOOP DELAY and FEEDBACK were at 0, that's all that would happen. But LOOP DELAY allows you to set a time at which that signal will regularly repeat, from 0 to 800 ms.; and FEEDBACK sets how long the repeats will continue before fading away. Because the FEEDBACK is all digital, you can have a lot of repeats without accumulating noise in the echoes.

The higher the FEEDBACK percentage, the longer the repeats will continue. A negative setting will get your brighter, crisper echoes, a positive setting will increase the low frequencies and round out the tone, softening it (although the longer the delay time the less important FEEDBACK polarity is, and by 50 ms. it's essentially irrelevant.)

CH 2 PRE-DELAY, LOOP DELAY, and FEEDBACK

As described above, but on an independent channel which can be set to different times than channel 1, creating extremely complex and rhythmic echo patterns.

SOFTKEY:

ZERO FEEDBACK

This SOFTKEY command shuts off all feedback, leaving only the initial echo of each delay line audible.

LONG DIGIPLEX

BASIC DESCRIPTION:

LONG DIGIPLEX is a MONO, reduced bandwidth version of DUAL DIGIPLEX. The reduced bandwidth is the price of extending the delay length out to over three seconds.

Aside from these differences, though, it operates exactly like its stereo counterpart.

FACTORY PROGRAM NAME: (4)

INPUT MODE: MONO

BANDWIDTH: 8 kHz

PARAMETERS:	RANGE:	DEFAULT:
CH 1 OUTPUT PRE-DELAY	0.0 to 3276.5 ms	100.0 ms
CH 2 OUTPUT PRE-DELAY	0.0 to 3276.5 ms	200.0 ms
LOOP DELAY	0.0 to 3276.5 ms	2000.0 ms
FEEDBACK	−99.9% to + 99.9% (of total feedback)	49.9%

SOFTKEY FUNCTIONS:	DEFAULT:
ZERO FEEDBACK	

LOSSLESS ROOM

BASIC DESCRIPTION:

Nothing ever leaves the LOSSLESS ROOM — not unless you let it out, anyway. You can let sounds in to reverberate briefly, or lock them up almost forever: a place for hauntings and mysterious chants and space washes of sound.

(and we bet you guessed we think it's neat)

It also works just a bit differently than the others. We'll cover the details after the spec list.

BUG NOTICE: *although LOSSLESS ROOM comes up in STEREO INPUT mode, when you try and use the program it shifts to MONO INPUT mode. So LOSSLESS ROOM can only be used as a MONO program, and you should completely ignore the INPUT parameter.*

FACTORY PROGRAM NAME: #31

INPUT MODE: 16 kHz

BANDWIDTH: MONO

PARAMETERS:	RANGE:	DEFAULT:
DECAY RATE	milliseconds, one second, seconds, minutes, hours	hours
HI FREQ ABSORPTION	1 (none) to 8 (max)	4
INPUT	stereo, mono (0 & 1)	stereo (see note above)

SOFTKEY FUNCTIONS:	DEFAULT:
ENABLE INPUT alternating with DISABLE INPUT	enable

WHAT'S GOING ON:

When you activate this program, it isn't on yet. Ready to go, yes. Triggered, no. To trigger it you press the SOFTKEY, which ENABLES INPUTS and lets your signal into the LOSSLESS ROOM.

BE CAREFUL! The default decay is HOURS, so if you are playing a lot of loud notes when you activate the room you could overload it and get distortion. Much better to 1) activate it, set the decay time to the level you want by changing the first parameter, and then play; or 2) play little and lightly at the moment of activation until you get a feel for the room's input limits.

Once you are in the room, you can change its character radically (with the parameter controls) without ever having to leave it. When you do want to get out, just press SOFTKEY again.

PLEASE NOTE: leaving the LOSSLESS ROOM with SOFTKEY doesn't turn off any existing reverb. *Because of this you can get a massive ambience going, disable the inputs, and then play new and unreverberating material against the existing ambience. To turn off that hanging ambience you must either adjust the DECAY TIME parameter or leave the program.*

DECAY RATE

This parameter sets how long it takes sound to decay inside the room. The choices are HOURS, MINUTES, SECONDS, ONE SECOND, and MILLI-SECONDS.

At HOURS, the decay is so long you should play sparsely to keep from overloading the unit. At MILLISE-CONDS the decay is so fast you'll only notice a slight "rounding," a softening, of the input signal. And in between ... well, experiment and find the setting that works best for you. SECONDS is particularly good for providing a floating ambience that doesn't obscure the input. (An effective way to do the same with HOURS is to keep the effect and the original signal on different tracks, balancing them against one another by adjusting relative volume and EQ.)

You can use the ADJUST SELECT slider to conveniently control the LOSSLESS ROOM. Any time the signal builds up beyond the level you want, just clear it out by shifting to a faster decay time.

HI FREQUENCY ABSORPTION

This setting determines how the high frequencies in the reverb are going to die away. At 1 they don't get absorbed at all and the ambient effect is quite bright. At the other end of the scale, 8, they fade so fast that the ambience acquires a considerably darker and muted timbre.

INPUT

This parameter should be ignored. See the notice under Basic Description for details.

SOFTKEY:

ENABLE INPUT/DISABLE INPUT

This SOFTKEY is first used to turn on the LOSSLESS ROOM, and after that toggles between turning it off and back on again.

GATED REVERB

BASIC DESCRIPTION:

GATED REVERB combines a high quality digital reverb with a programmable digital noise gate. While this effect is most often used on snare drums, it is applicable to all transient-rich program material. The noise gate's amplitude threshold and attack/release time constants can be adjusted allowing for a wide variety of dynamic effects. A complete palette of reverb control parameters, identical to those found in STEREO ROOM, is also provided.

GATED REVERB accepts a MONO input signal and returns a pseudo-stereo output.

INPUT MODE: MONO BANDWIDTH: 16kHz

PARAMETERS:	RANGE:	DEFAULT:
THRESHOLD	0 to 99	50
GATE ATTACK	0 to 99	75
GATE RELEASE	0 to 99	50
REVERB DECAY	.2 to 99 seconds (.1, .2, .5 and one second steps)	4.0 sec
PRE DELAY	1 to 250 ms (in 1ms steps)	2ms
LOW FACTOR (lo freq scaling)	−8 to +4	−4
LOW ROLLOFF (rolloff point)	50 to 500Hz (in 50Hz steps)	150Hz
HIGH FACTOR (hi freq scaling)	−8 to 0	−2
HIGH ROLLOFF (rolloff point)	1.0 to 8kHz (in 500Hz steps)	8.0kHz

SOFTKEY FUNCTIONS	DEFAULT:
DISABLE INPUT alternating with ENABLE INPUT	DISABLE
CLEAR REVERB alternating with RESUME REVERB	

WHAT'S GOING ON:

THRESHOLD sets the amplitude at which the signal opens and closes the gate. This adjustment is both important and somewhat tricky. Important because if set incorrectly there will be no gating or no signal. Tricky because it is dependant on a number of variables: the dynamics of the signal, the reverb decay time and the input signal level. Note that if the input signal level is changed, the threshold should be readjusted.

GATE ATTACK/RELEASE parameters allow the response time of the digital gate to be modified over a wide range (0-SLOW/99-FAST). Fast attack/slow release settings tend to sound natural, somewhat restoring the dynamics of the original signal. A wide variety of interesting effects can be achieved with other attack/release combinations.

part six: EXPANDING THE SYSTEM

CONTROL VOLTAGE/TRIGGER

No programs make use of the CV feature at this time. The Trigger input works the same way as the front panel softkey.

INSTALLING NEW ROMS

If you can use a screwdriver, you can customize or upgrade your SP2016. All you have to do is add to (or change) its complement of PROGRAM ROM chips. A range of optional program ROMs are available directly from Eventide.

ROM stands for "Read Only Memory." These are integrated circuits that can be permanently imprinted with programs. In this case, all the computer commands and signal processing tricks that make up factory programs like DUAL DIGIPLEX or STEREO ROOM or MULTITAP DELAY.

What ROMs you have will determine what factory programs your SP2016 can run.

Here's how to change them by yourself:

1) IMPORTANT FIRST STEP: Turn the unit off and unplug it from the outlet. This will prevent needless accidents. (ROMs are easily damaged by voltage surges.)

2) Remove the SP2016 from its mounting so that you have access to the front half of the unit's top panel.

3) Using a Philip's head screwdriver, take out the six screws holding the front section of the top panel in place. Put them someplace safe. Be careful not to lose the screws; they're small.

4) Lift the front section off and place it to one side.

5) Touch the metal chassis of the SP2016 to ground out any static charge you might have. THIS IS IMPORTANT. Do this *before* you touch any of the exposed electronic components, otherwise you might damage some of them. (Static carries a lot bigger zap than you might guess, and ROMS in particular are susceptible to damage from it.)

6) Examine the row of ROMs that are now exposed.

ON THE CARE AND FEEDING OF ROM CHIPS

DON'T TOUCH THE ROMS UNTIL AFTER YOU'VE TOUCHED THE SP2016 CHASSIS (TO PREVENT STATIC DAMAGE)

BEFORE PLACING THEM IN THE SP2016, PLACE THEM ON THE PROCESSOR'S METAL LID (SAME REASON)

WHEN STORING OR SHIPPING THEM, INSERT THEM INTO CONDUCTIVE FOAM OR WRAP THEM IN ALUMINUM FOIL.

DON'T BEND THE LITTLE LEGS ON THE CHIPS.

IF YOU DO BEND THE LITTLE LEGS BY ACCIDENT, DON'T PANIC. INSTEAD, GENTLY AND CAREFULLY BEND THEM BACK AS CLOSE TO STRAIGHT AS POSSIBLE. UNLESS THEY ACTUALLY BREAK OFF, ODDS ARE YOU'LL BE ALL RIGHT.

7) ... and especially take a good look at the mounts holding them in place. These are ZIF (Zero Insertion Force) sockets. They are called that because they let you insert and remove ROMS without having to push or pull on the chips themselves, thus eliminating any risk of bending the legs.

ZIF sockets work by clamping onto the legs from the side. Look at a ROM that is already installed. You will see, on the side of the ROM towards the back of the unit, a little plastic bar marked OPEN.

8) To open the socket and release that ROM, just take your small screwdriver and gently lever the OPEN bar toward the SP2016's front panel (you'll see the other end of the bar pop into view from underneath the ROM). The socket is now open, and the ROM easily lifted out by hand.

9) To put a new ROM in the socket, just drop it in, making certain of two things: first, that the little notch at one end of the ROM is facing towards the front of the SP2016, and second, that all the legs are in the socket.

10) To tighten the socket, take your thumb or screwdriver and carefully push in the socket's bar until it is all the way in on the socket's front side. The ROM will now be firmly clamped.

11) Put the lid back in place and screw it on.

12) Remount the SP2016, plug it in, turn it on, and enjoy your new program.

PLEASE NOTE: *If you've taken out one ROM so you can use another, store the first one in a safe place. You may want to use it again someday. In the case of UPGRADE ROMs, pack the old ones safely, as described above, and return them to Eventide.*

REMOTE CONTROL

We've got a tidy little remote control unit available for the SP2016, just in case you've had to place your unit somewhere inconvenient to get to (there's not so much space available within arm's reach of the mixing board).

Contact us if you want one. It plugs into the REMOTE CONTROL jack on the SP2016's rear panel, is small and unobtrusive, and duplicates all the processor controls on the unit's front panel, so that you don't sacrifice any ease of operation.

Except

Well, something had to go, or the remote control would have been as big as the front panel, and therefore pretty useless. Because your INPUT and OUTPUT levels should be set and left alone, those are not on the remote. Neither are the INPUT level bargraphs, the ALPHANUMERIC DISPLAY, or the DEFINE key.

For maximum effectiveness you do need to see the displays. So we suggest you put the SP2016 in easy sight of your work area. That way you have the information in the displays and the mobility of the remote available to you all the time.

COMPUTER CONTROL (Interface Optional)

Linking the SP2016's digital processor to a personal computer can greatly expand its already huge potential. Want to synchronize the operation of the SP2016 to the other synths and audio gear you're using? Make changes in settings and functions faster than you could do it by hand? Even create new SP2016 programs that are whole new uses for the digital signal processor, instead of variations on the factory-supplied programs? A computer makes all that possible.

We don't recommend this to the non-programming types out there, because right now it's definitely a case where you'll have to write your own programs and routines. There's nothing "off the shelf" available. But if you are knowledgeable, and are of the programming inclination...

...just take any of the following computers, equipped with the right interface cables and/or cards, plug them into the optional IEEE 488 GENERAL PURPOSE INTERFACE on the SP2016's rear panel, and go to it. (Check out the manual's glossary, too; there's a lot of basic, useful information in the definitions you'll find there.)

Here's an admittedly incomplete list of compatible machines (if your computer isn't on it, never fear; call us and we'll try to let you know if a link can be worked out):

Manufacturers	Models	Requirements
Hewlett-Packard	HP-83, 85, 87 and nearly all others	I/O ROM and interface
Apple	Apple II, II-plus, IIE, III	Require special card available from Apple and others
Radio Shack	TRS-80 model 1, 3	Require special card not available from Radio Shack
IBM	PC PC AT	Require special card

THE IEEE-488 INTERFACE

What makes it possible to link the SP2016 and these various personal computers is the use of a standard interface, the IEEE-488. This interface is also known as the "GP-IB" (General Purpose Interface Bus) and the "HP-IB," after Hewlett-Packard, the company which invented it.

The IEEE-488 interface has a number of powerful advantages. A major one is that you can throw a lot of units together with no loss of data transfer speed; the specs allow for up to 15 processors or remote devices to be controlled simultaneously from one computer. (Those of you with Eventide H949 Harmonizers units and 1745M Digital Delay Lines can take advantage of this; remote control interfaces are available for both devices. Contact us for more information.)

GETTING IT ALL TOGETHER

In order to computer control the SP2016, you need to know three things:

1) How to make your computer talk to the SP2016.

2) How to ADDRESS the SP2016.

3) What COMMANDS the computer should be giving.

The best way to learn #1 is to prowl through your computer's technical manuals. #2 and #3 we'll tackle here.

ADDRESSING

You need some way of telling the processor (or processors—remember, the IEEE-488 allows for up to 15) that you are trying to talk to it; in essence, a way of shouting "Hey, you!" that it will understand.

You do this as part of the bus protocol, in a way which will ideally be invisible to you in use. The basic trick is that whenever you send a command, you precede it with a number between 0 and 30; that's the command's ADDRESS. (More advanced computers allow you to use words as addresses, so that a command could be prefaced with the actual name of the unit—SP2016, in this case—you are sending it to, and the computer would know what to do. Obviously that is easier to keep track of.)

The DEFAULT ADDRESS for the SP2016 is 15. Whenever the SP2016 sees that number coming in over the bus, it knows to pay attention to the next command. But because more than one processor might be on the bus at one time, you might need to change that default to some other number. Here's how:

1) Successively press the COMMAND key until BUS ADDRESS appears in the display window.

2) Move the ADJUST/SELECT slider until NEW ADDRESS XX appears in the display. (XX will be the new default address.)

3) When the number in the display is right, press the EXECUTE key.

4) Turn the POWER OFF, wait ten seconds, then turn it ON. Don't skip this step—it locks the address in, so you can't accidentally change it by remote control (which would effectively make the unit deaf to all further outside commands.)

This new default address will remain until the battery runs out or you change it, whichever comes first.

If you've lost track of what the current default address is, just go through steps 1) and 2) above. When the number of the current default shows up in the display it will be accompanied by the words OLD ADDRESS.

CONNECTING

Now that the SP2016 knows what address to respond to, it's time to connect it to the computer. Use a standard IEEE-488 interface connector and whatever extra electronics your computer requires on its end.

CAUTION: Don't connect or disconnect any equipment (including the SP2016) while power is on in any of the units in the bus network.

CAUTION: The SP2016 uses Metric fasteners (black). Do not use English screws—you could damage your threads. Only use connectors with black retaining screws ... and, as when connecting anything to anything, be gentle.

TALKING BACK AND FORTH

Now that the SP2016 is hooked up and knows its address, you can use the computer to tell it what to do. There are two kinds of commands: STANDARD and RESERVED. The STANDARD commands basically duplicate the SP2016's front-panel functions. The RESERVED commands control internal SP2016 functions (and require the use SPUDsystem™, software from Eventide which makes it possible to create sound processing programs for the SP2016 yourself).

We've deliberately kept the subset of IEEE-488 commands being used on the simple side, so that even computers without full IEEE-488 compatibility can be used for outside control of the processor.

All the basic operations are available through the commands READ and WRITE.

WRITE

If you tell the computer to perform a WRITE command, it:

1) sends an ADDRESS message out over the bus, alerting the appropriate processor to listen up;

2) sends the command itself, followed by a "delimiter" (a signal which tells the processor the command is over);

3) sends a signal which tells the processor it can stop listening to the bus now.

READ

If you tell the computer to perform a READ command, it:

1) sends out over the bus an ADDRESS message that tells the appropriate processor it's supposed to talk.

2) configures itself to listen to whatever the processor says.

3) receives a signal (followed by a delimiter) from the processor.

4) Upon receipt of the delimiter, it throws an "okay, go back to not talking" signal at the processor . . . and proceeds with its program, whatever that may be.

DELIMITERS

A computer is a brick. There's no intelligence in there, and unless it is told when a message is ended, it will listen forever and nothing will ever get done. Which is why there are delimiters. Remember the old telegrams that said STOP at the end of each sentence? That's a perfect example of a pre-computer delimiter.

Standard delimiters include use of a CR (carriage return) followed by a LF (line feed), and the use of a special line in the IEEE-488 cable called the "EOI" ("End or Identify"). Because not all computers fully implement using the EOI signal, the SP2016 has been designed to handle incoming and outgoing messages in the following ways:

When SENDING: on receiving a "TALK" signal, the SP2016 sends a message followed by—in order—a CR (decimal 13) and an LF (decimal 10). At the same time as the LF character is sent, the EOI line in the IEEE-488 cable will be asserted.

When RECEIVING: the SP2016 considers a message ended when it receives an LF character. It ignores all CR signals and the EOI line.

THE SP2016 COMMANDS

The following commands, when sent to the SP2016 through the IEEE 488 bus, will control its operations.

The syntax shown for all commands is that of the Hewlett-Packard HP 85; it's one of those we use here in the shop. You will have to adapt the commands as necessary for your own brand of computer.

Some useful information:

- You can put spaces within the commands ("CO" is the same as "C O" or even "C   O") but not within data fields.

- You can use either lower or upper case interchangeably.

- Non-delimiter characters after a command string are ignored, and data must be delimited by either a comma or a space.

- Any command that returns data does so as a sequence of ASCII characters in the selected radix.

- Hex output format right-justifies the data and zero fills a 4-character field.

- All output is terminated by a CR LF sequence, with EOI asserted by the LF command.

- The 6809 outputs a string at the end of every command. If the most recent command doesn't generate any output, the process sends back "OK." If the controller doesn't read the string in the output buffer and another command is issued, the original string is lost without any error being generated.

CODE	DEFINITION	EFFECT
front panel key control commands		
CO	COMMAND	Same as pressing COMMAND key
CO *n*	COMMAND *n*	Display command *n*
PR	PROGRAM	Same as pressing PROGRAM key
PR *n*	PROGRAM *n*	Display program *n*
PA	PARAMETER	Same as pressing PARAMETER key
PA *n*	PARAMETER *n*	Display parameter *n*
SK	SOFTKEY	Same as pressing SOFTKEY
DE	DEFINE	Same as pressing DEFINE key
EX	EXECUTE	Same as pressing EXECUTE key
OU	OUTPUT	Same as pressing OUTPUT MONITOR button
DC	DEC COMMAND	Scroll back on the Command list
DP	DEC PROGRAM	Scroll back on the Program list
DR	DEC PARAMETER	Scroll back on the Parameter list
front panel fader control commands		
SF *n*	SET FADER *n*	Move ADJUST SELECT fader to position *n*
SI 1 *n*	SET IN 1 *n*	Set channel 1 input level *n* (0-255 range)
SI 2 *n*	SET IN 2 *n*	Set channel 2 input level *n* (same)
SO 1 *n*	SET OUT 1 *n*	Set channel 1 output level *n* (same)
SO 2 *n*	SET OUT 2 *n*	Set channel 2 output level *n* (same)
SM 1 *n*	SET MIX 1 *n*	Set channel 1 effects mix *n* (0=dry, 255 =full effect)
SM 2 *n*	SET MIX 2 *n*	Set channel 2 effects mix *n* (same)
commands that send data to controller		
RK	READ KEY	Send any key presses to remote
RM	READ MODE	Send system mode bytes
RL	READ LEDS	Send status of front panel LEDs
RF	READ DEFINE	Send the current define string
RD	READ DISPLAY	Send the text in the display
RI	READ LIMIT	Return number of parameter fader entries
RP *n*	READ PARAM *n*	Send the value of parameter *n*
RS	READ SERIAL #	Send the unit's serial # as 4-decimal digits
RN	READ NAME	Send device identifying string "SP2016"
RV	READ REVISION	Send software revision string (i.e., 2.1)

general control commands

LD *n*	LOAD PROGRAM *n*	Load Program n from Program list.
SP *n m*	SET PARAM *n m*	Write new value m to Parameter n
RE	REMOTE	Disable all front panel controls.
LO	LOCAL	Restore front panel controls.
BL	BLANK	Inhibit any further O.S. writing to screen.
UB	UNBLANK	Allow display writing from O.S. and remote.
WR "..."	WRITE "text"	Clear display and display text.
DM	DECIMAL	Change base of data I/O to decimal.
HE	HEX	Change base of data I/O to hexadecimal.
KE *n*	KEY *n*	Send RESERVED command key.

appendices

1. A/D ADJUSTS

All front panel faders are digital controls. They send out a voltage, from 0 to 5 volts, which is digitized by a multiplexed 10 bit Analog to Digital (A/D) converter. (Multiplexing means that a single A/D converter is shared by the various analog signals.) The A/D is also used to measure various audio levels.

For proper operation the A/D must be adjusted so that it precisely tracks the 0 to 5 volt analog signal range. A 5 volt input should convert to 1111111111 (ten ones). If the A/D is misadjusted, you're likely to run into some unexpected behavior on the part of the faders or level indicators: it may be impossible to find all the installed programs using the ADJUST/SELECT fader; parameters may not vary over their full range; the faders may seem to lose some resolution. Obviously, none of this is desirable. If you encounter any of these problems, try adjusting the A/D.

Here's how:

1. Set INPUT LEVEL fader to MAX. *(channel 1 only)*
2. Set all other faders to minimum—the ADJUST/SELECT fader should be all the way to the left.
3. Remove the small top panel. You'll find two printed circuit boards immediately behind the front panel: a short board in front of a longer one.
4. Find the pair of test points along the top right edge of the long board.
5. Find the small trim pot behind the SOFTKEY button (actually it's slightly to the right) on the top of the long board. The trim pot can (and should) be turned with your fingers.
6. Short the test points (a copper coin will do nicely). The leftmost bargraph should light.
7. Keep shorting the test points, while you turn the trim pot until all but the top bargraph segment are lit.
8. Turn the trim pot slightly, until the top segment is continuously lit (no flicker).

That's it. Put the cover back on and keep the penny.

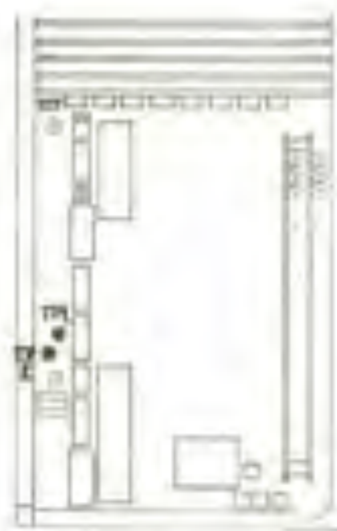
2.THE I/O MODULE

All audio inputs and outputs on the SP2016 are contained in a sub-chassis, the I/O Module, which is located on the left rear side of the main chassis, under the PD44 board (see Getting into the Box). This system protects the analog circuitry, and permits some flexibility in I/O formats, both analog and digital.

The model AD22A I/O module is a self-contained, two channel, analog-to-digital, digital-to-analog subsystem. Linear PCM techiques are used to ensure transparent performance, free of the nonlinearities introduced by other schemes, e.g. floating point, companding, etc.

The model AD22A features:

* two independent I/O channels
* balanced input/output
* 16 bit LINEAR PCM analog to digital converters
* 16 bit digital to analog converters
* selectable sampling rate: 20.5 kHz or 41 kHz
* input anti-aliasing filters for both 8 kHz and 16 kHz input signals
* output deglitching filters for both 8 kHz and 16 kHz input signals
* programmable audio level controls



In order to reach the I/O Module you will have to remove both top covers (see "Getting into the Box" in *Preventive Maintenance*). Remember to keep track of the small screws! Lift the PD44 board. On the right side of the chassis is the power supply. Do not open it! The box on the left hand is the I/O module. Remove the screws and lift off the cover. You will see eight printed circuit boards which handle various tasks. There are three kinds of adjustments you can make to the AD22A.

note: refer to the adjustments in step one (page A/3) before proceeding.

BARGRAPH ADJUSTMENT

If the bottom segments of the input level bargraph light when no input signal has been applied, you'll need to adjust the Input Monitor Bargraph. Note: *this adjustment is somewhat temperature sensitive, so it is best done when the SP2016 is at operating temperature.*

To make BARGRAPH adjustments you will need to locate the trim pot on circuit board PA54. With the rails facing you, the PA54 board will be the first of the standing boards which faces you. The trim pots are in the lower left corner of the board. Now load Loop Edit program and then load Dual Delay.

1. Set both input level faders to OFF.

2. Use a narrow flathead screwdriver to adjust the upper trim pot on circuit board PAS4 until the bottom LED of channel 1's input bargraph flickers.

3. Back-off the trim pot adjustment until the LED turns completely off.

4. Repeat steps 2 and 3 for the second input channel using the lower trim pot on PAS4.

PLEASE NOTE: The BARGRAPH adjustment affects the OFFSET adjustment of the A-to-D CONVERTER, which could result in more noticeable 'pops' when switching programs. Therefore, you should readjust the A-to-D after completing this adjustment.

A-to-D CONVERTERS

Each input channel of the AD22 has its own A-to-D converter mounted on the AD-1 boards. There are four trim pots on each board, one each for input LEVEL and OFFSET, and a pair for DISTORTION adjustment. Be particularly careful in adjusting the OFFSET trim pot; a misadjustment can cause loud 'pops' when you switch effects programs.

To make A-to-D adjustments, open the SP2016 and remove the top from the I/O module. Then:

1. Locate the the AD-1 boards: with the front of the SP2016 facing you, they're the standing boards on the right side of the module; channel 1's AD-1 will be the left-hand board and channel 2's AD-1 will be the right-hand.

2. Apply a 1kHz sine wave to the appropriate rear panel input XLR connector. Set the corresponding input level fader so that the top bargraph segment is constantly illuminated. (Unit in Dual Delay program.)

3. Set the output LEVEL fader to MAX and the ouput MIX fader to full EFFECT.

4. Observe the signal level at the appropriate output and adjust the rear-most trim pot for an 8 volt peak-to-peak level.

5. Look at Test Point 1 (TP1) on PA 563 board, a digitized sine wave should be observed. Turn input level fader to minimum position and the sine wave should decrease in amplitude until no signal is observed (a straight line), with no d.c. offset. If a d.c. offset is observed it can be zeroed by using the offset pot on AD-1 (second from rear trimpot). For CH2 use TP2.

6. Decrease the input signal level by 40dB and observe the output signal on an oscilloscope (a distortion meter can be used for this adjustment but isn't necessary). Adjust the fourth trim pot, a ten turn type, while observing the output signal. You will notice that at either extreme of the pot's rotation the signal will become distorted. This distortion should disappear for several turns near the middle of the pot's range. Set the pot near the middle of this acceptable range.

7. Increase the input signal level to approximately 3dB below clipping and measure the distortion of the output signal. Adjust the third from the rear trim pot for minimum distortion.

You repeat this procedure for the other channel, using the other AD-1 board.

D-to-A CONVERTERS

Each DAC has a trim pot for adjusting distortion. They're located on circuit board PA563. Each converter provides two outputs: IC 13 outputs a pair of 16kHz bandwidth signals, and IC 14 a pair of 8kHz signals. A trim pot is located to the left of each DAC, which can be adjusted to minimize distortion, as follows:

1. Locate circuit board PA563; it's the board that lies flat along the bottom of the I/O Module.

2. Apply a 1kHz sine wave to the appropriate rear panel input XLR connector. Set the corresponding input level fader so that the top bargraph segment is constantly illuminated.

3. Set the output LEVEL fader to MAX and the output MIX fader to full EFFECT.

4. Adjust the trim pot located to the left of IC 13 for minimum distortion.

5. Load the LONG DELAY program and adjust delay parameter for minimum delay.

6. Adjust the trim pot to the left of IC 14 for minimum distortion.

3. PREVENTIVE MAINTENANCE

Most of the circuitry of the SP2016 will never require adjustment. Still, there are a few checks and some routine maintenance you may want to perform from time to time.

LOW BATTERY

The SP2016 uses an internal battery to maintain storage of the user-created presets when the unit isn't on. It is continually recharged by use, so you may get a LOW BATTERY message on the front panel if you haven't turned the SP2016 on for a long period of time.

If you get a LOW BATTERY message for no obvious reason you should check the battery to make sure no chemicals are leaking. The battery is located on the front of the PC-24 board; to get to it you will have to remove both pieces of the unit's lid. (See "Getting Into The Box", below.)

FAN INTAKE

The SP2016 uses forced-air cooling, and the fan intake should be inspected every six months (or more often, if your studio is prone to dust). If there is a significant collection of dust, clean it up. The fan must not get blocked or the resultant heat increase might damage the SP2016.

MECHANICAL COMPONENTS

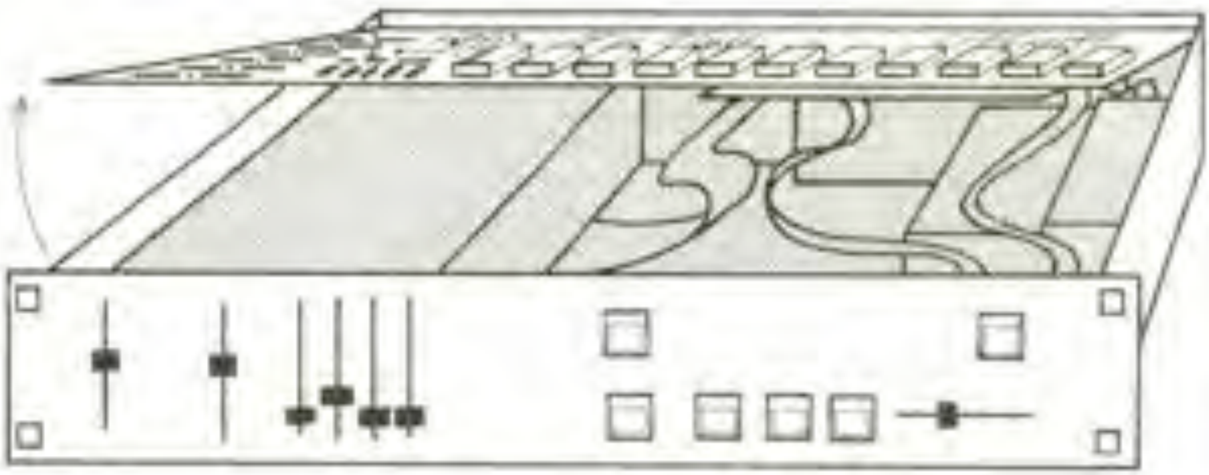
The SP2016 has been designed to be stable and rugged— but it is not proof against sudden stops, awkward roadies, and other studio hazards. If you move the SP2016 around a lot, it should be inspected periodically for signs of loose hardware or other physical damage.

PLEASE NOTE: *The SP2016 is not designed to be rack-mounted only by its front panel; if you hang the unit without rear support sooner or later the front panel will succumb to (very expensive!) metal fatigue.*

GETTING INTO THE BOX

You may need, from time to time, to get to the circuitry of the SP2016. Here's how:

If you're changing PROGRAM ROMs, only the front section of the top cover needs to be removed; this is described in part six of this manual. However, if the SP2016 is indicating an error and you need access to ICs or other circuitry, you will probably need to remove both top covers. Use a small Phillips-head screwdriver; remove all of the screws from the top of the unit, and lift off both cover plates. (Keep track of the screws—they're small and can get lost easily.)



Quick tour: the PD44 board is the large board at the top of the unit. It is immediately visible when the top covers are removed. The PC-24 board (see "Low Battery", above) is mounted to the underside of the PD44 board. In order to reach that board or to de-solder ICs from the PD44 board, both top cover plates must be removed, allowing the PD44 board to swing upward on its hinges.

To free the PD44 board, remove the three screws located on the front edge of the board. One is in the right hand corner, one is in the left hand corner and the last about two inches in from the left hand edge. The front edge of the PD44 will pivot upward and back. Below the PD44, on the bottom right hand side of the chassis, is the Power Supply.

WARNING! WARNING!

Do not remove the power supply shield! The Power Supply is not a user serviceable component.

Dangerous voltages are isolated by the shield!

No voltages higher than normal line voltages exist outside of the screened off area.

Across from the Power Supply is the I/O module. This houses the Analog-to-Digital and Digital-to-Analog converters which handle audio signals going into and out of the SP2016. For more information on adjustments to the I/O Module, see The AD22A I/O Module earlier in this section.

4. PROBLEMS (REAL AND IMAGINED)

Some problems aren't problems.

Sometimes it's just that you've missed a step, or a power surge has caused the circuitry in the SP2016 to "lock up", or perhaps an external cord has gotten worked loose. However, if you think you have a problem, check the list below and try the remedies suggested. If things don't clear up, go through the "Self Tests" described in the *Troubleshooting* section.

If you have gone through the Self Tests and still have not turned up the bug, re-examine all external connections and use the LINE OUT command to bypass the unit. This is accomplished by depressing the COMMAND key until LINE IN appears in the display window, and moving the ADJUST/SELECT slider until the display shows LINE OUT. This switches the internal relay to its bypass position. If there are still problems when the unit is bypassed, then the SP2016 is not at fault.

Finally, if you do find a problem, remember to unplug the SP2016's power cord *before attempting any repairs*. Better safe than sorry!

COMMON PROBLEMS

- Power surges or intermittent power may cause the SP2016's circuitry to "lock up". Simply turn the power off for a few seconds, then turn it back on, to correct it.
- If you hear a slight clicking when you use the front panel sliders, don't worry. This is normal for digitally operated controls of this type, and is explained fully in part two of this manual.
- You might get a display message that reads "E11 A/D WAIT". This is a selftest message which occurs when the microprocessor power-up sequence has been disturbed—rapidly flipping the power switch on and off will do it. To fix the problem, turn the SP2016 off, wait three seconds, and power it up again.
- If, with no signal sent through the processor, the bottom segments of the input level bargraph display flicker, or stay on continually, you'll have to make an adjustment to input monitor bargraph. See *The AD22A I/O Module* for information on how to make that adjustment.
- If you notice excessive switching noise when you load a program, check "Analog to Digital Converter Adjustment" in *The AD22A I/O Module* to find out how to correct the problem.
- If the ADJUST/SELECT fader no longer accesses the full complement of available programs or the full range of a given parameter, you should check for one of two causes. Either the fader is dirty and should be cleaned with contact cleaner, or the A/D has drifted out of adjustment. See *The AD Adjusts* for the adjustment routine.
- Once in a while you may come across a specific combination of program or parameter settings that doesn't work, or works very strangely indeed. You may have stumbled on a "bug" in the program or operating system! The SP2016 is powerful and versatile—and complex. Given the huge number of possible combinations of parameters, it's impossible to catch every potential bug in the operating software. However, any bugs you may encounter at this stage of systems development should be both benign (i.e. harmless) and certainly quite rare, requiring unusual settings or combinations of settings.
- If you do come across such a bug, let us know about it so we can correct the software for future updates and releases.

- If the problem only seems to crop up when the processor has been running for some time, you may have a "thermal intermittent", that is, a problem caused by temperature. The quick cure is to turn the unit off and let it cool. While it's off, check to make sure that its fan isn't blocked and there is adequate ventilation: some of the components give off a lot of heat during normal operation, and it has to have room to get out. At least an inch of clear space at the top, bottom, and back is recommended if your unit is showing signs of heat-related trouble. (Don't be concerned if the top of the box gets warm to the touch. That's normal.)

- If none of the front panel lights will come on, the SP2016 is probably not getting line power. Check the fuse in the back of the box. If the fuse is not blown, and the outlet is definitely delivering power, but the front panel lights still won't go on, or if the front panel lights work but the controls do not respond, then refer to "Getting into the Box" in the Maintenance section, for instructions on opening the unit. Check over both sides of the circuit boards, and check all wiring; look for connectors and ICs pulled slightly out of their sockets, frayed or broken wiring, foreign objects, etc.

REPAIRS

If you cannot locate the source of the problem by making the checks above, or by using the Self-Tests described in the Troubleshooting appendix, you should contact Eventide about repairs.

Before you do, make sure you have the Serial Number of your unit at hand, as well as a complete list of the symptoms of the problem, and the tests and checks you have made. The more information you give us, the more quickly and easily we can resolve the problem.

The Serial Number is on the rear panel plate. You can also cause the SP2016 to flash it on the display by pressing the COMMAND key until CONFIGURATION is visible, then pressing EXECUTE. But keep your eyes open—the Serial Number goes by pretty fast.

PLEASE NOTE: *Eventide does not require a formal return authorization. However, equipment returned for service without a complete and detailed written trouble report will not be accepted for repairs.*

The information in your trouble report may prove to be critical in helping us locate and fix the problem (especially if it happens only intermittently, or under specific circumstances). Also, a detailed trouble report helps us make certain we have fixed all manifestations of a problem. The last thing either you or we want is for you to get your unit back in anything less than 100% working order. Please help.

5. TROUBLESHOOTING

The SP2016's diagnostic Self-Tests will isolate and identify possible problems within the processor's circuitry. This allows on-location servicing in cases where test equipment is not available.

These tests are contained within a specific SELF-TEST ROM. If you do not have it, then your SP2016 will not perform the tests. If you do have it, then how it functions will depend upon which ROM socket you place it in.

- Place the ROM in the socket second from the right (as you face the front of the unit), and the SP2016 will perform a standard self-test every time you turn it on.

- Place the ROM in the socket all the way the left and the SP2016 will do nothing but Self-Test. All other functions are unavailable until the ROM is moved.

- Place the ROM in any of the other sockets and the SP2016 will only perform the self-test when commanded to.

A standard self-test begins with the display message "**SELF TEST**". After a few seconds this message will be replaced by another that indicates the number of programs and presets currently available. Finally, the name of the program last in use will be displayed.

If the Self-Test routine turns up any problems, a display message or indicator light will tell you so (some examples: BAD IC27/PC24 or LOW BATTERY).

A message such as BAD IC27/PC24 is a code which identifies the suspect IC (in this case, IC number 27) and the printed circuit board on which it is located (in this case, PC number 24).

The LOW BATTERY message example is self-explanatory.

There are two different Self-Test routines available in the SYSTEM COMMANDS. They freeze operation of the SP2016 only briefly, and if you interrupt a program to run a Self-Test it will resume from that point as soon as the tests have been completed. The one exception: ANY STORED AUDIO DATA IN THAT PROGRAM (such as delay loops) WILL BE LOST.

TESTING—THE SHORT FORM

To run a Short Self-Test, press the COMMAND key until SHORT SELFTEST is displayed, then press the EXECUTE key. This series of tests is similar to the power-up routine.

If you encounter any problems in running the test then turn the SP2016 off, check all power connections, turn the unit back on, and try again.

TESTING—CONTINUOUSLY

The Continuous Self-Test command contains a whole range of tests which review specific sections of internal circuitry. To set up the Continuous Self-Test routine, and to display a list of the tests available, do the following:

1) Press COMMAND until CONTINUOUS TEST is displayed.

2) Press EXECUTE to display TEST START.

3)Use the ADJUST/SELECT slider to scan the list of tests available. These are:

TEST START
TEST ALL
TEST NONE
SKIP KEYS
TEST ANALOG CPU
TEST CPU RAM
SKIP PRESET RAM
TEST OS ROMS
TEST PROG ROMS
TEST ADR CNTR
TEST R/W LATCH
TEST PROGRAM LATCH
TEST TEST PORT
TEST ALU
TEST ALU-LSS
TEST CACHE LATCH
TEST D RAM LATCH
TEST CACHE RAM
TEST DATA RAM

TO TEST EVERYTHING BUT FRONT PANEL KEYS AND PRESET RAM

1)Call up TEST START in the display.

2)Press EXECUTE.

TO TEST EVERYTHING INCLUDING FRONT PANEL KEYS

1)Call up TEST ALL in the display.

2)Press EXECUTE.

3)Adjust the slider until TEST START appears in the display.

The Test routine will prompt you to press each of the front panel keys in turn. (This panel test is performed only once: all other tests are performed continuously.)

TO SELECT A SPECIFIC SET OF TESTS

1) Call up TEST NONE in the display. Press EXECUTE.

2) Now, as you move through the list of tests with the slider, each test will be preceded by SKIP-- If you wish to perform that particular test, press EXECUTE. SKIP will change to TEST.

3) When you are through selecting tests, move the slider left to display TEST START, and hit EXECUTE.

WHAT THE TESTS DO

The first four entries in the list of tests, above, have just been explained or are self-explanatory. Here is a list of the circuitry tested by tests 5—19.

TEST ANALOG CPU

The ANALOG CPU controls the analog module, including the input and output gain control circuitry, the IN/OUT relay, the front panel controls, and the panel indicators. Selecting this test tells this CPU to initiate its own self test routine, which reports its results to the main CPU.

TEST CPU RAM

This test checks the random access memory used by the main CPU.

TEST PRESET RAM

The Preset RAM is the area in which User Presets and other "permanent" data are stored. Whenever the data changes, a new "checksum" is calculated from the data present and compared with the checksum that is actually stored. During Continuous Selftest, the data are actually transferred to a different RAM, after which the Preset RAM is exercised. After this test, the original data are restored. WARNING: INTERRUPTING IN MID-TEST COULD RESULT IN LOST PRESETS.

TEST OS ROMS

The operating system ROMs each have a checksum stored in them. This is compared with a newly calculated checksum, and any error is reported.

TEST PROGRAM ROMS

Each Program ROM also has a checksum. This is compared with the newly-calculated version.

TEST ADR COUNTER

This test checks that the address counter is operating correctly for the various possible processor bandwidths, and that the 6840 chip on the PC-24 board is functioning correctly.

TEST R/W LATCH

This test checks the Data Latch, the Read Latch, and the WRMODE Latch. A failure of this test displays DATA PATH FAULT. If this test fails, subsequent failure messages in the Processor RAM tests may be invalid.

TEST TEST PORT

This test checks the ability of the controller to read data from the ALU (high speed processor).

TEST ALU

This test checks that the most significant 16 bits of the ALU and the high speed multiplier are working correctly.

TEST ALU-LSS

The accuracy of the least significant 9 bits of the ALU are checked in this test.

RAM TESTS

The processor contains two sets of data memory: Dynamic RAMs and Cache RAMs. The Cache is faster in operation and stores data needed for rapid calculation. The Dynamic RAMs are somewhat slower, but hold much more data. These Dynamic RAMS give the processor its ability to act as a long delay line. The following four tests verify RAM operation.

TEST CACHE LATCH

This tests the latch that reads/writes data to the CACHE RAM.

TEST D RAM LATCH

This tests the latch that reads/writes data to the Dynamic RAM.

PLEASE NOTE: *If either latch test fails, a corresponding failure message for the RAM portion may not be valid. Make sure the latch is fixed before troubleshooting any RAM chips.*

TEST CACHE RAM

This test checks the Cache RAM for gross defects and bad bits.

TEST DATA RAM

This test checks the Data (Dynamic) RAMS for gross defects and bad bits.

IF YOU FIND TROUBLE

Almost all Self-Test errors indicate a significant problem with the SP2016. However, because of the nature of signal processing, some errors do not indicate a serious problem. For example, continuous testing over many hours or days may turn up "bad bits" in the Cache or Dynamic RAMs. A truly "bad bit" (that is, one that consistently gives an error reading on every pass) indicates that the chip should be replaced. If the bit is in one of the more significant RAM locations, the problem may show up audibly as a periodic clicking noise. However, if you get a random error indication only after many hours of testing, it simply means that an alpha particle or other random disturbance may have affected one bit of the RAM. If the error doesn't repeat, or other errors with no pattern occur very infrequently (every few hours to every few days) it is probably nothing to be concerned about.

All the other tests should always pass. These circuits should not be subject to any random disturbances, and failures do indicate problems.

If a test does fail, repeat it to confirm that it wasn't just a power glitch. If seemingly unrelated tests fail frequently and randomly, suspect a common cause such as power supply problems or intermittent connectors.

Once a problem has been found in the circuitry, the suspected component can be located by pressing the COMMAND key and reading the displayed code:

BAD ICxx *yyyy* indicates a bad integrated circuit, where xx is the number of the IC and *yyyy* represents the designation of the circuit board where the IC will be found. The IC numbers are screened on the tops of the circuit boards. Socketed ICs will be easy to replace; the others require the use of a soldering iron and a desoldering tool, or wicking, to pull solder away from the IC leads without excessive heating. Local electronic repair or supply houses should be able to provide advice on desoldering techniques, as well as the necessary equipment. Always be aware of the orientation of the IC being replaced, by observing which direction the indentation at the end of the IC should go.

WARNING! WARNING!

The new IC must be mounted in the same direction as the one being replaced. Failure to do so will probably destroy the replacement IC, and can seriously damage the processor!

Most of the ICs are standard industrial parts, but DO NOT replace an IC with anything but an identically numbered equivalent. Certain ICs are critical and must be replaced with another IC from the same manufacturer as the original. These ICs are listed below, with codes for acceptable manufacturers as follow:

N = National Semiconductor
TI = Texas Instruments
F = Fairchild
AMD = Advanced Micro Devices
M = Motorola

Component Number	Industry Number	Manufacturer
T2	74HC374	M
A2	2901C	AMD
A3	2901C	AMD
A4	2901C	AMD
A5	2901C	AMD
A6	2901C	AMD
A7	2901C	AMD

The printed circuit code listed in the error message will usually be PD44 or PC24. These are the two main circuit boards in the Signal Processor. Access to these boards is described in the *Preventive Maintenance* appendix.

PD 44

6. GLOSSARY

ANTHROPOMORPHISM — The tendency of human beings to ascribe human qualities to non-human species or inanimate objects.

This first entry is really a small warning to be wary of the way people normally approach computers, electronic equipment, and musical instruments. We all say "the computer thinks such and such" or "the SP2016 will *forget it* ..." because it's convenient. Just don't get to believing it. A computer doesn't think; you do. That's your edge.

It also happens to be an edge that cuts both ways. Don't rely on the SP2016 to understand things you don't, or to remember things you haven't told it. And if something seems to go wrong, check what you've done before you assume there's a problem with your unit.

The following are musical and computer terms you may find in this manual. Some of them you may already be familiar with, though not necessarily with the way they are used here, or in regard to signal processing or music synthesis. You will find some of the information here particularly useful if connecting your SP2016 to a personal computer, as described in part six.

ADC — Analog-to-Digital Converter. The SP2016's ADC circuit converts incoming analog signals to digital signals which the processor then manipulates.

ANALOG CIRCUITRY — Circuitry which carries signals of indeterminate voltage. Analog signals can transfer smoothly between voltages without having to remain within set ON and OFF levels (as digital circuitry must).

ASCII CODE — American Standard Code for Information Interchange. A standard set of characters by which all data (other than address commands) is sent between the SP2016 and computers or other remote controllers. Each character has a unique value which corresponds to a recognized binary bit pattern.

BANDWIDTH — The range of frequencies which can be reproduced or processed accurately. The 8kHz/16kHz LEDs on the front panel show the current bandwidth of the SP2016.

BINARY — Numeric system with two digits, 0 and 1. All computers and microprocessors deal in bits, or *Binary digiTs*, 0 and 1.

BUS — A system used to convey information between different sections of electronic equipment, particularly computer circuitry. The SP2016 is capable of interfacing with the IEEE-488 bus, which is in common use in the industry. Data within the processor is also carried on busses, such as the DATA bus, the 6809 address bus, etc.

BYTE — A binary number made up of eight binary digits (0 or 1). Bytes make up the language the processor or computer uses (not what *you* see in your display or monitor, but the way the computer talks to itself and programmers talk to it).

CARRIAGE RETURN — As an ASCII character, it is abbreviated as CR, and is usually part of a terminating sequence, CR/LF ("carriage return/line feed") which signals to the microprocessor that a command or piece of data is complete.

CHARACTER — Any single number transmitted over the interface bus. In this context it is usually an ASCII character, represented by a number from 0 through 127.

CLIPPING	A type of distortion which results when the the output capability of a circuit is exceeded. The waveform of the output signal will be 'clipped' when the circuit is asked to deliver a higher voltage than that which is available from the power supply. The maximum voltage at clipping is usually a factor in determining dynamic range.
COMMAND	One of the available operations obtained by pressing the COMMAND key on the front panel. Commands can also be generated by a remote controller (see Computer Control).
COMPUTER-CONTROLLER	In terms of the IEEE-488, a device which manages the interface bus. Usually a general purpose computer or remote controller designed specifically to be a bus controller. In this manual, "controller" will always mean "the device in charge of the interface bus." "Computer" will usually mean the same, but it may also refer to the microcomputer in the remote control interface itself, depending upon the context.
DAC	Digital-to-Analog Converter. These circuits take the data generated by the SP2016's circuitry and convert it back to the normal analog representation of sound, i.e., an analog audio signal.
DATA	The information passed between the computer and the remote control interface. Each piece of data (datum) is a byte. A string of bytes, usually terminated with a CR/LF, forms a complete message or command, which is then implemented by the computer or processor. DATA also refers to the digitized audio signal acted upon by the processor ALU circuitry.
DECIBEL	A ratio used in comparing sound levels. 20 dB corresponds to a 10 to 1 voltage ratio.
DEFAULT VALUES	The factory-preset values which will be used for any setting when no new values have been specified.
DIGITAL	Information represented by a set of numbers, or digits. Standard digital information is a code made up of only two numbers, 0 and 1, representing the ON and OFF states of electronic circuits. Digital coding breaks information up into discrete steps, instead of continuous spectrums (unlike analog encoding).
DRY	When referring to sound, unprocessed. The SP2016 uses the MIX faders to determine the proportion of Dry signal to Effect signal at each output.
DYNAMIC RANGE	The ratio (in decibels) of the highest output before clipping to the lowest usable signal level. The dynamic range will usually be a good indication of noise levels, and in digital audio equipment, will indicate the degree of accuracy achieved in 'digitizing' the audio signal. See Digital, DAC, ADC.
EOI	"End or Identify". This is a particular signal in the IEEE-488 scheme, and may be used optionally to indicate the completion of a data or command transmission. See Line Feed.
EPROM	Erasable Programmable Read Only Memory. This is the type of memory used for storage of the processor's factory programs, and for the operating system. An EPROM can be recognized by the small quartz windows on top of the IC (integrated circuit) chip. The window allows ultraviolet light to erase the contents

of the memory chip, so do not remove the labels from the EPROMs in the SP2016. The memory cells in the chips retain information until specifically erased.

FACTORY PROGRAM NUMBER — An identifier assigned to the Factory Programs which are included with the SP2016. The User Presets, which are modified Factory Programs, are assigned their own numbers, but remain "tagged" to the number of the original program. That original Factory Program Number appears as an "F-Number" when the DEFINE key is pressed with the program name in the window.

FORMAT — Describes how data are sent. For example, some computers send a number with two digits following the decimal point (e.g., the number 123 would be transmitted as "123.00".) The remote control interface has certain rules for interpreting numbers, and violations of these rules can lead to incorrect interpretations. Most computers have internal facilities for formatting outputs. If yours does not, it may be necessary for you to do so explicitly.

GPIB (HP-IB) — General Purpose Interface Bus, also known as the Hewlett Packard Interface Bus, the implementation of the IEEE-488 1975 standard.

INTEGER — A number, specifically any number comprised of the digits 0 through 9, without any decimal points or fractions. In communicating with the SP2016 remote control, all integers of interest will be positive.

LEADING ZEROS — A leading zero in a number increases the length of the number without changing its value. Example: 00012 = 12. Trailing zeros do change the value of a number: 12000 is not 12! CAUTION: decimal points are ignored by the remote unit, so that 12.000 is evaluated as 12000.

LINE FEED (LF) — A signal represented by an ASCII character, which acts as a "terminator" for the remote control interface. When the interface receives a "LF" character, it knows that the computer is through sending data and that it may act on what it has received.

LOADED — A program has been Loaded when it is in the operational portion of the SP2016's program memory. When a program is Loaded, it is being Executed; that is, its steps are being carried out sequentially.

MACHINE LANGUAGE — The "lowest level" at which one may communicate with most computers. Most machine instructions comprise from 1 to 4 bytes. The internal remote control program is in machine language, stored in ROM, or Read Only Memory. If your computer cannot cope as-is with IEEE-488 protocol, it may be necessary to program the computer in machine language. A typical BASIC instruction (BASIC is a "high level" language) can involve from ten to many thousands of machine language instructions.

MEMORY — The portion of the SP2016 that stores programs and data, in this case, the audio signal itself. For instance, up to 3.2 seconds of data is stored in numerical form in the more than 64000 memory locations of the processor data memory.

MONITOR — This button allows you to check the output level of the processor. Pressing this button connects the level measuring circuitry to the two outputs. Pressing the button again reconnects it to the input. The input is also automatically reconnected after about ten seconds.

MOST SIGNIFICANT DIGIT	That digit which appears first in a transmitted stream of ASCII data, or appears at the left of a written or printed representation is the MSD. For instance, "1" is the most significant digit of 12345.
NOISE FLOOR	All electronic equipment generates a certain amount of "noise" which is unavoidably mixed with the desired signal. The processor's noise floor is dictated by the limited resolution of the AD and DA converters. For best operation, it is desirable to keep the signal as high as possible with respect to the noise floor.
OPERATING SYSTEM	The "central program" in a computer, necessary for maintaining control over each program which runs in the computer. The operating system in the SP2016 was designed for control of sound, and is conveniently packaged in EPROMs. These can be interchanged if it ever becomes necessary to update the system or adapt it to other purposes, such as laboratory analysis.
PARAMETER	A characteristic or value, usually represented numerically, which varies with the circumstances of its application. For example, a program designed to produce delayed repeats would have two parameters, Delay and Feedback. DELAY, represented in milliseconds, governs the length of time between repeats. FEEDBACK, a ratio, governs the length of time it takes the signal to decay by a given amount.
PORT	A physical and electrical location that allows one access to a given signal or group of signals. Usually used when referring to a computer signal bus connector.
PROCESSOR	A group of electronic circuits capable of performing various mathematical operations on the digital representation of an analog signal. These operations are either analogous or equivalent to the corresponding operations performed by resistors, switches, capacitors, etc.
PROGRAM	A list of instructions which are to be followed by a computer. They may be as simple as "Set Delay to 0 and Then Stop", or far more complex giving many conditions under which delay, level, mode, etc., may vary. An Application program is one written by the user (YOU) to achieve some desired result with the processor, such as "Continuously Read and Log the Front Panel Settings vs. Time and Then, at Some Later Time, Reproduce Them under Computer Control." Programs are written in the language characteristic of the computer being used. Typical languages are BASIC and PASCAL, as well as various dialects of machine language. Examples in this manual are given in the BASIC "dialect" used by the Hewlett-Packard mode 83, 85, and 87 computers. The examples are usable, usually with minor changes, by most BASIC language computers, and, conceptually, by any computer with standard architecture.
RAM	Random Access Memory. Unlike ROM, data in RAM is lost when power is removed. RAM is used primarily for storing the actual audio signal (after it has been digitized) and for storing the program that the processor is currently executing.
RESET	Resetting a computer is usually a last-ditch effort to get it to behave. This is sometimes necessary after it has tried to carry out an illegal instruction (something it cannot do, but is too literal minded to know is impossible). If all else fails, the simplest Reset is to turn the power OFF, wait several seconds, then turn it ON again.

ROM

Read Only Memory. This is a type of integrated circuit (IC) on which the data and instructions for the microprocessors are stored. The ICs in the SP2016 are ultra-violet erasable, programmable read-only-memories, or EPROMs. The data in the EPROM cannot be lost except under extraordinary conditions, such as exposure to excessive voltage or short-wave ultraviolet light. There is no ordinary operation or sequence of data that can cause the EPROM to "forget" and so it is impossible to erase the factory programs, operating system, or remote control instructions under normal operating conditions, no matter what programming mistakes may be made. Other ROMs in the processor typically contain a small number of instructions that must be accessed very rapidly by the ALU.

SOFTWARE

Computer programs. The programs executed by the processor are generally considered "firmware" because they are not wired into hardware, but are not as readily altered as software in RAM.

SYNTAX

The individual elements (words, characters, numbers, etc.) in a software command, and the order in which they must occur for the command to be read without confusion or error. While you might tell a human operator "Give me 30 milliseconds of delay" or "set the delay at 30 milliseconds," and be perfectly understood, a computer, reading the number 30 before "delay", might think that "30" referred to an earlier and quite unrelated command. Instructions to a computer, therefore, must have the proper Syntax; if they are not, unexpected (but not unpredictable) operation will result.

TERMINATOR

A signal that lets the computer know a command or piece of data is complete, so that the computer can start working on it. Because the absence of data will not imply the completion of a command, all commands must be explicitly terminated. The Line Feed (LF) character serves as a terminator for most computers and for this remote control. EOI (see above) may be employed optionally. The word Delimiter is a synonym for a termination character.

USER NUMBER

A number for a user program. It can be chosen by the user or will be assigned by the operating system of the processor. This provides a specific identification which eliminates doubt about which program is in memory or is about to be deleted.

USER PRESET

A user-modified version of any of the Factory Programs supplied in ROM by Eventide. The version is created by the User, using his values of all the parameters. When a user program is loaded, it is the equivalent of loading a factory program after the parameters have been set.