ADR 68K DIGITAL REVERB + EFFECTS

OWNER'S MANUAL

Revision 4.0

By Paul D. Lehrman and Christopher Moore

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Corporate Statement

AKG Acoustics' Digital Products Division is dedicated to producing innovative, useful, and lasting products of superior value for professional audio users.

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Sincerely,
Christopher Moore

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Introduction to Revision 4.0

This manual is divided into two parts. The first part, comprising Chapters 1 through 3, covers those aspects of the ADR 68K which are unchangeable: i.e., the hardware, installation, the basic design concepts, service and upgrades, and general information about digital signal processing

The second part, comprising Chapters 4 through 7 and the Appendices, deals with the software and programs in the ADR 68K. This information will change every time a software upgrade is released, and so you will be supplied with new versions of these chapters when you receive your upgrades. You will also be supplied with a new Table of Contents, so your manual will never go out of date.

— Chapter 1 — Installation and Description

— Precautions —

⇒ Before connecting the ADR 68K to your AC power line, read the sections explaining the correct internal voltage selection. Connecting the unit to the wrong voltage may damage it.

⇒ Before installing your unit, read the section on mounting and be sure to allow enough room for ventilation. Also, pay attention to the warning about providing extra support for the rear of the chassis if the unit will be transported in a rack.

⇒ This unit is for use with line level audio sources, on the order of a few volts maximum. It is not to be connected to the output of audio power amplifiers. Doing so may damage the input circuits.

⇒ *Always* turn off any monitors, both speakers and headphones, that are connected to the ADR 68K when turning the unit on or off. When the ADR 68K is powered up or down, audio "pops" will appear at its outputs which can be quite loud, and may cause damage to speakers or headphones.

⇒ The 1/4" jacks on the remote control must not be subjected to greater than +5 volts or less than 0 volts (i.e., a negative voltage), or the remote may be rendered unusable.

⇒ Although the unit is quite tolerant of warm environments, you should nevertheless avoid excessive moisture and high temperatures to reduce any fire hazard.

"WARNING: This equipment generates, uses, and can radiate radio frequency energy and if not installed and used in accordance with the instructions manual, may cause interference to radio communications. It has been tested and found to comply with the limits for a Class A computing device pursuant to Subpart J of Part 15 of FCC Rules, which are designed to provide reasonable protection against such interference when operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference in which case the user at his own expense will be required to take whatever measures may be required to correct the interference."

Unpacking and Inspection

As soon as you receive the carton containing your ADR 68K, 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.

NOTE: 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 ADR 68K. *<u>This is IMPORTANT!</u> In order to keep you informed of software updates and other changes or enhancements to your ADR 68K in the years ahead, we must have your warranty card on file. If we do not, we will have no way of getting in touch with you!*

Wiring and Setup

Power

The ADR 68K operates on 115 or 230VAC, 50 or 60Hz. A label on the rear panel indicates how your unit was set at the factory.

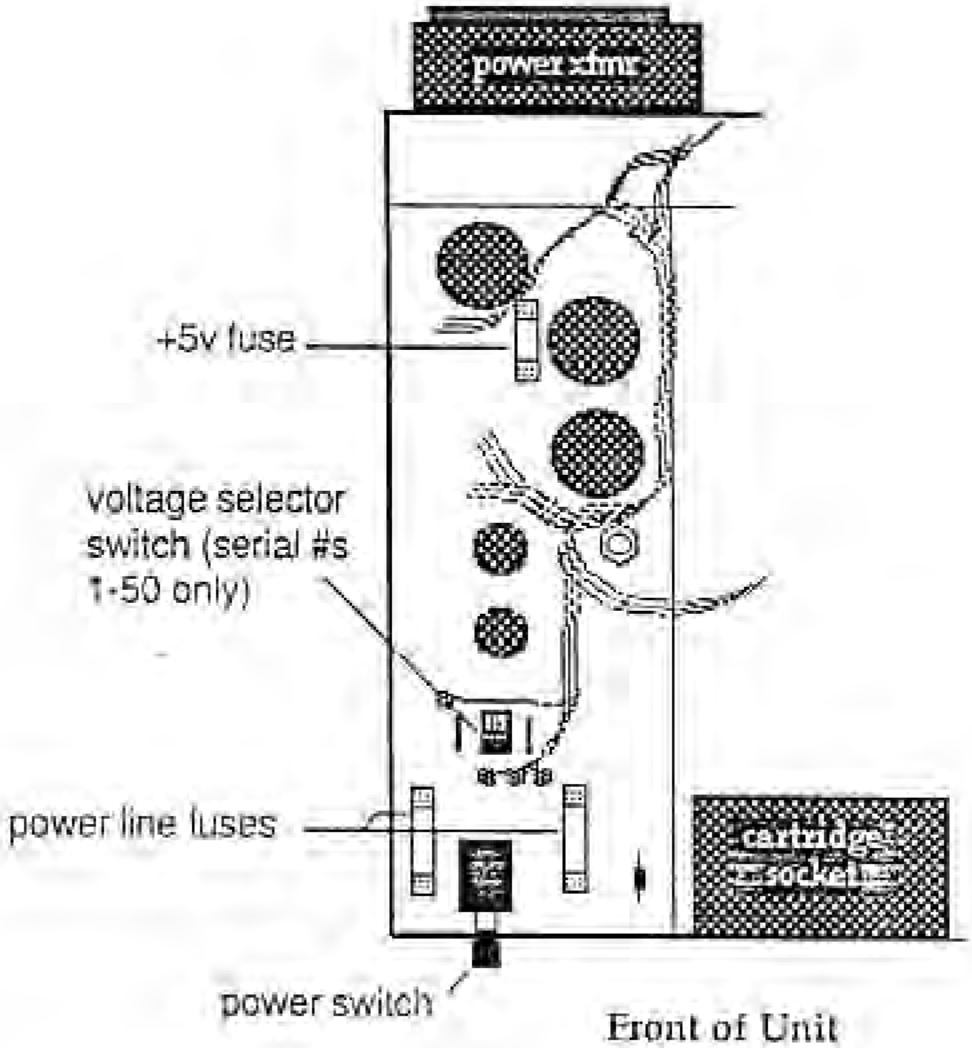
<u>IMPORTANT!</u> If you have any doubts about the internal voltage setting, and you are about to plug the unit into a 230VAC line, DON'T!! Operating an ADR 68K set to 115V at 230V will damage the unit. Remove the AC cord, unscrew the 10 screws holding the bottom cover in place, and ckeck the setting of the internal voltage selector switch. Correct if necessary and reinstall the bottom cover.

<u>IMPORTANT!</u> 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 ADR 68K chassis to power ground. DO NOT DEFEAT IT!

The AC receptacle is a standard 3-pin IEC connector for a detachable line cord. The center pin is tied to the ADR 68K chassis for electrical safety reasons. The first 50 ADR 68K's are equipped with a transformer and switching arrangement suitable for most countries, except Japan, where the nominal line voltage is 100V. (If you have one of these units and must use it in Japan, please contact your dealer.) These units have a two-position slide switch mounted internally on the PWR PC board that straps the transformer primaries for either 115V operation (windings in parallel) or for 230V operation (windings in series).

Later units have a rotary four- or six-position switch and a tapped transformer, allowing selection of nominal line voltages of 100V, 120V, 220V, and 240V. Set the switch to the position nearest the nominal line voltage of your country. When in doubt, chose the setting just higher than your nominal line voltage, or consult your dealer for advice.

Also inside are two fuses for the mains (US type 3AG SLO BLO 1.5A, 1-1/4 x 1/4"), and a fuse for the +5VDC supply (US type 8AG 8A fast blow, 1 x 1/4"). You can access these by removing the bottom cover — remove four small screws from each side of the cabinet, and two from the back. Always replace the bottom cover when you are done to avoid shock hazard.



Mounting

The ADR 68K may be rack mounted (height is "2U," or 3.5" [8.8cm]). Be sure to maintain adequate clearance for air flow at the top, bottom, and sides of the unit. One inch (2.5cm) all around is recommended. Do not install the ADR 68K directly above a high-power component such as a power amplifier. The ADR 68K should not be operated on a table or bench-top, as there are ventilation holes in its underside. Bench-top operation also interferes with convection air-flow cooling of the heat sink and power transformer.

For brief uses, on the order of an hour or so (such as demonstrations), there is no problem with bench operation. For periods on a bench of up to three hours or so, it is best to place the ADR 68K on blocks 1" (2-3cm) high, to allow convection cooling.

You will notice that much of the weight of the ADR 68K is in the power transformer on the rear panel. Therefore, if you install the unit in a rack that will be transported and subjected to physical jolting, you must arrange for extra support of the chassis at the rear. If that is especially inconvenient, you may obtain an accessory pair of reinforced mounting rails from AKG that screw into the rack from the front and give the chassis extra support.

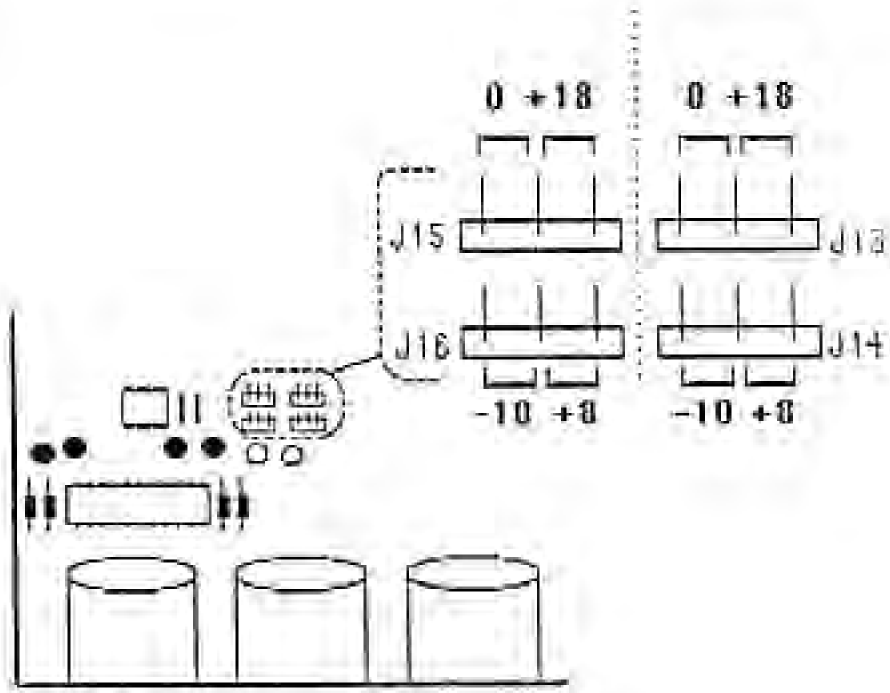
Audio Connections

Inputs

The ADR 68K has stereo inputs. Whether to use one or both of the inputs will depend on the program chosen, but in general, in order to utilize fully the unit's sound processing capabilities, both of these inputs should be connected. Feed the ADR 68K inputs from either balanced (preferable) or unbalanced sources of 600 Ω or less impedance. Maximum signal level is 7Vrms, while minimum signal for normal operation is -10dBV (316mV).

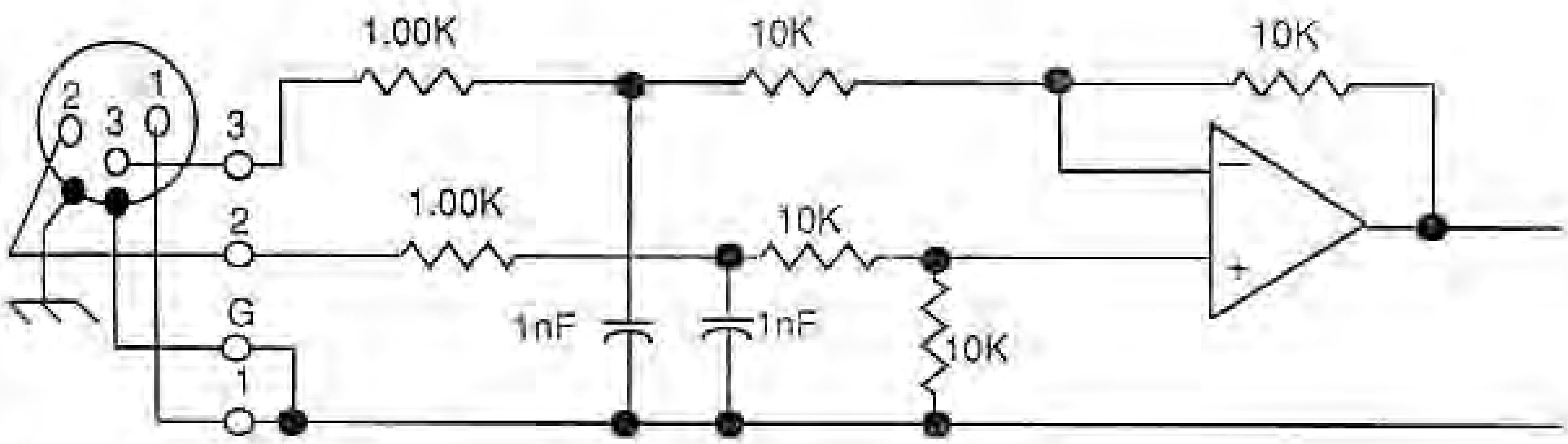
The ADR 68K can be set for four different input sensitivities: -10, 0, +8, and +18dBV. Input sensitivity is defined as the rms sine wave level of a signal that exactly corresponds to the saturation point of the analog to digital converter; i.e., when the sensitivity is set to "0dBV", a sine wave at the input of 1V rms will have its peaks converted to +/- full scale of the 16 bit ADC.

The ADR 68K is shipped from the factory set to +8dBV input sensitivity. To adjust the sensitivity, remove the AC line cord from the ADR 68K, and remove the bottom cover by removing four screws on each side of the cabinet and two on the back. Now locate the four jumper blocks, labelled J13 through J16, near the input connectors on the ANA-3 board. There is a pair of jumper blocks for each of the two input channels. For each pair of blocks, there are four valid positions for the shunt. Refer to the drawing (or to the designations screened on the PC board itself) to determine the shunt position for the input sensitivity you would like to use. Be careful not to turn the shunts upside down before putting them back on the pins: the end of the shunt with the holes is the one that gets pushed onto the pins. Once you have moved the shunts, reinstall the bottom cover.



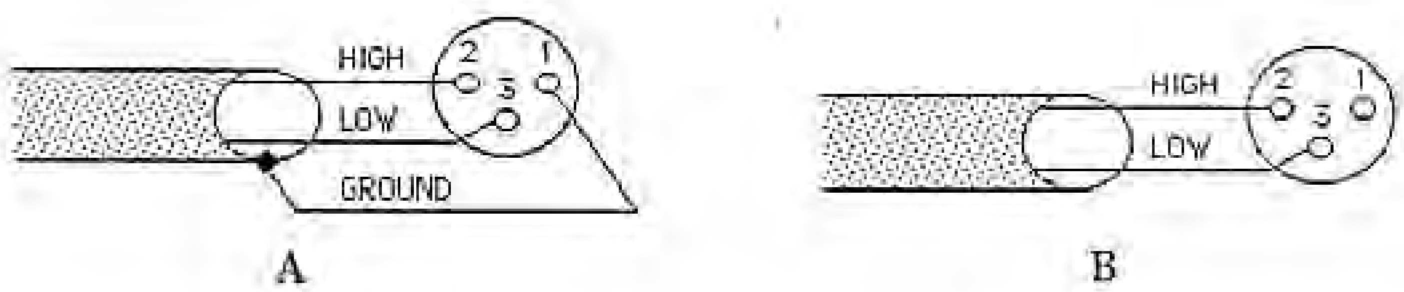
Adjusting the input sensitivity

The inputs are electronically balanced (differential amplifier): Pin 2 high, Pin 3 low, and Pin 1 ground. Recommended source impedance is 600 Ω or less. This is what the input stage looks like schematically:



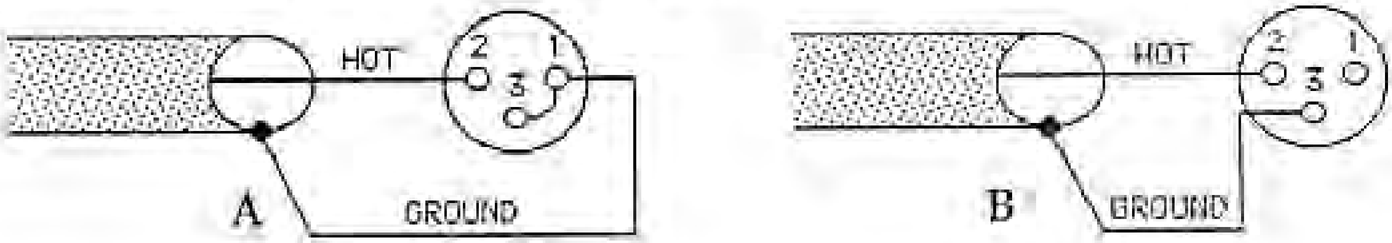
The ADR 68K input and output circuits are are quite versatile; for example, they're compatible with balanced or unbalanced equipment and they will also work perfectly well in either "pin 2 high" or "pin 3 high" systems. If your installation wiring uses pin 3 as "high" or "hot", simply interchange the connections to pins 3 and 2 in any of the connection figures given here. Just make sure you follow the same convention at both input and output of the ADR 68K, or you'll get an overall phase reversal.

There are two ways to connect the ADR 68K to a balanced source. You should use diagram "A" if your external source and load do not share a common ground, or if you are unsure about whether they do. If you're sure there is a common ground between source and load (e.g., you've connected the ADR 68K to a console echo send/receive circuit), then diagram "B" may offer some advantages in reducing potential ground loop problems.



Connecting the Inputs to a Balanced Source

There are also two ways to connect the ADR 68K to an unbalanced source. Diagram "A" is the normal connection, but diagram "B" may offer some advantages in dealing with ground loops. When you are connecting the ADR 68K to unbalanced equipment, be sure always to ground the unused pin of the ADR 68K input or output right inside the connector shell at the ADR 68K, not at the other, unbalanced end of the cable. Failure to do this can result in signal loss or hum pickup.



Connecting the Inputs to an Unbalanced Source

Outputs

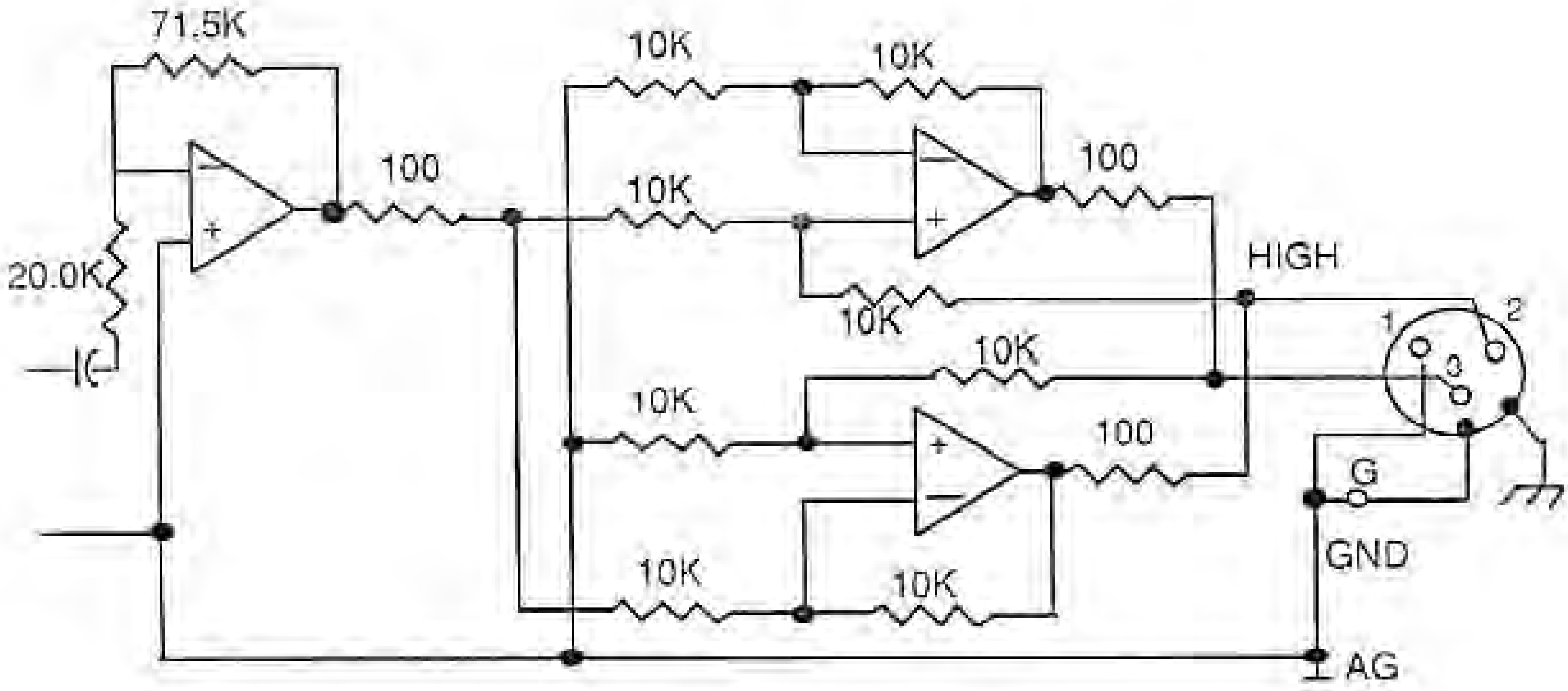
The ADR 68K has four outputs: Main Left and Right, and Aux Left and Right. For some applications, only the Main outputs are used, while for others you will want to have all four connected. The program descriptions later in this manual describe how each program uses the various outputs.

Each output is driven by an active differential circuit. Like the inputs, pin 2 is high, Pin 3 is low, and Pin 1 is ground. Source impedance on pin 2 and 3 is 100 Ω. The minimum recommended load impedance is 600 Ω, while the maximum output level is +17dBV nominal.

Unlike the inputs, the output levels are not adjustable, but are fixed at a +17 dBVfor analog-to-digital convertor full-scale; i.e., when the ADR 68K's signal processor delivers a full-scale digitized signal to the digital-to-analog converter, that signal is delivered to the output at +17dBV.

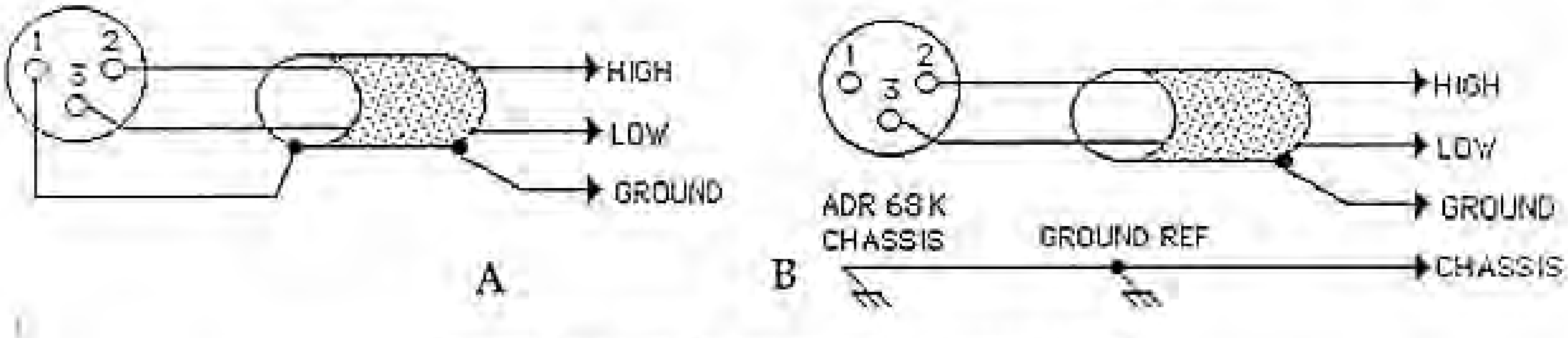
If +17dBV is too high for your other equipment to deal with, we recommend using a resistive attenuator pad between the ADR 68K's outputs and your system inputs. Consult your dealer if you need more help.

The ADR 68K output stage is an active differential design, with 100 Ω source impedance of each output pin, and will drive inputs with 600 Ω or (preferably) higher impedance. On the following page is a schematic of the ADR 68K output stage.



ADR 68K Output Stage

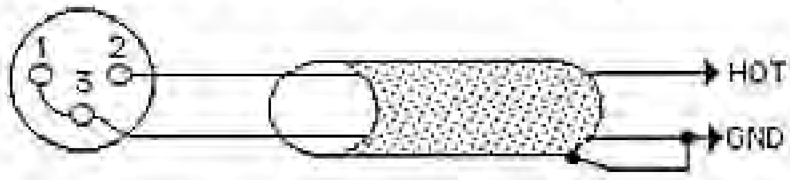
There are two alternative methods of connecting to a balanced load. Diagram "A" is the straightforward method, but diagram "B" may offer advantages in dealing with possible system ground loops. In "B" both the ADR 68K and the load chassis are grounded to a good system reference, and there is no direct connection via output cables. This prevents ground currents from flowing in the cable shields.



Connecting the Outputs to a Balanced Load

For connection to an unbalanced load, use two-conductor shielded cable wired as shown here, with the hot lead connected to pin 2, and pins 1 and 3 tied together to the shield. Note that the connection from pin 3 to pin 1 must be made inside the shell of the XLR-3 plug going into the ADR 68K output connector, not at the other, unbalanced end of the cable. Failure to do this can result in weak or distorted output, or to output stage oscillation.

NOTE: Do not tie the left and right outputs directly together to form a mono sum signal.



Connecting the Outputs to an Unbalanced load

Mainframe Controls and Features

Power Switch

The power switch connects the mains power source to the ADR 68K's power transformer so that the unit will function. When "off," this switch disconnects *both* sides of the power line from the transformer.

Data Cartridge

The data cartridge supplied with the ADR 68K is a non-volatile, removable, memory storage device, intended to hold user-defined presets. After adjusting the parameters of a program to suit your needs, you can name that preset and store it in either in internal memory or in the data cartridge for later use. Each 64K cartridge can hold up to 50 presets. You can buy additional cartridges from AKG.

Note that the data cartridge is not intended to hold new *algorithms* (these can only be changed by swapping the internally-mounted EPROMS — see the chapter on updates for more), nor can it be used to hold sampled sounds recorded by the ADR 68K. It is only for storing user program presets.

The data cartrige can be inserted or removed with the power on, without harm either to it or to the mainframe unit.

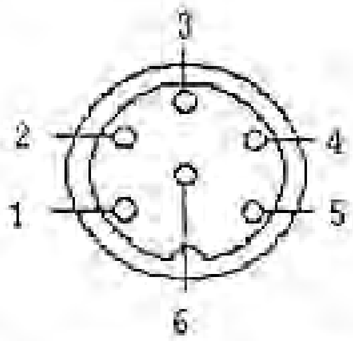
To insert the cartridge, position it in the slot with the label facing up. Push firmly on it, jiggling it around slightly until you feel the socket "give". When it is all the way in, it will not let you push any more. To remove the cartridge, grasp it firmly and pull straight out.

The cartridge has a "write-protect" switch which allows you to make sure that any presets you have stored on it cannot be changed or overwritten. You can use a pen or small screwdriver to switch it on and off (don't use a pencil — the point may break off and cause damage).

Remote Control Connector

The ADR 68K will not work without a properly connected remote unit. A 6-pin female DIN connector on the rear panel receives the connector from the remote control cable. The cable is double-shielded, four-conductor. The unit is supplied from the factory with a 25-foot cable, but if you would like to make your own, longer cable, lengths of up to 100 feet should present no problem. Both ends of the cable are identical, using 6-pin male DIN connectors, with the following pinouts:

pin	signal	wire type and color
1	digital ground	shield
2	digital ground	shield
3	data from remote	green
4	+9 Volts from mainframe	red
5	data to remote	white
6	digital ground for ELP	black



Remote connector wiring
(from wiring side)

MIDI Connectors

The ADR 68K has a full MIDI hardware implementation with standard interface devices and electrical characteristics. There are the usual MIDI In and MIDI Thru connectors, as well as a MIDI Out connector. The MIDI connectors are wired in accordance with the MIDI 1.0 specification, i.e., pins 4 and 5 signal, and pin 2 ground.

As the ADR 68K evolves and develops, the MIDI implementation will be come more flexible and more important.

The Remote Control

Except for those functions which are controlled via MIDI, all ADR 68K control functions must be carried out from the remote. The ADR 68K has no front panel controls except the Power switch.

If either end of the cable connecting the remote to the mainframe is accidentally pulled out while the ADR 68K is operating, there will be no damage (even with the power on), and the cable can be plugged back in.

The remote unit's operating features and protocols are almost entirely defined in software, and so most of them are discussed later in this manual.

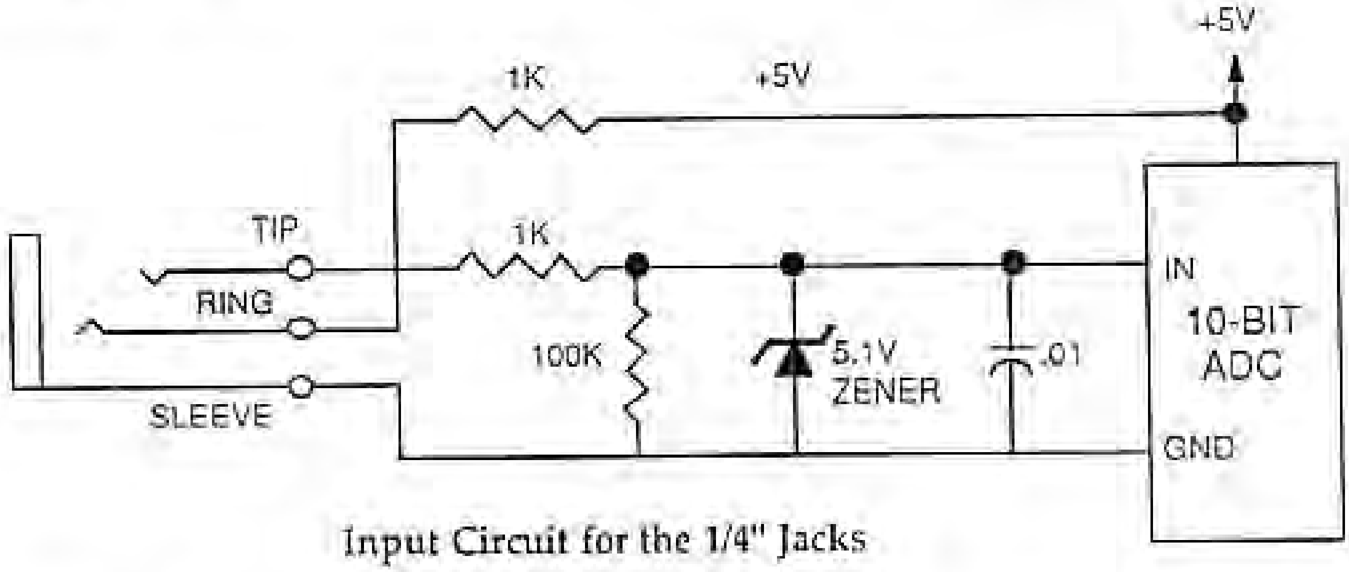
Adjusting the Viewing Angle

You can change the way the characters in the Liquid Crystal Display on the remote appear, to accomodate for different viewing angles. If the display is going to be located below you — say, down at the very front of a mixing desk — move the viewing angle up by holding the Shift button and holding the Up Arrow (alsolabelled "View Up") button until the display becomes clear. If the display is going to be above you — like on a shelf above a console — then hold the Shift button and the Down Arrow ("View Down") button until the display becomes clear. The viewing position is retained in memory when the ADR 68K is turned off.

1/4" Jacks

Four 3-circuit, 1/4" jacks (tip-ring-sleeve type, as used with stereo headphones) are on the rear panel of the remote. These allow foot switches, hand- or foot-operated potentiometers, or synthesizer CV outputs to control the ADR 68K's program parameters or to trigger various types of events. Software in the various programs handles which jacks control what functions.

The jacks require special care in wiring to the control source. This figure shows the wiring of the jacks to an analog-to-digital converter inside the ADR 68K.

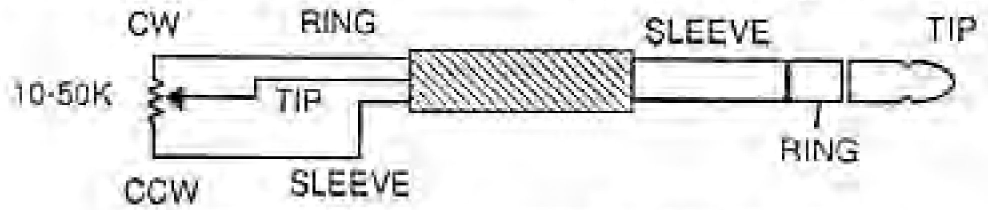


Input Circuit for the 1/4" Jacks

The ADC converts signals in the range of 0 to +5V into a 10-bit word, while software provides hysteresis and rounds the 10-bit result to the nearest 7-bit word. In practice, the ADR 68K usually offers 128 discrete parameter values. Notice that a capacitor provides low pass filtering for noise immunity, and a pull-down resistor holds unused inputs at ground. In addition, a diode clamp to ground will provide some protection from negative input voltages.

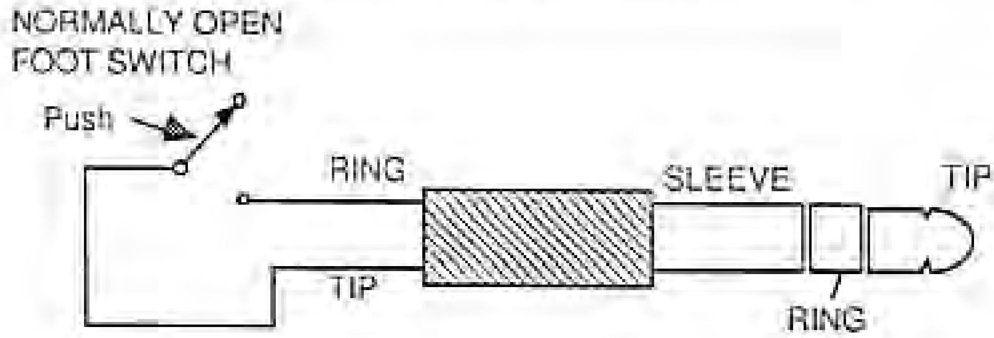
IMPORTANT! *Do not connect any source to either the tip or ring of the 1/4" jacks which can provide greater than +5 volts or less than 0 volts (i.e., a negative voltage). Voltages outside this range could destroy the analog-to-digital convertor, with the result that the ADR 68K becomes totally non-functional.*

Some programs will allow continuous external control over a parameter, from a device such as a foot pedal. A +5V source fed through a 1KΩ resistor is connected to the ring of the 1/4" jacks to serve as a voltage source for such a device. The pedal's potentiometer should be in the range of 10K (preferable) to 50KΩ, with linear taper. Connect the pot clockwise (cw), counter-clockwise (ccw), and wiper as shown.



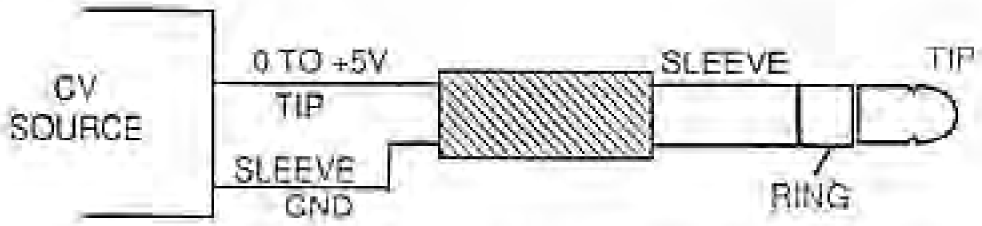
Connecting a Potentiometer to the 1/4" Jacks

For controlling parameters of a binary, on/off nature, you can connect a foot switch, with normally-open style contacts, to the jack as shown here.



Connecting a Foot Switch to the 1/4" Jacks

Many electronic-music synthesizers have "CV" (Control Voltage) outputs, which are generally in the 0 to +5V range. Before connecting such a CV source to the 1/4" jacks, you should verify that it is in fact operating within this range by measuring its extreme positive and its "off" voltages with a digital multimeter. If the CV source does not go above +5V or below 0V (into negative voltages), you can wire it in as shown.



Connecting a Synthesizer's CV Output to the 1/4" Jacks

— Chapter 2 —
General Information

Product Philosophy and Design Goals

The ADR 68K is a multifaceted digital audio processor. Although it produces superb reverberation, it is far more than just another digital reverb. It produces a wide range of useful time-based effects, but it is more than just a ddl. It can store, edit and recall samples, but it is more than a sampler.

In order to produce the most sophisticated, most flexible, and highest-quality digital processing system yet available, the ADR 68K's design team adopted a number of specific goals:

⇒ To have enough power and speed in the signal-processing chain, and in the controlling microprocessor, to support the most ambitious software needs, including concurrent effects processing, and smooth transitions between programs.

⇒ Excellent ergonomics, with a comprehensive but clear user interface.

⇒ Full MIDI capability, to take advantage of the growing importance of MIDI-based music and sound production.

⇒ Sound quality second to none.

⇒ A totally software-based, fully-updatable system, which will allow the unit to remain technologically current over a long lifetime.

⇒ Integrated processing and control software for maximum flexibility and upgradability.

⇒ Plenty of signal memory, with room for further expansion.

⇒ Plenty of EPROM memory space in the controlling microprocessor, to allow for indefinite software expansion.

⇒ Reliability and ease of maintenance.

⇒ Affordable cost.

System Design

The AKG engineers began by choosing a 68000 microprocessor (as used in the Apple Macintosh, the Atari ST, the Commodore Amiga, and most modern CAD workstations) as a central processing unit, and clocked it at 8MHz to obtain enough controlling power so that it could be counted on to fully support the signal-processing and user-interface needs for the long product life intended for the ADR 68K.

The signal processor has 256K 16-bit words of Dynamic RAM for audio memory. This allows over 8 seconds of sound sample storage, at a 16 bit/15kHz performance level; moreover, it is expandable to 1M words, for a potential of 32 seconds of internal sound memory.

A large alpha-numeric display was designed, to keep the user fully informed and in complete control over the unit's great power and variety of features. The engineers chose an easy-to-read four-line LCD with back-lighting, and then equipped the remote unit with convenient scrolling and paging software, allowing easy access to the many virtual pages of control information.

Anticipating the explosion of interest in MIDI — not just in music performance, but in serious recording and video production — the engineers included the capability of full MIDI parameter control (not just register recall), with complete input and output mapping. This extensive MIDI control allows a degree of precision, speed, and repeatability previously unattainable in a digital effects unit.

Sampling and Effects — Separately and Together

Stereo or mono sound sources can be recorded into the ADR's memory and edited internally. Up to 8 seconds of sound memory is available, using the highest quality 16-bit 32-kHz PCM techniques, and this memory can be divided up into as many as four completely independent sections.

A particularly useful innovation of the ADR 68K is the ability to replay samples in memory *and* process them in reverb or effects algorithms concurrently (simultaneously), all internally with no external mixing or looping necessary. Moreover, the internal samples can be mixed with external sound sources before being sent to the processor. For example, a sampled snare drum can be mixed with another percussive source (perhaps using the percussive source as the trigger for the sampled snare drum) and reverberated. Finally, digital mixers at the processor output can mix in a dry portion of the sampled sound as well — all before the signals leave the ADR 68K.

Another way to use the ADR 68K's concurrent-program capability is to split the unit into two separate processors, such as two independent stereo reverbs, each 1-in/2-out. But if bringing two stereo echo returns back to your mixer is inconvenient, the ADR 68K allows you to mix the two reverberators internally, along with the dry signals as well, into a single stereo output. Simply scroll to the Main Output Mixer page and use the mixing controls you'll find there.

As software for the ADR 68K develops, there will be effects programs that will allow a number of "virtual" digital signal processors to be combined and adjusted concurrently.

Sound Quality

The ADR 68K conforms to a high standard of audio design, using compact-disc-quality 16-bit converters, coupled to quiet, high-slew rate op amps and well-behaved sample and hold circuits. Anti-aliasing and output filters are 11th-order low-ripple hybrid low-pass designs. No pre-emphasis has been used, thus ensuring full-bandwidth performance at high signal levels. A proprietary analog dither circuit is included to improve the ADC's linearity at low levels, so that the residual noise with small signals is always Gaussian.

The digital signal processor retains 32-bit internal precision for most intermediate results, and uses 16-bit storage for other internal signals. Coefficients can be specified to a full 16 bits of resolution, enabling very accurate transfer functions, high-Q filtering, and long decay times.

Fully Software-Based Signal Processing

To ensure a long product life in today's era of rapid technological change, a product like the ADR 68K must cast as little as possible of its processing power in the "concrete" of digital hardware. This means that the signal processor has to be fully controlled by microcode of substantial width, coming completely from RAM, under control of the system microprocessor. In the ADR 68K, not only the microcode, but *all* of the critical signal processor data (coefficients, delay offsets, sampling rate, etc.) are software-based and therefore freely modifiable. The result is a general-purpose signal-processing computer with infinite room for growth. The ADR 68K is truly a computing engine, a hardware bed, that will support a long future of sound-processing development.

Integrated System and Sound Software

The ADR 68K is unique among digital effects units in that the software for the microprocessor that controls the *system operation*, and the software for the micrprocessor that handles the actual *signal processing*, are totally integrated. In other systems this is not so, which inevitably leads to several limitations.

In other units, the inflexibility of the original operating system eventually imposes a limit on how far the effects software can be improved and expanded. Some of the problems that come back to haunt the manufacturers of some systems are inadequate displays, not enough controls, controls with painted-on labels, and internal hardware rigidity. Any of these can lead to an earlier-than-necessary obsolesence of an expensive piece of equipment.

The ADR 68K, on the other hand, uses a large display with soft-labeled switches and slide controls that can be configured and used — in perhaps unanticipated ways — over an extended lifetime. All of the ADR 68K's software resides in the 68000 EPROM set. As additional sound programs are developed, the operating and control software can be simultaneously updated and extended to harness the full potential of the new programs.

Reliability and Maintenance

Today's audio products are sophisticated and complex, yet they must work day after day, sometimes in demanding environments. The ADR 68K has been designed to work hard and work long. For example, knowing that heat is the enemy of reliability, AKG's engineers used low power CMOS IC's wherever possible to keep power consumption to a minimum.

A detailed heat-transfer and convection airflow analysis led to a design that requires no noise-producing fan, yet keeps the internal circuitry comfortably cool. To accomplish this, ventilation holes, PC planes, and an internal baffle steer convection currents over the PC boards and out the top cover. Eschewing a switching power supply and its attendant noise and reliability problems, a simple, reliable linear power supply was incorporated.

Furthermore, the major heat-producing components of the power supply — the transformer and 5-volt regulator — are mounted externally so that they radiate into the outside world, not into the internal circuitry.

The ADR 68K uses three main PC boards, each removable with hand tools and no soldering iron, allowing practical module-level field maintenance. All IC's are socketed, making chip-level repair easy and non-destructive.

Each unit is tested, burned-in for seven days, subjected to 15 minutes of vibration stress, and then completely re-tested. During burn-in, power is cycled on and off, and air flow is restricted, to subject the units to tougher conditions than they are ever likely to encounter in the field.

Needless to say, all components are chosen for high quality and for conservative ratings.

Specifications

Sampling Rate: 32kHz, for 15kHz bandwidth.
Conversion format: True 16-bit linear PCM, for ≥86dB dynamic range; utilizes dither for improved low level performance.
Frequency Response: 20Hz-15kHz, +0/-2dB in straight delay mode.
Noise and Distortion: ≤0.03% total, 0.01% typical, in straight delay mode, @ 400 Hz, @ operating level.
Filters: 11th-order at both input and output.

Internal Audio Memory: 256K 16-bit available for audio delay (8 seconds of sound).
Internal Program Memory: holds 50 user and/or factory presets. Retention: 10 years estimated.
Data Cartridge: 64k, holds 50 user presets. Retention: 10 years estimated.

Size: standard rack mount, two units high, 19 x 3.5 x 13" deep (48 x 8.8 x 33 cm), excluding XLR-connector protrusion.
Weight: approximately 30 lbs (fully boxed for shipment).

Inputs: stereo, electronically balanced (differential amplifier). Pin 2 high, Pin 3 low, and Pin 1 ground.

- **Input impedance:** pin 2, 11KΩ, pin 3, 21KΩ.
- **Recommended source impedance:** ≤600Ω.
- **Maximum source voltage before input stage overload:** +17dBV (7Vrms).
- **Minimum input for operation of display** (@ 100Hz): -10dBV (.316Vrms).
- **Sensitivity settings:** -10, 0, +8, and +18dBV, available by jumper selection.
- **Connectors:** XLR-3 female.

Outputs: two stereo pairs, active differential circuit. Pin 2 high, Pin 3 low, and Pin 1 ground.

- **Source resistance, pins 2 and 3:** 100 Ω.
- **Minimum recommended load impedance:** 600 Ω.
- **Maximum output level:** +17dBV nominal.
- **Connectors:** XLR-3 male.

Power: 100, 120, 220, or 240VAC nominal voltage (selectable via internal switch), 50 or 60Hz.

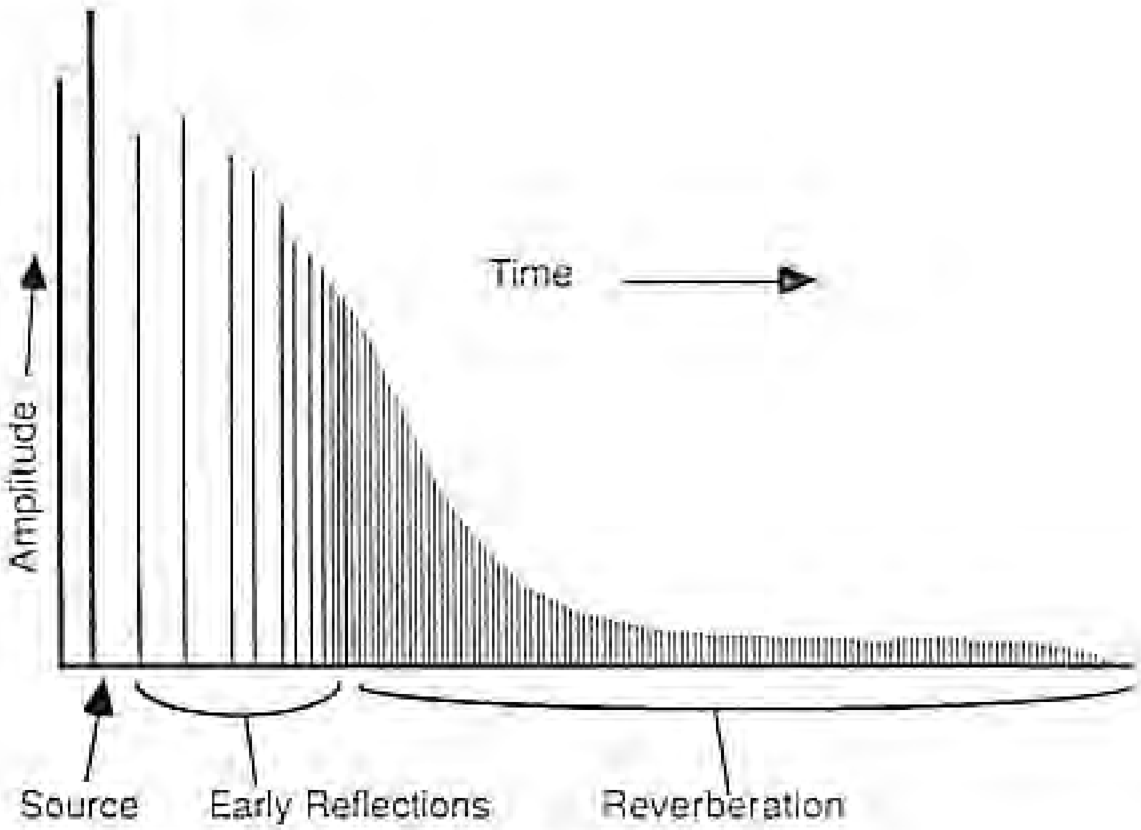
- **Minimum voltage:** power supply maintains regulation to approximately 80% of the nominal setting.
- **Maximum voltage:** 130/260VAC.
- **Power consumption:** approximately 90 watts.
- **Power cord:** detachable IEC standard.
- **Fuses:** All internal; two mains (US type 3AG, SLO BLO, 1.5A); one for the +5VDC supply (US type 8AG, 8A). All supplies are current and power limited.

Environment: 10-50° C. operating, 0-70° C. stored. RH (relative humidity), up to 95% non-condensing.

About Reverb

Reverberation is a term used to describe the effects of space on a sound. Acoustic space is a crucial component in human auditory perception — the human brain/ear "psychoacoustic" system is one of the most sophisticated in nature — and can be as important in the way we perceive a sound as the nature of the sound itself.

Strictly speaking, reverberation is only one of three types of sound information we hear when "processing" a space — but the word is also used to describe the overall phenomenon. The other components involved are direct sound and early reflections.



The Three Components of Reverberation

Direct sound from the source, because it travels the shortest path from the source to the ear, arrives first. It is a "pure" sound, in that it has not been influenced by anything other than its source. It lasts only as long as the source continues to produce it, and its frequency response is relatively true to the original. Transients are transmitted faithfully. Direct sound is responsible for telling us the apparent source position of a sound, which is calculated by the brain/ear system using time and amplitude differences between the two ears as the sound arrives at them.

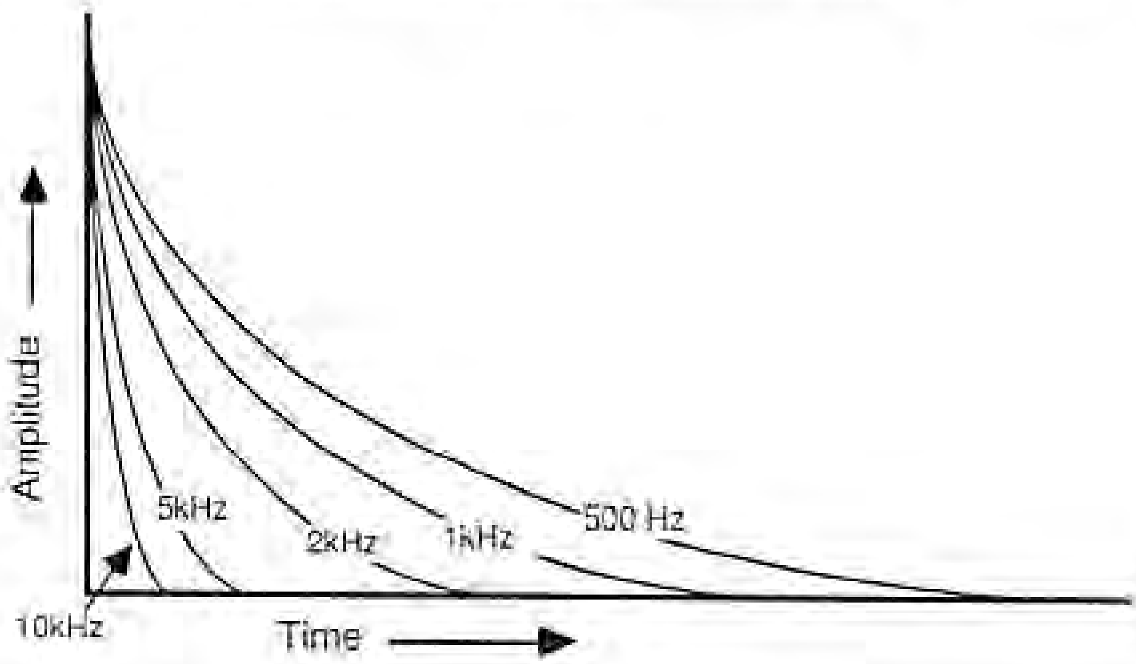
Following immediately after the direct sound are the early reflections. The early reflections tell us quite a bit about the space the sound source is located in. The time of the first reflection shows the distance of the sound source from the nearest wall, as well as the distance between our ear and that wall. Subsequent reflections, which come increasingly closer together after the first one, define the position of the other surfaces.

Because all sound-reflective surfaces absorb different frequencies to one degree or another, the frequency-response characteristics of the reflections will change depending on the nature of the surfaces: hard, flat walls reflect higher frequencies better, while softer, rougher surfaces absorb the higher frequencies and so will seem to reflect lower frequencies better. Air itself has some high-frequency absorption characteristics, and so the later reflections will have less high-frequency content regardless of the nature of the space.

The early reflections also tend to smear and fuse with the direct sound, due to the "Haas effect", which causes us not to hear the reflections as distinct sounds, but rather as part of the original direct sound. The influence the reflections have on the direct sound is to make it louder and more diffuse, giving it warmth and ambience, and actually making it "stereo", by placing it in a defined space.

Before very long, the reflections come fast and furious enough that they blend together into one smooth, incoherent overall sound that outlasts the original source by a considerable length of time. This, strictly speaking, is what we call reverberation. Except in a very dry room, most of the energy a sound produces goes into making reverberation, and so unless we are very near the source, and therefore getting a high proportion of direct sound, most of what we hear in a music performance is reverberation.

Reverberation has a frequency spectrum over time that looks like an increasing low-pass filter, as the high-frequency energy is gradually lost by air and surface absorption. Again, the nature of the room surfaces will determine the frequency balance of the reverberated signal, while the size of the room, as well as its shape, surface materials, and the presence of any openings, will determine its overall length. The decay time of a reverberation signal is expressed as "RT_{60}", which is the time the signal takes to fade down to 60 dB below its maximum level.



Reverb decay as a function of frequency in a typical space

In recording and reinforcement situations, artificial reverberation is used to heighten a sound's sense of "reality", by adding warmth, depth, ambience, and spatial diffusion. Of course, artifical reverberation can also be used to produce "unreal" effects, like shifting spaces; "double" rooms with more than one reverb characteristic; "gated" spaces with a long-sounding decay time at the beginning of the sound but a very short overall decay; or other types of acoustic spaces that would never exist in nature.

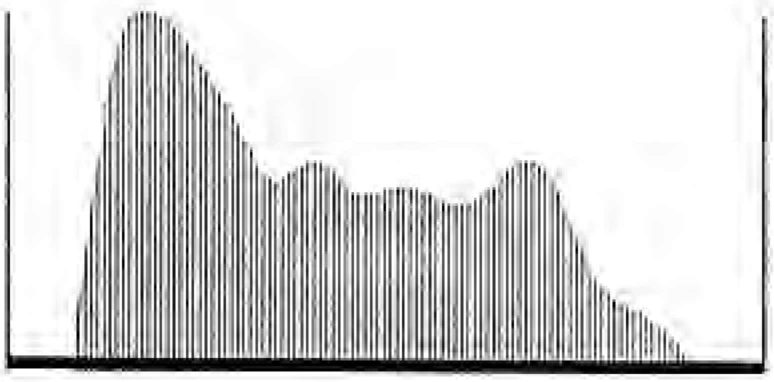
One of the traditional ways reverberation is generated artificially is by electro-mechanical means: a transducer carrying the signal to be reverberated is attached to a device like a spring or metal plate, which is "excited" by the signal, and a complementary transducer is placed at the other end to pick up the reflections produced by the metal. Although a metal plate or spring cannot possibly have the same acoustic characteristics as a real room, these devices — especially when combined with dynamics and/or frequency processors — can do a creditable job of producing reverberation. Another way is to use an "acoustic chamber": a small, live room with a speaker at one end and a pair of microphones at the other. Although they represent an intrinsically much more natural-sounding approach to reverb, acoustic chambers are not always practical, especially where space is at a premium or complete acoustic isolation is difficult.

With the advent of cheap microprocessors, however, the preferred method of producing reverb has become to use digital circuitry. With sophisticated hardware and software, digital reverbs can now achieve a sound quality and degree of control previously unattainable. Perhaps most importantly, units like the ADR 68K, with their "open" hardware and software architecture, can reflect the changing needs and tastes of a volatile audio industry without ever becoming obsolete.

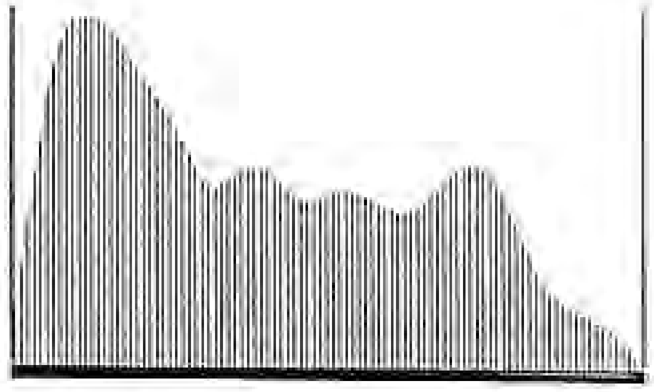
About Sampling

Sampling, the purely electronic recording and playback of "real" sounds, has in recent years become one of the most powerful and popular tools in the recording engineer's arsenal. Sampling consists of converting an analog signal to digital data, storing the data in memory (sometimes off-loading it onto a disk or other solid medium), and then on cue, reading it out from the memory and converting it back to analog. Depending on the type of device storing the samples, playback can be initiated with any of a number of triggers, including audio signals, control signals, or MIDI commands.

In its simplest form, sampling is a one-step "write/read" operation, in which the sound produced is unchanged from the sound recorded. At its most complex, a sampling device can include facilities for editing the beginning and end of the recorded sound; changing its pitch (playback rate); setting of "loops", so that a portion of a sound can continue indefinitely; dynamic filtering, to impose synthesizer-like timbral changes; envelope overlay, to change the dynamic characteristics of the original sound; and even multi-timbral splicing or layering of more than one sample.



Raw sample

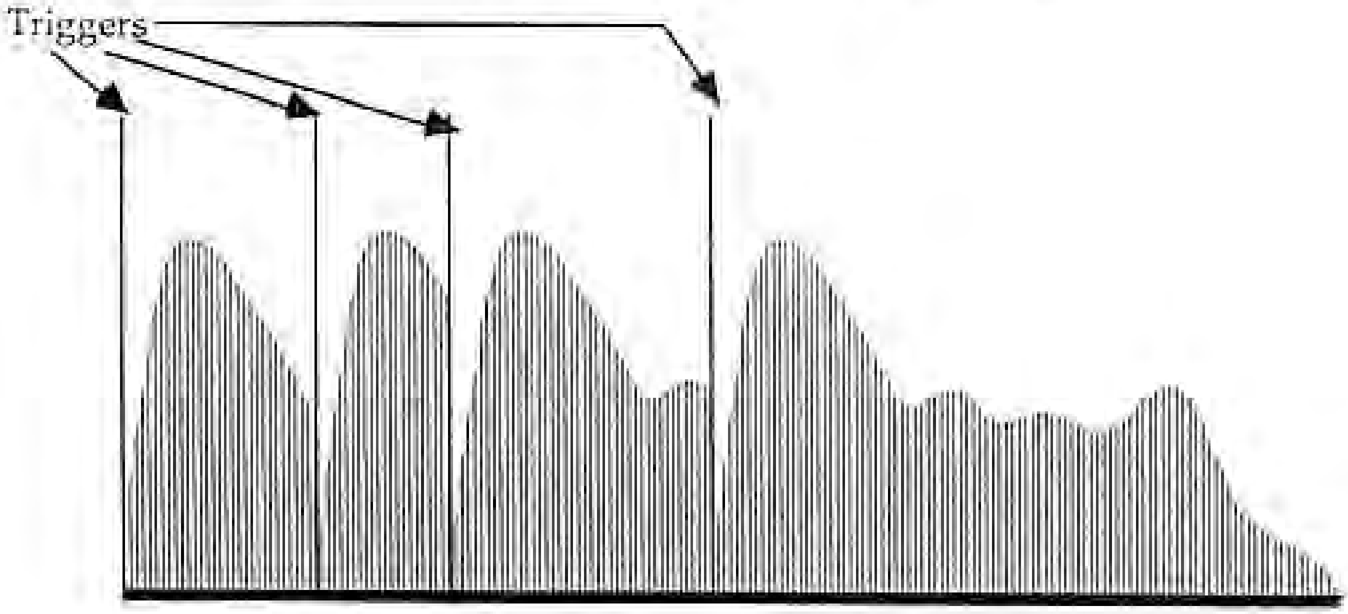


Edited sample

As with all forms of digital audio, the quality and length of a sampled sound depend on the sampling rate; the number of bits in each sample; the characteristics of the filters used on the analog-to-digital and digital-to-analog converters; and the amount of memory available. The Nyquist Theorem states that the sampling rate of a sound must be at least twice that of the highest frequency of the sound, so that if a sound is sampled at 20 kHz, none of its harmonic components above 10 kHz will be recorded. In practice, the upper frequency limit is always *less* than half of the sampling rate (or "Nyquist number") to allow for the action of the filters. The filters are necessary to insure that no frequencies above the Nyquist number get into the converters, where they can cause mathematical errors, known as "aliasing". Because no filter has an infinite slope (although it can be very steep), the usable audio bandwidth is necessarily reduced.

The number of bits in each sample determines the dynamic range. A 16-bit sampler has a dynamic range of approximate 96 dB. The amount of available memory, divided by the sampling rate and the word length, determines the maximum length of the samples. A 16-bit sample, one second long, recorded at a rate of 40 kHz, will fill 80,000 (eight-bit) bytes of RAM — about 78K.

Samples have many uses. There is a well-known story of a famous '70s rock group making an '80s "comeback" album who could not seem to get a good snare sound in the studio, so they sampled a snare shot from one of their old albums, and had it play on the track, triggered by the live snare. One successful English producer made a practice for a while (until it got to be clichéd) of including sampled "orchestral hits" — chords played by a full orchestra — on his rock records. Avant-garde and rap musicians love the "stuttering" effect produced by a triggering a sample multiple times before it has a chance to play all the way through.



Re-triggering a sample for the "stuttering" effect

Samples can also be used to enhance sounds being played on other instruments. A gong sample, for example, can be triggered by a cymbal crash, but only when the cymbal is played at a certain volume. A scream can be added to a guitar passage when the guitar reaches a specified pitch. In the world of jingles, samplers can be used to store vocal passages, phrases, or "logos", which can then be repeated and inserted into a mix much more easily and accurately than they can if they were on a separate piece of tape.

Because the technology is relatively new, many of the potential uses for sampling have not been thought up yet. We hope the ADR 68K will make high-quality sampling simple enough for users to come up with many unique and exciting applications.

About MIDI

MIDI, the Musical Instrument Digital Interface, was introduced in June, 1983, after several years of research and discussions by a consortium of Japanese and American music-hardware manufacturers. As a communciations standard for use among sound generators, processors, computers, and other music-related electronic devices, it has met with great success, and it is now in wide use in live performance, music recording, and film and video sound production.

What it is

MIDI is a digital data stream consisting of 10-bit bytes, with a data rate of 31,250 bits per second. A MIDI command or "message" consists of two or more bytes, and the definition of each command is set by MIDI Specification 1.0. All MIDI-equipped devices respond to MIDI messages in essentially the same way.

Originally, MIDI's primary use was to get sound-generating devices (synthesizers) to "talk" to each other, by means of specific digital commands for turning on and off notes, moving real-time controllers like pedals and wheels, and changing programs (or "patches"). More recently, MIDI capability has been developed for sound-processing equipment, including digital delays and reverbs, samplers, mixers, and equalizers; studio gear like tape recorder locaters; and even lighting equipment.

The most common generator of MIDI data is a musical keyboard — either one built into a synthesizer, or a "dedicated" keyboard not associated with a synthesizer; but this too is changing, as MIDI guitars, drums, vibraphones, wind instruments, and other devices are developed, as well as computer programs that can generate MIDI data all by themselves.

MIDI is not music — it is a digital description of "events" that have musical meaning. By itself it has no sound, and it cannot be recorded on standard audio tape. It can, however, be recorded by a device known as a "sequencer". A sequencer, which remembers MIDI data and allows it to be edited and played back much as a tape machine records music, can be built into a synthesizer, or it can be a dedicated stand-alone unit, or it can be a computer equipped with a MIDI interface and appropriate software.

MIDI connectors

Most MIDI equipment has three five-pin DIN plugs marked "In", "Out", and "Thru". The "MIDI In" jack accepts data from an outside source, such as a keyboard or sequencer. The "MIDI Out" jack sends data generated by the unit itself. Data coming into "In" jack also appears at the "MIDI Thru" jack, usually without modification.

The "Thru" function is important because, unlike audio signals, MIDI signals cannot be split with a simple "Y" connector. If the output of a MIDI source is to feed two different devices, the data stream must be "daisy-chained", by first feeding one unit's "In" jack, and then connecting the "Thru" jack to the second unit's "In" jack. An alternative is to use a splitting device, known as a "Thru box".

The opposite problem —in which the output from two devices is to feed one input — can be solved with a "MIDI mixer" (not to be confused with a "MIDI-controlled mixer", which is an audio mixer that responds to MIDI data). Some MIDI devices, like keyboards or drum pads, offer mixing capabilities in the form of a "merge" function, in which the MIDI data coming into the "In" jack is combined at the "Thru" jack with the data being generated by the device itself.

MIDI messages

Channels and modes: The MIDI specification allows for 16 discrete data channels on one MIDI line. It also describes four "modes", or states, for equipment that receives MIDI data. These are "Omni On/Poly", "Omni Off/Poly", "Omni On/Mono", and "Omni Off/Mono". The last two are special modes for synthesizers capable of playing multiple timbres.

When a device is set to "Omni On", it will read and respond to all MIDI data being sent to it, regardless of which channel it is being sent on. With "Omni Off", the device will respond only to data on one specific MIDI channel (or in the Mono mode, several specific channels), and will ignore the rest. Therefore, 16 different devices, each with Omni Off and each reading a different channel, can be sent separate MIDI instructions over the same data line.

Note-ons and -offs: These are the most common MIDI messages, and generally correspond to the pressing and releasing of keys on a synthesizer. Each note command is followed by a "velocity" value, which refers to how *quickly* (almost the same, but not quite, as how *hard*) the key is pressed or released. The range of notes is ten octaves plus a perfect fifth, and they are numbered 0 to 127, with 60 generally corresponding to Middle "C". Velocity values similarly range from 0 to 127. A special provision of the MIDI specification allows a note-on command with 0 velocity to substitute for a note-off command.

Controllers: These refer to "real-time" controllers, which can either be simple on-off switches, such as a sustain pedal, or "continuous", such as a modulation wheel, a volume pedal, or a breath controller. Keyboard aftertouch, the pressure on a key after it is played, fits into the latter category as well. Controllers have an identifying *number* (from 0 to 127), and an immediate *value* (also from 0 to 127). Controller values are absolute — they define the *position* of a switch or wheel, not its relative movement. Many controllers, such as pitch bend, aftertouch, modulation depth, and sustain, are "pre-assigned" in the MIDI specification, but the majority are not. Each MIDI channel can have its own full complement of controllers.

Controllers expand the expressive capabilities of MIDI significantly, but suffer from one serious drawback. They often generate data very densely: a simple motion of a pitch-bend wheel can produce 128 consecutive messages in a small fraction of a second. Because the bandwidth of MIDI is finite, it is possible to "choke" the MIDI data stream with excessive controller movements. This results in the data momentarily slowing down, and instruments' responses being delayed, which can lead to audible "hiccups", especially in highly rhythmic, multi-channel performances. There are various solutions to this problem in theory, but the most practical one is to use controllers sparingly in such a situation.

Program changes: These messages generally instruct a synthesizer to change "patches" — to go from one type of sound to another — or a processor to change from one type of processing program to another. They are also used in some MIDI-controlled audio mixers to change the mixer from one "state" to another.

System Exclusive: These messages contain a "header" with a specific identification code exclusive to a specific MIDI device. Each manufacturer of MIDI equipment is assigned its own ID code by the MIDI Manufacturers Association, and many manufacturers then develop their own "sub-IDs" to distinguish between the various models in their line. When a piece of hardware receives a system-exclusive message containing its ID code, it responds. If the message contains a different ID code, it is ignored.

System-exclusive messages are most often used for transferring complete descriptions of patches (as opposed to a patch *number*, which is a single integer) or even banks of patches, between identical synthesizers or between a synthesizer and a storage device, such as a computer. A description of a patch can consist of several hundred separate parameters, and a bank of patches can contain 128 such descriptions. Therefore these messages are rarely sent in real time as part of a performance. However, system-exclusive messages can also be sent in small amounts, to change a single parameter in a patch, and these can theoretically be used in real time. Few sequencers, however, allow recording and/or editing of system exclusive messages, and many synthesizers will "glitch" if they receive them while they are actually playing.

Local Control: The local control function allows a MIDI instrument to de-couple its keyboard (and on-board controllers) from its sound-generating circuitry. This allows the keyboard to be used to control another device, while it makes no sound itself. However, the sound-generating part of the synthesizer can be controlled from a remote device, such as an external sequencer. With "Local Control On", an instrument will play itself. With "Local Control Off", it will not.

The advantage of this feature is in a multi-synthesizer sequencing system, in which one keyboard serves as a master. When a track is being recorded for a synthesizer *other* than the master, it is less confusing if the master synthesizer is not sounding at the same time.

Other commands: The MIDI specification includes several other commands such as System Reset, Tune Request, Pointers, MIDI Time Code, and Sync, which need not concern us here. They are used in large automated setups and computer- or sequencer-based studios.

More information on MIDI can be obtained from the International MIDI Association, 11857 Hartsook Street, N. Hollywood, CA 91607.

Glossary of terms used in this manual

Algorithm

is an "rule" which defines how the ADR 68K will interact with the outside world. Among other things, algorithms define a *range* of control functions, and a specific setting of those functions is called a Program or Preset (see). Programs that follow the same algorithm are usually found within a single Bank (see), and behave somewhat similarly. When a Preset is displayed on the LCD, the name of the algorithm that it is associated with is displayed along with it.The ADR 68K will have 50 or more presets, but only about a dozen algorithms.

Auto Rpt

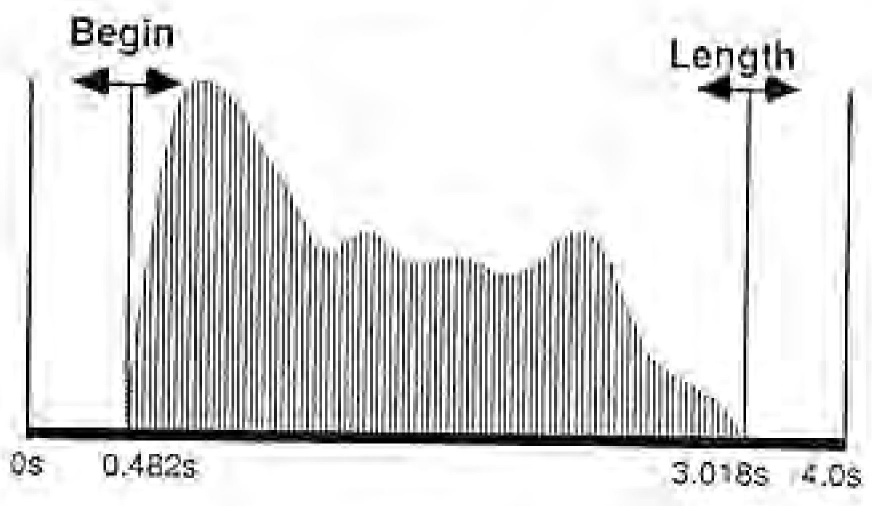
see Play mode

Bank

is a group of numbered Registers (see) in which Programs or Presets (see) are stored. The ADR 68K's internal memory is divided into six banks. Generally speaking, presets within a bank are related to each other in some functional way.

Begin

is used when editing samples in the sampler programs. The "Begin" value is the point within the sample at which it will start playing when triggered. The value can be anywhere from 0 (the starting point of the sample as originally recorded) to the length of the segment. Both coarse- and fine-adjustment controls are provided. See Length.

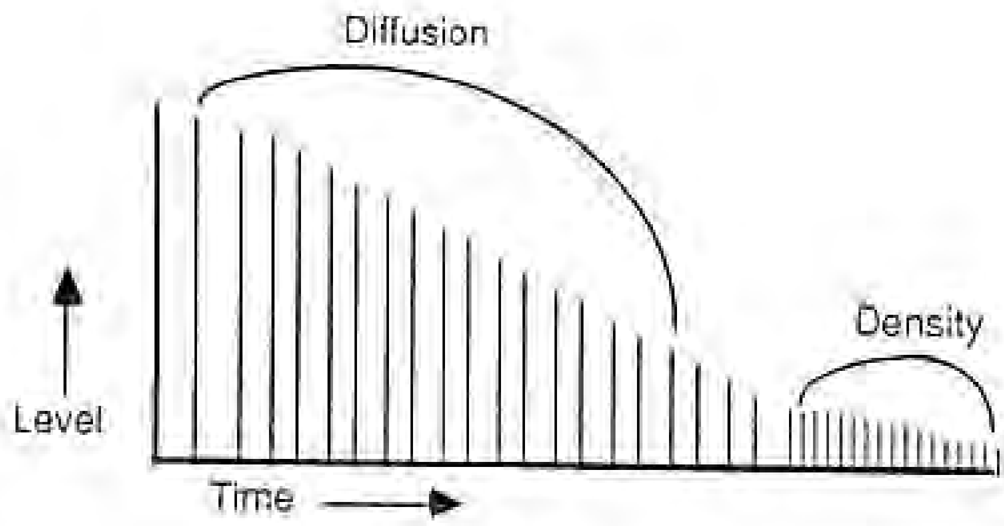


Adjusting Begin and Length to edit a sample

Density refers to the concentration of echoes at the end of the reverb — the "tail". Using too low a value of Density will prevent the reverb from ever reaching an echo density high enough to blur impulse sounds into a swish of reverberation, and therefore will make the sound "lumpy". On the other hand, too high a value, when used with vocal sounds or solo instrumental lines, tends to make them blurry and less intelligible. See also **Diffusion**.

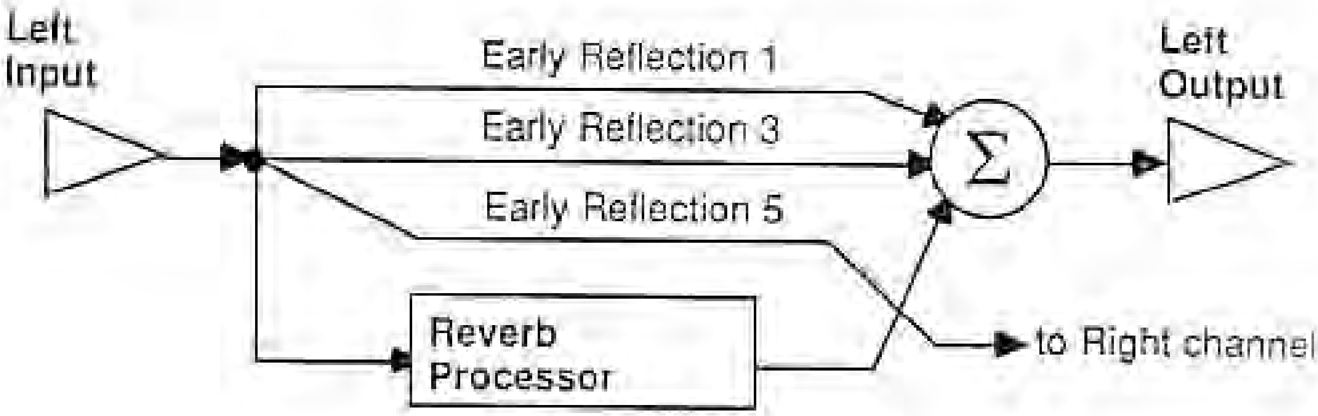
Depth represents the apparent distance of the listening position from the apparent source in a hall. Large values give the impression of being farther back in the hall, thereby hearing a more incoherent, more diffuse reverberation process.

Diffusion (Diffus) as opposed to **Density**, refers to the concentration of echoes at the *beginning* of the reverb. A Diffusion setting of 100 will give a very dense, incoherent reverb in which impulses spread into a rushing, almost white noise kind of sound.



Early Reflections (E-Rs)

in some programs, can be adjusted individually for time delay and level. Once the individual reflection levels are adjusted, the overall level of all the reflections can be set with the E-R's control of the Output Mixer page. In reverb programs with true stereo inputs, some reflections originate from the left input path, and others from the right input path. The reflections are then sent out in stereo as well, although there is some crossing of the reflections from one channel to the other as they are processed.



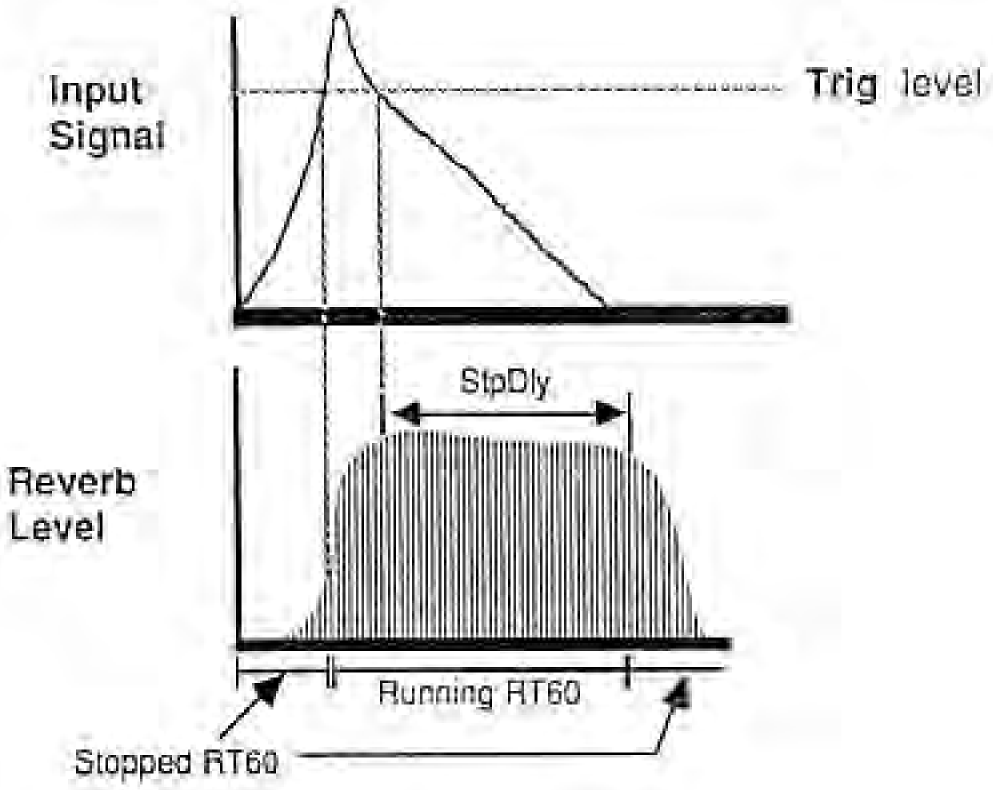
Early Reflections

Gated Reverb

is a type of reverb which has two distinct characteristics. In common practice, a gated reverb program starts out sounding as if it has a very long decay time (RT60), but then it cuts off abruptly after the source signal is removed. It can also work the other way, however — sounds can start off with short decay times, but as they die down the decay time lengthens.

In the ADR 68K, the two components of gated reverb are called "Running Reverb", which comprises the characteristics of the reverb while the input signal is present; and "Stopped Reverb", which are the reverb characteristics after the input level drops below a prescribed level (see Trig). In addition, after the input signal drops below the Trig level, there is an adjustable amount of time before the Stopped Reverb "kicks in" — see **StpDly**. All of the reverb programs in the ADR 68K can be gated.

— see diagram next page



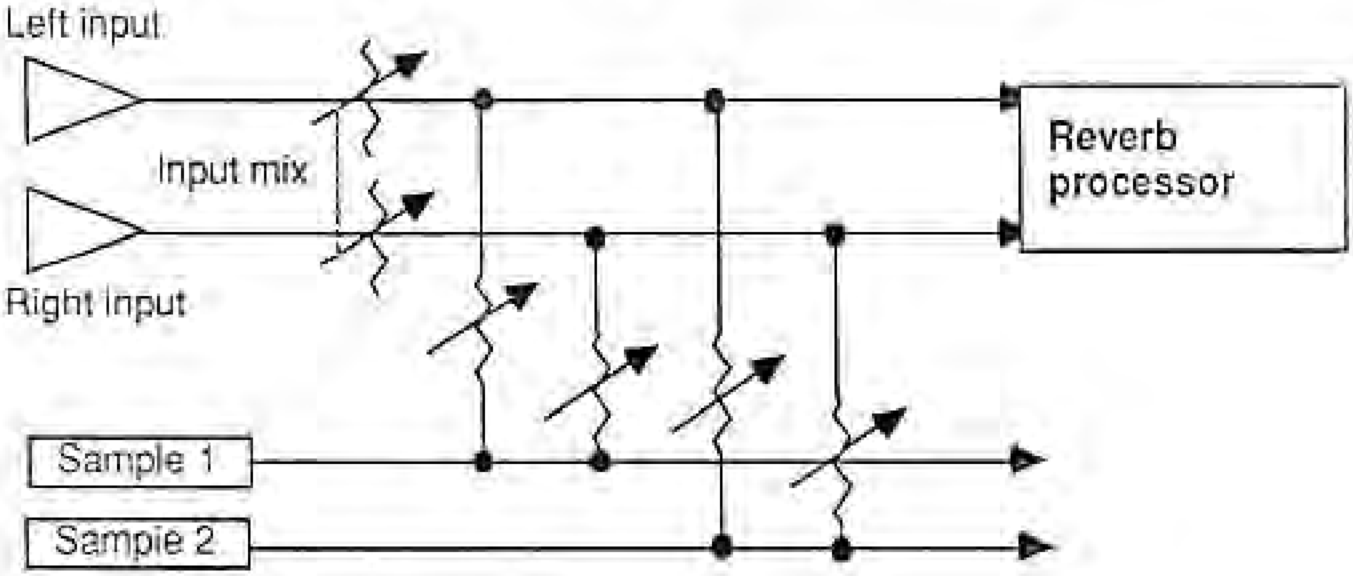
Gated Reverb

HF Dcy is the factor by which the reverb time (RT60) is reduced (or increased) from its mid-band value for frequencies above approximately 2 kHz. An RT60 of 3 seconds with HF Dcy factor of X0.2 implies a high-frequency decay time of 0.6s. In gated reverb modes, there is an HF Dcy for Running reverb, and another for Stopped reverb. See **LF Dcy**.

HF BW is the corner frequency of a low-pass filter, generally at the input of the reverberator. Reducing the HF BW will give a more mellow sound.

Input Mixer

is a software-controlled internal audio mixer which gives level control over the various sound sources (live inputs, stored samples, etc.) before they reach the reverberator.



Typical Input Mixer

Length

like Begin (see), is used when editing samples in the sampler programs. The "Length" value is the amount of the original sample that will be played when it is triggered. The value can be anywhere from 0 to the original segment maximum. Both coarse- and fine-adjustment controls are provided. If the Length is kept long enough, and the "Begin" value is made fairly high, the sample will "wrap around" when it is played, with the end of the original sample sounding first, followed by the beginning.

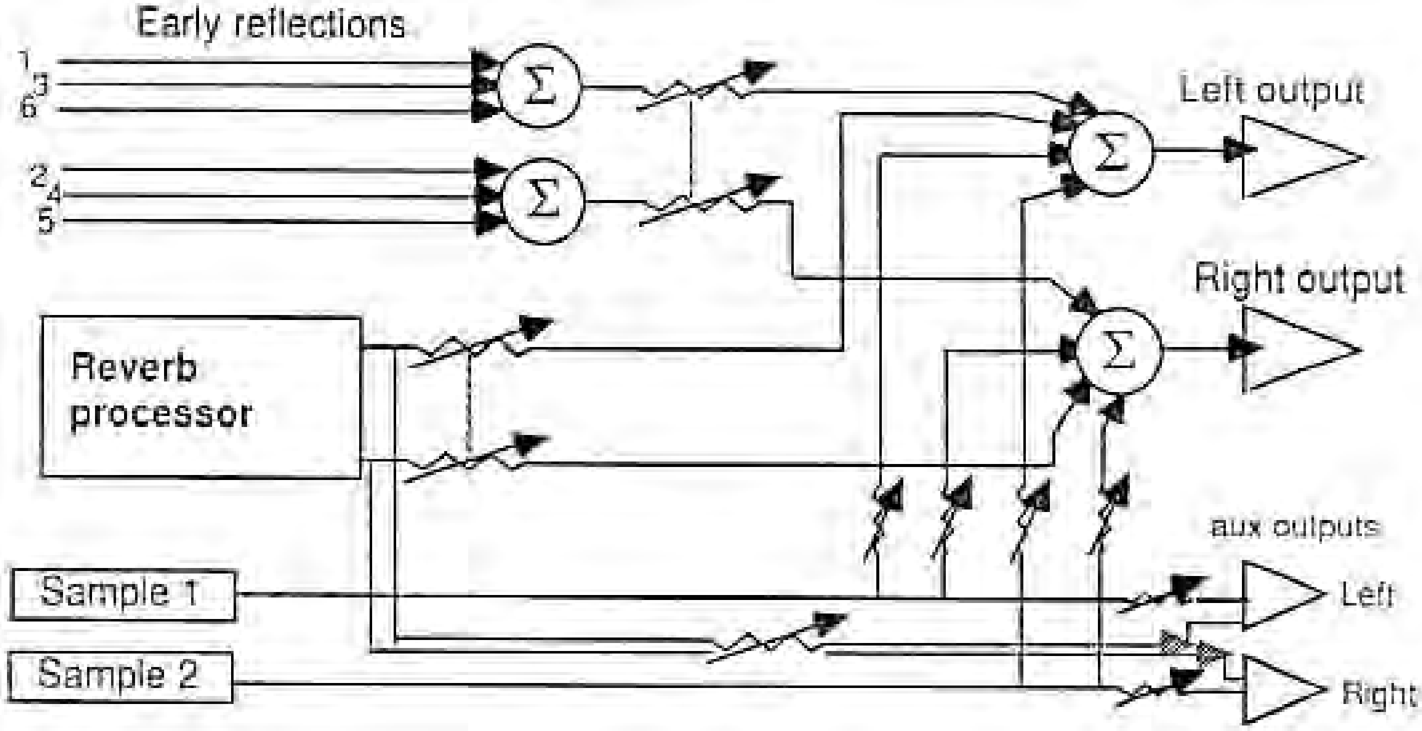
Level

is the adjustment from Off, or a minimum value, to 0dB of the level of sounds — incoming, outgoing, or internal.

LF Dcy

is the factor by which the reverb time (RT60) is reduced (or increased) from its mid-band value for frequencies below approximately 500 Hz. An RT60 of 3 seconds with LF Dcy factor of X0.2 implies a low-frequency decay time of 0.6s. In gated reverb modes, there is an LF Dcy for Running reverb, and another for Stopped reverb. See HF Dcy.

Output Mixers software-controlled internal audio mixers that, in reverb programs, give control over the various sound sources before they reach the various ADR 68K outputs. Separate mixers are available for the main and aux outputs. There are individual adjustments for the early-reflections cluster ("E-R"), the reverb ("Rev'b"), and any samples in storage that are available to the reverb program. The samples, being mono sources, also have pan controls.



Typical Output Mixer

Play Mode (Auto Rpt ON/OFF) in sampler programs, determines whether sample playback will be continuously looping or triggered.

Play Trig in sampler programs, adjusts the threshold of input signal that will cause a stored sample to be replayed.

Pre-Delay (P'Dly) is a time delay given to a signal before it goes into the reverb processor. A P'Dly of 0 does not mean that the reverb begins immediately: there is always an inherent delay in the build-up of a reverb sound. The P'Dly does not apply to the Early Reflections.

Preset — also referred to as "Program", is a particular application of an algorithm. While an algorithm defines a *range* of control settings, a Preset defines a set of *specific* settings. A preset also includes MIDI assignments and perhaps some system settings. Presets are stored in numbered Registers, which are grouped together in Banks, either in Internal memory or on the data Cartridge.

Randomization (Random) — is an effect used within some reverb programs to smooth out the end of the reverb — the "tail" — and make it less repetitious. The feature uses multiple taps within the reverb, and moves them back and forth in time, with a low-frequency oscillator.

Record Mode (Safety ON/OFF) — in sampler programs, sets up record readiness. When this control is ON, recording cannot begin, so stored samples are kept safe. When it is OFF, recording can begin.

Register — is a numbered location referring to a portion of memory, Internal or Cartridge, where a user or factory Preset can be stored.

Running Reverb — is the reverberation heard in a gated reverb program while the source sound is still present. It is also the *only* reverb in a gated program in which the gate has been disabled (Trig set to OFF). See **Gated Reverb and Stopped Reverb.**

RT_{60} — is the formal name for reverberation decay time. It is the time it takes for reverberation to drop 60dB from its running reverb sound level. In gated reverb programs, there is an RT60 for running reverb, and another for stopped reverb.

Safety — see **Record Mode**

Samp1, Samp2, etc. — refers to the level of the stored sample(s) as they are mixed into the input or output mixer when they are being replayed.

Sample Trig sets up the method, if any, by which a stored sample will be commanded to be played. Triggering may be Manual, using the appropriate Soft Button on the ADR 68K remote unit; it may be controlled by an external footswitch plugged into one of the 1/4-inch jacks on the remote; or it may be automatic, initiated by a circuit sensing the level of the input signal coming into one or both audio inputs. The sensitivity for automatic triggering can be set anywhere from a very low value, such as -40dB — in which case almost any sound coming into the input will cause the sample to be played — up to a high value, such as 0dB — in which case only very loud sounds will trigger the sample.

Size is a relative measure of the size of the space being simulated by a reverb program, expressed in percent. It is a fairly complex parameter, affecting several different aspects of the reverb simultaneously. Put simply, for a given RT60, a higher Size value decreases the density of the reflections within the reverb, while a lower value increases the density. The maximum and minimum RT60s that can be set for a particular preset will depend on the value of the Size parameter in that preset.

Stopped Reverb is reverberation heard in a gated reverb program after the source signal level has dropped below the Trig level, and after the StpDly interval has passed. See **Gated Reverb and Running Reverb**.

StpDly in a gated reverb program, is the amount of time after the source signal drops below the Trig threshold level and before the Stopped Reverb takes over. It is adjustable from almost instantaneous to several seconds. See **Gated Reverb**.

Trig is a "threshold" setting, used in gated reverb programs. The Trig level, adjustable over a wide range, is the level of input signal which will "kick" the program into its Running Reverb state. When the input level drops below the Trig level, the program can go into its Stopped Reverb state. When the Trig level is set to OFF, the program will remain in the Running Reverb state at all times. (See also **Play Trig** and **Sample Trig**.)

— Chapter 3 — Upgrading and Servicing

Installing New Software Versions

One of the ADR 68K's most important features is its upgradability. As new algorithms and configurations of the system are developed, they will be coded into EPROMs and sent to every ADR 68K owner for field installation. This way, every owner can be assured that he or she has the latest operating version.

Note: Every time you are issued a software upgrade, you will also be given a set of instructions as to how to install the new software. This is because the procedure may change as the ADR 68K evolves as a product. For convenience, the procedure as of the initial release of the ADR 68K is detailed here, but you should check the instructions that come with your upgrade before going ahead.

The installation procedure can be handled by any competent technician, but it still requires care. The terms of the warranty state that AKG will not be responsible for repairing any damage done by improper installation of software EPROMs.

Before you start, verify that the upgrade kit you have is complete, with all EPROM's, new sections of this Owner's Manual, and instructions. The instructions will probably call for you to have the following tools on hand:

1. Philips screwdriver, size #1.
2. Small regular-bladed screwdriver, blade width about 1/8" to 3/16" (3-4.5mm).
3. IC extractor tool.
4. Pair of needle-nose pliers.

Read the instructions that come with the new software to see if you will have to perform a System Reset or Cold Reset after replacing the EPROM's. You may also have to make copies of your presets in a data cartridge to be sure that none are lost during this procedure.

WARNING! *There are dangerous voltages present inside the ADR 68K which present a hazard when operating the unit with either cover removed. Be sure to restore the covers before operating the unit as you perform these tests and inspection procedures.*

To install the new software, proceed as follows:

1. Unplug the AC line cord from the rear of the ADR 68K.

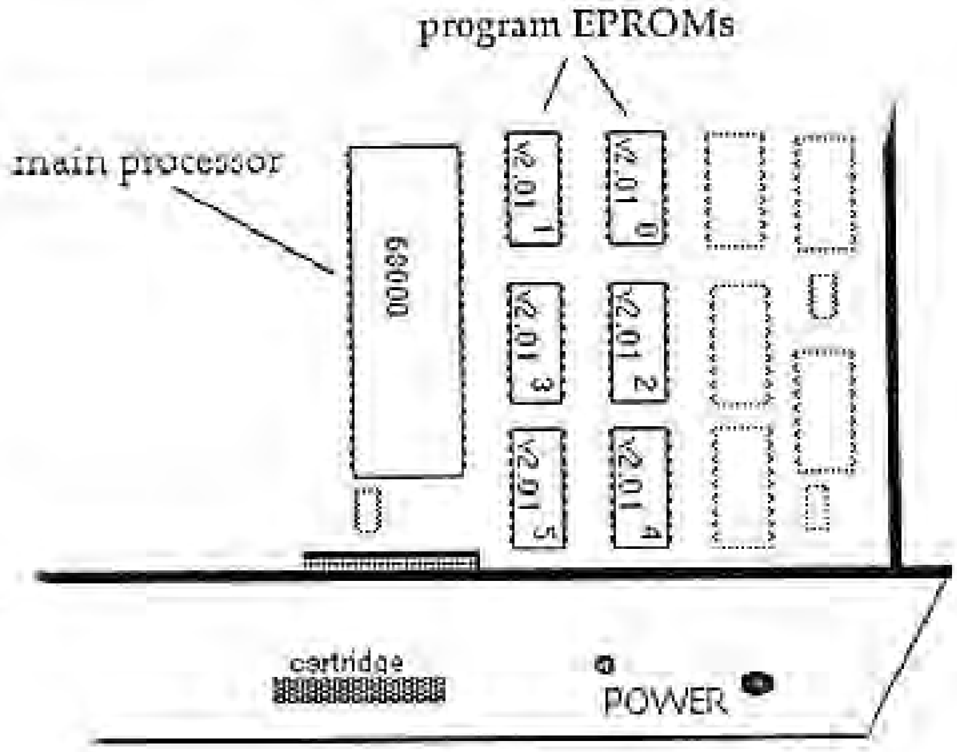
2. Remove the 11 screws holding the top cover in place.

3. Set the cover and screws safely aside.

4. Place the unit on a bench covered with a non-scratch antistatic mat. Orient the ADR 68K right side up, with its front panel toward your stomach.

5. Locate the 68000 microprocessor area. It's toward the front, with a 64-pin IC and six nearby sockets for 28-pin EPROM ICs. Four or more of these sockets will already have software EPROMs in them, each with a label such as "2.01" and "0" on it.

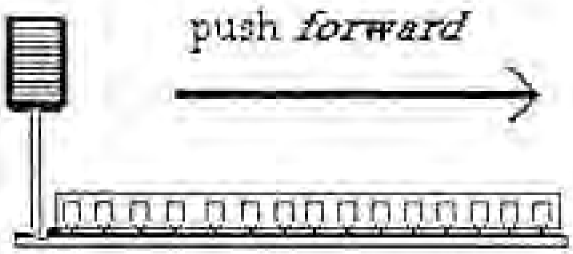
6. Note that each of these EPROM's is installed with its pin 1 end (the end with the notch or dot) toward the rear of the ADR 68K. You will need to install the new parts with the same orientation.



7. Remove the EPROM's using one of the following methods:

a) **If your mainframe serial number is 1-75,** the EPROMs will be in special sockets with a small tab or lever at the pin 1 end. Place the tip of the small screwdriver down in the slot just underneath this tab and use it gently as a lever to push the tab *in the direction of the front panel* until it clicks softly. *Do not use the screwdriver as a lever to pry upward on the tab or the IC — this will break the socket!* Do not force the tab, as excessive force can break it.

You should now be able to remove the EPROM easily, taking care not to bend the leads as you do so. (If the little tab breaks, get your IC extractor out and go to the next paragraph.)



b) **If your mainframe serial number is 76 or higher,** the EPROMs will be in conventional sockets. Place the IC extractor jaws under the pin 1 and opposite ends and lift the IC straight up and out of the socket. Use care to pull evenly so that the EPROM pins are all removed at once.

c) Set the EPROMs aside on a piece of conductive foam.

8. Now you can install the new EPROMs. The rest of these instructions apply no matter regardless of your mainframe serial number. Take the "0" EPROM from its foam mat and inspect the pins. They should all be perpendicular to the body, not splayed outward, and in an even row on each side. If not, gently straighten them. If they're all splayed outward, you can bend them all at once by placing the EPROM on a table surface and gently pressing. If a few are uneven, use the needle nose pliers to straighten them.

Identify the IC end with pin 1, and locate the correct socket for this "0" EPROM. Carefully position the IC with the pin 1 end toward the rear of the unit. Be very careful about the pin 1 orientation. *If you put the IC in backwards it will be destroyed when you power up your ADR 68K!*

Mesh the pins of the IC in the 28 pin socket, jiggling, squirming, looking very carefully to be sure that *every* pin is lined up and already part way in its hole of the socket. If you have a pin misaligned, so that it is aimed at a solid plastic area of the socket, it will be bent under or out when you push the EPROM in and the unit will not work.

Now press down on the entire top surface of the EPROM, gently at first, then a little bit more firmly if necessary, to seat the IC in the socket. There will be a reassuring "scrinch" sound as the IC is pressed home. Inspect carefully *all* 28 pins to be sure they have gone into the socket, not been bent under or outward.

Repeat this for each of the remaining EPROMs, taking care to put them in the correct sockets — the ADR 68K will not work if you scramble the EPROM locations.

9. Reinstall the top cover and plug the power cord back in. Be sure the remote is properly plugged in at each end.

10. Turn on the power and follow any special instructions that come with the new software for resetting the system.

11. Replace the appropriate sections of this manual with the new versions that came with your EPROMs. That's it!

Software-Related Problems

Our software engineers have made every effort to ensure that there are no software "bugs" in your ADR 68K. However, no software should ever be confidently described as bug-free. Software engineers are known to cautiously intone the statement "As of the present time, there are no known bugs." That is as close to a guarantee as one can hope.

The point is that it is not inconceivable that your ADR 68K might someday "crash," with strange display, blank screen, unresponsive remote, etc. If this happens, you will most likely be able to restore things to normal by using procedures described in the chapter on System Software, in the sections entitled System Functions, Reset and Test (SYS), and Remote Self Test and Cold Reset.

Hardware Problems

The ADR 68K is a complex, modern electronic system with two microprocessors and a dedicated high-speed digital signal processor. Although each unit is carefully tested and burned in at the factory, failures may occur. Because the failure modes are many and range from the simple to the complex, and because the availability of test equipment, service information, and trained technicians will vary greatly from location to location, it is impossible in these pages to present a thorough service manual.

Therefore, whenever possible, we recommend returning your unit to the local dealer, who will assist in obtaining the repair.

However, ADR 68K owners with conventional test equipment, a competent technician, and this manual will be able to make some tests and checks to see if there is a readily identifiable fault that can be repaired on the spot. We intend this section to give enough information to help with repairs *only* at this level.

For owners with good digital test equipment and a well-trained technician, who may wish to perform more complex repairs themselves, a copy of the separate Service Manual is available from AKG. The Service Manual contains full schematics, theory of operation, parts list, and scope photos of key waveforms.

NOTE: Please make sure you know what you're doing before you try to work on your ADR 68K. Under the terms of the warranty, AKG will not be responsible for any damage caused by an unauthorized person or facility attempting to service the unit.

User Servicing

Fuses and AC Supply

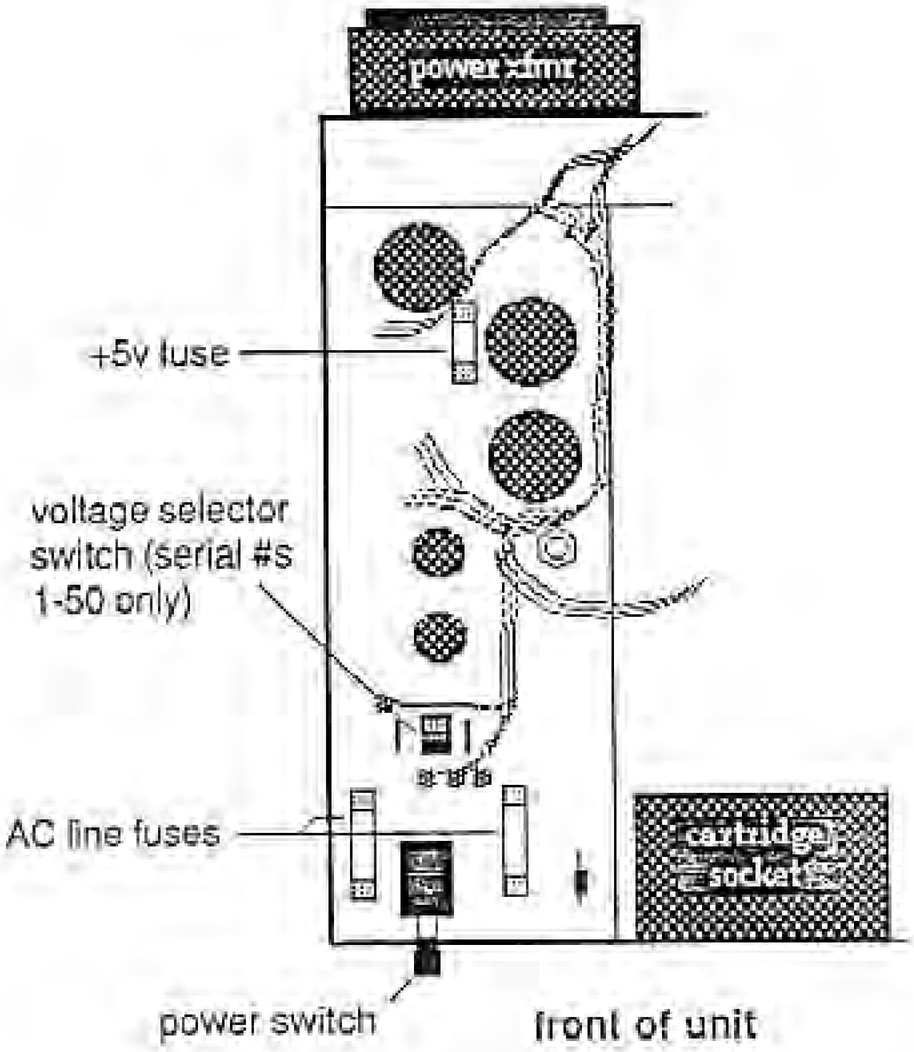
If the unit is totally dead — no lights, doesn't get warm, and emits absolutely no sound (not even background noise) — there is probably no AC getting to the power transformer. Check the power cord at both ends for good contacts, and try another one if possible. Verify the electrical outlet by operating another piece of equipment from it.

Next, check the two line fuses, at the front edge of the PWR-1 PC board, near the power switch). If the fuses appear to be OK, it is still a good idea to check them for continuity with an ohmmeter. If one or both of the fuses are open, replace it with a fuse of exactly the same type and value, replace the bottom cover, and try again.

<u>*WARNING!*</u>

THERE ARE DANGEROUS VOLTAGES PRESENT INSIDE THE ADR 68K WHICH PRESENT A HAZARD WHEN OPERATING THE UNIT WITH EITHER COVER REMOVED. BE SURE TO RESTORE THE COVERS BEFORE OPERATING THE UNIT AS YOU PERFORM THESE TESTS AND INSPECTION PROCEDURES.

If the power cord, outlet, and fuses check out OK, remove the power cord from its connector at the rear of the ADR 68K. Remove the bottom cover of the ADR 68K (four screws one each side and two on the back) and locate the AC line line voltage selector switch. In earlier units, this is a slide switch with positions 115 and 230, while in later units it is a rotary switch, with positions labeled 100, 120, 220, and 240. Be sure the setting corresponds to the voltage in your country (if your line voltage falls between the values labeled on the switch, use the next highest voltage setting). Even if the setting appears to be OK, move the switch back and forth in case it has moved and is sitting in between settings.



Power Supply

If the ADR 68K seems warm, and there is residual circuit noise in the output, but no audio, check the fuse for the +5VDC supply. Remove the power cord from its connector at the rear of the ADR 68K, and then remove the bottom cover. Locate this fuse at the rear of the PWR-1 PC board, near 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 bottom cover, and try the ADR 68K again.

IMPORTANT! If you replace a fuse, you must use exactly the same type and value in order not to create a safety hazard. Replacing a fuse with a short sircuit may permanently damage the unit and/or cause a fire!

Verifying Power Supplies

If a fuse is not the problem, you can follow the steps below, using a scope or voltmeter, to measure the three power supply voltages, and also to verify the power wiring. The internal wiring of the ADR 68K consists of wiring harnesses that connect the power transformer and AC connector to the PWR-1 board, as well as wiring carrying the regulated voltages from the PWR-1 board to the other circuit boards. The wire colors and connector pin-out are given on the schematic for the PWR-1 board at the end of this section. Since there is so little wiring in the ADR 68K, it's unlikely that wiring will cause a problem. Note: the PWR-1 and ANA-3 boards are accessible by removing the bottom cover, while to get to the DSP-1 you must remove the top cover.

1. Verify +5 volts by checking between pad 18 and 15 on the PWR-1 PC board. The tolerance is 4.75 to 5.25 volts.

2. Verify +15 volts by probing the wires of J12 on the ANA-3 PC board. The tolerance is 14.25 to 15.75 volts.

3. Verify -15 volts by probing the wires of J12 on the ANA-3 PC board. The tolerance is 14.25 to 15.75 volts.

4. Verify +5 volts on the ANA-3 PC board, measuring at U100, where ground is at pin 10 and +5 is at pin 20.

5. Verify +5 volts on the ANA-3 PC board, measuring at the test point near U120.

6. Remove the top cover, and verify +5V at a typical 20 pin IC on the DSP-1 PC board, such as UK1, where ground is at pin 10 and +5 is at pin 20.

7. Verify +15 on the DSP-1 board, at UA10-14.

8. Verify -15 on the DSP-1 board, at UA10-1.

9. Verify +9 volts on the DSP-1 board, at J9 pin 3.

Remote Unit

If the display and/or controls on the remote don't work normally, it's crucial to establish whether the remote or the mainframe is at fault. Ideally, if another ADR 68K is nearby, you can try a known good remote with your mainframe and isolate the failure to either your remote or your mainframe. The remote also has its own internal self test mode, which can be used even if the mainframe has serious problems (all the mainframe must provide in order to do this test is the +9 volt and digital ground signals). This procedure is described in the chapter on System Software, and is mentioned here briefly for convenience:

Turn the power off. Wait 5 seconds.
Press and hold the Help button on the remote.
Turn the power on, still holding the Help button.
Select Remote Self Test from the menu.
Try each button and slider and check the response on the display.
It is normal for the display to read "COMM:Fail."

We *do not* recommend disassmembly of the remote due to the delicate nature of the flexible circuit inside.

General Troubleshooting Hints

It's always a good idea to look a malfunctioning unit over carefully before sending it in for service. Look for the following:

— Are all the connectors pushed all the way home? Tug gently on all the power wires where they enter connectors to see that they're secure.

— Are the remote cable connectors pushed fully in?

— If someone else has already tried to fix the unit, check that they've not removed any IC's, or failed to push them fully into the socket, or returned them to the wrong sockets, or inserted them upside down, or with a pin bent under or outside the socket.

— Are there signs of severe mechanical stress, such as a broken PC board, or bent sheet metal?

— If liquid has spilled in the machine, try to clean the boards and sockets to remove the residue.

— Be sure the audio connections to and from the ADR 68K are good. Try plugging the input connector of your cable set into the output connector to verify the cables and associated equipment.

— Review the audio cable wiring, and refer to the figures in the Owner's Manual main section to be sure the cables are wired correctly.

— Check that no pieces of metal or hardware have fallen into the mainframe or remote. Shake them and listen for rattles.

Servicing by AKG

In the USA

In the US, refer service questions first to your dealer. If he cannot help you, contact Service Department, Digital Products Division, AKG Acoustics, 2 Calvin Rd., Watertown, MA 02172.

Outside the USA

In countries other than the United States, refer service questions first to your local Service Center, as listed below. If he cannot help, please contact the Service Manager, AKG Acoustics, Schanzstrasse 20, A-1150 Wien, Austria.

ARGENTINA: Magneto Sonora SRL, Corrientes 316-50-569, Buenos Aires 1314.

AUSTRALIA: AWA Amalgamated Wireless (Australasia) Ltd. Ashfield Div., PO Box 24, Ashfield NSW, 2131

BELGIUM: Inelco Belgium SA, Avenue des Croix de Guerre, 94, B-1120 Brussels

BOLIVIA: Fernando d'Achiardi-Schohaus, Casilla 274, La Paz

BRAZIL: Importechnica SA, PO Box 6134, Av. Dr. Abrahao Ribeiro 740, CEP 01133, Sao Paulo

CANADA: Philips Electronic, Ltd., Consumer Service, 601 Milner Ave., Scarborough, Ontario M1B 1M8

CHILE: Importadora Prov. Ltda., Casilla 91, Correo 10, Las Condes/Santiago de Chile

COSTA RICA: Electronic Corporation, Apartado 10304, San Jose

CYPRUS: Music-Sound A.G. Tillirides, POB 291, Limassol

DENMARK: SC Sound APS, Malervej 2, DK-2630 Taastrup

EL SALVADOR: Radio Parts SA, PO Box 1262, 4A Av. Sur 125, San Salvador

FINLAND: Nores OY, PO Box 889, SF-101 Helsinki 10

FRANCE: Harman France S.A., 33 Ave. du Marechal de Lattre de Tassigny, F-94 120 Fontenay-sous-Bois

GERMANY (WEST): Akustiche u. Kino-Gerate GmbH, Bodenseestrasse 226-230, D-8000, Munchen 60

GREECE: Asteriadis Nicolaos, 54 Tsimiski St, GR-546-23, Thessaloniki T.T.18

GUATEMALA: Almacen Magno Mercantil SA, 9a, Ave. 8-26 Zona 1, Guatemala City

HOLLAND: Audioscript B.V., Postbus 82, NL-1230 Loosdrecht

HONG KONG: The Radio People Ltd., 7/F News Building, 635 Kings Road

INDIA: Peico Electron. & Electric Ltd, PO Box 6598 Bombay 400 018

ITALY: M. Casale Bauer S.p.A., Via IV Novembre N. 6-8, I-40057 Cadriano di Granarolo, Bologna

JAMAICA: Stanley Motta Ltd., POB 341, Kingston

JAPAN: AKG of Japan Service Co, Ltd., Room 401-3, Daiichi-Kanon Bldg., 2-12-1 Kita-Otsuka, Toshima-ku, Tokyo

JORDAN: Zaki M. Nassar & Brother, POB 808, Amman, 01 Street 85

KOREA: Young Nak Trading Co., 182, Jansa-Dong, Jongro-ku, Seoul

KUWAIT: Jupiter Trading Company, PO Box 24566, Safat

KENYA: Aeradio (K) Ltd., Rehema House, Standard St. PO Box 46215/NAIROBI

LEBANON: Projects S.A.L., PO Box 11-5281, Beirut

LUXEMBOURG: Beaulieu Luxembourg S.A.R.L., Boite Postale 29, 2, Rue de Luxembourg, Petange

MALAYSIA, WEST: Asia Sound Equipment (M) PTE LTD, 80 Jalan Pasar Off Pudu, 55100 Kuala Lampur

MEXICO: Casa Veerkamp S.A., Mesones 21, Box M-851, Mexico 1 D.F.

NEW ZEALAND: AWA New Zealand Ltd., PO Box 50-248, Porirua

NIGERIA: David Hughes & Co. Ltd., P.O. Box 4007, Ikeja

NORWAY: J.M. Foiring Elektronikk A/S, Nils Hansens Vei 3/7, POB 101-Bryn, Oslo 6

PANAMA, REPUBLIC OF: Tropelco, Apartado 8466, Panama 7

PAKISTAN: Pakwestrex Co. Ltd., Rm 55, 4th Floor, Ghafoor Chambers, Abdullah Haroon Rd., Karachi

PARAGUAY: Electronicasa S.R.L., Casilla de Correo 66, Asuncion

PERU: Telewatt S.A., R. Rivera Navarrete No. 797, San Isidoro/Lima 27

PHILIPPINES: Avcom Electronics Inc., Matrinco Bldg., 2178 Pasong, Tamo, Makati, Metro Manilla

PORTUGAL: Philips Portuguesa SARL,Apartado 55, 2795 Linda-A-Velha, Lisboa

SAUDI ARABIA: Said Bin Saad Trading Est., PO Box 2318, Riyadh 11451

SINGAPORE, REPUBLIC OF: Asia Sound Equipment (S) PTE LTD, 100 Beach Road No. 02-13/14, Shaw Tower, Singapore 0718

SOUTH AFRICA: Studer Revox South Africa (PTY) Ltd., Downing House, 70 Central Avenue, Mayfair, Johannesburg 2092 - PO Box 31282, Braamfontein, 2017

SPAIN: Neotecnica SAE, Marques de Urquijo 44, 28008 Madrid 8

SRI LANKA: The HiFi Centre Ltd, 514, R. A. De Mel Mawatha (Duplication Road), Colombo 3

SWEDEN: Allba Ljud, Gelbgutarevagen 4, S-171 48 Solna

SWITZERLAND: Audio Systems PAS AG, Munchensteinerstr. 270, CH-4053, Basel

SYRIA: Technical Trading & Industrial Center, PO Box 3515, 57 Jamhouria Str., Damascus

TAHITI: Oceanic Garage, PO Box 39, Papeete

TAIWAN, REPUBLIC OF CHINA: Yang's Audio-Visual Labratories, 4th Floor, 58 Min-Chuan West Road, Taipei 104

THAILAND: Vichai Trading Co. R.O.P., POB 1512, Bangkok 10100

TUNISIA: Soger Electronique, 32 Rue Garibaldi, TN-1001 Tunis

UNITED ARAB EMIRATES: Eros Electricals, PO Box 1184, Dubai

UNITED KINGDOM: AKG Acoustics Ltd., Vienna Court, Catteshall Wharf, Catteshall Lane, Godalming, Surrey GU7 1JG

URUGUAY: Swissaudio, PO Box 584, Montevideo

VENEZUELA: ABS Assaf Breidy & Sons, CA, Boulevard de Sabana Grane, Apartado 60 228/ Caracas 1060

ZIMBABWE: Radio Frequency Communications (PVT) Ltd., PO Box 8357, Causeway, Zimbabwe.

Shipping Instructions

Should your ADR 68K 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 always present always or only intermittent; whether the unit was OK originally and then developed the problem, or whether it was always defective. If the problem is intermittent, tell us if it seems to depend upon mechanical shock and vibration, or if it seems to come and go with time or temperature changes. If possible, include a cassette recording of the unit's sound when exhibiting the failure.

2. When and where you purchased the unit and its serial number. Include a copy of your sales slip if possible.

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.

5. When you need the unit back, and how you want it shipped.

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 in your own materials, be sure the ADR 68K is completely surrounded by at least 3 inches of padding on all surfaces, and that the box is strong. Be sure to protect the mainframe front panel mounting ears, as they are especially vulnerable.

Don't return your power cord or manuals to us with your ADR 68K. If you and/or your dealer have determined positively that the fault is in either the remote or the mainframe (by testing the remote with another mainframe, or the mainframe with another remote), but not both, then it is only necessary to return the defective unit.

We recommend that you insure your ADR 68K shipment to us, because it is otherwise completely uninsured against damage or loss, and we will not be responsible for any damages incurred in shipping. You must bear the cost of shipping to us even if the unit is still under warranty.

If the unit is still under warranty, we will return it shipping prepaid after we repair it. On out-of-warranty repairs in the United States, and some other countries, the unit will be returned COD, for both shipping and service charges.

As of December, 1986, the shipping address to use is:

Digital Products Division
AKG Acoustics, Inc.
28 Calvin Rd.
Watertown MA 02172
USA

If you are returning a unit a long time after this date, it would be wise to check with us to be sure that the address above is still current. Our mailing and telephone addresses are:

Digital Products Division
AKG Acoustics, Inc.
2 Calvin Rd.
Watertown, MA 02172 USA
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Warranty Information — USA

AKG Acoustics, Digital Products Division, warrants AKG products against defects in material or workmanship for a period of one year from the date of original purchase for use, and agrees to repair or, at our option, replace any defective unit without charge for either parts or labor.

IMPORTANT: This warranty does not cover damage resulting from accident, misuse or abuse, lack or reasonable care, the affixing of any attachment not provided with the product, loss of parts, or connecting the product to any but the specified receptacles. This warranty is void unless service or repairs are performed at an authorized service center.

No responsibility is assumed for any special, incidental or consequential damages. However, the limitation of any right or remedy shall not be effective where such is prohibited by law.

Simply take or ship your AKG product prepaid to our service department. Be sure to include your sales slip as proof of purchase date, and be sure to insure it. (We will not repair transit damage under the no-charge terms of this warranty).

This warranty gives you specific legal rights, and you may also have other rights which vary from state to state. Some states do not allow the exclusion or limitations of incidental or consequential damages or limitations on how long an implied warranty lasts, so the above exclusion and limitations may not apply to you.

Products are sold on the basis of specifications applicable at the time of manufacture. AKG Acoustics, 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.

Warranty Conditions — Countries other than the United States.

AKG warrants AKG products against evident defects in material and workmanship for a period of one year from the date of original purchase for use. This warranty does not cover damage resulting from misuse or abuse, or lack of reasonable care, and inadequate repairs performed by unauthorized service centers.

Performance of repairs or replacements under this warranty is subject to submission of the Warranty Form at the front of this manual, completed and signed by the dealer on the day of purchase, and the sales slip. Shipment of the defective item for repair under this warranty will be at the customer's own expense. This warranty is valid for the original purchaser only.

Theory of Operation

For those of you interested in how the ADR 68K works on a more basic level, we have provided some of the theory behind its design. Schematic diagrams for the following sections appear at the end of this chapter. Schematics of other sections of the ADR 68K can be found in the Service Manual.

Power Supply

The PWR-1 PC board holds three voltage-regulator circuits. U300 and U301 are fixed 15V regulators for the + and -15V supplies. U3 is the +5V regulator, and is located on the rear panel heat sink. U3 is followed by a "crowbar" circuit which detects an overvoltage condition and, when found, clamps the +5V output line to digital ground. This blows the fuse F302, thus protecting the TTL IC's and other logic parts from damage. C300 and C301 are actually MOV varistors that suppress fast, short line-voltage transients. This board comes in two versions: serial numbers 1-50, and serial numbers above 50.

Analog Circuits

The ANA-3 board handles all the analog input/output and analog-digital-analog conversion functions for the ADR-68K. It processes two input channels, converting them to 16-bit PCM signals at a 32kHz sampling rate, and delivers four corresponding outputs after digital processing.

Sheet 1

The XLR input jacks are received by an electronic, differential, transformerless op amp input stage, with low-pass RC filters for RF pickup suppression. U130, a programmable gain stage, follows, providing four input sensitivities, from -10 to +18dBV. Next, U140 provides aperture correction, which is a slight high-frequency boost to correct for hf rolloff of the output sample-and-holds. Hybrid 15kHz low-pass anti-aliasing filters follow, leading via an analog switch into U138, a buffer and voltage gain stage. (An alternate path around the aperture-correction stage and 15kHz low pass filter can be selected by the ADR 68K operating software when full 20kHz bandwidth is desired).

U124, a digital white noise generator IC, feeds a high-Q bandpass filter, shaping the noise to a very narrow band signal centered at 16kHz (alternately, at 21kHz under software control). This is greatly attenuated and applied to U138 to serve as a dither signal for the analog-to-digital converter.

Sheet 2

The four sample-and-hold outputs serviced by the DAC are filtered by M100-103 to eliminate the sampling rate and signal sidebands. The filter outputs feed an analog switch and gain stage (alternately, the filters can be switched out under software control for 20kHz performance). Finally, the four XLR output jacks are fed from active differential, transformerless op amp output stages.

Sheet 3

Incoming signals from Sheet 1 Lin and Rin are fed to sample-and-holds and a buffer stage before reaching U134. U134, an analog switch, is a multiplexer that selects the left or right inputs alternately for analog-to-digital conversion. The U134 output enters the DAC which performs a 16-bit successive approximation conversion, with the aid of clamp U129, comparator U128, and comparator front-end gain stage Q101. The clamp shorts the conversion node briefly after each bit test to speed up conversion, while Q101 makes up extra gain needed by the comparator.

The comparator output signal CMP guides the 16-bit successive approximation registers U127 and U132 through the tests, and U126 and U131 buffer the conversion result from the ADC bus sent to the digital signal processor. A clamp offset control, RV100, is set for minimum noise and distortion, while the ADC offset control, RV101, is set to slightly off center scale to control idle channel noise.

Sheet 4

The 16-bit data bus sent from the digital signal processor, DAC0-15, is buffered by U100 and U109 and applied to the 16-bit DAC. This is a current-output DAC, which is buffered by U102 and applied to four sample-and-hold circuits which also act as demultiplexers, separating the four output channels.

Sheet 5

Three four-bit, complementary-output D flip-flop registers receive 12 timing and control signals from the DSP-1 board and deglitch and regenerate them with clean waveforms, for use on the ANA-3. They are clocked by the signal TIM CLK2 at 8MHz. U120 develops a local, clean +5V supply for use on this board, and other decoupling circuits decouple the 15V power supplies.

rev 4.0, 15 December 1987

— Chapter 4 — The ADR 68K System Software

Introduction

This chapter will deal with basic operation of the ADR 68K under the current software. It will describe the structure of the software, show you how to use many of the controls, and other general operational topics.

Specific information about the reverb and effects programs is contained in the next chapter, while using the samplers is covered in Chapter 6, and controlling the unit with MIDI is covered in Chapter 7.

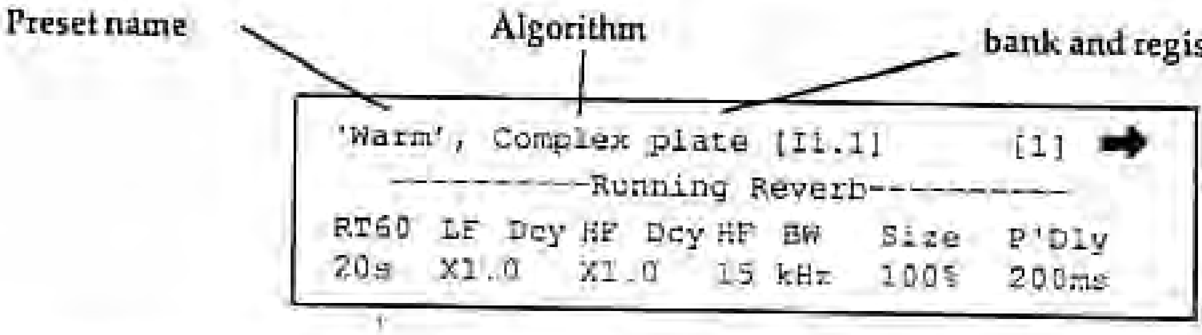
Remember that the ADR 68K's operating system is entirely in software. This means it is upgradable, and in fact will change at even the most basic levels as the product evolves and new ways of using it are developed. As these changes take place, you will be supplied with new versions of this chapter of the manual, as well as those other chapters and appendices that are affected.

Algorithms, Presets, Banks, and Registers

An ***algorithm***, in general parlance, is a rule that is followed to accomplish some function. In the ADR 68K, the definition of an algorithm is a little more involved than that. First, it is indeed a rule, written in microcode and programmed into the system EPROM, for dealing with audio signals and other information coming into the unit.

More than that, however, an algorithm in the ADR 68K also includes a set of control functions, including the layout of the controls and the pages, the rules for interpreting and displaying parameters, the organization of memory and function within the system, and the various ways that the system will interact with itself and the outside world. For example, an algorithm will not specify a specific control setting, e.g., decay time, but instead it will specify a *range* of settings, and will define how the user can adjust the setting. The algorithms are unchangable by the user — they are burned into the system EPROM and can only be altered by replacing the EPROM.

A *preset* is defined as a specific way of using an algorithm. It is comprised of an algorithm and set of specific control settings within that algorithm, including operating parameters and perhaps some system settings. When a preset is called up on the ADR 68K, its name and the name of the algorithm associated with it appear on the display together. For example, the factory preset "Warm" uses the algorithm "Complex Plate", and the two names appear on the LCD when the preset is called.



A preset is also sometimes referred to as a "program". Presets are completely definable by the user. While the ADR 68K's software contains about a dozen algorithms, a unit can hold 200 or more presets.

Banks are a way of organizing the presets. The ADR 68K has 11 banks. Generally speaking, presets within a bank share the same overall functions and sound-processing characteristics, which usually means they also use the same algorithm. The banks are organized as follows:

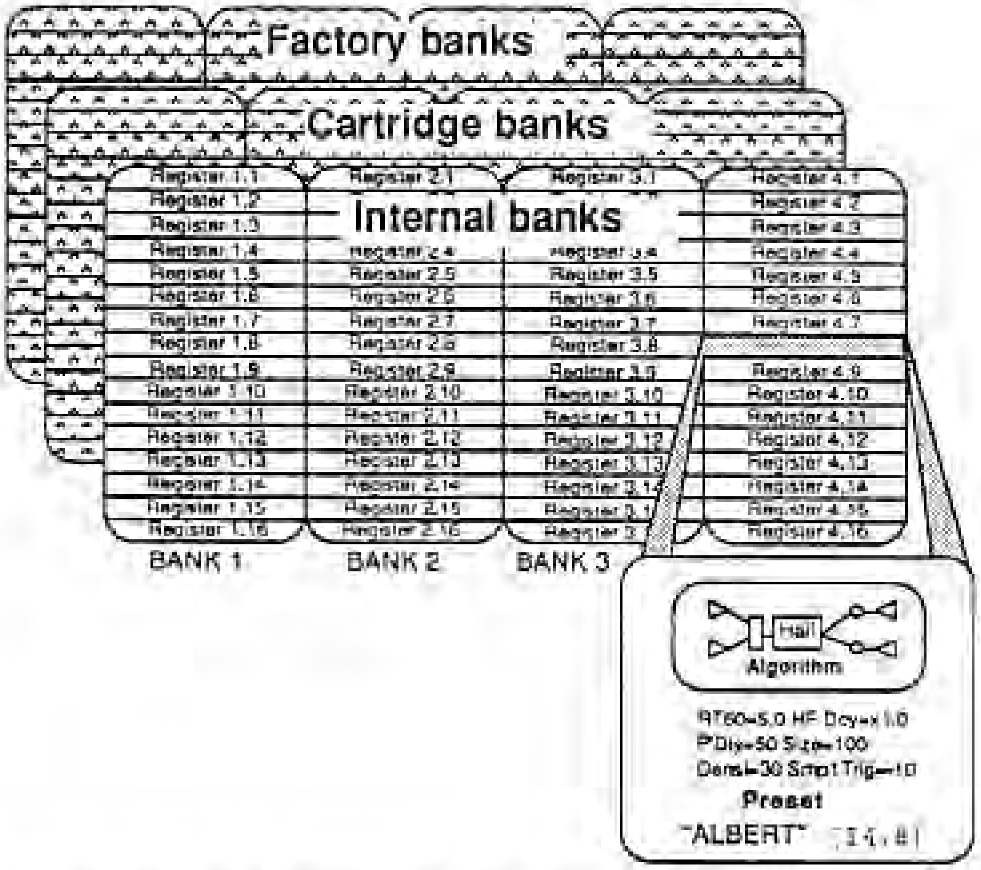
Bank #	Bank Name	Sound Processing Character
1	Plate	reverberation
2	Chamber	reverberation
3	Room	reverberation
4	Hall	reverberation
5	Splits	two independent processors
6	Sample	samplers, including those used in other banks
7	Revers	reverse reverb
8	Multi	multi-effects
9	Stereo	stereo processing of mono inputs
10	Chorus	chorusing
11	DDL	dual digital-delay lines

Each bank contains 50 *registers*, or individual preset locations. A preset is loaded, recalled, and edited by addressing its register location.

Registers exist in three memories: Factory, Internal, and Cartridge. A register number, when referred to in this manual or on the ADR 68K's display screen, shows which memory the register is in, which bank it is a member of, and its number within that bank. For example:

"F6.7" means: Factory, Sampler Bank (6), register #7
"I3.11" means: Internal, Room Bank (3), register # 11
"C5.3" means: Cartridge, Split Bank (5), register # 3

The following diagram shows the relationship between algorithms, presets, registers, banks, and memories.



AKG provides a variety of presets at the time of shipment. These are stored in system EPROM, where they will remain intact and unalterable. You can recall the factory presets and edit them, and if you want to store the edited versions, you must do so in one of the Internal or Cartridge registers — you can't write to a Factory register.

Each preset has a name as well as a register number. Unlike the algorithm names, preset names can be altered by the user. Names may be up to six characters long, and are assigned when storing a preset. Names of factory presets are in mixed upper- and lower-case letters (e.g. "Warm"), while user-named presets will be all upper-case ("WARM2"). *Note:* It is possible to take a factory preset, alter it without changing its name, and then store it under the original name in a user register, but to avoid confusion this practice is not recommended.

Since most or all of the presets within any one of the ADR 68K's factory memory banks reflect different uses of the same algorithm, it follows that any preset within a specific factory bank can be reconfigured by movement of the sliders to any other preset within that bank (banks 5 and 6 being exceptions to this). It is strongly suggested that, except under special conditions which will be discussed later, user presets be similarly organized, with all of the presets in a bank using the same algorithm.

Note: Although each user bank can theoretically hold 50 presets, this will almost certainly never happen. That's because the memory capacity of the ADR 68K is limited: a *total* of 50 presets can be stored in the internal memory and another 50 in the cartridge.

Note: It's worth mentioning at this point that what this manual refers to as an "algorithm," is what in more traditional digital reverb and effects units is called a "program". However, in MIDI applications the word "program" is used to mean a set of control positions (a "preset" or "patch"). Because the ADR 68K is a MIDI-compatible device, we have decided to adopt the latter meaning of "program" in our documentation. However, in order to avoid confusion, we will use the word sparingly, and instead refer to sets of control settings as "presets" and the structures within which presets are built as "algorithms".

Displaying the Presets in a Bank

To display a menu of the presets in any of banks 1 through 5, press and hold the Shift button and press the appropriate bank button (e.g., "Shift-Plate"). You will see a display like this:

```
Plate    1,1   1,2   1.3   1.4   1.5  ➡⬆
 CART    —     —     —     —     —
  INT    —     —     —     —     —
 FACT➡  Warm  Snare1Snare2Kick  HiHat
```

The second, third, and fourth lines show the names assigned to the first 5 registers of the cartridge, internal, and factory memories. To see register #'s higher than 5, scroll the display window rightward using the "→" button. To scroll back, use the "←" button. You can jump to the first menu page from anywhere in the menu by pressing Shift-"←", and you can jump to the last menu page by pressing Shift-"→". To look at the menu for another bank, hold the shift button and press the button corresponding to the new bank.

Banks 6 through 11 are accessed somewhat differently. First press "Shift-EFX", and a list of the banks will appear in the display:

```
  Select effects                  ⬆
  bank:
Bank6 Bank7 Bank8Bank9 Bank10 Bank11
SampleReversMultiStereo Chorus  DDL
```

Press the Soft Button (with the triangle inside it) underneath the bank that you wish to look at, and its menu will appear. To look at the menu for another bank in the group, press "↑" and the above display will reappear.

Calling up a Preset

To call up a preset for use, first find its menu page by pressing Shift and the bank name, or Shift-EFX and the appropriate Soft Button. Then choose the memory type (CART, INT, or FACT) where the preset is located by pressing the first Soft Button or moving the first fader so that the indicator arrow at the left is pointing to the correct selection. Now press the Soft Button below the desired register. If you have called a FACTory preset, that preset will be immediately recalled. Its algorithm will be loaded (if necessary), the stored parameter values will be put into operation, and the display will change to show the first page of control parameters for that preset.

If you have chosen an INTernal or CARTridge preset, the display will look like this:

```
Warm, Complex Plate [I1.1]    41 free ↑


Clear  Recall  Save
```

Press the Soft Button below the word "Recall" to call the desired register. (If you don't see "Recall", you've chosen an empty register. Press the "↑" key to get back to the bank menu.)

If this seems like a fairly lengthy procedure just to change a preset, there is a much shorter way of doing it, *if* you know exactly which preset you want to change to. On the numeric keypad, press the bank number, the decimal point, and the register number of the preset you want. The display will change to this:

```
Quick Register Select:  INT 1.10      ↑
Use CE or ↑ to abort, Enter to recall.
(Note: Type decimal point after bank #.)

FACT    INT    CART                  Enter
```

The memory type indicated will be the same as the last memory you called up. If you want to change the memory type, press the appropriate Soft Button.

When you're set, press the "ENT" key on the numeric pad or the Soft Button under the word "Enter". If you make a mistake or change your mind before you press "ENT", press "BSP" and the last character you typed will be erased, or press "CE" or "↑" and the whole thing will be cancelled. (If you press a bank number and "ENT" without pressing a register number, you will call up the first Internal register in that bank. If that Internal register is empty, you will call the first Factory register.)

You can use this "Quick Register Select" function from any menu page, but when you get to the new preset, you will always be on Page 1 of the preset.

When you recall a preset, the settings that you had active before recalling the preset are not lost — instead they are stored in a special register called the Last register. We'll discuss this in a moment.

Even Faster Recall of "Preferred" Presets

No matter what the display is showing, if you push one of the bank buttons along the front edge of the remote, you will recall the preset in the first Internal register of the corresponding bank (e.g., pressing the Chamber button recalls I2.1). If that register is empty, you will recall the first Factory preset of the bank.

The factory presets in the first registers are all good "general-purpose" programs. To take full advantage of this feature, it is recommended that any presets put in the first Internal registers also be good "general-purpose" programs.

Note that this function only applies to the first six banks. Pressing the "EFX" button by itself will call register I (or F) 6.1.

One possible use of this feature in the studio goes contrary to the recommendation of keeping user presets in the banks they originated from. If you have a mix which will be using several different presets (which may or may not use the same algorithm), load them into the ".1" positions of the first six internal banks. That way, you can have immediate access to six different presets, regardless of which algorithm they use.

Changing Presets on the Fly — avoiding glitches

All programmable effects units let you call up new programs from memory almost instantaneously, but with many devices doing so can have very strange results. Metallic glitches, unsavory pops, or gaping holes in the sound are some of the anomalies that switching programs in the middle of a piece of music can produce.

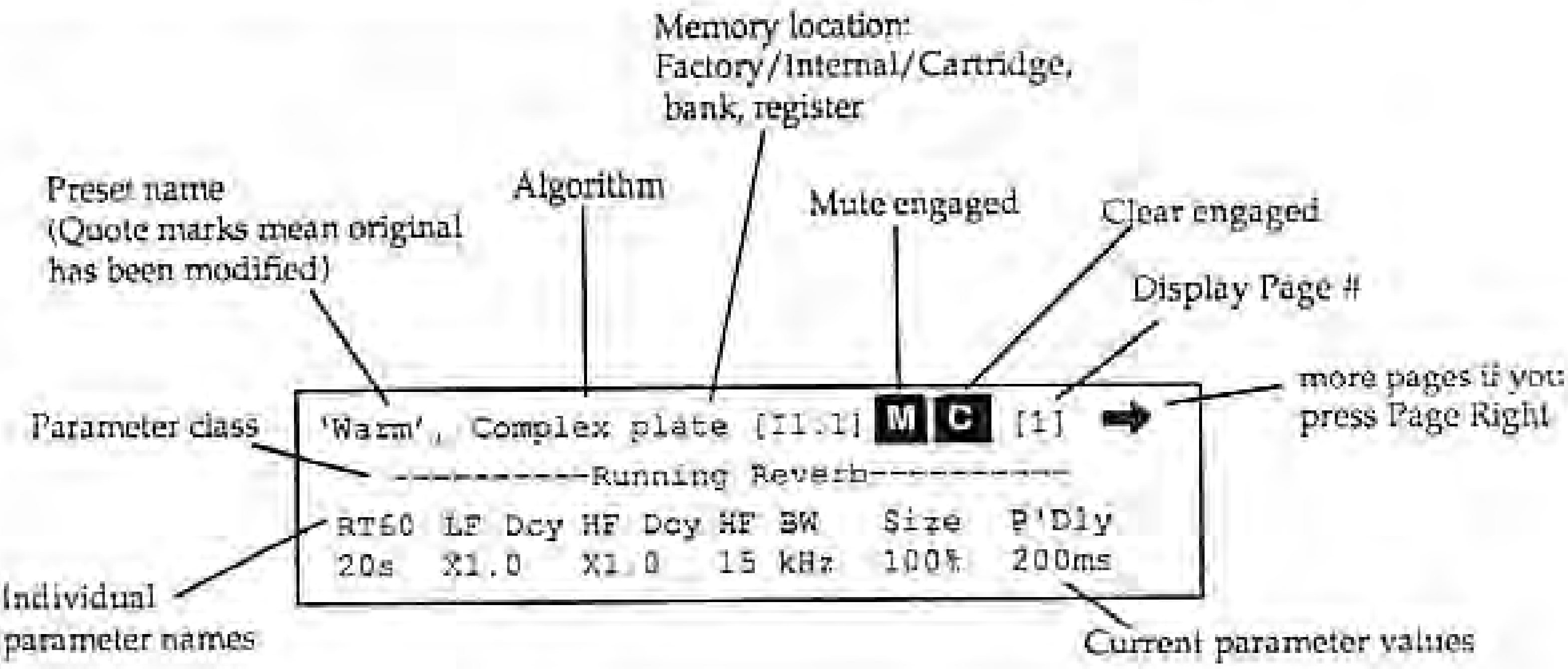
The ADR 68K has been designed to minimize these problems. A great many of the programs in the ADR 68K, whether they are factory- or user-designed, can be switched among rapidly with almost no audible artifacts — the first program decays naturally, and the second starts its work immediately, resulting in a completely smooth overlap. Generally speaking, switching between two programs that use the same algorithm (i.e., that are in the same bank — and being able to keep track of what's what is one good reason to group similar sounds in banks) will give the best results. If you switch between two programs that use different algorithms, there may be a momentary clearing of the internal memory before the second program takes over. However, the ADR 68K fills in the gap that is produced by quickly fading in a measure of dry signal, and then fading it out again as the second program starts to be heard.

The quality of the transition will be the same no matter how the program change is effected: through the bank menus, from the numeric keypad, using the "preferred preset" function, or via MIDI.

Modifying Presets

Scrolling through Pages

All of the presets have a number of user-modifiable parameters. Groups or classes of related parameters are arranged together into display "pages." A preset will have as many as eight pages, each page containing up to six parameters. The parameters have names, shown on the third line of the display, and are set to specific values, shown on the fourth line. To modify a preset, you must first move the display window to the appropriate page, and then locate the fader below the parameter you wish to change. A typical first-page display looks like this:



You can move or "scroll" the display window right and left through the various pages of parameters. A right arrow in the upper right of the display column, next to the page number, indicates the existence of more pages; use the "→" button to scroll the display right to Page 2, Page 3, etc. You can continue scrolling right beyond the highest page and wrap around to Page 1. When you go to Page 2 and beyond, you will see a left arrow appear next to the page number in the display; use the "←" button to scroll back. When you get to the last page used by that preset, the right arrow will disappear.

You can also use Shift-"←" to jump immediately back to Page 1, and Shift-"→" to jump immediately to the last page in a preset. The "←" and "→" buttons have "auto-repeat" — if you press and hold one of them, the screen will keep scrolling in the appropriate direction.

Using Faders

A parameter whose name appears on a page is adjusted by moving the fader control directly under its name. Watch the display and listen to the change in sound that occurs until you've got the setting you want. As soon as you have made a modification in a preset that has been called from memory, the top line of the LCD display is changed, with quotation marks placed around the preset name, e.g., 'Sizzle'.

It's important to know that when you first go to a page (or if you've just recalled a preset), the parameter values you see, which are the values in use, will probably *not* correspond to the present physical position of the faders. The ADR 68K knows this, and does not change any values until you physically take hold of a fader and move it. When you move the fader, the system software automatically and silently changes the value of the parameter associated with that fader to the value corresponding to the new fader position. The initial change may be slight, or it may be gross, depending on the original value and the fader position. Note that the parameter values entered by a fader normally correspond to the *absolute* position of the fader, not by relative movements or "nudges" of the slide control (an exception to this occurs when editing samples, which will be discussed in Chapter 6).

Using Soft Buttons

The Soft Buttons are used with menu and register functions, as described above, and certain system-level functions, which will be described later. They are also used to trigger samples within a reverb or effects program, and to perform certain other tasks within the Sampler presets themselves. See Chapter 6 for a full discussion.

Note that in some contexts, the Soft Buttons have an "auto-repeat" function, which means if you hold one down, the parameter or other item it affects will keep changing. When you are using a Soft Button with the Shift key in auto-repeat mode to decrement a value, you can let go of the Shift key after you have pressed (and begun to hold) the Soft Button, and the value will keep decreasing. The auto-repeat function has an acceleration characteristic built in, so the longer you hold a button down, the faster the numbers will change.

The Current register and the Last register

To help in editing programs, the ADR 68K maintains two "invisible" registers: one for the preset currently running — the "Current" register — and one for the last preset you used before loading in the present one — the "Last register".

When you call up a preset from a register, its contents are copied into the Current register, and from there control the machine operation. When you modify one or more parameters, these changes are remembered in the Current register, but not, of course, copied back into the source register. The quotation marks that appear around a preset name when you adjust a parameter are an indicator that the contents of the Current register are no longer identical to the contents of the source register.

Whenever you recall a new preset, and thereby overwrite what is in the Current register, before the new preset is loaded in, the contents of the Current register are automatically copied into the Last register. This register can then be recalled by pressing the Last button.

When you press the Last button, the ADR 68K *swaps* the contents of the Last and Current registers. You now hear whatever was most recently copied into the Last register, and have a safe copy of what you *were* hearing in the Last register. It's a little tricky to conceptualize, but remember that you are always hearing and seeing the values in the Current register.

The Last register lets you quickly compare different versions of a preset while you are editing it. For example, if you have recalled a preset, say from a register called "WARM," and modified in one or more ways, you can compare the new version with the original by recalling the original "WARM" (using either the bank menu, the Quick Register Select, or — if it is in the .1 "preferred" position — the bank button). You haven't lost your variation — it now resides in the Last register.

To get the variation back, press Last. Now the variation is in the Current register and the original is in the Last register. You can go back and forth between the two indefinitely, and can continue to change the program, constantly comparing it to the original if you wish. If you are worried about losing track of which is which, remember that whenever the modified version is operating, the register name will be in quotes.

The Current register is always remembered when you shut off the ADR 68K, and its settings will come up when you turn the unit back on. The Last register, however, is not remembered — when you power up, the Last register will contain the contents of the Current register.

Storing Presets

Once you have adjusted the parameters of a preset to your liking, you can store it in a register as a new preset. You should store the preset in the appropriate bank (the one the source program came from), but you can choose any register in either the Internal memory or Cartridge memory. You can either choose a new, empty register, or you can use a register that currently has another preset — if, for example, you had a favorite preset which you've just made better, and you don't mind losing the old version.

Remember that when you store a preset, it is always the preset in the Current register that is being saved.

This is how you store a preset in a register other than the last one called:

Press Shift-Menu to get to the bank menu. If necessary, use the first Soft Button or fader to select the correct memory type (Internal or Cartridge). If necessary, use the "→" and "←" buttons to get to the screen with the register you would like to place the preset in. You will now see a display like this:

```
Plate   1.1   1.2   1.3   1.4   1.5   →↑
 CART   —     —     —     —     —
  INT   —     —     —     —     —
 FACT → Warm  Snare1Snare2 Kick  HiHat
```

Press the Soft Button below the target register. If you choose an empty register, you will get this display:

```
 Empty Register [I1.15]      41 Free    ↑


 —     —      Save    —
```

If the register already has a program in it, you will see these options:

```
 Reggae, Complex Plate [I1.14]   41 free ↑


 Clear  Recall  Save
```

Either way, you can now save the current program by pressing Save. If you want to get back to the menu, you can press "↑". (The other options are to Recall the preset whose register you have chosen [if there is one] or Clear [empty out] that register.)

Naming Presets

If you choose Save, you will get this screen:

```
Reggae, Complex Plate [11.14]                  ↑
?ABCDEFGHIJKLMNOPQRSTUVWXYZ0123456789.!-
       ↑
 ←→                                         Save
```

If you press Save from this screen *right away*, the preset will be saved into the chosen register with the original name. If you want to change the name, move the first fader. The original name will disappear from the display and a letter corresponding to the position of the fader will appear in the first space instead. At the same time, the cursor underneath the "alphabet" will move horizontally, showing you which letter you have chosen. If the fader moves the cursor too fast for your liking, you can increment the letter by using the Soft Button underneath the double-headed arrow. Pressing the button moves the cursor one letter to the right, and pressing Shift and the Soft Button moves it to the left.

When you have the first letter, press the "ENT" button, and move the first fader again to choose a letter for the second space. Keep going until you have spelled out the name you want. You don't have to fill all six spaces. You can edit what you've done by pressing "BSP", which resets the last letter you've chosen, or by pressing "CE", which clears out all the letters. When you have the name you want, press the Soft Button under Save, and you will save the preset under the new name. Once again, the "↑" button will let you leave this menu without taking any action.

Fast Storage — the Set function

There's a second method for storing a preset, which is very useful if you have just tweaked an existing preset and want to leave it in the register it originally came from, with the same name (thereby erasing the original version). Pressing the "Set" button (Shift-Last) takes the Current register (i.e., the program currently operating) and copies it into the source register it came from, without changing its name.

For obvious reasons, this function will not work with a preset in a Factory bank.

Other Control Functions

LCD Viewing Angle (VIEW UP, VIEW DN)

You can optimize the LCD character display for best readability from your viewing position. Press Shift-↑ ("View Up") to improve readability from a viewing angle overhead or from the rear. Press Shift-↓ ("View Dn") for better viewing from the front. The new setting is remembered in nonvolatile internal RAM of the ADR 68K mainframe until you again change it.

Input Mute (MUTE)

This button at the front left edge of the remote allows you to "insert" a mute directly after the ADR 68K's input ADC, before the processor. As long as Mute is "on", no *new* sound can reach the processor; however, any sound *already* echoing or reverberating in the processor is not affected, and will continue to be audible as before.

Mute can be used in an immediate, momentary fashion by simply pressing and holding the button, or it can be "latched" on by pressing Shift-Mute. To unlatch it, press Mute again. When Mute is on, a reverse video character "M" appears toward the end of the first line of the display.

Mute can be used in a number of ways. If a sound is being processed with a reverb with a long decay time, turning Mute on allows the reverberating sound to be heard all by itself as it continues to die out. This can be nice at the end of a musical phrase, or as a special effect.

If a really large (or infinite) decay time is being used, Mute can be used to open and close the input audio path, allowing new sound to be layered on top of the previously entered sound still held in the reverberation memory. In fact, when running with an infinite decay time, you should turn Mute on and leave it there until you want to "layer" or overdub another sound into the memory. This prevents any residual noise from the external source or the unit's own ADC from piling up continuously. When you want to add another layer, turn Mute off, add the sound, and then latch Mute back on.

Mute operates in a special way with presets that use gated reverb. If Mute is on in such a program, then the gate function will be disabled and the decay time values of the *stopped reverb* page will be operational at all times. The gate will open in response to changes in input level only after Mute is turned off. See the next chapter for a detailed discussion of gated reverb.

Mute will also turn off any samples that are playing within a reverb or effects program, treating them as if they were input signals. In a Sampler preset, Mute disables the audio feed from the inputs to the samplers.

Reverb Clear (CLR)

Clear is basically the inverse of Mute. When you press Clear, the audio input path stays connected to the reverb or effects processor, but internally a decay time of zero is forced into the reverb RT60 parameter. Any old reverberating sound is "flushed" out or dumped from audio memory, clearing the way for new reverberant sound to build up as soon as the Clear button is released. Like Mute, Clear can be used in an immediate, momentary fashion by simply pressing and holding the button, or it can be latched on by pressing Shift-Clear. When Clear is on, a reverse video character "C" turns on toward the end of the first line of the display.

In reverb and effects programs that use samples, pressing Clear while a sample is playing will cut off the sample as well as the reverb. (If more than one sample is playing, they will all be cut off.) In Chorus and DDL programs, pressing the Clear button will halt all modulation of the delay taps, but will not turn off the taps themselves, so the levels remain constant. This can create some useful effects, which will be discussed in the next chapter.

Page Up (↑)

Page Up is used to go backwards from a menu to the prior, higher level menu. If pressed several times, it will eventually return the display to showing the currently running preset. Whenever you see an up-arrow on the display, it means that you can back up one or more levels with the "↑" button, always returning to a control page of the current preset. (You will be returned to whichever page of the preset you were on when you last left it.) It is often useful to think of "↑" as an "Escape" key — if you find yourself in a menu where you're not sure what to do, except that you want to get out, pressing "↑" will usually get you back to more familiar territory.

The only places the "↑" key will *not* get you out of are the Help screens, discussed below. To leave a Help screen and get back to a control page, press the Help button.

Back Space (BSP), Clear Entry (CE), Enter (ENT), and the Numeric Key Pad

All of these are used in the Quick Register Select function discussed previously. The numbers specify banks and registers; errors can be corrected with BSP (delete previous character) and CE (delete all characters); and the command is executed when ENT is pressed. Also as discussed previously, these buttons are used similarly when naming Presets.

System Functions, Reset, and the BPM calculator (SYS)

The SYS button along the front edge gives you access to certain system functions, which are used only rarely. Pressing SYS gives the following display:

```
System menu          Software Rev #4.01      ↑
                     Last revision 13 Dec 87
                  Auto-
Reset   BPM       Erase
```

The top line gives the revision level of the software in your ADR 68K. You will need to know this when you ask for technical support from AKG. The first digit (before the period) is incremented only when significant changes or additions are made to the programs or system software. The digits following the period are incremented in between significant updates as smaller features are added or bugs are fixed.

From this menu you can "reset" the ADR 68K. *This will clear all Internal registers and reset all system and MIDI functions to the state they were in when the unit left the factory.* If you want to reset the unit and not lose all the presets in the Internal memory, you should first copy them onto the Cartridge (see below for instructions).

Pressing the Reset button gives the following display:

```
Warning, this will erase all internal
registers, shall I continue?

                          Yes     No
```

If you press Yes, the display shows:

```
Initializing system...
```

And then returns you to Page 1 of the Factory preset 1.1.

The second Soft Button turns on a BPM (Beat-Per-Minute) calculator. This function has nothing to do with the audio path — it is provided as a convenience for setting delays to fit the rhythm of a song.

It is very simple to use. Set the first fader at the metronome setting of the song you are working with, over the range 32 to 286 beats per minute. The fader will only access even numbers; if you need to set it to an odd number, increment the number using the Soft Button. Now move the second fader so it corresponds with the note value of the delay you want to set up, from a sixteenth note (1/4-beat) to a whole note (four beats). Like magic, the proper setting for the delay will appear on the second line of the display.

```
Compute delay time from beats/minute:     ↑
      -- Set delay to: 468ms --

  BPM     Note Value
128bpm       1/4
```

Finally, there is the "Auto-Erase" slider, which affects how samples are handled when switching sampler programs or turning the unit off and on. When this control is "On", then every time you go from one sampler program to another, or every time you turn the ADR 68K's power *on*, the sample memory is erased completely. When the control is "Off", the memory is not erased, but instead remains intact to one degree or another. Exactly how intact it will be depends on the nature of the sampler programs you are switching between, or the length of time between turning the power off and turning it back on again. This will be covered in more detail in the section on Using Samples in Chapter 6.

When you receive the ADR 68K from the factory, or when you perform an upgrade to software revision 4.0, this switch is set to "On".

Using the Cartridge (CART)

The cartridge supplied with the ADR 68K is used for storing presets, giving the user an extra 50 registers for that purpose. It is *not* used for storing samples.

Pressing the CART button brings these options on screen:

```
64K Cartridge installed, 43 of 50 free  ↑

Cart➜     Int ➜
 Int      Cart    Swap              Init
```

The top line identifies the size of the cartridge presently installed, and how many of its registers are free (i.e., have not had presets written into them). (Cartridges with more than 64K of memory will be available from AKG in the future.)

However, if there is no cartridge installed, you will see:

```
Cartridge not installed                 ↑



  —      —      —                 —
```

To get out of here, press "↑".

If the *write-protect switch* on the data cartridge is set to on, the CART menu will say so, and will not display the Int→ Cart, Swap, or Init options.

Cart→Int, Int→Cart, and Swap

These functions allow you to copy all the internal registers of the ADR 68K into its cartridge or vice versa. Of course when you do so, you overwrite the registers of the destination memory.

Pressing Cart→Int gives the display:

```
Warning, this will overwrite all user   ↑
registers, shall I continue?

                          Yes     No
```

Pressing Int→Cart instead gives this display:

```
Warning, this will overwrite all        ↑
cartridge registers, shall I continue?

                          Yes     No
```

Answering Yes to either of these options will complete the copying process. Pressing No or "↑" will return you to the first Cartridge menu.

If you press Swap, the contents of the cartridge and the internal memory are exchanged. This command does not ask for verification.

INIT

This command will erase the cartridge completely. Be really sure you are prepared to lose all the registers in your cartridge before choosing this option. If you do press Init, you are given one more chance to change your mind:

```
Warning, this will overwrite all
cartridge registers, shall I continue?

                    Yes      No
```

Help Function (HELP)

The Help button brings short explanations of controls or procedures up and displays them on the LCD. The Help screens are "context sensitive"; that is, they give information relevant to whatever operations or parameters are currently in use. When you enter the Help section from the first page of a preset, a description of the current algorithm appears, as well as descriptions of the controls on the page. The algorithm description does not appear, however, if you call for Help from a page of a preset other than the first, or from anywhere else.

When you press Help, there is no change in the ADR 68K's operating mode or sound processing — only the LCD display is changed. You return the LCD display to its former status *not* by pressing "↑", but instead by pushing the Help button a second time.

When one screen isn't sufficient to show all the help information, you will find more information by using "↓" or "↑" to scroll through all the screens available.

If You Have Trouble — Remote Self Test and Cold Reset

Although it is highly unlikely, it is just possible that a hardware failure or an as-yet-undetected software bug could cause your ADR 68K to lock up. Symptoms of this would be a sudden lack of response from the faders and/or buttons on the remote, or a frozen signal-level display. Most likely the ADR 68K would continue to give normal audio — you would just have lost control of all functions.

If this situation occurs, the first thing to try is to disconnect the remote, wait five seconds, then reconnect it. This may clear the problem, and you can go on. If not, then shut off the power switch on the mainframe (remember to turn down the monitor first), wait five to ten seconds, and then turn it back on. Chances are all will be restored to normal.

If this doesn't work, you should next perform a Self Test of the remote. This self test requires only that the remote be connected to the ADR 68K mainframe, and that the mainframe +9V power supply is functioning. None of the other mainframe functions or circuits are used, limiting the test to the remote itself.

To perform a remote self test:

1. Turn the ADR 68K power off and wait 10 seconds.
2. Press and hold down the Help button on the remote.
3. While you are holding, turn the ADR 68K power back on.
4. Continue holding the Help button down until the display changes to:

```
Choose one:

      COLD for system cold reset
      SELF for remote self test
                          COLD   SELF
```

If you lose your nerve here, or want to exit for other reasons, shut the power off, wait five seconds, and power up again.

5. Press the soft button labeled SELF.
6. The display will change to this:

```
comm:—— rev:2.00 view:60 key:__
 ———fader———         ———jack——
 F0  F1 F2 F3  F4 F5  J1  J2 J3 J4
127  0  0  0   0  0   0   1  1  0
```

and the input-level and internal-level LCD's will start jumping up and down in step.

You can now test every function of the remote control. Here's how:

1. "faders" shows the position of each fader, with a number from 0 to 127. Move each fader and observe the display.

2. "jacks" shows the values obtained by the internal analog-to-digital converters from the four 1/4" jacks on the rear. If nothing is connected to a jack, this will read 0.

3. "view:" will change in response to using the VIEW UP and VIEW DN buttons to change the viewing angle. Try it.

4. "key" will show the name of each button as you press them one by one, thereby letting you test all of them. Try the buttons one at a time. (The Shift button by itself will give no response.)

5. "comm: FAIL" or "comm: —" is normal, and simply indicates that the remote is not trying to communicate with the mainframe. Rather, it is looking for a "loop-back" connection of its data-out and data-in connector pins so that it can self-test its communications chips and ports (this is used at the factory).

NOTE: the only way to leave the Remote Self Test mode is to turn off the power. Be assured this will not affect any stored presets, but any samples you may have in memory will be lost or scrambled.

If the remote tests out okay in Self Test mode, then there is one final step available to you: a Cold Reset. Be aware that a Cold Reset, like a System Reset, will erase any user presets you have in the internal registers, so if you value them you should copy them onto a cartridge first (if you can).

Here's how to do a Cold Reset:

Follow the first four steps in the procedure above for the remote self test. When the first display appears, press the button under "Cold". You will see:

```
COLD
System cold reset Warning: this will erase
all internal registers:
Do you want to continue?
                                YES    NO
```

Press the soft button labeled YES.

If the Cold Reset is successful, you will next see the first page of parameters for Factory register 1.1, and you will be able to proceed normally. If you're still having trouble, you most likely have a hardware problem. Refer to the section on Servicing in Chapter 3, or consult your dealer for help.

Audio Test

An audio test mode is available to verify basic audio functioning of the ADR 68K and the signal processor. In this mode, the left input feeds the left main and auxiliary outputs, and the right input is feeds the right main and auxiliary outputs, with unity gain and no delay. Although it seems like the signal is being unaffected, it is actually going through the analog-to-digital convertor, the signal processor bus, the multiplier/accumulator, the audio scratch pad memory (temporary registers), and the digital-to-audio convertor, as well as all the audio circuitry of the ANA-3 PC board.

To use this mode, follow the first four steps in the procedure above for the remote Self Test. When the first display appears, you are in the audio test mode. Put a signal into each of the inputs, and test it at the outputs.

rev 4.0, 15 December 1987

— Chapter 5 —
Reverb and Effects

While the ADR 68K is loaded with over 100 useful programs when it comes from the factory, most users are going to want to design their own programs. To help you in this, this chapter will describe the reverb and effects algorithms used in the unit, and how to use them, with particular emphasis on how to take advantage of the unit's special features. Using the samplers is covered in the next chapter, and MIDI applications in Chapter 7.

For a general outline on using reverb, sampling, and MIDI, please read Chapter 2 of this manual if you haven't already done so. A glossary of the terms used here appears at the end of the manual, as Appendix A. Following that, in Appendix C, are detailed menus for each of the algorithms, block diagrams showing how they are configured, and lists of the factory presets that use each algorithm.

Although this chapter contains lots of information, often the best way to teach a new concept is not by explanation but by example. Therefore, we suggest that you spend some time examining the presets — especially the more exotic ones — to see how certain effects are accomplished. We also encourage experimentation; remember, you cannot destroy the Factory presets, and there's plenty of room in the Internal and Cartridge memories for storing your experiments.

The Reverb algorithms

The ADR 68K contains six different reverb algorithms. Although each of these algorithms represents a different "kind" of reverb, they are all extremely flexible, thanks to the number and range of adjustable parameters within them. They also (with some exceptions) take advantage of the unit's stereo inputs, and can be "gated" (a discussion of which comes later in this chapter).

The primary parameters generally appear on Pages 1 and 2 of a preset. The first parameter you will encounter, at the left-most side of Page 1, is "**RT60**", which is the length of the reverb. Strictly speaking, it is the time it takes the reverb level to decay to 60 dB below its initial level. It is adjustable, depending on the algorithm, from a small fraction of a second to over a minute, and can even be set to "infinity", in which case there is no decay at all.

The RT60 of the low and high portions of the audio spectrum can also be adjusted separately: the "**LF Dcy**" parameter affects frequencies below about 500 Hz, while "**HF Dcy**" affects frequencies above about 2 kHz. Both controls adjust their respective decays as ratios of the nominal RT60, from (RT60)x0.1 to (RT60)x2.0. These parameters can help determine the apparent surface characteristics of a space: a hard-surfaced room will have a longer HF Dcy time, while a more absorbent room will have a shorter HF Dcy, and/or longer LF Dcy.

Also adjustable is the bandwidth of the signals coming into the reverb processor. The cutoff, which is controlled by the "**HF BW**" parameter, can range from 15 kHz down to 2 kHz. A delay can be imposed on a signal before it reaches the reverb processor, delaying the build-up of reverb, while not changing the early reflections of the apparent space (which are handled separately, as explained below). The parameter controlling this is "**P'Dly**", which is adjustable from 0 to 500 ms.

Very important contributions to the sound of a reverb — particularly concerning the definition of a stereo space — come from the early reflections. The ADR 68K gives you control over as many as six distinct early **Reflections** (also referred to as "E-Rs"). Three early reflections are derived from the Left input, and two of these go to the Left output, while the third crosses over to the Right output. Three other reflections are generated from the Right input, and two of these go to the Right output, while the third crosses to the Left. Control over the reflections usually appears on Pages 4 and 5 of a preset. Each of the early reflections can be individually delayed up to 500 ms, and their levels are likewise individually adjustable over a very wide range.

"Diffusion", "Density", "Punch", "Depth", "Randomize", and "Size" are parameters that appear in various combinations, depending on the algorithm. **Diffusion** and **Density** refer to the relative concentration of echoes at the beginning and end of the reverb, respectively. Low Diffusion values can make small rooms more interesting, with more discrete echoes sounding at the beginning of the reverb. Low settings of Density are good for improving the intelligibility of vocal lines and for making solo instrumental lines stand out, but the sound isn't as smooth as it is at higher settings. Higher settings make the most sense for drums.

Punch increases the *proportion* (not the density) of early echoes within the reverb processor, making the reverb effect more immediate and dramatic. **Depth** controls the ratio of the dry and reflected sound in a complex manner (it doesn't just balance and mix them), thereby controlling the apparent distance between the listener and the sound source. At the lowest values, the direct sound is highest, putting the listener close to the source.

Randomize is used to smooth out the reverb "tail" and make it less repetitious and more interesting. It uses multiple taps, and moves them back and forth with a low-frequency oscillator. It is a feature that should be used carefully, as it can add a granular noise to high-frequency signals, or sometimes a slight pitch-shifting effect.

Size is a complex parameter that simultaneously changes the RT60 and the density of the reflections within the reverb. It works semi-interactively with the RT60; changing the size alters the RT60 setting, but not vice versa. It's best to think of Size in the following way: for a given RT60, a Size setting of 100% represents a typical room. Increasing the Size, while it increases the RT60, also *reduces* the density of the reflections, thereby reducing frequency coloration, but increasing the chance that individual echoes will be heard. Decreasing the Size has the opposite effect — fewer echoes will be audible, but frequency coloration will increase.

As mentioned in the previous chapter, the ".1" positions in each of the factory reverb banks contain good "general-purpose" sounds using the appropriate algorithm, and are good starting points for users wanting to experiment and design their own presets.

Now a few words about each of the algorithms. For details about the parameters used in each algorithm and their ranges, as well as block diagrams and lists of the factory presets that use each algorithm, see Appendix C.

The **Plate** algorithm in bank 1, whose formal name is "Complex Plate", is a simulation of a metal-plate reverberator, which features a high echo density, fast diffusion of the sound, and little coloration. It has a wide range of applications, especially in pop music and with percussion.

Chamber, or "Optimal Chamber" (bank 2), is useful for simulating a range of spaces, from large rooms to small halls.

Room, or "Medium Room" (bank 3) is good for simulating smaller acoustic spaces, such as film and video interiors, and can be used with most kinds of non-symphonic music.

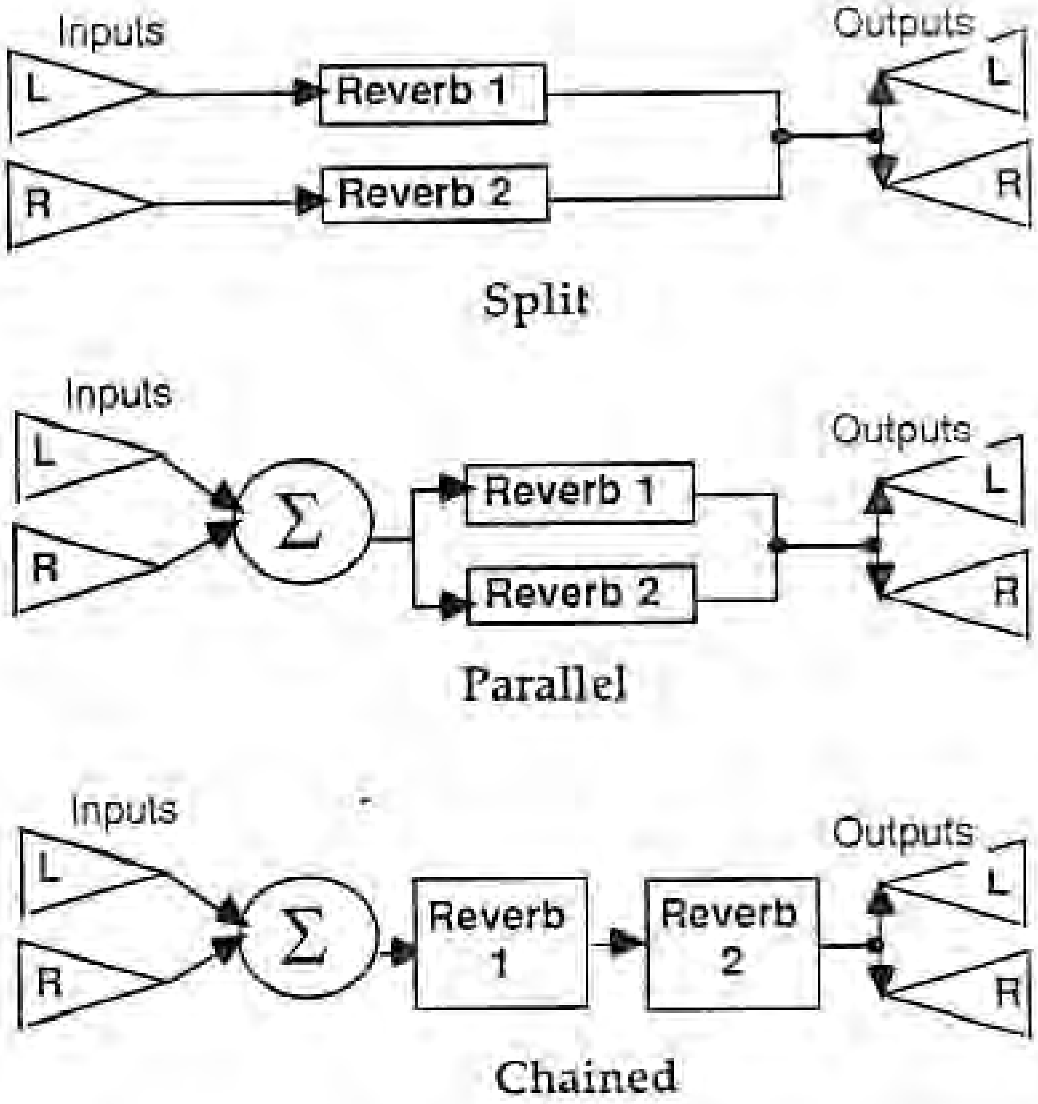
Hall, or "Natural Hall" (bank 4), is best for the largest acoustic spaces, ranging from small concert halls to canyons. It creates a spacious, open-sounding reverb, good with orchestral and "spacey" electronic music.

Split, (bank 5), is actually several different algorithms combined in one bank. What they all have in common is that they allow two different processing functions to be active at one time, effectively splitting the ADR 68K into two distinct devices. Each side of the split has one (mono) input and stereo outputs. A representative preset using each of the five Split algorithms appears on the first page of Factory presets. On subsequent pages, all of the presets on a given page use the same algorithm.

One algorithm, which appears on the second page of Factory presets, is "Plate/Hall". In this algorithm, the left audio input feeds the "Plate" side, while the right input feeds the "Hall" side. The Plate side can be gated, but not the Hall side. Only two early reflections are available in each half. Pages 1 through 3 control the Plate characteristics, and Pages 4 through 6 control the Hall. The output from both halves is available at the main audio outputs, while only the output from the second half appears at the Aux audio outputs.

The other algorithms are even more flexible. They are, going through the pages of presets, "Plate/Plate", "Hall/Hall", "Hall/Chorus", and "Hall/DDL". Each half of a split is essentially identical to the algorithm described previously that bears its name, except that one or more less-crucial features may be altered or left out. In all cases, the first side of the split is more complex than the second side, i.e. it allows gating, and has more parameters available.

The "Chorus" half of the "Hall/Chorus" algorithm is identical to the Poly-Chorus algorithm used in bank 10, and described later in this chapter, except that there is no "Diffusion" parameter, and the number of taps available is three, not six. The "DDL" half of the "Hall/DDL" algorithm is identical to *one* of the delay lines in the Dual Delays algorithm used in bank 11.



The splits in these four algorithms are configurable, with a switch on Page 6 of each preset. When the switch is set to "Split", the left audio input feeds the first half of the split, while the right input feeds the second half (as in the "Plate/Hall" algorithm just described). When the switch is set to "Para", the left and right inputs are merged and the combined signal feeds both halves of the split. When the switch is set to "Chain", the two inputs are combined and feed the first half of the split, and then the outputs from that half go to the second half of the split.

Normally, the output from the first half of the split appears at the main audio outputs, while the output of the second half appears at the Aux outputs. However, a switch is included to allow the signal that comes up at the Aux outputs to also appear at the main outputs. See the sections later in this chapter and in Chapter 6 on output mixing for more details.

Revers, which is short for "Reverse Reverb" (bank 7), simulates a backwards reverberator, whose output starts low and rises over time. It uses just a few parameters: the RT60 controls the total length of the reverb event, while Punch controls the level of the *last* sound heard (which is the image of the *first* sound going in). HF Dcy modifies the decay characteristics. There is no gating, or control over early reflections. The program sums the inputs into mono, and produces a stereo image. When a signal is being processed, it comes out at first in stereo, and then collapses over time into mono.

A note about stereo inputs

Reverb algorithms with true stereo inputs give special benefits, and also need special care. Many digital reverbs have two inputs, but often the two input signals are combined ("summed") internally before they get to the actual reverberation processor. While this can create "stereo" reverb, that reverb is going to be exactly the same regardless of whether the input has been applied to the left input, the right input, or to both inputs.

A reverb with true stereo inputs, on the other hand, will sound slightly different if sound enters only one input. The first four reverb algorithms of the ADR 68K all have true stereo inputs. In these programs, the early reflections and the first portion of the reverb build-up will be different for left and right inputs, giving a stereo perspective to the initial reverb sound.

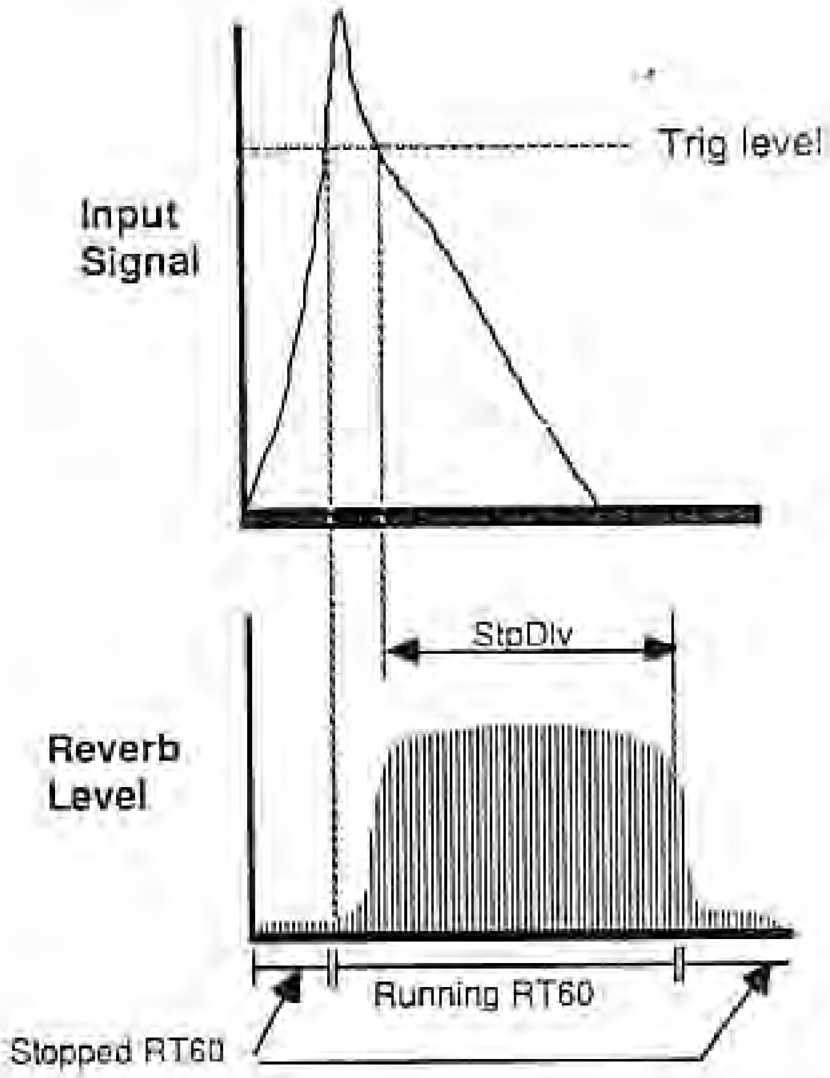
This is easy to deal with if you have a stereo signal to reverberate, like a final mix or a stereo track to process in mixdown. But if you are working with a mono source, you'll need to send it to *both* reverb inputs. If you don't, the initial reverb sound will be unbalanced (off to one side), and the later reverb will not achieve a proper echo density.

Gated Reverb

As mentioned above, all of the reverb programs in the ADR 68K (except "Revers") allow for gated reverb. The term "gated" reverb is a holdover (like "flanging") from an earlier recording technique in which the outputs of a spring or plate reverb, with its relatively fixed decay characteristics, were fed into a noise gate so that the sound would decay more rapidly and dramatically.

In a software-driven digital processor like the ADR 68K, the capabilities and operation of gated reverb are somewhat different. Each reverb program has two different decay characteristics, both under user control. The first, or primary, decay characteristics are those of the Page 1 "Running Reverb" parameters. The second decay characteristic is that of the Page 3 "Stopped Reverb" parameters.

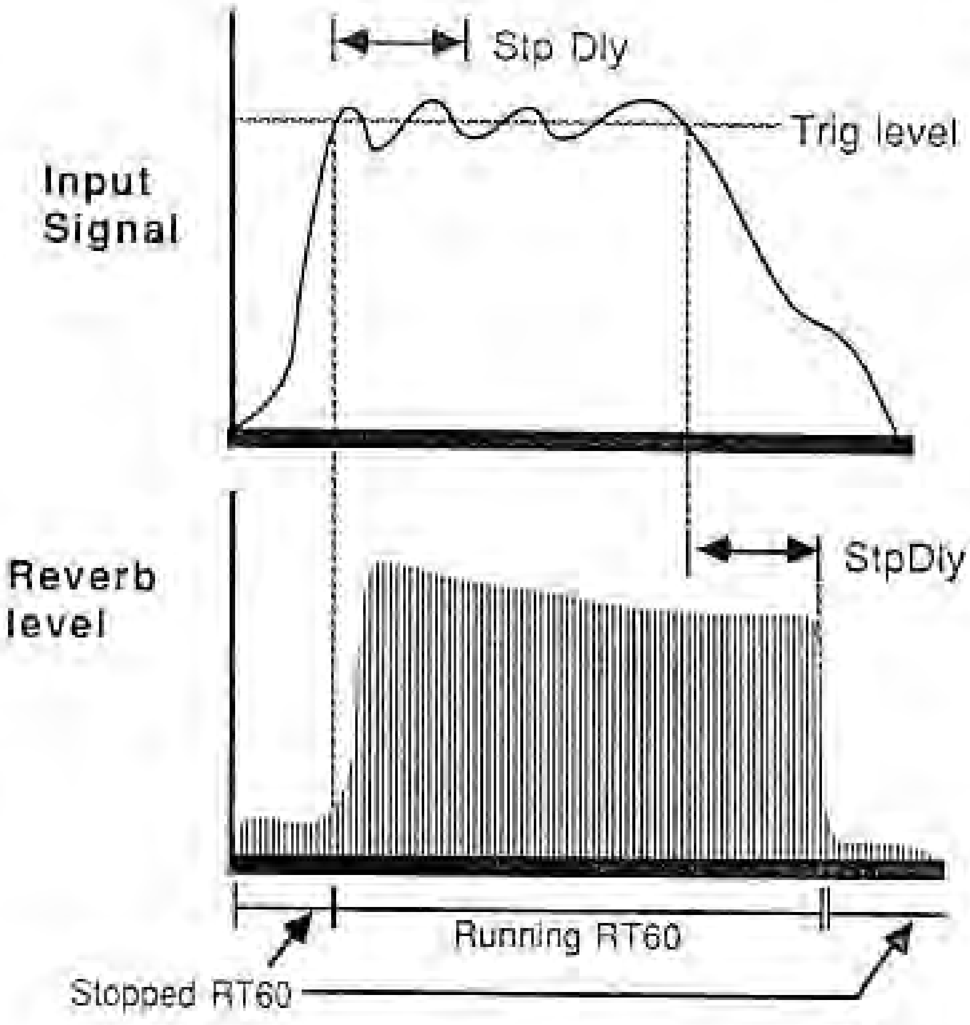
Page 3 is where the action of the gated reverb is controlled. The "Trig" parameter sets a threshold level for the input signal. When the signal level (at either the left or right input) reaches the threshold, the gate is "opened", and the Page 1 Running Reverb parameters take effect. When the signal drops below the threshold, the gate can "close" and the Page 3 Stopped Reverb parameters take over. The threshold can be set from -42dB to the input-level limit ("LIM") of the unit, in steps which correspond to the steps of the Input Level LEDs on the remote control.



If you select a very low threshold, say -42dB, almost any signal coming along will be high enough to open the gate, and thus the Running Reverb settings will always be in effect. With a setting of, say, -4 dB, only the loudest signals will open the gate, and so the Stopped Reverb parameters will be in charge most of the time, except at the highest peaks. You will probably find a setting of -18dB or so to be about right for percussive sounds. Note that setting the Trig control to OFF (its default setting) will disable the gate entirely, leaving it open at all times (and rendering the other parameters on Page 3 irrelevant).

When the input signal drops below the threshold, the gate does not close immediately. The amount of delay between the two events can be adjusted from 10 ms to 5 seconds, using the "StpDly" parameter. If the StpDly is too long — for example, if it is longer than the time between percussion hits — the ADR 68K will never switch to the Stopped Reverb settings. If it is too short, then you will lose most of the effect of the Running Reverb.

Therefore, to achieve gated reverb the input level must exceed the Trig level at one point, and then must fall below the Trig level for a period of time longer than the StpDly setting.



Input level dipping below threshold for periods shorter than StpDly interval

One popular use of gated reverb is to fatten up a percussive sound by giving it lots of reverb when it's loud, and little when it's soft. The effect is to lengthen or broaden the sound, without making it "reverby". This calls for a large RT60 for Running Reverb and a short RT60 for Stopped Reverb.

Here's a typical setting for use with percussion:

```
GROOVY, Complex Plate [I1.4]          [1] →
——— Running Reverb ———
RT60  LF Dcy  HF Dcy  HF BW   Size  P'Dly
20s   X1.0    X1.0    15kHz   100%  500ms
```

```
GROOVY, Complex Plate [I1.4]       ← [3] →
—Gate—            —Stopped Reverb—
Trig    StpDly    RT60    LF Dcy   HF Dcy
-18dB   0.30s     0.8s    X1.0     X1.0
```

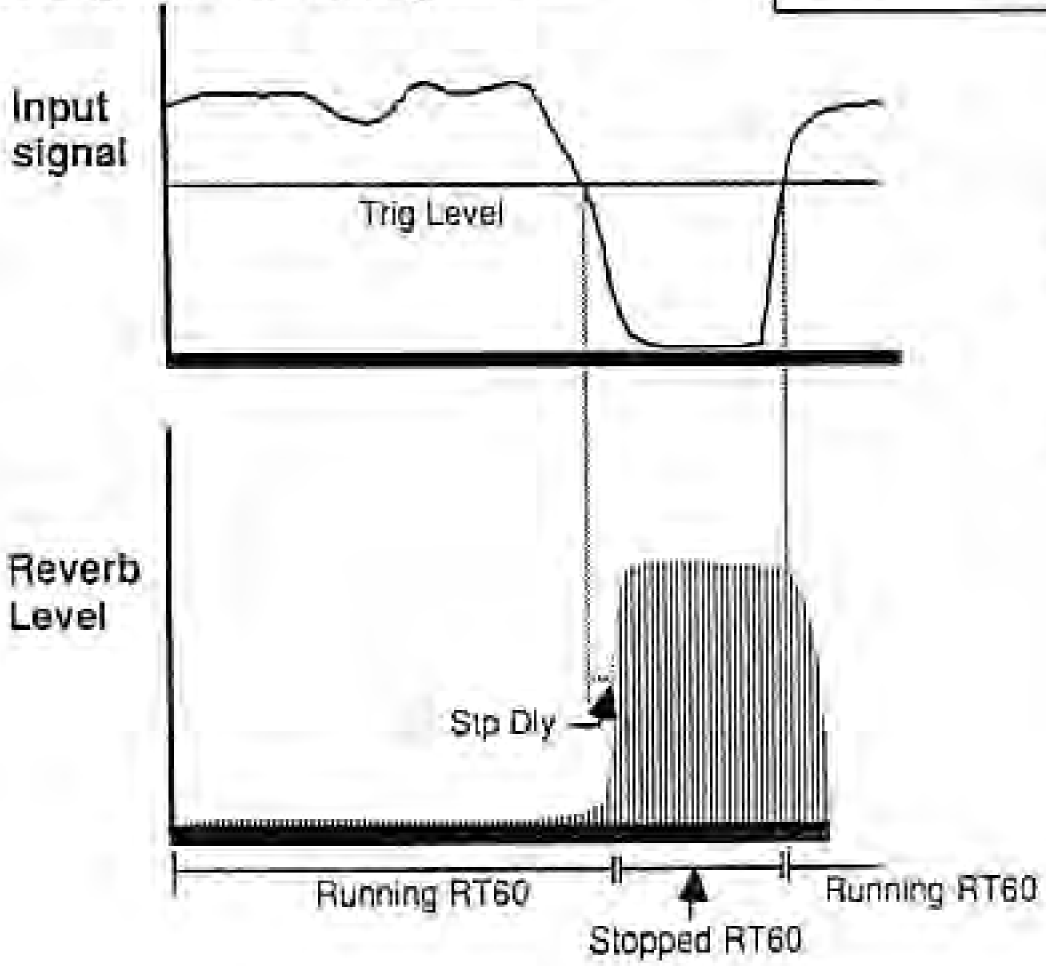
This is how it works: Let's begin with no input signal. The ADR 68K's gate is closed, and the Page 3 Stopped Reverb RT60 of 0.8s (a very short decay time) is in effect. Along comes a snare drum hit with a level of -4dB. This exceeds the -18dB Trig threshold, so the ADR 68K switches to Running Reverb settings of Page 1, with RT60 of 20s (a very long decay time!). The drum hit is grabbed and almost "frozen" in the reverberator.

The snare drum has a short duration, so it ceases, and the input level drops to below -18dB. When it does, the microprocessor starts counting out the StpDly interval of 0.3s. Assuming that during that interval another drum hit doesn't come along, then after 0.3s the ADR 68K switches to the Stopped Reverb RT60 value of 0.8s, and the "frozen" drum hit is quickly dropped out due to the short RT60. We are now back where we began.

Another way to use gated reverb is exactly the opposite of the percussion effect just described. Here we want very little reverb to accompany the louder sounds so as not to clutter or muddy the main musical passages, yet we want a good amount of reverb to linger as a "tail" during the gaps in the music. The settings here will accomplish this:

```
SPACEY, Complex Plate [I1.4]              [1]➡
———Running Reverb ———
RT60   LF Dcy   HF Dcy   HF BW    Size   P'Dly
1.0s   X1.0     X0.5     15kHz    100%   500ms
```

```
SPACEY, Complex Plate [I1.4]          ⬅[3]➡
—Gate—               —Stopped Reverb—
Trig    StpDly       RT60     LF Dcy    HF Dcy
-12dB   0.20s        2.5s     X1.0      X0.5
```



In this mode, the reverb time is short (1.0s) as long as there is an input signal higher than -12 dB, but as soon as the input drops out, a longer reverb (2.5s) takes over quickly (in 0.2s).

Gated reverb with short Running RT60 and long Stopped RT60

You can also turn the ADR 68K into a "true" gated reverb. If you set the RT60 level on Page 3 to MUTE, then when the signal level goes below the Trig threshold, the audio outputs of the unit will shut down completely (after the StpDly interval, of course).

Because the ADR 68K provides separate LF and HF Dcy parameters for the Running and Stopped Reverb, you can also achieve interesting effects based on differences in these parameters alone. For example, if you set the HF Dcy on Page 1 to x1.0 and the HF Dcy on Page 3 to X0.1, you can cause the Running Reverb to be bright and sibilant, with a rapid loss of high-frequency energy when the Stopped Reverb takes over during gaps in the input sound. Conversely, if you set the Stopped Reverb HF Dcy much higher than the Running Reverb HF Dcy, then after the gate closes, you can get that "escaping steam" effect so popular in some circles.

The Effects algorithms

Besides reverb, the ADR 68K is capable of a very wide range of digitally-produced effects. They are broken down into four categories, each with its own bank (some effects also appear within the split programs). We'll deal with these effects in order of complexity.

Dual Delays (DDL)

The Dual Delays algorithm (abbreviated "DDL"), which resides in the factory presets in bank 11, contains two full-bandwidth digital delays, with delay times adjustable from 160 µsec to 2 seconds. The first two pages are for adjusting the parameters of each delay.

Range and **Delay** set the length of the delay. Range sets the minimum and maximum values and the resolution of the Delay fader, and gives the values a descriptive name. For example, if Range is set to "Long", the maximum value is 2000 msec, and the fader resolution is 40 msec, while if Range is set to "Sampls", the maximum delay is 1760 µsec, and the resolution is 32 µsec (which is equal to the length of one sample created by the ADR 68K's analog-to-digital convertor). In table form, the Range settings are as follows:

Range	min value	max value	fader resolution
Sampls	160 µsec	1760 µsec	32 µsec
Flange	0.0 msec	10.0 msec	0.2 msec
Chorus	0 msec	50 msec	1 msec
Echo	0 msec	200 msec	4 msec
Long	0 msec	2000 msec	40 msec

The **Gain** control handles feedback of the delay to itself. "0%" means no feedback — the delay just plays once — while "100%" means the signal will recirculate forever, without changing level. Negative gain values mean that the feedback is 180° out of phase.

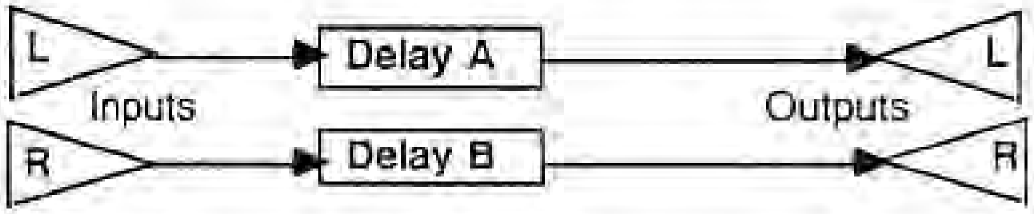
HF Dcy is a low-pass filter in the feedback path.

Depth and **Rate** control a low-frequency oscillator that is used to "modulate" the delay lines by altering their lengths. The Depth parameter controls how much modulation is used. For example, if a delay time is set for 100 msec, and the depth control is set to 50%, the delay time will vary between 50 msec and 100 msec. The modulating waveform is a triangle. Rate, of course, is the frequency of the modulation. In the longer delay ranges (Chorus, Echo, and Long), the Rate is adjustable from from 0.01 Hz (one cycle every 100 seconds) to 1 Hz, while in the shorter delay ranges, it is adjustable from 0.05 to 9.99 Hz. These controls can be used to create some fascinating pitch-shifting, vibrato, or frequency-modulation effects.

On the third page, **HF BW** is a bandwidth-limiting control that acts on the entire unit, adjustable down to 2 kHz.

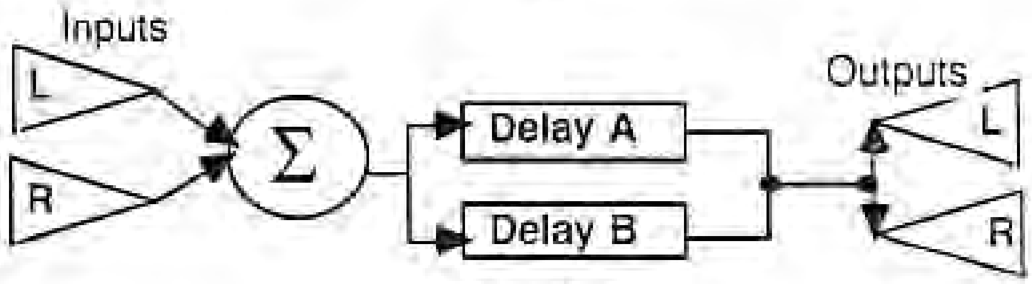
Config determines the relationship of the two delays to each other. There are three choices:

In "Split" configuration, the Left input feeds delay A, which feeds the Left output, and the Right input feeds delay B, which feeds the Right output.



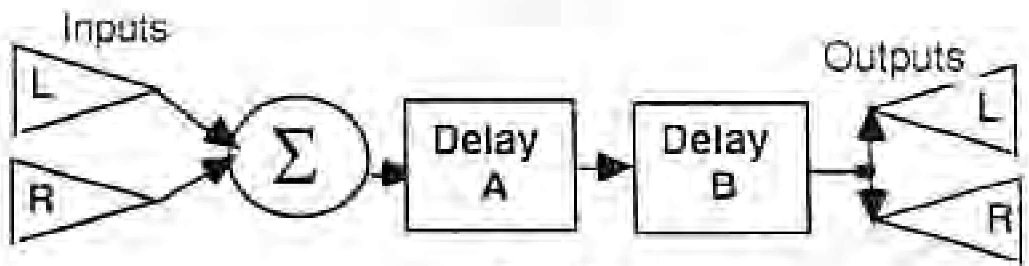
Split

In "Parallel" configuration, the Left and Right inputs are summed, and then fed to both delays. Then, the outputs of the delays are summed, and sent to both outputs.



Parallel

In "Chained" configuration, the Left and Right inputs are summed, and sent to delay A. The output of delay A then goes to delay B, and the output of delay B goes to both audio outputs.



Chained

The CLR button has an interesting effect when used with this algorithm. It stops the modulation process, but does not kill the delays themselves, so that any modulation effects being created are stopped, but the signal, and its delays, keep going. This means that such effects can be brought in and out instantaneously, and very dramatically — in "mid-flange", as it were — with the CLR button.

Samples created with the * Sample programs can be used in ddl programs, and can appear before and/or after the delays. When you are feeding multiple monaural samples (created with the "Mono*" preset) into a delay program, odd-numbered samples are treated as if they were appearing at the left audio input, and even-numbered samples are treated as if they were at the right input. Therefore, wherever the Left audio input is going (which, depending on the configuration, will be the either delay A or both delays), that's where the odd-numbered samples will go.

The outputs of the ddl do not appear at the aux outputs of the ADR 68K, but the (dry) samples do.

Poly-Chorus (Chorus)

The chorusing algorithm, whose formal name is "Poly-Chorus", is found in the factory presets in bank 10. This algorithm can produce chorusing, pitch-shifting, automatic panning, and many other effects. A version of the algorithm makes up the second half of the "Hall/Chorus" Split algorithm. All of the controls described here are applicable to both algorithms, except as noted.

The algorithm creates up to six delay taps, and moves them in time, modulating them with a triangle wave. The taps always move in opposition to each other, and the phase relationship of their modulating waveforms can be determined by dividing 360° by the number of taps; i.e., if you are using two taps, their modulation will be 180° out of phase with each other, while if you are using five taps, each will be 72° out of phase with the next.

The first parameter on Page 1 sets the number of taps, or **Voices**, from one to six. With one voice, the vibrato and panning effects can be heard, while as the number of voices increase, the more pronounced the chorusing effect becomes. (In the Split algorithm, the number of voices is limited to three.)

The **Depth** parameter sets the range over which the taps will be moved; the delay time of the taps will constantly be moving from 0 (or another "base point", which we'll get to in a moment) to the time indicated by this parameter. The maximum Depth is 10 ms. The **Rate** determines how fast the taps will move. If this control is set to "OFF", the taps don't move at all. Slower rates impose a vibrato on the sound, while faster ones impart almost a ring-modulation effect, by actually frequency-modulating the input signal. The rate is adjustable from 0.05 to 9.99 Hz.

The amount of pitch shift you hear will be dependent on both of these parameters. Pitch shift is determined by the speed of a moving tap, which in turn is determined by both the distance the tap has to move (the Depth) and the amount of time it has to get there (the Rate).

Delta is only effective when you are using more than one voice. It specifies the offset in time between the various taps, by adding extra delay to each tap. With a Delta of greater than 0, the "base point" for any tap other than the first will no longer be 0, but will be [Delta *x* (tap#-1)]. For example, if you are using two voices, with a Depth setting of 5 msec, and a Delta of 20 msec, tap 1 will be going back and forth between 0 and 5 ms, while tap 2 will be offset 20 msec, and hence will be modulating between 20 and 25 msec. With three voices, and the same settings, the third tap will be modulating between 40 and 45 msec.

If Delta is 0, you will get a flanging effect, as the taps move against each other and create moving comb filters. If Delta is greater than 0, but still short, the offset will create a resonant-filter effect.

Pan takes the output of the processor and pans it back and forth between the Left and Right audio outputs at a fixed rate, again using a triangle wave for modulation. Its range is 0.01 Hz (one cycle every 100 seconds) to 2 Hz. It is most effective with fewer voices — with too many voices the complexity of the sound masks the effect.

Diffusion spreads the sound out slightly at the input, to give it a little more body. (This control is not included in the Split algorithm.)

As in the DDL programs, the CLR button can be very useful in a chorus effect. When you press it, it will stop the modulation process in "mid-flange", and will set all the taps to zero, but not turn them off, so the level won't change. Therefore, a chorusing effect can be turned on and off instantaneously, and very dramatically, with the CLR button.

Samples created with the * Sample programs can be used in chorus programs, and can appear before and/or after the chorus processing. The outputs of the chorus do not appear at the aux outputs of the ADR 68K, but the (dry) samples do. When using multiple monaural samples, the "Odd/Left, Even/Right" rule, as described with the Dual Delays algorithm, applies.

Stereo Processor (Stereo)

The Stereo Processor algorithms (for indeed, there are more than one) are to be found in the factory presets in bank 9. They are used for converting monaural sound sources into stereo, using a variety of techniques and allowing for a wide range of customization. All of them feature a high degree of compatibility when collapsed back into mono, which makes them useful for a number of purposes for which other types of stereo synthesis would be unsuitable.

The Stereo algorithms use a number of delays, or taps, spread out in time and space according to certain formulas, or modes. The first fader chooses which mode to use. The **Depth** control adjusts the relative distance in time between the various taps, in some modes giving the stereo image a front-to-back dimension. The **Width** control adjusts how far off to the Left and Right outputs the taps will appear. When this control is set to 0, the outputs are identical, i.e., mono. **LF Eq** and **HF Eq** controls are shelving equalizers at about 250 Hz and 6 kHz respectively, providing up to 10 dB of cut or boost.

Each preset within this algorithm has *two* separate mono-to-stereo convertors, labelled Processor A (on page 1) and Processor B (on page 2). They do not have to be the same: Processor A, for example, can use the MSP mode, while Processor B is using COMB2. They can also be configured, with regard to the inputs and outputs, any way the user likes — the input of either convertor can be either the Left or Right input (but not both), and the output of each convertor can be sent to the main or aux outputs, or both. In addition, samples created with the * Sample programs can be used, and can appear before and/or after the stereo processing.

The Stereo modes, in the order they appear on the first fader, are as follows:

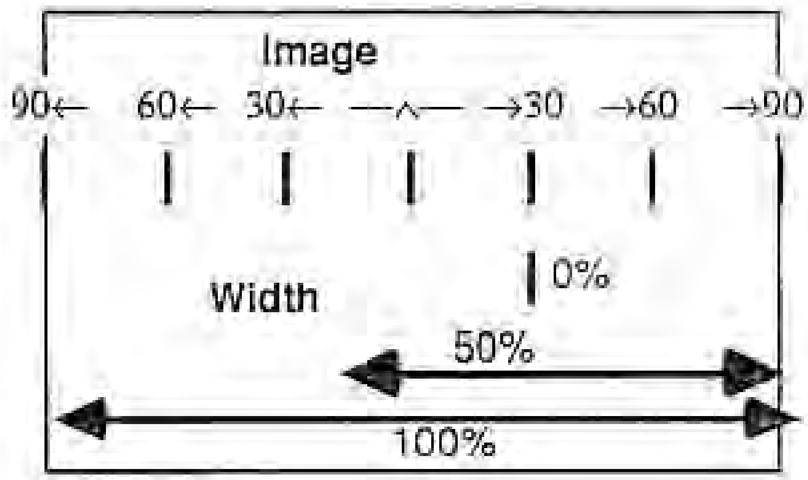
ROOM: uses relatively long delays to create the early reflections of a room or concert hall. Frequency coloration is quite low. The Width parameter is not available in this mode.

MSP: for "Multitap Stereo Processor", this mode uses shorter delays to create a very stable stereo image, which features flat frequency response at *each* of the output channels *and* the summed outputs. On highly transient sounds, if the Depth parameter is set at 50% or higher, it is possible to hear separate echoes.

COMB1: this mode uses a pair of complementary left and right comb filters. The summed mono output is flat, but the individual left and right channels are highly colored. The Depth control adjusts the time delay used to create the comb filters, while the Width control sets the depth of the filter nulls, and therefore the intensity of the effect.

COMB2: this mode is similar to COMB1, but instead of using identical delay values to create the comb filters in the left and right channels, each side has its own delay values. The stereo effect, therefore, is created by both frequency-response and arrival-time differences. This mode is the most dramatic method of stereo processing of the three available. As with MSP, larger Depth settings can create discrete echoes on transient sources. The Width parameter is not available in this mode.

PAN: uses the Haas effect to create apparent changes in image position of the source. Regardless of the apparent position of a sound, both channels are always at the same level. The Depth control changes to "Image" in this mode, and allows the sound to be placed anwhere from 90° Left ("90←") to 90° Right ("→90") of center ("—^—"). The Width control adjusts the apparent width of the sound being panned. A width of 100% means the sound takes up the entire stage, and the panning effect isn't heard. Smaller widths make the panning more dramatic, but also induce colorations.



Samples created with the * Sample programs can be used in Stereo programs. Any sampler, mono or stereo, can be used at any point in the signal path (although why one would want to stereo-ize a stereo sample is an open question...).

Multi-Effects (Multi)

The Multi-Effects algorithm ("Multi"), which is used in the factory presets in bank 8, in many ways puts all of this together. Since the ADR 68K is entirely software-controlled, it made sense to design an algorithm in which a number of processing modules are available simultaneously, leaving up to the user which ones are used, and how they work on each other.

Multi-Effects starts with a pair of independent digital delay lines (DDLs), one connected to the Left input (and the Left, or odd-numbered, samplers), and the other connected to the Right input (and even-numbered samplers). The delay on each input is adjustable from 0 to 766 ms, using a logarithmic taper. Feedback gain is adjustable up to 100% positive or negative (out of phase), and the high frequencies can be attenuated down to x0.1.

On the second page are two Gate parameters, which are adjusted the same way as the gate parameters in a reverb program — a threshold (Trig) level for when the gate will open, and a Stopped Delay time for how long the gate will remain open after the input level falls below the threshold. This gate is a "true" gate (there is no "Stopped Reverb" page), located after *all* of the processing in this algorithm, so that when it closes, all of the processed output shuts off at the main outputs. The gate has no effect on the aux outputs, however.

On the same page is a two-band **equalizer**, providing ± 10dB of shelving at approximately 250 Hz and 6 kHz. The equalizer works identically on both inputs. A bypass switch allows you to take it out of the circuit completely.

The next four pages give you control over three processors that work in parallel. Page 3 is **Stereo Chorus**, and is similar to, although somewhat simpler than, the Poly-Chorus algorithm used in bank 10. This chorus has up to four voices. The modulation depth is adjustable, using the Range parameter, from 1 to 16 ms, in .1 ms increments. The speed of modulation, instead of being in Hertz, is expressed by the Speed parameter in terms of percentages, from 0% to 100%, with lower values resulting in longer cycle times. The Mode parameter lets you determine whether the taps are moving parallel or contrary to each other. The chorus has a bandwidth filter, which can be useful for minimizing "clicking" noises that are sometimes generated when high frequencies are being processed. There is also a mute switch that takes it out of the circuit.

Next is the **Stereo Multi-Tap** page, which generates a series of up to eight discrete echoes, and lets you control them as to time, amplitude, and stereo placement. There are nine echo patterns, or **Modes**, to choose from.

"Echo 1" produces up to four echoes, evenly spaced. "Echo 2" produces up to four echoes, with the time between them increasing. "Echo 3" produces up to four echoes with the time between them decreasing (like a bouncing ball).

"L←→R" generates up to *eight* echoes, placing the first in the Left channel, the second in the Right, the third in the Left, etc. "R←→L" does the opposite, placing the first echo in the Right channel, etc. "R→L" generates up to *five* echoes, and pans them across the stereo field from right to left. "L→R" pans the echoes from right to left.

"R,L" only generates even numbers of echoes, up to eight. The echoes are broken up into two equal groups, with the first half coming out of Right output followed by the second half coming out of the Left output. "L,R" reverses the process. To keep all this straight, think of R→L as "Right to Left" and R,L as "Right, then Left".

The number of echoes is controlled by the "# Taps" parameter. The "Envlp" parameter determines the volume envelope of the echoes, whether they increase over time, decrease, or stay the same. The amount of increase or decrease is approximately 6 dB per echo. "Length" controls the amount of time the entire series of echoes will take (*not* the time between individual echoes), up to 2 seconds. Again, this page has a low-pass filter and a mute switch.

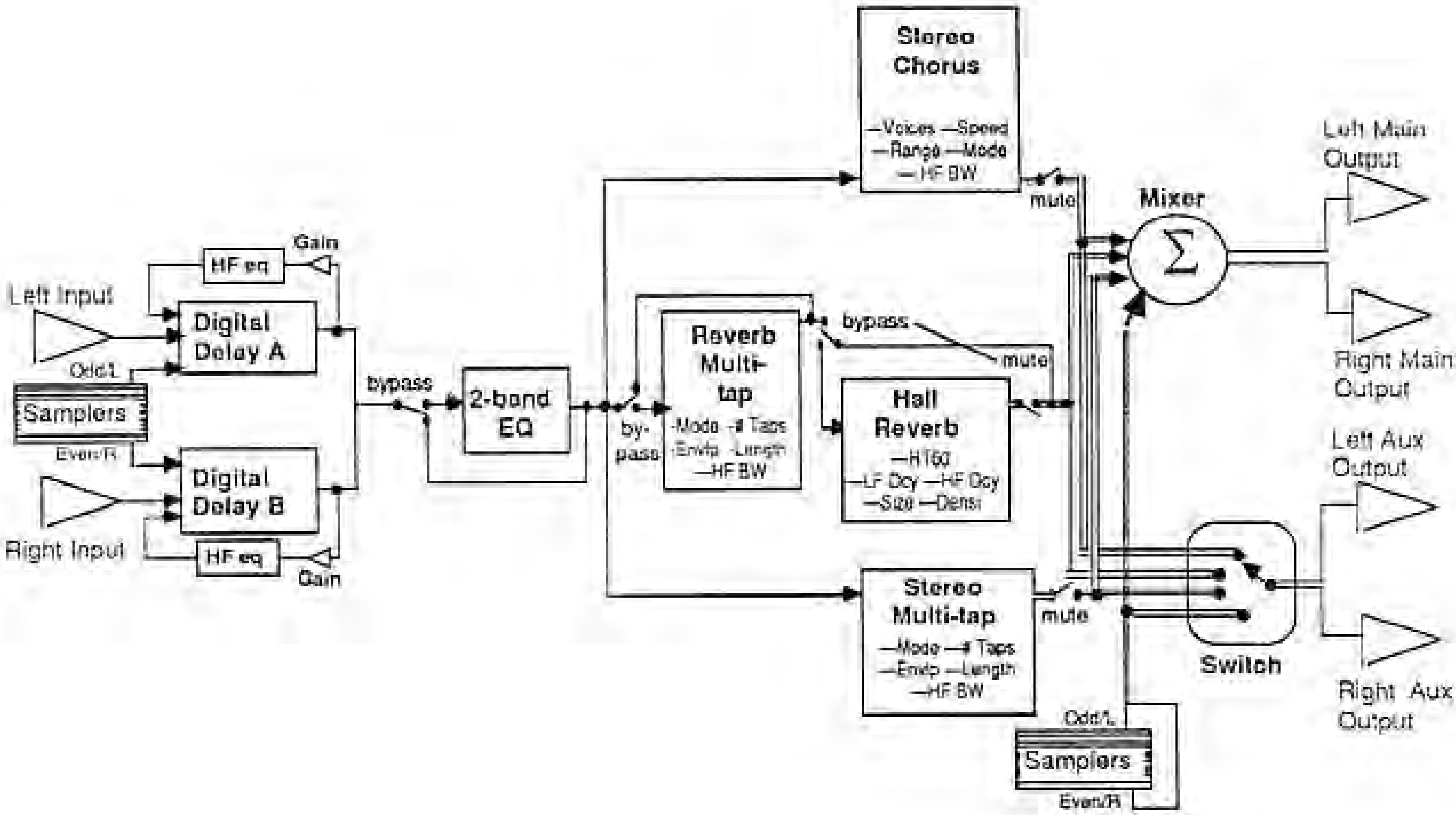
The next page is also a multi-tap echo generator, but it is designed to produce early reflections that will be used by the reverb processor on the next page, and so it is called "**Reverb Multi-Tap**". The controls are the same as on the previous page, with the following exceptions: only the three Echo modes are available, and these can produce up to six echoes, not four. A fourth mode, "P'Dly", produces a single echo. This page has an adjustable low-pass filter, which affects both these taps and the reverb processor that follows, and a bypass (not a mute) switch.

The **Reverb** page is functionally equivalent to the "Hall" side of a preset using the Split algorithm, except that it has a Size control. The other controls — RT60, LF Dcy, HF Dcy, and Densi — behave as they do in all other reverb programs. A final control on the page switches the reverb on, or mutes the output, *or* bypasses it, so that the early reflections produced on the previous page can get through.

The last two pages are the input and output mixers. There is no mixer for the aux outputs. Instead, the right-most control on page 8 acts as a multi-position switch, which allows you to choose whether one (and only one) of the effects — Reverb, Chorus, or Stereo Multi-Tap — or the dry samples will go to the aux outputs.

Samples created with the * Sample programs can be used in Multi-Effects programs, and can appear before the ddls and/or after all of the processing.

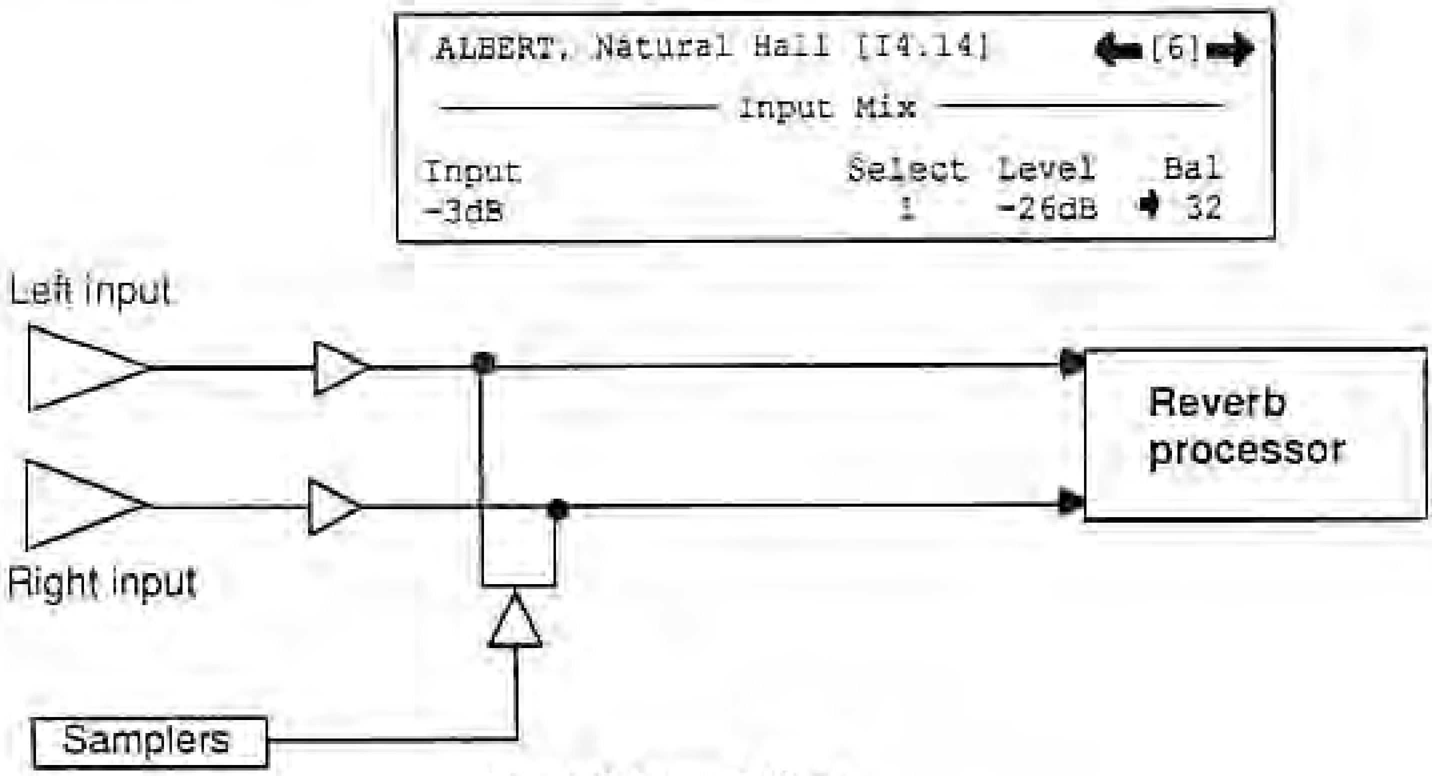
Here is a diagram of the Multi-Effects processors and their relationship to each other:



Mixing Inputs and Outputs

With so many different signals coming in and out of the ADR 68K, it's essential that their routings and relative levels be carefully controlled. The ADR 68K handles all of this in software, and offers several pages of input and output mixers for each program.

Each algorithm has its own mixer configuration, tailored for that algorithm, but all of the mixers include the same general design elements and features. Let's take, as an example, the mixers from a reverb algorithm. Here is the input mixer, which appears on Page 6:



Typical Input Mixer

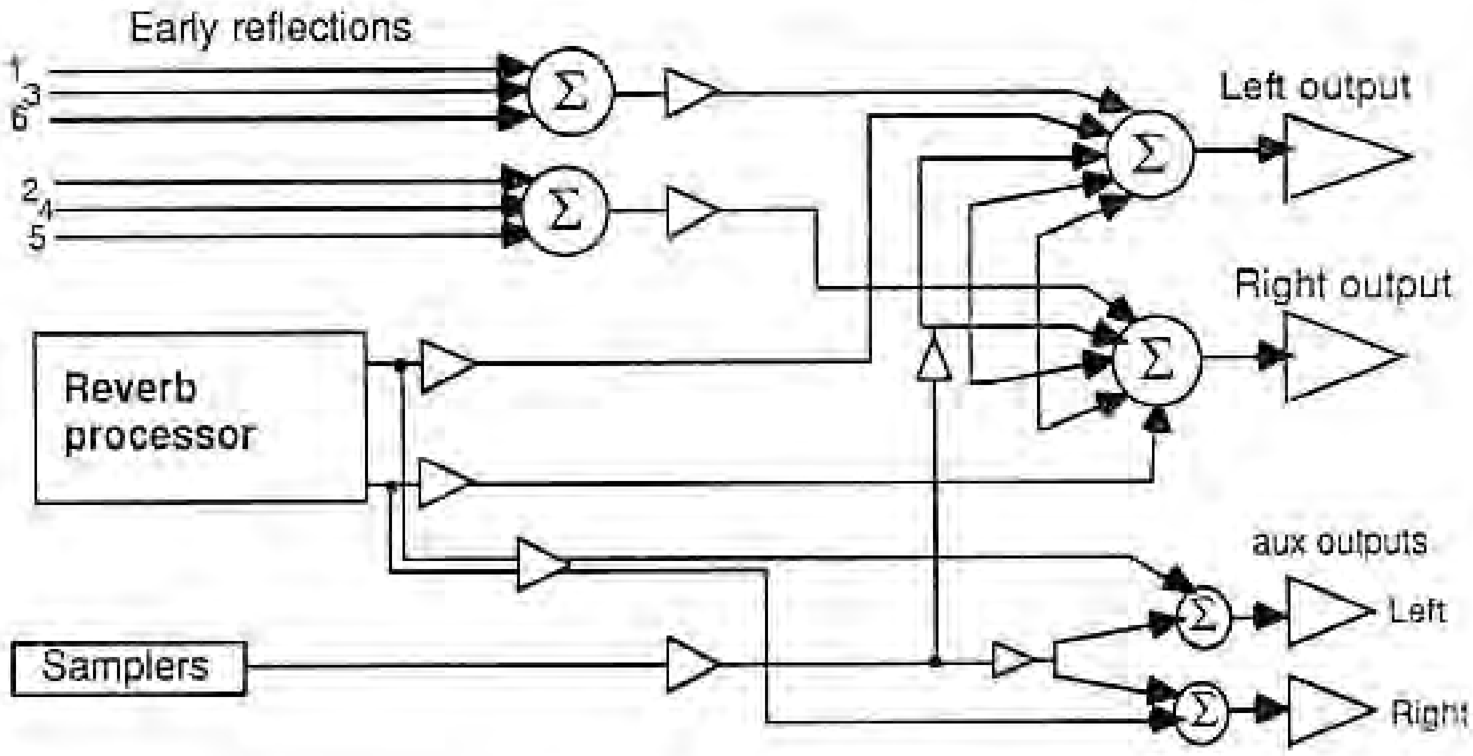
This mixer is electronically located before the reverberation processor. The Input parameter controls the attenuation, if any, between the ADR 68K's audio inputs and the reverberator. The maximum value is 0 dB. The input path is true stereo. You can actually turn the Input OFF so that the ADR 68K's inputs never reach the reverberator. This has it uses, as will be seen in the discussion of samples in the next chapter. Other algorithms, like Splits and Multi-Effects, have more faders on the input page for the various sections of the program.

The "Select", "Level", and "Balance" parameters deal with samples coming into the reverberator, and will be covered in the next chapter.

Here are the output mixers for the same algorithm. The main and auxiliary outputs each have their own mixer, located on Pages 7 and 8:

```
ALBERT, Natural Hall [I4.16]          ⬅[7]➡
———————————— Main Output Mix ————————————
E-R's Rev'b               Select  Level    Bal
-6dB  0dB                    1    -26dB  ➧ 80
```

```
ALBERT, Natural Hall [I4.16]          ⬅[8]
—————————— Aux Output Mix ——————————
 Rev'b   Select  Level
 0dB        2    -12dB
```



Typical Output Mixer

The mixer for the main outputs is stereo. The first fader is a master controller for the six stereo Early Reflections (E-R's) of the signal, whose *relative* levels are adjusted on Pages 4 and 5. The second fader controls the stereo sound coming from the reverb processor. The three sampler controls on this page control the dry (unprocessed) sampler signals from the samplers at the outputs. The mixer page for the aux outputs contains a stereo fader for the output of the reverberator, and also sampler controls. Again, the more complex algorithms have somewhat different output-mixing pages. Look through the presets and at the program charts in Appendix C for details.

The mixers give the ADR 68K a great deal of flexibility. Within many of the reverb programs, the various components of the reverb, i.e. the early reflections and the main reverb, can be split off to two different sets of outputs, by setting the controls like this:

```
 MONTY, Natural Hall [I4.17]        ← [7] →
 ————————————— Main Output Mix —————————————
E-R's Rev'b                 Select  Level    Bal
0dB   OFF                      1     OFF   ▸ 80
```

```
 MONTY, Natural Hall [I4.17]        ← [8]
   ——————————— Aux Output Mix ———————————
  Rev'b   Select  Level
  0dB        1      OFF
```

In the more complex programs, the mixers are essential for balancing and keeping track of the various components of the sound. In the Split programs, the mixing pages allow the two stereo reverb outputs to be sent to the outputs separately (i.e., Plate→Main, Hall→Aux), or if you are not using the aux outputs, you can mix them both into the main outputs.

In the Multi-effects programs, there is no mixer for the aux outputs. Instead, the right-most control on the last page acts as a multi-position switch, which allows you to choose whether one (and only one) of the effects — Reverb, Chorus, or Stereo Multi-Tap — or the dry samples will go to the aux outputs.

```
 ZOUNDS, Multi-Effects [I8.12]      ← [8]
 ——————— Main Output Mixer———————    —Aux—
Chorus  Mtap   Reverb  Select Level  Source
 -6dB   -21dB   OFF      1     OFF   Reverb
```

— Chapter 6 — Using Samples

Samples can be generally defined as discrete audio events to be played back on demand. They are handled by the ADR 68K in a number of ways: they can be played just as they were recorded or they can be edited; they can sound by themselves or in conjunction with reverb or effects; they can be played back at their original pitch or a different one; and they can be triggered manually, remotely, and/or automatically.

The ADR 68K has enough memory to store more than 32 seconds of sound, configured either as one large sample or as several smaller ones, either in mono or stereo. The device uses 16-bit linear coding for the best possible dynamic range, and a bandwidth of 15 kHz.

Samples are recorded into the ADR 68K using the sampling programs (or presets) normally located in Factory bank 6, although it is possible to move those programs into other banks (more on this later). They can be played back from controls within those programs, from controls in other (reverb and effects) presets, or from various external triggers including MIDI. Samples reside in volatile RAM, and so they normally disappear when the power is turned off, but they can be stored off-line in a number of ways.

The sampling programs fit into three categories:

1) programs that let you work with only one long sample at a time, and allow pitch change on playback,

2) programs that allow multiple samples of variable length,

3) programs that allow multiple samples to be used with the reverb and effects programs in other banks.

In each of these categories there is one mono and one stereo sampling program. Obviously, stereo samples take up twice as much room as mono ones, so within each category the stereo samples are either shorter or there are fewer of them available. Programs in which the samples are not "portable" to reverb and effects programs allow for longer sample storage, because memory does not have to be reserved for the reverb or effects processing.

The sampling programs and their major characteristics are as follows:

Name	register	maximum # of playable samples	inputs	outputs	maximum length each sample
Mono*	F6.1	12	mono (L or R)	main & aux†	2.04 seconds
Ster*	F6.2	6	stereo	main & aux†	2.04 seconds
M	F6.3	1	mono	main only	33.5 seconds§
St	F6.4	1	stereo	main only	16.8 seconds§
M-12	F6.5	12	mono	main & aux	variable
St-6	F6.6	6	stereo	main & aux	variable

** samples recorded with these two programs can be used in reverb and effects programs*

† the aux outputs are mixed in the reverb or effects programs that use the samples created with these programs, not in these programs themselves

§ samples recorded with these programs can be played back at a different pitch

Normally speaking, samples recorded with one program cannot be "ported" to another, and if you have in memory a sample recorded with one sampling program and then call up a different sampling program, the information you've recorded will be lost. In addition, samples recorded with the "non-portable" sampling programs (the ones that don't have a "*" in their name) will be lost whenever you leave the sampling program, either to go to a different sampling program or to a reverb or effects program. There are exceptions to this rule, which we'll explain later in this chapter.

Recording a sample

Using the numeric keypad, call up Factory register 6.1, the Mono* sampling program. Move the "Select" fader on Page 1 to "1". Then, with the "Source" fader, choose whether the signal you want to sample is coming into the Left input or the Right input. Because the ADR 68K has no user-accessible input-level control, adjust the level of the incoming signal at its source, watching the LED indicators on the ADR 68K's remote control for maximum level without distortion. If Safety is ON, use the fader or Soft Button immediately underneath it to turn it OFF.

```
Mono*, Efx-12-Sampler [F6.1]            [1]→
———————— Record Functions ————————
Select  Source  Safety  Rec     Stop   Play
  1      Left    OFF   -18dbB          ——
```

You can now either start the recording manually, or have it trigger automatically. If you want to start the recording manually, set the fader under the label "Record" to read "MAN". Then, when you are ready, push the Soft Button above the fader, and start the sound.

To trigger recording automatically, look at the LCD level display to see if there is any residual input noise registering, and set the Record fader to the next increment above the noise level. For example, if there is room noise that is peaking at around -30 dB, set the Record fader to -24 dB. If the noise is uneven, you may have to put up with a couple of false starts (in which the system starts recording by itself) while you adjust the level. When you have a good setting, start the sound. The recording will start as soon as the signal level reaches the threshold you have set.

Once the recording starts, the display will start to count down from the maximum time available for the sample, which in this case is 2.0 seconds. When the countdown reaches zero, recording will stop, and Safety will reset to ON. You can stop recording before the countdown gets to zero by pushing the Soft Button labelled "Stop".

You can now play back the sample by pressing the Soft Button labelled "Play". You can stop the playback by pressing "Stop". If you like, you can hear the sample over and over by moving the Play fader to RPT, and you can stop it playing by pressing Stop.

If the recording is unsatisfactory, turn off Safety and try it again.

There are minor differences in recording procedures among the various sampler programs. In stereo programs, there will be no Source fader, because both inputs are active. In the "M" and "St" programs, there will be no Select fader, as you can only deal with one sample at a time. Also in those programs, the recording countdown will start at about 16.7 or 33.5 seconds, because that is the amount of time in the sampler memory. In the M-12 and St-6 programs, the countdown time will be different depending on the number of samples in memory. We'll cover this in detail later on.

Editing a sample

Press the Play button. The sample you have just recorded will play back, and a "countup" will appear above the Play button. You can stop the sound at any time by pressing Stop.

Now Go to page 2 by pressing →. Make sure the Select fader is still on"1". The second and fourth faders on this page, marked "Begin" and "End" are used to find the precise beginning and ending point of the sound that you want to hear within the sample memory. The faders work differently on this page than anywhere else in the ADR 68K's operation. Instead of setting them to an absolute position, the faders are used to increase or decrease the beginning and end points in a *relative* manner.

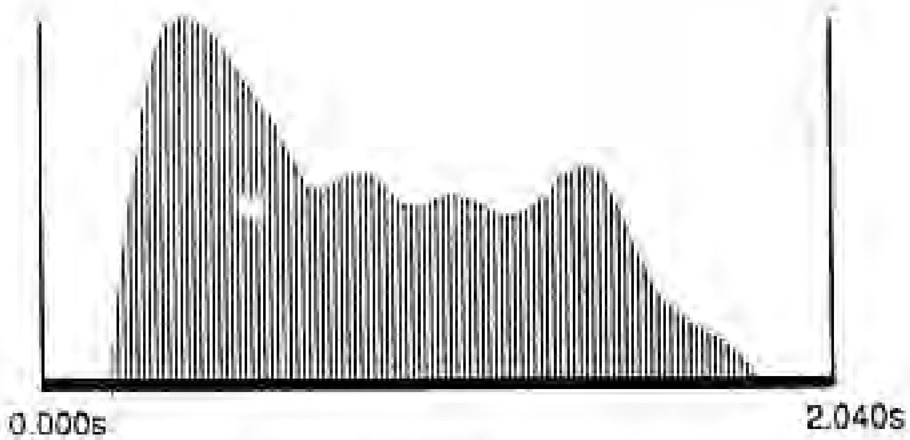
Here's how you use the "Begin" fader to find the beginning of a sound. When you first come to this page, the value above the fader will read "0.000". Move the fader upwards, and as you pass the center point, you will see the numbers start to change. You will also start to hear 20 millisecond-long "slices" of the sample, playing at a rate somewhat slower than what they were recorded at. In other words, at the moment the display reads "0.760", you are hearing the section of the sample from 760 to 780 ms after the beginning of the sample memory. Note that the position of the fader does not directly correspond with the value of the numbers — instead the fader position controls the *rate of change* of the numbers: as you move the fader further up, the numbers change faster, and the slices of the sample come more rapidly.

When you find the beginning of the recorded sound (if you have used automatic recording, this should happen very quickly), move the fader back to the center position, and the numbers will stop changing, and the sample slices will stop sounding. You will probably have gone past the beginning of the sound, so in order to get the Begin value to go down, move the fader slightly *below* the center point, and you will hear the sample start to play "backwards". (It's not really playing backwards — the 20ms slices are playing in the right direction, but the slices themselves are coming out *in reverse order*.) Again, the distance of the fader below the center point will determine how fast the Begin value will change and how fast the sample slices will be played.

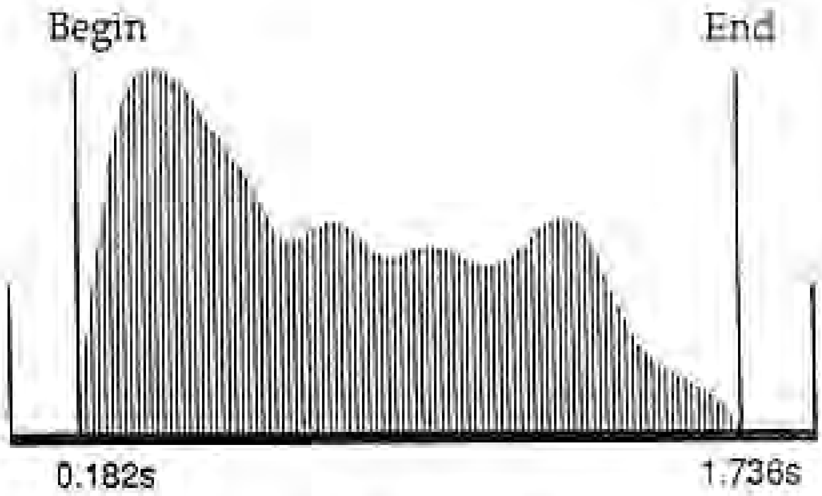
When you again hear where the sound begins, move the fader back to the center. If you can't quite get the motion of the numbers to stop precisely where you want, you can use the "trim" fader immediately to the right of the Begin fader. The trim fader increments the Begin value in 1-ms steps, and as you move it, you will hear 1-ms slices of the sample. The trim fader works like a normal ADR 68K fader, in that its value is fixed to its position. Its center point is zero, and it has a range of ±63 ms.

Note that when you are moving the faders, the slices of the samples you hear are being played at the same *pitch* they were recorded at — in other words, each 20-millisecond slice takes exactly 20 milliseconds to sound. This makes it very easy to hear exactly where in the sample you are, because you don't have to identify a sound that has been transposed by two octaves or more. What does change as you move the fader is *how fast* those slices will be played one after another. At the slowest rate, when the fader is closest to the center, the sample plays at about 1/25th normal speed. As you move the fader further away from the center, you will find a point at which the sample plays back at normal speed. You can go past that point, and at the extreme position of the fader, the playback rate is about four times normal.

Setting the end point of the sample is done exactly the same way, using the "End" fader. As you move the End fader, 20-millsecond slices of the sample will play, *starting* at the point indicated as the End value, and you should clearly hear the end of the sample give way to silence. You can fine-tune the end point with the trim control next to the End fader. Note that because you are listening to a sample slice that *starts* at the End value, once you find the proper end point, if you want to be really precise, you might want to adjust the End value with the trim control so that it reads 20 milliseconds *below* the current value.



Original Sample



Sample after Editing

If you get confused during this process, and lose track of where the beginning and ending of the sample are, you can go back to Page 1 and press the Play button, and observe the "countup" as the sample plays. Watching a moving display while the sound plays is a good way to help you get oriented as to what happens when.

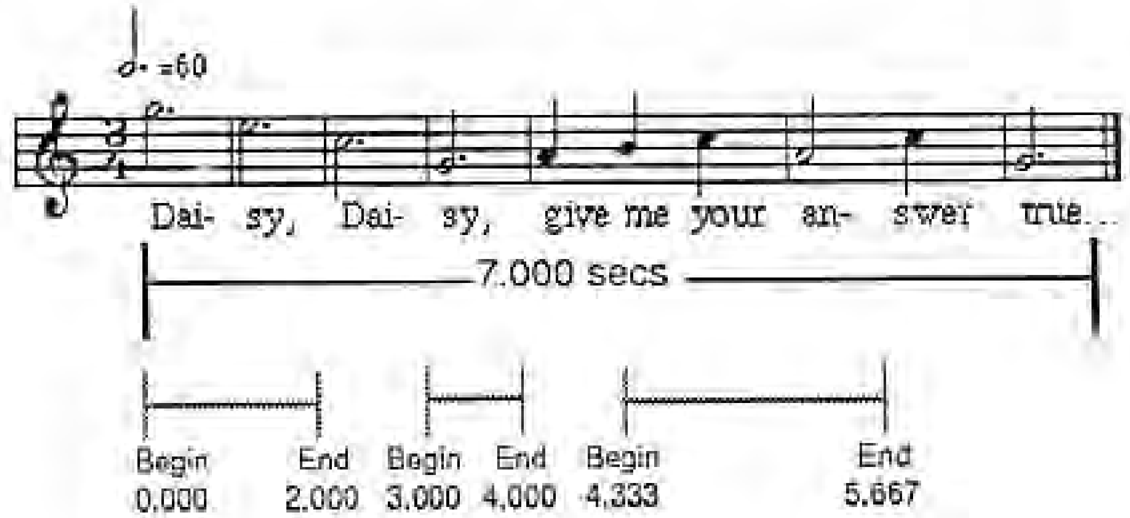
At all times during the editing process, even while the values are changing, the Play button is active, so you can check your editing "in context". The sound you hear will reflect the current Begin and End values at the precise moment you hit the button. (You can also hear the sample by pressing the Select button.) You can even put the Play fader into RPT mode, which can lead to some interesting effects.

The Soft Buttons above the various faders are active during the editing process, although, like the faders, their function is a little different. Pressing the Soft Button above the Begin or End fader immediately sets the fader value to the end of the sample register, e.g., 2.096, 33.524, etc., while pressing Shift-Soft sets the value to the beginning of the sample, i.e., zero.

The Soft Buttons above the trim controls behave much more conventionally: pressing one of them increments the appropriate value by 1 ms, and pressing Shift-Soft decrements the value by 1 ms. Note that these Soft Buttons have both "auto-repeat" and "accelerating" characteristics: if you hold one of them down, it will continue to increment (or decrement) the starting point, and if you continue to hold it, the *rate* of change will increase.

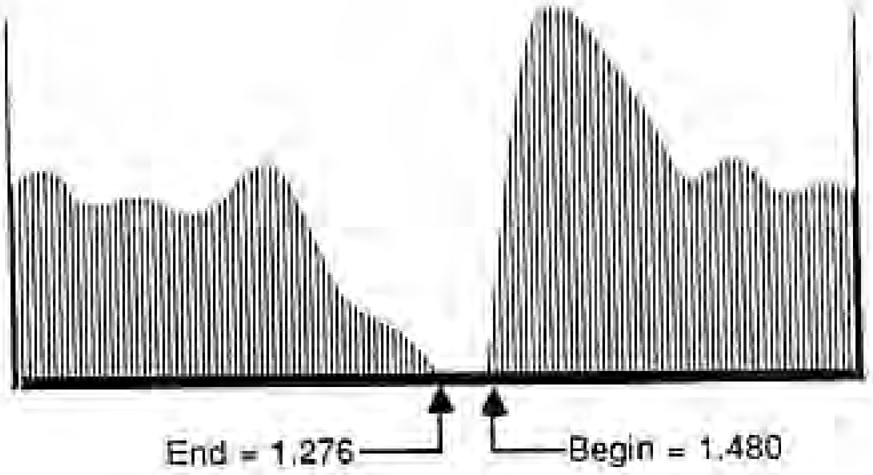
Once you set Begin and End points for a sample, those points stay with the sample until you specifically reset them or erase the sample (e.g., by going to another sampler program). When you edit a sample, the parts you edit out are not lost — they stay in memory as long as the power stays on (or until you record over them), and they can be retrieved at any time simply by moving the Begin and End points apart, so that the sounds are once more included in the edit "window". If you are using a portable sampling program, you can even leave the program and go to a reverb or effects preset, then come back to the sampling program and re-edit the sample(s).

Sample editing does not have to be limited merely to clipping silent ends off. You can also extract a single note from a musical line, or a single syllable from a spoken phrase. Because the edited portions are not lost, you can record a long phrase in one of the longer samplers, and play back different portions of it at different times, using the Begin and End faders to determine the current "window".



Once you set the Begin and End faders for a sample and you re-record it, the actual section of memory the new sample will occupy will be the section defined by the Begin and End values. Therefore, if you start with a 6-second sample, then edit it down to 4 seconds, and *then* want to re-record it, you'll find that when you go into Record, the countdown will start at 4.0 seconds. If you want to record over the *entire* original sample, you will have to set the Begin and End faders back to their original values.

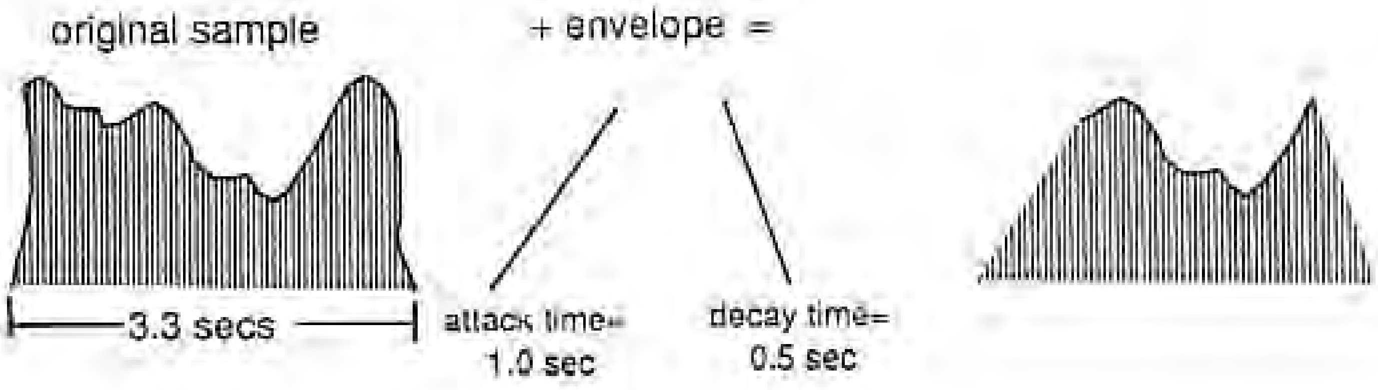
You will notice an interesting effect if the Begin value is higher than the End value: you will hear the end of the sample first, and then the beginning. This is not "backwards sampling" — the unit is simply "wrapping" the sample around itself. Although this feature is not one we expect will be employed frequently, it does have its uses, such as when two words in a spoken phrase need to be reversed.



Sample "Wrapped" by setting Begin higher than End

Sample envelopes

Besides adjusting the Begin and End points of a sample, you can also impose a volume envelope on it, similar to an envelope generated by a synthesizer, to allow for smoother attack and decay of the sound. On Page 3 or 4 of all of the sampler programs is a group of three faders marked Select, Attack, and Decay. Select chooses the sample to work on (if there are more than one), and Attack and Decay do just what they say. The attack and decay curves are linear, with a range of 0 to 970 milliseconds.



Modifying a Sample with
the Attack and Decay faders

The envelopes are linked to the Begin and End points, so that moving the Begin time will alter the point within the original sample at which the attack starts, but the slope of the attack will not be changed. If an attack time is longer than the sample length itself, then the sample will never reach peak volume. The decay time *cannot* be set longer than the length of the sample. If you set a decay time and then edit the sample so that it is shorter than the decay time, the decay time will automatically change to fit.

If you stop the playback of a sample that has a decay envelope on it, the sample will not stop immediately, but instead the decay envelope will take over. This is explained more fully a little later on.

Playing back samples

Manual Playback

The simplest way to play back samples recorded in the ADR 68K is from the sampler presets themselves. If there is more than one sample in memory, first choose which sample to play by moving the Select fader, and then starting the playback by pressing either the Select or the Play button. (Note that these controls appear on every page of the sampler presets.) If there is no sample recorded in a selected memory then the Play button for that memory will be disabled.

As mentioned above, a sample can be set to repeat endlessly by selecting it and then moving the Play fader to RPT. A sample playing this way will keep going until the Play fader is moved or until you press the Stop button (which only appears on Page 1), the CLR button, or Shift and the Play button.

When samples are recorded using the portable sampler programs, then they can also be played manually from within reverb and effects programs, using the Select and Play faders and buttons that appear in those programs (often on more than one page).

But the ADR 68K would be a very dull device if it did not allow sample triggering more sophisticated than this. Therefore, there are several other ways to play back samples, using information external to the unit itself.

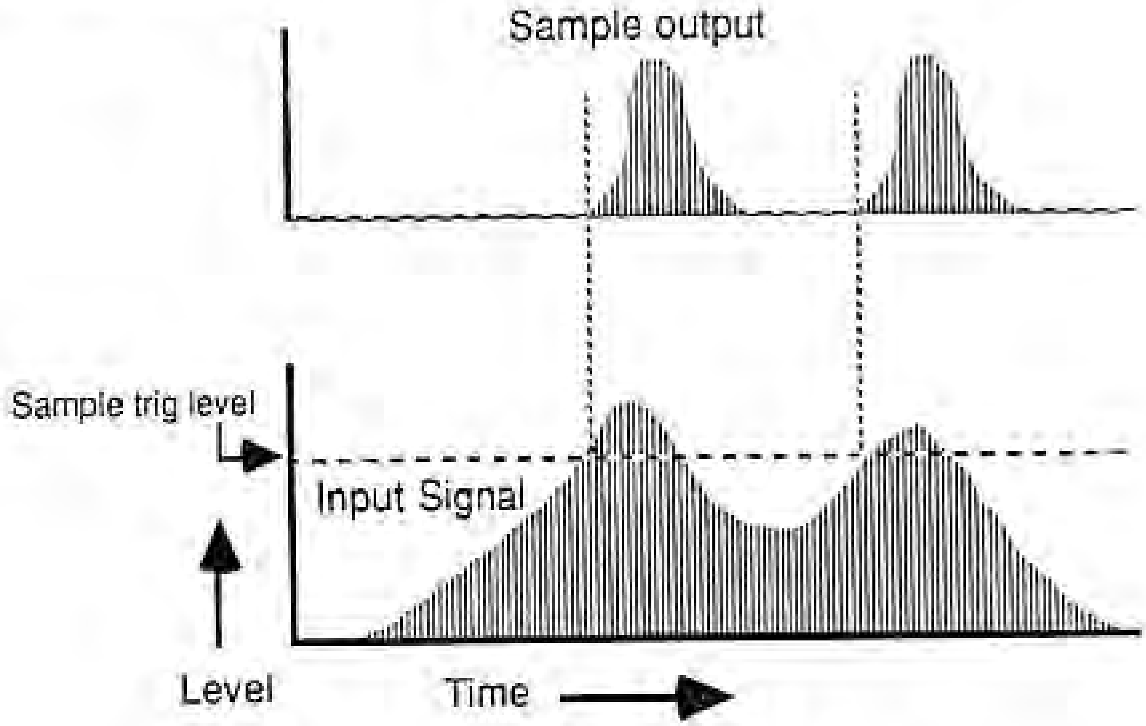
Playback from Audio Triggers

The use of sampled sounds to replace "live" sounds — for example, having a car crash occur every time the drummer hits his third floor tom-tom — is a very popular practice. It is often accomplished by allowing a sampled sound to be triggered in real time by an audio signal, either played live or on tape.

In the ADR 68K, *any* recorded sample can be set to respond to an input trigger. Let's try triggering the sample you've just recorded. Go back to Page 1 of the Mono* preset, and set the Play fader to "-12db". Send a signal into the Left input of the ADR 68K. As soon as that signal level reaches -12 dB (as indicated on the LED display on the remote control), the sample will sound. It will not stop sounding if the signal level drops below -12dB, but will continue until it has played through. You can stop it manually at any time, however, by pressing Stop, CLR, or Shift-Play.

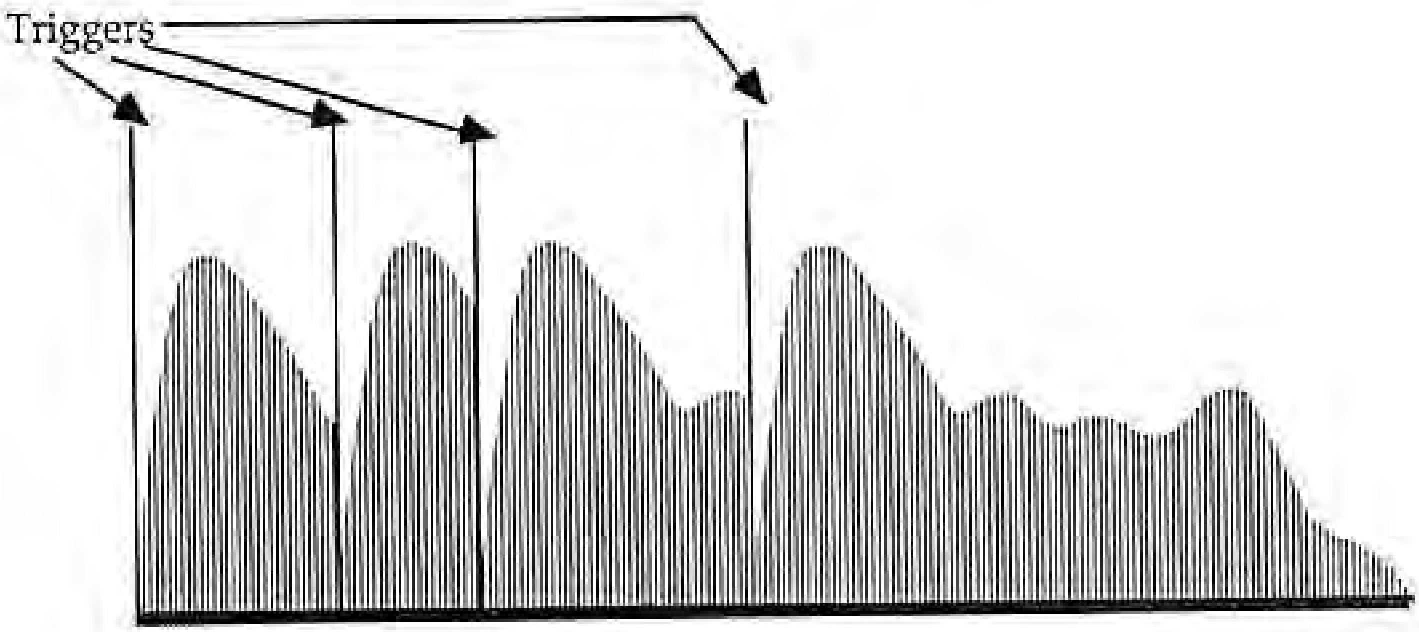
The Play fader on any page of the sampler preset can be used to set the input trigger level. (Note that there is only one Play fader per sample — it just appears in more than one place. If you set it on one page, that setting will remain when you change pages.) Setting the fader to -42 dB means that almost every time a signal comes into the ADR 68K, the sample will trigger, while setting it to "LIM" means only the very loudest input signals will trigger the sample. The trigger levels all correspond exactly to the levels shown on the LEDs of the remote.

The sample will be played when the input level crosses the threshold in an *upward* direction. If the input level falls below the threshold, and then goes back up above it, the sample will trigger again.



Triggering a Sample from an Audio Source

If the sample is still playing from the first trigger, it will be interrupted. Rapid changes in the input signal will produce the popular "stuttering" effect.



Sample being re-triggered before it finishes

A sampler will respond to an input trigger on only one input channel. The Source fader on Page 1 of the sampler preset will determine which channel will trigger which sampler. If a sample has been recorded with the Source fader set to Left, i.e., from the Left input, then it can only be triggered from a signal at the Left input. *However*, this can be changed after the recording is made by moving the fader, which means that the fader essentially has two independent functions. Therefore, for example, a sound can be recorded from the Right input into sampler #4, and then the Source fader for sampler #4 can be reset to "Left", so that only signals at the left input will trigger the sample playback.

The exception to the above rule is when you are using the *stereo* samples. In that case, odd-numbered samples (1, 3, and 5) can only be triggered from audio at the Left input, while even-numbered samples (2, 4, and 6) can only be triggered from the Right input. In the "St" program, there is only one stereo sample to be triggered, and it reads the left input.

Except in the portable samplers, any number of stored samples can be made to sound simultaneously. This has all sorts of creative possibilities, especially under MIDI control, which we'll get into in the next chapter.

If you press the Mute button, the inputs triggering the Samplers are disabled for as long as you hold the button. You can latch the button by pressing Shift-Mute.

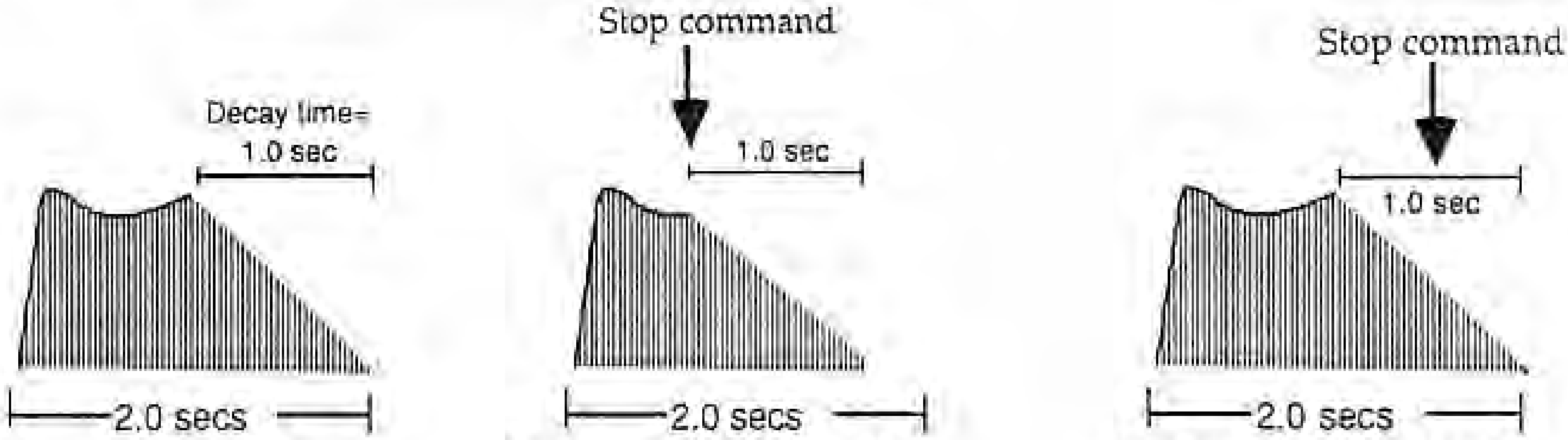
Playback from MIDI and other Triggers

Samples can also be triggered from MIDI sources, like a keystroke on a keyboard, a press of a MIDI pedal, or a note coming from a MIDI sequencer. In addition, they can be triggered from other electrical sources, like a footswitch, potentiometer, or synthesizer control voltage, using the 1/4-inch jacks on the back of the ADR 68K's remote control. These functions are set up from the MIDI menu, which is accessed by pressing the "MIDI" button on the remote control. Using this menu will be discussed in the next chapter.

As noted above with input triggering, if before a sample finishes playing, it is retriggered in *any* manner from *any* source, the playback will be interrupted and the sample will start again.

Stopping playback

Playback of a sample can be halted at any time by pressing the Stop button on the first page of the Sampler program; by pressing Shift-Play or Shift-Select on any page in any program; or by external or MIDI control. Normally, when the sample is stopped this way, it stops playing immediately. However, if the sample has a Decay envelope, stopping the sample instead starts the Decay envelope, so the sample dies away instead of ceasing abruptly. This does *not* happen, however, if the Stop command occurs after the Decay envelope *has already started* by itself. In other words, if you have a two-second sample with a decay envelope of one second, if you tell the sample to Stop 1.5 seconds after it has started, the Decay envelope will already have started, so the Stop command will be ignored.



Original sample with Decay envelope

Stop command received before Decay starts shortens sample

Stop command received after Decay starts has no effect

Setting Sample Output Levels

The last page or two of each of the sampler presets handles output mixing. Here you can set the levels and balances of the various samplers and the various outputs. Each sampler (if you are using more than one) has its own level control for the main outputs, which is accessed with the Select fader on the mixing page (if you are using a sampler preset with multiple samples). If the sampler is mono, there will also be a balance control, which reads from hard left ("←100"), to center ("—^—"), to hard right ("100→"). In addition, the M-12 and St-6 programs include a separate set of controls for the aux outputs, so that you have a choice of four destinations for each sampler.

As noted above, you can mix the the portable samplers into the aux outputs, but that function is handled within the appropriate reverb or effects program, not within the sampler program itself. Also, in some programs, there will be no balance controls available when mixing samples into the aux outputs, in which case odd-numbered samples will appear at the Left aux output, and even numbered samples at the Right. See Appendix C for details on how each particular program handles the auxiliary outputs.

When you are using portable samplers in other programs, any level and/or balance settings created within the sampler presets are ported over to the reverb or effects programs along with the samplers, and appear on the output mixing page of the reverb or effects program. Conversely, if changes to the main sample output levels are made within a reverb or effects program, they will show up in the *sample* preset when it is recalled.

Sample mixing parameters are not stored with any preset, but stay active as long as the sample is intact. There is only one set of "master" controls for each sampler, and it stays the same whether you change from a reverb program back to the sampler program, or go from one reverb or effects program to another. More details on using the output mixers with samples will be given later in this section.

Note that the level and balance controls can be moved while the sample is playing, and will affect the level and/or stereo placement of the signal in real time.

Pitch-Changing Samples

Two of the sampler presets,"M" (register F6.3) and "St" (register F6.4) allow the sample to be played back at a different pitch than the one at which it was recorded, up to one octave above or below the original pitch. This function is handled on Page 3 of each preset.

```
M, Mono 32sec Sampler [F6.3]            ⬅(3)➡
—— Tuning ——

Method  Amount   Fine                 Play
  %      -32%    + 8                  MAN
```

The first fader on the page determines how the pitch change will be displayed: as a percentage, in musical half-steps, or as a musical interval. The second fader determines the "Amount" of change. In the "%" Method, the resolution of the Amount fader is 2% going positively and 1% going negatively (there is also an extra-large "0%" area on the fader, to allow easy centering). In the other two Methods, the resolution is a half-step (or minor 2nd) in both directions.

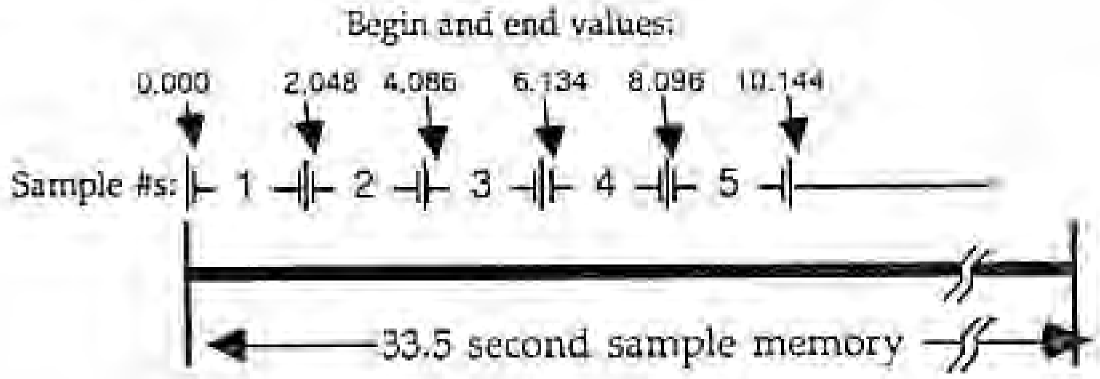
The third fader, "Fine", is calibrated in cents, or 1/100ths of a half-step (1200 cents = one octave), with a range of ±200 cents (a whole step or Major 2nd) and a resolution of 4 cents. Note that the maximum pitch change in any direction is always 1 octave, so if the Amount fader, for example, is set to +100%, positive values of the Fine fader will have no effect.

By the way, there is a minor inconsistency in the way the ADR 68K handles samples whose pitch is being drastically changed. At extreme settings of the tuning parameters, the output of any sampler destined for the left channel (that is, a mono sample whose balance is set to ←100 or the left side of a stereo sample) drops off somewhat, while the output destined for the right channel (a mono sample balanced to 100→ or the right side of a stereo sample), migrates over to the left audio output. This effect is less pronounced on the Aux outputs than on the main outputs.

Multiple Samples

The ADR 68K lets you work with up to 12 samples at the same time, depending on the program. This section will deal primarily with recording and editing multiple samples. Triggering and playing back more than two samples at a time is a job for MIDI, and so that subject will be covered in the next chapter. There are restrictions on the number of samples that can be triggered within a reverb or effects program, which are outlined later in this chapter.

Recording multiple samples into the portable sampler programs, Mono* and Ster*, is a straightforward procedure, similar to recording single samples. The length of each sample is fixed at 2.04 seconds, and when you access a particular numbered sampler with the Select fader, you always are working with the same segment of memory. You can shorten a sample as it plays, by moving the Begin point up and/or the End point down, but you can't lengthen it or make it overlap another sample.



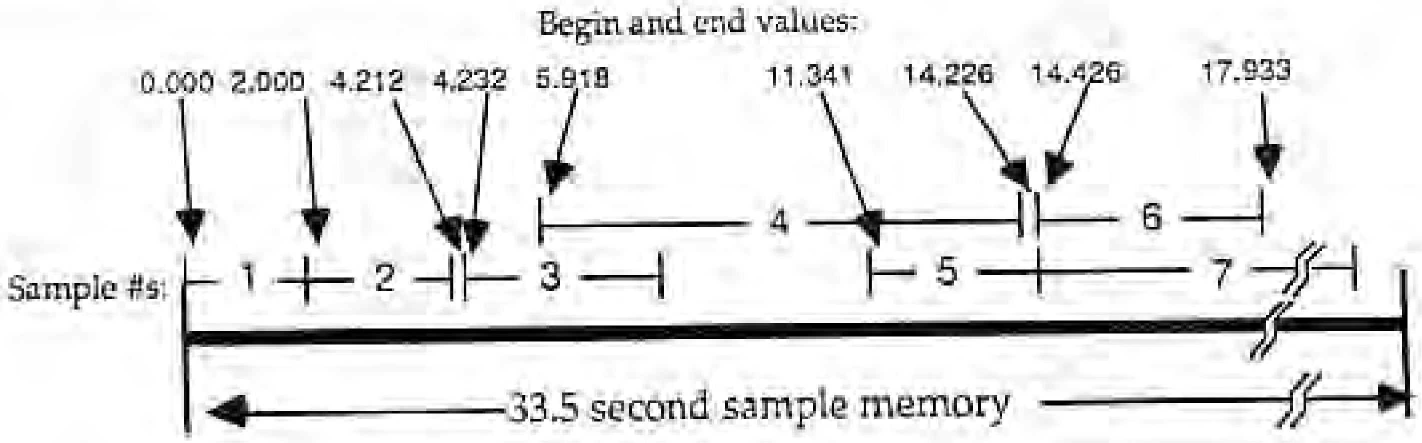
Portable ("*") Sampler Programs have fixed memory segments

However, there are far fewer restrictions on the other sampler programs. These programs perform the functions of what is often known as a "window recorder", by giving you a large amount of memory to record sound into, and then letting you access any portion of that memory you like.

The programs M-12 and St-6 are multiple-window recorders, in which each window can be resized. When you record samples into one of these programs, initial values for all of the windows are automatically calculated based on the length of each recorded segment. Here's how this is accomplished in preset M-12:

When you record into sampler 1, you have a blank 33.5-second memory available. When you start recording, the display starts counting down from 33.5 seconds. Let's say you press Stop after 6 seconds. Now if you move the Select fader to sampler 2 and start recording, the countdown will start at 27.5 seconds, which is equal to 33.5 *minus* the amount of time occupied by the first sample. If you go to the edit page and move Select to sampler 2, you will notice that the Begin time no longer reads "0.000", but instead has set itself to about 6 seconds. (In actuality, the ADR 68K sets up a 20-millisecond blank "buffer" between samples recorded this way, so the new Begin time will be the precise moment you hit the Stop button, plus 20 msec.)

Keep recording samples 3, 4, 5, etc., and the countdown will keep starting at a lower value as you gradually use up the available memory. Every time you go to Page 2 to edit a sample, the Begin value for that sample will reflect where in the memory you started recording. (Note, however, that the end times do not change automatically, but instead stay at the maximum value.)



Other multiple-sampler programs have movable memory segments

After you've recorded the samples, you can edit them with the Begin and End faders as described above. You can also, if you like, adjust the faders so that samples overlap; so that a long sample is divided up into shorter ones; so that two or more samples can be combined sequentially; so that two samples start at the same point in memory; or even so the samples come up in the wrong order (e.g., by setting the Begin and End for sample 3 before the Begin and End for sample 2). The ADR 68K will remember the Begin and End times that you set for each sample. If you want to re-record a sample, you can do so, and *only that section of memory* between the Begin and End points that you have set for *that particular sample* will be affected, and the recording countdown will start at the number of seconds available in that section (End value minus Begin value). Of course, if the Begin and End points you are recording between happen to overlap any other samples, those samples will be affected as well.

If you use up all of the memory in the first few samples, e.g., the end of sample 3 occurs at 33.5 seconds, the ADR 68K will let you continue to record samples, but they will be almost infinitely short, with their Begin time set at 33.523, which is 1 msec before the end of the sample memory. You can change that value by going to the edit page, but remember if you then try to record, you will erase some sound already in the memory.

Activating unrecorded sample memories

There are times when you are going to want to use a sampler memory without actually recording anything into it. For example, say you have recorded a spoken phrase, and want to be able to access different words within it. The ADR 68K allows you to "activate" empty sample memories. Here's how:

Record the phrase into sample 1, using up as much memory as needed. Now move the Select fader to sample 2. Instead of pressing Record, however, *turn the Safety switch ON*. Now sample 2 is active, even though it has had nothing recorded in it. You can get it to play back sound by moving the Begin and End faders on page 2 to define a part of memory in which something *has* been recorded. Again, sections defined in this way can be set to overlap or come up in the wrong order if desired.

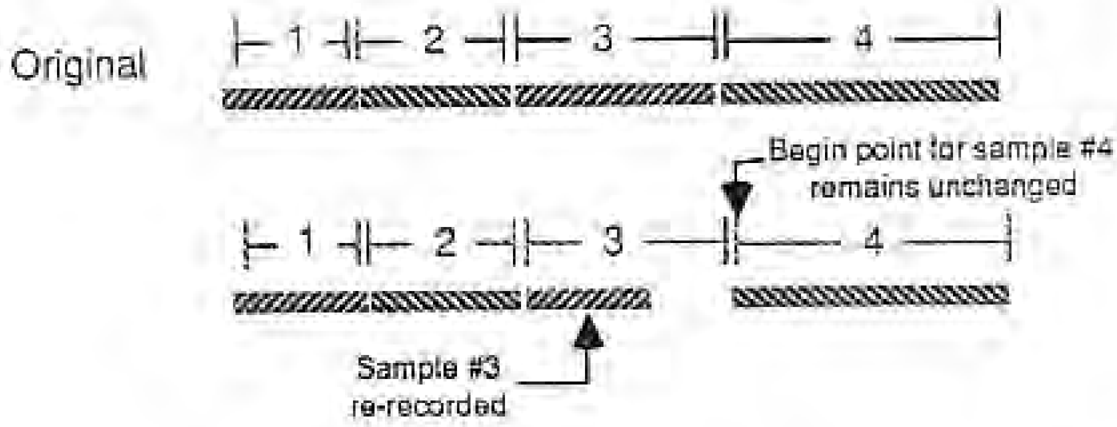
(Note that this "activation" function, although it is present in the portable sampler programs, is designed for use primarily in the non-portable programs, which allow overlapping memory boundaries.)

Here's a situation where all this might come in handy: you have a recorded sample of a long, complex sound, like a pile of tin cans falling down, starting at 1.000 second. You can set up three different samplers that access that same sound, but use different envelopes, End points, and trigger levels. Now you can run a signal from a snare drum microphone into the inputs (you'll need both inputs because you're triggering both odd- and even-numbered samplers) of the ADR 68K, and set up the sampler as follows:

Sampler #	Begin	End	Attack	Decay	Play (trig)level	Main output level
1	1.000	1.200	0	70	-24	-16
2	1.000	2.100	50	20	-12	-8
3	1.000	6.000	250	1000	-6	0

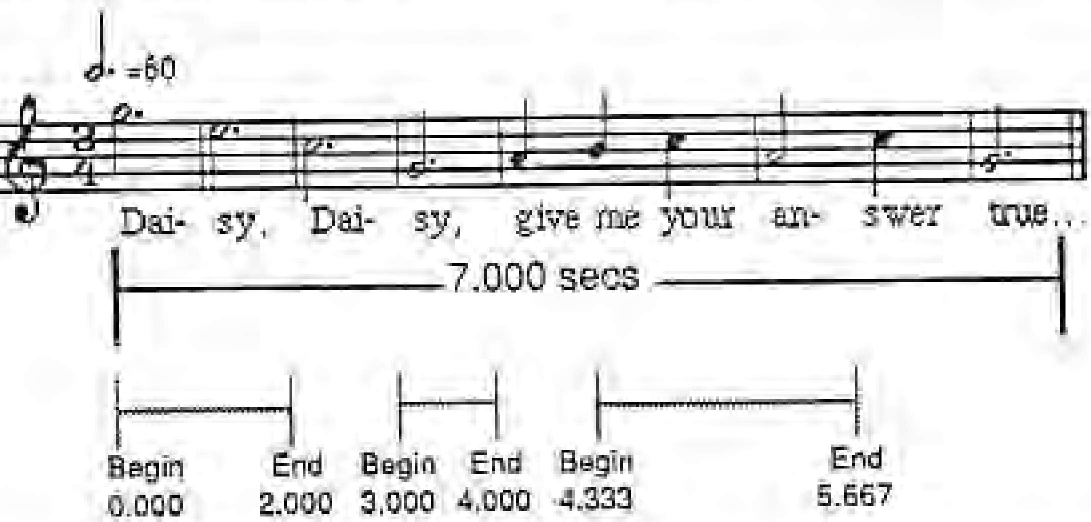
With this setup, hitting the drum relatively softly will trigger a brief, quiet "dink" out of the sampler. Hitting it a bit harder will get a somewhat longer, louder crash, while hitting it very hard will result in a long, loud, slowly dying catastrophe.

Note that although the Begin points for the individual samples set themselves automatically when you are first "filling up" a sample memory, they do not do so *after* an individual sampler has been recorded into, i.e., when you are editing or re-recording a sample. For example, say you have recorded samples 1, 2, and 3. Sample 3 started at 3.000 seconds and initially lasted 1.5 seconds. Sample 4, which is still empty, will be set to Begin at 4.520 seconds (3.000 + 1.5 + 20msecs). If you go back and re-record sample 3 and stop the recording before the full 1.5 seconds are used up, the Begin time for sample 4 will *not* reflect the new shorter value of sample 3, but instead will remain at 4.520 seconds. If you want to take advantage of the extra memory freed up by shortening sample 3, you will have to reset sample 4's Begin fader by hand.



Dividing Up Long Single Samples

The M and St programs only allow you to work with one sample, but they give you full access to the sampler memory. For example, as mentioned earlier, you can record a long musical phrase in the M program, and then by moving the Begin and End faders, extract individual notes or measures for use at different times.



You can also "punch in" a section of the sample memory by setting a Begin and End time corresponding to the section you want to replace and pressing Record. The recording will stop automatically when the End point is reached.

Remember that there is a "countup" function on Page 1 of all the sampling programs, which shows elapsed time from when the sample started playing (either manually or from a trigger). As mentioned earlier in this chapter, this feature can be very useful when dealing with long samples in helping you keep track of what events within the sample happen at what times.

It may seem to you at this point that the M-12 and St-6 programs can handle long samples just as well as the M and St programs. This is true, but don't forget that only the M and St programs allow you to change the sample's pitch on playback.

Porting Samples between programs

We mentioned earlier in this chapter that samples recorded with one sampler program cannot be used in another sampler program. Although this rule was inviolable in earlier versions of the ADR 68K software, it is no longer strictly true.

On the "SYS" menu is a function labelled "Auto Erase", which is "ON" when you first unpack your ADR 68K or first install the Rev. 4 software in an older unit. When this function is enabled, then moving between sampler programs, or moving from a non-portable sampler program to a reverb or effects program, will indeed erase the sampler memory. If you turn this switch "OFF", however, the sampler memory is retained to a certain degree.

If you go from one non-portable sampler program to another, e.g., from M-12 to St, all of the sampler memory will be retained, and you can still use the sounds recorded. However, all of the pointers — the Begin and End points that define each individual sampler — and all other settings, such as pitch change, envelopes, output levels and balances, triggering, etc. will be lost, and will have to be redefined in the new preset. You can, if you want record over a portion of a ported sample memory without affecting the rest of it. Moving from a stereo program to a mono one will cause other changes, as the two modes apportion the memory somewhat differently.

Going between portable and non-portable programs is more difficult. If you have recorded a sample with a non-portable program and go to a portable one, some of the sample memory will be erased, and what remains will be fragmented. Going the other way, from portable to non-portable, will give you extra memory to play with, but that memory may initially contain various forms of audio garbage.

Turning the power off and then on again quickly while samples are in memory can have an interesting effect, as the memory will scramble itself in unpredictable ways. It actually takes quite a long time (20 seconds or more) before the memory wipes itself completely. (If the Auto-Erase function is on, the memory will clear immediately on power-up.)

Storing Samples

If you do need to use the same samples in more than one program, and certainly if you will need them for more than one session, it is often best to store them off-line. One method is simply to record them on tape. Obviously you should use the highest-quality recording system you have, digital if possible.

The other method is to download them to a digital storage device, like a computer or hardware sequencer, via MIDI. The ADR 68K provides for two types of MIDI sample transfer, one using system-exclusive data, and the other using the MIDI Sample Dump standard. These are discussed in the next chapter.

Storing Sampler Programs

If you have used previous versions of the ADR 68K software, you may be surprised to learn that sampler presets can now be saved in the Internal and Cartridge registers. This is to allow different MIDI maps (which are described in the next chapter) associated with different sampler programs to be stored and recalled. The procedure for storing a sampler program is the same as storing a preset from any other bank: access the appropriate menu with the Shift and bank keys, find the location you want to save in, rename the preset if you want, and Save it.

As always, it's a good idea to keep all of your sampler programs stored within one bank so as to avoid confusion. It might also make sense to tag all portable sampler programs that you store with some special character to help you keep track of them (the asterisk that adorns the Factory portable samplers is not available, unfortunately). Remember that sample mixing parameters are not stored as part of a preset, so if you have a complex sample mix that you want to save, it's best to write down the various level and balance settings. Likewise, edit and envelope settings are not stored, so they should be written down if you want to save them.

Using Samples with Reverb and Effects Programs

One of the ADR 68K's most powerful features is its ability to play back samples at the same time it is performing reverb and effects processing on incoming signals. The samples themselves can also be processed at the same time. All of the reverb and effects algorithms allow this concurrent use of samples, but the only samples that can be used in this way are those recorded with the "portable" sampler programs, Mono* and Ster*.

All of the settings made from the sampler presets — i.e., edit points, envelopes, trigger levels, and output levels and balances — will be carried over and remain active within the reverb or effects preset (the samplers which employ pitch change are not portable). Triggering a sample through a reverb or effects preset is handled the same way as within the original sampler program — it can be done from the remote, from an audio signal, or via MIDI or the 1/4-inch jacks. Remember that stereo samples are only sensitive to triggers from one audio input; odd-numbered ones Left, and even-numbered ones Right.

When the samples are played, they can be sent to the inputs of the processor, where they will be treated just like any other input signal, or they can go "dry" directly to the unit's outputs, or both. Where the samples end up is handled by the input and output mixers, which will be discussed in the next section. As noted above, there is only one set of "master" mixing parameters for each sample, and they are shared by all presets — if you change a parameter affecting a sample within one preset, that same parameter will be changed in all other presets.

Remember you can always go back to one of the portable sampler programs to edit a sample, and then get right back to the reverb or effects program you were working in by pressing the Last button.

All of the Sample Trig controls in the reverb and effects presets have a RPT position, which works the same as the RPT position on the Play faders within the Sampler programs themselves. Move the fader to that position, and the sample will start to play. Move the fader up and it will stop. There is no Stop button in the reverb and effects programs, but you can stop the sound by pressing Shift-Play, Shift-Select, or CLR, or moving the fader.

There are some interesting uses for this function. One is that it can give a constant, reliable sound source for adjusting and testing control parameters within an effects preset. Another is that it can create a sonic bed or rhythmic ostinato, which will keep going regardless of what's happening around it.

Note: There is a very important restriction when using multiple samples within a reverb or effects program, or within one of the portable sampler programs, which will be of great significance when you start to work with MIDI, as detailed in the next chapter: Although you can trigger as many samples as you like simultaneously, *only two mono samples can actually be sounding at any one time,* and then only if one of them is odd-numbered and the other even-numbered. If, while two samples are playing, you trigger another one, one old sample (the odd-numbered one if the new sample is odd, the even-numbered one if the new sample is even) will be cut off while the new one plays. Similarly, *only one stereo sample can be sounding at a time,* and new sample triggers will cut off old samples.

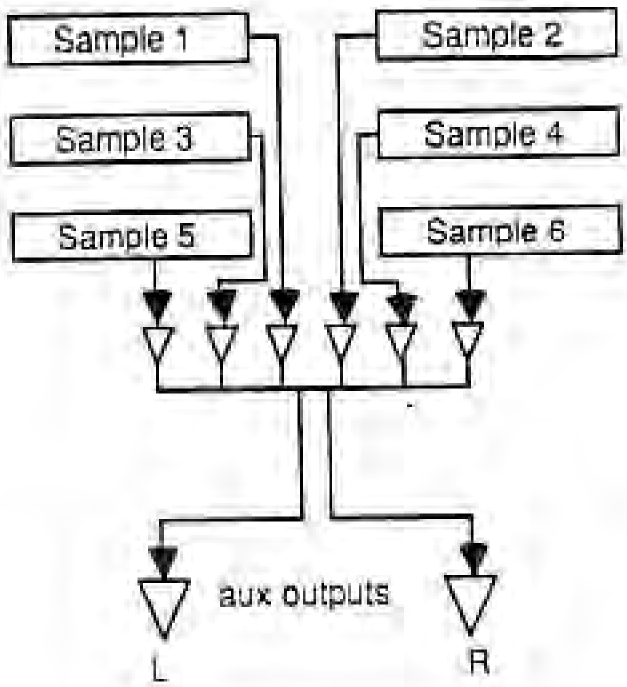
Mixing Samples

Controls for the samplers show up in the reverb and effects mixers. On the Input Mix page of each preset, the "Select", "Level", and "Balance" parameters control how much signal from the samples in memory will be mixed into the reverberator or effects processor. These controls operate precisely the same as the controls within the sample presets. The ADR 68K knows how many, and what kind of samples have been recorded: if you try to Select a sample that doesn't exist, the Select fader will not access that number, and if you try to change the balance of a stereo sample, you'll find the balance fader is disabled.

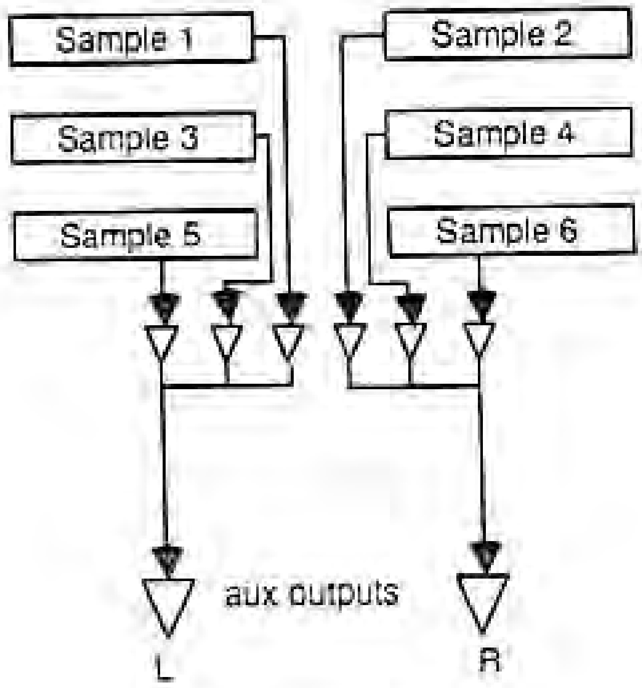
You can play a sample, and thereby check its level and placement, at any time from this page by pressing the Soft Button above the select fader, and you can stop the sample playing either by pressing Shift and the Soft Button or Clear.

On the output pages are more controls, whose nature will vary according to the program. They determine the level and stereo placement of the dry (unprocessed) signal from the samplers at the outputs. In some programs, there is a full complement of controls for the aux outputs, while in others there are limited controls. If a balance control is not available for a particular path, the display will read "LEFT" or "RIGHT" where the balance control would normally be, indicating which channel the signal will appear at. If a path itself is not available, the display will read "N/A".

If a balance control does not appear on the aux output page, odd-numbered mono samples will appear only at the Left aux output, while even-numbered ones appear only at the Right aux output. In the DDL and Chorus algorithms (Factory banks 10 and 11) there is no aux output page at all, but the dry samples do show up at the aux outputs. In the Split algorithms, the configurations for sending samples to the main and aux outputs change from one sub-algorithm to the next — refer to the diagrams in Appendix C for details.



Sample outputs with balance controls



Sample outputs without balance controls

The mixers allow you a great deal of flexibility when performing tasks like triggering samples. Here's an example. Let's set up a program which will allow us to replace a "live" snare drum and tom-tom with sampled drums.

First, use the Mono* program to record the replacement snare drum in sampler 1 and the tom-tom in sampler 2. Set the snare so that it plays slightly left of center in the output mix, and the tom slightly right of center. Finally, add some stereo reverb to the samples as they are played back.

Here's the setup:

```
RPLCE, Natural Hall, [14.14]          ⇐[6]⇒
-----------Input Mix-----------
  Input         Select    Level     Bal
   OFF            1        0dB      28⇐
```

```
RPLCE, Natural Hall, [14.14]                 ⇐[7]⇒
-----------Main Output Mix-----------
E-R's   Rev'b     Select     Level        Bal
-20dB   -15dB       1         -3dB        28⇐
```

```
RPLCE, Natural Hall, [14.14]          ⇐[6]⇒
-----------Input Mix-----------
  Input         Select    Level        Bal
   OFF            2        0dB       ⇒ 28
```

```
RPLCE, Natural Hall, [14.14]                 ⇐[7]⇒
-----------Main Output Mix-----------
E-R's   Rev'b     Select     Level        Bal
-20dB   -15dB       2         -3dB      ⇒ 28
```

The input mixer is off, so the live drums don't get into the reverberator at all. The sampler outputs, however, go right into the reverberator with no attenuation, and are panned slightly left and right on their way in. The samples appear at the output, mixed fairly high, and panned again. (For this to work properly, of course, the Sample Trig levels on Page 2 have to be set up correctly.)

Here's another example: You have a sung vocal phrase, in stereo, which you would like to have play whenever a bass drum hit appears at the ADR 68K's left input. You want it to play without reverb. At the same time, you have a background vocal track coming into to the ADR 68K's right input, and would like stereo reverb on that track. Plus, you want to preserve the bass drum coming into the left input, and put a little reverb on it.

First, record the sample and edit it using the Ster* preset. Go to the reverb preset you want to use, and set the Trig level for sampler 1 moderately high, say -18 dB. (Make sure all the other samplers are turned off.) Now set the mixing pages like this:

```
VOCLOG, Natural Hall, [I4.15]        ←[6]→
 ──────────── Input Mix ────────────
 Input          Select    Level   Bal
 -2dB             1        OFF    ---
```

```
VOCLOG, Natural Hall, [I4.15]        ←[7]→
 ────────── Main Output Mix ──────────
E-R's    Rev'b      Select    Level    Bal
-6dB     0dB          1       -3dB     ---
```

These are very simple setups. More complex ones might involve multiple samplers (for instance, a whole drum set panned across the stereo field), or might involve the auxiliary outputs in some novel ways. Certainly the more complex programs, like Splits and Multi-Effects, invite exploration of the possibilities of the input and output mixers. With a little thought and practice, you should be able to come up with many new ways to use them.

Keep in mind that the mixing parameters for the samples will not change when you move from one reverb preset to another, or when you go back and forth between a reverb preset and a sample preset. They will, however, reset themselves when you change *sampler* programs (unless you go between two programs that use the same sample format, for example, "M" and a variation on M that you have stored in an Internal register).

Also remember that the mixing parameters are not stored as part of a preset — either in a reverb or effects program, or a sampler program — although the *reverb* input and output mixes *are* stored. Therefore, if you have concocted a complex multi-sample mix which you would like to preserve, you should write down the various balance and level values before you shut off the power or load in a new set of samples.

— Chapter 7 —
MIDI

The Rev. 4.0 software of the ADR 68K has an extensive MIDI implementation, allowing comprehensive remote control over the unit in a variety of modes:

• It can receive program changes, and assign them to specific presets.

• It can send and receive system-exclusive dumps of all or some of its memory registers, and can also send and receive samples, using either system-exclusive or the new MIDI Sample Dump Standard.

• It can act as its own MIDI "instrument", sending all fader and button moves to a sequencer in real time using MIDI continuous controller commands, where they can be recorded and played back.

• Every control on the ADR 68K can be assigned to respond to virtually any type of MIDI command, from a live performer or a sequencer.

A MIDI implementation chart appears later in this manual in Appendix B. The ADR 68K's system-exclusive data format is available from AKG Digital.

MIDI functions are accessed by pressing the "MIDI" button on the remote control. This brings up a display with six choices.

The **Setup** menu has controls to enable overall MIDI reception; to select a MIDI channel to work with or put the unit in Omni (all channels) mode; to enable reception of program change commands; to enable reception of System-exclusive data generated by an external editor; and to enable a special "MIDI Remote Control" mode for recording control movements on the ADR 68K into a sequencer and playing them back.

The **Map** menu allows you to assign specific registers in the ADR 68K to specific MIDI program change numbers.

The **Send** and **Receive** menus enable bulk dumps over MIDI of the control parameters that define one or more presets, and of samples recorded with the ADR 68K.

The Parameter menu allows specific MIDI events, including notes, velocities, and continuous controllers, to be assigned to specific faders and buttons within a preset. In addition, this menu determines the functions of the four 1/4-inch jacks at the rear of the ADR 68K's remote control.

Finally, the Test button allows you to perform a check of the unit's MIDI In and Out jacks, and also provides a way of testing MIDI cables.

The MIDI functions are accessed from within a preset, and you can always get back to the preset by pressing "↑" one or more times.

Main functions

Pressing "Setup" accesses most of the main MIDI functions. These functions are not Preset-specific — i.e., they do not change when you change presets. All of the controls on this page are operable with either the fader or the Soft Button underneath their labels.

```
 Basic MIDI Setup                                  ↑

                              —Auto-Midi—
Master   Chan   SysEx   Xmit    Recv    PgChng
  ON      11     OFF     ON      OFF     ON
```

"Master" enables and disables the MIDI functions. If it is Off, it means that the ADR 68K will not respond to MIDI commands at all. (However, any MIDI data appearing at the "MIDI In" jack on the back of the mainframe will still be echoed to the "MIDI Thru" jack.)

"Chan" sets the MIDI channel on which the unit will send and respond to information, from 1 to 16. Setting this control to "Omni" means that the unit will respond to MIDI data on any channel, and will send data on Channel 1.

Turning on "SysEx" allows the unit to accept a single preset definition, via MIDI system-exclusive, into the current register, in real time. This feature will be useful with any editing software that may become available for the ADR 68K and a personal computer, in that it will allow complete presets to be sent from the computer and tried out on the unit itself, without storing them permanently in memory. If you want to store the preset permanently, you must use the customary procedure of picking a register and Saving. There are other sets of controls for transmitting preset definitions, or for receiving bulk parameter "dumps", which we'll get to in a moment.

"Xmit", when turned on, enables a special mode called "Auto MIDI". This function is used for real-time recording of fader and other movements of the ADR 68K's controller into a sequencer. We will cover this in detail a little later.

"Recv" is the flip side of "Xmit". When it is enabled, information in a sequencer that was previously recorded from the ADR 68K can now be played back to control the unit. Both Recv and Xmit can be enabled at the same time.

"PgChng" enables incoming MIDI program changes to call up ADR 68K registers. *Which* program changes call which registers is the job of the "Map" menu, described below.

All settings in the Setup menu are permanently retained in non-volatile memory automatically. If "Xmit" is left on when the power is turned off, the ADR 68K will start transmitting MIDI data as soon as it is powered up again, and if "SysEx" is left on, the unit will be externally controllable immediately on power up.

The other main MIDI function is **Test**, which is accessed from the main MIDI page. If you connect a MIDI cable between the unit's MIDI Out and In jacks, this test will tell you if there is continuity. (Use a straight cable — any splitters or "Thru boxes" may interfere with the test.) If the test fails, either the cable is no good, or there is something amiss with one or both of the MIDI jacks. If you know that the MIDI jacks are operating properly, then this feature can be a convenient MIDI cable checker. To get back out of the test, press the "↑" button one or more times.

Assigning program changes

```
Midi Map Albert, Natural Hall
 -Midi-   ----ADR Register----
 Program  Type    Bank    Register
    3      INT      4        15       Save
```

The "Map" menu is used for determining how the ADR 68K responds to MIDI program-change commands. This function allows you to specify which register is called when the unit receives a particular program change. MIDI program changes are numbered 1 through 128, and each of those numbers can be mapped to a corresponding ADR 68K register. (Some manufacturers of MIDI equipment label their program changes 0-127, which actually conforms closer to the MIDI specification. If you have such a piece of equipment, simply add 1 to all of your program change commands when you are constructing a map for the ADR 68K.)

The ADR 68K comes from the factory with a "default" MIDI map. In this map, the first 10 presets in each of the 11 Factory banks are assigned MIDI program numbers in order; i.e., registers F1.1 through F1.10 are assigned MIDI program numbers 1-10, F2.1-F2.10 are assigned 11-20, F3.1-F3.10 are assigned 21-30, etc. The first fader on the Map page lets you scroll through and view the entire map. As you scroll, you'll see the number of the program change, and the name of the assigned preset, its register number, and its algorithm.

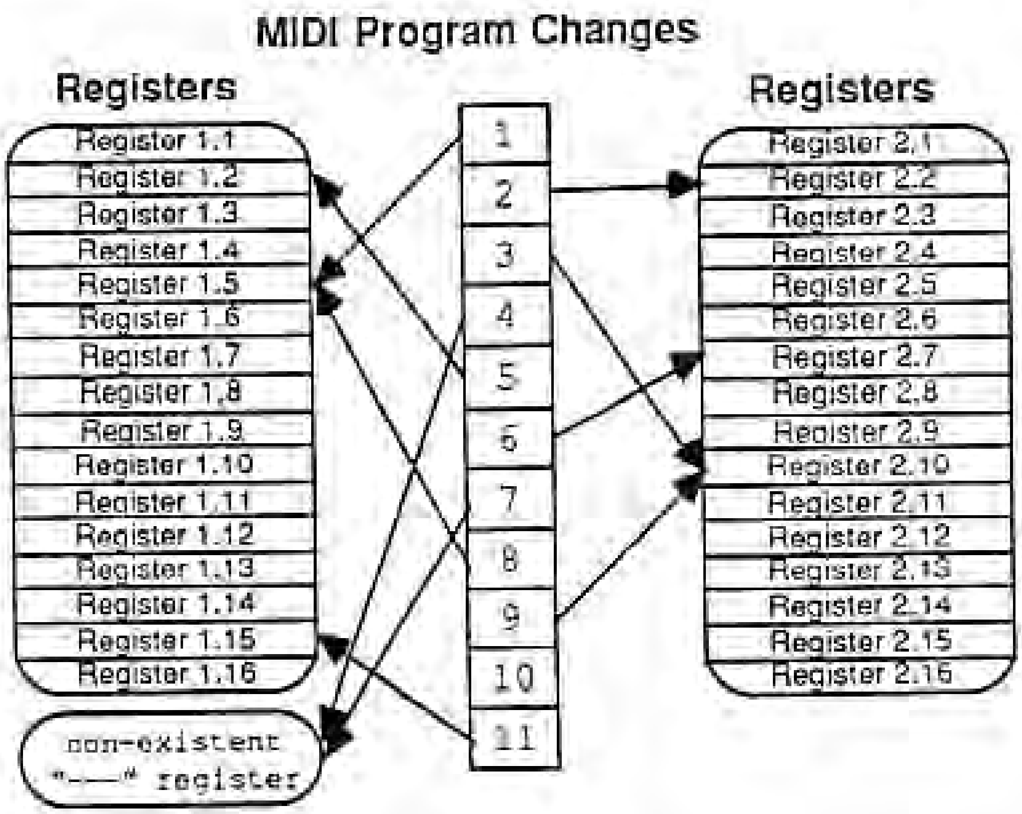
If you want to change the map, here is the procedure: Suppose you want MIDI program 3 to recall register I4.15. Move the first fader so that the number 3 appears under the word "Program". Move the second fader until the word "INT" appears. Then move the third fader until Bank reads "4", and the fourth until it reads "15".

You will notice that as soon as you move the *second* fader, the word "Save" starts to blink, indicating that you have made a change. If you press Save, the change will be recorded in non-volatile memory, and will stay there until you edit it again (or Reset the machine), and the blinking will stop. Note that *each change in a MIDI map has to be saved individually;* if you don't press Save after you make a change, then as soon as you move the Program fader, or leave the menu, the change will be lost.

For the sake of convenience, particularly in live situations, more than one MIDI program change can be assigned to a particular register. This is so when a synth player, for example, who is slaving the ADR 68K to his master keyboard, switches between low brass and high brass patches, he does not have to change the reverb setting as well.

In addition, an empty register can be assigned a MIDI program change. If the unit receives a program change that is mapped to an empty register, it is ignored, and the current program is maintained. This is useful in situations where you need to control the programs on an ADR 68K and another MIDI device simultaneously, but you have only one free MIDI channel. You can set up the ADR 68K so that it responds only to program changes below 65, assigning all other program changes to empty registers, and set up the other device to respond only to numbers above 64.

Another way to ensure that a MIDI program change will have no effect is to set the first fader to the program change number, then move the fader under "Type" all the way up until "—" appears, and press Save. The program change will now be mapped to a *non-existent* register, and therefore ignored.



The MIDI program change map can be completely disabled, without altering it, by turning off the "PgChng" switch on the MIDI Setup page.

Parameter dumps

All of the parameters that describe a particular preset or an entire bank of presets can be sent over MIDI to another device, using system-exclusive data. This is useful for storing the preset information "off-line" in a computer or hardware "librarian" that can deal with that kind of data, and can restore it to the ADR 68K later on. It allows you to design and save an unlimited number of presets. It's also useful if you have two ADR 68Ks and want to get presets from one to the other without entering all the information and adjusting all the controls by hand.

The "Send" page lets you send all of the data in every register in the Internal bank (press "All INT"), or just the current register (press "Current"). If you want to send the Cartridge registers, you must first Swap them with the Internal registers, as described in Chapter 4. (There is no need to send Factory registers, as they are always there, in all ADR 68Ks.)

```
Send Samples or Registers via Midi        ↑
Current register is: ALBERT, [I4.15]

SysEx   MMA SDS->12  Current       All INT
```

After data has been sent, a message appears telling you so.

The "Receive" switch on the main MIDI page opens up the unit's Internal memory to receive bulk parameter data over MIDI. The Receive function will only accept a complete internal memory, not just one register or bank, and so will erase any existing internally-stored registers. (If you want to receive just one preset, use the SysEx switch on the Setup page, described above.) The program gives you a chance to change your mind after you have chosen "All Regs".

```
Receive Samples or Registers over Midi ↑


 SysEx   MMA SDS                  All Regs
```

If you have a MIDI sequencer with the ability to handle system-exclusive events within a sequence, you could conceivably store an *entire preset* (not just a program-change number) in a sequence using the Xmit function, and have it play back into the ADR 68K in real time, with the SysEx switch on the Setup page enabled (the SysEx switch on the Receive page is for samples — don't get confused). However, this amount of data coming out of a sequencer could clog the MIDI data line, so it is not recommended except under special conditions.

Sample dumps

With the Rev. 4 software, samples recorded with the ADR 68K can now be off-loaded over MIDI. The Send and Receive pages have controls for sending samples as well as parameters.

Samples can be sent (downloaded) only from within a Sampler program. With the numeric keypad or from the bank menu choose the sample program that was used to create the current sample (or a variation of it). Choose which sample you want to download with the Select fader. Now press "MIDI" and "Send".

The first two Soft Buttons control how the sample is going to be sent. Pressing "Sysex" sends the information as MIDI system-exclusive data. "MMA SDS" sends it in the Midi Manufacturers Association "Sample Dump Standard" format. When you use the SDS format, the third fader lets you choose whether to send the sample in 16-bit or 12-bit form.

There are advantages and disadvantages to using each of these formats, which will determine which one you will choose.

System Exclusive ("Sysex") Pros: it is the fastest way to transfer the samples; stereo samples can be sent and received; full 16-bit fidelity is always maintained; many software products are available that let you receive, store, and send system-exclusive data; attack and decay envelopes are stored along with the sample; open-loop operation (in which only one MIDI cable is used, going from the ADR 68K to the receiving device) is possible.

Cons: there is (as of this writing) no way to edit samples which have been downloaded in system-exclusive format; samples downloaded this way cannot be uploaded to any other device, like a sampling keyboard, except for another ADR 68K.

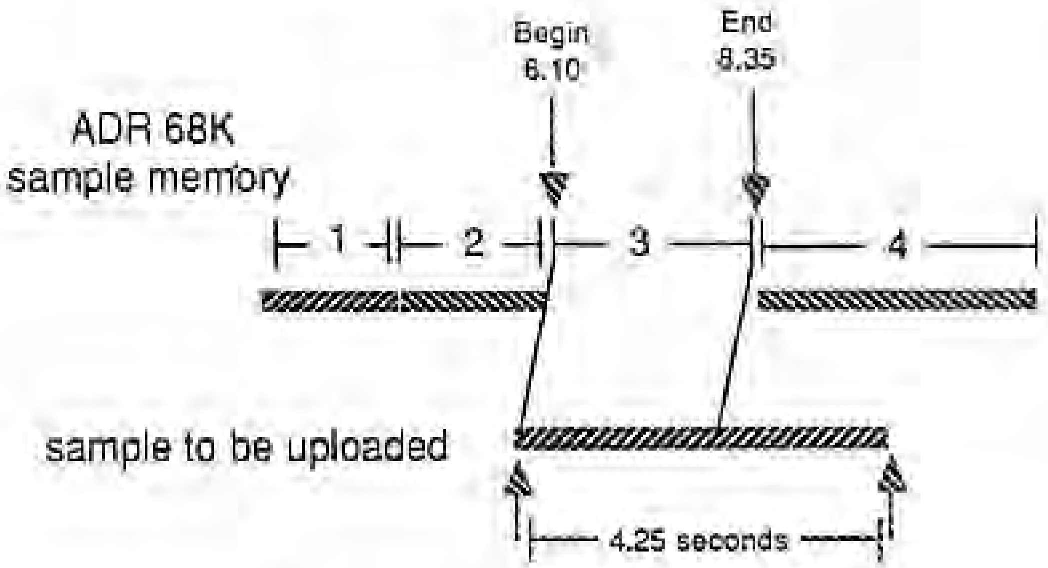
Sample Dump Standard ("SDS") Pros: Excellent software exists for manipulating and editing samples with a personal computer; samples can be transferred to a large number of keyboard and rack-mount samplers; you can initiate a sample dump externally, for example, from a computer running a sample-editing program.

Cons: Existing sample-editing software only deals with 12-bit samples, so some fidelity will be lost; the standard has no provisions for stereo samples, so if you try to download a stereo sample only the Left channel will be sent; any attack and decay envelopes you have created are lost; the process is somewhat slower than using system-exclusive; closed-loop operation (in which two MIDI cables are used, one from the ADR 68K to the receiver and the other from the receiver to the ADR) is essential, or the transfer will take a very long time and some deterioration of the sample may result.

Once you initiate a sample dump, a message appears on the screen telling you how much time is remaining in the procedure. You can press any button on the remote to cancel the operation at any point. To avoid confusion, if you cancel a sample dump halfway through, make sure that you *also* cancel the operation on the receiving device — most storage devices cannot handle "partial dumps". Press the "↑" key to get back out of this menu.

The procedure for reloading ("uploading") samples back into the ADR 68K is quite similar. First make sure that a sampler program is in the current register (you cannot upload samples into a reverb or effects program), and use the Select fader to choose which sampler you are loading into. (This number does not have to match the number of the sampler as it was downloaded; e.g., you can download a sample from sampler number 12 and upload it back into sampler 1.) Go the "Receive" page from the main MIDI menu, and press either the "SysEx" or "MMA SDS" button, depending on the format of the sample you wish to upload (if you are uploading an SDS file, the unit will automatically know whether it is 12- or 16-bit). Now send the sample data from the source into the ADR 68K's MIDI input.

Before you upload a sample, it's a good idea to make sure that there is enough room to hold the whole thing in the sample memory you've chosen. Go to Page 2 of the sampler program, and adjust the Begin and End points so that they are far enough apart to accomodate the entire sample. If there is not enough room, the upload will still work, but the end of the sample will be cut off.



Sample will be truncated if there is not enough room to upload it

Also watch out that when you upload multiple samples, you don't upload more than one into the same portion of memory. It's a good idea to upload samples *in numerical order* to avoid confusion, but note that unlike when you are *recording* multiple samples, the ADR 68K will not automatically set Begin points for subsequent samples when you are uploading them — you will have to set each Begin point by hand before you upload.

A downloaded sample does not know very much about where it came from — specifically what type of ADR 68K program was used to create it (e.g., mono, stereo, fixed-length, etc.) — so you can create a sample with any of the sampler programs, download it, and then upload it into any other program. Be aware of the following discrepancy, however: if you download a stereo sample, and then upload it into a mono sampler, the right channel will be lost; similarly, if you upload a mono sample into a stereo sampler, the right channel will have no data in it.

Note also that when you download a sample, *only the portion of the sample that lies between the Begin and End points* on the second page of the sampler program get sent. In other words, if you edit out the beginning of a sample, download it, and then upload it again, the portions you have edited out will be lost permanently. If you want to save *all* of the original sample, make sure the Begin and End faders are at their original value when you download.

Note: During sample up- and downloading, you may hear spurious noises from the ADR 68K's audio outputs. This is perfectly normal — turn down your monitor outputs if you find them overly annoying.

Live MIDI Control of the ADR 68K — the Auto-MIDI Mode

As MIDI sequencers, controllers, and processors become more sophisticated, more demands are being put on the standard, particularly in the realm of studio automation. Recognizing this, the ADR 68K has been designed so that all of its functions can be controlled externally by MIDI, either live from a keyboard or other controller, or from a sequencer, which may in turn be controlled from audio or videotape, or some other timing source.

The Auto-MIDI mode is a special feature of the Rev. 4 software that allows the unit to be used in conjunction with a MIDI sequencer, with a minimum of effort by the user.

In this mode, each of the six faders and 38 buttons on the remote controller is automatically assigned a specific MIDI command. Moving a fader, or pushing or releasing a button, sends that particular command out the MIDI Out jack, and it can then be recorded in a sequencer. (The MIDI channel which the data is assigned to is determined by the "Chan" fader on the main MIDI page — if it is set to "Omni", the transmit channel is 1.)

Similarly, the unit will accept this data from the sequencer, and "perform" (electronically, not physically) the fader movements and button pushes accordingly. Although the faders and buttons do not move when the unit is receiving MIDI data, the displays do change.

This mode is turned on by the "Xmit" and "Recv" controls on the MIDI Setup page. Either or both switches can be turned on at any time.

Any operation possible with the ADR 68K can be controlled via MIDI through this mode. Not only can parameter values be changed in real time, but inputs can be cleared and muted, outputs can be remixed, registers can be loaded and saved, program configurations can be changed, MIDI assignments can be restructured, and even samples can be recorded, edited, and played back, all automatically. The data sent out reflects the *absolute* positions of the faders and buttons, so if you make a minor adjustment in some parameter value, it will not affect subsequent fader movements.

However, there is a need for caution when using this mode. Remember that all of the ADR 68K's controls are context-sensitive — that is, the effect that moving a particular fader will have is dependent on what preset and page you happen to be on. The Auto-MIDI controls are "dumb", in that they know that a fader is supposed to moving, but they know nothing of its context, or what it's *really* doing.

Therefore, recording a sequence in this mode should be regarded essentially as a linear operation — start recording the sequence, perform all the program, parameter, etc. changes you want in the sequence, and then stop. Overdubbing is possible, as long as movements you overdub do not contradict or otherwise interfere with movements previously recorded. Punching-in on a previously recorded sequence is only recommended if the punch-in continues all the way to the *end* of the sequence, since the ADR 68K will have no idea where it's supposed to be if you punch-out in the middle.

Adhering to the following principles will help to prevent problems:

1) Always start the sequence playing from the beginning. If you start in the middle, the ADR 68K will most likely be in the wrong state, and the fader and button actions will have all the wrong effects.

2) The first command that the ADR 68K should see at the beginning of a sequence should establish a "reference point" for it to work from. Therefore, when you start recording the control sequence, immediately call up a stored register using the numeric keypad on the ADR 68K's remote. This is an unambiguous command, which will always put the ADR 68K in the same place, i.e., page 1 of a particular preset, every time you start the sequence.

3) Stay away from commands that may have more than one possible response. For example, if you decide to call a sampler preset from the numeric keyboard, what will happen next will depend on whether there is data already recorded in another sampler program. If there is, you'll get the "WARNING: Samples will be lost..." message, but if there is no data, you will not get the message. The sequencer has no way of knowing this is happening, and cannot possibly know the proper way to respond. Similarly, if you want to record a sample, the sequencer has no way of knowing whether the particular sample you've chosen already has data in it, and therefore doesn't know whether the Safety switch needs to be turned off before you can start recording.

4) When "Recv" is enabled, the ADR 68K's controls are still "live". You can make adjustments, but be careful that what you do does not interfere with the sequence of events, i.e., don't go to a different page or register if you know that there are fader movements coming up.

5) If you have "Xmit" and "Recv" enabled at the same time, for overdubbing or punching in, make sure any "Echo" or "Thru" function in the sequencer is turned off, or you'll end up getting doubled commands, which will cause severe problems.

6) Don't play sequences back significantly faster than their original tempos. The reason for this is that sometimes very large fader movements take a few milliseconds to take effect and it is possible, if you are playing a sequence that has a large fader movement followed closely by a button push, that the button may get pushed before the fader movement is complete. An example of this would be a page change following a long fader movement. If the page change gets executed in the middle of the fader movement, you'll get two fader moves: the first half will end up on the old page while the other half will be on the new page. If you play back sequences at (or not too far from) the tempo at which they were recorded, this should not be a problem.

The Auto-MIDI mode is meant to be invisible to the user, and sequences recorded with using it are not really meant for close editing. If you find yourself in a position where you must edit such a sequence, however, the actual MIDI commands generated by the faders and buttons in this mode are listed in Appendix B.

Live MIDI Control of the ADR 68K — the Parameter Mode

There is another way to control the ADR 68K via MIDI, known as the Parameter Mode. In this mode, which is equally useful for live performance work as for sequencing, particular MIDI commands are assigned to specific ADR 68K parameters, entirely under user control.

Unlike the Auto-MIDI mode, in which the *physical* faders and buttons are assigned to MIDI commands, in the Parameter mode it is the *parameters* themselves that are under MIDI control — while they are being controlled, the physical faders and buttons may be doing something completely unrelated. This mode only works in one direction — the ADR 68K can receive MIDI parameter messages, but *does not send* them.

Determining which MIDI commands control which parameters is the job of the Parameter menu, which is accessed from the main MIDI page. The Parameter menu lets you map up to ten different MIDI commands to ADR 68K parameters. These ten "maps" (which should not be confused with the Program Change Maps discussed previously) are not "universal" — instead, they are part of the Preset within which you create them, and each preset has its own set of MIDI parameter maps. When you save a preset, its MIDI maps are saved along with it, and when you recall a preset, its MIDI maps replace whatever MIDI maps were previously in use. You can have two presets which are identical except for their MIDI maps. (All of the factory presets have blank MIDI maps.)

To set up a MIDI map, go to the Parameter menu, which you will notice is in two pages.

```
ADR 68K Parameter Control                ↑[1]→
                Target:RT60

Map#     Source                 Target Target#
 1        OFF                   FADER     1
```

```
ADR 68K Parameter Control                ←[2]↑
── Source ──                  Target:RT60
 Low     High                     Low     High
 24      98                      3.85s  14.85s
```

The first fader on the first page is used to select which map you would like to work on. Unlike with the samplers, there is no reason to work on the maps in any particular order — any combination of the ten can be active or not at any time.

Sources

The second fader is labelled "Source". This determines the identity of the MIDI command that will be used in this map. If this control is set to "OFF", the particular map is disabled. As you move the fader up, or press the Soft Button, you will see the various types of commands appear in the display under the word "Source".

The MIDI command sources available from this fader are as follows:

• **Continuous Controllers 1-31** and **64-128.** In the MIDI Specification, many of these are assigned particular functions, such as Mod Wheel, Breath Control, Detune, Volume, General Purpose Controller (GPC) 1, All Notes Off, Omni Mode On, etc. When you move the fader to the position of such an assigned controller, the display shows the name of the controller. Unassigned controllers are simply referred to by their numbers. (Controllers 32-63 are generally paired with controllers 1-31 as "Least Significant Bytes" of a double-precision message, and so they are not used here.)

• **Pitch Wheel.** Only the Most Significant Byte of a Pitch Wheel message is read.

• **Channel Pressure** (or Aftertouch).

• **Poly Pressure** (or Polyphonic Aftertouch) Event and Note#. Poly Pressure is, at the current time, a fairly esoteric and not widely-used command. It contains separate bytes for the event itself and for the number of the key being pressed. The ADR 68K can treat those two bytes as separate commands.

• **MIDI Clocks,** which reflect the speed of a sequence, and can be useful when matching delay times to musical tempos. The units used in the display are quarter-notes per minute, in which a quarter-note is defined as 24 MIDI Clocks, as per the MIDI specification. The range is 20-274 beats per minute, in 2-beat increments. If you have a sequence that uses eighth-notes as beats, then the tempo will be the number displayed times 2. The BPM calculator on the "SYS" page can be very useful in working with MIDI Clocks and delay times.

• **Note-on Number and Velocity.** A Note-on Number command indicates the number of a key being pressed. The velocity value indicates how quickly the key is being pressed. The ADR 68K treats these as separate events.

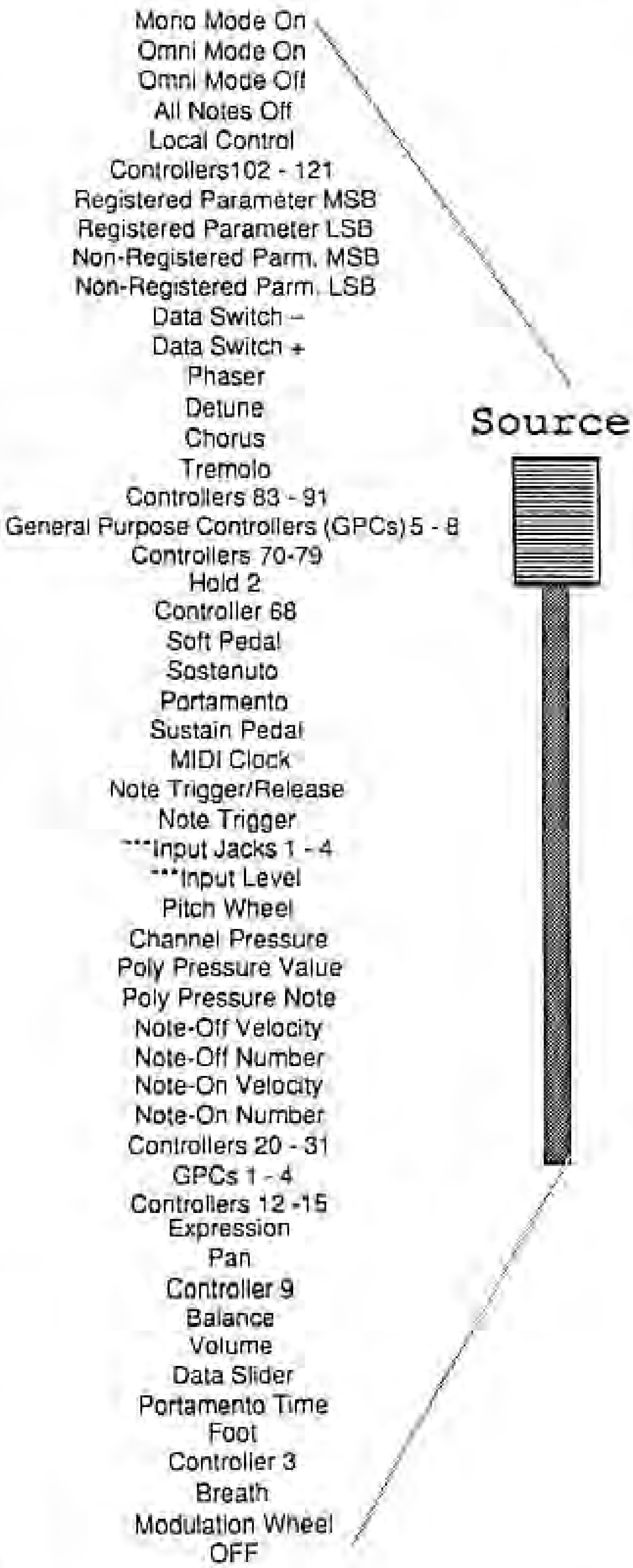
• **Note-off Number and Velocity.** Likewise.

In addition, several non-MIDI commands can be accessed as Sources in a MIDI map, and assigned to operating parameters and controls. These are:

• **Input Level**, which senses the averaged level of audio signals coming into the Left and Right inputs.

• The four **Input Jacks** on the back of the remote controller. They are capable of reading control voltages, as described in Chapter 1 of this manual, and generating a value of 1 to 128. (Remember not to input signals into these jacks greater than +5 volts or less than 0v [negative voltages] or you may do serious damage to the ADR 68K.)

To simplify setting up a parameter map, there are two "meta-commands" that have been included in the software, which we'll get to in a moment.



The MIDI Commands on the Source fader, from top to bottom.

The commands marked with "*" are non-MIDI commands.**

"Note Trigger" and "Note Trigger/ Release" are combinations of MIDI commands called "Meta-Commands".

Targets

The third fader on the second Parameter page is labelled "Target", and the fourth is labelled "Target #". These determine where in the ADR 68K the data generated by the Source will go. The "Target" fader determines the type of target, while the "Target #" fader determines the specific target. Possible targets are:

• **Faders.** The number of the fader refers to its position *within the preset*. That is, the first fader on the first page of the preset is Fader #1, but the first fader on the second page is Fader #7 (regardless of whether all six faders on the first page are functional). The number of faders available will depend on the preset you are working in: if the preset has eight pages of menus, there will be 48 faders to choose from, but if it has only four pages, there will be 24 faders to choose from.

As you select a fader with the Target # control, the function of that fader within the preset appears on the second line of the display. If that fader has no function, that part of the display will remain blank. (You can assign a MIDI parameter to a non-functioning fader, if you like, but there is no reason to do so.)

• **Soft Buttons** (or **Keys**). Their number and identity likewise depend on the current preset. The identity of each button is shown as you scroll the Target # fader.

• **Shift-Soft Buttons.** Ditto. These are considered as separate items from the Soft Keys. The Shift Button itself does not have its own target assignment, so if you want to use a Shift-Soft Key command as a Target, this is how you set it up.

• **Mute** and **Clear** buttons. Since there is only one of each of these buttons, the Target# fader is disabled when you select either of them.

Meta-Commands

There are two "meta-commands" in the Source list which have specific uses, and are designed to simplify the mapping operation. They are:

• **Note Trigger**, in which a Note-On Number command is automatically sent to two different targets: one a fader, and the other the Soft Button immediately above the fader. You can only assign a Note Trigger event to a fader — the setting of the first "Target" fader is ignored.

Although this command, like all other source commands, is mappable to any fader in a preset, it will most likely only be used to play multiple samples from a keyboard or other controller; therefore you would normally asssign it to a "Select" fader in either a sampler or a reverb or effects program. When assigned this way, the note number controls the position of the fader, and therefore the sample number, and also pushes the Soft Button, thereby triggering the sample.

• **Note Trigger/Release**, in which a Note-On Number command is assigned as in the Note Trigger meta-command, *and* a Note-Off Number command is assigned to the same fader/button combination. When the note-off command is received, however, instead of pushing the Soft Button, it pushes Shift and the Soft Button, thereby cutting off the sample. (If there is a Decay envelope associated with the sample, it will start as soon as the note-off command is received.) The note-off number command also moves the fader to the proper position, in case an intervening command, like another note-on, moved it somewhere else during the time since the original note-on occurred.

In other words, when you assign a Note Trigger command to a Select fader, pressing the proper note will trigger a sample, and make it play all the way through. When you assign a Note Trigger/Release command to a Select fader, pressing the key will start the sample, and releasing it will stop it.

In both of these cases, the note-on and note-off velocities are handled as separate commands, and if you want them to affect the sample playback, they must be mapped separately to an output level.

When the ADR 68K receives note-on and note-off commands, whether they are mapped separately or as part of a meta-command, it always knows to process them in the right order — i.e., the fader selecting the sample number is activated first, and then the button is pressed to trigger the sample. You will never have a situation where an attempt is made to play a sample *before* it is selected.

Setting ranges

The next step in designing a MIDI parameter map is to determine the operating ranges of both the sources and the targets. Being able to set a source range allows the MIDI control of the ADR 68K to be tailored to particular performing and recording situations. For example, assume you want the pre-delay of a reverb program to increase dramatically when you play a MIDI keyboard loudly. However, let's say you only want the effect to occur with notes above a velocity of 60 (more or less equivalent to a dynamic level of *mezzo-forte*), while all notes softer than that have no effect on the pre-delay. To accomplish this, you set the source range so that only velocity events between 61 and 128 will affect the pre-delay.

Another example would be if you want the pitch wheel to have one effect when it's pushed up, and a different effect when it's pushed down. In that case, one parameter map would have a source range of 0 to 4096, while the other would have a range of 0 to –4096.

Any incoming MIDI data outside of the source range is ignored. If, for example, you have notes from two octaves of a keyboard, say C5 to C7, mapped to trigger 12 different samples, any notes below C5 or above C7 will have no effect. This allows you to set aside a particular range of a MIDI keyboard for ADR 68K control, and you can still use the rest of the keyboard for other functions, like accessing notes on a drum machine.

The target range can be adjusted similarly. To continue with the first example, let's say that you want the minimum pre-delay, at the lowest velocity levels, to be 35 ms, and the maximum, at the the highest velocities, to be 450 ms. Set the target range so that the minimum source value sets a pre-delay of 35 ms and the maximum source value sets a pre-delay of 450 ms.

One thing to watch out for when using something like velocities in this way is that if you send the ADR 68K a note with a velocity of less than 61, the pre-delay will not automatically go to the minimum value, but instead will stay at the *last* value it was given. If you play a note with a velocity of 128 followed by one with a velocity of 40, the pre-delay will remain at 450 ms, until you send another note whose velocity is inside the source range. Similarly, if you have the upper, positive, half of the range of the pitch wheel assigned to a parameter, and you move it quickly down past the zero point, the parameter may not "zero out", because the pitch wheel may not actually send a zero value, but may skip over it as it travels.

Source and target ranges are set up on Page 2 of the MIDI Parameter menu. The first fader in each pair sets the lower limit of the range, and the second fader sets the higher limit. Both sets of faders are context-sensitive — that is, they reflect the real or "native" values of the source or target being used in the map you are working on. While with most sources the range will be 1 to 128, if you have chosen the source to be "Note-on number", for example, then the range available on the source faders will be "C 0" through "G 10". Similarly, if the source is "Input Level" the range available will be "-70dB" to "LIM".

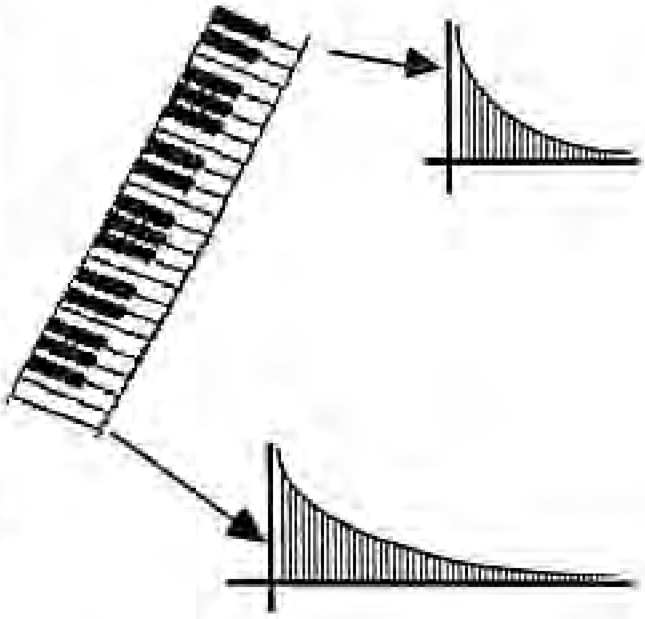
```
ADR 68K Parameter Control                ← [2] ↑
── Source ──                   Target:RT60
 Low     High                     Low     High
 24      98                      3.85s   14.85s
```

In the same way, the target faders reflect the native values of the ADR 68K parameter they are working with. For example, if you are in a preset that uses the Complex Plate algorithm, and you have set the target to "Fader 2", which is the LF Decay of the Running Reverb in that algorithm, the available range will be "X0.1" to "X2.0". Note also that the name of the target fader appears on the second line of the display of this page, just as it does on the first page of the Parameter menu, to help you keep track of where you are. If you are targetting a fader that has no function within the preset you're in, the range values will be blank.

Note that you can set a source or target range so that it is "upside down", i.e., so that increasing Source values results in decreasing parameter values. For example, this setup will cause RT60 values to shorten as you play notes higher up the scale:

```
ADR 68K Parameter Control            ↑[1]→
              Target:RT60

Map#    Source               Target Target#
 3      Note-On Number        FADER     1
```

```
ADR 68K Parameter Control            ←[2]↑
── Source ──                 Target:RT60
Low     High                    Low    High
C 2     B 5                    5.50s   1.32s
```



Note that when the value of one fader is dependent on the value of another — e.g., the "Delay" and "Range" faders within the Dual Delay programs, or the "Size" and "RT60" faders in a reverb program — MIDI control of the "slave" fader will also be dependent on the setting of the "master" fader. Therefore, changing the value of the Range fader will affect the way that MIDI controls the Delay fader, and the Target high and low settings of the Delay fader will change accordingly.

It is completely permissible to map the same source to two different targets, or to map two sources to the same target. In the case of the former, the two target parameters will simply move in parallel (or, if one is mapped "upside down", in contrary motion). In case of the latter, the parameter value will always reflect the *last* value sent to it, regardless of which source sent it.

Remember that Note-On and Note-Off are treated as completely separate entities (except in the Note Trigger/Release meta-command), so if you want the action of one to negate the other (which will occur often), you must set up the proper map for each of them.

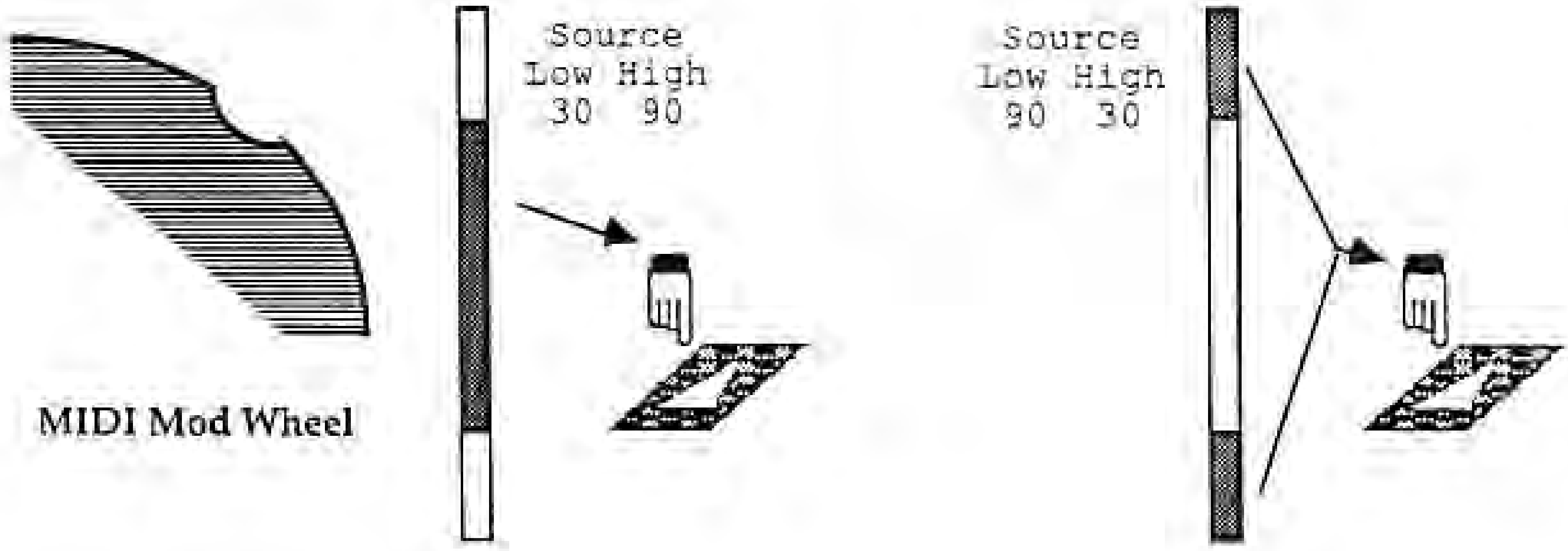
All sources, targets, and ranges will remain active within a preset until you leave the preset. If you want to save the values, you have to save the preset, or else they will be lost when you leave the preset.

You can easily get back and forth between the control pages of a preset and its MIDI parameter page without disturbing either. To get from the preset to the MIDI page, press "MIDI" and then "Parameter". To go the other way, press "↑" twice.

Note: When the first MIDI Parameter page is showing on the display, the MIDI parameter mode is disabled. To enable MIDI parameter control, you must leave that page, either by going to the Range page, or back to the preset (by pressing "↑"). This feature is to prevent the ADR 68K from getting into an endless MIDI feedback loop, which could have nasty consequences.

Buttons as Targets

If you select a Soft Button or Shift-Soft Button as a Target in a map, there will be no Target Range parameters on Page 2. If any Source command is received whose value is within the Source range, the button will be "pressed", and all commands outside the range will be ignored. *However*, if the Source range is set up so that the Low value is *higher* than the High value, than any commands *outside* the range will press the button, and any notes inside the range will be ignored.



Things are slightly different if the Mute or Clear button is a Target. In this case, any command within the Source range pushes *and holds* the button, while any command outside the Source range releases it. Again, if the source range is upside-down, these functions are reversed.

Live Control under MIDI

The physical faders and buttons remain "live" while the ADR 68K is under MIDI control, just as they do in the Auto-MIDI mode. Since the external control is over specific parameters, and is not related to the current state of the ADR 68K's display, you can operate the controls with much more freedom than you can under Auto-MIDI mode.

What About Program Changes?

Notice that there is no Source setting for recognizing MIDI program changes, and no Target setting for changing ADR 68K registers. If you want to change a register through MIDI, you must do it with a program change. How MIDI program changes correspond to registers is determined with the "Map" function on the main MIDI page, described earlier.

It might be a good idea, whenever you are working on a parameter map, to disable the Program Change map, so that you don't inadvertently change presets and lose your work. For example, if you are using a synthesizer as a master keyboard and you want to hear what the effect you were working on sounds like with a different synth patch, changing the synth patch will also send out a MIDI program change, which might call a new preset. You can avoid this by turning off the "PgChng" switch on the MIDI Setup page. This will not disturb any Program Change maps you have created, it will merely turn them all off temporarily.

If you do inadvertently send the ADR 68K a MIDI program change, remember that all is not lost — you can easily get back to the preset you were working in by pressing the "Last" button. Although you will not land back on the MIDI Parameter page, you can get to it quickly enough, and there you will find your parameter map intact.

Using the Parameter mode — examples

Since the ADR 68K software allows ten distinct parameter maps for each preset, there is a tremendous amount of flexibility and creative potential for MIDI control of the device.

Here are some examples of interesting possible uses of MIDI parameter control. Don't forget when trying these examples that you must leave the first MIDI Parameter page of the display for the incoming MIDI data to have any effect.

***Example 1*: Triggering Multiple Samples from a keyboard, and retriggering with Sustain Pedal.**

This is a typical mapping for use when you want to have the ADR 68K emulate a drum machine, with different samples playable from different keys. There are more notes (24) than there are samplers, so the ADR 68K will "interpolate" the extra notes, with the result that each sampler can be triggered by either of two adjacent notes. The second map provides velocity control over the main output level of each sample. The third map allows a press on the sustain pedal to re-trigger the last sample played, for a stuttering effect.

Original Preset: `M-12 [F6.3]`
Algorithm: `Mono-12-Sampler`

Map 1:
Source: `Note Trigger` — Target: `Fader 1 (Select)`
Source Low: `C 5` — Target Low: `1`
Source High: `B 6` — Target High: `12`

Map 2:
Source: `Note-On Velocity` — Target: `Fader 14 (Level)`
Source Low: `40` — Target Low: `-48dB`
Source High: `110` — Target High: `0dB`

Map 3:
Source: `Sustain Pedal` — Target: `Soft Key 6 (Play)`
Source Low: `65` — Target Low: —
Source High: `128` — Target High: —

If you wanted to control the duration of each sample as it played, instead of using "Note Trigger" in Map 1, you would use "Note Trigger/Release". Note that within the Mono-12-Sampler algorithm, you can have all 12 samples playing at once, while in the Mono* algorithm, or in any reverb or effects program in which samples are being used, only two samples can sound at a time — one odd-numbered sample, and one even-numbered one.

Example 2: **Sample Playback with Keyboard-Controlled Pitch Change, and Aftertouch-Controlled Vibrato, and Note-Controlled Stereo Position.**

This example uses MIDI control over sampling, if you'll excuse the expression, to the max. Pressing a note between C5 (Middle C) and B6 starts the sample playing (Map 1). The number of the note will change the playback pitch of the sample, up to one octave above or below its original pitch (Map 2). The playback level of the sample will be determined by the key velocity (Map 3). If the key is let go before the sample finishes playing, the sample will stop (Map 4). Any aftertouch generated by the keyboard will raise the playback pitch slightly, which can give the sample a bit of vibrato (Map 5). Finally, the stereo position of the signal is dependent on the pitch (Map 6), with the lowest note being sent to the Left channel and the highest to the Right.

Original Preset: `M [E6.1]`
Algorithm: `Mono 32sec Sampler`
Set the Following Control on Page 3: `Method:1/2st`

Map 1:
Source: `Note-On Number` — Target: `Soft Key 6 (Play)`
Source Low: `C 5` — Target Low: —
Source High: `B 6` — Target High: —

Map 2:
Source: `Note-On Number` — Target: `Fader 14 (Amount)`
Source Low: `C 5` — Target Low: `-1 Oct`
Source High: `B 6` — Target High: `+1 Oct`

Map 3:
Source: `Note-On Velocity` — Target: `Fader 19 (Level)`
Source Low: `40` — Target Low: `-46 dB`
Source High: `117` — Target High: `0 dB`

Map 4:
Source: `Note-Off Number` — Target: `Soft Key 5 (Stop)`
Source Low: `C 5` — Target Low: —
Source High: `B 6` — Target High: —

Map 5:
Source: `Channel Pressure` — Target: `Fader 15 (Fine)`
Source Low: `1` — Target Low: `0`
Source High: `128` — Target High: `+40`

Map 6:
Source: `Note-On Number` — Target: `Fader 20 (Balance)`
Source Low: `C 5` — Target Low: `→50`
Source High: `B 6` — Target High: `50←`

As previously noted, the behavior of the "Tuning" fader is not linear when "Method" is set to "%", but instead the control has an enlarged area around zero, to allow it to be centered easily. Therefore, MIDI note values should not be mapped to that fader in that mode, as the pitch-change intervals will not be even over the entire mapped scale. Mapping pitch wheel to the fader in that mode, however, works just fine.

Example 3: **Using Keyboard Aftertouch to control Chorusing.**

In this simple example, aftertouch (channel pressure) controls the Rate parameter of a chorus program, thereby bringing the chorused voices in as the key is pressed harder.

Original Preset: `Chor2 [F10.2]`
Algorithm: `Poly-Chorus`
Map 1:

Source: `Channel Pressure`	Target: `Fader 3 (Rate)`
Source Low: `1`	Target Low: `0.05 Hz`
Source High: `128`	Target High: `6.20 Hz`

Note that setting the Target Low limit to "OFF" would disable the chorussing completely, but it would also cause an audible click to be generated every time the channel pressure moved above zero. Therefore, leaving the chorussing on, but at a very slow rate, is the preferred solution.

Example 4: **Moving Resonant Filters with Pitch Wheel and "vibrating" them with Mod Wheel.**

Here a pair of resonant filters are generated, and then moved contrary to each other with the pitch wheel. At the same time, they can be set vibrating, out of sync and at different speeds, with the application of modulation wheel.

Original Preset: `Out-In [F11.1]`
Algorithm: `Dual Delays`
Set the Following Controls on Page 1:
Range: `Samples`, Gain: `96%`, Rate: `5.80 Hz`
Set the Following Controls on Page 2:
Range: `Samples`, Gain: `96%`, Rate: `2.40 Hz`

Map 1:

Source: `Pitch Wheel`	Target: `Fader 2 (Delay, DDL A)`
Source Low: `-4032`	Target Low: `1449us`
Source High: `4096`	Target High: `353us`

Map 2:

Source: `Pitch Wheel`	Target: `Fader 8 (Delay, DDL B)`
Source Low: `-4032`	Target Low: `928us`
Source High: `4096`	Target High: `1760us`

Map 3:

Source: `Mod Wheel`	Target: `Fader 5 (Depth, DDL A)`
Source Low: `1`	Target Low: `0%`
Source High: `128`	Target High: `60%`

Map 4:

Source: `Mod Wheel`	Target: `Fader 11 (Depth, DDL B)`
Source Low: `1`	Target Low: `0%`
Source High: `128`	Target High: `60%`

Example 5: Controlling pre-delay and early reflection times with tempo.

In this example, the pre-delay and early reflections are controlled by MIDI clocks generated by a sequencer. The pre-delay occurs one-quarter of a beat (a 16th-note) after the original signal, and the early reflections are spaced in quarter-beats after that.

Original Preset: `Voice [F2.1]`
Algorithm: `Optimal Chamber`

Map 1:
Source: `Midi Clock` | Target: `Fader 6 (P'Dly)`
Source Low: `120` | Target Low: `125ms`
Source High: `240` | Target High: `65ms`

Map 2:
Source: `Midi Clock` | Target: `Fader 22 (Left Refl 2)`
Source Low: `120` | Target Low: `250ms`
Source High: `240` | Target High: `125ms`

Map 3:
Source: `Midi Clock` | Target: `Fader 24 (Left Refl 3)`
Source Low: `120` | Target Low: `370ms`
Source High: `240` | Target High: `185ms`

Map 4:
Source: `Midi Clock` | Target: `Fader 20 (Right Refl 2)`
Source Low: `120` | Target Low: `250ms`
Source High: `240` | Target High: `125ms`

Map 5:
Source: `Midi Clock` | Target: `Fader 22 (Right Refl 3)`
Source Low: `120` | Target Low: `500ms`
Source High: `240` | Target High: `250ms`

Example 6: **Adjusting Stereo Processing Position and Width with Note Number and Velocity.**

In this map, the Stereo Processor algorithm is used to convert a mono input signal to stereo, using the Haas-effect "Pan" mode. The stereo position is determined by the note number, with lower notes appearing at the left side of the image. The width of the stereo image is determined by the note velocity, so playing a note harder will make it appear to come from a "larger" sound source. Both processors are active, so faders for both of them are mapped.

Original Preset: `Pan (F9.5)`
Algorithm: `Stereo Processor`

Map 1:
Source: `Note-On Number` — Target: `Fader 2 (Pan, Proc. A)`
Source Low: `C 3` — Target Low: `90←`
Source High: `C 7` — Target High: `→90`

Map 2:
Source: `Note-On Number` — Target: `Fader 8 (Pan, Proc. B)`
Source Low: `C 3` — Target Low: `90←`
Source High: `C 7` — Target High: `→90`

Map 3:
Source: `Note-On Velocity` — Target: `Fader 3 (Width, Proc. A)`
Source Low: `20` — Target Low: `0%`
Source High: `110` — Target High: `49%`

Map 4:
Source: `Note-On Velocity` — Target: `Fader 9 (Width, Proc. B)`
Source Low: `20` — Target Low: `0%`
Source High: `110` — Target High: `49%`

Example 7: **Playing Back a Sample with Reverb Mix dependent on velocity.**

This map gives the effect of a sampled sound moving closer as it gets louder. As note velocity increases, the reverb level goes down and the dry-sample level goes up, while as the velocity goes down, the reverse effect occurs.

Original Preset: `Voice [F2.1]`
Algorithm: `Optimal Chamber`
Map 1:
Source: `Note-On Velocity`
Source Low: `20`
Source High: `110`

Target: `Fader 35 (Input Level)`
Target Low: `-9dB`
Target High: `-34dB`

Map 2:
Source: `Note-On Velocity`
Source Low: `20`
Source High: `110`

Target: `Fader 41 (Main Output Level)`
Target Low: `-37dB`
Target High: `0dB`

Some words to the wise regarding MIDI parameter control: Some parameters work better under real-time control than others. In the chorus example above, we saw that turning on and off the chorussing in real time can cause clicks. Moving the Begin and End faders in a sampling program with MIDI controllers does not have the effect of starting the sample at different times — instead, it produces the effect of *editing* the sample while you play it. While there might be some potential in this effect, it is probably rather limited. Be careful not to create maps so complex that you lose track of what's doing what, or that the various parts start to get in each others' way.

When using multiple samples, remember the restrictions on sample playback in the "*" programs or within reverb and effects programs: *Only two mono samples can be sounding at a time, and then only if one is odd-numbered and the other even-numbered. Only one stereo sample can be sounding at a time. New sample triggers will cut off already-sounding samples.*

These examples just scratch the surface. The creative engineer will be able to think of many more innovative ways to use MIDI with the ADR 68K. We at AKG Digital welcome your examples, your suggestions, and your ideas for any changes or improvements you would like to see.

rev 4.0, 15 December 1987

— Appendices —

In these three appendices you will find various useful items.

Appendix A is an extended glossary containing definitions of many of the terms used in this manual and on the ADR 68K's display.

Appendix B contains a standard MIDI implementation chart for the rev 4.0 software, and a list of the MIDI commands that are generated when the ADR 68K is in the "Auto-MIDI" mode.

Appendix C contains charts and diagrams of the various algorithms resident in this version of the software. Each algorithm is covered in two pages. The first page gives general information about the algorithm, and the location of the various parameters that are used in that algorithm, as well as their maximum and minimum values. Also on the first page is a list of all factory presets which use that algorithm. The second page of the pair is a block diagram of the algorithm, showing the paths from all inputs, through the controllers and processors, to the various outputs.

— Appendix A — Extended Glossary

Algorithm is a way of defining the ADR 68K's behavior. It includes the type of processing that is done on an audio signal, as well as definitions of controls, parameters, displays, how memory is organized, and various other aspects. It is a structure for building presets; a specific set of control settings within an algorithm constitutes a **preset** (see also).

Amount determines the change in pitch of a sample as it is being played back. A sample can be pitch-shifted up to one octave up or down. The units of operation of this fader are set with the **Method** fader, and can be percentages, half-steps, or musical intervals, e.g., minor 2nd, augmented 4th, etc. The only sampling programs that have this control are the ones that work with just one sample at a time, e.g., M and St. See also **Tuning**.

Attack is a parameter on the **Envelope** page (see also) of the Sampler programs that allows you to impose a rising volume characteristic when playing back a pre-recorded sample, much as a synthesizer has an attack setting. The range is 0 to 970 msec. See also **Decay**.

Auto-MIDI is a special "automation" mode of the ADR 68K in which each fader movement or button press of the remote controller is sent out as MIDI continuous controller data, in which form it can be recorded and played back by an external sequencer, which can then in turn control the ADR 68K. It is designed to be invisible to the user, and should only be used in a linear fashion — i.e., with little or no editing or overdubbing. It is enabled from the Setup Page of the MIDI menu; **Xmit** starts the data transmitting, and **Recv** allows the unit to be controlled by incoming MIDI data.

Aux mix is a software switch in the Split programs that allows the auxilliary output mix to be routed to the main outputs. Normally, the output from one half of the split is available only at the main outputs, and the output from the other half is available only at the aux outputs, but this switch makes both halves available at the main outputs.

Aux outputs are the second set of stereo outputs on the mainframe. They can be configured to put out the same signal as the main outputs, or something completely different. In most algorithms, the aux outputs have their own mixer page.

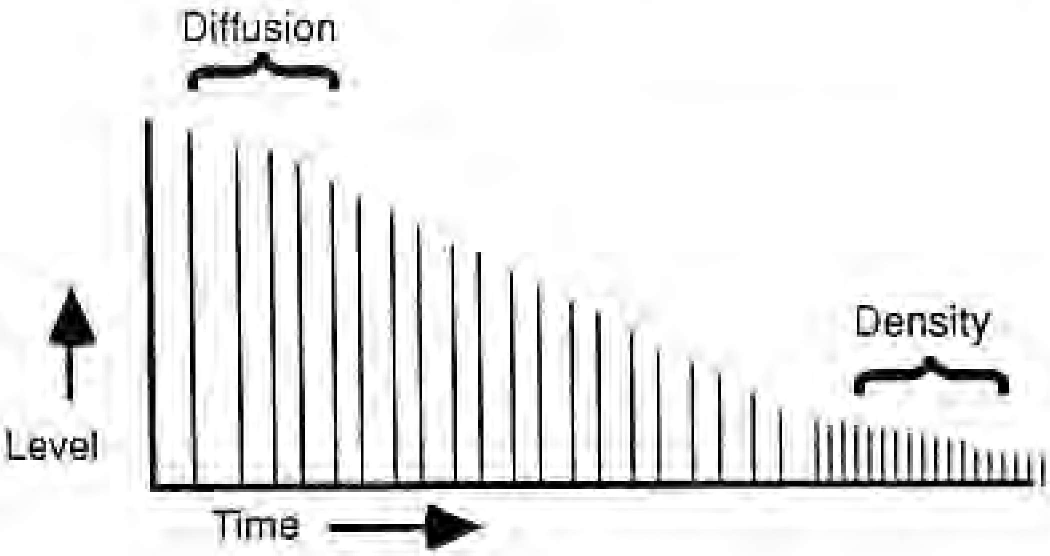
Balance (Bal)	is a control for handling mono samples that appears on most mixer pages, both inputs and outputs. It is calibrated in percentages, from "100←" (full Left), through "–^–" (center), to "→100" (full Right). Do not confuse this with the **Pan** control that appears in the Stereo Processing programs.
Bank	A group of registers in the Internal, Cartridge, or Factory memories, in which are stored presets. In general, presets within a single bank should all share the same algorithm, although this rule can be violated for special purposes. (See **Preferred Presets**.)
Begin	in a Sampler program, refers to the beginning point of an edited sample. When the Begin fader is moved, 20 msec-long "slices" of the sample are played, starting at the time indicated in the display above the fader. Moving the fader above or below the center point of its travel causes the start time of the slice to move forwards or backwards, respectively. The further away from the center point the fader is moved, the faster will be the playback speed of the sample slices. Better resolution can be had by using the "trim" fader next to the Begin fader, or the Soft Button above it, which give 1 msec resolution and play 1 msec slices.
Bypass	in the Multi-Effects algorithm, takes various modules out of the circuit, and allows signal to pass through them unaffected. Contrast this with **Mute**, which shuts off the output of a module.
Cold Reset	is a way to "restart" the ADR 68K if you run into trouble. It will erase any internal registers, MIDI maps, and samples, so it is not recommended it be performed indiscriminately. It is performed by turning off the power, waiting 10 seconds, and then holding the "Help" button while turning on the power, and pressing the "COLD" button when the display appears.
Config	In the DDL algorithm and some of the Split algorithms, determines how the two processor sections are to be aligned: in Parallel as two separate delays treating the same signal, Chained with one delay feeding the other, or Split with each delay processing its own signal.
Current Register	is an area of memory in which are located the control settings that appear on the display at any time. When you save a preset into a numbered register, it's the contents of the Current register that get saved. When you call up a preset, the contents of the register you specify are loaded into the Current register. When you press the Last button, the contents of the **Last register** (see also) and the Current register are exchanged.

Decay is a parameter on the Envelope page (see also) of the Sampler programs that allows you to impose a falling volume characteristic at the end of a pre-recorded sample, much as a synthesizer has a decay setting. The range is 0 to 970 msec. See also **Attack.**

Delay in the DDL and Multi-Effects algorithms, the length of the delay of the first echo.

Delta The distance between taps, or voices, in a Chorus preset. Short Delta values give resonant-filter effects, while longer ones can be heard as discrete echoes.

Density refers to the concentration of echoes at the end of the reverb (the "tail") in some reverb algorithms. At a setting of 0, one is liable to hear more discrete echoes, and the sound can get "lumpy". At 100, the sound is smoothest. Lower density settings are good for making vocals more intelligible and solo instrument lines more distinct. Higher settings are best for drums. See also **Diffusion.**



Depth has various meanings, depending on the context.

In reverb algorithms, it controls the apparent distance of the listener from the source, by mixing the direct and reflected sound. At a setting of 0, the direct sound is highest, putting the listener right at the front of the stage. At 100, the reverb is at its highest, putting the listener at the back of the hall.

The control does much more than just balance wet and dry sound; it ramps the various taps back and forth in a complex manner that maintains the naturalness of the sound.

In some modes of the Stereo Processing algorithm, it gives the synthesized stereo space a sense of distance. At 0, the image is two-dimensional, with left and right imaging only. At higher values, it gives the illusion of a third dimension.

In the Poly-Chorus algorithm it refers to the time span within which the taps will be modulated.

Diffusion	refers to the concentration of echoes at the *beginning* of the reverb in many of the reverb algorithms. Although it is usually best left at a high value, low values can make smaller rooms more interesting, with discrete echoes at the beginning of the reverb becoming more audible. (See also **Density**.) Diffusion is also used in the Poly-Chorus algorithm to spread out the sound before it gets to the chorus processing.
End	in a Sampler program, refers to the end point of an edited sample. When the End fader is moved, 20 msec-long "slices" of the sample are played, starting at the time indicated in the display above the fader. Moving the fader above or below the center point of its travel causes the start time of the slice to move forwards or backwards, respectively. The further away from the center point the fader is moved, the faster will be the playback speed of the sample slices. Better resolution can be had by using the "trim" fader next to the End fader or the Soft Button above it, which give 1 msec resolution and play 1 msec slices.
Envelope	is the page or portion of a page within the sampler programs that allows you to impose an envelope on a recorded sample. The parameters it comprises are **Attack** and **Decay**.
Envlp	in the Stereo Multi-Tap and Reverb Multi-Tap modules of the Multi-Effects algorithm, refers to the volume envelope of the echoes; whether they will be increasing, decreasing, or steady (even). The rate of change for increasing or decreasing is approximately 6 dB per echo.
Equalizer	in Multi-Effects and Stereo Processing presets, is a two-band shelving equalizer, with 10 dB of boost or cut at approximately 250 Hz and 6 kHz.
E-Rs	see **Reflections**
Fader	refers to one of the six slide controls on the remote unit. The action of the faders is controlled entirely by the system software, so what they do at any one time will be completely dependent on the context.
Gain	in DDL and Multi-Effects programs, the level at which the delays feed back to themselves. At 100%, the delays repeat indefinitely without changing level. At 0%, they don't repeat at all. At negative settings, the repeats are reversed in phase.

Gate — comprises two parameters, Trig and StpDly, which together determine at what signal level, and how fast, the reverb characteristics of a preset will change between Running Reverb and Stopped Reverb.

HF BW — adjusts a low-pass filter which can be electronically located in any of several places, depending on the algorithm. The range of this filter is 2 - 15 kHz.

Image — is a parameter in the Stereo Processing algorithm that only appears in the "Pan" mode. Image sets the apparent position of the signal, from 90← (90° to the left of center), through 0 (center), to →90 (90° right of center).

Init — erases all of the presets in the Cartridge. It is accessible from the Cart menu.

Input — on a mixer page, determines how much, if any, the input signal will be attenuated before it reaches the processor.

Last register — is an "invisible" register wherein reside the contents of whatever preset was in use *previous* to the preset in use now. It is useful when editing presets for comparison purposes, in that it allows instant switching between a new version of a preset and its previous version. It is also useful for editing samples within the context of a reverb or effects program, as it allows instant switching between the reverb program and the sampling program. See also **Current register.**

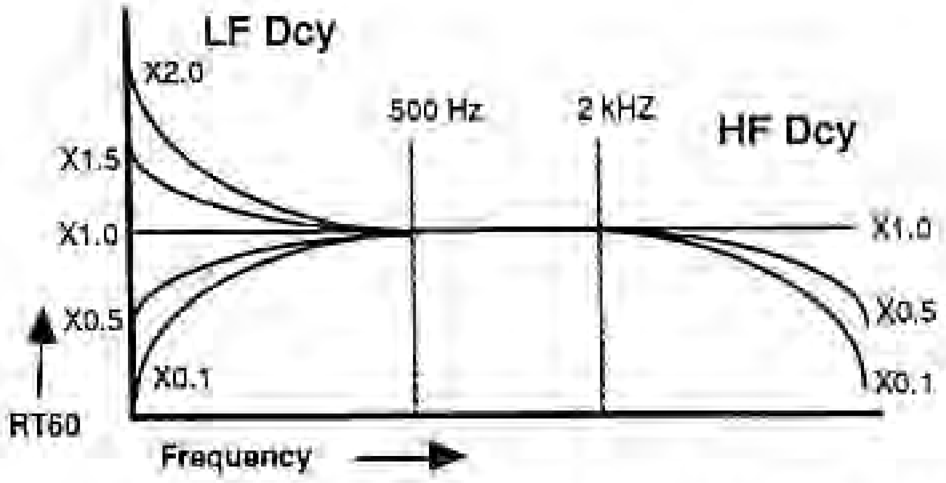
Length — in the Stereo Multi-Tap and Reverb Multi-Tap modules of the Multi-Effects algorithm, refers to the overall length of the series of echoes produced.

LF and HF — are the labels for the two-band equalizer in the Stereo Processing algorithm. See **Equalizer.**

LF and HF Dcy — are factors by which the relative RT60 can be changed for two specific frequency bands. The HF band is the spectrum above approximately 2 kHz, while the LF band is the spectrum below approximately 500 Hz.

An HF Dcy of "x0.6" in a program whose RT60 is 3.0s means that the high frequencies will decay in 1.8 seconds. In all programs, the Dcy factors range at least from x0.1 to x1.0; in some programs the LF Dcy can be increased to x2.0.

In Gated reverb programs, the Running reverb and Stopped reverb each have their own LF and HF Dcy adjustments, so that completely different room characteristics can be set up for the two sections.



HF Dcy is also used in the DDL and Multi-Effects algorithms, where it controls the ratio of high frequencies in the delays.

Map refers to two different MIDI functions.

From the main MIDI page, pressing "Map" allows you to specify which presets within the ADR 68K will be called up when the unit receives particular MIDI program change numbers. More than one program change number can be assigned to the same preset, but not *vice versa*. A MIDI program change number can also be mapped to a "non-existent" register, in which case it is ignored. Program change maps remain in non-volatile memory.

In the MIDI Parameter menu, the meaning is quite different. Within a preset, up to 10 parameter maps can be created, each one with a MIDI **Source**, which is a type or class of MIDI command; a **Target**, which is a control parameter within the ADR 68K; and **Ranges** for the Target and Source. The "Map#" fader allows you to choose which map to work on. Parameter maps are stored as *part of a preset*, and if you configure a set of maps and then leave the preset without storing it, the maps will be lost.

Memory refers to the *type* of storage location of a **register**, in which resides a preset. The ADR 68K has a Factory memory, which contains 11 **banks** of registers, the presets in which are unmovable and uneditable. It also has Internal and Cartridge memories of 11 banks each, where presets can be edited and stored. The number of presets in the Factory memory is determined at the factory. The number of presets in the other memories is determined by the user, but each memory can hold a maximum of 50 presets.

Method determines how the pitch change of a recorded sample will be calculated. The choices are percentages, musical half-steps, or musical intervals, e.g., minor 2nd, augmented 4th, etc. The only sampling programs that have this control are the ones that work with just one sample at a time, e.g., M and St. See **Tuning** and **Amount**.

Mode has several meanings, depending on the context.

In the Stereo Chorus module of the Multi-Effects algorithm, it specifies whether the various taps will move in parallel or opposite (contrary) motion when modulated.

In the Multi-Tap modules of the Multi-Effects algorithm, it determines the pattern of echoes produced, in terms of relative volume, timing, and stereo position.

In the Stereo Processing algorithm, it determines the type of processing (multitap, comb filtering, Haas-effect image shift, etc.) used.

Mute has two meanings. It refers to the button on the remote control that shuts off all input coming into a processor or a sampler recorder.

It also can refer to a parameter setting, in which case it means that the effect in question is inactive. It is used in gated reverb programs, in which case it disables the Stopped Reverb parameters, and shuts down the output completely when the input signal goes below the Trig level. It is also used to disable and turn off the outputs of the three "parallel" modules in the Multi-Effects program. (See also **Bypass**.)

Pan has two meanings. Within the Stereo Processing algorithm, it refers to a mode that produces an apparent position shift of a monaural signal across a stereo field, but it uses time delay (the Haas effect) instead of volume, so that the energy in both channels is always the same.

In the Poly-Chorus algorithm, it refers to the frequency of a triangle wave that modulates the outputs, panning them back and forth across the stereo field. (Don't confuse this with **balance**, which refers to stereo placement of the output signals on the mixer pages.)

Play refers to both a Soft Button and a fader in Sampler programs, which determine how samples will be played back. If a Play fader is set to MAN, then the corresponding sample can be triggered by pressing the Play Soft Button. If it is set to RPT, the sample will play indefinitely until the fader is reset or the CLR button is pressed. If it is set to a level between -42 dB and "LIM", an incoming signal on the appropriate channel (as determined by the **Source** fader — see also) which reaches that level will start the sample.

Whenever a sample is playing, whether in a sampler or effects program, pressing Shift-Play will get it to stop immediately.

Pre-Delay (P'Dly)	is a time delay given to a signal before it goes into a reverb processor. A P'Dly of 0 does not mean that the reverb starts immediately; there is always an inherent delay in the build-up of a reverb sound. The P'Dly setting does not affect the Early Reflections. In the Multi-Effects algorithm, P'Dly is one mode of operation of the Reverb Multi-Tap page, in which only a single delay is produced.
Preferred Preset	is the preset that is located in the first, or ".1" position of each of the Internal banks. It can be called very quickly by merely pressing the appropriate bank button. This feature can be very useful in a mixing session, in which case the presets that will be used most often can be loaded into the "preferred" positions. This is one time when violation of the "one-bank, one-algorithm" rule is advisable. If the .1 register of an Internal bank is empty, pushing the bank button will call up the .1 register in the corresponding Factory bank.
Preset	is a set of specific control settings (i.e., RT60, HF Dcy, Sampl Trig, etc.) within an algorithm. Put another way, it is a specific way to *use* an algorithm. Presets have names, and are stored in registers, which have numbers. Presets can be in the Factory memory, in which case they are unmovable and uneditable, or in the Internal or Cartridge memories, where they can be edited.
Program	Although some engineers are used to another definition of this word, for purposes of this manual, it is to be considered a synonym for Preset.
Punch	is a parameter in some of the reverb algorithms that increases the proportion of early echoes within the reverb processor (not the separate early reflections — see below), making the reverb effect more immediate and dramatic. In the Reverse Reverb algorithm, Punch controls the level of the *last* sound heard — which is the image of the first sound going in.
Quick Register Select	is a function allowing you to get to a specific register or preset without going through the menu pages. Type the bank number of the register you want, and the Soft Button for the appropriate memory, if necessary. Then type the decimal point and the register number, and then "ENT".

Range has two meanings. In the Stereo Chorus module of the Multi-Effects algorithm, it determines how far in time the various taps will move when modulated. This parameter, in conjunction with **Speed**, will determine the cycle time of a vibrato or other effect produced in this module.

In the DDL algorithm, range is a "coarse delay" control, which sets the range and the resolution of the "Delay" fader next to it. The lowest setting for Range is "Sampls", in which the Delay fader operates from 160 µsec to 1760 µsec, in steps of 32 µs; the highest setting is "Long", in which the Delay fader goes from 0 msec to 2000 msec in 40 ms steps.

Rate in the Chorus and DDL algorithms refers to the frequency of a triangle wave that modulates the various taps.

Rcv on the MIDI Setup Page, enables real-time reception in the Auto-MIDI mode. Not to be confused with...

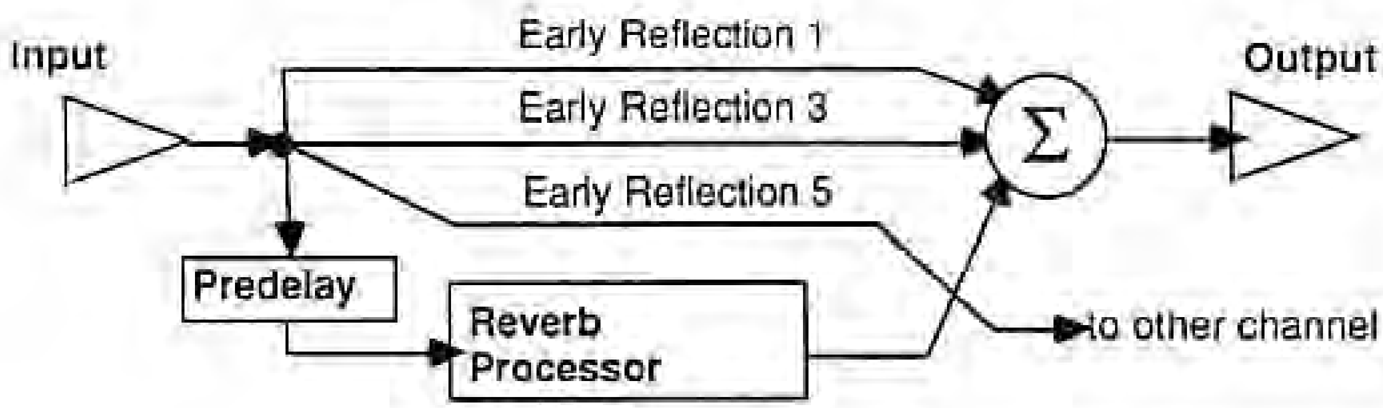
Receive is a MIDI function that allows the ADR 68K to receive bulk MIDI information. This can be a complete set of Internal registers from an outside source, such as a computer or another ADR 68K, in which case all the presets currently in the Internal memory will be erased; or it can be a sample, sent to the unit via the MIDI Sample Dump Standard or System Exclusive. The ability to receive MIDI data in *real time* is not controlled by this parameter.

Record in a sampler program, refers to both a fader and a Soft Button. If the fader is set to MAN, recording will take place *only* when the Soft Button is pressed. If the fader is set to a signal level, from -42dB to "LIM", then recording will start automatically as soon as an incoming audio signal (on the appropriate channel) reaches that level, *or* whenever you press the Soft Button.

Reflections or **Early Reflections (E-Rs)** are up to six discrete echoes within a reverb algorithm that are completely separate from the reverb processors. Early reflections play a large part in determining the apparent size, shape, and surface hardness of the generated space, as well as enhancing our sense of "stereo".

In programs with true stereo inputs, there are three separate early reflections, each adjustable for delay time and level. Two of these signals are then sent to the output of the same channel, while the third is sent to the opposite channel. In Split programs, each input generates two reflections, one going to the Left output and the other going to the Right ouput.

In each preset that uses reflections, there are one or more pages for balancing the individual presets, and a separate control for balancing the *overall* level of the reflections with the rest of the sound.



Register — is a memory location, in which a **preset** is stored. Registers reside in **banks**, which in turn reside in one of three **memories**. When it is being called up from memory, or stored to memory, a register is referred to by its number; e.g. "C9.13" means the 13th register in bank 9 of the Cartridge memory.

Reset — is a way to "restart" the ADR 68K if you run into trouble, without turning off the power. It will erase any internal registers, MIDI maps, and samples, so it is not recommended it be performed indiscriminately. It is performed from the SYS menu.

RT60 — is the overall reverb time in a reverb algorithm. More specifically, it is the time it takes for the reverb level to decay to 60 dB below its initial level. In Gated reverb programs, there are two RT60s, one for the primary, or "Running" reverb, and one for the secondary, or "Stopped" reverb. In the Reverse Reverb algorithm, the RT60 is the length of the overall reverb, from start (at low level) to finish (at high level).

Running Reverb — refers to the "primary" reverb settings in a reverb preset that uses gating. The running reverb parameters are active when the input signal level exceeds the "Trig" threshold. When the "Trig" parameter is set to "OFF", there is no gating action, and the running reverb settings are the only ones active. (See also **Stopped Reverb**).

Safety — is a switch used in Sampler programs. When a sampler is empty, the switch is off, meaning that recording can take place. Once signal is recorded, however, the Safety automatically turns on, and before the sampler can be recorded into again — thereby wiping out its existing contents — it must be turned off. It is also used to "activate" an unrecorded sample, thereby making that particular sample memory available for use with other parts of the sample RAM.

Sample Trig

refers to controls within reverb and effects programs that are used to determine how pre-recorded samples will be played back. They serve the same function as the Play buttons and sliders within the sampler programs themselves. Each sample in an effects program can be triggered separately: the first Sample Trig ("Select") fader selects which sample you want to adjust, and the second fader does the actual adjusting. When you set the second fader to MAN, the sample in question can be triggered *only* by pressing the Soft Button over the Select fader. You can also set it to RPT, in which case it will play indefinitely until the second fader is reset or the CLR button is pressed; or to a level between -42 dB and "LIM", in which case an incoming signal on the appropriate channel (as determined within the sampling program itself, using the "Source" fader) that reaches that level will start the sample.

Select

This fader and Soft Button appear whenever multiple samples have to dealt with — when recording, editing, mixing, or playing back, either within a sampling program or in a reverb or effects program. The fader chooses which of the samples to work on, i.e., which one will be affected by the rest of the controls on the page or portion of the page. The Soft Button does *not* increment the Select number — instead, it acts as a "Play" button for the sample selected by the fader. This gives you the ability to hear a particular sample in context without constantly going back and forth between pages and programs. Setting the Select fader on any page also selects it on all other pages within the current preset.

Send

is a MIDI function that allows the ADR 68K to send bulk MIDI information. This can be the parameters that determine presets (either the current register or all Internal registers) in system-exclusive format, or a sample, either in system-exclusive or MIDI Sample Dump Standard format. The function does not affect the internal memory of the ADR 68K at all. The ability to send MIDI data in *real time* is not controlled by this parameter; that is the function of the **Xmit** switch (see also).

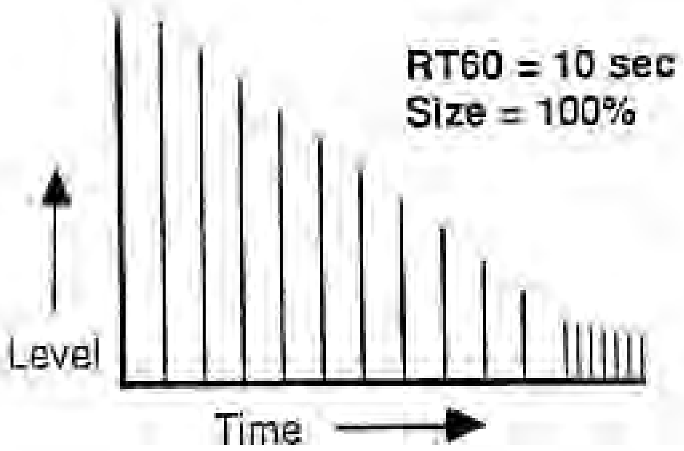
Set

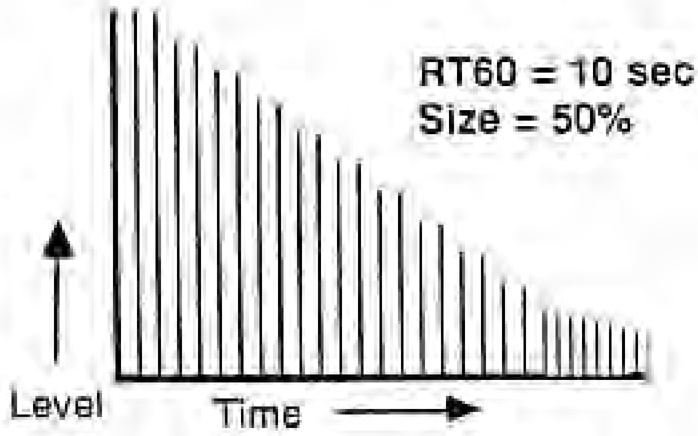
is a "quick-save" command that lets you save the contents of the Current register into the register from which it came — i.e. the register whose number is showing in the top line of the display. If that register is in a Factory bank, this function will not work.

Size is a complex parameter used in reverb algorithms, and expressed in percent. It simultaneously changes the RT60 and the density of the reflections within the reverb (not the early reflections).

For a particular setting of RT60, a Size setting of 100% represents a typical room with that decay time. Increasing the Size increases the RT60, but it also reduces the density of the reflections. In this case, it is more likely that individual echoes will be heard within the reverb, but at the same time the frequency coloration of the reverb will be minimized. Decreasing the Size lowers the RT60, increases the density, reduces the chance of echoes being audible, and increases the coloration.

Depending on the algorithm, the Size control can be set anywhere from 15% to 200%.





S-Mtap stands for "Stereo Multi-Tap", and refers to that module (Page 4) in the Multi-Effects algorithm on the algorithm's output mixer page.

Soft Button refers to one of the six buttons (with the triangles inside) immediately above the faders. They are used to select menu items and sometimes to augment the functions of the faders below them, by acting as switches, or incrementing controls. When a button is acting to increment a value, often pressing the Shift key and the button will make it decrement the value. Depending on the context, Soft buttons will often have auto-repeat and acceleration characteristics.

Source has two meanings:

In the sampler programs, it determines from which audio input, Left or Right, a sample program will record signal *or*, once the sample is recorded, which input will trigger playback.

On the MIDI Parameter Control page, it refers to what type of MIDI data will be used to control what function of the ADR 68K (see **Target**).

Speed in the Stereo Chorus module of the Multi-Effects algorithm, refers to how fast modulation of the various taps will occur. This parameter, in conjunction with Range, will determine the cycle time of a vibrato or other effect produced in this module.

Stop is a control that appears on the first page of all sampling programs. It is used to cut off either recording or playback of a sample before it reaches its endpoint. Samples can be stopped from other pages by pressing the Shift key and any "Play" button.

Stopped Reverb refers to the "secondary" reverb settings in a reverb preset that uses gating. The stopped reverb parameters are active before the input signal level exceeds the "Trig" threshold, as well as after the level drops below the threshold, and the StpDly interval has passed. If the "Trig" level is set to "OFF", there will be no Stopped reverb, and these settings will have no effect. (See also **Running Reverb**).

StpDly In a gated reverb preset, after the input signal level drops below the Trig threshold, a specified period of time before the Stopped Reverb parameters take over. Adjustable between 0.01 and 5 seconds.

Swap a function, called up from the Cart menu, that exchanges the contents of the Internal registers with those of the Cartridge registers in a non-destructive way.

SysEx is a MIDI function that allows the ADR 68K to be controlled via System-Exclusive data externally. The usual way of operating the ADR 68K over MIDI in real time is to use normal ("System Common") messages such as notes and controllers, but System-Exclusive control can be used as well, in conjunction with a dedicated software program. This function and the **Send** and **Receive** functions are completely separate.

Target, Target # In the MIDI Parameter mode, a Target is an ADR 68K parameter that is being controlled by a specific MIDI command. A target can be a fader, a Soft Button, a Soft Button with Shift key, the Mute button, or the Clear button. The "Target" fader on Page 2 of the MIDI Parameter menu chooses the type of target, while the "Target#" fader chooses the specific target, if there is more than one of that type, e.g., Soft Button 12, which refers to the sixth Button on Page 2 of the current preset. When the Target# fader is moved, the display shows the actual identity of the chosen target within the context of the current preset, e.g. selecting Fader 15 will display "Bal". See **Source**.

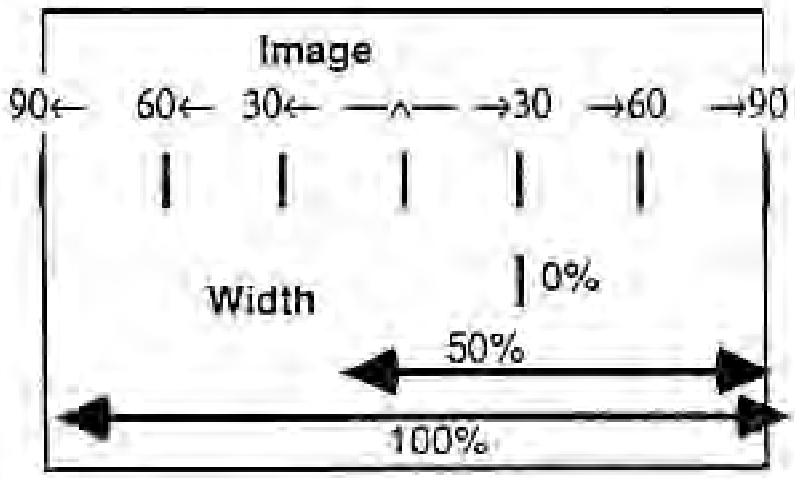
Trig — sets a level, between -42 dB and "LIM", at which an incoming audio signal will open the gate in a reverb algorithm, and therefore allow the Running Reverb parameters to take over. (Not to be confused with **Sample Trig.**)

trim — See **Begin** and **End**

Tuning — refers to the parameters within a sampling program that control pitch change on playback. They are **Method**, **Amount**, and **Fine**. The only sampling programs that have this control are the ones that work with just one sample at a time, e.g., M and St.

Voices — in a Chorus or Multi-Effects program controls the number of voices, or taps, that will be generated. Generally speaking, the more voices the "thicker"-sounding the effect.

Width — in some modes of the Stereo Processing algorithm, is used to control how wide the apparent stereo spread of the synthesized signal will be. A width of 0% means that the output of the algorithm will be monaural, while a width of 100% means that the full stereo field will be used. In the PAN mode of this algorithm, Width has a slightly different meaning: it determines how wide the apparent sound source will be. At 0%, the signal will appear to be a point source, while at 100%, it will take up the full stereo stage (and the pan effect will be lost).



Xmit — turns on the ADR 68K's **Auto-MIDI** mode (see also) from the Setup Page. When this is enabled, any movement of a fader or press of a button is sent out as MIDI controller data.

— Appendix B —
MIDI Charts

Function	Transmitted	Recognized	Remarks
Basic Channel	x	1-16	memorized
Mode	x	1 (Omni On, Poly) 3 (Omni Off, Poly)	memorized
Note Number	x	o	
Velocity	x	o	
Aftertouch	x	o	
Pitchbender	x	o	
Control Change	o	o	Only General Purpose Controllers transmitted
Program Change: true #	x	o 0-127	
System Exclusive	o	o	
System Common	x	x	
System Real Time	x	x	
Aux messages	o	o	MIDI Sample Dump Standard

o : Yes
x : No

ADR 68K MIDI Implementation Chart
Version: 4.00
Date: 15 December 1987

Auto-MIDI values

The following are the MIDI values sent by the faders and by the pressing and releasing of buttons on the ADR 68K remote when the Auto-MIDI mode is active (i.e., "Xmit" is turned on).

The faders link directly with the following controllers, and their position is indicated by a value of 0-127:

Fader 1	=	General Purpose Controller #1 (Continuous Controller #16)
Fader 2	=	General Purpose Controller #2 (Continuous Controller #17)
Fader 3	=	General Purpose Controller #3 (Continuous Controller #18)
Fader 4	=	General Purpose Controller #4 (Continuous Controller #19)
Fader 5	=	General Purpose Controller #5 (Continuous Controller #80)
Fader 6	=	General Purpose Controller #6 (Continuous Controller #81)

The buttons use General Purpose Controllers #7 and #8 (Continuous Controllers #82 and #83), to signify a button push and release, respectively, and each button is assigned a particular *value*. For example, pushing Soft Button 1 sends a Controller #82 value of 28 (decimal), while releasing it sends a Controller #83 value of 28.

Button	Decimal	Hex
0	0	00
1	1	01
2	2	02
3	3	03
4	4	04
5	5	05
6	6	06
7	7	07
8	8	08
9	9	09
decimal pt.	10	0A
BSP	11	0B
CE	12	0C
ENT	13	0D
Shift-0	14	0E
Shift-1	15	0F
Shift-2	16	10
Shift-3	17	11
Shift-4	18	12
Shift-5	19	13
Shift-6	20	14
Shift-7	21	15
Shift-8	22	16
Shift-9	23	17
Shift-dec pt.	24	18
Shift-BSP	25	19
Shift-CE	26	1A
Shift-ENT	27	1B

Button	Decimal	Hex
Soft 1	28	1C
Soft 2	29	1D
Soft 3	30	1E
Soft 4	31	1F
Soft 5	32	20
Soft 6	33	21
Shift-Soft 1	34	22
Shift-Soft 2	35	23
Shift-Soft 3	36	24
Shift-Soft 4	37	25
Shift-Soft 5	38	26
Shift-Soft 6	39	27
↑	40	28
↓	41	29
←	42	2A
→	43	2B
Shift-↑	44	2C
Shift-↓	45	2D
Shift-←	46	2E
Shift-→	47	2F
MUTE	48	30
CLR	49	31
Shift-MUTE	50	32
Shift-CLR	51	33

Button	Decimal	Hex
PLATE	52	34
ROOM	53	35
CHAMB	54	36
HALL	55	37
SPLIT	56	38
EFX	57	39
LAST	58	3A
Shift-PLATE	59	3B
Shift-ROOM	60	3C
Shift-CHAMB	61	3D
Shift-HALL	62	3E
Shift-SPLIT	63	3F
Shift-EFX	64	40
Shift-LAST	65	41
MIDI	66	42
SYS	67	43
CART	68	44
HELP	69	45
Shift-MIDI	70	46
Shift-SYS	71	47
Shift-CART	72	48
Shift-HELP	73	49

BANK 1 — Complex Plate (PLATE)

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √
Inputs: true stereo

Gating Capability √
Aux outputs: samples only

Complex Plate is a broadly applicable fast-diffusing, high-density, uncolored simulation of a plate reverberator. It is a good choice for percussion and popular music, for which the gated settings are especially useful. True stereo inputs allow for stereo perspective of early reverb sound if desired. Only the dry samples appear at the aux ouputs — they are not sent the reverb or early reflections.

The following are the Factory presets using this algorithm:

1.1 **Warm** — nominal plate setting, a good "starter" sound for using or customizing.

1.2 **Short1** — short decay time, good for percussion.

1.3 **Short2** — slightly longer than above.

1.4 **Crisp** — moderate decay with emphasized high end.

1.5 **Sizzle** — an effect like frying bacon.

1.6 **Gate1** — boomy gated sound, with low-end emphasis.

1.7 **Gate2** — tighter, shorter gated sound.

1.8 **Gate3** — similar to above.

1.9 **Gate 4** — similar, with more emphasized high frequencies.

1.10 **Gate5** — extremely boomy gated sound.

1.11 **Thud** — dull, boomy room.

1.12 **TakOff** — special effect reverb, small space with extremely long decay.

1.13 **BHolly** — Moderate reverb, good for electric guitar.

1.14 **Thick** — a reverb with very dense early reflections.

1.15 **Cavern** — a long, hollow-sounding space.

1.16 **Reggae** — a special effect, with reflections set to long delays for multiple "slaps".

1.17 **Jump!** — bombastic reverb.

1.18 - 1.20 **Gate6, Gate7, and Gate8** — tight gates with muting.

Page 1 — Running Reverb

	RT60	LF Dcy	HF Dcy	HF BW	Size	P'Dly
max	INF	X2.0	X1.0	15kHz	150%	500ms
min	0.22s	X0.1	X0.1	2kHz	15%	0ms

Page 2 — Details / Sample Trig

	Punch	Diffus	Select	Play
max	100%	100%	12	MAN/LIM
min	0%	0%	1	-42dB/RPT

Page 3 — Gate / Stopped Reverb

	Trig	StpDly	RT60	LF Dcy	HF Dcy
max	LIM	5.00s	INF	X2.0	X1.0
min	OFF/-42dB	0.01s	0.22s/MUTE	X0.1	X0.1

Page 4 — Reflections for Left Outputs

	1 (Lin)		2 (Rin)		3 (Lin)	
max	0dB	500ms	0dB	500ms	0dB	500ms
min	OFF	0ms	OFF	0ms	OFF	0ms

Page 5 — Reflections for Right Outputs

	1 (Rin)		2 (Lin)		3 (Rin)	
max	0dB	500ms	0dB	500ms	0dB	500ms
min	OFF	0ms	OFF	0ms	OFF	0ms

Page 6 — Input Mix

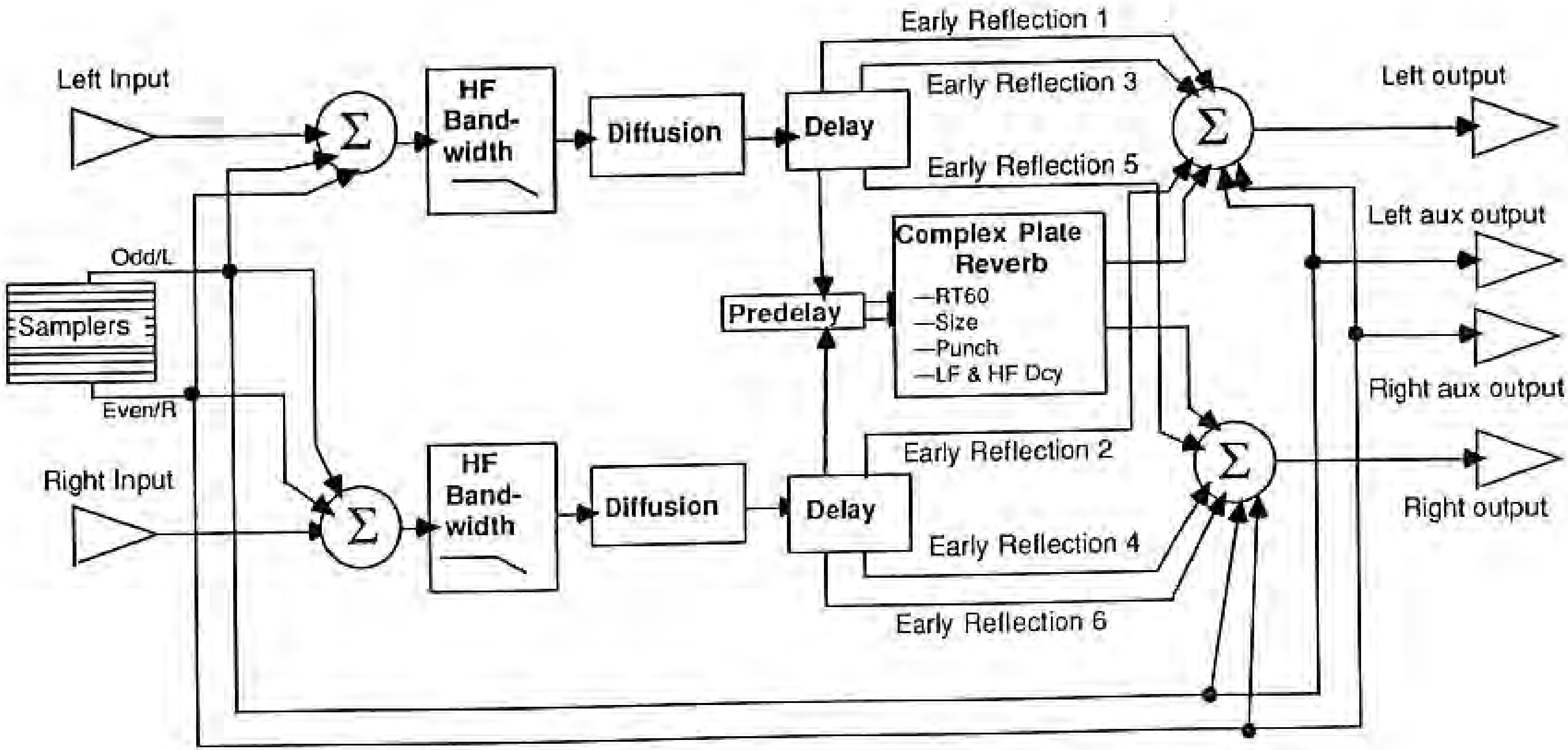
	Input	Select	Level	Bal
max	0dB	12	0dB	100←
min	OFF (/-68dB)	1	OFF	→100

Page 7 — Main Output Mix

	E-R's	Rev'b	Select	Level	Bal
max	0dB	0dB	12	0dB	100←
min	OFF (/-68dB)	OFF (/-68dB)	1	OFF	→100

Page 8 — Aux Output Mix

	Select	Level
max	12	0dB
min	1	OFF (/-68dB)



Bank 1 — Complex Plate (Plate)

BANK 2 — Optimal Chamber (CHAMBER)

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √
Inputs: true stereo

Gating Capability √
Aux outputs: reverb and samples

Optimal Chamber is useful for simulating spaces ranging from large rooms to small halls. True stereo inputs allow for stereo perspective of early reverb sound if desired.

The aux outputs are sent the reverb sound, but not the early reflections. The samples at the aux outputs are not pannable.

The following are the Factory presets using this algorithm:

2.1 **Voice** — nominal chamber setting, a good "starter" sound for using or customizing.

2.2 **Small** — shorter reverb.

2.3 **Nice** — longer, very pleasant decay.

2.4 **Open** — a warm, transparent space.

2.5 **ErVerb** — long decay with early reflections.

2.6 **Gated** — nominal gated chamber preset.

2.7 **Shower** — a very short decay time, simulating a small, hard enclosure.

2.8 **Slap** — a reverb with close "back wall" reflection.

2.8 **Slap2** — even closer than above.

2.8 **Pan** — special effect, pans the signal across the channels.

2.11 **Coin** — long gated program with many discrete echoes, like a coin dropping on the floor.

2.12 **Thunder** — reverb with heavy low-end sustain.

2.13 **Quake** — a special effect with lots of LF boost.

2.14 **Chmber** — uncolored moderate-sized room.

2.15 **Gated2** — tight gate with muted tail.

2.16 **Coin2** — variation on Coin, with shorter decay.

Page 1

Running Reverb

	RT60	LF Dcy	HF Dcy	HF BW	Size	P'Dly
max	INF	X2.0	X1.0	15kHz	145%	500ms
min	0.30s	X0.1	X0.1	2kHz	60%	0ms

Page 2

	Details			Sample Trig	
	Punch	Diffus	Random	Select	Play
max	100%	100%	100%	12	MAN/LIM
min	0%	0%	0%	1	-42dB/RPT

Page 3

	Gate		Stopped Reverb		
	Trig	StpDly	RT60	LF Dcy	HF Dcy
max	LIM	5.00s	INF	X2.0	X1.0
min	OFF/-42dB	0.01s	0.30s/MUTE	X0.1	X0.1

Page 4

Reflections for Left Outputs

	1 (Lin)		2 (Rin)		3 (Lin)	
max	0dB	500ms	0dB	500ms	0dB	500ms
min	OFF	0ms	OFF	0ms	OFF	0ms

Page 5

Reflections for Right Outputs

	1 (Rin)		2 (Lin)		3 (Rin)	
max	0dB	500ms	0dB	500ms	0dB	500ms
min	OFF	0ms	OFF	0ms	OFF	0ms

Page 6

Input Mix

	Input	Select	Level	Bal
max	0dB	12	0dB	100←
min	OFF (/-68dB)	1	OFF	→100

Page 7

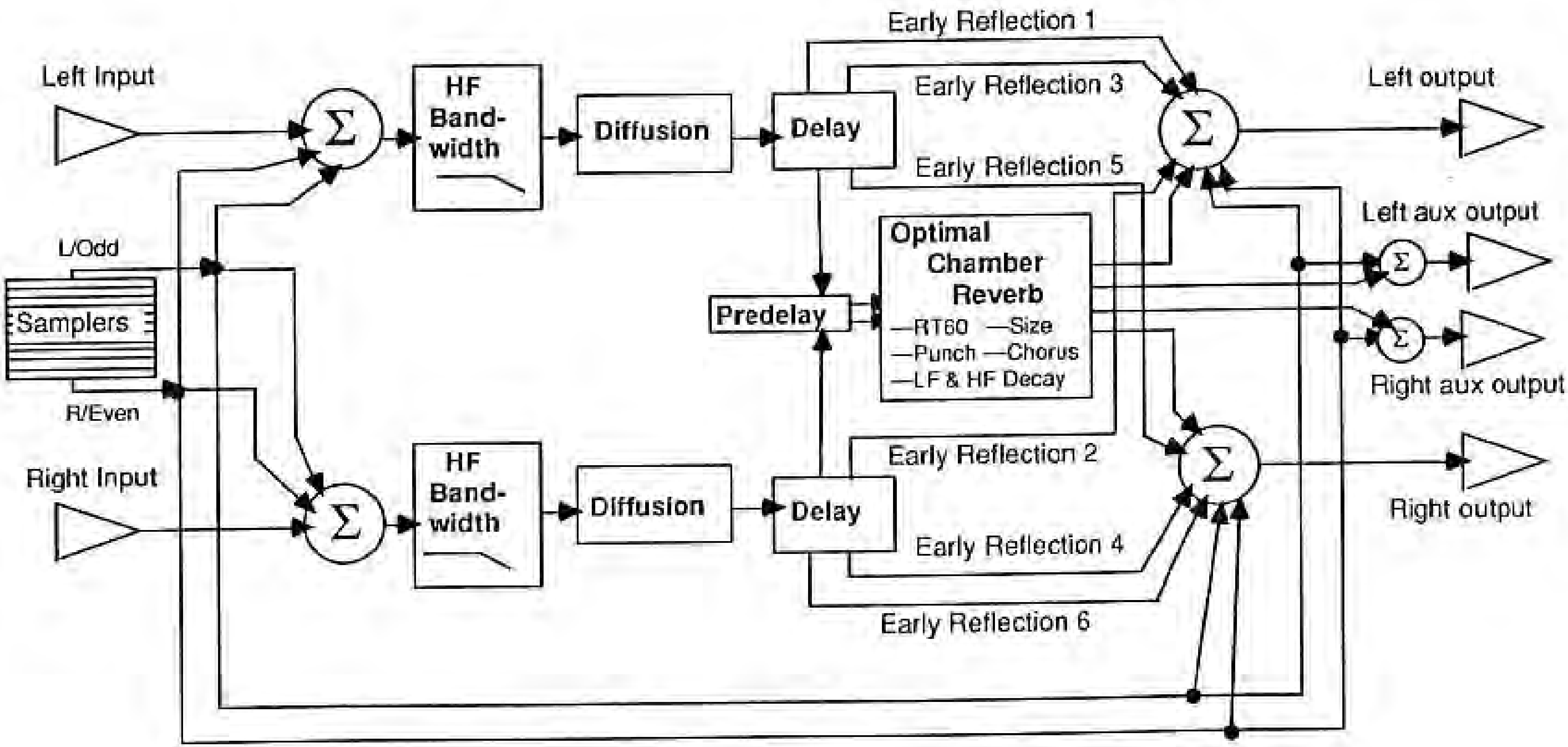
Main Output Mix

	E-R's	Rev'b	Select	Level	Bal
max	0dB	0dB	12	0dB	100←
min	OFF (/-68dB)	OFF (/-68dB)	1	OFF	→100

Page 8

Aux Output Mix

	Rev'b	Select	Level
max	0dB	12	0dB
min	OFF	1	OFF (/-68dB)



Bank 2 — Optimal Chamber (Chmb)

BANK 3 — Medium Room (ROOM)

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √
Inputs: true stereo

Gating Capability √
Aux outputs: reverb and samples

Medium Room is useful for simulating small to medium size acoustic spaces. It will sound good as the reverberator for film and video interiors, and for most kinds of music. True stereo inputs allow for stereo perspective of early reverb sound if desired.

The aux outputs are sent the reverb sound and the samples, but not the early reflections.

The following are the Factory presets using this algorithm:

3.1 Club — nominal room setting, a good "starter" sound for using or customizing.

3.2 Short — room setting with very short decay.

3.3 Room — longer than Club.

3.4 Good1 — medium reverb and reflections.

3.5 Good2 — variation of above.

3.6 Gated — basic gated room preset.

3.7 Closet — simulation of a very small enclosure.

3.8 Echoes — discrete-sounding reverb and reflection echoes.

3.9 DblEko — two slapback echoes that pan.

3.10 Build — early reflections build up and blossom into reverb.

3.11 Gong — a long decay with lots of punch at the beginning.

3.12 Weird — special effects reverb.

3.13 U.F.O. — special effects reverb.

3.14 Boom — long reverb with lots of LF boost.

3.15 Tank — giant oil drum.

3.16 Large — very, very large room.

Page 1

Running Reverb

	RT60	LF Dcy	HF Dcy	HF BW	Size	P'Dly
max	INF	X2.0	X1.0	15kHz	200%	500ms
min	0.90s	X0.1	X0.1	2kHz	40%	0ms

Page 2

Details / Sample Trig

	Depth	Diffus	Densi	Random	Select	Play
max	100%	100%	100%	100%	12	MAN/LIM
min	0%	0%	0%	0%	1	-42dB/RPT

Page 3

Gate / Stopped Reverb

	Trig	StpDly	RT60	LF Dcy	HF Dcy
max	LIM	5.00s	INF	X2.0	X1.0
min	OFF/-42dB	0.01s	0.90s/MUTE	X0.1	X0.1

Page 4

Reflections for Left Outputs

	1 (Lin)		2 (Rin)		3 (Lin)	
max	0dB	500ms	0dB	500ms	0dB	500ms
min	OFF	0ms	OFF	0ms	OFF	0ms

Page 5

Reflections for Right Outputs

	1 (Rin)		2 (Lin)		3 (Rin)	
max	0dB	500ms	0dB	500ms	0dB	500ms
min	OFF	0ms	OFF	0ms	OFF	0ms

Page 6

Input Mix

	Input	Select	Level	Bal
max	0dB	12	0dB	100←
min	OFF (/-68dB)	1	OFF	→100

Page 7

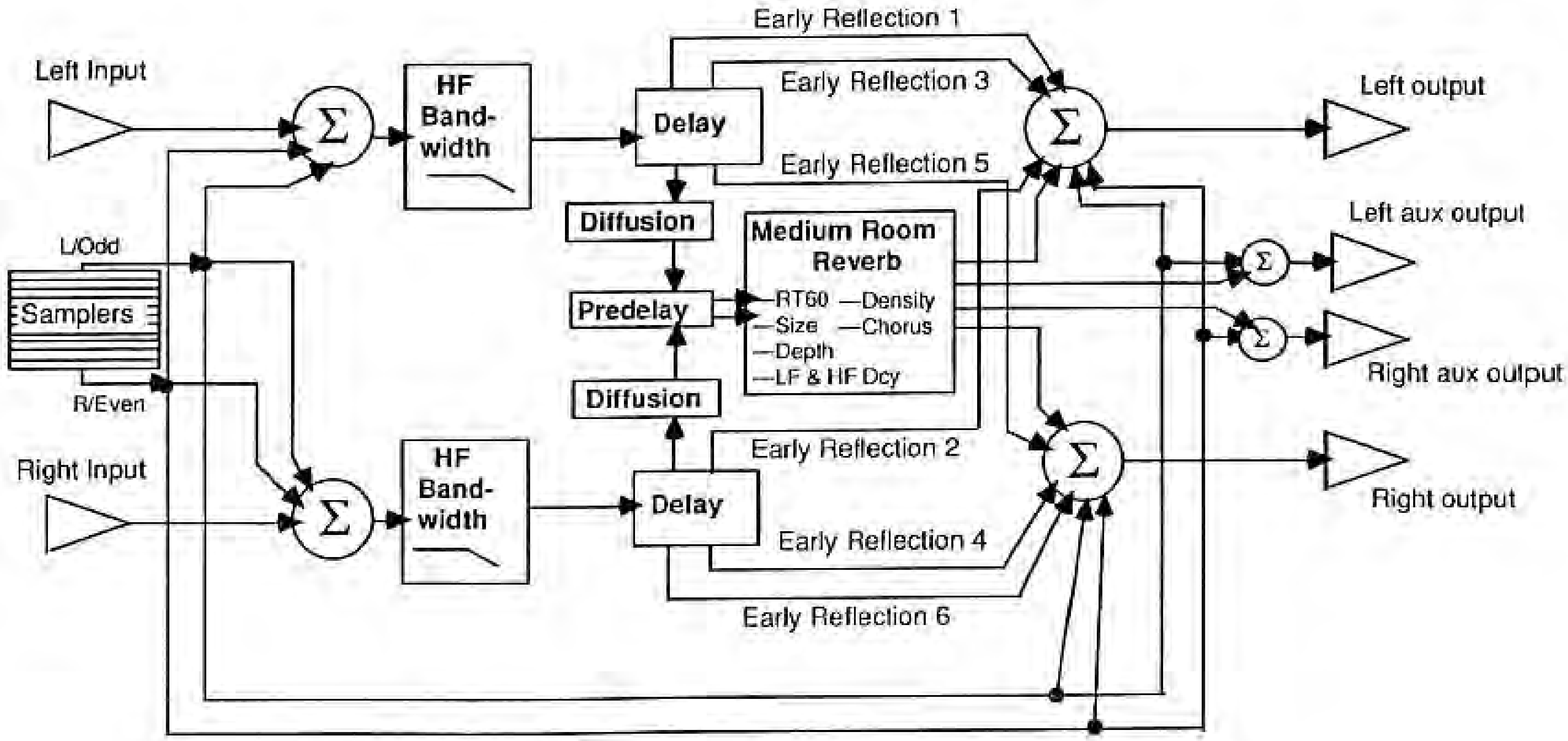
Main Output Mix

	E-R's	Rev'b	Select	Level	Bal
max	0dB	0dB	12	0dB	100←
min	OFF (/-68dB)	OFF (/-68dB)	1	OFF	→100

Page 8

Aux Output Mix

	Rev'b	Select	Level
max	0dB	12	0dB
min	OFF	1	OFF (/-68dB)



Bank 3 — Medium Room (Room)

BANK 4 — Natural Hall (HALL)

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √
Inputs: true stereo

Gating Capability √
Aux outputs: reverb and samples

Natural Hall is useful for simulating medium to large acoustic spaces. It allows for a more spacious, open sounding reverberation, pleasing with orchestral or spacy synthesizer music. True stereo inputs allow for stereo perspective of early reverb sound if desired.

The aux outputs are sent the reverb sound, and the samples, but not the early reflections.

The following are the Factory presets using this algorithm:

4.1 Concrt — nominal hall setting, with early reflections specially designed by Norbert Sobol. A good "starter" sound for using or customizing.

4.2 BallRm — smaller enclosure than above.

4.3 Arena — giant hockey rink.

4.4 Church — another large enclosure.

4.5 Studio — large soundstage.

4.6 Choir — moderate decay, with early reflections used for doubling.

4.7 Sustn — short running reverb, long gated reverb, for sustaining sound.

4.8 HiGate — a gated effect for high hat.

4.9 QikPan — special effect, pans sound rapidly across channels.

4.10 Cellar — a "damp"-sounding reverb.

4.11 Fugue — a hall with delusions of grandeur.

4.12 Tight — tightly gated hall.

Page 1

——— Running Reverb ———

	RT60	LF Dcy	HF Dcy	HF BW	Size	P'Dly
max	INF	X2.0	X1.0	15kHz	200%	500ms
min	0.6s	X0.1	X0.1	2kHz	40%	0ms

Page 2

	—Details—				—Sample Trig—	
	Depth	Diffus	Densi	Random	Select	Play
max	100%	100%	100%	100%	12	MAN/LIM
min	0%	0%	0%	0%	1	-42dB/RPT

Page 3

	—Gate—		—Stopped Reverb—		
	Trig	StpDly	RT60	LF Dcy	HF Dcy
max	LIM	5.00s	INF	X2.0	X1.0
min	OFF/-42dB	0.01s	0.6s/MUTE	X0.1	X0.1

Page 4

——— Reflections for Left Outputs ———

	1 (Lin)		2 (Rin)		3 (Lin)	
max	0dB	500ms	0dB	500ms	0dB	500ms
min	OFF	0ms	OFF	0ms	OFF	0ms

Page 5

——— Reflections for Right Outputs ———

	1 (Rin)		2 (Lin)		3 (Rin)	
max	0dB	500ms	0dB	500ms	0dB	500ms
min	OFF	0ms	OFF	0ms	OFF	0ms

Page 6

——— Input Mix ———

	Input	Select	Level	Bal
max	0dB	12	0dB	100←
min	OFF (/-68dB)	1	OFF	→100

Page 7

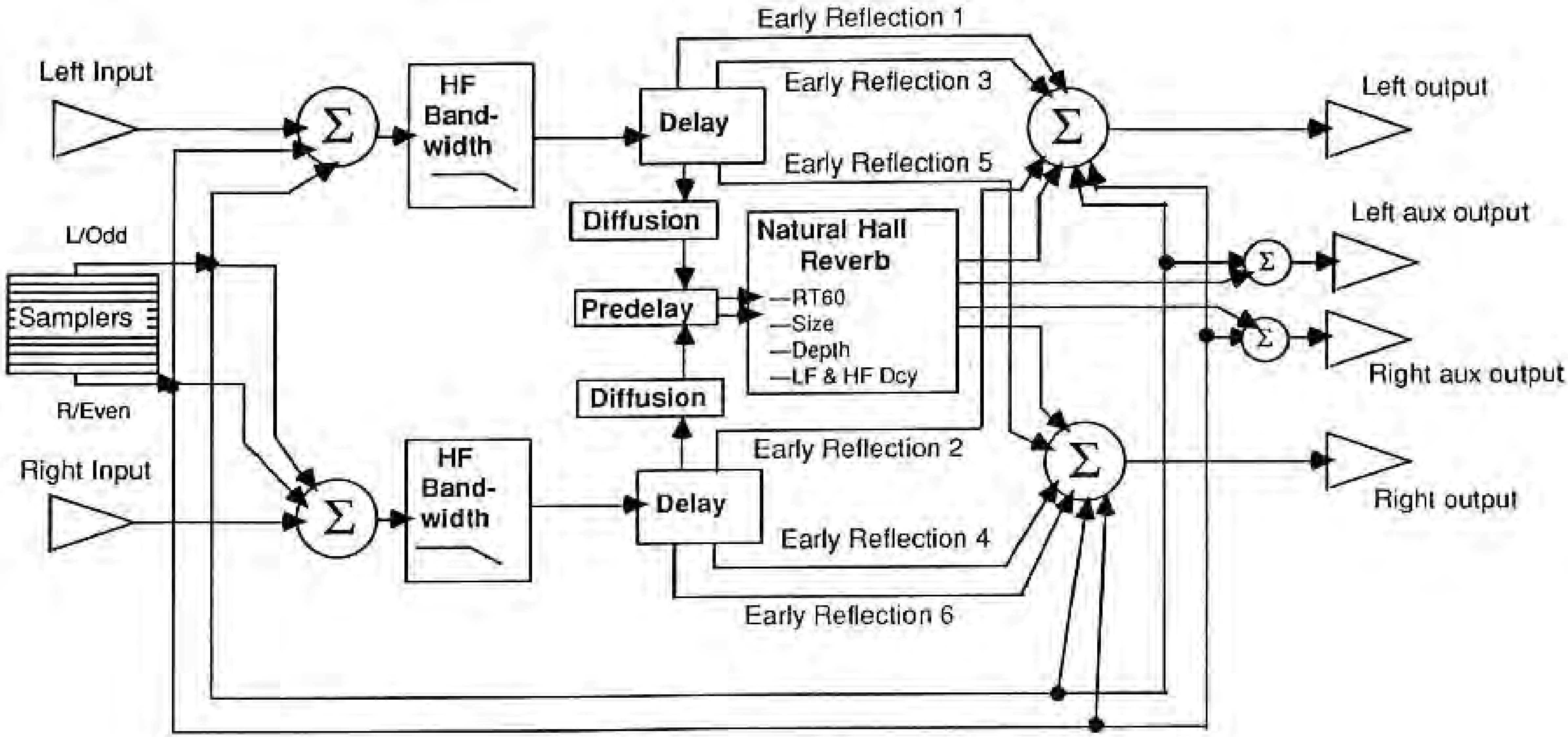
——— Main Output Mix ———

	E-R's	Rev'b	Select	Level	Bal
max	0dB	0dB	12	0dB	100←
min	OFF (/-68dB)	OFF (/-68dB)	1	OFF	→100

Page 8

——— Aux Output Mix ———

	Rev'b	Select	Level
max	0dB	12	0dB
min	OFF	1	OFF (/-68dB)



Bank 4 — Natural Hall (Hall)

Register F5.1 — Plate-Hall Split

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √ Gating Capability √(plate only)
Inputs: each reverb has one mono input
Aux outputs: sent Split 2 (Hall) only, and samples

Plate-Hall Split gives two reverbs, a Plate and a Hall, each with one input and two outputs. The Left input feeds the Plate side, and the Right input feeds the Hall side. The main parameter screen for the Plate side is Page 1, while the main screen for the Hall side is Page 4. Neither side has an adjustable Size parameter — it is defaulted at 100%. Gating is available on the Plate side only, and is adjusted on Page 3. The main outputs receive both reverbs, as mixed on page 7, while the aux outputs receive only the Hall reverb, not the Plate.

Samples recorded with the portable ("*") sampler programs are available for use with this program. The main outputs receive only odd-numbered (or Left) dry samples. At the aux outputs, odd-numbered mono samples appear only at the Left output, while even-numbered samples can be panned to either side.

The Factory presets using this algorithm are Register 5.1, plus the presets that appear on Page 2 of the Split menu:

5.1 PltHal — nominal split setting, for two independent mono-in, stereo-out reverbs; a good "starter" sound.

5.6 PH-A1! — useful, generic split.

5.7 1-Gate — gated Plate/generic Hall.

5.8 Lg-Sml — large Plate/small Hall.

5.9 NeGate — inverse, sustaining gate on Plate/generic Hall.

5.10 Ook!!! — weird special effect with lots of echoes.

Page 1

—Plate—

	RT60	LF Dcy	HF Dcy	HF BW	P'Dly
max	INF	X1.0	X1.0	15kHz	500ms
min	0.7s	X0.1	X0.1	2kHz	0ms

Page 2

—Plate Details—

	Random	Reflection1→L		Reflection2→R	
max	100%	0dB	500ms	0dB	500ms
min	0%	OFF	0ms	OFF	0ms

Page 3

—Plate Gate— —Plate Stopped Reverb—

	Trig	StpDly	RT60	LF Dcy	HF Dcy
max	LIM	5.00s	INF	X1.0	X1.0
min	OFF/-42dB	0.01s	0.7s/MUTE	X0.1	X0.1

Page 4

—Hall—

	RT60	LF Dcy	HF Dcy	HF BW	Densi	P'Dly
max	INF	X1.0	X1.0	15kHz	100%	500ms
min	0.5s	X0.1	X0.1	2kHz	0%	0ms

Page 5

—Hall Details—

	Random	Reflection1→L		Reflection2→R	
max	100%	0dB	500ms	0dB	500ms
min	0%	OFF	0ms	OFF	0ms

Page 6

—Input Mix— —Samplers—

	Lin	Rin	Select	Play	Level
max	0dB	0dB	12	MAN/LIM	0dB
mix	OFF	OFF	1	-42dB/OFF	OFF

Page 7

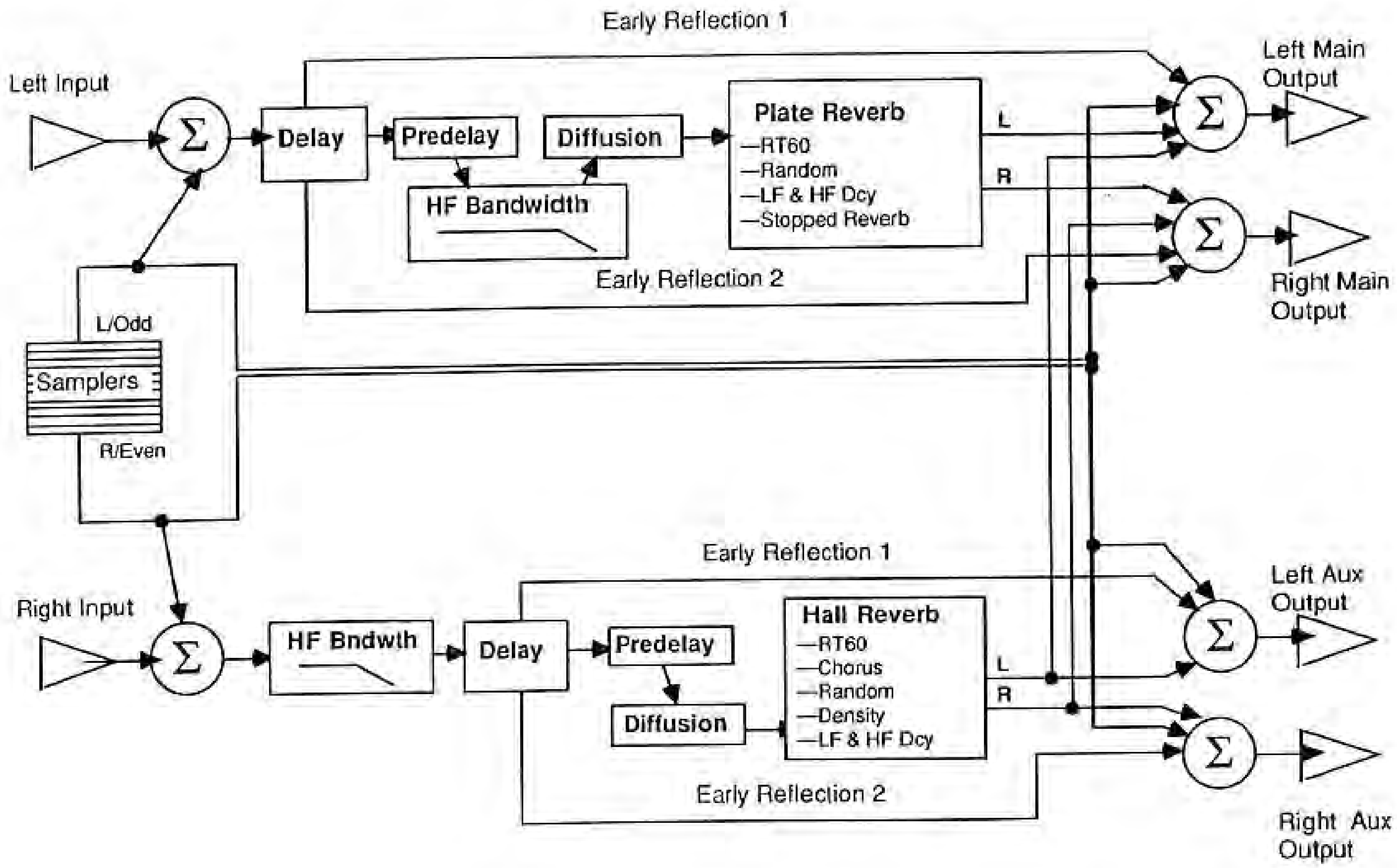
—Main Output Mix— —Samplers—

	Plate	Hall	Select	Level	Bal
max	0dB	0dB	12	0dB	100←
mix	OFF	OFF	1	OFF	→100

Page 8

—Aux Output Mix— —Samplers—

	Hall	Select	Level	Bal
max	0dB	12	0dB	100←
mix	OFF	1	OFF	→100



Register F5.1 — Plate/Hall Split

Register F5.2 — Plate/Plate Split

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √
Inputs: configurable
Aux outputs: sent Plate 2 only, and samples

Gating Capability √(plate 1 only)

Plate/Plate Split gives two reverbs. Plate 1 is the more complex, offering gating and a size control. The inputs are configurable, in Split, Parallel, or Chain modes. The Plate 1 outputs appear at the main outputs, while the Plate 2 outputs appear at the aux outputs. However, the auxilliary mix (containing the Plate 2 outputs) can be fed, with a software switch, to the main outputs.

Samples recorded with the portable ("*") sampler programs are available for use with this program. The main outputs receive only odd-numbered (or Left) dry samples, but all samples appear at the aux outputs, and can be panned to either side.

The Factory presets using this algorithm are Register 5.2, plus the presets that appear on Page 3 of the Split menu (configuration is "Split" unless otherwise indicated):

5.2 **2Plate** — nominal split setting; a good "starter" sound.

5.11 **PP-A1!** — useful, generic split.

5.12 **A-Gate** — gated Plate 1/generic Plate 2.

5.13 **BigSml** — large Plate 1/small Plate 2.

5.14 **Sustan** — inverse sustaining gate on Plate 1/very small space on Plate 2.

5.15 **Grapes** — (Chain configuration) grainy, bright reverb.

Page 1

—Plate 1 Running Reverb—

	RT60	LF Dcy	HF Dcy	HF BW	Size	P'Dly
max	INF	X2.0	X1.0	15kHz	150%	500ms
min	0.10s	X0.1	X0.1	2kHz	50%	0ms

Page 2

—Plate 1 Details—Samplers—

	Punch	Diffus	Random	Select	Play
max	100%	100%	100%	12	MAN/LIM
min	0%	0%	0%	1	-42dB/RPT

Page 3

—Plate 1 Gate— —Plate 1 Stopped Reverb—

	Trig	StpDly	RT60	LF Dcy	HF Dcy
max	LIM	5.00s	INF	X2.0	X1.0
min	OFF/-42dB	0.01s	0.10s/MUTE	X0.1	X0.1

Page 4

—Plate 2—

	RT60	LF Dcy	HF Dcy	HF BW	P'Dly
max	INF	X2.0	X1.0	15kHz	500ms
min	0.20s	X0.1	X0.1	2kHz	0ms

Page 5

—Plate 2 Details—

	Diffus	Random	Reflection1→L		Reflection2→R	
max	100%	100%	0dB	500ms	0dB	500ms
min	0%	0%	OFF	0ms	OFF	0ms

Page 6

—Input Mix— —Samplers—

	Config	Rev1	Rev2	Select	Level	Bal
max	Chain	0dB	0dB	12	MAN/LIM	100←
mix	Para/Split	OFF	OFF	1	-42dB/OFF	→100

Page 7

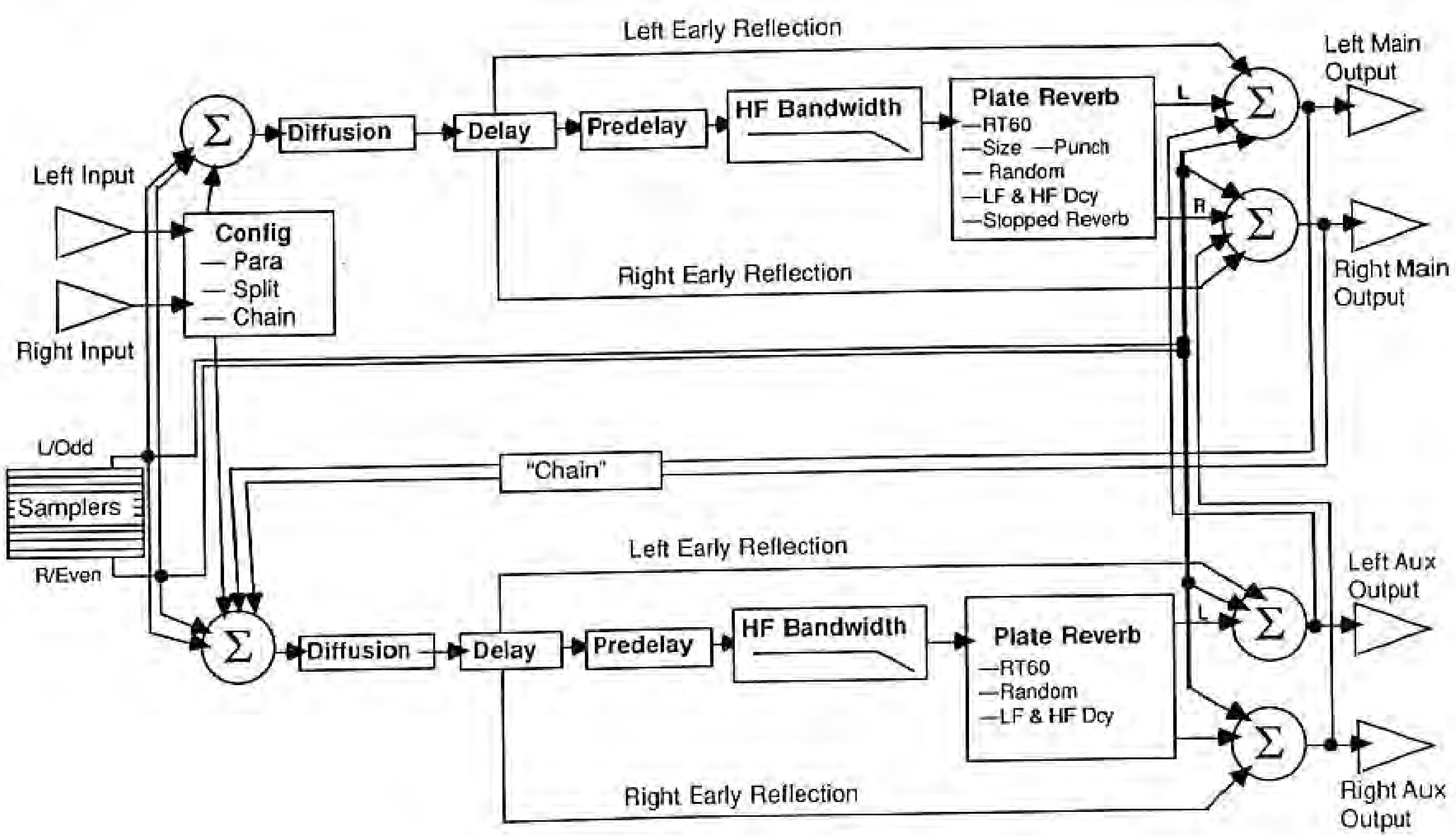
—Main Output Mix— —Samplers—

	Rev1	Aux Mix	Select	Level	Bal
max	0dB	ON	12	0dB	100←
mix	OFF	OFF	1	OFF	→100

Page 8

—Aux Output Mix— —Samplers—

	Rev2	Select	Level	Bal
max	0dB	12	0dB	100←
mix	OFF	1	OFF	→100



Register F5.2 — Plate/Plate Split

```
                                                    Page 1
        ————————Hall 1 Running Reverb————————
      RT60     LF Dcy HF Dcy HF BW  Size      P'Dly
max   INF      X1.0   X1.0   15kHz  200%      500ms
min   0.07s    X0.1   X0.1   2kHz   15%       0ms

                                                    Page 2
            ———————Hall 1 Details—Samplers——
max    Punch   Diffus Densi  Random Select    Play
max    100%    100%   100%   100%   12        MAN/LIM
min    0%      0%     0%     0%     1         -42dB/RPT

                                                    Page 3
       —Hall 1 Gate———Hall 1 Stopped Reverb—
       Trig         StpDly   RT60        LF Dcy   HF Dcy
max    LIM          5.00s    INF         X1.0     X1.0
min    OFF/-42dB 0.01s       0.07s/MUTE  X0.1     X0.1

                                                    Page 4
              ——————— Hall 2 ———————
       RT60     LF Dcy  HF Dcy   HF BW     Densi  P'Dly
max    INF      X1.0    X1.0     15kHz     100%   500ms
min    0.50s    X0.1    X0.1     2kHz      0%     0ms

                                                    Page 5
           ———————Hall 2 Details———————
       Diffus Random   Reflection1→LReflection2→R
max    100%   100%     0dB      500ms 0dB        500ms
min    0%     0%       OFF      0ms   OFF        0ms


                                                    Page 6
       —Input Mix———      ——Samplers——
       Config       Rev1 Rev2 Select Level        Bal
max    Chain        0dB  0dB   12       MAN/LIM   100←
mix    Para/Split OFF    OFF   1        -42dB/OFF →100

                                                    Page 7
       —Main Output Mix———Samplers——
       Rev1        Aux Mix     Select Level Bal
max    0dB         ON          12       0dB   100←
mix    OFF         OFF         1        OFF   →100
                                                    Page 8
       —Aux Output Mix———Samplers——
       Rev2                    Select   Level       Bal
max    0dB                     12       0dB         100←
mix    OFF                     1        OFF         →100
```

Register F5.3 — Hall/Hall Split

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √ Gating Capability √(hall 1 only)
Inputs: configurable
Aux outputs: sent Hall 2 only, and samples

Hall Split gives two reverbs. Hall 1 is the more complex, offering gating and a size control. The inputs are configurable, in Split, Parallel, or Chain modes. The Hall 1 outputs appear at the main outputs, while the Hall 2 outputs appear at the aux outputs. However, the auxilliary mix (containing the Hall 2 outputs) can be fed, with a software switch, to the main outputs.

Samples recorded with the portable ("*") sampler programs are available for use with this program. The main outputs receive only odd-numbered (or Left) dry samples, but all samples appear at the aux outputs, and can be panned to either side.

The Factory presets using this algorithm are Register 5.3, plus the presets that appear on Page 4 of the Split menu (configuration is "Split" unless otherwise indicated):

5.3 **2Halls** — nominal split setting; a good "starter" sound.

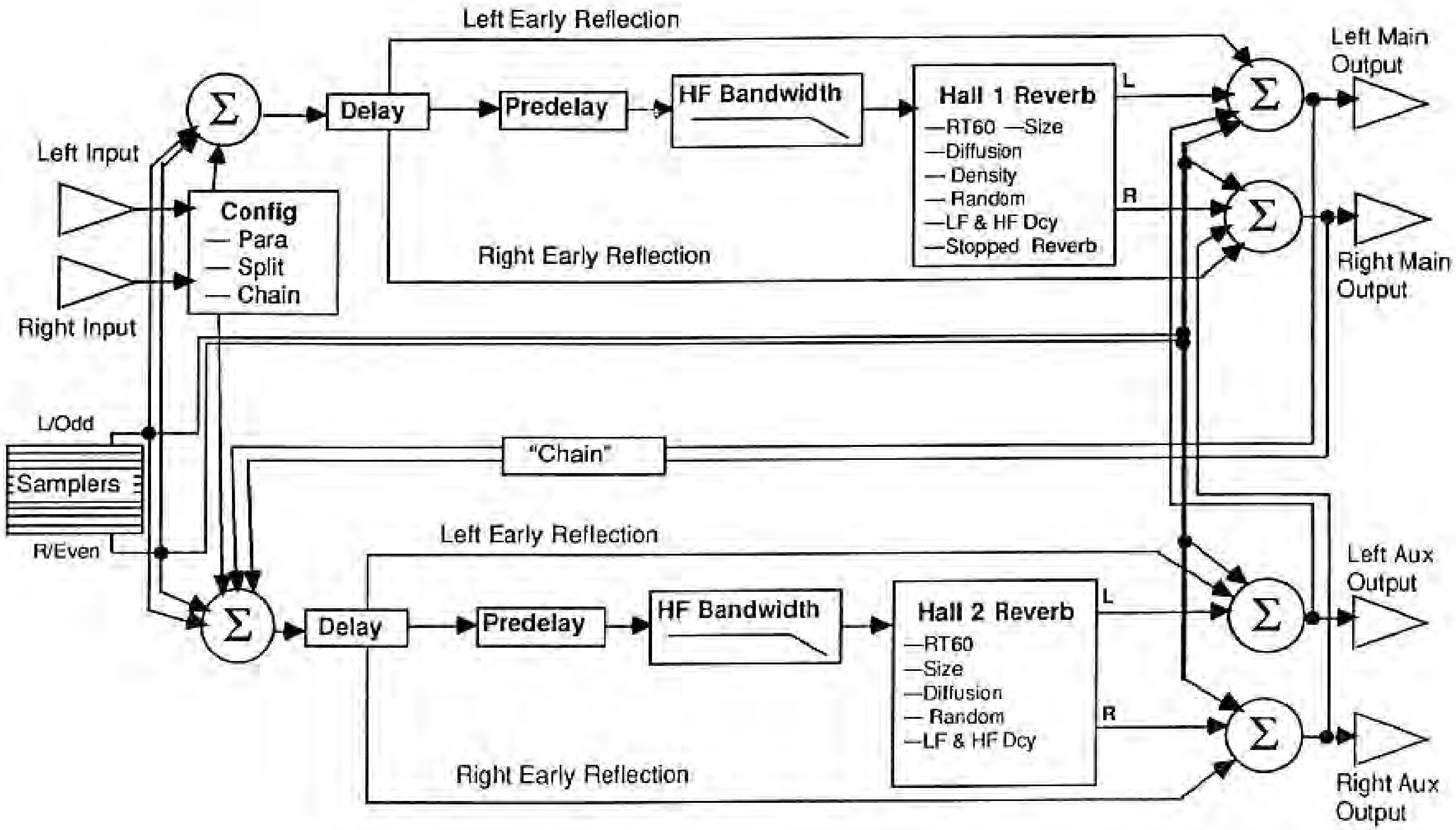
5.16 **HH-A1!** — good generic split.

5.17 **HHgate** — gated Hall 1/generic Hall 2.

5.18 **BigLit** — large Hall 1/small Hall 2.

5.19 **!!!...** — inverse, sustaining gate on Hall 1/generic large Hall 2.

5.20 **La-La-** — (Chain configuration) tinny special effect with much randomization.



Register F5.4 — Hall/Hall Split

Register F5.4 - Hall/Chorus Split

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √ Gating Capability √
Inputs: configurable
Aux outputs: sent Chorus only, and samples

Hall/Chorus Split gives a Hall reverb and a chorussing program. The latter is identical to the Poly-Chorus algorithm in bank 10, except that it has only three voices. The reverb side has gating. The inputs are configurable, in Split, Parallel, or Chain modes. The Hall outputs appear at the main outputs, while the Chorus outputs appear at the aux outputs. However, the auxilliary mix (containing the Chorus outputs) can be fed, with a software switch, to the main outputs.

Samples recorded with the portable ("*") sampler programs are available for use with this program. The main outputs receive only odd-numbered (or Left) dry samples, while all samples appear at the aux outputs, and can be panned to either side.

The Factory presets using this algorithm are Register 5.4, plus the presets that appear on Page 5 of the Split menu (configuration is "Split" unless otherwise indicated):

5.4 HalCho — nominal split setting; good "starter" sound.

5.21 HCnice — short Hall reverb on one side/friendly Chorus with short echo and a little auto-panning.

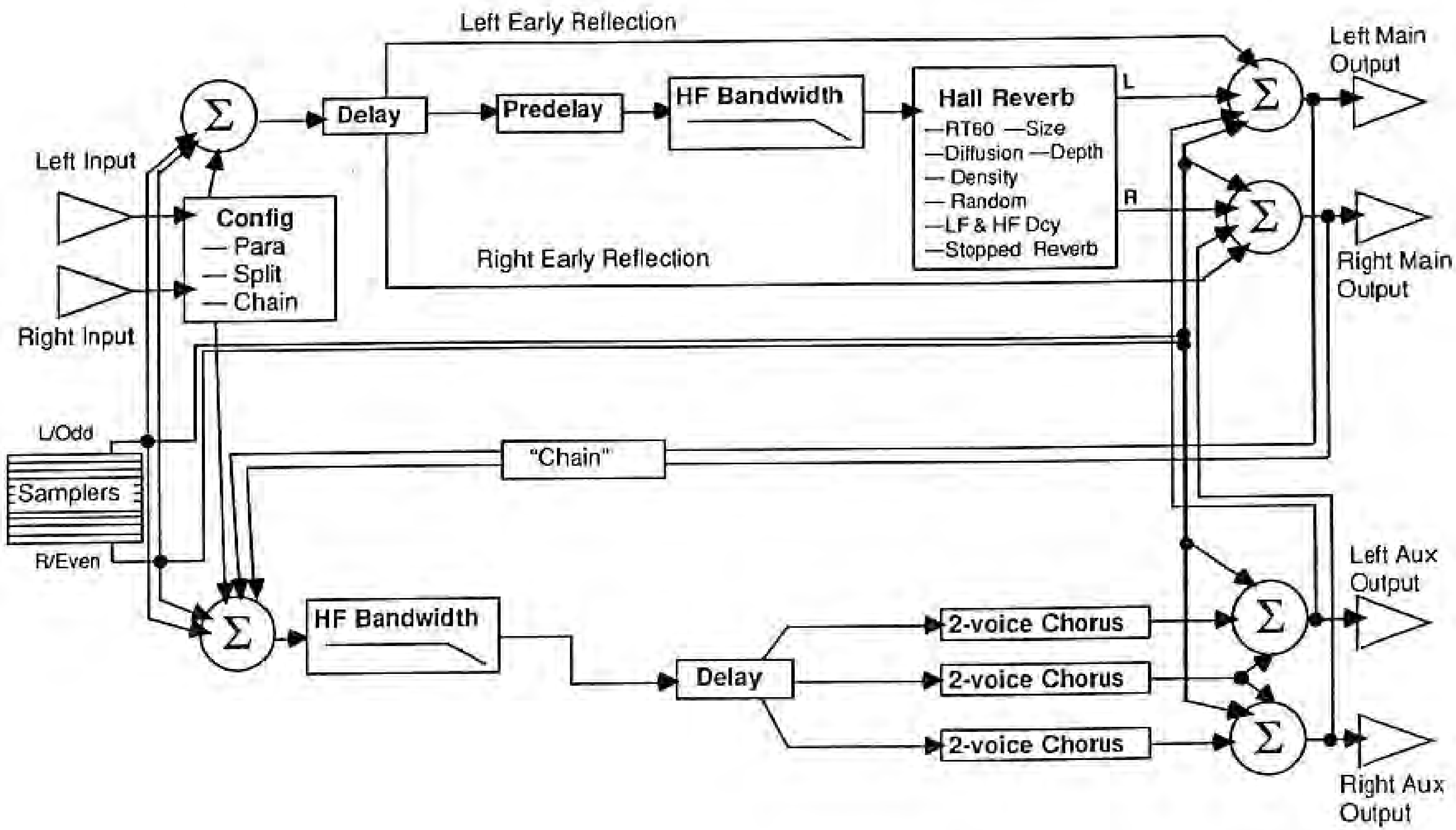
5.22 HCgate — gated Hall/panner.

5.23 I-Gate — gated Hall/quick-panning "Leslie" effect Chorus.

5.24 Roverb — (Chain configuration) reverb with massive detuning.

5.25 PanRev — (Chain configuration) long reverb tail with auto-panning applied to it.

```
                                                        Page 1
             ————————Hall Running Reverb————————
     RT60     LF Dcy HF Dcy HF BW  Size       P'Dly
max  INF      X1.0   X1.0   15kHz  200%       500ms
min  0.07s    X0.1   X0.1   2kHz   15%        0ms

                                                        Page 2
              ————Hall Details—Samplers——
max  Depth   Diffus Densi  Random Select      Play
max  100%    100%   100%   100%   12          MAN/LIM
min  0%      0%     0%     0%     1           -42dB/RPT

                                                        Page 3
     —Hall Gate—      —Hall Stopped Reverb—
     Trig        StpDly   RT60          LF Dcy   HF Dcy
max  LIM         5.00s    INF           X1.0     X1.0
min  OFF/-42dB   0.01s    0.07s/MUTE    X0.1     X0.1

                                                        Page 4
            ———— Hall Reflections ————
     Reflection1→L   Reflection2→R
max  0dB        500ms 0dB       500ms
min  OFF        0ms   OFF       0ms

                                                        Page 5
                 ————Chorus————
     Voices Depth  Rate    Delta Pan            HF BW
max  3      10.0ms 9.99Hz  250ms 2.00Hz         15kHz
min  1      0.0ms  OFF     0ms   0.01Hz/MONO    2kHz

                                                        Page 6
     —Input Mix——      ——Samplers——
     Config     Hall Chorus Select Level        Bal
max  Chain      0dB  0dB    12     MAN/LIM      100←
mix  Para/Split OFF  OFF    1      -42dB/OFF    →100

                                                        Page 7
     —Main Output Mix———Samplers——
     Hall      Aux Mix         Select Level Bal
max  0dB       ON              12     0dB   100←
mix  OFF       OFF             1      OFF   →100

                                                        Page 8
     —Aux Output Mix———Samplers——
     Chorus                    Select  Level     Bal
max  0dB                       12      0dB       100←
mix  OFF                       1       OFF       →100
```



Register F5.3 — Hall/Chorus Split

Register F5.5 — Hall/DDL Split

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √ Gating Capability √
Inputs: configurable
Aux outputs: sent DDL only, and samples

Hall/DDL Split gives a Hall reverb and a digital delay line, identical to one-half of the "Dual Delays" algorithm in Bank 11. The inputs are configurable, in Split, Parallel, or Chain modes. The Hall outputs appear at the main outputs, while the DDL outputs appear at the aux outputs (with separate levels available at the Left and Right outputs). However, the auxilliary mix (containing the DDL outputs) can be fed, with a software switch, to the main outputs.

Samples recorded with the portable ("*") sampler programs are available for use with this program. The main outputs receive only odd-numbered (or Left) dry samples, while all samples appear at the aux outputs, and can be panned to either side.

The Factory presets using this algorithm are Register 5.5, plus the presets that appear on Page 6 of the Split menu (configuration is "Split" unless otherwise indicated):

5.5 **HalDDL** — nominal split setting; a good "starter" sound.

5.26 **HDgood** — generic Hall/short repeating delay.

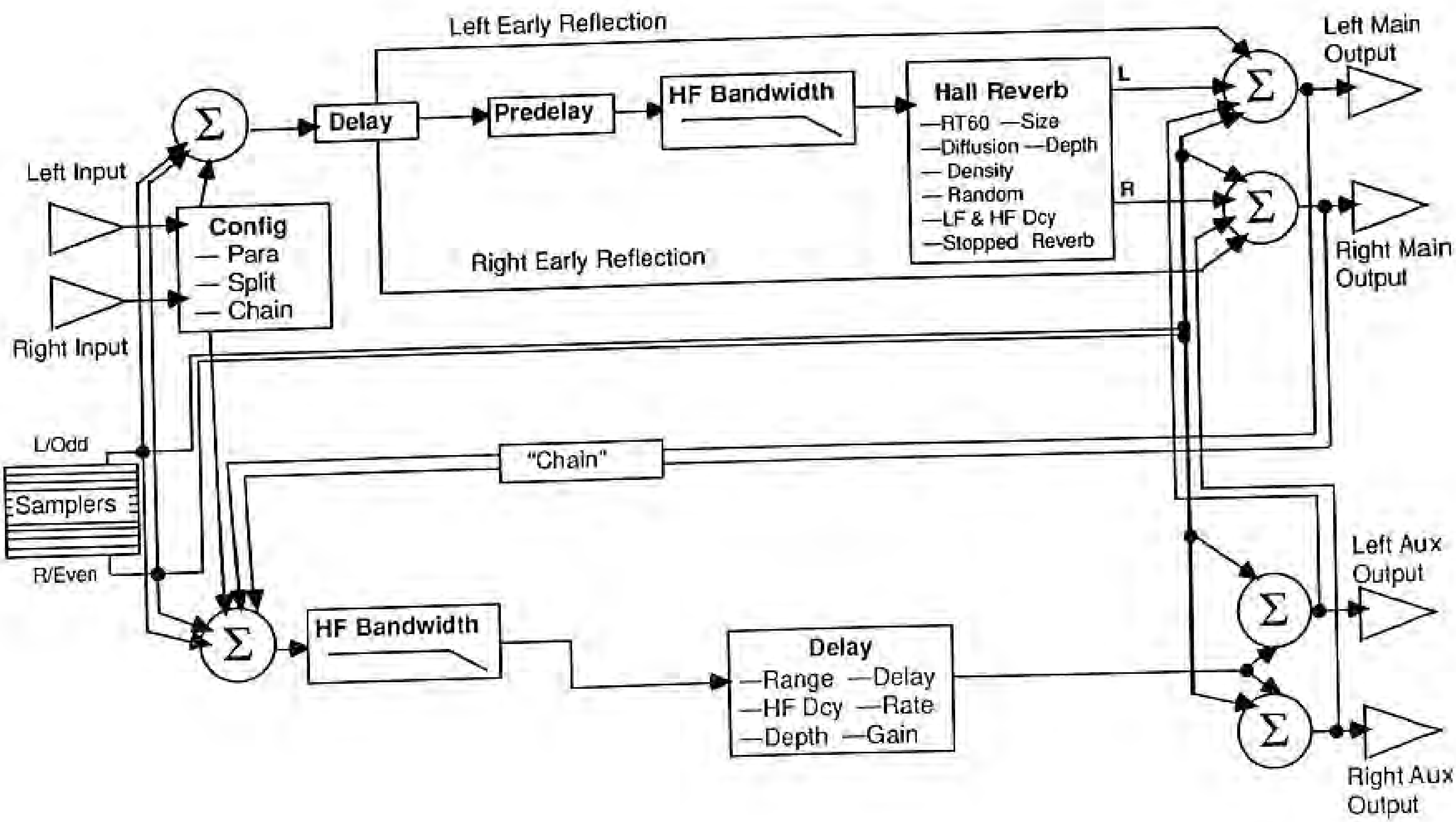
5.27 **1-Gate** — short, tight gated Hall/repeats with modulation applied.

5.28 **Oh-Boy** — long Hall/tight flange with cycling resonant filter.

5.29 **Wowzer** — "floppy"-sounding Hall/"bouncing ball" delay.

5.30 **RevEko** — (Chain configuration) uncolored reverb with tail sent through short repeating delay.

```
                                                         Page 1
              ————Hall Running Reverb————
     RT60     LF Dcy HF Dcy HF BW  Size    P'Dly
max  INF      X1.0   X1.0   15kHz  200%    500ms
min  0.07s    X0.1   X0.1   2kHz   15%     0ms

                                                         Page 2
          ————Hall Details—Samplers——
max  Depth  Diffus Densi  Random Select  Play
max  100%   100%   100%   100%   12      MAN/LIM
min  0%     0%     0%     0%     1       -42dB/RPT

                                                         Page 3
     —Hall Gate—   —Hall Stopped Reverb—
     Trig        StpDly  RT60        LF Dcy  HF Dcy
max  LIM         5.00s   INF         X1.0    X1.0
min  OFF/-42dB   0.01s   0.07s/MUTE  X0.1    X0.1

                                                         Page 4
          ———— Hall Reflections ————
     Reflection1→L  Reflection2→R
max  0dB      500ms 0dB      500ms
min  OFF      0ms   OFF      0ms

                                                         Page 5
               ————DDL————
     Range       Depth   Gain   HF Dcy  Depth  Rate
max  Long        2000ms  100%   X1.0    100%   9.99Hz
min  Samples160us -100%  X0.1   0%      OFF/0.05Hz

                                                         Page 6
     —Input Mix———     ——Samplers——
     Config      Hall  DDL  Select Level       Bal
max  Chain       0dB   0dB  12     MAN/LIM     100←
mix  Para/Split  OFF   OFF  1      -42dB/OFF   →100

                                                         Page 7
     —Main Output Mix————Samplers——
     Hall      Aux Mix    Select Level Bal
max  0dB       ON         12     0dB   100←
mix  OFF       OFF        1      OFF   →100

                                                         Page 8
     —Aux Output Mix————Samplers——
     DDL→L     DDL→R      Select  Level  Bal
max  0dB       0dB        12      0dB    100←
mix  OFF       OFF        1       OFF    →100
```



Register F5.5 — Hall/DDL Split

F6.1

Page 1

Record Functions

	Select	Source	Safety	Record	Stop	Play
max	12	Right	OFF	MAN		MAN/LIM
min	1	Left	ON	-42dB		-42dB/RPT

Page 2

Edit Functions

	Select	Begin—	trim	End—	trim	Play
max	12	2.096	63ms	2.096	63ms	MAN/LIM
min	1	0.000	-63ms	0.000	-63ms	-42dB/RPT

Page 3

Main Output — Envelope

	Select	Level	Bal	Attack	Decay	Play
max	12	0dB	100←	970ms	970ms	MAN/LIM
min	1	OFF	→100	0ms	0ms	-42dB/RPT

F6.5

Page 1

Record Functions

	Select	Source	Safety	Record	Stop	Play
max	12	Right	OFF	MAN		MAN/LIM
min	1	left	ON	-42dB		-42dB/RPT

Page 2

Edit Functions

	Select	Begin—	trim	End—	trim	Play
max	12	33.524	63ms	33.524	63ms	MAN/LIM
min	1	0.000	-63ms	0.000	-63ms	-42dB/RPT

Page 3

Main Output — Aux Output

	Select	Level	Bal	Level	Bal	Play
max	12	0dB	100←	0dB	100←	MAN/LIM
min	1	OFF	→100	0dB	→100	-42dB/RPT

Page 4

Envelope

	Select	Attack	Decay	Play
max	12	970ms	970ms	MAN/LIM
min	1	0ms	0ms	-42dB/RPT

Register F6.1 — Efx-12-Sampler (Mono*)

Register F 6.5 — Mono-12-Sampler (M-12)

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Samples: 12, 2.096 seconds each (Mono*); variable length (M-12)
Inputs: selectable Left or Right   Aux outputs: mixed separately

Mono* is a program for setting up monaural samples for use within a reverb or effects program in one of the other banks (although it can also be used by itself). It allows for recording, editing, and replaying 12 sound samples, each up to two seconds in length. The samples are retained in memory and are "portable" to reverb and effects programs.

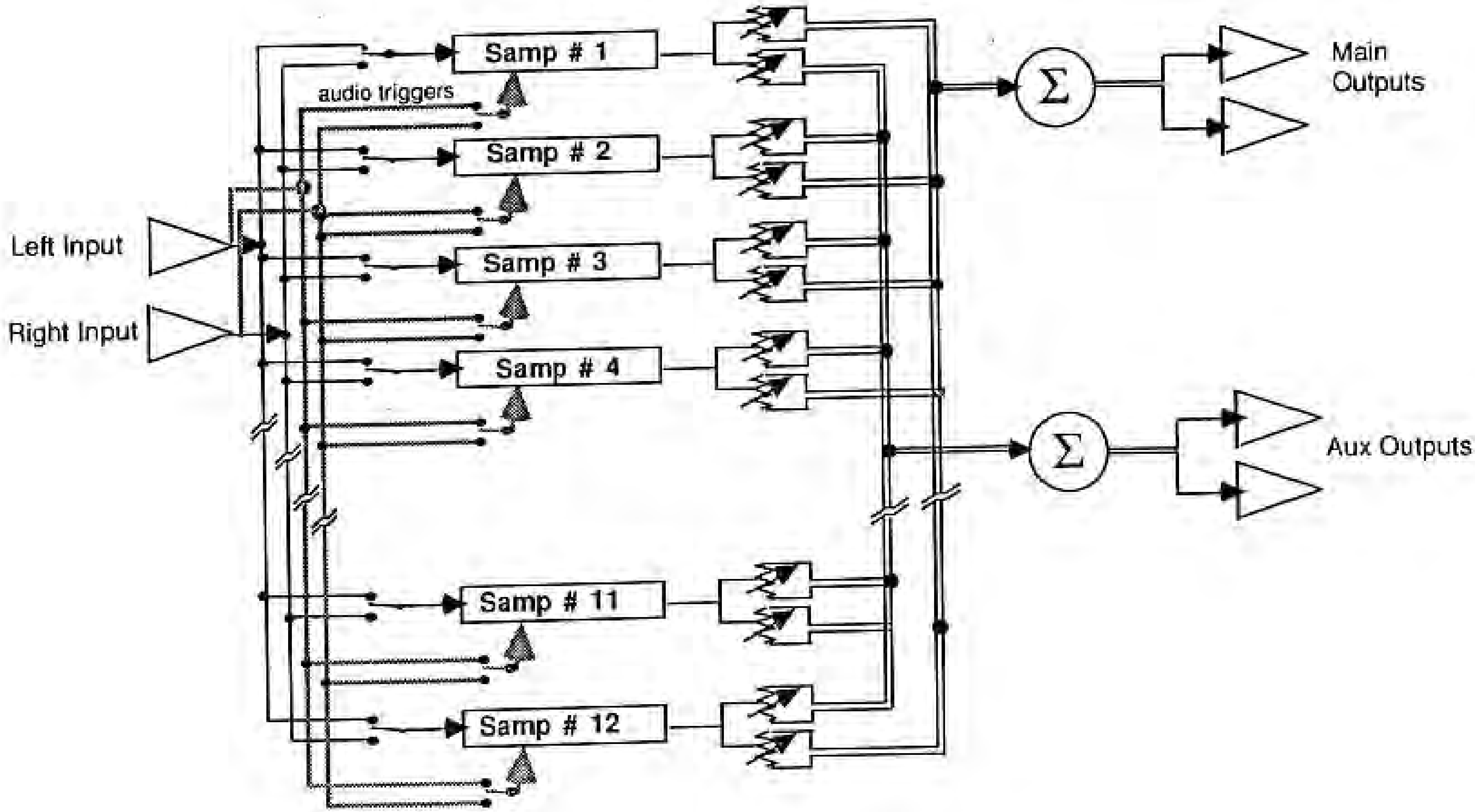
Playing back a sample through a reverb or effects preset can be done from the remote, with an external switch, from an audio signal, or via MIDI. Setting the level of an external audio trigger can be done within this preset or in the reverb or effects preset, and the settings will carry over fom one to the other. Setting auxilliary output levels can only be done in the reverb or effects preset.

Determining which input *channel* will trigger a sample is done from within this preset, by adjusting the Source fader after the sample has been recorded. You can go from an effects program to this preset without losing any settings by using the menus or Quick Register Select to call this preset, adjusting the Source fader, and then recalling the effects preset by pressing "Last".

M-12 is for setting up monaural samples that will be used by themselves, without reverb or effects. The samples can be any length, but the total length of all samples together cannot exceed 33.524 seconds.

These presets can be assigned one or more MIDI parameter maps, and can be stored in Internal or Cartridge registers with those maps. Editing, envelope, and mixing parameters are not stored.

If the Auto-Erase function is turned off, samples recorded with these programs may be used in other sampling programs, except that all edit and envelope information will be erased. Under certain conditions, the sample data may be altered or scrambled.



Registers F6.1 and 6.5 — Mono* and M-12 (Multiple Mono Samplers)

F6.2

Page 1

Record Functions

	Select	Safety	Record Stop	Play
max	6	OFF	MAN	MAN/LIM
min	1	ON	-42dB	-42dB/RPT

Page 2

Edit Functions

	Select	Begin	trim	End	trim	Play
max	6	2.096	63ms	2.096	63ms	MAN/LIM
min	1	0.000	-63ms	0.000	-63ms	-42dB/RPT

Page 3

Main Output — Envelope

	Select	Level	Attack	Decay	Play
max	6	0dB	970ms	970ms	MAN/LIM
min	1	OFF	0ms	0ms	-42dB/RPT

F6.6

Page 1

Record Functions

	Select	Safety	Record Stop	Play
max	6	OFF	MAN	MAN/LIM
min	1	ON	-42dB	-42dB/RPT

Page 2

Edit Functions

	Select	Begin	trim	End	trim	Play
max	6	16.762	63ms	16.762	63ms	MAN/LIM
min	1	0.000	-63ms	0.000	-63ms	-42dB/RPT

Page 3

Main Output — Aux Output

	Select	Level	Bal	Level	Bal	Play
max	6	0dB	100←	0dB	100←	MAN/LIM
min	1	OFF	→100	0dB	→100	-42dB/RPT

Page 4

Envelope

	Select	Attack	Decay	Play
max	6	970ms	970ms	MAN/LIM
min	1	0ms	0ms	-42dB/RPT

Register F6.2 — Efx-6-Sampler (Ster*)
Register F 6.5 — Stereo-6-Sampler (St-6)

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Samples: six, 2.096 seconds each (Ster*); variable length (St-6)
Inputs: stereo Aux outputs: mixed separately

Ster* is a program for setting up stereo samples for use within a reverb or effects program in one of the other banks (although it can also be used by itself). It allows for recording, editing, and replaying six sound samples, each up to two seconds in length. The samples are retained in memory and are "portable" to reverb and effects programs.

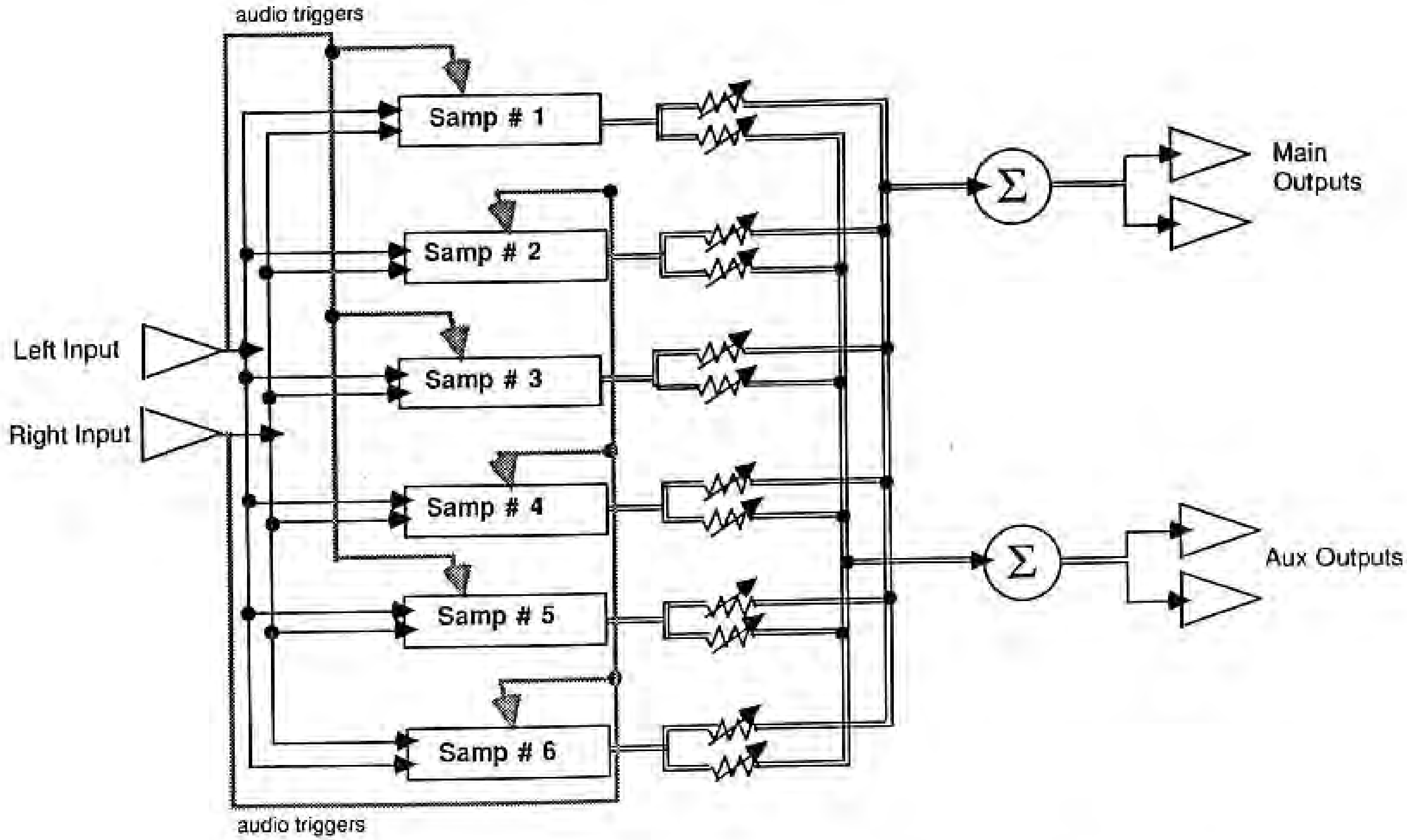
Playing back a sample through a reverb or effects preset can be done from the remote, with an external switch, from an audio signal, or via MIDI. Setting the level of an external audio trigger can be done from within this preset or in the reverb or effects preset, and settings will carry over from one to the other. Setting auxilliary output levels can only be done in the reverb or effects preset. You can go from an effects program to this preset without losing any settings by using the menus or Quick Register Select to call this preset, adjusting the Source fader, and then recalling the effects preset by pressing "Last".

St-6 is for setting up stereo samples that will be used by themselves, without reverb or effects. The samples can be any length, but the total length of all samples together cannot exceed 16.762 seconds.

In both presets, odd-numbered samples are triggered from signals at the Left input, and even-numbered ones are triggered from the Right.

These presets can be assigned one or more MIDI parameter maps, and can be stored in Internal or Cartridge registers with those maps. Editing, envelope, and mixing parameters are not stored.

If the Auto-Erase function is turned off, samples recorded with these programs may be used in other sampling programs, except that all edit and envelope information will be erased. Under certain conditions, the sample data may be altered or scrambled.



Registers F6.2 and F6.6 — Stereo* and St-6 (Multiple Stereo Samplers)

Register F6.3 — Mono 32sec Sampler (M)

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Samples: one, 33.524 seconds long
Inputs: selectable Left and Right
Aux outputs: same as main outputs

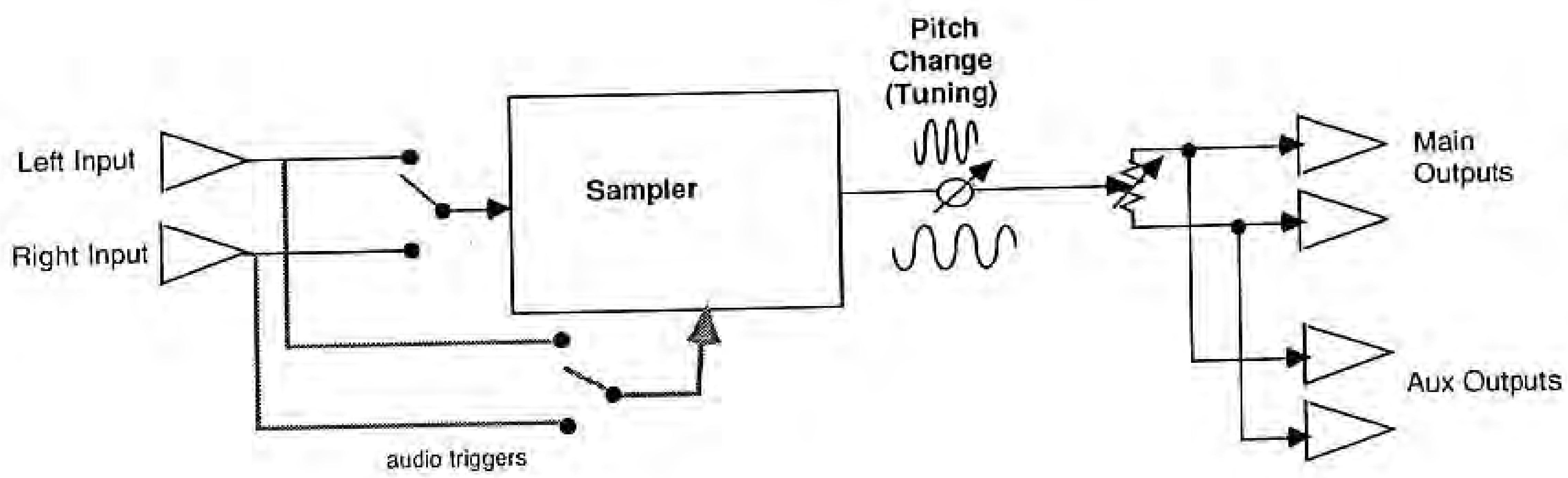
M is a program for recording and playing one monaural sample. The sample memory can be broken up into segments of any length, and recorded separately, but only one such segment can be played back at a time.

A sample can be triggered from an audio source, whose level is determined from the "Play" fader, and whose channel is determined from the "Source" fader.

This preset can be assigned one or more MIDI parameter maps, and can be stored in an Internal or Cartridge register with those maps. Editing, envelope, and mixing parameters are not stored.

If the Auto-Erase function is turned off, samples recorded with this program may be used in other sampling programs, except that all edit and envelope information will be erased. Under certain conditions, the sample data may be altered or scrambled.

```
                                                     Page 1
              ——— Record Functions ———
              Source  Safety  Record Stop    Play
max           Right   OFF     MAN            MAN/LIM
min           Left    ON      -42dB          -42dB/RPT

                                                     Page 2
              ——— Edit Functions ———
              Begin—trim      End— trim      Play
max           33.524 63ms     33.524 63ms    MAN/LIM
min           0.000  -63ms    0.000  -63ms   -42dB/RPT

                                                     Page 3
          ———Tuning———
          Method  Amount         Fine          Play
max       %       +1 Oct/100%    +200 cents    MAN/LIM
min       Intvl   -1 Oct/50%     -200 cents    -42dB/RPT

                                                     Page 4
          ———Main Output——— ———Envelope———
          Level   Bal            Attack  Decay  Play
max       0dB     100←           970ms   970ms  MAN/LIM
min       OFF     →100           0ms     0ms    -42dB/RPT
```



Register F6.3 — M (Single Mono Sampler)

Register F6.4 — Stereo 16s Sampler (St)

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Samples: one, 16.762 seconds long
Inputs: stereo
Aux outputs: same as main outputs

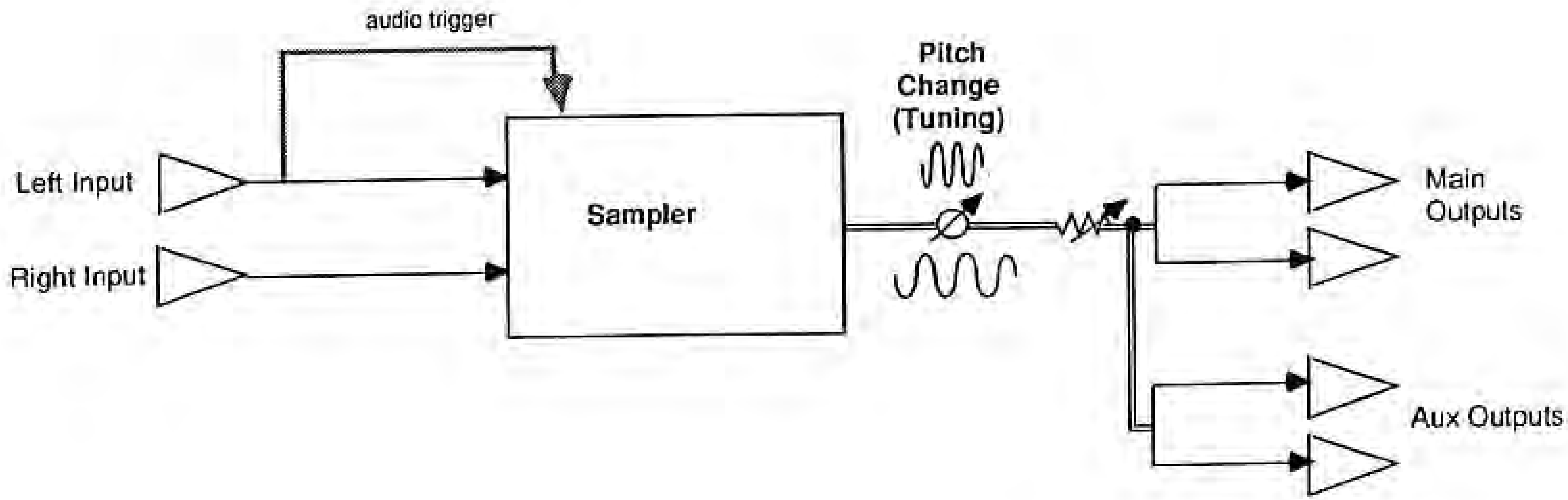
St is a program for recording and playing one stereo sample. The sample memory can be broken up into segments of any length, and recorded separately, but only one such segment can be played back at a time.

A sample can be triggered from an audio source on the Left input channel, whose level is determined from the "Play" fader.

This preset can be assigned one or more MIDI parameter maps, and can be stored in an Internal or Cartridge register with those maps. Editing, envelope, and mixing parameters are not stored.

If the Auto-Erase function is turned off, samples recorded with this program may be used in other sampling programs, except that all edit and envelope information will be erased. Under certain conditions, the sample data may be altered or scrambled.

```
                                                    Page 1
            ——— Record Functions ———
                    Safety  Record Stop    Play
max                 OFF     MAN            MAN/LIM
min                 ON      -42dB          -42dB/RPT

                                                    Page 2
            ——— Edit Functions ———
          Begin—trim     End— trim     Play
max       33.524 63ms    33.524 63ms   MAN/LIM
min       0.000  -63ms   0.000  -63ms  -42dB/RPT

                                                    Page 3
       ———Tuning———
       Method  Amount       Fine                Play
max    %       +1 Oct/100%  +200 cents          MAN/LIM
min    Intvl   -1 Oct/50%   -200 cents          -42dB/RPT

                                                    Page 4
       ———Main Output———Envelope———
       Level                Attack  Decay  Play
max    0dB                  970ms   970ms  MAN/LIM
min    OFF                  0ms     0ms    -42dB/RPT
```



Register F6.4 — St (Single Stereo Sampler)

Page 1

	RT60	HF Dcy	Punch	Select	Play
max	1.00s	X1.0	100%	12	MAN/LIM
min	0.20s	X0.1	0%	1	-42dB/RPT

Page 2

Input Mix

	Input	Select	Level
max	0dB	12	0dB
min	OFF/-68dB	1	OFF/-68dB

Page 3

Main Output Mix

	Rev'b	Select	Level	Bal
max	0dB	12	0dB	100←
min	OFF/-68dB	1	OFF/-68dB	→100

Page 4

Aux Output Mix

	Select	Level
max	12	0dB
min	1	OFF/-68dB

BANK 7 — Reverse Reverb (Revers)

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √
Inputs: summed into mono

Gating Capability x
Aux outputs: dry samples only

Reverse Reverb simulates a "backwards" reverb, whose output level starts low and rises over time.

The RT60 is the total length of the reverb event. HF Dcy modifies the decay characteristics, and Punch controls the level of the *last* sound you hear (which is the image of the first sound going in).

The program sums the inputs into mono, and produces a stereo image. After a signal is applied to the input, it comes out at first in stereo, and then evolves over time into mono.

Samples recorded with the Two2s* and St-2s* programs are usable within this program.

Only the dry samples appear at the aux outputs, odd-numbered ones at the Left output and even-numbered ones at the Right.

The following are the Factory presets using this algorithm:

7.1 **Revers** — nominal reverse reverb preset, with RT60 of 0.6s.

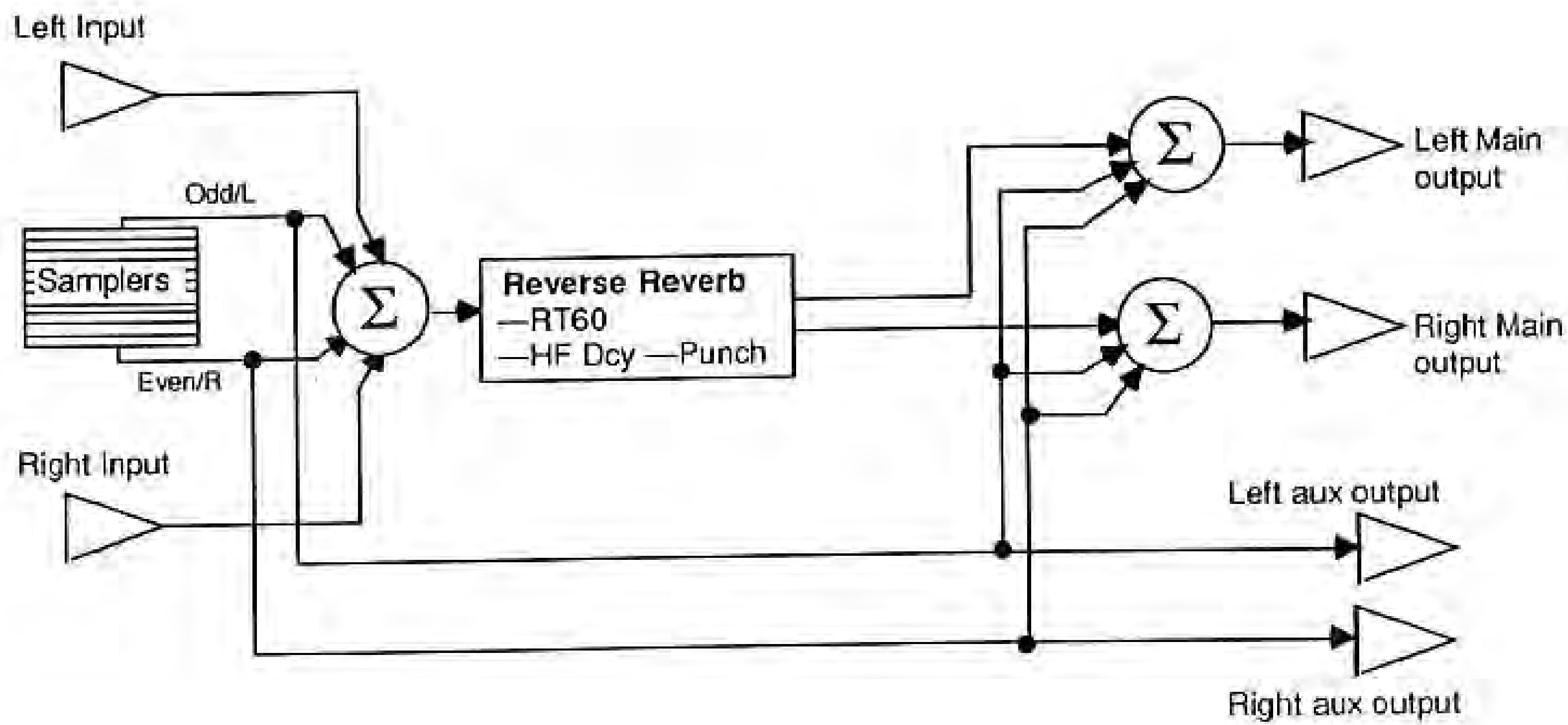
7.2 **Crsndo** — has Punch and HF Dcy rolled back for smooth orchestral crescendo.

7.3 **Echo** — short RT60 for an unusual "slapback" effect.

7.4 **Jimi** — long decay time, lots of Punch for maximum dramatic effect.

7.5 **Short** — short and dull reverse reverb.

7.6 **Motown** — very short effect with sharp high-frequency rolloff.



Bank 7 — Reverse Reverb (Revers)

Page 1

	Left DDL			Right DDL		
	Delay	Gain	HF Dcy	Delay	Gain	HF Dcy
max	766ms	100%	x1.0	766ms	100%	x1.0
min	0ms	-100%	x0.1	0ms	-100%	x0.1

Page 2

	Gate		Equalizer		
	Trig	StpDly	LF EQ	HF EQ	Effect
max	LIM	5.00s	+10dB	+10dB	ON
min	OFF/-42dB	0.01s	-10dB	-10dB	BYPASS

Page 3

	Stereo Chorus					
	Voices	Speed	Range	Mode	HF BW	Effect
max	4	100%	16.0ms	Oppo	15kHz	ON
min	1	0%	1.0ms	Para	2kHz	MUTE

Page 4

	Stereo Multi-Tap					
	Mode	# Taps	Envlp	Length	HF BW	Effect
max	L,R	8	Decr	2000ms	15kHz	ON
min	Echo1	1	Even	0ms	2kHz	MUTE

Page 5

	Reverb Multi-Tap					
	Mode	# Taps	Envlp	Length	HF BW	Effect
max	P'Dly	6	Decr	2000ms	15kHz	ON
min	Echo1	1	Even	0ms	2kHz	BYPASS

Page 6

	Reverb					
	RT60	LF Dcy	HF Dcy	Size	Densi	Effect
max	INF	x1.0	x1.0	100%	100%	ON
min	0.07s	x0.1	x0.1	15%	0%	BYPASS/MUTE

Page 7

	-L DDL In-	-R DDL In-	Samples		
	Lin	Rin	Select	Play	Level
max	0dB	0dB	12	MAN/LIM	0dB
min	OFF	OFF	1	RPT/-42dB-68dB/OFF	

Page 8

	Main Output Mixer					-Aux-
	Chorus	Mtap	Reverb	Select	Level	Select
max	0dB	0dB	0dB	12	0dB	Samps
min	OFF	OFF	OFF	1	-68dB/OFF	Reverb

BANK 8 — Multi-Effects (Multi)

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √ Gating Capability √
Inputs: summed into mono after initial ddl's
Aux outputs: switchable among effects or dry samples

In the **Multi-Effects** algorithm, a number of processing modules are available simultaneously, leaving up to the user which ones are used, and how they work on each other.

On Page 1 is a pair of independent digital delay lines (DDLs), one connected to the Left input (and the Left or odd-numbered samplers), and the other connected to the Right input (and Right or even-numbered samplers). The delay fader uses a logarithmic taper. Gain controls feedback within the digital delay; a negative gain indicates that the feedback is out of phase. After it passes these DDLs, the input signal is summed into mono.

On Page 2 are the Gate parameters. The gate in this algorithm is located after all of the processing, so that when it closes, all of the processed output shuts off at the main outputs. However, the gate has no effect on the aux outputs. Also on Page 2 is a two-band equalizer, whose corner frequencies are approximately 250 Hz and 6 kHz.

The processors on Page 3, 4, and 5 work in parallel. Page 3 is Stereo Chorus. The Range parameter adjusts modulation depth, and the Speed parameter determines cycle time. The Mode parameter determines whether the chorused voices move in parallel or opposite to each other.

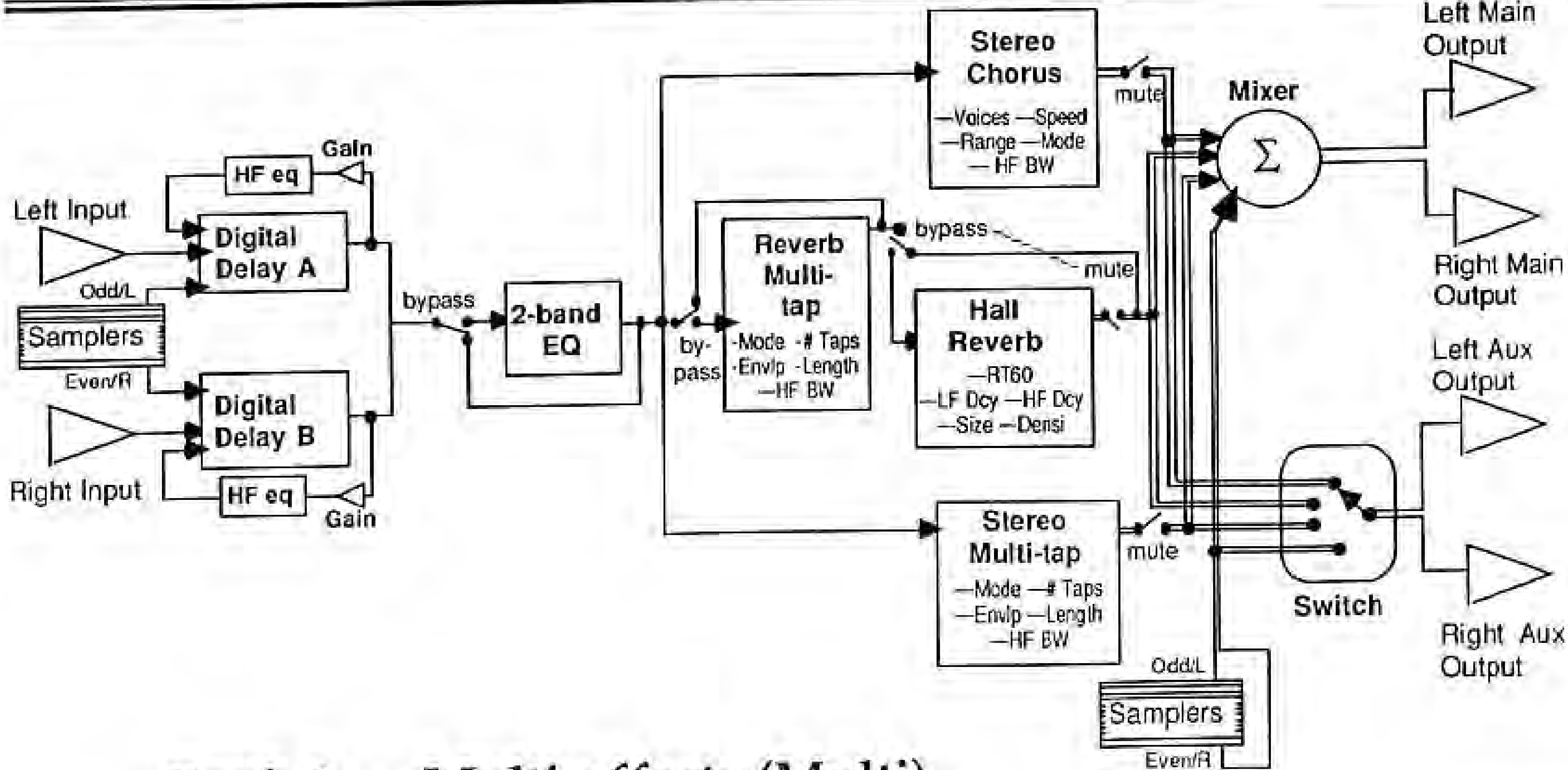
Page 4 is Stereo Multi-Taps, with control over the timing, amplitude, and stereo placement of the echoes. Page 5 is another set of taps, with a slightly reduced set of options, which are used to feed the reverb. The Reverb on Page 6 is the equivalent of the Hall side of the Plate-Hall Split algorithm, with the addition of an adjustable Size parameter.

There is no mixer for the aux outputs. Instead the Aux-Select fader on Page 8 is a multi-position switch which sends the dry samples or one of the three parallel effects to the aux outputs.

Samples created with the portable sampler programs can be used in Multi-Effects programs, and can be inserted before the ddls and/or after all of the processing. Samples are triggered from Page 7.

The Factory presets that us[illegible]is algorithm are listed below:

8.1 Mefx — all modules active at nominal settings.
8.2 Pan — panning echoes.
8.3 Bounce — echoes which alternate between channels.
8.4 Jump — echoes start on one side, jump to other side.
8.5 Circle — a sensation of 360° movement.
8.6 UpEcho — echoes and reverb increase in volume.
8.7 Gately — weird gating effect.
8.8 ChEcho — echoes going up and down, timed with chorusing.
8.9 Jewels — strange delays with HF boost.
8.10 Awful — bizarre multiple comb filters.
8.11 Nulso — echoes out of control (as was the programmer).
8.12 EkoRev — reverb and echoes.
8.13-8.15 EkoRv2, EkoRv3, and EkoRv4 — variations on above.
8.16 Moverb — Reverb and panning echoes.
8.17 Vibrio — vibrating reverb.



Bank 8 — Multi-effects (Multi)

BANK 9 — Stereo Processor (Stereo)

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √ Gating Capability √
Inputs: each processor has mono input
Aux outputs: mixed separately from main outputs

Stereo Processor is used to convert monaural sound sources into stereo. Presets using this algorithm contain two discrete stereo processors, A and B, which can use the two inputs as two different sources.

The Depth control adjusts the relative distance in time between the various taps used to generate the stereo image, in some modes giving the image a front-to-back dimension. The Width control adjusts how far off to the Left and Right outputs the taps will appear. At 0 (except in the PAN mode), the output will be mono. LF Eq and HF Eq are shelving equalizers with corner frequencies at about 250 Hz and 6 kHz respectively.

The processing modes, in the order they appear on the first fader, are ROOM, MSP, COMB1, COMB2, and PAN. See Chapter 5 for full descriptions of the modes.

Samples created with the portable sample programs can be used in Stereo presets. Odd-numbered samples, and the Left channel of stereo samples, can only feed Processor A. Even-numbered samples, and the Right channel of stereo samples, can only feed Processor B. The samples can also be sent directly to the outputs dry.

The following are the Factory presets using this algorithm:

9.1 **TwoMSP** — shows MSP mode on both sides.

9.2 **TwoCM2** — shows COMB2 mode on both sides.

9.3 **TwoCM1** — shows COMB1 mode on both sides.

9.4 **Room** — shows ROOM mode on both sides.

9.5 **PAN** — shows 90°R panning, using one processor only.

Page 1

	Stereo Processor A					
	Mode	Depth/ImageWidth		LF Eq	HF Eq	Samp1
max	PAN	100%/→90	100%	+10dB	+10dB	MAN/LIM
min	ROOM	0%/90←	0%	-10dB	-10dB	-42dB/RPT

Page 2

	Stereo Processor B					
	Mode	Depth/ImageWidth		LF Eq	HF Eq	Samp2
max	PAN	100%/→90	100%	+10dB	+10dB	MAN/LIM
min	ROOM	0%/90←	0%	-10dB	-10dB	-42dB/RPT

Page 3

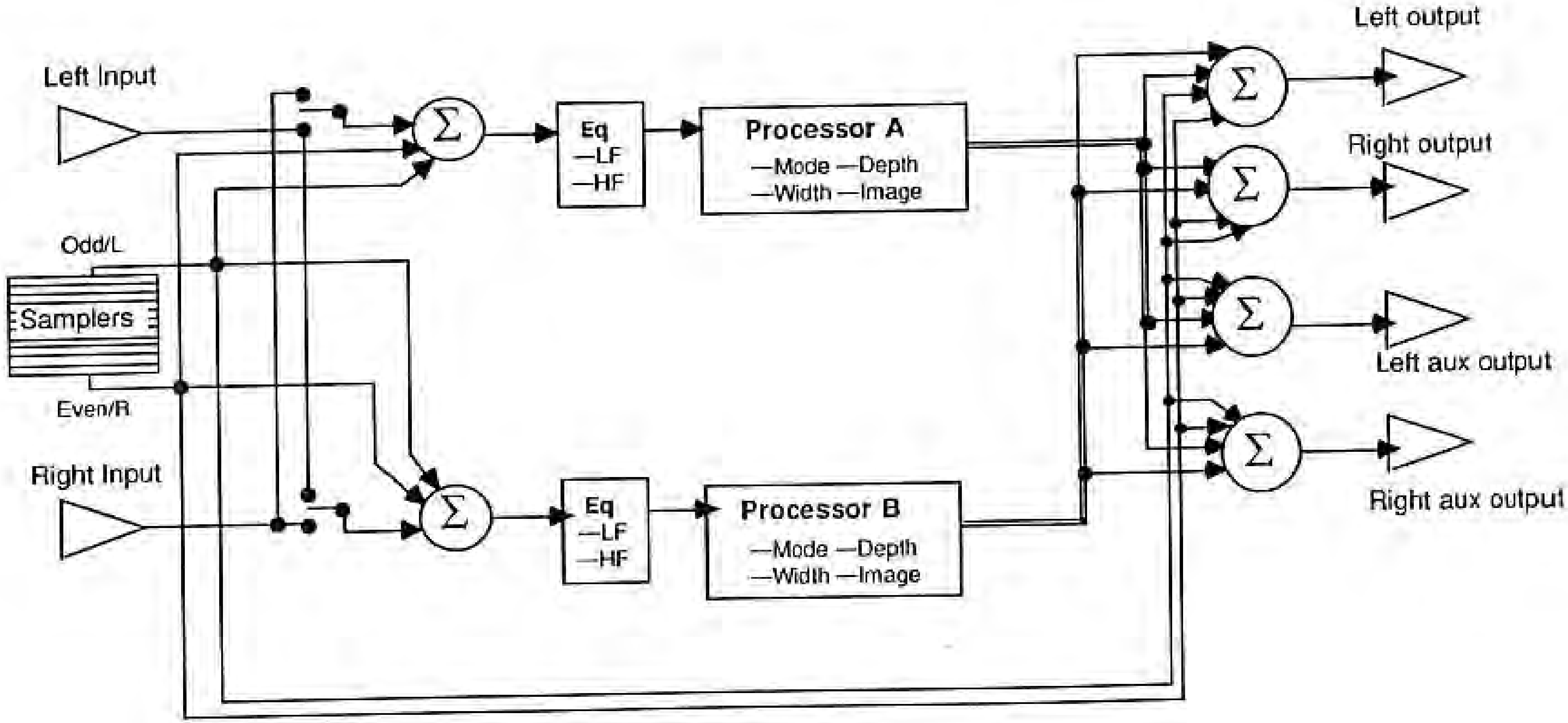
	—A—	—B—	—Input Mix—		—Samplers—	
	Input	Input	Select	Play	Level	Bal
max	L/R	L/R	12	MAN/LIM	0dB	100←
min	OFF	OFF	1	-42dB/RPT	OFF	→100

Page 4

	Main Output Mix				
	SP-A	SP-B	Select	Level	Bal
max	0dB	0dB	12	0dB	100←
min	OFF	OFF	1	OFF	→100

Page 5

	Aux Output Mix				
	SP-A	SP-B	Select	Level	Bal
max	0dB	0dB	12	0dB	100←
min	OFF	OFF	1	OFF	→100



Bank 9 — Stereo Processor (Stereo)

Page 1

	Voices	Depth	Rate	Delta	Pan
max	6	10.0ms	9.99Hz	250ms	2.00Hz
min	1	0.0ms	OFF	0ms	MONO

Page 2

	Diffus	HF BW	Select	Play
max	100%	15kHz	12	MAN/LIM
min	0%	2kHz	1	-42dB/RPT

Page 3

Input Mix

	L+R	Select	Level	Bal
max	0dB	12	0dB	LEFT
min	OFF	1	OFF	RIGHT

Page 4

Main Output Mix

	Chorus	Select	Level	Bal
max	0dB	12	0dB	LEFT
min	OFF	1	OFF	RIGHT

BANK 10 — Poly-Chorus (Chorus)

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √
Inputs: summed into mono

Gating Capability x
Aux outputs: dry samples only

Poly-Chorus creates multiple delay taps, and moves them in time, creating chorusing, vibrato, pitch-shifting, and many other effects. Depth sets the range over which the taps will be moved, while Rate determines how fast the taps will move. Faster ones impart almost a ring-modulation effect, by actually frequency-modulating the input signal. The amount of pitch shift is dependent on both of these parameters.

Delta specifies the offset in time between the various taps, if you are using more than one. If it is 0, you can get a flanging effect. At low values it can create a resonant-filter effect.

Pan sweeps the output of the processor between the two audio outputs. Diffusion spreads the sound out slightly at the input, to give it a little more body.

Samples created with the Two2s* and St-2s* programs can be used in Chorus presets, and can appear before and/or after the chorus processing.

Only the dry samples appear at the aux outputs.

The following are the Factory presets using this algorithm:

10.1 **Chor1** — basic Chorus setting.

10.2 **Chor2** — slower Chorus rate, faster panning.

10.3 **Chor3** — faster Chorus and panning.

10.4 **Pan1** — basic setting for sweeping sound over the stereo field.

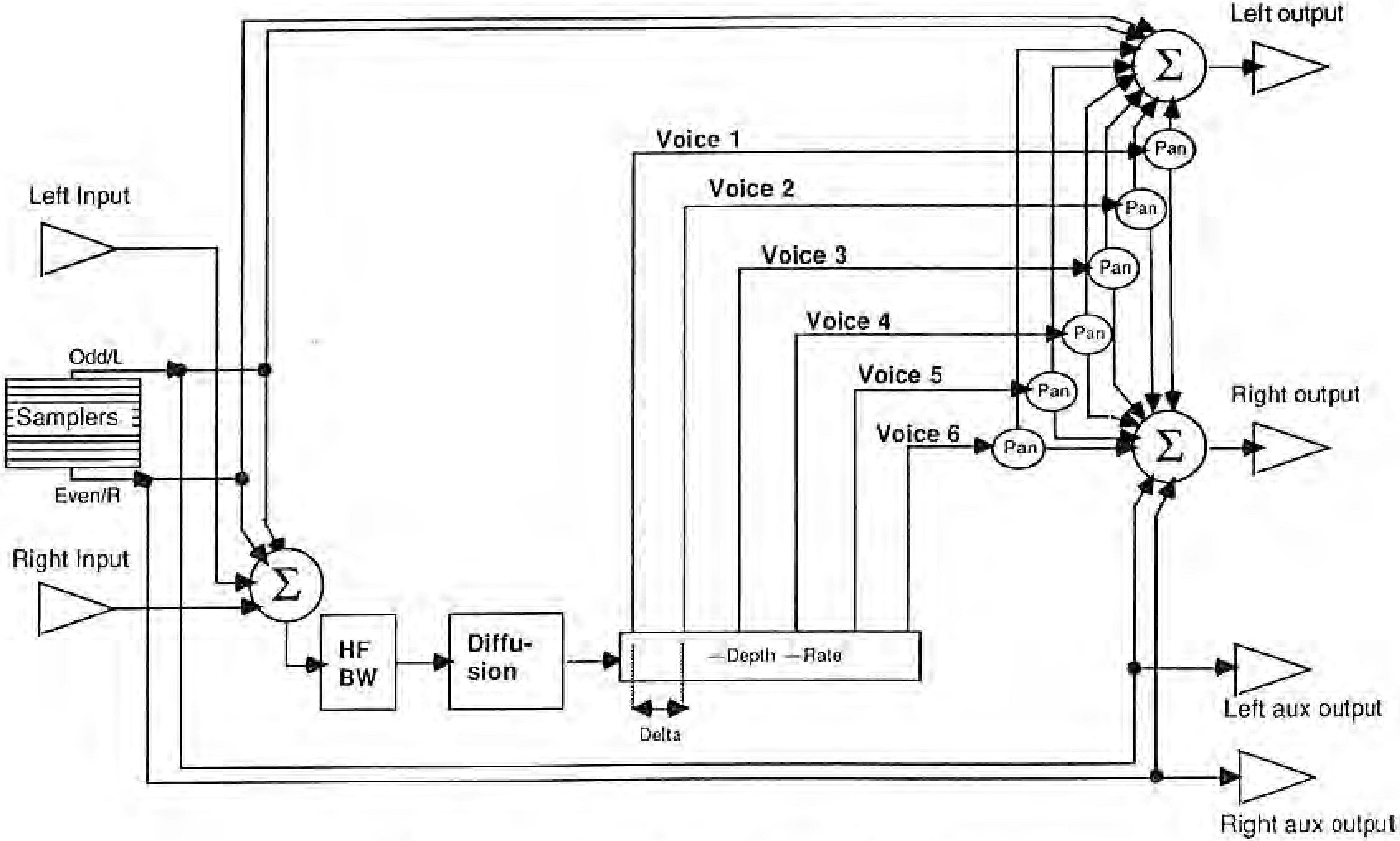
10.5 **Pan2** — faster panning rate, and more voices for a more complex sound.

10.6 **Pan3** — slow but more complex than above.

10.7 **Echo** — Delta set high for discrete echoes, which are also chorused.

10.8 **Pan4** — another panning preset.

10.9 **Lesly** — simulated "Leslie" effect.



Bank 10 — Poly-Chorus (Chorus)

BANK 11 — Dual Delays (DDL)

RELEASE 4.0, 15 DECEMBER 87

Sampling Rate: 32 kHz
Concurrent Sampling √   Gating Capability x
Inputs: separate or summed, depending on configuration
Aux outputs: dry samples only

The **Dual Delays** algorithm contains two separate digital delays, each with adjustable delay times, feedback, and modulation. They can be used to create a wide variety of effects, including pitch-shifting, vibrato, frequency modulation, tunneling, etc., etc. The delays can be arranged in a parallel, chained, or split configuration.

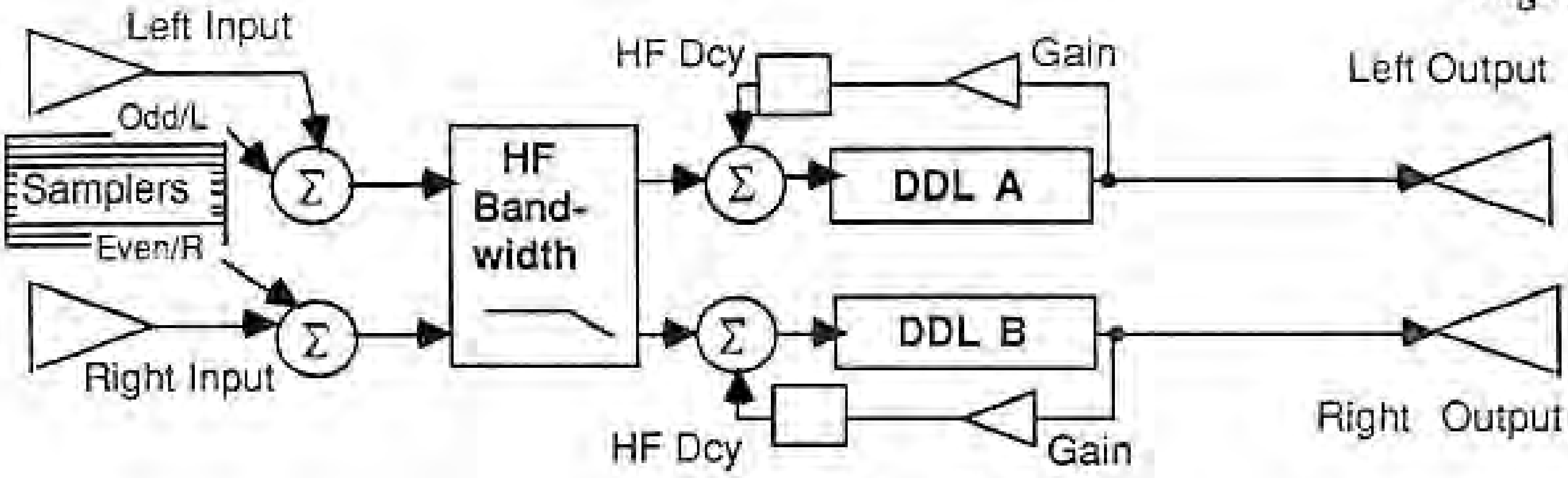
Range and Delay are the "coarse" and "fine" adjustment for the delay times. The ranges are described in detail in Chapter 5. Gain controls feedback. Negative gain values mean that the feedback is 180° out of phase. Depth and Rate control the modulation of the delay taps. In the longer delay ranges (Chorus, Echo, and Long), Rate is adjustable from from 0.01 Hz (one cycle every 100 seconds) to 1 Hz; otherwise it is adjustable from 0.05 to 9.99 Hz.

Samples created with the portable sampler programs can be used in DDL presets, and can appear before and/or after the delays. Odd-numbered and Left-channel samples can feed Delay A, while Even-numbered and Right-channel samples feed Delay B. Only the dry samples appear at the aux outputs.

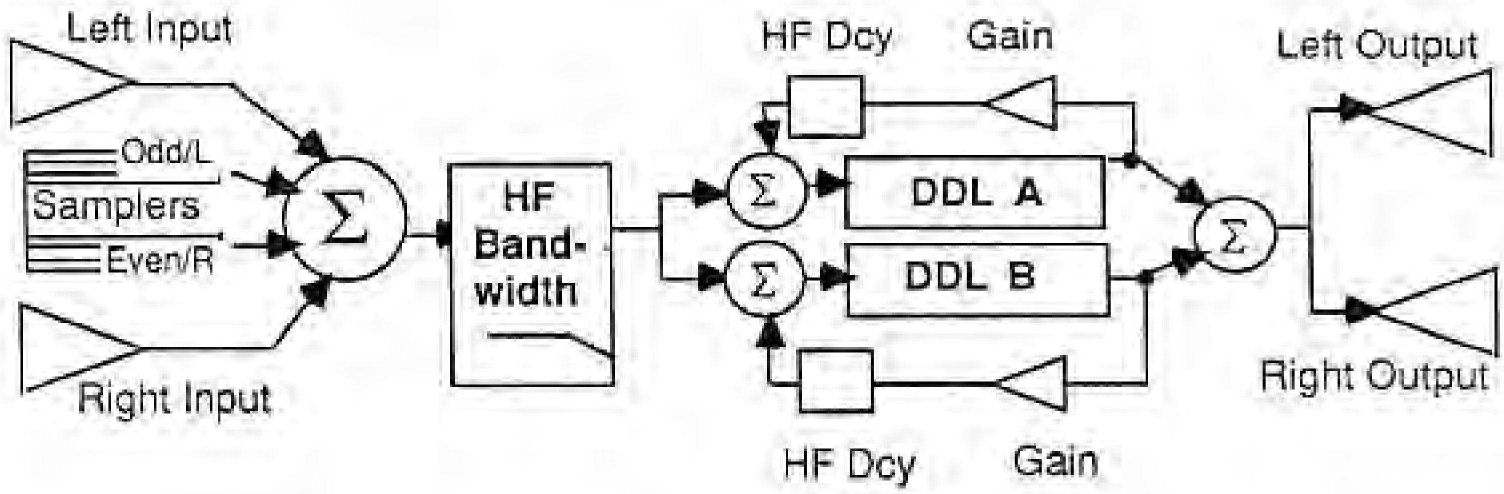
The following Factory presets use this algorithm:

11.1 **Out-In** — long echo with movement from center to sides.
11.2 **Juggle** — delays jump around in space.
11.3 **Pitch** — non-harmonic alternating pitch transpositions.
11.4 **Tunnel** — echoes that alternate in transformation: higher and closer, then lower and further.
11.5 **Stunnl** — stereo version of above, with moving image.
11.6 **Flnge1** — basic stereo flange preset.
11.7 **Flnge2** — mono flange.
11.8 **Flnge3** — two flangers running at different speeds, feeding each other.
11.9 **Flnge4** — extremely short stereo flange.
11.10 **Flnge5** — combination flanging and delayed echo.
11.11 **Bacon** — time-scrambled sound cruncher
11.12 **Trplts** — stereo delays with a triplet feel.
11.13 **Swims** — thick stereo flanging.
11.14 **Swims2** — variation on above.

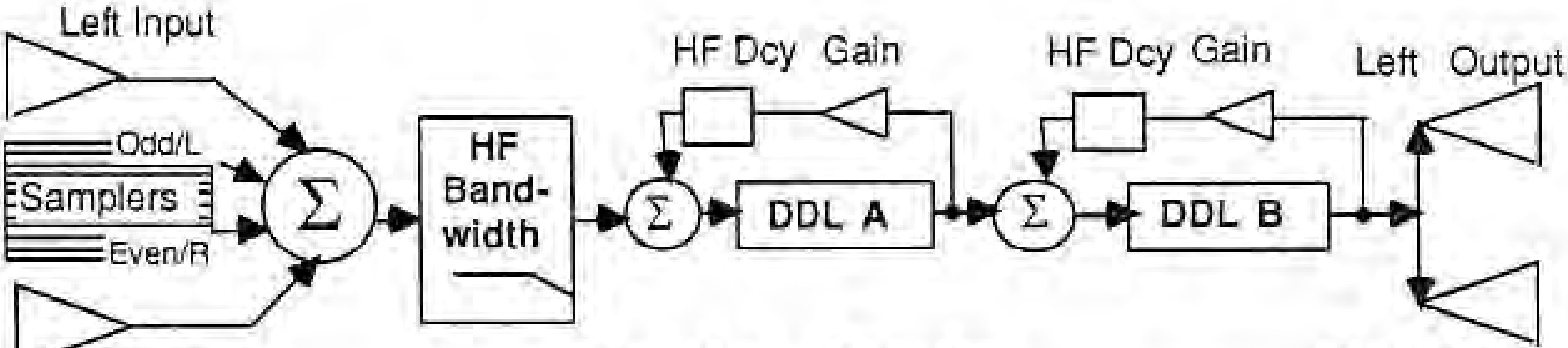
```
                                                  Page 1
              ————Digital Delay A————
     Range   Delay   Gain    HF Dcy   Depth   Rate
max  Long    2000ms  100%    X1.0     100%    9.99Hz
min  Sampls  160us   -100%   X0.1     0%      OFF/0.01Hz

                                                  Page 2
              ————Digital Delay B————
     Range   Delay   Gain    HF Dcy   Depth   Rate
max  Long    2000ms  100%    X1.0     100%    9.99Hz
min  Sampls  160us   -100%   X0.1     0%      OFF/0.01Hz

                                                  Page 3
     ——————————————————————————————————————
     Config           HF BW      Select     Play
max  Chained          15kHz      12         MAN/LIM
min  Split/Parallel   2kHz       1          -42dB/RPT

                                                  Page 4
     ——————————————Input Mix——————————————
     Lin        Rin              Select     Level
max  0dB        0dB              12         0dB
min  OFF        OFF              1          -68dB/OFF

                                                  Page 5
     ——————————————Main Output Mix————————————
     DDL-A   DDL-B                 Select   Level
max  0dB     0dB                   12       0dB
min  OFF     OFF                   1        -68dB/OFF
```



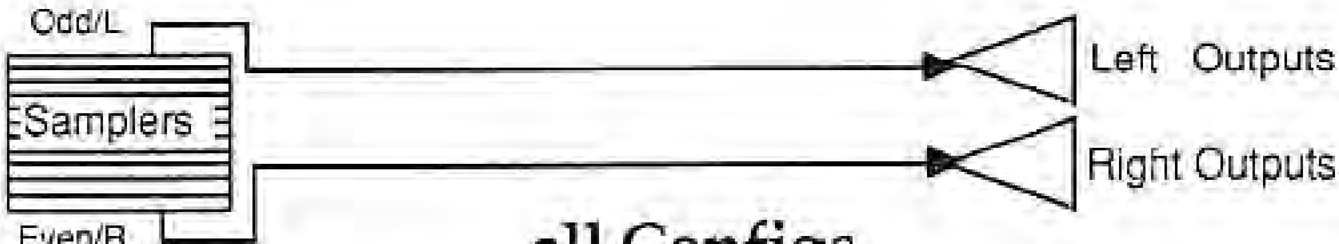
Config = Split



Config = Parallel

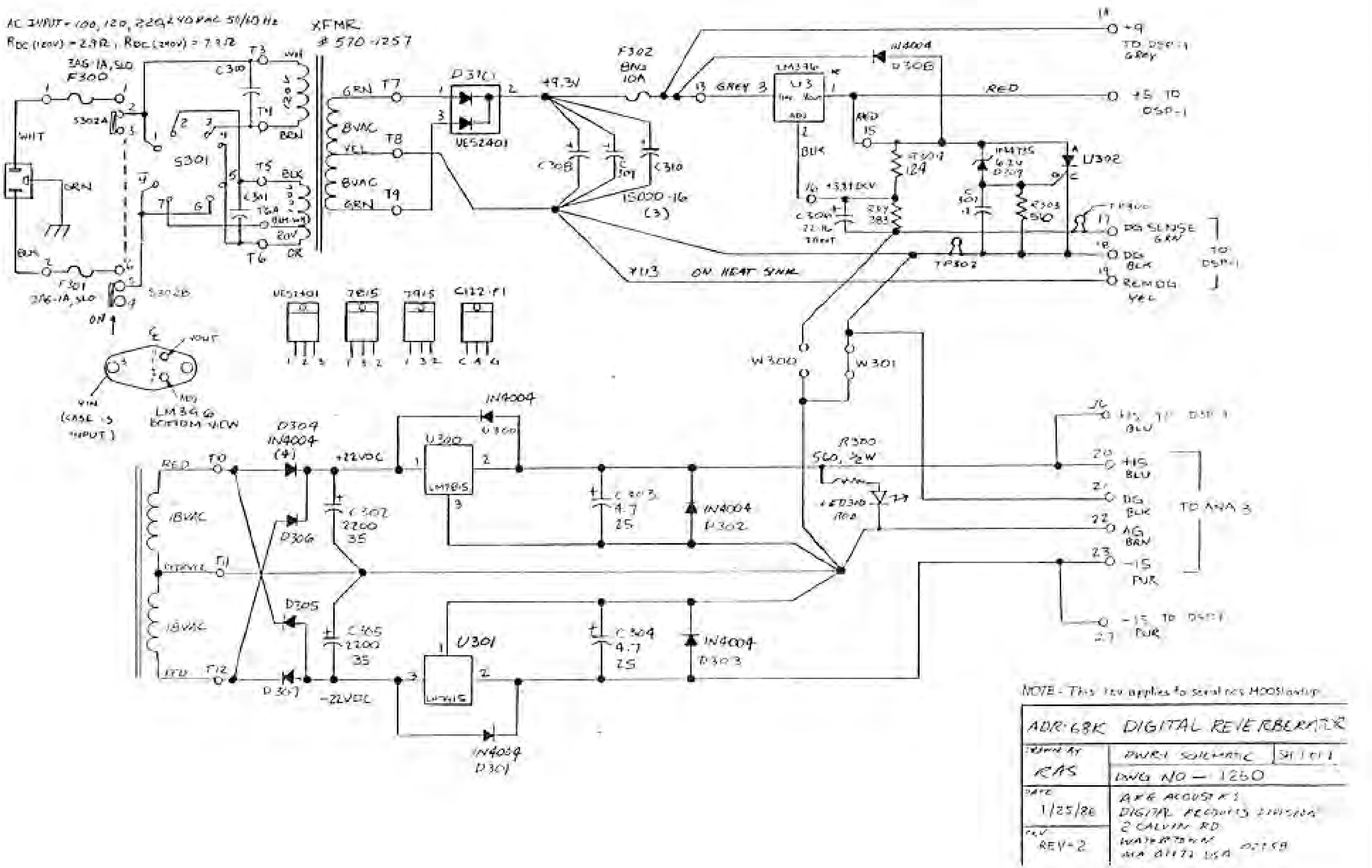


Config = Chain



all Configs

Bank 11 — Dual Delays (DDL)



AC INPUT = 100, 120, 220, 240 VAC 50/60 Hz
RDC (120V) = 2.9Ω, RDC (240V) = 7.7Ω
XFMR
570-1257
3AG 1A, SLO
F300
F301
3AG-1A, SLO
S302A
S302B
S301
ON
WHT
GRN
BLK
C300
C301
T3
T4
T5
T6
T7
T8
T9
BRN
GRN
YEL
8VAC
20V
OR
D310
UES2401
+9.3V
F302
8AG
10A
C308
C309
C310
15000-16
(3)
ON HEAT SINK
LM376
U3
ADJ
BLK
GREY 3
RED
+5 TO
DSP-1
+9
TO DSP-1
GREY
IN4004
D306
R304
124
R303
510
IN4735
6.2V
D307
C301
.1
U302
TP400
TP302
C306
22-16
TANT
DG SENSE
GRN
DG
BLK
REM DG
YEL
TO
DSP-1
UES2401
7815
7915
C122-F1
1 2 3
1 3 2
C A G
VOUT
VIN
(CASE IS
INPUT)
ADJ
LM336
BOTTOM VIEW
W300
W301
RED
18VAC
T10
T11
T12
D304
IN4004
(4)
D305
D306
D307
+22VDC
-22VDC
C302
2200
35
C305
2200
35
U300
LM7815
U301
LM7915
IN4004
U300
C303
4.7
25
C304
4.7
25
IN4004
D302
IN4004
D303
IN4004
D301
R300
560, ½W
LED310
RED
+15 TO DSP-1
BLU
+15
BLU
DG
BLK
AG
BRN
-15
PUR
-15 TO DSP-1
PUR
TO ANA 3
NOTE - This rev applies to serial nos H005 and up.
ADR-68K DIGITAL REVERBERATOR
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REV-2