THE URSA MAJOR
MSP-126
OWNER'S MANUAL

JANUARY 1985

WARRANTY REGISTRATION & T-SHIRT OFFER

We are pleased and grateful that you have chosen an URSA MAJOR product for your use. We will stand behind your MSP-126 and hope that it will serve you well. Please fill out this form completely to register your MSP-126 for its one year warranty (refer to the Owner's Manual for warranty details). The information you provide will help us serve you better. When we receive this completed form from you, we will send you a complimentary URSA MAJOR T-shirt. It has our sleeping bear logo and name on the front.

Specify T-shirt size:

Small_____Medium_____
Large_____X-Large_____

Where do we send the shirt?
Name:______________________________
Company Name:________________________
Street Address:_______________________
City/State/Zip:_______________________
Country:____________________________

Serial Number of your Unit:

____

Dealer Name:_________________________
Street Address:_______________________
City/State/Zip:_______________________
Country:____________________________

How did you first hear about the MSP-126?________________________

What's your application? recording_______broadcast___________
live performance________sound reinforcement________home_________
other____
Describe in a few words how you use it________________________

Which modes do you find most useful? why?_______________________

With what sources?_______________________________________

What do you like about the MSP-126?___________________________

Any suggestions for other products?___________________________

Please mail the completed form to:

URSA MAJOR, Inc.
Box 28
Boston MA 02258
USA

MSP-126

MULTI-TAP STEREO PROCESSOR

OWNER'S MANUAL

By
Christopher Moore

January 1985

Last Revision: 1985 January 14

CORPORATE STATEMENT

URSA MAJOR is dedicated to producing useful, lasting products of superior value for professional audio users, and to building a satisfying, equitable workplace for its people.

ACKNOWLEDGEMENT

I would like to thank the following URSA MAJOR workers for their support, direct and indirect, during my development of the MSP-126. They are:

Gerard Abeles
Mark Bruckner
David Goldstein
Lauren Weisman
Charles Anderson
David Drolette
Donna Kallman

Thanks also to XyQuest Inc for their XyWrite word processor. XyWrite made this manual a pleasure to produce.

Sincerely,

Christopher Moore

The MSP-126 is made in the USA by URSA MAJOR, Inc, Box 28, Boston MA 02258 USA. tele: 617 924 7697 telex: 921405 URSAMAJOR BELM

TABLE OF CONTENTS

INTRODUCTION

A Warranty Registration Form has been included in the front of this manual. Please fill it out and return within 2 weeks of receiving your 126. The warranty for your unit is only valid if we receive this completed form promptly.

********************* IMPORTANT! ************************

IN ORDER TO KEEP YOU INFORMED OF SOFTWARE UPDATES AND OTHER CHANGES OR ENHANCEMENTS TO YOUR 126 IN THE YEARS AHEAD, WE MUST HAVE YOUR WARRANTY FORM IN OUR FILE!

****

DESIGNER'S STATEMENT

William Blake coined the phrase "to find a universe in a grain of sand." The 126 has a seemingly narrow province: 370 milliseconds of memory, with one input and two multi-tap outputs. This is far simpler than my other digital processor designs optimized for reverberation synthesis, yet it contains a fascinating world of possibilities.

The 12 tap algorithm of the 126 is more complex and, to my mind, far more interesting than that most popular of delay devices, the digital delay line or "DDL". With only one tap, the DDL is unable to achieve convincing stereo perspective. However, with two groups of six taps, the 126 delivers rich, full stereo in all its signal processing modes.

Although all listening experiences are stereo, in modern recording, broadcasting, film sound recording, and even in musical composition and performance, the closely miked or purely electronic sources are severely monaural.

I believe that a richer, more natural kind of stereo ambience is vital in all these media. That is why I've based my MSP-126 design on a stereo time delay algorithm with multiple taps. Even in the 126's effects modes, where the sound processing is deliberately dramatic, the result is satisfyingly stereo. I believe you will find these modes new, exciting, and extremely musical.

SPECIFICATIONS

Bandwidth: 20kHz, 44.1kHz sampling rate

Noise: 15 bit PCM conversion for 80+dB dynamic range

Modes: 8 Processing Modes

Bypass Function: Front button or rear panel jack

Size: Rack mount, 1 unit high (19" (48cm) wide, 1.75" (4.4cm) high), and 11" (28cm) deep, excluding XLR connector protrusion. Weight, fully boxed for shipment, approximately 12 lbs (5.5kg).

INPUTS

Stereo or mono Input Modes, switchable by rear panel switch. In stereo Input Mode, the processor is fed from the sum of left and right inputs, and with Bypass "on", the left input feeds the left output and the right input feeds the right output. In mono Input Mode, the processor is fed from the left input only, while with Bypass "on" the left input feeds through to both left and right outputs.

The inputs are electronically balanced (differential amplifier): Pin 3 high, Pin 2 low, and Pin 1 ground. In stereo Input Mode, input impedance of pin 3 is 11K ohms and pin 2 is 21K ohms. In mono Input Mode, input impedance of pin 3 is 5.5k ohms, pin 2 is 10.5k ohms. Recommended source impedance is 600 ohms or less.

Maximum source voltage before input stage overload is +17dBV (7Vrms). Minimum input for operation of "0 LED" (Input Level fully clockwise, input frequency 100Hz, mono

Input Mode, left input only driven), is -10dBV (.316Vrms). In stereo Input Mode, with both inputs driven in phase, sensitivity rises to -16dBV (.159Vrms). Connectors are XLR-3 female.

OUTPUTS

Stereo outputs, active differential circuit. Pin 3 is high (100 ohm source resistance), Pin 2 is low (100 ohm source resistance), and Pin 1 is ground. Minimum recommended load impedance is 600 ohms. Maximum output level (Input driven so that "0 LED" flashes occasionally, Output Level full clockwise) is +12dBV nominal. Output stage maximum level is +17dBV feeding an unbalanced load, and +23dBV feeding a balanced load. Connectors are XLR-3 male.

POWER

115 or 230VAC nominal voltage (selectable via internal switch). Supplies maintain regulation down to approximately 95VAC (182VAC). Consult with factory about special version with transformer for 100/200VAC supply. Unit operates with 60 or 50Hz power line frequency. Power consumption approximately 30 watts. Detachable IEC standard power cord.

There are two internal fuses, one for the mains, and one for the +5VDC supply. All supplies are current and power limited.

The storage environment range is 10-50 degrees C operating, while the storage environment range is 0-70 degrees C. RH (relative humidity) may be up to 95% non-condensing.

INSTALLATION, POWER SOURCE, AND SETUP

UNPACKING

As soon as you receive the carton containing your 126, inspect it carefully for signs of shipping damage. Report any shipping damage to the carrier immediately and file a claim. Although in most cases we insure our shipments, it is the consignee's (ie., your) responsibility to initiate a claim for shipping damage. Save the carton and all packing material in case return to the factory is ever necessary.

POWER

The Model 126 operates on 115/230VAC, 50/60Hz. A sticker on the rear panel indicates how your unit was set at the factory.

******************** IMPORTANT ************************

IF YOU HAVE ANY DOUBTS AS TO THE INTERNAL VOLTAGE SETTING AND ARE ABOUT TO PLUG THE 126 INTO 230VAC, DON'T! OPERATION OF A 126 SET TO 115VAC ON A 230VAC SUPPLY MAY DAMAGE THE UNIT. REMOVE THE AC CORD, UNSCREW 10 SCREWS HOLDING THE TOP COVER IN PLACE, AND CHECK THE SETTING OF THE INTERNAL VOLTAGE SELECTOR SWITCH. CORRECT IF NECESSARY AND REINSTALL THE TOP COVER.

****

******************** IMPORTANT ************************

IF YOU MUST CHANGE THE POWER CORD PLUG TO SUIT THE STANDARD IN YOUR COUNTRY, BE SURE TO CONNECT THE GREEN WIRE TO THE GROUND PIN OF YOUR PLUG. THE GROUND PIN IS A SAFETY FEATURE DESIGNED TO CONNECT THE 126 CHASSIS TO POWER GROUND. DO NOT DEFEAT IT!

****

AC CONNECTOR AND INTERNAL VOLTAGE SELECTOR

The AC receptacle is a standard 3 pin IEC connector, for a detachable line cord. The center pin is tied to the 126 chassis for electrical safety reasons. A nearby sticker shows the factory setting of an internal slide switch at the time of shipment. This switch straps the transformer primaries for 115V operation (windings in parallel) or for 230V operation (windings in series). Also inside are a fuse for the mains (US type 3AG SLO BLO .75A, 1-1/4 x 1/4"), and a fuse for the +5VDC supply (US type 8AG 2.5A fast blow, 1 x 1/4").

INSTALLATION

The Model 126 may be rack mounted (height is 1.75" [4.4cm]). Be sure to maintain adequate clearance for air flow at the sides and top of the unit. One inch (2.5cm) all around is recommended. We do not recommend installing the 126 directly above a high power component such as a power amplifier. The 126 may be operated on a table or bench-top, as there are no ventilation holes in its underside.

CONNECTIONS TO THE UNIT

INPUT MODE SWITCH

This rear panel switch selects whether the 126 responds to both its inputs, or only to the left input. When the button is "out" (stereo Input Mode), the unit takes the left and right input signals, sums them, and feeds them to the processor. With the button "in" (mono Input Mode), the unit takes the left input only and sends it to the processor. The right input is ignored.

This switch also determines what happens in the Bypass mode. It is possible to insert the MSP-126 in-line in either a full stereo or a mono/stereo situation. If, for example, a broadcast feed is normally stereo, but occasionally mono, and you want to selectively replace the mono feed coming down both lines with processed stereo, you would use the 126 in the stereo Input Mode. Then, with Bypass mode "on", the stereo source material would pass through the 126 as stereo, unaltered. On the other hand, when mono material comes along, Bypass can be switched "off", and processed stereo will drop in to replace the mono source.

In the mono Input Mode, the MSP-126 can be dropped into a mono signal path in order to split it into stereo. Simply connect the source feed into the left input and push the Input Mode button "in". Now when Bypass is "on" the mono input will be bypassed around the processor and go to both left and right outputs unaltered. If Bypass is then switched "off", the output will be processed stereo.

INPUT CHARACTERISTICS

Feed the 126 Inputs from either balanced (preferable) or unbalanced sources of 600 ohm or less impedance. Maximum signal level is 7Vrms, while minimum signal for normal operation is -10dBV (316mV). If both inputs are driven with mono or in-phase stereo components, the sensitivity increases to about -16dBV.

The inputs are electronically balanced (differential amplifier): Pin 3 high, Pin 2 low, and Pin 1 ground. In stereo Input Mode, input impedance of pin 3 is 11k ohms and pin 2 is 21k ohms. In mono Input Mode, input impedance of pin 3 is 5.5k ohms, pin 2 is 10.5k ohms. Recommended source impedance is 600 ohms or less. A detailed schematic of the MSP-126 input stage is reproduced below for your reference.

IS PIN 3 "HIGH" OR IS PIN 2 "HIGH"?

We're pretty pleased with our input and output circuits. For one thing, they're compatible with balanced or unbalanced equipment. Another advantage is that you can use them in pin 3 "high" or pin 2 "high" systems. If your system has pin 2 defined as "high" or "hot", simply interchange the connections to pins 2 and 3 in any of the connection figures given here. But be sure to follow the same convention at both input and output of the 126, or you'll get an overall phase reversal.

OVERALL MSP-126 GAIN AND OPERATING LEVEL

The 126 is also versatile with regard to operating level and gain. You can adjust the Input Level control to accomodate a variety of nominal levels, from -10dBV to +17dBV. Likewise, you can then independently set the Output Level control for -10dBV to +12dBV levels. This allows for a range of gain from roughly 22dB of gain to 27dB of loss. Of course, you can also set the controls for unity gain.

No matter what you do, level matching between processed and bypassed audio is within about 1dB regardless of setting (a little tricky to specify, due to the comparison between the bypassed signal and the processed signal made up of 6 time delay taps!).

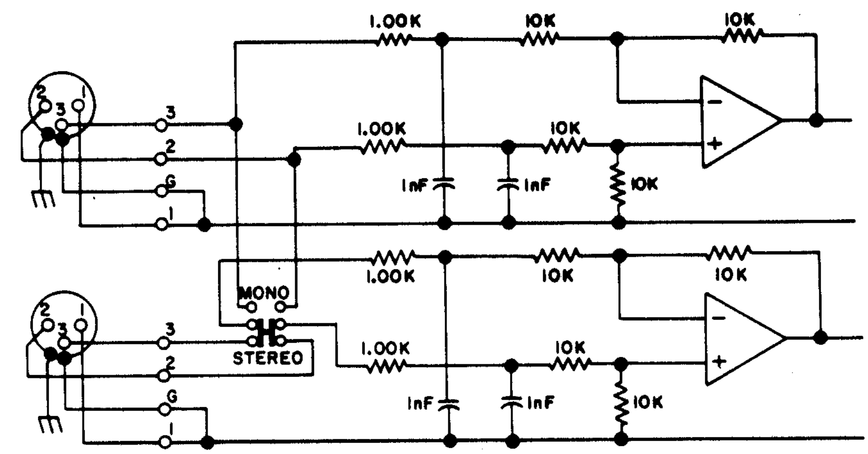

FIG 1: 126 Input Stage, Detailed Schematic

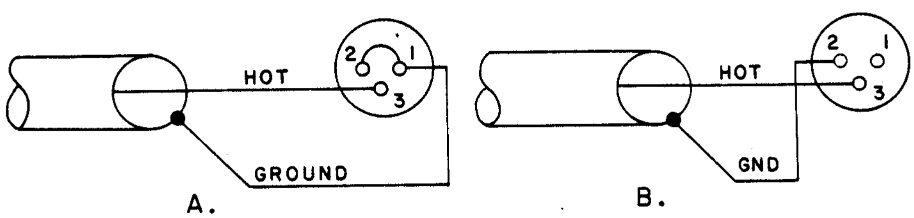

FIG 2: Connecting an Unbalanced Source to 126 Input

To connect the 126 to an unbalanced source, refer to Fig 2 above. You should use "A" if your external source and load do not share a common ground, or if you are unsure about this. If you're sure of the common ground (ie., you've connected the 126 to a console echo send/receive circuit), then "B" may offer some advantages in reducing potential ground loop problems.

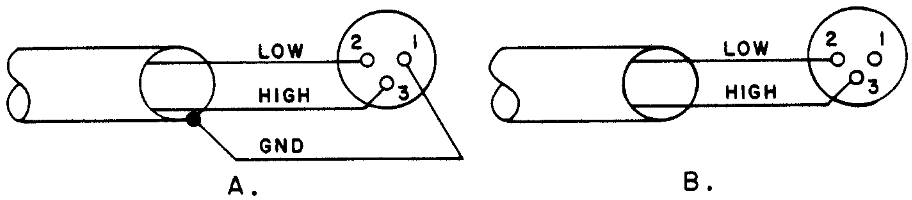

FIG 3: Connecting a Balanced Source to 126 Input

To connect the 126 to a balanced source, refer to the two diagrams above. "A" is the normal connection, but "B" may offer some advantages in dealing with ground loops.

OUTPUTS

Send the 126 outputs to balanced (preferable) or unbalanced inputs. Do not tie the left and right outputs directly together to form a mono sum signal. The 126 output stage is an active differential design, with 100 ohm source impedance of each output pin, and will drive inputs with 600 ohm or (preferable) higher impedance. A detailed schematic of the MSP-126 output stage is reproduced below for your reference.

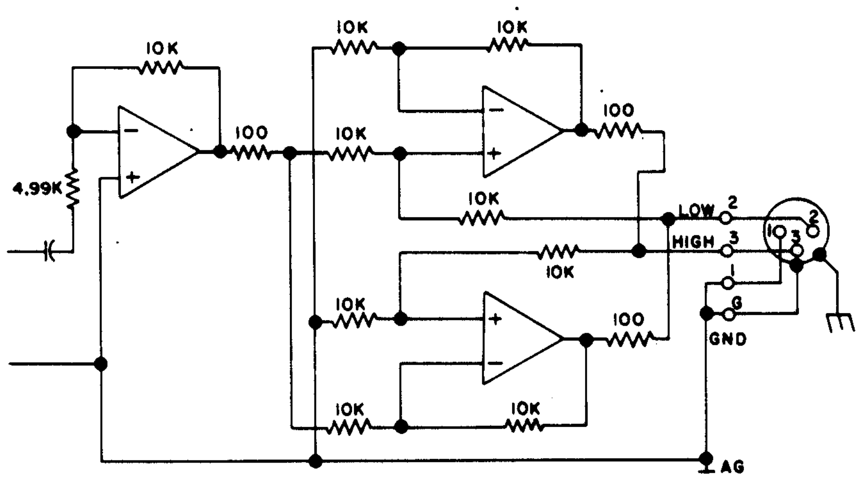

FIG 4: 126 Output Stage, Detailed Schematic

******************** IMPORTANT ***************************
IF THE 126 OUTPUT MUST FEED AN UNBALANCED INPUT, CONNECT HOT LEAD TO PIN 3, AND TIE PIN 2 TO PIN 1 INSIDE SHELL OF XLR PLUG GOING INTO 126 CONNECTOR. CONNECT GROUND LEAD TO PIN 1. FAILURE TO GROUND PIN 2 AT ALL, OR GROUNDING AT THE FAR END OF THE CABLE, MAY LEAD TO WEAK OR DISTORTED OUTPUT, OR TO OUTPUT STAGE OSCILLATION.
****

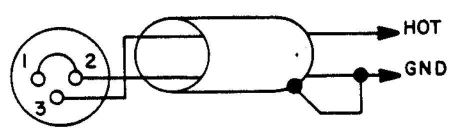

FIG 5: Connecting 126 Output to an Unbalanced Load

For connection to an unbalanced load, use two conductor shielded cable wired as shown above. Note that the connection from pin 2 to pin 1 must be made inside the shell of the XLR-3 plug going into the 126 output connector. See the note just above.

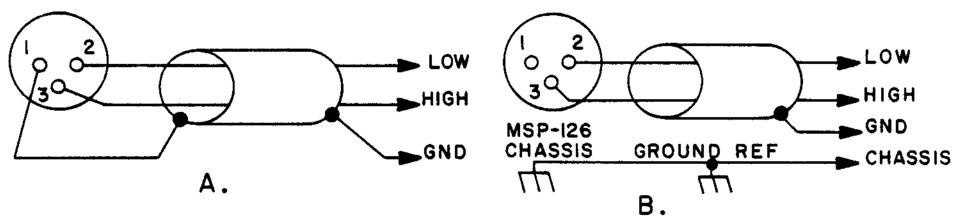

FIG 6: Connecting 126 Output to a Balanced Load

To connect to a balanced load, refer to the figure above. "A" is the straightforward connection, but "B" may offer advantages in dealing with possible system ground loops. In "B" both the 126 and the load chassis are grounded to a good system reference, and there is no direct connection via output cables. This prevents ground currents from flowing in the cable shields.

BYPASS JACK

This jack allows remote control of the Bypass function. The sleeve of the 1/4" jack connects to ground, while the tip goes to an LSTTL logic input pulled up by a 3.3K ohm resistor to +5VDC. To gain remote control of the Bypass function, connect a 1/4" mono (two circuit) plug to either a switch (single pole, single throw), as shown in "A", or to the output of a LSTTL gate, as shown in "B". The gate should be capable of sinking 1.5mA at 0.4V or less.

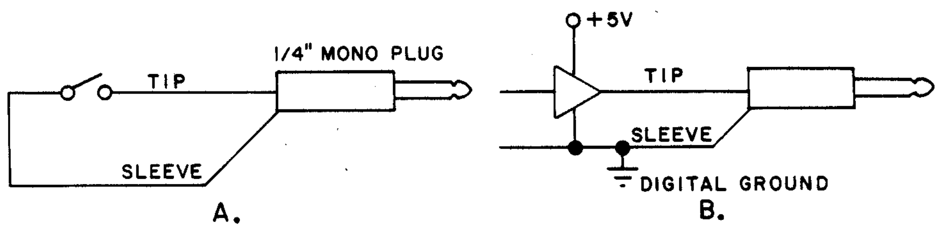

FIG 7: Remote Bypass Jack Schematic and Wiring

This jack and the front panel button are "OR'd"; that is, activating one or the other sets Bypass "on". If the front panel button is in, then the rear panel jack will seemingly have no effect, and vice versa.

CONTROL DESCRIPTIONS

INPUT LEVEL

This dual potentiometer is an overall level control that operates on the stereo inputs (or mono input) to compensate for differing source loudnesses. Adjust so that the green -6 LED flashes frequently, the yellow -3 LED flashes occasionally, and the red 0 LED flashes only rarely.

PEAK LEVEL DISPLAY

These 5 LED's show the peak signal level at the processor input. "Peak hold" captures even the shortest peaks. All circuits in the 126 operate with good headroom and slew rate reserve when this indicator never reaches red.

OUTPUT LEVEL

This dual potentiometer adjusts the level of the stereo processed or "bypassed" signal reaching the stereo output.

MODE SELECTOR

This rotary switch selects which of the eight built-in processing modes is in use. This switch, like the other two, rotates through a full 360 degrees with no end stop so that you can more quickly get to any desired setting. Because a Mode change alters most, if not all of the 12 tap delay times and gains, it is likely that some "glitching" will be heard if signal is present when the switch is turned. This is a normal consequence of sudden delay changes with attendant skips on the sound signal waveform. For silent changes, wait for a pause in the program material.

PARAMETER 1 SELECTOR

This rotary switch selects one of the 16 delay variations of the selected Mode. The display indicates the setting in use with a convenient mnemonic chosen to correspond to the psychoacoustic effect that Parameter 1 has in each mode. Although Parameter 1 affects only the tap time delays, this will generally be enough of a change to result in glitching if signal is present when the switch is turned.

PARAMETER 2 SELECTOR

The right hand rotary switch selects one of the 16 gain variations for the chosen Mode. In each Mode, the display shows mnemonics indicative of the effect this control has on the signal processing. The gain changes effected by this switch are the most benign with regard to glitching, but may still be audible if program material is present.

BYPASS SWITCH

In stereo Input Mode, Bypass "on" feeds the left input directly to the left output and the right input directly to the right output. In mono Input Mode, Bypass "on" feeds the left input through to both left and right outputs.

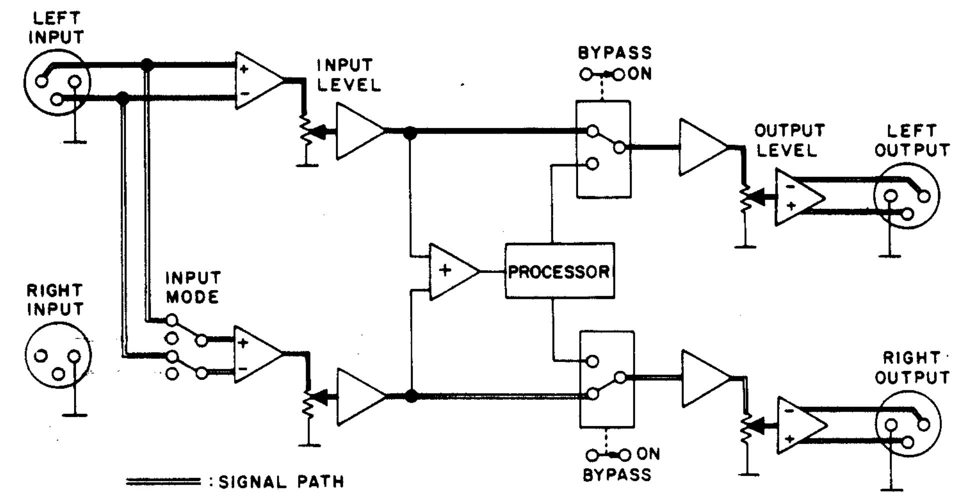

FIG 8: Signal Flow, Bypass "on", mono Input Mode

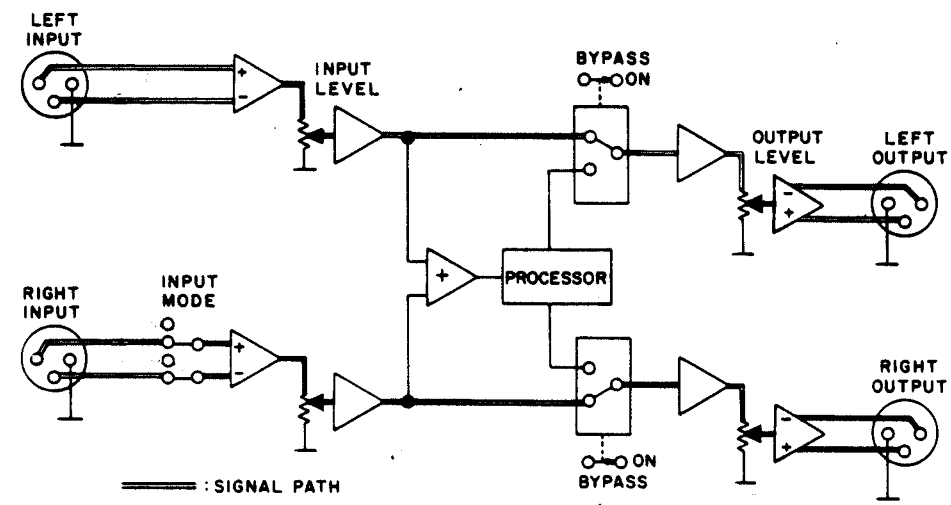

FIG 9: Signal Flow, Bypass "on", stereo Input Mode

POWER SWITCH

The power switch connects the mains power source to the 126's power transformer so that the unit will function.

When "off," this switch disconnects both sides of the power line from the transformer.

INSTALLATION IN BROADCASTING

It is possible to insert the MSP-126 in-line in either a full stereo or a mono/stereo situation. If, for example, a broadcast feed is normally stereo, but occasionally mono, and you want to selectively replace the mono feed coming down both lines with processed stereo, you would use the 126 in the stereo Input Mode. Then, with Bypass mode "on", the stereo source material would pass through the 126 as stereo, unaltered. On the other hand, when mono material comes along, Bypass can be switched "off", and processed stereo will drop in to replace the mono source.

In the mono Input Mode, the MSP-126 can be dropped into a mono signal path in order to split it into stereo. Simply connect the source feed into the left input and push the Input Mode button "in". Now when Bypass is "on" the mono input will be bypassed around the processor and go to both left and right outputs unaltered. On the other hand, when Bypass is switched "off", the output will be processed stereo.

INSTALLATION IN RECORDING STUDIOS

The 126 is best treated as an outboard effects processor, not as an "echo" device. Arrange the 126 input and output connections so you can send selected tracks or sub-mixes or final mixes into and through the 126 before they go onto tracks or into the final mix. It is unwise to connect the 126 in an echo send/return loop, as this increases the chance of mixing the direct signal with the 126 output within the console. Mixing the 126 outputs with the direct signal may, in some modes, have undesirable comb filter effects due to the short delays used. In those modes of the 126 where it is sensible to mix the processed sound with the direct sound, the Parameter 2 switch controls the mix ratio. It is very important not to mix any direct signal with the processed signal when using the MSP mode, as it will destroy the flat frequency response. Likewise, the CSP mode will suffer a reduction of the comb filter effect. In those modes of the 126 with a mixing function, the "direct" signal mixed in is actually a zero delay tap from digital memory, not a non-digitized analog signal, so it does have an inherent transit delay on the order of 50-70 microseconds.

THEORY OF OPERATION

The MSP-126 is a digital audio delay system. It converts sound into a digital format and writes it into memory. It reads signals from 12 different memory locations and creates a left output and a right output, each consisting of six delay signals. The time delay, phase (+/-), and amplitude of each tap are all set by firmware stored in EPROM's. Three front panel switches allow you to select from many pre-programmed possibilities stored in the EPROM's. The Mode switch has the highest priority and sets up the "menus" of choices for the other two switches. Parameter 1 accesses 16 variations of the time delay placements for the taps, while Parameter 2 accesses 16 variations of the amplitude/phase values.

Fig 10 shows the MSP-126 from a signal processing perspective. The left and right inputs are summed internally (but when the Input Mode is set to mono, only the left input is used) and fed into the RAM (random access memory). A 16K RAM and our sampling rate of 44.1kHz permit time delays up to 376ms. The 6 left taps are read out with time delay values T1L...T6L, and subjected to gains g1L...g6L, and the right taps are similarly read out. The two groups of taps are then summed and fed to the MSP-126 outputs.

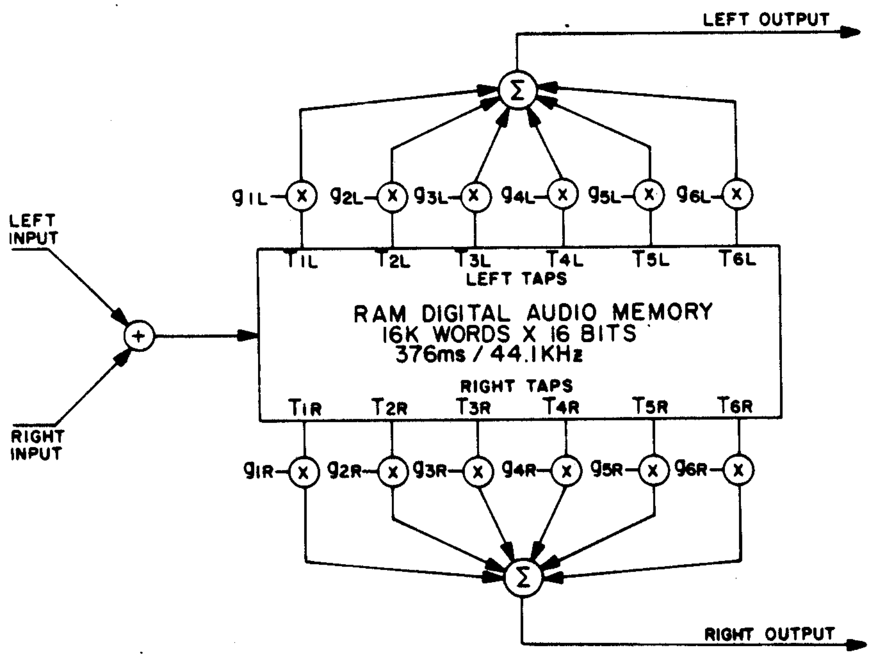

FIG 10: Signal Processing Block Diagram

Many new and useful audio processing functions are possible with this technique. If all the time delays are short, the effect will tend to be audible as frequency response changes. If the delays are long, the taps will be heard as echoes. The MSP-126 would be easier to explicate if this technique were more generally utilized commercially, and thus more familiar to sound engineers. Sound engineers have grown up with a variety of reverb devices, which produce a very high number of echoes quickly and sustain them for several seconds. They are also familiar with DDL's, with only one tap and a very limited number of echoes even when used with feedback. But the MSP-126 is in a world of its own, and it will require some reading of the material in this manual and several hours of experimentation before most sound engineers reach the point of saying, "Ah hah! I hear what you mean."

In our first Mode set, MSP-11, we have given 8 modes that range from CSP and its short delays to RPTS, with echoes up to 360ms. Each mode is a coherent processing function with appropriate range of adjustment over the time and amplitude parameters of the 12 taps. In some modes, one tap per channel is placed at zero delay and serves as a "direct" or "dry" signal that can be mixed with the other delay signals under control of Parameter 2. In other modes, Parameter 2 turns taps on and off selectively to alter the number of taps contributing to the effect. Although it is theoretically interesting to know that Parameters 1 and 2 control time delays and amplitudes, respectively, you will not always get that impression from listening. For example, in PAN mode, Parameter 1 moves the apparent image from left to right, while in ROOM mode, it alters the apparent size of a room. Perhaps it's better to just get used to each mode and its variations and not think too much about the theory.

IN CASE OF DIFFICULTY

USER SERVICING

There are no user serviceable parts in the 126 except for the two fuses, and they are extremely unlikely themselves to be the cause of a failure. If the unit is totally dead--no lights, doesn't get warm, and emits absolutely no sound (not even background noise)-- check the power cord at both ends for good contacts, and try another one if possible. Verify that the electrical outlet can operate another piece of equipment. If the power cord and outlet check out OK, remove the power cord from its connector at the rear of the 126. Remove the top cover of the 126 (10 screws) and locate the AC line fuse (it's at the front right edge of the PC board, near the power switch). If the fuse appears visually to be OK, it would still be a good idea to check it for continuity with an ohmmeter. If the fuse is open, replace it with a fuse of exactly the same type and value, replace the top cover, and try again. If your 126 is still inoperative, refer servicing to a qualified technician.

If the 126 seems warm, and there is residual circuit noise in the output, but the display characters are blank, check the fuse for the +5VDC supply. Remove the power cord from its connector at the rear of the 126, and then remove the top cover. Locate this fuse at the right rear of the PC board, to the left rear of the power transformer. Look closely for evidence that the fuse has opened. If it appears OK, use an ohmmeter to check for continuity, and replace if necessary, being sure to use exactly the same type. Reinstall the top cover, and try the 126 again.

********************* IMPORTANT *************************
IF YOU MUST REPLACE A FUSE, YOU MUST USE EXACTLY THE SAME TYPE AND VALUE IN ORDER TO NOT CREATE A SAFETY HAZARD. REPLACING A FUSE BY A SHORT CIRCUIT MAY PERMANENTLY DAMAGE THE UNIT OR CAUSE A FIRE HAZARD!
****

SERVICING

There are roughly 70 IC's in the 126, so servicing is not feasible for the typical owner. Please consult with your dealer or with URSA MAJOR if you have a service problem. We will do our best to quickly resolve the problem.

SHIPPING INSTRUCTIONS

Should your 126 require service, return it with a note, placed inside the carton on top of the unit, that tells us:

1. Exactly what's wrong--any symptoms you observed, how the unit is connected to other equipment, whether the problem is present always or only intermittently, whether the unit was OK originally and then developed the problem, or whether it was always defective. If possible, include a cassette recording of the the unit's sound when exhibiting the failure.

2. When and where you purchased the unit and its serial number.

3. The name and phone number of someone we can call if we have questions or difficulty duplicating your symptoms.

4. The full name and street address we should use when returning the unit to you.

Hopefully, you have saved the original carton and will re-use it when shipping to us. If not, you may obtain another carton from us for a nominal charge, or you may pack it yourself. If you pack it yourself in your own materials, be sure the 126 is completely surrounded by at least 3 inches of padding on all surfaces, and that the box is strong. We recommend that you insure your 126 shipment to us because it is otherwise completely uninsured against damage or loss. You must bear the cost of shipping to us even if the unit is in warranty still.

After we repair your unit, we will return it postage prepaid. On out-of-warranty repairs in the United States, and some other countries, the unit will be returned COD (covering shipping and service charges).

The shipping address for URSA MAJOR is, at the time of writing (December 1984):

URSA MAJOR, Inc.
28 Calvin Rd.
Watertown MA 02172
USA

It would be wise to check with us to be sure that the address above is still current if you are returning a unit in 1986 or later. Our mailing and telephone addresses are:

URSA MAJOR, Inc.
Box 28
Boston MA 02258
USA
telephone: 617 924 7697
telex: 921405 URSAMAJOR BELM

WARRANTY INFORMATION

LIMITED ONE YEAR WARRANTY

URSA MAJOR, Inc. warrants each Model 126 to be free from defects in materials and workmanship, under normal use and service, for one year. This warranty begins on the date of delivery to the purchaser or his authorized agent or carrier. If you sell your unit or give it as a gift, the warranty is automatically transferred to the new owner and remains in effect for the original one year period. During the warranty period, we will repair, or, at our option, replace at no charge, components that prove to be defective, provided that the equipment is returned, shipping prepaid, to the factory or a designated service facility.

This warranty is null and void under any of the following conditions:

a. Abuse, neglect, alteration, or repair by unauthorized personnel.

b. Damage caused by improper use, or operation from an incorrect power source.

c. Damage caused by accident, act of God, war or insurrection.

No other express warranty is given. The repair or replacement of a product is your exclusive remedy. ANY OTHER IMPLIED WARRANTY OF MERCHANTABILITY OR FITNESS IS LIMITED TO THE ONE-YEAR DURATION OF THIS WRITTEN WARRANTY. Some states, provinces, or countries do not allow limitations on how long an implied warranty lasts, so the above limitation may not apply to you. IN NO EVENT SHALL URSA MAJOR, INC., BE LIABLE FOR CONSEQUENTIAL DAMAGES. Some states, provinces, or countries do not allow the exclusion or limitation of incidental or consequential damages, so the above limitation or exclusion may not apply to you.

This warranty gives you specific legal rights, and you may also have other rights which vary from state to state, province to province, or country to country.

Products are sold on the basis of specifications applicable at the time of manufacture. URSA MAJOR, Inc., reserves the right to make changes or improvements in the design of the machine without obligation to make such changes or improvements in purchaser's machine.

Any out-of-warranty repairs are warranted against defects in materials and workmanship for a period of 90 days from date of service.

MODE SET MSP-11-- DESCRIPTION

The first "Mode" set, or program set, for the MSP-126 consists of the following:

MSP11 Multi-Tap Stereo Processing creates stereo
from mono with flat response
CSP Comb Filter Stereo Processing creates stereo
from mono with complementary comb filters
ROOM ROOM Early Reflections provides early
reflection pattern of rooms
DLAY DeLAY Cluster provides a cluster
of delays after an initial delay
PAN PAN "Pot" using Delays adjusts L-R image
position and width using delays
DDL Digital Delay Line provides a simple
adjustable delay for L, R, or L & R
RPTS REPEATS provides
adjustable, stereo delayed repetitions
SCALE Filter in musical SCALE steps provides stereo
comb filter with musical intervals between L and
R channels

MSP11 Multi-Tap Stereo Processing creates stereo
from mono with flat response
Parameter 1: "Depth" D5 (minimum depth) to D100 (maximum)
Parameter 2: "Width" W0 (minimum width) to W100 (maximum)

This mode uses special, adjustable time delay patterns to process mono into stereo with flat response of left, right, and left+right output signals. At minimum Width, the output is mono with a small time delay. At Depths above about D60, discrete time delays will begin to be audible with some kinds of program. Stay below D60 for general purpose broadcast processing.

CSP Comb Filter Stereo Processing creates stereo from mono with complementary comb filters
Parameter 1: "Frequency" 200Hz to 14kHz
Parameter 2: "Width" W0 (minimum width) to W100 (maximum)

This mode uses a pair of complementary left and right comb filters to process mono into stereo with high mono compatibility (their sum is flat response), but very colored response of left and right output signals. Parameter 1 adjusts the frequency of the first (lowest) comb peak (left channel) and null (right channel). Parameter 2 sets the depth of the comb nulls, and thus the intensity of the comb filter effect. At minimum Width, the output is mono with zero delay and flat response.

ROOM ROOM Early Reflections provides early reflection pattern of rooms
Parameter 1: Longest reflection time in ms, from 5 to 360ms
Parameter 2: % reflection in dry/wet output mix, from, 0% (all dry) to 100% (all reflections)

This mode adds the early reflections of rooms or concert hall to the dry signal. The apparent size of the space is adjusted with Parameter 1, while the apparent distance from the source is adjusted with Parameter 2. Coloration is quite low in this Mode, especially above 50ms.

DLAY DeLAY Cluster provides a cluster of delays after an initial delay
Parameter 1: Time to beginning of delay cluster, in ms, from 20 to 320ms
Parameter 2: % delay cluster in dry/wet output mix, from 0% (all dry) to 100% (all.wet)

This mode adds a short duration cluster of delays to the dry signal. The delay time before the cluster begins is set by Parameter 1, while the loudness of the cluster relative to the dry sound is set by Parameter 2.

PAN   PAN "Pot" using Delays   adjusts L-R image position and width using delays

Parameter 1: Apparent source azimuth in degrees, from 90<-- (full left) through --!-- (center) to -->90 (full right)

Parameter 2: Apparent image width, from W0 (minimum) to W100 (maximum)

This mode uses short clusters of time delays of adjustable duration (Parameter 2) to create the illusion of an apparent source at azimuth angles from 90 degrees left to 90 degrees right. The image position is established with time delay patterns: each channel has equal energy at all settings; furthermore, each channel has flat response at all settings. Finally, the mono sum (left+right) is also flat response at the higher width settings (about W40 and above).

DDL   Digital Delay Line   provides a simple adjustable delay for L, R, or L & R

Parameter 1: Delay multiplier, from 1 to 16

Parameter 2: Delay scale factor, from 1ms to 20 ms Left and Right, and from 0.1ms to 20ms L+R

This allows the 126 to be used as a simple DDL. The delay value indicated is given to the named output channel(s), while the unnamed channel (if any) is given a zero delay signal.

RPTS   REPEATS   provides adjustable, stereo delayed repetitions

Parameter 1: Sets the delay time of the last repeat, in ms, from 10 to 360ms.

Parameter 2: Selects the number of repeats from 2 to 10 repeats, and the gain profile, including equal, increasing, and decreasing loudness.

Intended for special effects, this mode gives roughly evenly spaced repetitions of the input sound, skipping from left to right.

SCALE Filter in musical SCALE steps provides stereo
comb filter with musical intervals L-R

Parameter 1: Sets the musical interval between the fixed filter (A=440Hz) in the left channel and the variable filter in the right channel (from A, "UNIson" up through a chromatic scale to the -3RD above A=880)

Parameter 2: Sets the intensity of the filters, from off (0) to on with peaks at all harmonics (+10) or to on with peaks at only odd harmonics (-10)

Intended for special effects, this mode uses evenly spaced delay values to create comb filters tuned at musical intervals. In the left channel, A=440 and its harmonics will be emphasized, while in the right channel the apparent pitch will rise relative to the left channel, going up 15 half-tones of a chromatic scale.

APPLICATIONS

BROADCASTING

The MSP-126 is superlative for broadcast processing because it gives broadcasters full control over an innovative form of processing based on multiple, short delays. Thirsting for good processing, broadcasters often use reverb units to add a dash of "time extension" to a program material to help fill in gaps and add warmth to sound. The problem is that most reverbs have a fixed decay time of 2-4 seconds. In order for the reverb to contribute any useful early support to the voice, it must be mixed in at a level so high that the long reverb decay tail is also audible. Now the sound of a newscaster in the reverberant space of a concert hall or church, while nice, isn't really appropriate. By using the MSP-126, with its shorter, more subtle delays, sound is improved without the distraction of a large pseudo-space reverberating around the listener.

ON-AIR PROCESSING-- MONO TO STEREO

The MSP-126 can be connected in-line, as discussed elsewhere in this manual, and will pass stereo programs unchanged when in the Bypass mode. When mono material, such as announcers, commercials on cart, older records (45's from the 50's), etc. come along you can switch Bypass off and substitute the 126's processed stereo for the mono. This can be remotely controlled with a switch, relay, or computer. We recommend the mode MSP11 for this kind of processing. In mode MSP11, the processed stereo is highly compatible: left channel, right channel, and the left + right sum are all uncolored, flat response signals. The left and right signals are of equal loudness, while the sound is at all times natural and balanced. The stereo sense imparted by the 126's processing is not dramatic or unnatural, is pleasing on speakers or headphones, and works with any kind of program material.

ON-AIR PROCESSING-- VOICE ENHANCEMENT

You can not only obtain a stereo version of the announcer's voice, you can also enhance the voice for extra body, punch, and carrying power. The MSP -126 does this by adding subtle, short, time-delays to the original signal. These, while so short as to be undetectible as "processing", nevertheless enhance the voice, and actually contribute to a perceived rise in loudness greater than any loudness increase that may show on a peak modulation meter. Again, mode MSP11 is first choice for this kind of processing. But you can experiment with some of the other modes as well, such as DLAY, ROOM, or even DDL and RPTS. Each of these modes can be adjusted to take the processing far into the realm of special effects to obtain dramatic repetitions, echoes, and acoustic simulations.

ON-AIR PROCESSING-- SPECIAL EFFECTS

You can place the MSP-126 in-line and have it set normally to Bypass on, passing program material unaltered. Now you can set the controls to give you whatever subtle or bizarre effect you like, while the MSP-126 sits there innocently while you wait for the right moment. Then you release the Bypass button, and voila, a dramatic alteration and highlighting of program material!

PRODUCTION PROCESSING

The processing discussed so far is all applicable in the production environment. For other production processing ideas, see the discussions that follow for recording and performance situations.

RECORDING STUDIOS

Although I expect that creative audio engineers will discover many uses for the MSP-126 that I haven't dreamed of, here are some ideas that may help you get started.

LAYING DOWN TRACKS

If you're recording an instrument or voice and want it to be stereo in the final mix, don't set up two mikes and waste an extra track to record it in stereo. Mike it carefully with one mike and later on use the MSP-126 to process it into stereo. Not only will you save some tracks, but the MSP11 mode stereo processing will be more mono-compatible than your stereo miking would have been (due to comb filter effects when the two mike signals combine in mono).

MONO TO STEREO

This very basic application of the MSP-126 will go a long way toward enhancing the many close-miked mono sound sources that need to be mixed down to stereo. You will achieve not only stereo, but also, in some modes, an enhanced body and richness. Although the 126 has some modes specifically designed for stereo processing, the entire design philosophy of the MSP-126 is inherently stereo. Unlike conventional DDL's or reverbs, the 126 gives you a lot of control over a lot of delay taps already arranged to create two channels of effective processing. All the processing modes are designed to utilize the stereo outputs to some degree.

For stereo processing that changes the sound the least, try mode MSP11. See the broadcasting application discussion above. Mode CSP, while doing dramatic things spectrally (akin to a flanger that's not sweeping), is very simple from a time delay standpoint as it only adds one very short time delay to the dry signal. PAN and DLAY also produce mono from stereo, and their coloration and mono-compatibility is good at most settings. ROOM yields an early reflection pattern in stereo with adjustable size, low coloration, and good mono compatibility.

DELAY EFFECTS-- SLAP AND BEYOND

Some modes can produce long delays and obvious time effects. DLAY, DDL, RPTS, and even ROOM can be adjusted for obvious echo effects. In DDL, you can get any of about 96 delay times and use it on the left signal, while the right is at zero delay (or vice versa). This gives delays up to 360ms, with the simplicity of a DDL (only one tap).

With RPTS, you select a maximum delay, and then the number of repeats you want. This gives a stereo richocheting pattern of echoes, the first at zero delay, the last at the selected maximum, and others filling in at even spacing. You can control not only the number of repeats, but their amplitude profile (whether they become louder or softer or stay the same amplitude). Try RPTS with percussion tracks or a drum machine. Or try it with vocals and the spoken word. Selecting a profile with increasing loudness will give the impression of a tape played backwards, destroying speech intelligibility.

The DLAY mode gives a cluster of delays after an adjustable pre-delay. The cluster itself adds a little body, in stereo, to the delayed sound, while the adjustable pre-delay can cause the cluster to come as a distinct echo: very nice with vocals. In DLAY you can also adjust the dry/wet mixing ratio with Parameter 2.

ROOM uses 10 delay taps starting near zero delay and extending to the displayed delay time as a maximum. As you increase the delay time, the individual taps will become audible as echoes. Taken to maximum delay, ROOM doesn't provide a space where you'd like to play your drum kit, but it can be very nice for legato passages of strings or synthesizer. As in DLAY, Parameter 2 gives you a dry/wet mixing control.

IMAGE LOCALIZATION

Since Haas and other psychoacoustic researchers began to unravel the mysteries of directional hearing, there has been controversy over how we determine apparent source location. We know that during stereo mix down we can "place" a source over to the left by giving the left speaker a louder signal than the right. We know that when we "pan a signal to the center" with our pan pot, we give both speakers the same loudness, and that a listener centered between the two speakers will localize the sound in the center. Unfortunately, we also know that as soon as the listener moves as little as a foot or so nearer to one speaker, he will now localize the sound at the nearer speaker, and may not even hear the other speaker.

Haas researched this phenomenon and determined the role of arrival time differences in cases where the same sound comes from two locations at once. He found that the earlier arriving sound source determines our localization of a sound, even when it is up to 20dB weaker than another, competing source. This has led to interest in using time delay as a new method of "panning." A DDL can do this easily, and it works-- almost. Putting the same sound in each speaker, but delaying one speaker's signal by at least 1ms, establishes an apparent location at the earlier speaker. And this localization holds pretty well, even if we depart from a central listening position and approach the delayed speaker. Of course at some point during our approach, as the relative delay difference decreases and the delayed speaker gets louder, we shift localization to the delayed speaker. In between, there is a region of confusion, where the source seems broader and hard to localize.

This "delay pan pot" works, but there is at least one potential problem. The left and right channels have the same sound, but with a time delay difference. All's well until a situation when the two signals get summed into mono (your clock radio, car radio, etc.): then the result is a comb filtered version of the original sound, when all we wanted was panning. Is there a solution? As you might expect, there is, and it's the PAN mode of the MSP-126.

The PAN mode uses two multi-tap time delay patterns to establish source azimuth (left-right localization between the two speakers) and image width. Parameter 1 controls the azimuth, from 90<-- to -->90, while Parameter 1 sets the image width from minimum (W0) to maximum (W100). Notice that I said the MSP-126 uses "time delay patterns": it takes more than one delay to accomplish what we have here in our PAN mode. At position W0, the pattern is reduced to one tap per channel, and Parameter 1 moves the image by shifting the relative time delays of left and right signals. This is just like the basic "delay pan pot" application discussed above, and has the same mono compatibility failure. But as you adjust Parameter 2 to higher width values, more taps come into play, the image width increases, and the localization continues to hold. And, lo and behold, as the width increases, so does the mono compatibility of the stereo output.

The flip side of localization is localization confusion. And we have a mode for this, too: MSP11. MSP11, the cornerstone of our mono to stereo processing, turns mono into stereo, but yields an image hard to localize (of course! would anyone want stereo where everything seemed to be to the left or right?). Use this mode when you want a sound in both channels, but not in the center. This confused localization is the product of some inspired juggling of time and amplitude cues when we conceptualized mode MSP11.

REVERB APPLICATIONS

Your venerable EMT plate is great, but a warm concert hall it is not. There is very little pre-delay ("fast diffusion") and no reflections in the first one or two hundred milliseconds. Similar shortcomings exist in all plate and spring reverbs and even, truth to tell, in some of today's much-vaunted digital reverbs. The MSP-126 can help.

Delaying the drive to reverb units is a well known technique used to help get a sound closer to concert halls. The 126 can provide pre delay in its DDL or DLAY modes, but that's no great trick. Where the 126 really shines is in giving multi-tap reflection patterns. Patch the reverb send into both the reverb and the MSP-126 and set the 126 to ROOM. Adjust the wet/dry ratio between the direct signal and reflections on the 126, and the reverb level on the console. You now have full control over direct, early reflection, and reverb sounds.

Another use for a 126 is as an echo density multiplier in front of a reverb. Patch the echo send through the MSP-126 on its way into the reverb, select ROOM or DLAY modes, and experience your reverb unit with a higher echo density, and with adjustable pre-delay.

SPECIAL EFFECTS-- SPECTRAL MODIFICATION

An equalizer or flanger the MSP-126 is not; most of its modes are intended to be uncolored. After all, it's easy to produce dramatic comb filter coloration with a 1 tap DDL/flanger, but impossible to get uncolored stereo imaging. Nevertheless, we put a few interesting modes in the 126 for fans of spectral modification and special effects. Try these modes with sounds that have a broad spectrum, such as noise-like or percussive instruments. Comb filters are a drag if your source is close to a pure tone. (How can you hear a peak in a filter if the sound passing through the filter is a tone nowhere near the peak?).

CSP, as observed before, is like a flanger that's standing still, except that there are two outputs from the 126, each with its own comb filter pattern. Regardless of the Parameter 1 setting, the left and right comb filters are exact complements of each other: wherever the left has a peak, the right has a null, and vice versa. Furthermore, the comb filter response curves are so well matched that when the two outputs are summed, the response becomes flat. The display shows you the frequency of the first peak in the left channel (null in the right channel). Other peaks and nulls occur at integer multiples of the displayed frequency. Parameter 2 sets the null depth, from no null to greater than 60dB. You may be tempted to move the delay setting to get a sweeping filter sound (flanging): be our guest, but know that as you go toward lower and lower frequencies the chance of audible glitching increases.

Mode SCALE is a special beast peculiar to the MSP-126. In SCALE, I simulated two feedback delay units, one left, the other right. The delay unit for the left channel has a fixed delay regardless of Parameter 1 setting such that it produces a resonant peaked comb filter with peaks at 440Hz and multiples of 440Hz. The right channel, however, uses a delay time adjustable by Parameter 1, a delay time that decreases as the control is turned from "UNI" to "^ -3RD". The right channel delay times are chosen to create a filter related by the displayed musical interval to the fixed left channel filter. Sound entering the 126 emerges tuned by these two filters so that a 2 note interval is created between the left and right channels. It's a very dramatic and useful effect with broadband source material, or with speech for special effects.

SPECIAL EFFECTS-- EXTERNAL FEEDBACK

We haven't tried this, but if you are inclined to experiment, you can probably get some interesting effects from it. Patch the 126 into your console in a way that allows you to externally couple one of its outputs back to one of its inputs. If you've got a spare DDL, patch it into the feedback path so you can delay the 126 output before it gets mixed back into its input. An equalizer or compressor/limiter in the feedback path would add to the possibilities.

If you don't have a DDL in the loop, there are some modes and settings that will probably be useless for feedback:

MSP11, CSP, PAN, SCALE at any settings
ROOM: stay away from the smaller rooms, and set Parameter 2 to 100%
DLAY: set Parameter 2 to 100%
DDL, RPTS: don't feedback very short delays

If you do have a DDL in the loop, all modes are fair game for your exploration.

In most modes and settings, the 126 output signal you're feeding back is a multi-tap signal. With a multi-tap signal, it is harder to know how much feedback you can use before instability occurs and the output rises to clipping level ("howl round", as the British call it). Nevertheless, some of the multi-tap patterns in RPTS, ROOM, and DLAY will probably give some very nice special effects. Good luck!

MUSICAL PERFORMANCE AND COMPOSITION

More and more musicians today are working in home studios, composing at the keyboards of sophisticated computer-based synthesizers. In many cases, an entire video or film sound track, or record album is produced and mixed in a home studio. Synthesizers and drum machines are basically mono, devoid of room ambience. "Stereo" in these situations often means pan pot left-right placement of different sources, and nothing more. Once again, the MSP-126 comes to the rescue!

Process the synthesizer or drum machine through the 126 and hear how it opens up in stereo width and depth. Now add some stereo reverberation and you'll have a warm, natural, and pleasing overall sound. Most of the MSP-126 modes are useful in performance or home production situations. With synthesizers, I think that MSP11, ROOM, and DLAY are especially useful. With drum kits, MSP11, DLAY, RPTS, and SCALE are good choices.

PA (STAGE SOUND)

Frankly, we haven't had an opportunity to try this one yet, and it is a little far out. If you are doing PA work in stereo, you have a dilemma. Pan pot image localization will work OK for the part of the audience seated exactly symmetrically between the speakers, but listeners off to either side will tend to only hear the nearer speaker and will miss whatever is panned to the opposite speaker. Our proposal is that you try two MSP-126's and use them as follows.

Develop the stereo mix you want by panning everything hard left or hard right. Feed the left console output into one MSP-126 and the right console output into the other MSP-126. Set the left 126 for PAN mode 90<-- and the right 126 for PAN mode -->90. You will want to experiment with the Parameter 2, Width, settings, but start at W100. Now combine (with an active mixer) the two 126 left outputs into a left signal for the left power amplifier, and the two right outputs into a right signal for the right power amplifier. What you should have is pretty good localization even off to the sides and very good localization for the center audience. And everyone, wherever seated, will hear all the sound sources, whether you assigned them to the left or to the right. Take a listen and let me know how it works.

FILM/VIDEO

One special film application of the MSP-126 has already been discussed under the section "Image Localization," above. The ability to place an actor's voice or sound effects to one side or the other while still maintaining equal energy in both speakers should prove very useful in motion picture and video sound.

All the other stereo processing and special effects modes will also prove useful under some circumstances. Robot and sci-fi voices can be created from mere mortals in modes SCALE and RPTS.

INDEX