

***PCM 81***  
*Digital Effects*  
*Processor* User Guide

**lexicon**

## Unpacking and Inspection

After unpacking the PCM 81, save all packing materials in case you ever need to ship the unit. Thoroughly inspect the PCM 81 and packing materials for signs of damage. Report any shipment damage to the carrier at once; report equipment malfunction to your dealer.

## Precautions

Save these instructions for later use.

Follow all instructions and warnings marked on the unit.

Always use with the correct line voltage. Refer to the manufacturer's operating instructions for power requirements. Be advised that different operating voltages may require the use of a different line cord and/or attachment plug.

Do not install the unit in an unventilated rack, or directly above heat producing equipment such as power amplifiers. Observe the maximum ambient operating temperature listed in the product specification.

Slots and openings on the case are provided for ventilation; to ensure reliable operation and prevent it from overheating, these openings must not be blocked or covered. Never push objects of any kind through any of the ventilation slots. Never spill a liquid of any kind on the unit.

This product is equipped with a 3-wire grounding type plug. This is a safety feature and should not be defeated.

Never attach audio power amplifier outputs directly to any of the unit's connectors.

To prevent shock or fire hazard, do not expose the unit to rain or moisture, or operate it where it will be exposed to water.

Do not attempt to operate the unit if it has been dropped, damaged, exposed to liquids, or if it exhibits a distinct change in performance indicating the need for service.

This unit should only be opened by qualified service personnel. Removing covers will expose you to hazardous voltages.

This triangle, which appears on your component, alerts you to the presence of uninsulated, dangerous voltage inside the enclosure... voltage that may be sufficient to constitute a risk of shock.



This triangle, which appears on your component, alerts you to important operating and maintenance instructions in this accompanying literature.

## Notice

This equipment generates and uses radio frequency energy and if not installed and used properly, that is, in strict accordance with the manufacturer's instructions, may cause interference to radio and television reception. It has been type tested and found to comply with the limits for a Class B computing device in accordance with the specifications in Subpart J of Part 15 of FCC Rules, which are designated to provide reasonable protection against such interference in a residential installation. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause interference to radio or television reception, which can be determined by turning the equipment OFF and ON, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient the receiving antenna
- Relocate the computer with respect to the receiver
- Move the computer away from the receiver
- Plug the computer into a different outlet so that the computer and receiver are on different branch circuits.

If necessary, the user should consult the dealer or an experienced radio/television technician for additional suggestions. The user may find the following booklet prepared by the Federal Communications Commission helpful:

"How to identify and Resolve Radio/TV Interference Problems."

This booklet is available from the U.S. Government Printing Office, Washington, DC 20402, Stock No. 004-000-00345-4.

Le présent appareil numérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils numériques de la class B prescrites dans le Règlement sur le brouillage radioélectrique édicté par le ministère des Communications du Canada.

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Dansk

### Vigtig information om sikkerhed

Gem denne vejledning til senere brug.

Følg alle anvisninger og advarsler på apparatet.

Apparatet skal altid tilsluttes den korrekte spænding. Der henvises til brugsanvisningen, der indeholder specifikationer for strømforsyning. Der gøres opmærksom på, at ved varierende driftsspændinger kan det blive nødvendigt at bruge andre lednings- og/eller stiktyper.

Apparatet må ikke monteres i et kabinet uden ventilation eller lige over andet udstyr, der udvikler varme, f.eks. forstærkere. Den maksimale omgivelsestemperatur ved drift, der står opført i specifikationerne, skal overholdes.

Der er ventilationsåbninger i kabinettet. For at sikre apparatets drift og hindre overophedning må disse åbninger ikke blokeres eller tildækkes. Stik aldrig noget ind igennem ventilationsåbningerne, og pas på aldrig at spilde nogen form for væske på apparatet.

Dette apparat er forsynet med et stik med jordforbindelse. Denne sikkerhedsforanstaltning må aldrig omgås.

Udgangsstik fra audioforstærkere må aldrig sættes direkte i apparatet.

Apparatet må ikke udsættes for regn eller fugt og må ikke bruges i nærheden af vand for at undgå risiko for elektrisk stød og brand.

Apparatet må aldrig bruges, hvis det er blevet stødt, beskadiget eller vådt, eller hvis ændringer i ydelsen tyder på, at det trænger til eftersyn.

Dette apparat må kun åbnes af fagfolk. Hvis dækslet tages af, udsættes man for livsfarlig højspænding.



Denne mærkat på komponenten advarer om uisoleret, farlig spænding i apparatet ... høj nok til at give elektrisk stød.



Denne mærkat på komponenten advarer om vigtig drifts- og vedligeholdelsesinformation i den tilhørende litteratur.

Suomi

### Tärkeitä turvallisuusohjeita

Säilytä nämä ohjeet tulevaa käyttöä varten.

Seuraa kaikkia yksikköön merkittyjä ohjeita ja varoituksia.

Käytä aina oikeaa verkkojännitettä. Tehovaatimukset selviävät valmistajan käyttöohjeista. Huomaa, että eri käyttöjännitteet saattavat vaatia toisenlaisen verkkojohdon ja/tai -pistokkeen käytön.

Älä asenna yksikköä telineeseen jossa ei ole tuuletusta, tai välittömästi lämpöä tuottavien laitteiden, esim. tehovahvistimien, yläpuolelle. Ympäristön lämpötila käytössä ei saa ylittää tuotespesifikaation maksimilämpötilaa.

Kotelo on varustettu tuuletusreiillä ja -aukoilla. Luotettavan toiminnan varmistamiseksi ja ylläpennemisen välttämiseksi näitä aukkoja ei saa sulkea tai peittää. Mitään esineitä ei saa työntää tuuletusaukoihin. Mitään nesteitä ei saa kaataa yksikköön.

Tuote on varustettu 3-johtimisella maadoitetulla verkkopistokkeella. Tämä on turvallisuustoiminne eikä sitä saa poistaa.

Älä kytkä audiotehovahvistimen lähtöjä suoraan mihinkään yksikön liittimeen.

Sähköiskun ja palovaaran välttämiseksi yksikkö ei saa olla sateessa tai kosteassa, eikä sitä saa käyttää määrässä ympäristössä.

Älä käytä yksikköä jos se on pudonnut, vaurioitunut, kostunut, tai jos sen suorituskkyky on huomattavasti muuttunut, mikä vaatii huoltoa.

Yksikön saa avata vain laitteeseen perehtynyt huoltohenkilö. Kansien poisto altistaa sinut vaarallisille jännitteille.



Tämä kolmio, joka esiintyy komponentissasi, varoittaa sinua eristämättömän vaarallisen jännitteen esiintymisestä yksikön sisällä. Tämä jännite saattaa olla riittävän korkea aiheuttamaan sähköiskuvaaran.



Tämä kolmio, joka esiintyy komponentissasi, kertoo sinulle, että tässä tuotedokumentoinnissa esiintyy tärkeitä käyttö- ja ylläpito-ohjeita.

Norsk

### Viktig informasjon om sikkerhet

Ta vare på denne veiledningen for senere bruk.

Følg alle anvisningene og advarslene som er angitt på apparatet.

Apparatet skal alltid anvendes med korrekt spenning. Produktbeskrivelsen inneholder spesifikasjoner for strømkrav. Vær oppmerksom på at det ved ulike driftsspenninger kan være nødvendig å bruke en annen ledning- og/eller støpseltype.

Apparatet skal ikke monteres i skap uten ventilasjon, eller direkte over varmeproduerende utstyr, som for eksempel kraftforstærkere. Den maksimale romtemperaturen som står oppgitt i produktbeskrivelsen, skal overholdes.

Apparatet er utstyrt med ventilasjonsåpninger. For at apparatet skal være pålitelig i bruk og ikke overopphetes, må disse åpningene ikke blokeres eller tildekkes. Stikk aldri noe inn i ventilasjonsåpningene, og pass på at det aldri søles noen form for væske på apparatet.

Dette apparatet er utstyrt med et jordet støpsel. Dette er en sikkerhetsforanstaltning som ikke må forandres.

Utgangsplugg fra audioforstærkere skal aldri koples direkte til apparatet. Unngå brannfare og elektrisk støt ved å sørge for at apparatet ikke utsettes for regn eller fuktighet og ikke anvendes i nærheten av vann.

Apparatet skal ikke brukes hvis det har blitt utsatt for støt, er skadet eller blitt vått, eller hvis endringer i ytelsen tyder på at det trenger service.

Dette apparatet skal kun åpnes av fagfolk. Hvis dekslet fjernes, utsettes man for livsfarlig høyspenning.



Komponenten er merket med denne trekanten, som er en advarsel om at det finnes uisolert, farlig spenning inne i kabinettet ... høy nok til å utgjøre en fare for elektrisk støt.



Komponenten er merket med denne trekanten, som betyr at den tilhørende litteraturen inneholder viktige opplysninger om drift og vedlikehold.

Svenska

### Viktiga säkerhetsföreskrifter

Spara dessa föreskrifter för framtida bruk.

Följ alla anvisningar och varningar som anges på enheten.

Använd alltid rätt nätspänning. Se tillverkarens bruksanvisningar för information om effektkrav. Märkväl, att andra matningsspänningar eventuellt kräver att en annan typs nätsladd och/eller kontakt används.

Installera inte enheten i ett oventilerat stativ, eller direkt ovanför utrustningar som avger värme, t ex effektförstärkare. Se till att omgivningens temperatur vid drift inte överskrider det angivna värdet i produktspecifikationen.

Behållaren är försedd med hål och öppningar för ventilering. För att garantera tillförlitlig funktion och förhindra överhettning får dessa öppningar inte blockeras eller täckas. Inga föremål får skuffas in genom ventilationshålen. Inga vätskor får spillas på enheten.

Produkten är försedd med en jordad 3-trådskontakt. Detta är en säkerhetsfunktion som inte får tas ur bruk.

Anslut aldrig audioeffektförstärkarutgångar direkt till någon av enhetens kontakter.

För att undvika elstöt eller brandfara får enheten inte utsättas för regn eller fukt, eller användas på ställen där den blir våt.

Använd inte enheten om den har fallit i golvet, skadats, blivit våt, eller om dess prestanda förändrats märkbart, vilket kräver service.

Enheten får öppnas endast av behörig servicepersonal. Farliga spänningar blir tillgängliga när locken tas bort.



Denna triangel, som visas på din komponent, varnar dig om en uisolerad farlig spänning inne i enheten. Denna spänning är eventuellt så hög att fara för elstöt föreligger.



Denna triangel, som visas på din komponent, anger att viktiga bruksanvisningar och serviceanvisningar ingår i dokumentationen i fråga.

Deutsch  
**Wichtige Sicherheitsanweisungen**

Heben Sie sich diese Sicherheitsanweisungen auch für später auf. Befolgen Sie alle auf der Vorrichtung stehenden Anweisungen und Warnungen. Immer nur mit der richtigen Spannung verwenden! Die Gebrauchsanweisungen des Herstellers informieren Sie über die elektrischen Anforderungen. Vergessen Sie nicht daß bei verschiedenen Betriebsspannungen ggf. auch verschiedene Leitungskabel und/oder Verbindungsstecker zu verwenden sind.

Stellen Sie die Vorrichtung nicht in ein unbelüftetes Gestell oder unmittelbar über wärmeerzeugende Geräte wie z.B. Tonverstärker. Halten Sie die in den Produktspezifikationen angegebene maximale Umgebungstemperatur bei Betrieb ein.

Schlitze und Öffnungen im Gehäuse dienen der Belüftung; um verlässlichen Betrieb sicherzustellen und Überheizen zu vermeiden dürfen diese Öffnungen nicht verstopft oder abgedeckt werden. Stecken Sie nie irgend einen Gegenstand durch die Belüftungsschlitze. Vergießen Sie keine Flüssigkeiten auf den Apparat.

Dieses Produkt ist mit einem 3-drahtigen Erdungsstecker ausgerüstet. Diese Sicherheitsmaßnahme darf nicht unwirksam gemacht werden.

Schließen Sie nie Tonverstärker unmittelbar an einen Anschluß des Apparates an.

Um elektrischen Schlag oder Feuer zu vermeiden, setzen Sie den Apparat weder Regen noch Feuchtigkeit aus und betreiben Sie ihn nicht dort wo Wasser eindringen könnte.

Versuchen Sie nicht den Apparat zu betreiben falls er fallen gelassen, beschädigt, oder Flüssigkeiten ausgesetzt wurde, oder falls sich seine Arbeitsweise derart ändert daß daraus ein Bedarf nach Reparatur zu schließen ist.

Dieser Apparat sollte nur von qualifizierten Fachleuten geöffnet werden. Das Abnehmen von Abdeckungen setzt Sie gefährlichen Spannungen aus.



Dieses Dreieck auf Ihrem Apparat warnt Sie vor nicht-isolierter, gefährlicher Spannung im Gehäuse ... stark genug um eine Berührungsfahrgefahr darzustellen.



Dieses Dreieck auf Ihrem Apparat bedeutet daß wichtige Betriebs- und Wartungsanweisungen in der mitgelieferten Dokumentation zu finden sind.

Español  
**Instrucciones importantes de seguridad**

Guarde estas instrucciones para uso posterior.

Utilice siempre el voltaje correcto. Dirijase a las instrucciones de operación del fabricante para obtener las especificaciones de potencia. Está al tanto de que voltajes de operación distintos requieren el uso de cables y/o enchufes distintos.

No instale esta unidad en un estante sin ventilación, ni tampoco directamente encima de equipos que generen calor tales como amplificadores de potencia. Fijese en las temperaturas ambientales máximas de operación que se mencionan en las especificaciones del producto.

Las aperturas y ranuras del chasis sirven para proveer la ventilación necesaria para operar la unidad con seguridad y para prevenir sobrecalentamiento, y por lo tanto no pueden ser obstruidas o cubiertas. No introduzca objetos de ningún tipo a través de las ranuras de ventilación, y nunca deje caer ningún líquido sobre la unidad.

Este producto está equipado con un enchufe de 3 clavijas con conexión a tierra. Éste es un elemento de seguridad que no debe ser eliminado.

Nunca conecte ningún tipo de salida de amplificadores de sonido directamente a los conectores de la unidad.

Para prevenir descargas eléctricas o incendios, mantenga la unidad alejada de la lluvia, humedad o cualquier lugar en el que pueda entrar en contacto con agua.

No trate de hacer funcionar la unidad si se ha caído, está dañada, ha entrado en contacto con líquidos, o si nota cualquier cambio brusco en su funcionamiento que indique la necesidad de hacerle un servicio de mantenimiento.

Esta unidad deberá ser abierta únicamente por personal calificado. Si usted quita las coberturas se expondrá a voltajes peligrosos.



Este triángulo que aparece en su componente le advierte sobre la existencia dentro del chasis de voltajes peligrosos sin aislantes ... voltajes que son lo suficientemente grandes como para causar electrocución.



Este triángulo que aparece en su componente lo alerta sobre las instrucciones de operación y mantenimiento importantes que están en los materiales de lectura que se incluyen.

Français  
**Instructions de Sécurité Importantes**

Gardez ces instructions pour référence future.

Observez toutes les instructions et tous les avertissements marqués sur l'appareil.

Branchez uniquement sur un réseau de tension indiquée. Consultez le manuel d'instruction du fabricant pour les spécifications de courant. N'oubliez pas que différentes tensions peuvent nécessiter l'utilisation de câbles et/ou de fiches de connexion différents.

N'installez pas l'appareil en un compartiment non-aéré ou directement au-dessus d'équipements générateurs de chaleur, tels qu'amplificateurs de courants, etc. Ne dépassez pas la température ambiante maximale de fonctionnement indiquée dans les spécifications du produit.

Des fentes et ouvertures sont prévues dans le boîtier pour l'aération; Pour assurer le bon fonctionnement et pour prévenir l'échauffement, ces ouvertures ne doivent pas être couvertes ou bloquées. N'insérez pas d'objets dans les fentes d'aération. Empêchez tout liquide de se répandre sur l'appareil.

Ce produit est muni d'une fiche à trois fils pour la mise à terre. Ceci est une mesure de sécurité et ne doit pas être contrariée.

Ne connectez jamais d'amplificateurs audio directement aux connecteurs de l'appareil.

Pour empêcher les chocs électriques et le danger d'incendie, évitez d'exposer l'appareil à la pluie ou à l'humidité, et ne le mettez pas en marche en un endroit où il serait exposé aux éclaboussures d'eau.

N'essayez pas de faire fonctionner l'appareil s'il est tombé à terre, a été endommagé, exposé à un liquide, ou si vous observez des différences nettes dans son fonctionnement, indiquant la nécessité de réparations.

Cet appareil ne doit être ouvert que par un personnel de service qualifié. En enlevant les couvercles vous vous exposez à des tensions électriques dangereuses.



Ce triangle, sur votre appareil vous avertit de la présence de tension dangereuse, non-isolée à l'intérieur du boîtier...une tension suffisante pour représenter un danger d'électrocution.



Ce triangle sur sur votre appareil vous invite de suivre d'importantes instructions d'utilisation et d'entretien dans la documentation livrée avec le produit.

Italiano  
**Importanti norme di sicurezza**

Conservare le presenti norme per l'utilizzo futuro.

Osservare tutte le istruzioni e le avvertenze apposte sull'unità.

Utilizzare esclusivamente con la tensione di rete corretta. Consultare le istruzioni operative fornite dal fabbricante per i dati riguardanti la tensione e l'assorbimento di corrente. Potrebbe essere necessario l'uso di cavi di rete e/o di spine diverse a seconda della tensione utilizzata.

Non installare l'unità in uno scaffale privo di ventilazione oppure direttamente sopra una fonte di calore, come, ad esempio, un amplificatore. Non superare la temperatura ambientale massima di funzionamento riportata nei dati tecnici del prodotto.

Le fessure e le altre aperture nella scatola servono alla ventilazione. Per un funzionamento affidabile, e per evitare un eventuale surriscaldamento, queste aperture non vanno ostruite o coperte in nessun modo. Evitare in tutti i casi di inserire oggetti di qualsiasi genere attraverso le fessure di ventilazione. Non versare mai del liquido di nessun tipo sull'unità.

Questo prodotto viene fornito con una spina a 3 fili con massa. Tale dispositivo di sicurezza non va eliminato.

Evitare sempre di collegare le uscite dell'amplificatore audio direttamente ai connettori dell'unità.

Per prevenire il pericolo di folgorazione e di incendio non esporre l'unità alla pioggia o ad un'umidità eccessiva; evitare di adoperare l'unità dove potrebbe entrare in contatto con acqua.

Evitare di adoperare l'unità se la stessa è stata urtata violentemente, se ha subito un danno, se è stata esposta ad un liquido o in caso di un evidente cambiamento delle prestazioni che indichi la necessità di un intervento di assistenza tecnica.

Ogni intervento sull'unità va eseguito esclusivamente da personale qualificato. La rimozione della copertura comporta l'esposizione al pericolo di folgorazione.



Il presente triangolo impresso sul componente avverte della presenza di tensioni pericolose non isolate all'interno della copertura... tali tensioni rappresentano un pericolo di folgorazione



Il presente triangolo impresso sul componente avverte l'utente della presenza nella documentazione allegata di importanti istruzioni relative al funzionamento ed alla manutenzione.

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	<b>7. Specifications</b>	

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Thank you for your purchase of the PCM 81, one of Lexicon's most powerful and versatile digital effects processors. The PCM 81 brings you exciting new effects with extensive processing and control capabilities, and uncompromising sonic clarity.

## Introduction

The PCM 81 contains a built-in library of 300 preset programs that provide a comprehensive array of effects ranging from beautiful and lush to completely wild. The presets are organized into 6 Banks of 50, and functionally grouped for a wide range of applications. Be sure to experiment with all 300 presets to get a feel for the full range of PCM 81 capabilities.

## The Presets

<b>Bank P0</b> Multi Effects Modulation Effects Special Effects	<b>Bank P1</b> Rhythmic Echo & Delay Effects Ambience Effects	<b>Bank P2</b> EQ Effects Spatial Effects Gain Effects Resonant Chord Effects	<b>Bank P3</b> Reverb Effects Processed Reverb Effects Remix Effects	<b>Bank P4</b> Vocal Shift Vocal Harmony Guitar Magic Instrument Shift & Harmony Percussion & Tempo	<b>Bank P5</b> Sci-Fi FX Pitch & Delay Pitch Sequences Pads & Drones Utility Programs Pitch Correct
--	--	--	--	---	---

Every preset has one or more of its parameters patched to the front panel ADJUST knob, giving you instant control over the primary aspect of the effect — without going into Edit mode. In the preset, *Concert Hall*, for example, Mid Reverb Time is patched to ADJUST. When you turn the knob, you will adjust the reverb decay. In *Prime Blue*, ADJUST is patched to several parameters, so that turning the knob changes the effect from a tight chorus, to a chorus with recirculating echoes and, finally, into a reverb.

The PCM 81 uses 17 stereo algorithms to create different types of effects. Each algorithm includes an uncompromised stereo reverb effect, as well as several voices of additional stereo effects. There are three general classes of algorithm: 4-Voice, 6-Voice and Pitch. The 4-Voice algorithms: Concert Hall, Plate, Chamber, Inverse and Infinite each combine a specific type of reverberation with a 4-voice, general purpose, stereo effect “toolbox”, as well as additional post-processing for the reverb. The 6-Voice algorithms: Glide>Hall, Chorus+Rvb, M-Band+Rvb, Res1>Plate and Res2>Plate each combine a specific type of reverberation with a specialized 6-voice stereo effect. The Pitch algorithms: Quad>Hall, Dual-Chmb, Dual Plt, Dual Inv, Stereo-Chmb, VSO-Chmb and Pitch Correct each offer a unique type of pitch shifting, combined with an uncompromised Lexicon reverb.

## The Algorithms

Many of the PCM 81 algorithms include parameters that make it possible to create dynamic spatialization effects for 2-channel or surround applications. Check out the Ambience Effects in Bank P1 and the Spatial Effects in Bank P2 for examples. The Dual FX Pitch algorithms have dedicated Submixer controls that allow instant reconfiguration of the reverb and effects blocks.

The PCM 81 gives you a unique set of tempo controls. Tempos can be tapped in with the front panel Tap button (or an assigned controller) or “dialed-in”, in BPM (beats per minute) on the display. The PCM 81 also lets you generate MIDI clock from your tempo, as well as receive MIDI tempo from an external sequencer or drum machine. In the PCM 81, tempo can control LFO speeds and

## Tempo Control

Time Switch controls, as well as all delay parameters, ensuring that all of your modulations are in tempo with your music. You can even set independent rhythmic values for different parameters within a single program.

Tempo can be set and displayed in either rhythmic value or time values. Many presets, particularly the Rhythmic Echo & Delay Effects in Program Bank P1, have delay times assigned to Tap tempo. Try loading some of these and pressing **Tap** twice in rhythm to change tempo.

**Editing** An enormous range of editing control is provided for each algorithm, with parameters organized in an edit matrix of as many as 100 main controls. In addition to providing this powerful sound design capability, the PCM 81 also allows you to customize these controls for your day-to-day editing needs, or to use a subset of controls specially designed for each preset.

The PCM 81 has two levels of Edit Mode control called **Go** mode and **Pro** mode. In **Go** mode, the most useful parameters within an effect are grouped for instant access via the front panel Edit button. Each preset has a specially selected set of **Go** mode parameters which let you make value changes to the effect without losing the character of the sound. **Pro** mode gives you access to the full parameter editing matrix for the algorithm of any loaded effect when you press Edit. In this mode, you can access a complete set of Modulation and Patching parameters, create your own ADJUST knob patch and assign your own **Go** mode parameters.

A unique Patching and Modulation system provides unprecedented control over your effects, with a versatile set of internal modulators: LFO, AR Envelope, Envelope Follower, Latch and Time Switches. These allow you to create modulation sweeps which move in time with music, or wildly animated effects. You can create as many as 10 patches per effect, each with as many as 8 pivot points. You can patch multiple parameters to a single controller, or patch multiple sources to a single destination.

**Memory Cards** The front panel memory slot will accept industry standard PCMCIA SRAM cards (up to 1Meg). Use these cards to store effects (as many as 2350 on a 1 Meg card), system setups, MIDI program maps, and more.

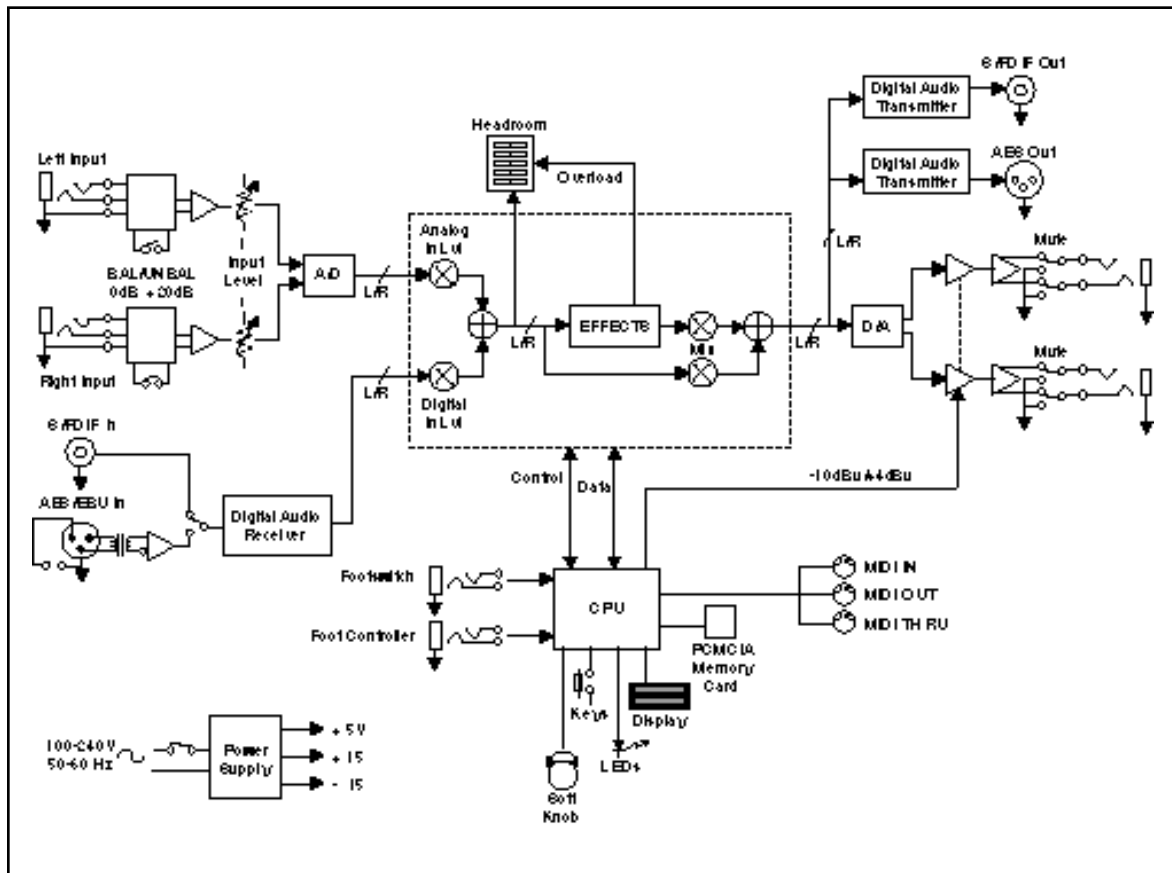
PCM 81 and PCM 80 SRAM cards are readable by either system.

**User Interface** For all of its programming power and flexibility, you'll find the PCM 81 simple to use. The large, 2-line fluorescent display is easy to see from any angle whether the surroundings are bright or dark. Separate SELECT and ADJUST knobs make program loading and editing quick and easy. We've even designed in a special Info mode — press and hold any button to find out what its function is, or to get status information such as the name of the running effect, current tempo rate, etc.

To get the most out of the PCM 81, we suggest that you invest the time to explore this manual. We think you'll agree that the time spent investigating will reward you with enjoyment of its full capabilities.

## Product Overview

### Block Diagram



## The Front Panel

### Headroom

5-position indicator for analog and digital signal levels and overload conditions.

### INPUT

Adjusts analog input level.

### Display

Two rows of 20 alphanumeric characters display effect names and ID numbers, and parameter names and values.

### ADJUST

In Edit mode, changes values of parameters chosen with SELECT. With Program Banks or Register Banks selected, behaves as a soft knob for patched parameters.

### SELECT

Scrolls through presets, registers or parameters. With Program Bank selected, scrolls through the 50 programs in the selected bank. With Edit selected, scrolls only through the parameters of the active row.

### POWER

On/Off.

### Memory Card

Slot for optional preset ROM or register RAM cards. Press Eject button to remove card.



### Up/Down

Press to move up and down through a program, register, or parameter matrix.

### Program Banks

Enables selection of factory presets. Press repeatedly to cycle selection of 4 internal preset banks. Press and hold to display the name of the current bank.

### Load/\*

In Program or Register mode, loads the selected program. In Edit mode, scrolls through any multi-field parameter.

### Register Banks

Enables selection of user memory. If a RAM card is loaded into the Memory Card slot, each press of this button selects a new register bank. Press and hold to display the name of the current bank.

### Store

Initiates register store function.

### Edit

Enables parameter selection for editing of values.

### Compare

Active in Program, Register, and Edit modes. Press to compare the active version of the current effect with the most recently stored version.

### Control

Enables selection of system and global parameters.

### Bypass

Bypasses or mutes audio, depending on the setting of each program's bypass parameter.

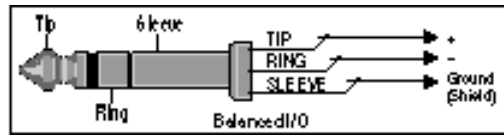
### Tempo

Press to display tempo rate and to initiate tempo functions. LED flashes in time with current tempo rate.

### Tap

Sets tempo. Press twice in rhythm to establish tempo rate. Press once to reset LFO.

## The Rear Panel



### AES/EBU and S/PDIF Inputs

AES/EBU format digital connectors conform to AES professional standards. S/PDIF format digital connectors conform to CP-340 Type II and IEC-958 consumer standards. Only one of these options (AES or S/PDIF) may be selected for input.

### Balanced Outputs

Output impedance is 125Ω, each side, balanced, and levels up to +18dBu maximum full scale. 1/4" phone connectors and XLRs provided. Both S/PDIF and AES outputs are active at all times.

### Input Level

2-position (In/Out) switch for matching input gain to the source being used. In position adds 20dB of input gain (unbalanced) to the input stages. Out position provides 0dB of gain (balanced).

### Balanced Inputs

Combined 3 pole XLR and 1/4" jacks, electronically balanced.

Input impedance is 50kΩ unbalanced, and 100kΩ balanced. Inputs accept input levels from -22dBu to +20dBu.



### AC Power

Standard 3-pin IEC power connector. 100-240V, 50-60Hz automatic switching to correct voltage range.

### MIDI

**IN**  
Receives MIDI information from other MIDI equipment such as master keyboard controllers, MIDI foot controllers, sequencers and synthesizers.

### THRU

Passes any MIDI data received without change.

### OUT

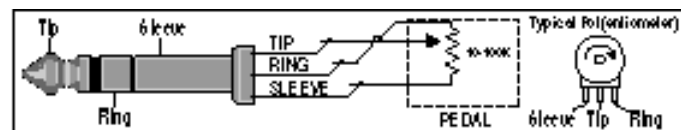
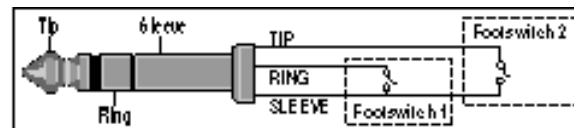
Transmits MIDI data to other equipment.

### Footswitch

1/4" Tip/Ring/Sleeve phone jack for two independent momentary footswitches

### Foot Controller

1/4" Tip/Ring/Sleeve phone jack provided for footpedal with 10kΩ to 100Ω impedance.



## Installation Notes

**Mounting** The PCM 81 uses one EIA-standard rack space, and can be mounted on any level surface or in a standard 19 inch (483 mm) rack. If the PCM 81 is mounted in a rack or road case, support the rear of the chassis to prevent possible damage from mechanical shock and vibration.

The maximum ambient operating temperature is 104°F (40°C). Provide adequate ventilation if the PCM 81 is mounted in a closed rack with heat-producing equipment such as power amplifiers.

**Power Requirements** The PCM 81 is equipped with a 3-pin IEC power connector and detachable cord.

The PCM 81 will operate with power sources from 100 to 240 volts AC, 50-60Hz. Power switching to actual line voltage is automatic.

### Audio Connections **Analog Audio**

For best performance, maintain balanced connections, and use high-quality, low-capacitance, twisted-shielded pair cable.

When connecting to single-ended, unbalanced devices, connect the low side to signal ground at the unbalanced piece of equipment. Output level does not change when connected to an unbalanced input.

#### **Mono Applications**

Use a Y-connector inserted at the analog inputs and outputs to have the signal summed to mono.

#### **NOTE**

Be careful to keep input and output to all channels wired consistently. Out-of-phase wiring can produce audible effects.

#### **Digital Audio**

S/PDIF (CP-340 Type II) Consumer Digital Audio connections require 75Ω coaxial cable suited for digital audio or video signals. Audio grade cable is *not* suitable.

AES/EBU connections require balanced connections using high-quality, low-capacitance, controlled-impedance, data communication, twisted-shielded pair cable. **Microphone cable may introduce a significant amount of jitter into the signal, causing distortion.**

**Footswitch/Foot Controller**

One 1/4 inch T/R/S phone jack is provided for 2 momentary footswitches. Another 1/4 inch T/R/S phone jack is provided for a footpedal (minimum 100Ω to maximum 10k impedance). Normally open or normally closed momentary switches are suitable. At power on, the PCM 81 assumes the switch is off. Use shielded, twisted-pair cable with shield connected to sleeve. See diagram on page 3.

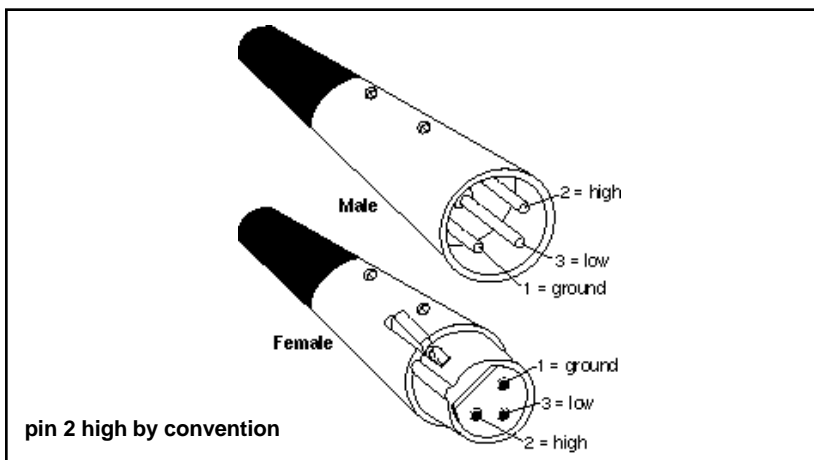
**Control Connections**

**MIDI**

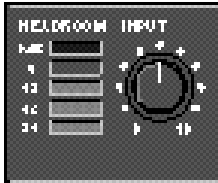
5-pin DIN connectors are provided for MIDI IN, THRU and OUT. Use standard 5-pin DIN MIDI cable assemblies, available from your local dealer.

Signal	Mating Connector	Description
L and R Analog Audio Input	XLR A3M	Active balanced, pin 2 high +2dBu min; +20dBu max at 0dB setting
L and R Analog Audio Output	XLR A3F	Active balanced, pin 2 high -2dBu to +18dBu at full scale output
AES/EBU Digital Input	XLR A3M	Balanced RS-422 pin 2 high
AES/EBU Digital Output	XLR A3F	Balanced RS-422 pin 2 high
S/PDIF CP-340 Type II Consumer Digital Audio Input and Output	1/4"	EIAJ Consumer Digital Audio Format tip high
MIDI In MIDI Out MIDI Thru	5-pin DIN	Standard MIDI Interface

**Connectors**



## Setting Audio Levels



The PCM 81, with both analog and digital input and output connections, requires some attention to proper setting of signal level.

Analog inputs are first gain-conditioned by the rear panel input gain switch, and then by the front panel INPUT knob. Proper setting of both the switch and knob are important for best performance of the A/D converter. Audio data from the A/D converter is level adjusted by the Analog Lvl parameter before reaching the effects processors. Digital inputs are also level adjusted before reaching the effects processors via the Digital Lvl parameter.

Analog and the selected digital source are mixed at the input to the effects processors. For example, setting both Analog Lvl and Digital Lvl to 50% will mix the analog and the selected digital input signals equally and send them to the effects. Creating a mix which exceeds 100% can cause overload.



Proper setting of Input level on the PCM 81 is dependent on:

- Proper signal level into the analog front end to avoid signals causing overload at the DSP input
- Proper adjustment of the signal level into the analog-to-digital converter to optimize noise and avoid overload
- Proper setting of signal level into the digital signal processor to optimize noise.

### Headroom Display

The headroom display provides both headroom and overload information from a variety of measurement points. The meters display the sum of both the analog and the digital input data. Examining either the analog or the digital level alone requires that the Level parameter of the subject data stream be set to 100%, while the Level parameter of the other is set to 0%.

The chart below illustrates the adjustment range that will set input levels for both balanced and unbalanced operation. When a choice can be made, it is best to operate at the higher amplitude end of the recommended range to optimize noise performance.

	 <b>Unbalanced</b>	 <b>Balanced</b>
overload:	> 0dBu	>+20dBu
acceptable:	0dBu to -22dBu	+20dBu to -2dBu
too low (noisy):	<-22dBu	<-2dBu

### Overload

The 0db (overload) indicators will light under the following conditions:

- A/D overload
- overload at any point in effects processing

For example, internal peaking of high Q filters, or level buildup from certain reverberation modes can result in overload, even when the input A/D or digital receiver data stream is not at full scale. Such conditions are most often caused by a combination of extreme parameter settings. Adjusting parameter/level settings can eliminate these overload conditions.

### Selecting a Digital Input Source

1. Press **Control**.
2. Press **Up** or **Down** until the leftmost digit in the lower lefthand corner of the display is **0**.
3. Turn SELECT to **0.0 Word Clock**, and turn ADJUST to display **Ext: XLR or Ext: Coax**, depending on the input you are using.

### Selecting Word Size

1. Press **Control**.
2. Press **Up** or **Down** until the leftmost digit in the lower lefthand corner of the display is **0**.
3. Turn SELECT to **0.3 Word Size**, and turn ADJUST to display desired Word Size.

NOTE: When using analog outputs as primary outputs, set **Word Size** to **20 bits**.

### Setting Analog and Digital Input Level

1. Press **Control**.
2. Press **Up** or **Down** until the leftmost digit in the lower lefthand corner of the display is **0**.
3. Turn SELECT to **0.2 Dig In Lvl**, and turn ADJUST to display **0%**.

NOTE: If you are not running digital audio, controlled by External Clock, into the PCM 81, the digital audio input will be disabled or muted. Until there is valid digital audio input, select **0.0 External** to enable the digital input level control. Until valid digital audio is connected, the **Dig In Lvl** control will remain muted.

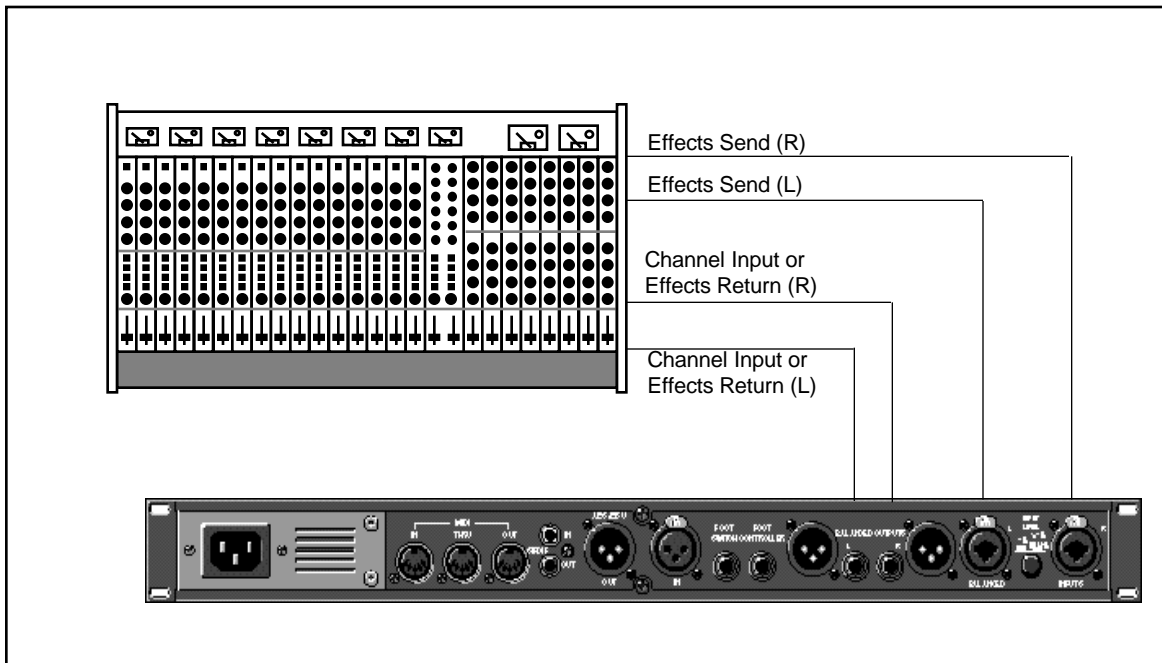
4. Turn SELECT to **0.1 Analog In Lvl**, and turn ADJUST to display **100%**.
5. Adjust the front panel INPUT knob so that program material level peaks cause the headroom display to reach the top of the column *without* lighting the overload indicators. An occasional large signal peak causing momentary flashing of the overload indicator is acceptable in most instances, but should be validated by listening to the actual result.
6. If you are running digital audio, turn SELECT back to **0.2 Dig In Lvl**, and turn ADJUST to the desired level. You may want to back off the **Analog In Lvl** setting to prevent the analog/digital mix from overloading the effects processor.

### Setting Output Level

While still in Control mode, turn SELECT to **0.6 Output Level**. The Output Level parameter has two range positions. The appropriate position depends on the level handling capability of the device connected to the analog outputs. Devices capable of handling outputs with peak levels of 18dBu require setting **Output Level** to the **+4dBu** setting. Devices which cannot handle peak levels greater than +4dBu require the **-10dBu** setting.

## Configurations

### Connection to a mixing console's effects sends



If you will be using a PCM 81 as your primary effects unit, and your system includes a console with one or more auxiliary (effects) sends, connect the PCM 81 as shown above. In most applications, it is preferable to connect the PCM 81 outputs to two of the console's input channel strips, panned full left and right, rather than to the effects returns. This allows the greatest flexibility in routing and equalization.

In this configuration the console controls are used to set the amount of effect heard—the PCM 81's MIX control should be set for 100% wet. To assign a global MIX setting:

1. Press **Control**.
2. Press **Up** or **Down** until **1.x** is displayed in the lower left of the display and **System** is displayed on the upper line.
3. Turn SELECT until **System Mix Mode** is displayed on the upper line. **1.1** will be displayed in the lower left.
4. Turn ADJUST until the lower line reads:  
**1.1 ✱ Global**
5. Press **Load / ✱** to show the current global setting of MIX; use ADJUST to set it to 100% wet.

You can use Memory cards to store as many as 2350 PCM 81 registers (47 banks of 50 — on a 1 Meg card). Registers stored on a properly formatted card will be recognized whenever the card is inserted, and can be accessed via the front panel **Register Banks** button, exactly as internal registers.

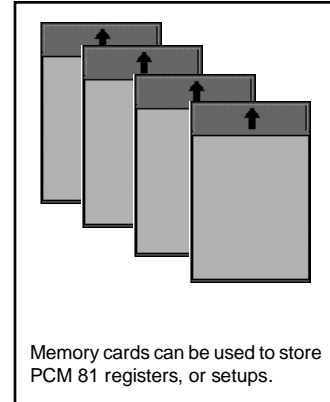
Memory cards can also be used to store "setups" (your system configuration, as set in Control mode). As many as 5 PCM 81 setups can be stored on a card, allowing you to transport not only your effects, but complete PCM 81 environments to another PCM 81. Cards also provide storage for additional program maps and effect chains.

See *Control Mode* Store and Load functions for details on saving setups on a card and reloading them.

Memory cards must be of the following type:

<b>PCMCIA SRAM Memory Card — 68 pin, Type I</b>	
Usable densities:	64 kByte 128 kByte 256 kByte 1 MByte
Access Time:	250 nsec or faster
Conforms to PCMCIA 2.0/JEDIA 4.1. Can use either 8-bit or 8/16-bit bus configuration. Attribute memory can be present, but is not used.	

## Memory Cards



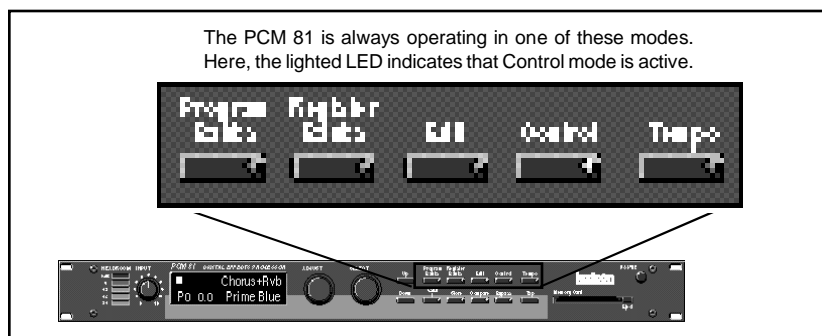


## Basic Operation

The PCM 81 provides a wide range of control over an extraordinary set of reverb, delay, pitch and modulation effects. All of the controls are easily accessed from the front panel and are described in detail in this section.

The PCM 81 has five basic modes of operation, each of which is selected by pressing a front panel button (**Program Banks**, **Register Banks**, **Edit**, **Control** and **Tempo**). Each of these first four mode buttons has an LED which lights when the mode is active. The Tempo LED (unless you elect to have this function turned off) flashes the current tempo. When Tempo mode is active, no other mode LEDs will be lighted.

## Modes of Operation

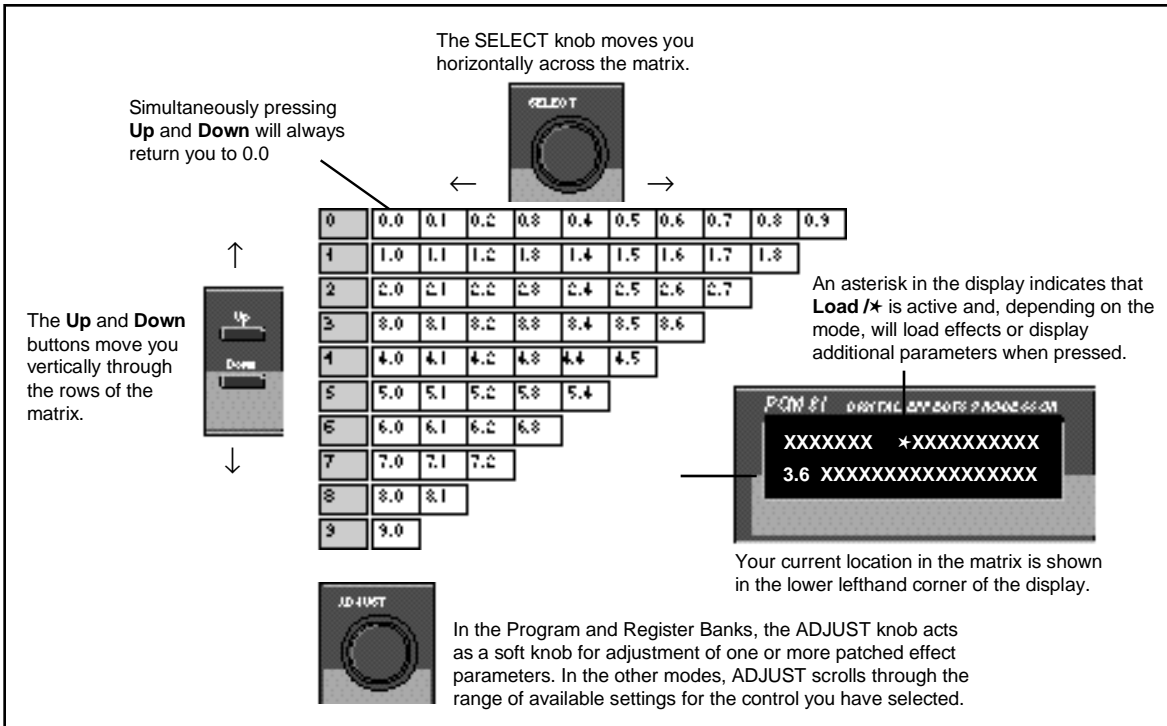


The five mode buttons give you the first level of access to all of the functions and parameters in the PCM 81.

- Press **Program Banks** repeatedly to access six banks of factory preset programs. Each bank contains 50 programs.
- Press **Register Banks** to access a bank of 50 memory locations, called *registers*, where you can store your customized effects. Memory cards can be used for storage of additional banks of registers. When a formatted memory card containing stored registers is inserted, pressing **Register Banks** repeatedly will cycle through all of the available register banks.
- Press **Edit** to access all of the available parameters for the currently running effect.
- Press **Control** to select system parameters, MIDI, card formatting, etc.
- Press **Tempo** to set tempo-related values that affect the delay time and LFO rate parameters of the currently-running effect. This is an exciting feature which is unique to the PCM 81, and which will be described in detail later in this chapter.

### Navigating a Matrix

All of the controls available in a mode are arranged in a matrix of up to 10 columns (numbered 0-9) and 10 rows (each numbered .0-.9). This arrangement allows any one of as many as 100 parameters to be selected simply by using the SELECT knob and the **Up** and **Down** buttons to select a position in the matrix.



**Go or Pro** The PCM 81 offers a choice between two levels of Edit mode parameter access. We call these **Go** mode and **Pro** mode.

**Go** mode makes use of an extra row in the edit mode matrix called the Soft Row, where you can assign as many as 10 effect parameters for easy access. Selecting **Go** mode (Control mode 1.0) limits the action of the **Edit** button to displaying only the Soft Row parameters assigned to the current effect.

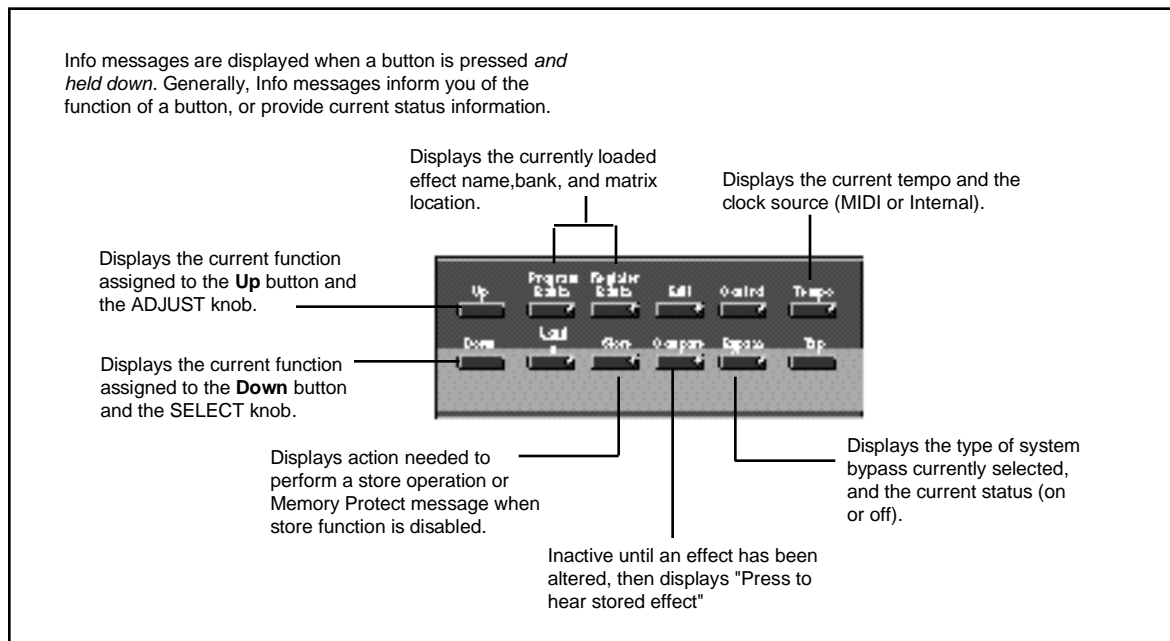
Each preset has a set of Soft Row assignments which we've selected for you (as well as an assignment for the ADJUST knob). When shipped, the PCM 81 will power up in **Go** mode with the first preset (P0 0.0) loaded. Press **Edit** to display the Soft Row of parameters.

**Pro** mode gives you access to the full parameter matrix, including the Soft Row. Use this mode when you want to do in-depth effects editing or patching, or when you want to customize Soft Row assignments.

**Go** mode and **Pro** mode selection is made in Control mode at matrix location 1.0.

The PCM 81 offers an extensive set of informative display messages which can be activated from the front panel. **Info**

The front panel switches perform various functions when pressed. Most of these functions are activated on *release* of the button. If you want to know more about the function of a particular button (without actually executing any action) *press and hold* the button down. While you are holding down the button, an explanatory message will appear on the display. The activation of an Info message overrides the normal function of the button, so that no action is taken on release.



**Control Mode** Selections of various system states and conditions are made in Control Mode. Press **Control** to enter this mode. The Control button LED will light to indicate that the mode is active. Note that Control Mode functions are not available when the **Compare** function is active.

The Control Mode matrix is shown below, followed by descriptions of each available selection.

Simultaneously press **Up** and **Down** to return to 0.0.

0 Audio	0.0 *Word Clock	0.1 Analog In Lvl	0.2 Dig In Lvl	0.3 Word Size	0.4 6-CH6	0.5 Emphasis Bit	0.6 Output Level			
1 System	1.0 Edit Mode	1.1 MultiMode	1.2 Tempo Mode	1.3 Bypass Mode	1.4 Pgm Bypass	1.5 Mem Protect	1.6 Auto Load	1.7 Patch Update	1.8 Initialize	
2 Card	2.0 Bank Copy	2.1 Load	2.2 Format							
3 MIDI	3.0 Reset	3.1 Receive	3.2 Transmit	3.3 *Pgm Change	3.4 Automation	3.5 *Send	3.6 In Clock	3.7 6.5Bit	3.8 Dump	3.9 Dump Speed
4 Setup	4.0 Store	4.1 Load								
Map:	Map 0	Map 1								
Chain	Chain Pgm Assign									
	Chain 0	Chain 1	Chain 2	Chain 3	Chain 4	Chain 5	Chain 6	Chain 7	Chain 8	Chain 9

An asterisk (\*) accompanying a parameter name indicates that there are subparameters available at that matrix location. The **Load/\*** LED will light whenever an asterisk appears in the display. Press **Load/\*** to step to the next subparameter. From any point in the matrix, press **Up** or **Down** together with **Load/\*** to backstep to the previous parameter.

**Row 0  
Audio**

0 Audio	0.0 *Word Clock	0.1 Analog In Lvl	0.2 Dig In Lvl	0.3 Word Size	0.4 6-CH6	0.5 Emphasis Bit	0.6 Output Level			
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**0.0 Word Clock**

The PCM 81 can use its own internal clock as a timing reference, or it can reference an external clock source from the rear panel S/PDIF or AES jacks. Use ADJUST to select **Ext XLR (AES)**, **Ext Coax (S/PDIF)**, **Int: 48kHz** or **Int: 44.1kHz**. When either Internal rate is selected, the digital input is disabled. To process audio from the digital input, you must select **Ext**.

When External clock is selected, and the PCM 81 detects valid digital audio, the rate of the External word clock will be displayed with a label indicating the digital audio format type: **Prf** (Professional [AES]) or **Cns** (Consumer [S/PDIF]).

Audio \*Word Clock  
0.0 Ext: XLR Prf 48

Audio \*Word Clock  
0.0 Ext: Coax Cns 44.1

When the PCM 81 is receiving valid digital audio, selecting **Word Clock** will display the audio format and the rate of the incoming signal.

When External clock is selected, any loss of lock detected in the incoming digital audio, or reception of non-audio data will cause the PCM 81 to immediately mute the digital input, and switch to Internal clock at the sample rate of the last valid external signal. An error message will be displayed if this occurs. The PCM 81 will continuously try to re-establish lock, returning automatically to External clock if and when lock is confirmed.

The following types of errors are detected when the PCM 81 is set to **Ext**:

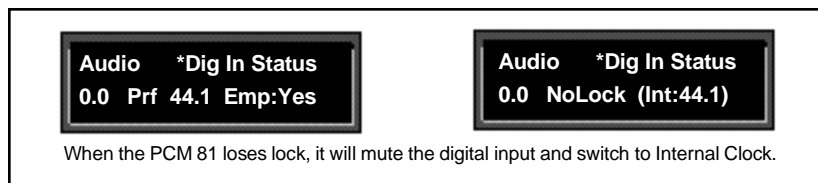
- No Lock: The PCM 81, at some point, lost lock to the incoming digital audio signal. Digital audio input is muted.
- Out of Range: The sampling rate of the incoming audio signal is outside of acceptable tolerance limits of  $\pm 4\%$ . Digital audio input is muted.
- Non Audio: Indicates transmission of non-audio data, such as from a CD ROM. Digital audio input is muted.

### Dig In Status

Pressing **Load/\*** from **Word Clock** will display the current digital input status. This status display is continuously updated, acting as a real-time monitor of the PCM 81 digital input. This display is active even when the PCM 81 is set to Internal clock. Note that in the case of an AES Pro format signal, "Emp:Yes" means either CCITT or 50/15 $\mu$ s emphasis.

If valid digital audio is detected, the display will show the external clock rate and format information, along with the status of the Emphasis bit(s) in the incoming audio signal. If the PCM 81 has lost lock, the display message will indicate "No Lock" and parenthetically show the internal clock rate now in use.

Upon loss of lock, or reception of non-audio data, the PCM 81 will mute the digital input and display the following messages when **Word Clock** or **Dig In Status** is selected:



Word Clock	Dig In Status
Not Locked	No Lock (Int 44.1)
Out of Range	No Lock (Int 44.1)
Non Audio: 44.1	Non Audio: 44.1

### Error Log

The following errors are continuously logged and are available for review by pressing **Load/\*** from the Dig In Status display and using **ADJUST** to scroll through the error list.

- Validity: A Validity error indicates that the Validity bit was set in a frame of incoming data and that the data attached to it may be corrupted. This bit may also be sent when the transmitting device is paused.
- Confidence: The PCM 81 is detecting excessive jitter or noise on the digital audio line. No data has been corrupted, but corrective action should be taken.
- SlipSample: Indicates that a single sample is misaligned with the window defined by the Word Clock. This may occur when an external master changes sample rate, or when it is just powering up, but should not occur in normal operation.
- CRC: Indicates a Cyclic Redundancy Check error in the incoming data.


**Parity, Biphase:** Indicate that at least one bit (and therefore at least one audio sample) was corrupted.

Parity, Biphase, and Confidence errors are most often caused by inappropriate cabling. Be sure to use 75Ω video-grade cable, kept as short as possible — standard audio cable will not work reliably.

Each error is reported by name, with the number of occurrences of that particular type of error. The display might show, for example "CRC: 4752". As many as 9999 instances of each error can be shown. If the number of actual errors exceeds 9999, the display will indicate ">9999". A special symbol (■) before the error type indicates the most recently received error.

Press **Load/★** from the **Dig In Status** display and use ADJUST to scroll through the Error Log.

A typical Error Log display showing that the last error received was a CRC error (■ CRC) and that there have been more than (>) 9999 instances of CRC errors since Word Clock was last set to **Ext**.



To clear the Error Log, reselect **Ext** from the Control Mode **Word Clock** display. This will cause the PCM 81 to attempt to lock to the current external source and will reset the Error Log. The log is also cleared on power up, and whenever it relocks (Auto Lock On).

### 0.1 Analog In Lvl

This is a master level control for analog left and right inputs. Use ADJUST to select values from 0-100%. If using only analog inputs, this should be 100% for best audio performance. Values between 0% and 100% are for mixing analog and digital sources.

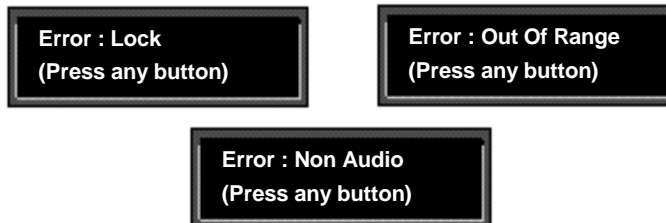
Note: If the analog signal being fed into the PCM 81 is too hot, turn it down by adjusting the front panel INPUT level control, or by changing the rear panel Input Gain switch. (See *Setting Audio Levels* in Chapter 1.)

### 0.2 Dig In Lvl

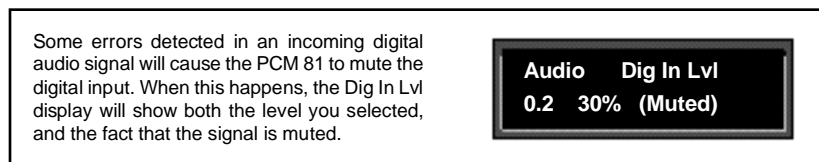
This is a master level control for the selected digital left and right input connector. Use ADJUST to select values from 0-100%. If using only digital inputs, this should be 100%, with **Analog InLvl** set to 0% for best audio performance. Values between 0% and 100% are for mixing analog and digital sources.

Note: Only one digital input connector can be active: XLR (AES/EBU) or Coax (S/PDIF).

If digital audio is interrupted by lock or range errors, or by transmission of non-audio data, the digital input will be muted. Whenever this occurs, the PCM 81 remains functional, while an error message, such as those shown below, is displayed.



Any such message will remain on the display until you dismiss it by pressing any front panel button. If the condition causing the error is corrected, and the digital input is unmuted, the message will still remain until you dismiss it to ensure that you are informed of a condition that has muted digital audio, even if only for a brief period of time. Once the message is dismissed, the display will return to its previous state.



If you select the **Dig In Lvl** parameter while there is a lock, range, or non-audio error, the display will continue to show the level you have selected, and parenthetically show that the signal has been muted.

Note, if Word Clock is set to **Int**, Dig In Lvl is disabled. You must select **Ext** to process digital audio.

### 0.3 WordSize

The WordSize control allows you to dither the PCM 81 24-bit word size to match the device receiving digital audio from the PCM 81. If you are recording to the digital input of a DAT machine, select **16 bits**. If you have a digital console with a 24-bit bus, select **24 bits**.

If you are using the analog outputs, select **20 bits** (the default setting).

### 0.4 SCMS

Digital audio signals, in order to comply with copyright standards, are encoded with control information which can limit the ability to copy audio data. This control information is generally known as SCMS (Serial Copy Management System). Under this system, you can choose to have the audio material processed by the PCM 81 encoded to allow one of three levels of copy restriction. To make your selection, use ADJUST to select **No Copy**, **One Copy**, or **Multi Copy**.

### 0.5 Emphasis Bit

The Emphasis control allows you to explicitly set the emphasis "flag" in the digital audio, or to pass along the incoming signal without changing its emphasis coding. (The PCM 81 does not perform any emphasis or de-emphasis as part of its signal processing. The AES input/output path will correctly pass either 15/50 or CCITT emphasis.) The choices available with ADJUST are: **Yes**, **No**, and **Pass Thru**.

### 0.6 Output Level

This control allows you to select the maximum output level at the PCM 81's analog outputs. Use ADJUST to select **+4 dBu**, or **-10 dBu**.

**Exercise care when switching this control, as a 14dB level change instantly occurs when going from -10dBu to +4dBu.**

## Row 1 System

1	1.0	1.1	1.2	1.3	1.4	1.5	1.6	1.7	1.8
System	Edit Mode	Mix Mode	Tempo Mode	E_pass Mode	Pgm E_pass	Mem Protect	Auto Load	Patch Update	Initialize

### 1.0 Edit Mode

The PCM 81 has been designed with a "plug and play" feature called **Go** mode. In this mode, the most useful parameters of each effect are grouped together in a single row which is available whenever you press **Edit**.

Each PCM 81 preset has a set of **Go** mode parameters which we've selected for you. When shipped, the PCM 81 will power up in **Go** mode, with the first preset (**P0 0.0**) loaded. Press **Edit** to display the first available parameter in the Soft Row.

If you want access to the full parameter matrix for any effect, including the Soft Row parameters, use ADJUST to select **Pro** mode. Now, when **Edit** is pressed, you can select any parameter for adjustment, and customize any effect with your own Soft Row assignments. For more information about the Soft Row, see *Editing an Effect* later in this chapter.

### 1.1 Mix Mode

Each PCM 81 effect has its own Mix parameter, with the Mix setting stored as an integral part of the effect. Mix Mode allows you to override these individual Mix settings and set a global Mix value for all effects. This is useful when using a mixing console's controls to set the amount of wet signal in a mix. In such a case, you can use this control to set all PCM 81 effects to 100% wet.

When shipped, the PCM 81 has the Mix Mode set to **Pgm**. This setting determines that effects will be loaded with their stored Mix settings, and allows the individual Mix controls in the edit matrix of each effect to be adjusted from 0-100% Wet. To set a global Mix value, use ADJUST to select **Global**, press **Load/★** to display the current value, and use ADJUST to assign any value from 0-100% Wet.

## 1.2 Tempo Mode

The PCM 81 gives you an exciting new approach to working with delay times and modulation parameters. Now you can set these parameters in *beats*, allowing you to control your effects in a completely musical way. Each PCM 81 effect has its own Tempo parameters, with tempo settings stored as an integral part of the effect. These include: Tempo Rate, Tempo Beat, Tempo Source (internal or MIDI), Tap Duration, and Tap Average. The **Global** setting here allows you to override individual Tempo Rate settings with a global value which can then be changed on the fly.

When shipped, the PCM 81 has the Tempo Mode set to **Pgm**, with each effect driven by its own stored tempo rate. To change to a global Tempo Rate, use ADJUST to select **Global**, press **Load/★** to display the current tempo in BPM (beats per minute), then use ADJUST to assign a global tempo value of 40-400BPM.

Whether Tempo Mode is set to **Global** or **Pgm**, you can set a new tempo rate by pressing the front panel **Tap** button twice. Alternatively, you can choose to have tempo set automatically from incoming MIDI clock. The rate you tap, or the MIDI tempo, will be displayed here.

For more information about working with the tempo parameters, see *Tempo Mode* later in this chapter.

## 1.3 Bypass Mode/Bypass Src

This control allows you to determine the behavior of the PCM 81 when the front panel **Bypass** button is pressed. You can also assign an external controller to perform identically to the front panel button. When the **Bypass** button is pressed, the LED will light, and a message indicating bypass type will be displayed. Pressing **Bypass** again will turn bypass off.

The choices available via ADJUST are:

- AllMute:** Mutes both the input and the output signal, giving complete silence.
- InputMute:** Mutes the input to the PCM 81, allowing the tail of the effect to ring out. (This is the default setting.)
- OutputMute:** Mutes the output. Audio signals are still being fed to the PCM 81, so processed audio returns immediately when **Bypass** is turned off.
- Bypass:** Completely bypasses the PCM 81, passing unprocessed audio directly through to the outputs.

To assign an external controller to perform the selected bypass function, press **Load/★** to display **Bypass Src**. Use ADJUST to select a footswitch or any MIDI controller (or Off). Once a source is selected, it will perform the same function assigned to the front panel **Bypass** button.

## 1.4 Pgm Bypass

This control allows you to determine the behavior of the PCM 81 when a new effect is loaded. The choices available are: **AllMute** or **Bypass**.

### 1.5 Mem Protect

The PCM 81 provides a memory protection feature to prevent accidental overwriting of your stored effects. When this control is set to **On**, pressing the front panel **Store** button will cause an error message to be displayed. To enable the Store function, turn ADJUST to select **Off**.

### 1.6 Auto Load

This control allows you to choose whether PCM 81 effects will be loaded immediately when selected with SELECT and the **Up** and **Down** buttons (**On**), or whether they will require a press of the **Load/★** button (**Off**).

### 1.7 Patch Update

When a controller is patched to an effect parameter, this control determines when the controller will take control of the parameter. If **Immediate** is selected, stored parameter values will jump to the current controller position when the effect is loaded. If **Delayed** is selected, the stored parameter value will remain in effect until the controller is moved. See *Patching* later in this chapter.

### 1.8 Initialize

Selecting this control arms the PCM 81 to revert to its factory settings.

**This will erase all registers and setups,  
and return the PCM 81 to its default states.**

If you press **Store**, the display will ask "Are you sure?" (Press STORE). If you don't want to reinitialize your unit, press any button to cancel the operation. If you press **Store** in response to this message, the display will flash "Restoring original factory settings" and your unit will be reinitialized.

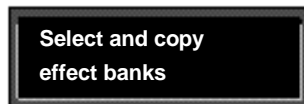
★	☰	☰	☰
Card	Bank Copy	Load	Format

**Row 2  
Card**

### 2.0 Bank Copy

This control allows you to copy banks of effects from one location to another. Banks can be copied internally, or to and from PCMCIA Memory Cards. Try, for example, copying Preset Bank 0 into the internal Register Bank.

1. Press **Store**. The following display will appear briefly.



The display will then change to show:



The asterisk indicates that **Src** is available for adjustment. ADJUST will scroll through all available banks, including internal preset and register banks, as well as any banks on inserted PCMCIA cards. Internal banks are labeled "P0...P3" and "R1". Card registers will be labeled "C0, C1, C2" etc.

2. Press **Load/★** to move the asterisk to **Dst**.



Now, use ADJUST to select the destination of your copy. **Selecting a register bank here will cause its contents to be erased and overwritten with the bank you have selected as the source when Store is pressed.**

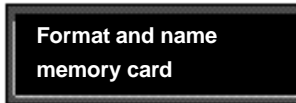
4. Press **Store** to copy the selected source (in this case Preset Bank 0) into the internal Register Bank. The display will ask "Are you sure?" (Press STORE). Press any button to cancel. Press **Store** to complete the store operation.

### 2.1 Load

This control allows you to load audio software from a Memory Card simply by inserting the new card and responding to the display prompts. The PCM 81 can load all PCM 80 algorithm cards. It can also share registers, maps, and chains with the PCM 80 via RAM cards. (Sharing Setups between the PCM 80 and the PCM 81 is *not* recommended.)

### 2.2 Format

This control allows you to format a Memory Card for PCM 81 use. Press **Store** and insert an unformatted card (or one you don't mind erasing). Make sure the Write Protect latch on the card is set to Off. Press **Store**. The display will ask "Are you sure?" (Press STORE). Press **Store**. The following display will appear briefly.



The display will then change as shown below.



This display allows you to assign a name (of 10 characters or less) to the card. A blinking cursor indicates that a particular character is available for changing. Use ADJUST to select the character you want in that position. Turn SELECT to move the cursor to another character. Press **Store** to execute. The display will ask "Are you sure?" (Press STORE). Press **Store** again to complete the operation. Press any front panel button to cancel.

### Row 3 MIDI

8 MIDI	8.0 Reset	8.1 Receive	8.2 Transmit	8.3 Pgm Change	8.4 Automation	8.5 Send	8.6 In Clock	8.7 Edit	8.8 Dump	8.9 Dump Speed
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### 3.0 Reset

This control resets all patched parameters to their previously stored values. When this control is activated, a MIDI "Reset All Controllers" message is also transmitted on the current channel by the PCM 81.

### 3.1 Receive

Turn ADJUST to select **OFF**, **1-16**, or **OMNI** for receipt of MIDI messages.

### 3.2 Transmit

Turn ADJUST to select **OFF**, or **1-16** for transmission of MIDI messages.

### 3.3 Pgm Change

This control specifies the PCM 81's response to incoming MIDI Program Change messages. There are four selections available via ADJUST: **On**, **Off**, **Map** and **Chain**.

**Pgm+** and **Pgm -**, are available as subparameters in each location. **Pgm+** will load the next higher effect in the current bank, map, or chain. **Pgm -** will load the next lower effect. You can select the following sources to activate **Pgm+** and **Pgm -**:

Off  
 Footswitch 1  
 Footswitch 2  
 •  
 •  
 •  
 MIDI Controller #119

### On

Program Change messages 0—49 correspond to PCM 81 Effects 0.0 —4.9 in the current bank. Program Change messages 50—127 are ignored. The current bank can be changed with MIDI Bank Select Messages as follows:

- 0–5: Program Banks 0–5
- 6: Internal Register Bank
- 7–11: reserved
- 12–58: Memory Card Banks. The number of banks available on a given card will vary with its size, as follows:

Card Size	# Banks
64	2
256	11
512	23
1 Meg	47

### Off

All Program Change and Bank select messages are ignored.

### Map

Program Change 0-127 can be mapped to any PCM 81 effect in any internal or card bank. Two 128 element maps are stored internally, additional maps may be stored on RAM cards. Once you have selected **Map**, press **Load/\*** to display:

```
MIDI Pgm Change
3.3 *MapSelect 0
```

Turn ADJUST to select the desired Program Change Map.

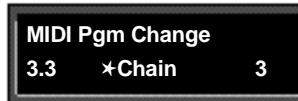
### Chain

Any Program Change number can be selected to load any one of ten customized effect "chains". Additional chains can be stored on RAM cards. Once a chain is loaded, effects in the chain are accessed by the controller patched to **Pgm +** and **Pgm -** (program increment and program decrement). Once you have selected **Chain**, press **Load/\*** to display:

```
MIDI Pgm Change
3.3 *Chain MIDI
```

Turn ADJUST to select the desired Program Chain. When set to **MIDI**, Program Chains will be loaded by MIDI Program Change messages according to the settings of Chain Pgm Assign in Row 6.

To load a specific Program Chain, without sending the PCM 81 a MIDI Program Change message, use ADJUST to display the desired chain number.



### 3.4 Automation

This control is provided for communication with one or more additional PCM 80s or PCM 81s. Select **On** to have values resulting from front panel operations sent out as SysEx messages. Press **Load/★** to select the ID (**0-126** or **All**) of the target PCM 81(s).

Automation sends all MIDI commands in PCM 80 format, which is also recognized by the PCM 81. As the PCM 80 has only four Banks (0-3), it cannot load programs from PCM 81 Banks 4 and 5 in Automation mode.

### 3.5 Send (Foot Pedal, Foot Sw 1, Foot Sw 2, ADJUST)

If **Transmit** is set to **On** these four controllers can transmit MIDI controller messages. Press **Load/★** to cycle through the controller selections. Use ADJUST to select the MIDI Controller message to transmit.

### 3.6 Int Clock

You can choose to have the PCM 81 transmit MIDI Clock at the current tempo rate by setting this control to **On** and Tempo mode **Source** (0.2) to **Internal**. If this control is set to **Off**, MIDI Clock will not be transmitted.

### 3.7 SysEx

This control is provided for communication with one or more additional PCM 81s or computer editor software. **On** (the default setting) allows SysEx messages to be received by the PCM 81. Press **Load/★** to select device ID (**0-126**).

### 3.8 Dump

Press **Store** to configure the PCM 81 to execute MIDI dumps of single effects, banks, maps, chains, or setups.

With the exception of Setup Dumps, all dumps are transmitted in PCM 80 format, which is also recognized by the PCM 81. You can dump from the front panel of either unit to the other. As the PCM 80 has only four Banks (0-3), references to Banks 4 and 5 will be ignored when transmitted to a PCM 80.

### 3.9 Dump Speed

Turn ADJUST to select dump speeds of **Slow**, **Medium** or **Fast** to achieve compatibility with the connected MIDI device.

4 Setup	4.0 Store	4.1 Load
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## Row 4 Setup

### 4.0 Store

Control mode Audio, System, and MIDI parameter settings, along with two settings from the Tempo matrix, comprise a "Setup." Five setups can be stored in the unit, or on a Memory Card, allowing you to transport not only your effects, but complete PCM 81 environments to another PCM 81. Press **Store** to initiate the Setup Store function.

When the PCM 81 is shipped (or when you reinitialize the unit) default values are assigned to these parameters. The following table shows the Setup parameters along with the factory default setting of each parameter.

Control Mode Matrix Location		System Parameter	Default Setting
<b>Audio</b>	0.0	Word Clock	Internal 48kHz
	0.1	Analog Lvl	100%
	0.2	Digital Lvl	0%
	0.3	Word Size	20 bits
	0.4	SCMS	Multi Copy
	0.5	Emphasis Bit	Pass Thru
	0.6	Output Level	+4dBu
<b>System</b>	1.0	Edit Mode	Go
	1.1	Mix Mode	Pgm
		Global Mix Value	100% Wet
	1.2	Tempo Mode	Pgm
		Global Tempo Value	120 BPM
	1.3	Bypass Mode	InputMute
		Bypass Src	Off
	1.4	Pgm Bypass	AllMute
	1.5	Mem Protect	On
1.6	Auto Lock	Off	
1.7	Patch Update	Delayed	
<b>MIDI</b>	3.1	Receive	OMNI
	3.2	Transmit	1
	3.3	Pgm Change	On
		Pgm+	Off
		Pgm-	Off
		Map select	0
	3.4	Chain	MIDI
		Automation	Off
	3.5	Footpedal	None
		Sw 1	None
		Sw 2	None
		ADJUST	None
	3.6	Int Clock	Off
3.7	SysEx	On	
	Device ID	0	
3.9	Dump Speed	Slow	
Tempo Mode Matrix Location		System Parameter	Default Setting
<b>Tempo</b>	0.2	Source	Internal
	<b>Tap</b>	1.3	Display

### 4.1 Load

This control allows you to load your stored setup, restore the factory default setup shown above, or load a setup from a Memory Card.

**Row 5**  
**Mapx**

Map:	Map 0	Map 1
------	-------	-------

**Map 0 and Map 1**

When Control mode 3.3 is set to **Map**, received MIDI Program Change messages will be mapped according to the selections made here. The selections available are: **MIDI Program Change #** (0-127), **Bank #** (PCM 81 preset, register, or card bank) and **Pgm #** (PCM 81 effect number 0-49). When shipped, the PCM 81 has the two internal maps configured to access all presets and registers:

<b>Map 0</b>	<b>Map 1</b>
MIDI 0 = P0 0.0	MIDI 0 = P2 2.8
MIDI 127 = P2 2.7	MIDI 121 = R 4.9

**Row 6**  
**Chain**

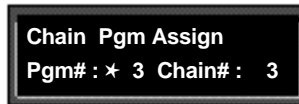
Chain	Chain Pgm Assign									
	Chain 0	Chain 1	Chain 2	Chain 3	Chain 4	Chain 5	Chain 6	Chain 7	Chain 8	Chain 9

The PCM 81 has 10 internal program chains, numbered 0-9. (Ten additional chains are available if a register card is inserted.) Each chain is made up of 10 "links" (numbered 0-9). You can assign any program or register to any link in the chain.

A chain can be loaded with a MIDI Program Change message, or by selecting its number directly at Control mode 3.3. Once a chain is loaded, the source assigned to **Pgm+** and **Pgm-** will load the next higher or lower program in the chain.

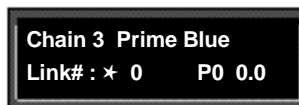
**Pgm Assign**

When Control mode 3.3 is set to **Chain**, received MIDI Program Change messages will be mapped according to the selections made here. Two assignments are available: **MIDI Program Change #** (0-127) and **Chain #** (0-9). When the PCM 81 is shipped, all program numbers are mapped to Chain #0. To change assignments, set **Pgm#**, with ADJUST, press **Load/★** to move the **★** to **Chain#** and set it with ADJUST. (You cannot assign more than one chain/MIDI #.)



**Chain 0-9**

Use SELECT to choose a chain. Once a chain is selected, any PCM 81 program or register can be assigned to any link in the chain. Press **Load/★** to move the **★** from **Link#** to the Bank ID, to the program number. Depending on the field marked with the **★**, ADJUST will select link numbers 0-9, Banks (Pn or Rn), or the desired program within the displayed bank.



The PCM 81 has 300 factory-designed programs, organized into six Program Banks of 50 each, and 50 memory locations, called registers, for storing your customized effects. One Register Bank is available in the PCM 81 itself. Additional Register Banks can be stored on PCMCIA cards.

## Program and Register Banks

The PCM 81 has 6 Program Banks, each with 50 factory-designed programs

All PCM 81 programs have one or more parameters patched to the ADJUST knob. In Program Banks or Register Banks mode, turn ADJUST to change the sound of the currently running effect. While ADJUST is turned, the patch name and value are displayed.

A Register Bank provides 50 memory locations for storing your own effects.

R	0.0	0.1	0.2	0.3	0.4	0.5	0.6	0.7	0.8	0.9
	1.0	1.1	1.2	1.3	1.4	1.5	1.6	1.7	1.8	1.9
	2.0	2.1	2.2	2.3	2.4	2.5	2.6	2.7	2.8	2.9
	3.0	3.1	3.2	3.3	3.4	3.5	3.6	3.7	3.8	3.9
	4.0	4.1	4.2	4.3	4.4	4.5	4.6	4.7	4.8	4.9

The procedure for loading effects is the same, whether you are choosing from banks of factory presets, or from your own banks of registers. When first turned on, the PCM 81 will load whatever effect was running when it was last turned off. When shipped from the factory, the first effect in the first Program Bank (**P0 0.0 Prime Blue**) is loaded. The Bank ID (**P0, P1, P2, P3, P4, P5**), matrix location, program name, and the algorithm from which the effect is derived are all displayed.

## Selecting Effects

Indicates that the ADJUST knob is patched to one or more parameters in the currently-running effect — in this case, Prime Blue.

Turn SELECT to display another effect.

The patch indicator remains to indicate that the currently running effect has an ADJUST knob patch, although the displayed effect may not.

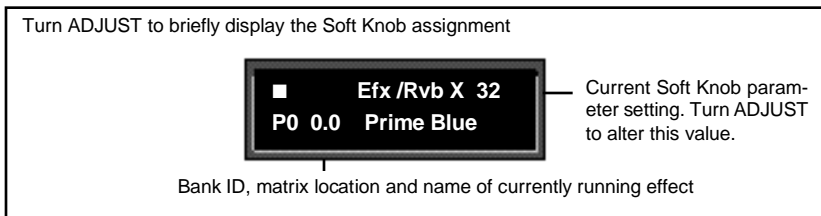
An asterisk in the display indicates that the displayed effect is not loaded.

Press **Program Banks** to cycle through the program banks. Simultaneously press **Program Banks** along with either **Up** or **Down** to backstep.

Turn SELECT to scroll through all 300 effects. Press **Program Banks** to reselect the last displayed effect in another bank. Press **Load/\*** to load the displayed effect.

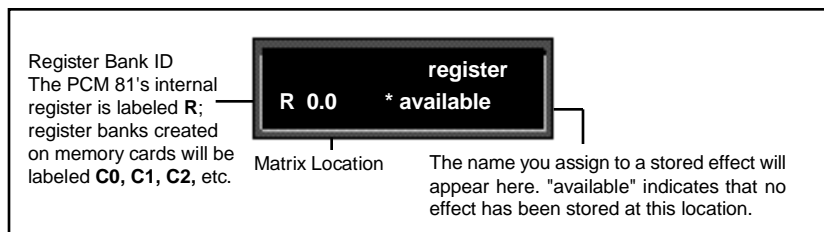
In the Program and Register Banks, ADJUST is a Soft Knob. Each of the factory-designed programs has one or more parameters patched to this knob, providing a quick way to make useful changes to the effect. For example, in **P0 0.0 Prime Blue**, ADJUST varies the mix of Chorus, Echo and Reverb effects.

The ■ symbol in the upper left corner of the display indicates that the currently running effect has an ADJUST knob patch. (When you display another program or register, this indicator continues to be displayed.) Turning ADJUST will cause the display to change to show the patch name and the current value, as well as the Bank, matrix location and name of the currently running effect. Continue turning ADJUST to alter the patched parameter's value. You can create your own ADJUST knob parameter assignments for your registers. (See *Editing*, later in this chapter.)



Note that scrolling through the effects in the Program or Register banks will not load the effects, but will merely display them (unless you have specifically turned on the Auto Load function in Control Mode). Displayed effect names will be preceded by an ★ indicating that they can be loaded by pressing **Load/★**. (The **Load / ★** LED will light to indicate that the load function is available.) To find out at any time what the currently running effect is, press and hold **Program Banks** or **Register Banks**. An Info message will be displayed providing the name bank and matrix location of the currently running effect.

To select an effect stored in a register, press **Register Banks**. If you have registers stored on a memory card, and have the card inserted, pressing **Register Banks** repeatedly will cycle through all of your stored bank selections. Simultaneously press **Register Banks** along with either **Up** or **Down** to backstep. Turn **SELECT** to scroll through all of the effects in the bank. As in Program Banks mode, an asterisk in front of the effect name indicates that the displayed effect is not loaded. Press **Register Banks** to reselect the last displayed effect in another bank. Unused registers are indicated by the message "available" on the display. Press **Load/★** to load the displayed effect.



The organization of programs in the six program banks and descriptions of the 300 preset programs are given in Chapter 3.

## Tempo Mode

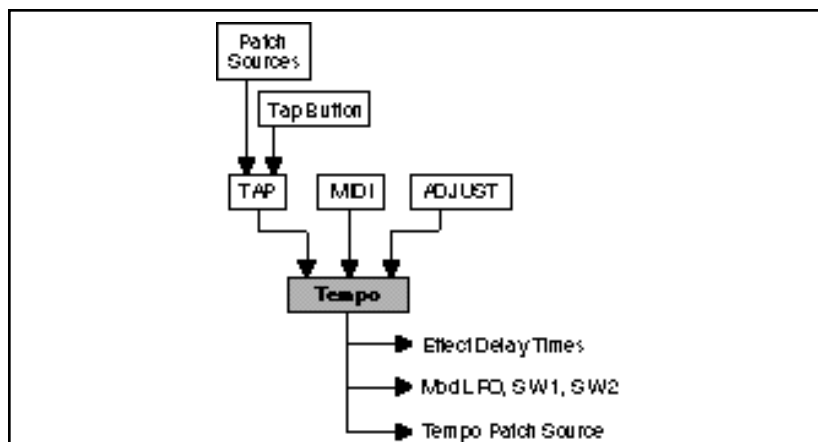
The PCM 81 gives you unique control over tempo. In the PCM 81, tempo is not just a matter of setting echo rates. Any delay parameter (as many as 10 in some effects) and any time-based modulator (LFO, Sw1 and Sw2) can be individually assigned to an absolute time value, or assigned to a tempo value.

For example, a delay time can be set to a specific number of milliseconds, and you will always get a delay of that number of milliseconds, regardless of tempo changes. Alternatively, a delay time can be set to a specific ratio of echoes to beats. Now, if you create a rhythmic echo pattern, delay times will be linked to tempo. When you change tempo, the delay time will change to maintain the same rhythm at the new tempo.

With the LFO and time-based switches, the rate of change can be an absolute value (such as once per second), or it can be linked to tempo (for example, once every four beats). Any delay parameter, or time-based modulator, can be set to its own individual rhythm, allowing you to set up an effect which will change in a rhythmically interesting way — evolving over time, for example, as opposed to being a mere series of repetitions. Once delays and time-based modulators are assigned, tempo rate can be easily changed in a variety of useful ways.

Tempo **Rate** can be set internally or via MIDI. If Tempo **Source** is set to **Internal**, you can dial in any tempo from 40-400 BPM at location 0.0 in the Tempo mode matrix. Alternatively, you can press the front panel **Tap** button twice in rhythm to establish the rate you want, or you can have the value of a patch source act as a tap trigger. (See *Patching*.) The Tap function, whether performed by the front panel **Tap** button, or by an assigned controller, is always active, allowing you to change tempo on the fly from any mode. You can also choose to have your tempo transmitted as a MIDI Clock signal to control the tempo of connected MIDI devices. (Control mode 3.6) If **Source** is set to MIDI Clock, PCM 81 tempo will sync to incoming MIDI Clock. Whether tempo is set internally or via MIDI, the LED in the Tempo button will flash at the current rate. (You can disable the Tempo LED flashing under Tempo mode 1.3.) Press and hold the **Tempo** button at any time to display the current Tempo **Rate** and **Source**.

Each effect in the PCM 81 has its own tempo rate setting which is stored with the effect. You can override these individual tempo rates with a global tempo rate at Control Mode 1.2. Tempo is also available as an independent patch source which can control any effect parameter. (See *Patching* later in this chapter.)



When **Compare** is on, **Tap** will not alter the tempo, but will still reset the LFO's. See Chapter 3 *Modulation*.

**The Tempo Mode Matrix** Press **Tempo** to access the following tempo parameters:

Simultaneously press **Up** and **Down** to return to 0.0.

0	0.0	0.1	0.2	
Tempo	Rate	Beat Value	Source	
1	1.0	1.1	1.2	1.3
Tap	Duration	Source	Average	Display

An asterisk (\*) accompanying a parameter name indicates that there are subparameters available at that matrix location. The **Load/\*** LED will light whenever an asterisk appears in the display. Press **Load/\*** to step to the next subparameter.

From any point in the matrix, press **Up** or **Down** together with **Load/\*** to backstep to the previous parameter.

**Row 0  
Tempo**

0	0.0	0.1	0.2
Tempo	Rate	Beat Value	Source

**0.0 Rate**

This is the current tempo (in Beats Per Minute). When 0.2 is set to **Internal**, turning ADJUST allows you to select a different rate (40-400 BPM). The Tempo LED will flash at the new rate. Note that fractional tempos can be tapped in, but are not available via ADJUST. The display will always show the nearest whole number value.

**0.1 BeatValue**

Tempo is expressed in BPM. This control allows you to specify the value of the beat. Eighth, dotted-eighth, quarter, dotted-quarter, half, dotted-half, and whole-note values are available. If, for example, the rate is 120 BPM, and you select eighth-note here, the tempo will be 120 eighth-notes per minute. If you select quarter-note here, the tempo will be 120 quarter-notes per minute. (The factory default is quarter-note.)

**0.2 Source**

You can choose to have tempo determined by the PCM 81 **Tap** and **Rate** controls (Internal), or by MIDI Clock. When MIDI Clock is selected as the tempo source, **Tap** acts as a reset, setting the downbeat of the LFO and the time-based switches.

1	1.0	1.1	1.2	1.3
Tap	Duration	Source	Average	Display

**Row 1  
Tap**

### 1.0 Tap Duration

This control determines how many beats will occur in a tap interval. The default setting (1 beat) is probably adequate for most applications. With the default setting of 1 beat, if the tempo is set to 120 bpm, and the beat value is set to quarter-note, each TAP = 1 quarter-note = 1 beat. Available values are: 1/8, 1/7, 1/6, 1/5, 1/4, 1/3, 1/2, 1-8 beats

### 1.1 Tap Source and Tap Level

Press **Load**/**\*** to toggle between these two controls.

**Tap Source** allows you to assign the Tap function to any of the PCM 81's Internal, MIDI, or MIDI controllers as listed under Patching.

**Tap Level** allows you to set the level at which the Tap function is triggered.

### 1.2 Average

This control allows you to average the last 2-8 taps. Higher numbers mean that the response to incoming taps will be more gradual. (The tempo is updated on every tap, but with a value which is the average of the last 2-8 taps.) Higher average values are more useful if you're trying to lock into a pre-recorded track.

### 1.3 Display

This control allows you to disable the flashing of the Tempo LED. Turn ADJUST to select **Off**. The Tempo LED will turn off.

### Editing an Effect

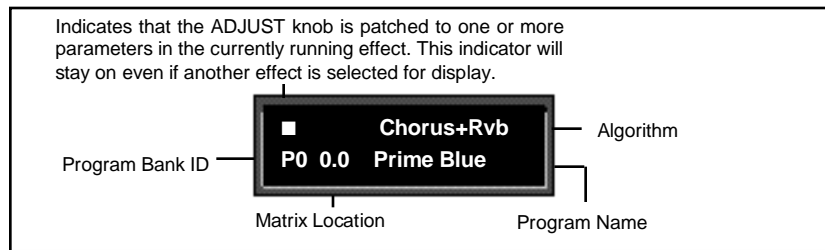
With 17 algorithms and 300 preset effects, the PCM 81 gives you a lot to play with right out of the box. An enormous range of editing control is provided for each algorithm, with parameters organized in an edit matrix of as many as 100 main controls. In addition to providing this powerful sound design capability, the PCM 81 is also designed to allow you to customize these controls for your day-to-day editing needs, or to simply use a subset of controls specially designed for each preset.

This section will describe three basic levels of editing, from the simplest "plug and play" method, through the full edit matrix.

### The Soft Knob

Every preset in the PCM 81 has one or more of its available parameters patched to the front panel ADJUST knob. This Soft Knob provides the first level of editing control. Once you have loaded a preset, simply turn ADJUST to alter the effect.

When shipped, the PCM 81 will power up with the first preset (**P0 0.0 Prime Blue**) loaded. The following display will appear:

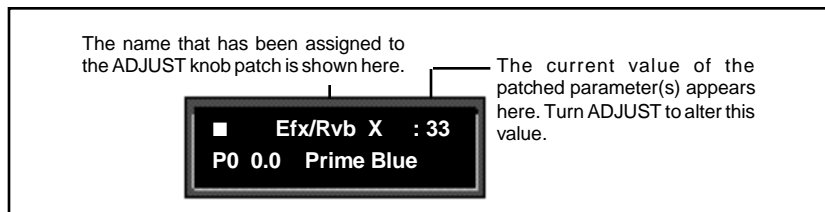


Turn ADJUST. The display will change to show the name assigned to the ADJUST patch, and the current value of the patch. Continue turning ADJUST to change the value of the patch along its entire range. The range of ADJUST knob control can be limited in Control Mode, making it possible, for example, to have a range of only 0...1, in order to have the ADJUST knob behave as an Off/On control. Many of the presets have range limits to make them more convenient to use. In some of the presets, the changes effected by ADJUST will be as simple as altering the wet/dry mix. In others, such as the example shown below, turning ADJUST will glide you all the way from a chorusing effect to a reverb effect over the complete control range of 0-127.

#### Controlling the Soft Knob with a Foot Pedal

If you have a foot pedal connected to the PCM rear panel Foot Controller jack, you can use it to control the soft knob patch. (Note that no MIDI connections are required to do this.)

To assign a foot pedal control over the soft knob patches, set both Control mode 3.5 **ADJUST** and Control mode 3.5 **Foot Pedal** to the same MIDI Controller. See Chapter 5 *MIDI Operation*.



Details on how to create your own ADJUST knob patches are given at the end of this chapter under *Patching*.

Beyond simple ADJUST knob editing, the PCM 81 offers two levels of editing control, called **Go** mode and **Pro** mode. **Go** mode is designed to be a basic "plug and play" mode, with easy access to a specific set of preset parameters. For each of the 300 presets, we have designed a Soft Row containing those parameters which allow you to make value changes to the effect *without* losing the character of the sound.

When shipped, the PCM 81 will power up in **Go** mode with the first preset (**P0 0.0 Prime Blue**) loaded. Press **Edit** to display the Soft Row of parameters which have been designed for this preset. In this example, 10 soft row parameters have been assigned. The name of each Soft Row parameter is displayed, along with a reference to its row in the Edit matrix.

Turn **SELECT** to scroll all of the available parameters in the Soft Row. Turn **ADJUST** to change the value of any displayed parameter along its entire range.

Each preset has a Soft Row of parameters which have been specifically selected to provide everything you need to play with the effect. This example shows the Soft Row for **P0 0.0 Prime Blue**.

An asterisk in front of a displayed parameter indicates additional parameters are available when you press **Load/\***.

Controls Mtr	Controls High Cut	Controls FX Width	Rub Time Mid R	Rub Time R HC	Rub Time Pre Delay	Rub Design Size	Chorus Mod Depth	Mod LFO Rate	Mod Sync Rate
0	1	2	3	4	5	6	7	8	9
0	1	2	3	4	5	6	7	8	9
1	2	3	4	5	6	7	8	9	
2	3	4	5	6	7	8	9		
3	4	5	6	7	8	9			
4	5	6	7	8	9				
5	6	7	8	9					
6	7	8	9						
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Turn **SELECT** to move horizontally across the Soft Row.

Use **ADJUST** to change the value of a displayed parameter.

The Soft Row display shows:  
 an Edit matrix row label...  
 ... the name of the parameter...  
 ... and the current parameter value.

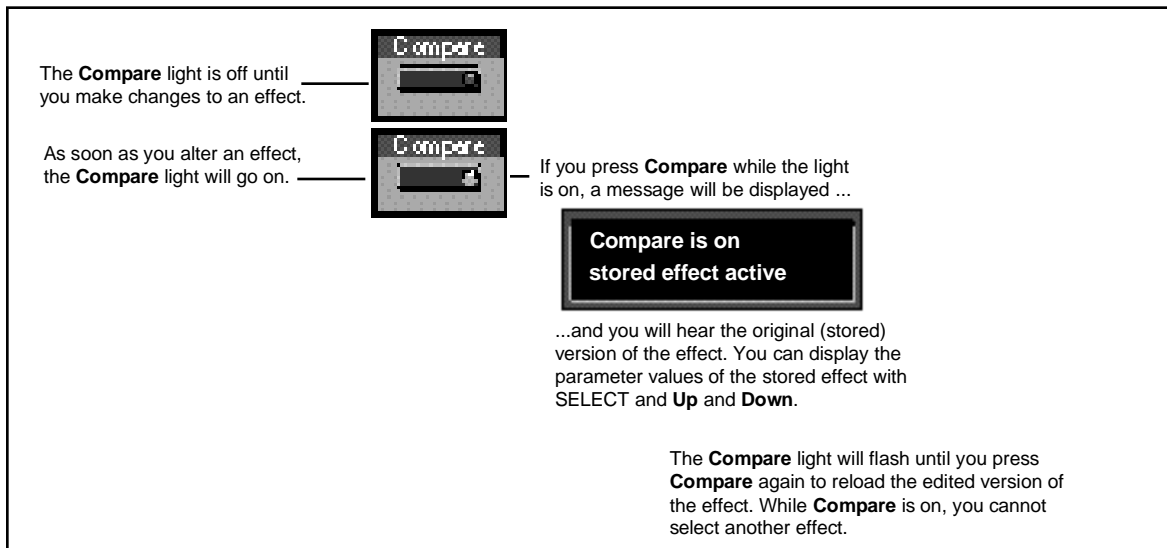
The symbol ■ in the upper left corner of the display indicates that a modulator is patched as a "source" controller.

The symbol ■ in the upper right corner of the display indicates that a parameter is a patch "destination".

**Compare** Whenever you edit a PCM 81 effect from the front panel, the LED in the **Compare** button will light. This lets you know that the effect has been altered since the last store operation, and that the edit compare function is active. Whenever this light is on, you can press **Compare** to hear the original version of the effect. A message will be displayed to inform you that the stored version of the effect is being loaded. Although this message is only displayed briefly, the **Compare** LED will flash to let you know the effect you are hearing is the stored version. While **Compare** is on, you can use the SELECT knob and the **Up** and **Down** buttons to view parameter values in the stored effect.

Press **Compare** again to reload your edited version. Another message will be displayed, and the **Compare** LED will stop flashing and remain lit until you store your edited version, or select and load another effect.

Note: Altering parameter values with patched sources other than ADJUST will *not* light the LED.



**Bypass** The front panel **Bypass** button is always active, and will turn on the type of bypass (**AllMute**, **InputMute**, **OutputMute**, or **Bypass**) selected in Control Mode 1.3.

When you press **Bypass**, the button LED will light and a message will be displayed to inform you that bypass is on. The display message will also indicate the type of bypass which is in effect. Press **Bypass** again to turn both the LED and the bypass condition off. A brief display message will inform you that bypass is off.

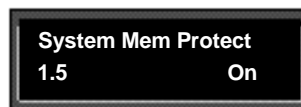
Press and hold **Bypass** to display the current state and type of bypass.

## Store Operations

The PCM 81 is shipped with its Memory Protection option on. When you press the front panel **Store** button, the following message will be displayed:



To turn Memory Protection off, press **Control** and use **Up** and **Down** and **SELECT** to display matrix location 1.5. The display should read:



Turn **ADJUST** to select **Off**.

### Turning Memory Protection Off

With Memory Protection disabled, pressing **Store** will cause a message like the following one to be displayed:

An asterisk (★) is positioned in front of the display label which is available for changing — in this case, the name of the effect.

The default Bank and register location will appear here.

"available" means that this location is empty. If an effect is stored at this location, its name will appear here.

Press **Load/★** to select the **ADJUST** patch for renaming. Press again to select the register bank. Press again to select the register location within the selected bank.

### Storing an Effect

If you press **Store** again, the display will ask "Are you sure?" (Press **STORE**). Pressing **Store** will cause the currently running effect to be saved, as is, in the location shown in the lower lefthand corner of the display. A brief message will inform you that the effect is being stored, then the PCM 81 will revert to the mode it was in before **Store** was pressed, with the newly-stored effect loaded.

When you store an effect, the following things are saved:

- Values of all Edit matrix parameters. This includes the initial values for any patch destinations when **Patch Update** (Control mode 1.7) is set to **Delayed**.
- Soft Row parameter assignments.
- Value of **ADJUST** when the effect was stored.
- Last parameter selected when the effect was stored.
- Tempo **Rate** and **BeatValue**.
- Tap **Duration** and **Average**.
- **ADJUST** name.
- Effect name.

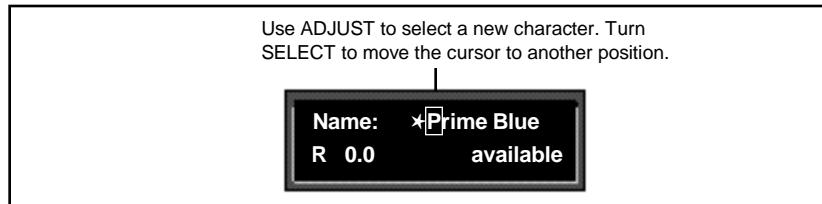
#### How the PCM 81 selects a default Bank and Register

If a register is loaded when **Store** is pressed, the default location will be the same as the loaded effect. If a preset is loaded when **Store** is pressed, the default location is determined as follows:

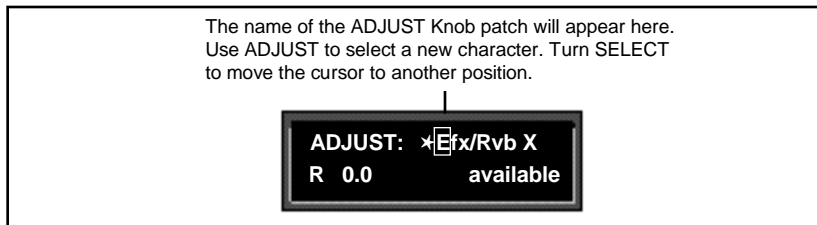
The PCM 81 always uses the last-stored register location as a starting point. Its default choice is the next "available" register within that bank. If there are no available registers at a higher location number within the same bank, the default location will be that of the last stored register.

If the last stored register is on a RAM card, and the RAM card is not inserted, the PCM 81 will search for available registers starting at 0.0 in the Internal Register Bank.

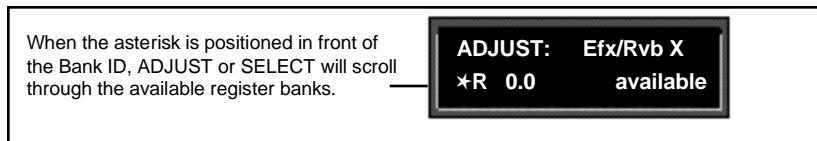
**Renaming the Effect** Renaming an effect is straightforward. With the asterisk and the cursor positioned as shown, turn ADJUST to select a new character. Press **Up** or **Down** to select a new *type* of character (upper case, lower case, numeric, symbolic, or blank). Simultaneously press **Up** and **Down** to clear all characters from the cursor to the end of the line. Turn SELECT to reposition the cursor over another character, and use ADJUST to change it. Continue in this manner until you have finished entering the new name. A maximum of 12 characters (including spaces) may be used.



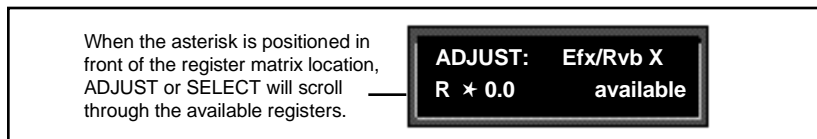
**Renaming the ADJUST Knob Patch** Press **Load/\*** to select the ADJUST patch for renaming. The display will change to show the following:



**Selecting a Bank and Register Location** Press **Load/\*** to move the asterisk to the Register Bank ID.



Press **Load/\*** again to move the asterisk to the register matrix location.



Once you have made all the changes you want to an effect, and have selected a register location, press **Store**. If you press **Store**, the display will ask "Are you sure?" (Press STORE). If you don't want to save the effect as shown, press any button to cancel the operation. Press **Store** to complete the operation.

Setting Edit Mode to **Pro** (Control Mode 1.0) gives you access to the full parameter matrix of the algorithm for any loaded effect whenever you press the front panel **Edit** button.

### The Full Edit Matrix

To select any parameter for adjustment, use **SELECT** to move horizontally across the matrix and the **Up** and **Down** buttons to move vertically. An asterisk (\*) appearing before a displayed parameter indicates that more than one parameter is available at that location. Press **Load/\*** to display these additional parameters.

Each of the 17 PCM 81 algorithms has a unique matrix, but many parameters are common to all effects, and their placement within the matrix is consistent. For example, all parameters within a given row of any matrix are related. This type of grouping is immediately apparent from the name of the row. For example the "Levels" row in any matrix will contain all of the available level controls for the effect's delay voices.

As far as possible, rows with similar controls will always appear in the same position in the matrix and, within each row, parameters will generally appear in the same position. For example, the first row (0) in every algorithm is "Controls". The **Mix** parameter, which is available in all effects, is always located in position 0 in this row. Similarly, the last two rows of every matrix are "Modulation" and "Patches". This makes it very easy to find your way around the large number of available parameters, and to anticipate where to find the controls you are looking for when you switch between effects.

<b>Concert Hall</b>														
0 Controls	0.0 Mix	0.1 FX AdjW	0.2 *Rel Lvl Pan	0.3 High Cut	0.4 Voice Dtl	0.5 FX Mix	0.6 FX MixIn	0.7 *RCU U6T						
1 Pre-Time	1.0 LowFR	1.1 Mid FR	1.2 Crossover	1.3 RH C	1.4 Pre Dela.	1.5 *Rel Lvl DL	1.6 *PH GM							
2 Pre-Design	2.0 Size	<b>Plate</b>												
3 Levels	3.0 Master	0 Controls	0.0 Mix	0.1 FX AdjW	0.2 *Rel Lvl Pan	0.3 High Cut	0.4 Voice Dtl	0.5 FX Mix	0.6 FX MixIn	0.7 *RCU U6T				
4 Delay Time	4.0 *Master	1 Pre-Time	1.0 LowFR	1.1 Mid FR	1.2 Crossover	1.3 RH C	1.4 Pre Dela.	1.5 *Rel Lvl DL	1.6 *BoRML DL	1.7 *PH GM				
5 Feedback	5.0 Master	2 Pre-Design	2.0 Size	<b>M-Band+Rvb</b>										
6 Panning	6.0 Master	3 Levels	3.0 Master	0 Controls	0.0 Mix	0.1 FX AdjW	0.2 *Rel Lvl Pan	0.3 FX Mix	0.4 FX MixIn	0.5 *RCU U6T				
7 Modulation	7.0 Mod L FD	4 Delay Time	4.0 *Master	1 Pre-Time	1.0 LowFR	1.1 Mid FR	1.2 Crossover	1.3 RH C	1.4 Pre Dela.	1.5 *Rel Lvl DL	1.6 *BoRML DL			
8 Patch++	Patch 0	5 Feedback	5.0 Master	2 Pre-Design	2.0 Size	2.1 Diffusion	2.2 Shape	2.3 Spread	2.4 Spin	2.5 Link	2.6 Pre Out			
		6 Panning	6.0 Master	3 Levels	3.0 Master	3.1 Voice 1	3.2 Voice 2	3.3 Voice 3	3.4 Voice 4	3.5 Voice 5	3.6 Voice 6			
		7 Modulation	7.0 Mod L FD	4 Delay Time	4.0 *Master	4.1 Voice 1	4.2 Voice 2	4.3 Voice 3	4.4 Voice 4	4.5 Voice 5	4.6 Voice 6			
		8 Patch++	Patch 0	5 Feedback	5.0 Master	5.1 *Voice 1	5.2 *Voice 2	5.3 *Voice 3	5.4 *Voice 4	5.5 *Voice 5	5.6 *Voice 6			
				6 Panning	6.0 Master	6.1 Voice 1	6.2 Voice 2	6.3 Voice 3	6.4 Voice 4	6.5 Voice 5	6.6 Voice 6			
				7 Modulation	7.0 Master	7.1 Voice 1	7.2 Voice 2	7.3 Voice 3	7.4 Voice 4	7.5 Voice 5	7.6 Voice 6			
				8 Patch++	Patch 0	8.1 Mod FR Env	8.2 Mod Env L	8.3 Mod Env R	8.4 Mod Latch	8.5 Mod SMI	8.6 Mod 6mC			
						Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6	Patch 7	Patch 8	Patch 9

Similar parameters can be found in similar positions in every effect matrix. Once you are familiar with one edit matrix, you will find it easy to anticipate where to find controls in another matrix.

Detailed information about each individual edit matrix, as well as parameter and effect descriptions are given in Chapter 3.

**Creating a Soft Row** In **Pro** mode you still have complete access to the Soft Row, which appears above row 0 of the full edit matrix. The parameters assigned here are duplicates of selected parameters in the matrix and can be adjusted from Row **S** (Soft Row), or from their matrix location.

The following example shows the edit matrix for the preset, **P 0.0 Prime Blue** (Chorus+Rvb algorithm). As you can see, parameters from locations throughout the matrix have been assigned to this Soft Row. In this preset (and in every preset) we have designed the Soft Row by assigning the parameters in numerical order, so that parameters from the Controls row are assigned in order to the first locations, followed by assignments from the Rvb Time row, and so on. As Soft Row parameters are also labeled with the name of the row from which they were taken, this makes it relatively easy to find the source parameter for any Soft Row assignment.

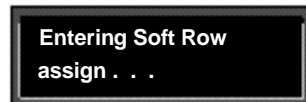
The Soft Row assigned to an effect appears above Row 0 in the full Edit matrix.

# (Full Row)	Control C-1	Control C-2	Control C-3	DL Time D-1	DL Time D-2	DL Time D-3	DL Time D-4	DL Time D-5	DL Time D-6	DL Time D-7	DL Time D-8	DL Time D-9	DL Time D-10
0 Control	R-1	R-2	R-3	R-4	R-5	R-6	R-7	R-8	R-9	R-10	R-11	R-12	R-13
1 Rvb Time	L-1	L-2	L-3	L-4	L-5	L-6	L-7	L-8	L-9	L-10	L-11	L-12	L-13
2 Rvb Control	C-1	C-2	C-3	C-4	C-5	C-6	C-7	C-8	C-9	C-10	C-11	C-12	C-13
3 Lamb	L-1	L-2	L-3	L-4	L-5	L-6	L-7	L-8	L-9	L-10	L-11	L-12	L-13
4 Delay Time	D-1	D-2	D-3	D-4	D-5	D-6	D-7	D-8	D-9	D-10	D-11	D-12	D-13
5 Chorus	C-1	C-2	C-3	C-4	C-5	C-6	C-7	C-8	C-9	C-10	C-11	C-12	C-13
6 Reverb	R-1	R-2	R-3	R-4	R-5	R-6	R-7	R-8	R-9	R-10	R-11	R-12	R-13
7 Panning	P-1	P-2	P-3	P-4	P-5	P-6	P-7	P-8	P-9	P-10	P-11	P-12	P-13
8 Modulation	M-1	M-2	M-3	M-4	M-5	M-6	M-7	M-8	M-9	M-10	M-11	M-12	M-13
9 Pitch	P-1	P-2	P-3	P-4	P-5	P-6	P-7	P-8	P-9	P-10	P-11	P-12	P-13

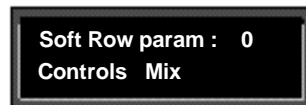
x  
to the  
line,  
|

Modifying the Soft Row, or creating a completely new Soft Row for an effect is easy:

1. From the full Edit matrix, press **Up** until you get to the Soft Row, indicated by an **S** in the lower left corner of the display.
2. With any Soft Row parameter displayed, press and hold down the **Edit** button. The following display will appear briefly.



When you release the **Edit** button, the display will change to the Soft Row Assignment display shown below. The **Edit** button LED will begin flashing and will continue to flash as long as you are in Soft Row Assign.



Here we show the first Soft Row parameter in our example — the actual Soft Row parameter selected will correspond to whichever one was displayed when you pressed and held **Edit**.

3. Turn ADJUST to scroll through all of the available parameters in the Edit matrix row by row, in numerical order. The entry "available" can also be selected to indicate that no assignment has been made at that position.
4. Turn SELECT to display another Soft Row position (0-9) for assignment. For each position, you can choose any effect parameter, including one(s) you have already assigned to a Soft Row position.
5. When you have arranged the Soft Row assignments as you want them, press **Edit** to exit Soft Row Assign and return to your last position in the Edit matrix. Pressing **Up** or **Down** once will also cause you to exit Soft Row Assign. Another push of either of these buttons will move you vertically through the Edit matrix.

When you return to **Go** mode, you will see the parameters in the order you assigned them — the spaces from any unassigned row positions will not appear.

**Patching** Patching is the ability to assign a control (Source) to any PCM 81 parameter (Destination). This allows you to alter the value of the parameter by manipulating the control Source. For example, you can select the front panel ADJUST knob as a Source and an effect's Mix parameter as a Destination. This simple patch will allow you to dynamically alter the mix of the effect whenever you turn ADJUST. You can create as many as 10 patches, each with as many as 8 pairs of pivot points. You can patch multiple parameters to a single controller, or patch multiple Sources to a single Destination.

**About Sources** All Sources are the same in the sense that each generates an output value in the range of 0-127. The output value is used to increase or decrease the setting of a Destination parameter. Sources differ in the manner in which they generate an output. Some generate values continuously (they're always "on"); some generate output based on the position of a particular external MIDI controller, or an external footpedal, or footswitch. Some Sources generate output based on aspects of physical performance such as how loud, how fast, or how hard you play. The PCM 81 allows you to choose from a selection of Sources as shown for each Destination. A list of the available Sources is shown on the following page.

**The Patch Row** Each PCM 81 effect has an identical Patch row at the end of its Edit matrix where you can make as many as 10 patches.



To make a patch, use SELECT and the **Down** button to move down through the Edit matrix to the Patch row. A display such as the following will appear.

The asterisk (\*) indicates that Sources will be selected when you turn ADJUST. Press **Load/\*** to change the selection to **Dst**. Press **Load/\*** again to bring up the Values display where you can select **Src Value** and **Dst Value**.

SELECT will move you across the Patch row (0-9).

The Source type is indicated here. (**Int**, **MIDI** or **001-119** to indicate a MIDI Controller.)

**Patch 0**    **\*Src**

**Int**                      **LFO**

The name of the current Source assignment is shown here. (The default Source assignment is **Int...Off**.)

Pressing the **Load/\*** button will change the display to allow you to sequentially select: **Src**, **Dst**, **Src Value** and **Dst Value**. Press **Up** or **Down** together with **Load/\*** to backstep to the previous parameter.

Three types of sources are available: Internal, MIDI and MIDI Controller. These types are indicated in the Source list by the labels: **Int**, **MIDI**, or a number (**001-119**). Turn ADJUST to scroll through the entire list of available sources. **Assigning a Source**

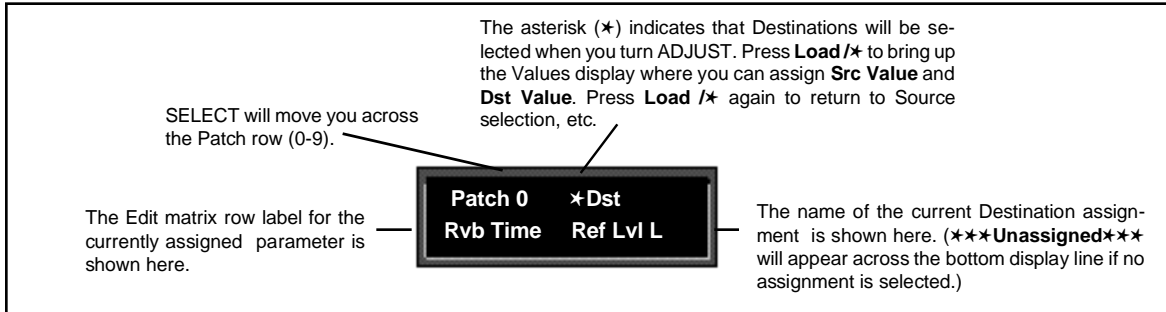
Internal
LFO
Sine
Cosine
Square
Sawtooth
Pulse
Triangle
Env L
Env R
AR Env
Latch
Sw 1
Sw 2
Sw 1 & 2
Mono Lvl
Left Lvl
Right Lvl
Footpedal
Foot Sw 1
Foot Sw 2
ADJUST
Tempo
On
Off

MIDI
P Bend
A Touch
Velocity
Last Note
Low Note
High Note
Clk Comnds

MIDI Controller Numbers		
(PCM 81 interprets 000 as Bank Select)	070	Sound Var
001 Mod Wheel	071	Timbre
002 Breath	072	Release
003 Ctl 3	073	Attack
004 Foot Ctl	074	Bright
005 PortaTime	075	Sound 6
006 Data Entry	076	Sound 7
007 Volume	077	Sound 8
008 Balance	078	Sound 9
009 Ctl 9	079	Sound 10
010 Pan	080	General 5
011 Xpression	081	General 6
012 Effect 1	082	General 7
013 Effect 2	083	General 8
014 Ctl 14	084	Porta Ctl
015 Ctl 15	085	Ctl 85
016 General 1	...	...
017 General 2	090	Ctl 90
018 General 3	091	FX1 Depth
019 General 4	092	FX2 Depth
020 Ctl 20	093	FX3 Depth
...	094	FX4 Depth
031 Ctl 31	095	FX5 Depth
(PCM 81 interprets 032 as Bank Select)	096	Data Inc
033 Ctl 33	097	Data Dec
...	098	NRPN LSB
063 Ctl 63	099	NRPN MSB
064 Sustain	100	RPN LSB
065 Porta On	101	RPN MSB
066 Sostenuto	102	Ctl 102
067 SoftPedal	...	...
068 Legato	119	Ctl 119
069 Hold 2		

**Patch Sources**

**Assigning a Destination** Once you have selected a Source, press **Load /\*** and the display will change to allow you to assign a Destination (**Dst**).



Turn ADJUST to scroll through all of the parameters in the Edit matrix of the currently loaded effect, including the Modulation parameters.

Once you have assigned a parameter as a Destination, the controller you have assigned “owns” that parameter. Adjustments made to this parameter from the Edit matrix, will only affect the initial value of the parameter when the program is loaded.

The behavior of the parameter on program load is determined by the setting of the Patch Update parameter in Control mode (1.7). This parameter can be set to **Immediate** or **Delayed**.

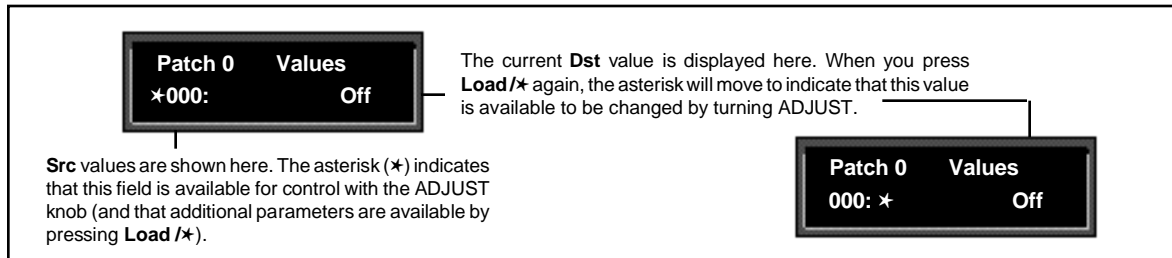
When **Immediate** is selected, the initial value of the parameter value will correspond to the controller position.

When **Delayed** is selected, the stored value of the parameter will continue to be in effect until the controller is moved. (It is, therefore, a good idea to set a sensible value to the parameter in the Edit matrix.)

Mod Row parameters that can be assigned as Patch Destinations	
<b>LFO</b>	Rate, Shape, P Width, Depth
<b>AR ENV</b>	Attack, Release, Mode
<b>Env L</b>	Release
<b>Env R</b>	Release
<b>Sw 1</b>	Rate, P Width, Mode
<b>Sw 2</b>	Rate, P Width, Mode

## Assigning Values

Once you have assigned a Destination, press **Load /\*** to get to the **Values** display.



This display allows you to assign Destination values to specific Source values. These assignments are made in pairs, each with a value for the Source and a value for the Destination. For example, the default is two pairs mapped as follows:

minimum Src value (0) = minimum Dst value  
 maximum Src value (127) = maximum Dst value

This establishes a linear relationship between the parameter and the controller. Inverse control is accomplished easily by reversing these settings. As many as eight pairs of Destination/Source values, or pivot points, can be assigned here, providing an exciting new level of dynamic control.

## Jump

When creating patches, there are situations in which you will want to leave the Patch row to adjust parameters. To make this convenient, a *Jump* command is available. Jumping is dependent on the current Patch display, and is activated simply by pressing **Edit** while a certain display is active. This will jump you out of the Patch row and to the location where you can make the necessary adjustments. Pressing **Edit** again will jump you back to the Patch Row. (Note that using any front panel controls other than those required to adjust the parameter to which you have jumped, will disable the jump. This is not catastrophic, but it will require you to return by using **Up** and **Down** and **SELECT**.) The following Jumps are available:

From the Patch row **Src** display:

- With ADJUST selected as the Source, press **Edit** to jump to the Controls row, where you can specify range limits for the ADJUST knob. Press **Edit** again to return to the Patch row.
- With any modulation parameter selected as a Source, press **Edit** to jump to the Modulation row position of the Source. For example, if the Patch source **LFO** is displayed, press **Edit** to jump to Modulation row position 0 (LFO) where you can edit any LFO parameter value. Press **Edit** again to return to the Patch row.

**Jump cont'd.**

From the Patch row **Dst** selection display:

- Press **Edit** to jump to the Edit controls for the parameter you have selected as the Destination. You will have complete access to all parameter controls, including any subparameters at that location. Press **Edit** again to return to the Patch row.

From the Patch row **Values** display:

- Press **Edit** to jump to the next **Src** or **Dst** value. Default values are 0...minimum, 127...maximum.

**Patching Examples**

The following examples illustrate how to create a patch, use the patch jump features, modify the default patch values and add an additional pivot point to the example patch values.

**Creating a patch with default values**

Load program **P0 0.1 EkoChorus**. Press **Edit** to enter Edit mode, then press **Up** to move to the Patch Row. Press **Load/★** until the display looks like this:



Turn SELECT to select Patch 1 (which is set to **Off**).



Turning ADJUST will scroll through the entire list of available patch Sources. Turn ADJUST counterclockwise until **LFO** is displayed in the lower right.



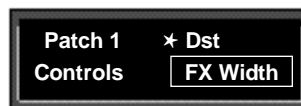
The LFO is now assigned as a patch Source.

Press **Load/\*** to bring up patch Destinations for selection. The display should show that Destination is unassigned.



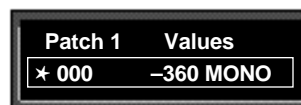
The ADJUST knob will now scroll through all of the available parameters of **EkoChorus**. The lower line of the display will show the edit matrix row label on the left, and the parameters in that row on the right.

Turn ADJUST clockwise until **FX Width** (in the effect's Controls row) is displayed in the lower right corner of the display.

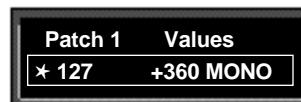


The FX Width parameter is now assigned as the patch Destination.

Now, press **Load/\*** to bring up the Values display. This will show the default Destination value setting (–360 MONO). This is the value assigned to FX Width when the LFO is at its minimum value (000).



Turn ADJUST one click counterclockwise to display the default Destination value (+360 MONO) assigned to FX Width when the LFO is at its maximum value (127).



That's all there is to setting up a default patch—select a Source and Destination, and the minimum and maximum patch values are set automatically.

Of course, you will often want to modify the patch further, either by adjusting the modulation source parameters, changing the default values or adding additional pivot points. In the following sections, we'll continue using this patch to demonstrate examples of these modifications. When we're done, the new patch will add dynamic spatialization to the EkoChorus program.

**Adjusting the modulation source parameters**

Continuing the previous example, we'll adjust the rate of the LFO by jumping to it from the Patch row.

Press **Load/\*** repeatedly to return to the Patch 1 Source selection display.



Press **Edit** to jump directly to the LFO parameters in the Mod row.

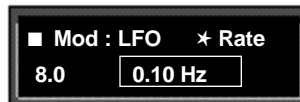
The asterisk (\*) indicates that LFO **Rate** will be altered when you turn ADJUST. Press **Load/\*** to change the selection to the other LFO parameters ( **Shape**, **P Width** and **Depth**).

Patch Source Indicator



The display now shows position 8.0 in the Edit matrix Mod Row. The small square in the upper left corner of the display is a *patch source indicator*. This indicator appears whenever a modulator (such as the LFO in this example) has been assigned as a patch Source in the effect being edited.

Turn ADJUST to change the LFO rate to **0.10Hz**.



Now, press **Edit** to jump back to your previous position in the Patch row.



Let's modify the patch further by adjusting the Destination values to a more useful range.

### Changing the default destination values

Press **Load/★** repeatedly until the Patch 1 Values screen is displayed.



Notice that the ★ is to the left of the Source value. This indicates that the Source value is selected and its value will be changed when you turn ADJUST.

Press **Load/★** once to move the ★ to the right of the Source value. When the ★ is in this position, ADJUST will change the Destination value.



The asterisk (★) indicates that the Source value will be altered when you turn ADJUST.



Press **Load/★** to move the asterisk to this position, where it indicates that Destination values will be altered when you turn ADJUST.

With the Destination value selected, turn ADJUST clockwise to set the value to **+0**.



Next, we'll want to adjust the Destination value when the LFO is at its maximum value. One way to do this is to press **Load/★** three times to cycle the ★ to the display of the Source value, and continue on from there — but we've provided a short cut! Press **Edit** to jump to the next assigned value ( in our example, the LFO maximum value).



Note that the ★ remains in the same position, so you can just turn ADJUST to set the new Destination value. Set the value to **+90**.



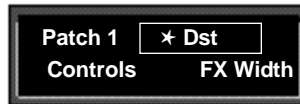
Now our example has been modified so that the LFO sweeps the FX Width Value from 0 to 90. This creates a dynamic alteration of the effect's spatial characteristics. Its stereo image changes smoothly from mono to stereo, to surround, and back again.

**Adding an additional pivot point to the patch**

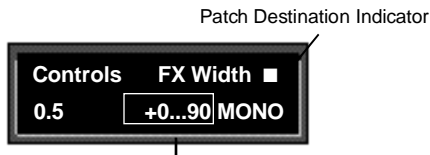
So far, our example uses only two pairs of patch values. The Destination parameter moves linearly between the value assigned at 000 and the value assigned at 127.

You can watch this change by displaying the Destination parameter. Here's how to jump directly to it from the patch:

Press **Load/★** repeatedly to return to the Patch 1 Destination selection display.



Press **Edit** to jump directly to the FX Width parameter in the Controls Row.

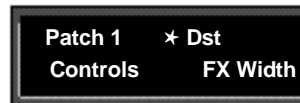


The FX Width value will be changing from 0 to 90 and back again.

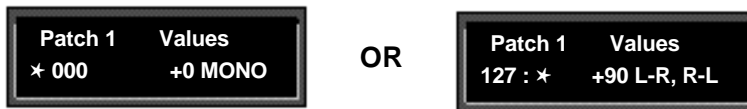
The display will change to show position 0.5 in the Controls Row. Note that the value is changing continuously from 0 to 90 and back again. Notice also the small square in the upper right corner of the display. This patch destination indicator appears whenever a parameter has been assigned as a patch Destination in the effect being edited.

Now let's return to the Patch row to add a pivot point to the effect.

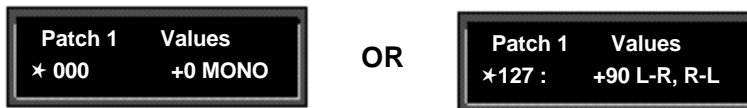
Press **Edit** to jump back to the Patch row.



Press **Load/✱** to bring up the Values display. The last value edited will be displayed, so you will see *either* the minimum or maximum value.



If the ✱ is not at the left of the Source value, press **Load/✱** three times to move it there. (You can take a short cut instead — simultaneously press **Down** and **Load/✱** to back-step.)



Turn ADJUST to display **64**. This will be the Source value of our new pivot point. The string of dots in the destination value portion of the display indicate that there is no Destination value assigned when the source value is 64.



Press **Load/✱** to move the ✱ to the right of the Source value, and turn ADJUST clockwise to set the Destination value for this point to **-45**.



By adding this pivot point, we have put a “kink” in the patch. The value of the Destination parameter no longer moves in a straight line between 0 and 90. Instead it moves from 0 to -45, and then from -45 to 90. This will produce a very different sounding spatial change from the original patch. You can see the difference by pressing **Load/✱** twice to display the patch Destination, then pressing **Edit** to jump back to the Destination parameter to watch its value change.

**Multiple Patches with the Same Destination**

If you create two or more patches with the same Destination, the Destination value will be the sum of all of the patches assigned to it.

For example, if Pedal and ADJUST are both assigned to Mix, the Mix value will be the sum of the patch Destination values for those two patches. When creating multiple patches to the same Destination, you should set the individual Destinations to values which, when added together, are less than or equal to the maximum value for the parameter. Pedal and ADJUST, for example, could each have a maximum value of 50%, or they could be assigned values of 25% and 75%, 60% and 40%, etc.

When the sum of multiple patched parameter Destination values is greater than the maximum value of the parameter, the parameter value will remain at maximum until the sum of the patches falls below it.

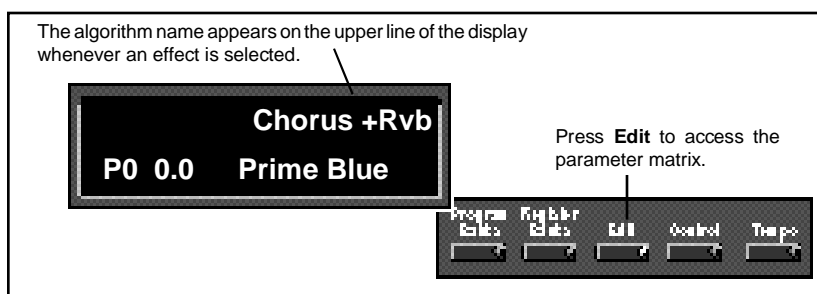
**Mod Row Patches**

**AR ENV**, **Latch**, **Sw 1** and **Sw 2** are each activated by assigning a threshold source to **T Src** that is used to turn them on and off. This assignment is a subparameter in the Mod row — not in the Patch row.

For an example, check out ADJUST and Latch in the preset **P0 3.1 Rotor Cabinet**.

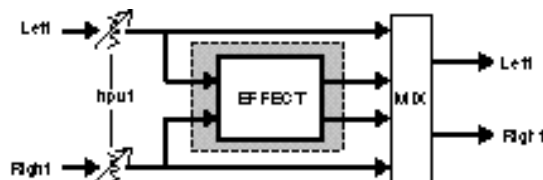
## The Algorithms and Their Parameters

The PCM 81 uses 17 algorithms to create different types of effects. Each algorithm includes an uncompromised stereo reverb effect, as well as several voices of additional stereo effects. When you select any effect, the name of the algorithm from which it was derived will appear on the upper display line. Pressing **Edit** will display the last edited parameter in that algorithm's parameter matrix.



Most PCM 81 algorithms share the same general structure, shown below. The shaded area of the diagram is detailed in the individual effect descriptions that follow.

## About the Algorithms



There are three general classes of algorithm: 4-Voice, 6-Voice and Pitch. The 4-Voice algorithms: Concert Hall, Plate, Chamber, Inverse and Infinite each combine a specific type of reverb with a 4-voice, general purpose stereo "effect toolbox" which we call the Reverb Shell. These algorithms also provide "post-processing" for the reverb. The 6-Voice algorithms: Glide>Hall, Chorus+Rvb, M-Band+Rvb, Res 1>Plate, and Res 2>Plate each combine a specific type of reverb with a specialized 6-voice stereo effect. The Pitch algorithms: Quad>Hall, Dual-Chmb, Dual-Plt, Dual-Inv, Stereo-Chmb, VSO-Chmb and Pitch Correct each offer a unique type of pitch shifting, combined with an uncompromised Lexicon reverb.

In this chapter, diagrams and descriptions of the Reverb Shell, and of the 4 and 6-Voice algorithms are presented first, along with pictures of each edit matrix. The diagrams are followed by a glossary of parameter descriptions, organized alphabetically by matrix row name. Within each matrix row, parameters are organized as far as possible in the sequence in which they appear in the row. The Pitch algorithms, most of which make use of a dedicated Submixer incorporated into each algorithm, are described in the following section along with operational tips on pitch shifting in general and the submixer in particular.

## The 4-Voice Algorithms

**The Reverb Shell** Each of the 4-Voice algorithms share a common set of controls and parameters built around one of five stereo reverb effects: Concert Hall, Plate, Chamber, Inverse and Infinite. The diagram below shows these common controls and parameters as they are structured around a reverb effect. We call this structure the Reverb Shell. (Individual reverb effects are described on the following pages.)

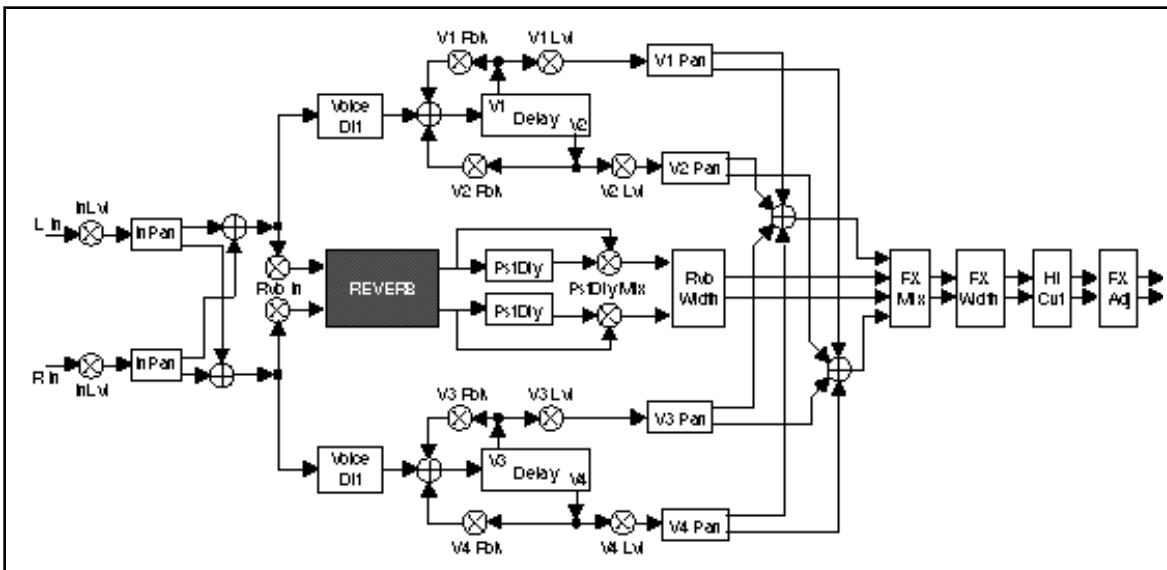
In the 4-Voice algorithms, input levels and pans determine the signal flow to left and right pairs of delay voices, and also to the reverb effect. Each delay voice has individually adjustable level, delay, feedback, and pan controls, as well as master controls for all of the voices. These algorithms can be used to create a wide variety of stereo delay and modulation effects in parallel with studio quality stereo reverberation.

The two delays after the reverb effect are referred to as post delays. The **PstDly Mix** control determines how much post delay is actually heard. The **Rvb Width** control affects only the output of the reverb portion of the effect.

The delays and the output of the reverb effect are summed at the **FX Mix** control. This control allows you to blend the relative amounts of delay and reverb.

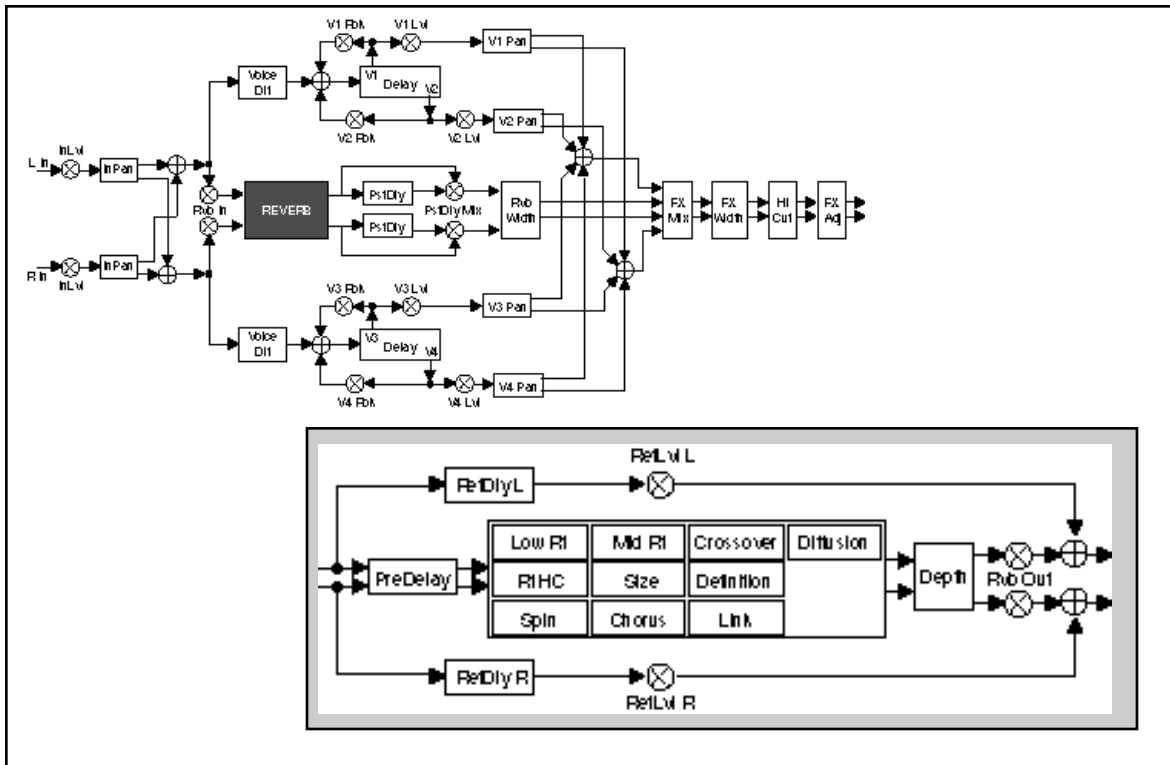
**FX Width** controls the spatial characteristics of the composite reverb and delay effect.

The **Hi-Cut** parameter provides high end rolloff, while **FX Adjust** determines the output volume of the entire processed signal.



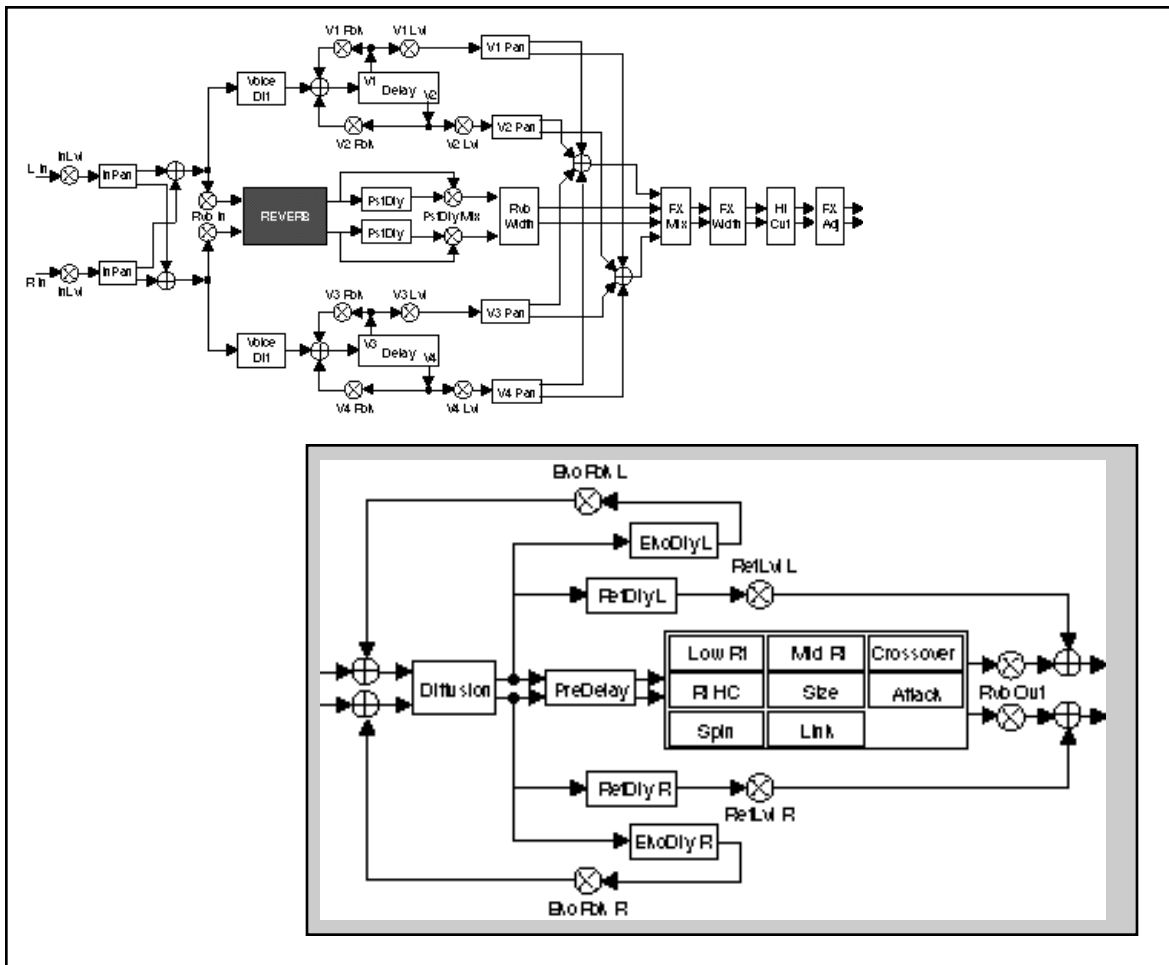
This algorithm emulates a real concert hall. The reverberation is very clean, and designed to remain behind the direct sound — adding ambience, but leaving the source unchanged. This effect has a relatively low initial echo density which builds up gradually over time.

### Concert Hall



0 Control+	0.0 Mix	0.1 FX Adj	0.2 *InLvlPan	0.3 High Cut	0.4 Voice Df	0.5 FX Mix	0.6 FX Width	0.7 *DCU6T	
1 Pvb Time	1.0 Low Rf	1.1 Mid Rf	1.2 Crossover	1.3 Rf HC	1.4 Pre Dela.	1.5 *Rel LvlL	1.6 *Ph Gf		
2 Pvb Design	2.0 Size	2.1 Diffusion	2.2 Del	2.3 Depth	2.4 Spin	2.5 Chorus	2.6 Link	2.7 Rub Width	2.8 *Rub h Du
3 Level+	3.0 Master	3.1 Voice 1	3.2 Voice 2	3.3 Voice 3	3.4 Voice 4				
4 Delay Time	4.0 *Master	4.1 Voice 1	4.2 Voice 2	4.3 Voice 3	4.4 Voice 4				
5 Feedback	5.0 Master	5.1 Voice 1	5.2 Voice 2	5.3 Voice 3	5.4 Voice 4				
6 Panning	6.0 Master	6.1 Voice 1	6.2 Voice 2	6.3 Voice 3	6.4 Voice 4				
7 Modulation	7.0 Mod L Fd	7.1 Mod RFR Env	7.2 Mod EnvL	7.3 Mod EnvR	7.4 Mod Lach	7.5 Mod Sml	7.6 Mod Smc		
8 Patch++	Patch 0	Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6	Patch 7	Patch 8
									Patch 9

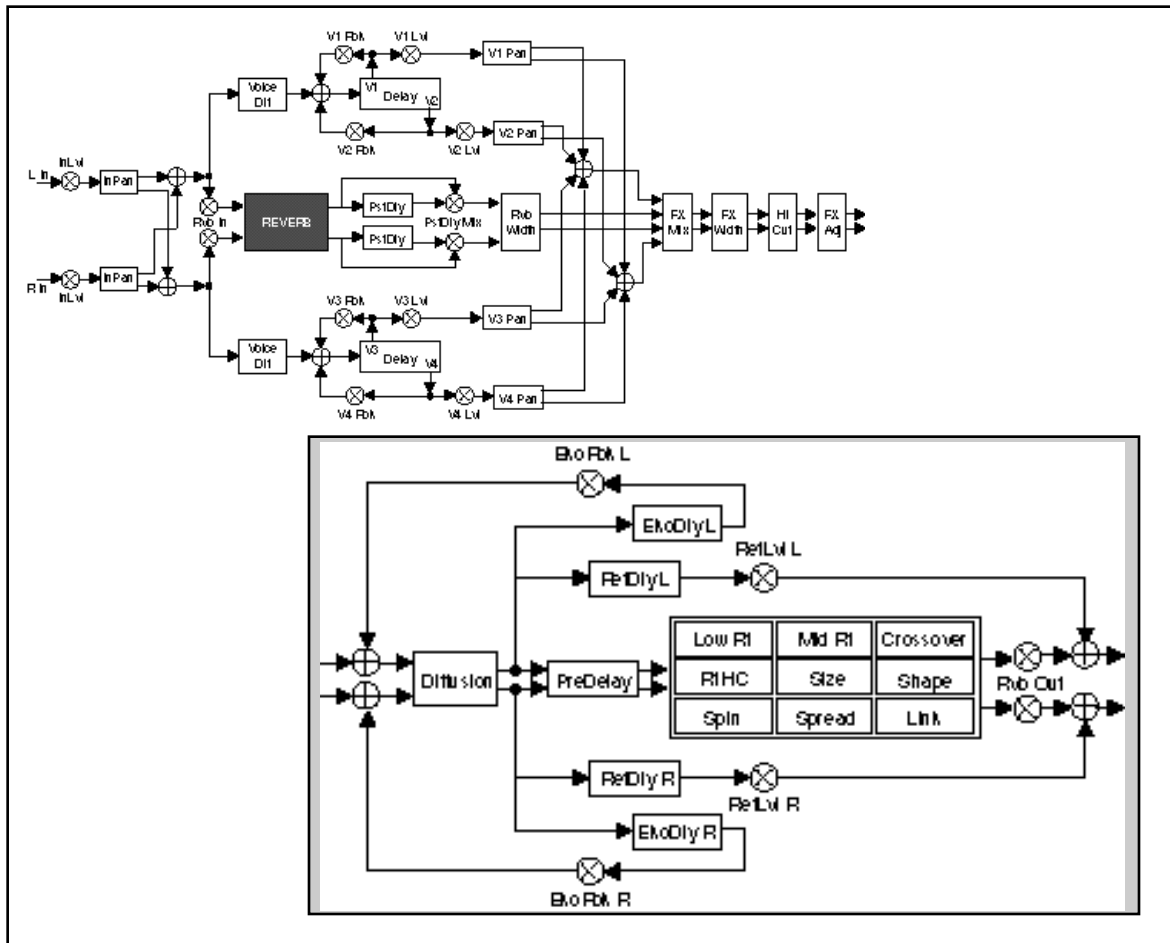
**Plate** The Plate algorithm mimics the sound of metal plates, with high initial diffusion and a relatively bright sound. This makes them a good choice for enhancing any type of percussion.



0 Control+	0.0 Mtr	0.1 FX Adj	0.2 *InLvl Pan	0.3 High Cut	0.4 Voice DI1	0.5 FX Mix	0.6 FX Width	0.7 *FDU St
1 Feedback	1.0 Low Rf	1.1 Mid Rf	1.2 Crossover	1.3 Rf HC	1.4 Pre Dela.	1.5 *ReLvlL	1.6 *ReLvlR	1.7 *Ph Gd
2 Feedback	2.0 Size	2.1 Diffusion	2.2 *Attack	2.3 Sphn	2.4 Link	2.5 *Rub Width	2.6 *Rub InOut	
3 Levels+	3.0 Master	3.1 Voice 1	3.2 Voice 2	3.3 Voice 3	3.4 Voice 4			
4 Delay Time	4.0 *Master	4.1 Voice 1	4.2 Voice 2	4.3 Voice 3	4.4 Voice 4			
5 Feedback	5.0 *Master	5.1 Voice 1	5.2 Voice 2	5.3 Voice 3	5.4 Voice 4			
6 Panning	6.0 *Master	6.1 Voice 1	6.2 Voice 2	6.3 Voice 3	6.4 Voice 4			
7 Modulation	7.0 Mod L RD	7.1 Mod Rf: Env	7.2 Mod: Env L	7.3 Mod: Env R	7.4 Mod Lch	7.5 Mod Sm 1	7.6 Mod Sm 2	
8 Patch+	Patch 0	Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6	Patch 7 Patch 8 Patch 9

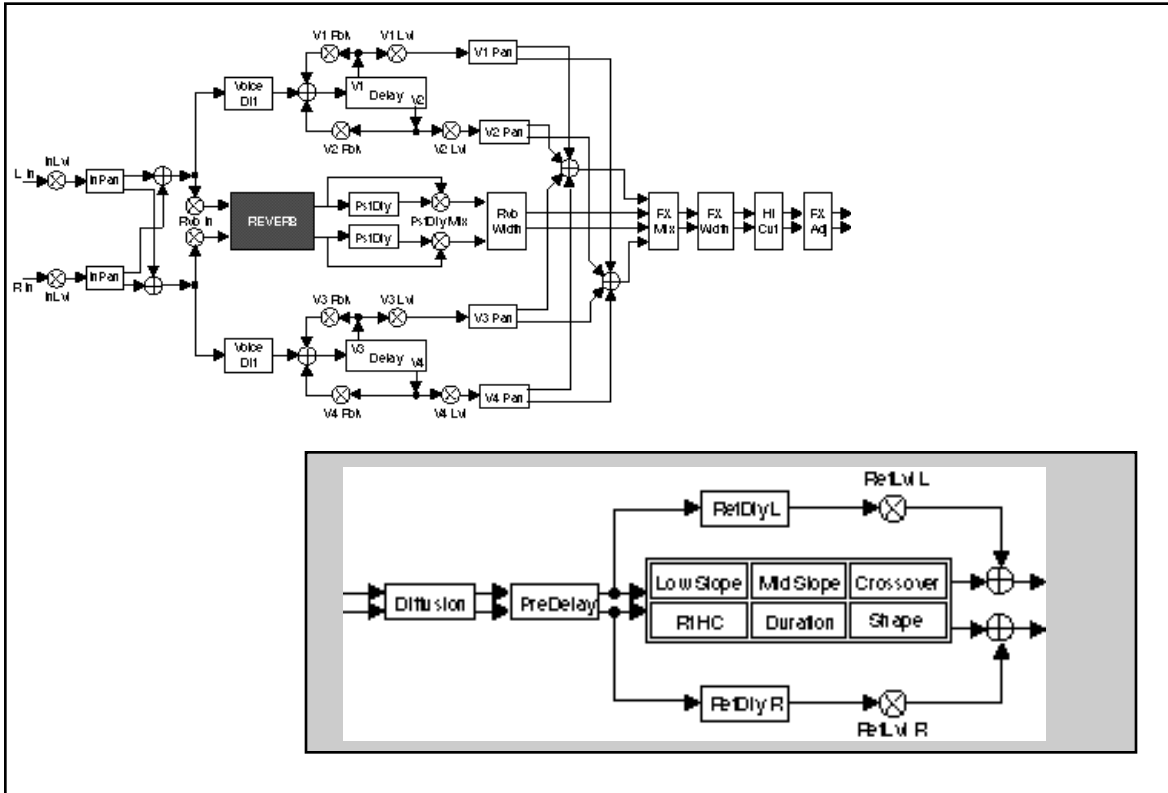
The Chamber algorithm produces an even, relatively dimensionless reverberation, with little change in color as the sound decays. The initial diffusion is similar to the Concert Hall algorithm, but the sense of space and size is much less obvious. This characteristic, along with the low color in the decay tail makes Chamber useful on a wide range of material. It is especially useful on spoken voice, giving a noticeable increase in loudness with very low color.

### Chamber



0	Controls	0.0 Mix	0.1 FX Adj	0.2 *InLvlPan	0.3 High Cut	0.4 Voice Dtl	0.5 FX Mix	0.6 FX Width	0.7 *ReDlyL		
1	Reverb Time	1.0 Low Rf	1.1 Mid Rf	1.2 Crossover	1.3 Rf HC	1.4 Pre Dela.	1.5 *ReLvlL	1.6 *EkoRfLvl	1.7 *Pan Rfd		
2	Reverb Design	2.0 Size	2.1 Diffusion	2.2 Shape	2.3 Spread	2.4 Spin	2.5 Link	2.6 Reverb Width	2.7 *Reverb InOut		
3	Level	3.0 Master	3.1 Voice 1	3.2 Voice 2	3.3 Voice 3	3.4 Voice 4					
4	Delay Time	4.0 Master	4.1 Voice 1	4.2 Voice 2	4.3 Voice 3	4.4 Voice 4					
5	Feedback	5.0 Master	5.1 Voice 1	5.2 Voice 2	5.3 Voice 3	5.4 Voice 4					
6	Panning	6.0 Master	6.1 Voice 1	6.2 Voice 2	6.3 Voice 3	6.4 Voice 4					
7	Modulation	7.0 Mod LFO	7.1 Mod RR Env	7.2 Mod EnvL	7.3 Mod EnvR	7.4 Mod Latch	7.5 Mod Env L	7.6 Mod Env R			
8	Path	Path 0	Path 1	Path 2	Path 3	Path 4	Path 5	Path 6	Path 7	Path 8	Path 9

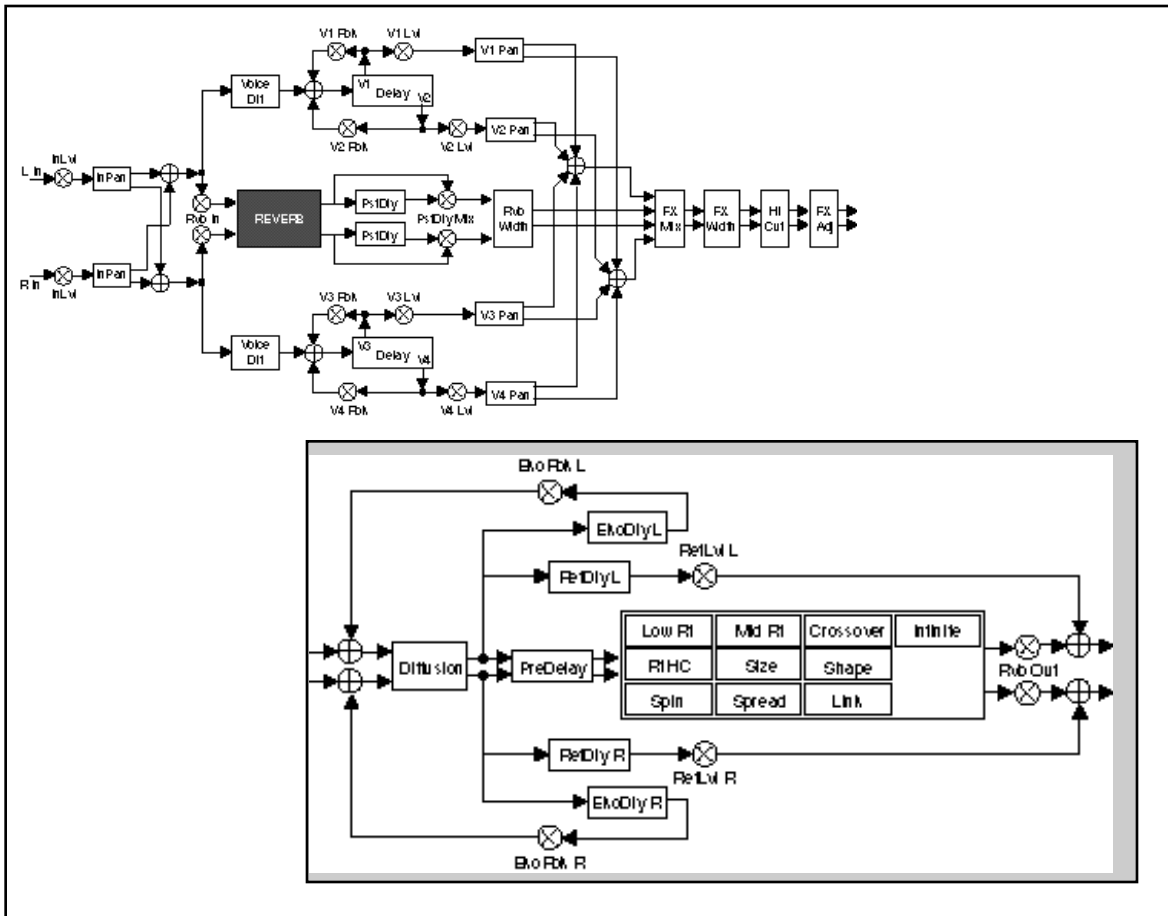
**Inverse** The Inverse algorithm allows you to vary the slope of the initial portion of the reverb envelope. The slope can decay, remain level, or rise over a variable time interval. When the time interval is up, the reverberation cuts off abruptly. The resulting effect is similar to a gate, but is not at all dependent on the level or complexity of the input signal. Slopes are adjustable over a negative, even, or positive slope. Positive slopes create inverse effects, while more even slopes create gated effects. Negative slope values have rather natural reverb tails.



0	Control+	0.0 Mtr	0.1 FX ADJN	0.2 *RL Ut Pan	0.3 High Cut	0.4 Voice DIT	0.5 FX Mtr	0.6 FX Width	0.7 *RDUST
1	Rfb Time	1.0 Low Slope	1.1 Mid Slope	1.2 Crossover	1.3 Rf HC	1.4 Pre Delay	1.5 *Rf Lvl L	1.6 *Rf Gld	
2	Rfb D+Ign	2.0 Duration	2.1 Diffusion	2.2 Shape	2.3 Rfb Width	2.4 Rfb In			
3	Level+	3.0 Master	3.1 Voice 1	3.2 Voice 2	3.3 Voice 3	3.4 Voice 4			
4	Delay Time	4.0 Master	4.1 Voice 1	4.2 Voice 2	4.3 Voice 3	4.4 Voice 4			
5	Pre+Dnck	5.0 Master	5.1 Voice 1	5.2 Voice 2	5.3 Voice 3	5.4 Voice 4			
6	Feedback	6.0 Master	6.1 Voice 1	6.2 Voice 2	6.3 Voice 3	6.4 Voice 4			
7	Modulation	7.0 Mod L RD	7.1 Mod RR Env	7.2 Mod Env L	7.3 Mod Env R	7.4 Mod Lch	7.5 Mod Sm L	7.6 Mod Sm C	
8	Patch++	Patch 0	Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6	Patch 7
									Patch 8
									Patch 9

Infinite is acoustically similar to the Chamber algorithm, with the addition of an Infinite parameter. When this parameter is turned on, the input to the reverberator ramps off. (Note that this still allows the Reverb Shell to be utilized.) With Infinite on, the reverb tail remains constant, creating strange and useful reverb effects. When Infinite is switched off, input to the reverberator is restored, and the current running reverb time is utilized.

**Infinite**



0 Control+	0.0 Mix	0.1 FX Adj	0.2 *InLvlPan	0.3 High Cut	0.4 Voice Dly	0.5 FX Mix	0.6 FX Width	0.7 *ReDlyT		
1 Reb Time+	1.0 Low Rf	1.1 Mid Rf	1.2 Crossover	1.3 Rf HC	1.4 Pre Del.	1.5 Infinite	1.6 *ReLvlL	1.7 *EnvRfL	1.8 *PreIGid	
2 Reb Design	2.0 Size	2.1 Diffusion	2.2 Shape	2.3 Spread	2.4 Spin	2.5 Link	2.6 Reb Width	2.7 *Reb InCu		
3 Level+	3.0 Master	3.1 Voice 1	3.2 Voice 2	3.3 Voice 3	3.4 Voice 4					
4 Delay Time	4.0 Master	4.1 Voice 1	4.2 Voice 2	4.3 Voice 3	4.4 Voice 4					
5 Feedback	5.0 Master	5.1 Voice 1	5.2 Voice 2	5.3 Voice 3	5.4 Voice 4					
6 Panning	6.0 Master	6.1 Voice 1	6.2 Voice 2	6.3 Voice 3	6.4 Voice 4					
7 Modulation	7.0 Mod LFO	7.1 Mod Rf Env	7.2 Mod EnvL	7.3 Mod EnvR	7.4 Mod Latch	7.5 Mod Sml	7.6 Mod Smc			
8 Patch++	Patch 0	Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6	Patch 7	Patch 8	Patch 9

## The 6-Voice Algorithms

Each 6-Voice algorithm is a combination of a specific 6-voice stereo effect and a specific reverb effect. These algorithms: Glide>Hall, Chorus+Rvb, M-Band+Rvb, Res1>Plate and Res2>Plate are each optimized for a particular class of audio processing effects in combination with studio quality stereo reverberation, bringing formidable processing power and flexibility to effects creation.

The 6-Voice algorithms use the same, uncompromised, stereo reverberation effects as the 4-Voice algorithms but, instead of the general-purpose Reverb Shell, each reverb is coupled with a unique 6-voice stereo effect: gliding delays, 6-voice chorus, multi-band EQ, and two 6-voice resonators. Voices 1-3 are connected to input audio panned to the left. Voices 4-6 are connected to input audio panned to the right. (Use the **InPan L** and **InPan R** controls at Control mode 0.2 to pan input audio.) Each voice has independent delay time, panning and level controls, in addition to other parameters specific to the particular effect.

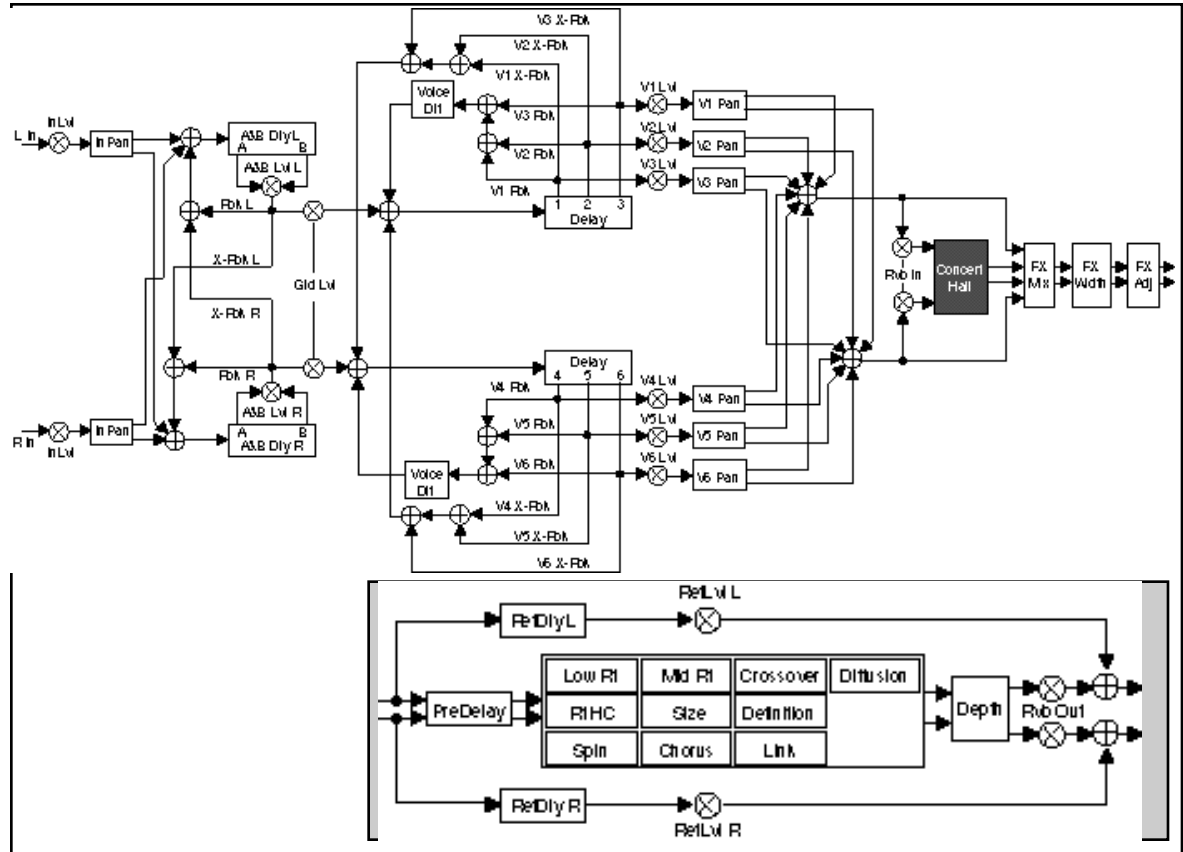
In the Glide>Hall, Res1>Plate and Res2>Plate algorithms, the reverb effect is in series with the 6-voice effect. Use **FX Mix** to set the relative level of dry and reverberated effect.

In the M-Band+Rvb and the Chorus+Rvb algorithms, the reverb effect is in parallel with the 6-voice effect. Use **FX Mix** to set the balance of the 6-voice effect and the reverb.

A stereo pair of 2-tap gliding delays feeds six individually adjustable delay voices. Each voice has its own level, feedback, delay, cross-feedback, and pan parameters. The output of these delay voices is fed into a Concert Hall reverb algorithm. Glide>Hall is useful for creating such effects as stereo flangers, loop samplers, pitch modulation, etc. which can then be fed into the reverb.

**Glide>Hall**

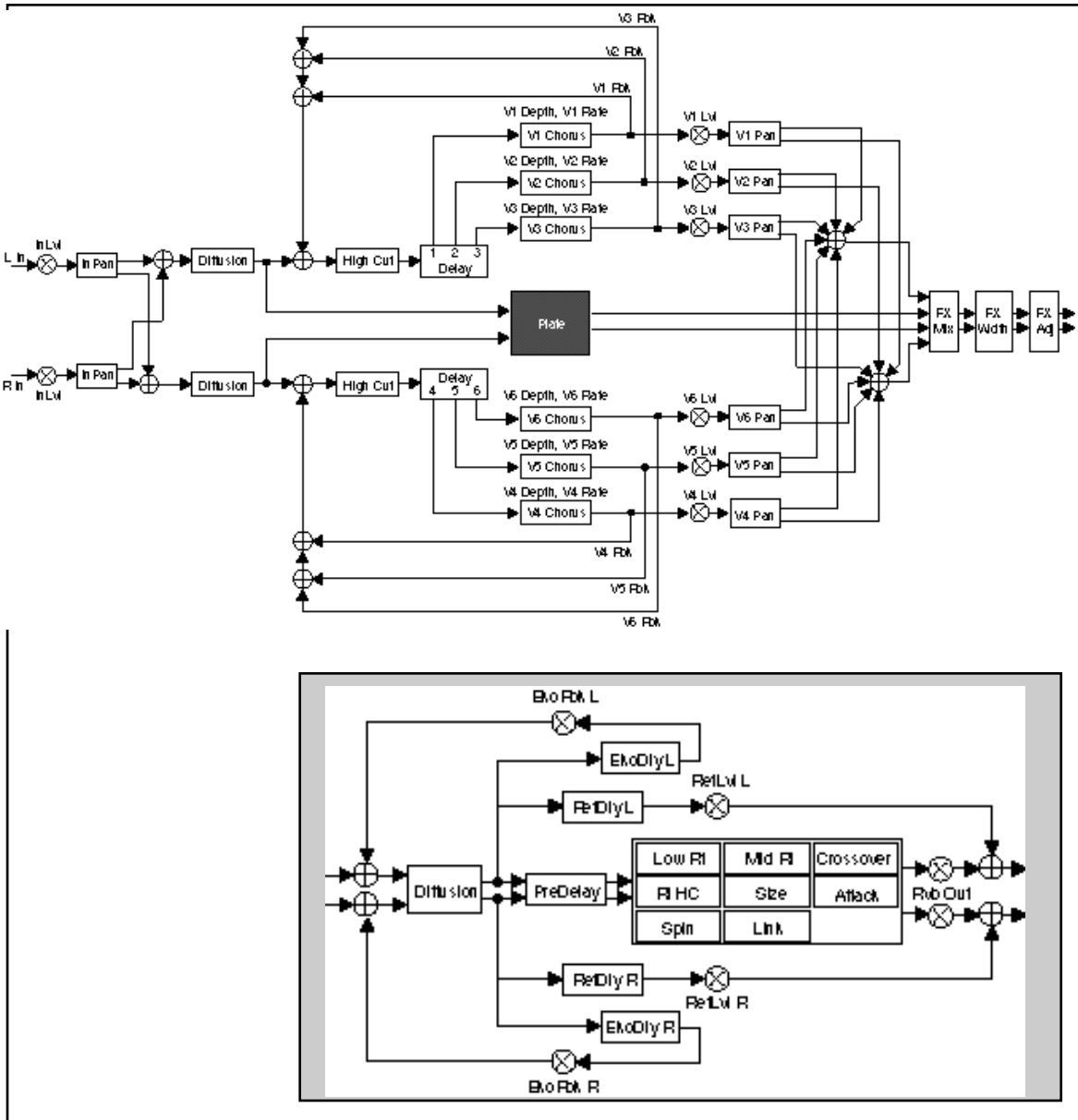
This algorithm can address up to 4 Meg of delay memory (with optional SIMMs added to the PCM 81), providing 42 seconds of stereo delay.



0 Control+	0.0 Mtr	0.1 FX Adj	0.2 In Lvl Pan	0.3 Wake Dtl	0.4 Fk Mtr	0.5 Fk Wdth	0.6 *RDU6T
1 Fbk Time	1.0 Low Rf	1.1 Mid Rf	1.2 Crossover	1.3 Rf HC	1.4 Pre Del.	1.5 *Rf Lvl DL	
2 Fbk Design	2.0 Size	2.1 Diffusion	2.2 Del	2.3 Depth	2.4 Spin	2.5 Chorus	2.6 Link
3 Glide FX	3.0 Gld Lvl	3.1 *Rf Lvl	3.2 *Rf Rvl	3.3 *Rf Lvl	3.4 *Rf Rvl	3.5 *Rf Lvl R	3.6 *Rf Rvl R
4 Levels	4.0 *Master	4.1 Voice 1	4.2 Voice 2	4.3 Voice 3	4.4 Voice 4	4.5 Wake 5	4.6 Wake 6
5 Delay Time	5.0 *Master	5.1 *Voice 1	5.2 *Voice 2	5.3 *Voice 3	5.4 *Voice 4	5.5 *Voice 5	5.6 *Voice 6
6 Feedback	6.0 *Master	6.1 Voice 1	6.2 Voice 2	6.3 Voice 3	6.4 Voice 4	6.5 Wake 5	6.6 Wake 6
7 Panning	7.0 *Master	7.1 Voice 1	7.2 Voice 2	7.3 Voice 3	7.4 Voice 4	7.5 Wake 5	7.6 Wake 6
8 Modulation	8.0 Mod L RD	8.1 Mod Rf Rv	8.2 Mod Env L	8.3 Mod Env R	8.4 Mod Lch	8.5 Mod Env 1	8.6 Mod Env 2
9 Patch++	Patch 0	Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6
							Patch 7
							Patch 8
							Patch 9

**Chorus+Rvb** The Chorus effect has six separately adjustable voices — allowing the PCM 81 to sound like a rack of six digital delay boxes. Each voice has its own independently adjustable chorus depth and rate, level control, delay time, feedback and panning control. The 6-voice chorus is in parallel with a plate algorithm, providing two independent stereo effects.

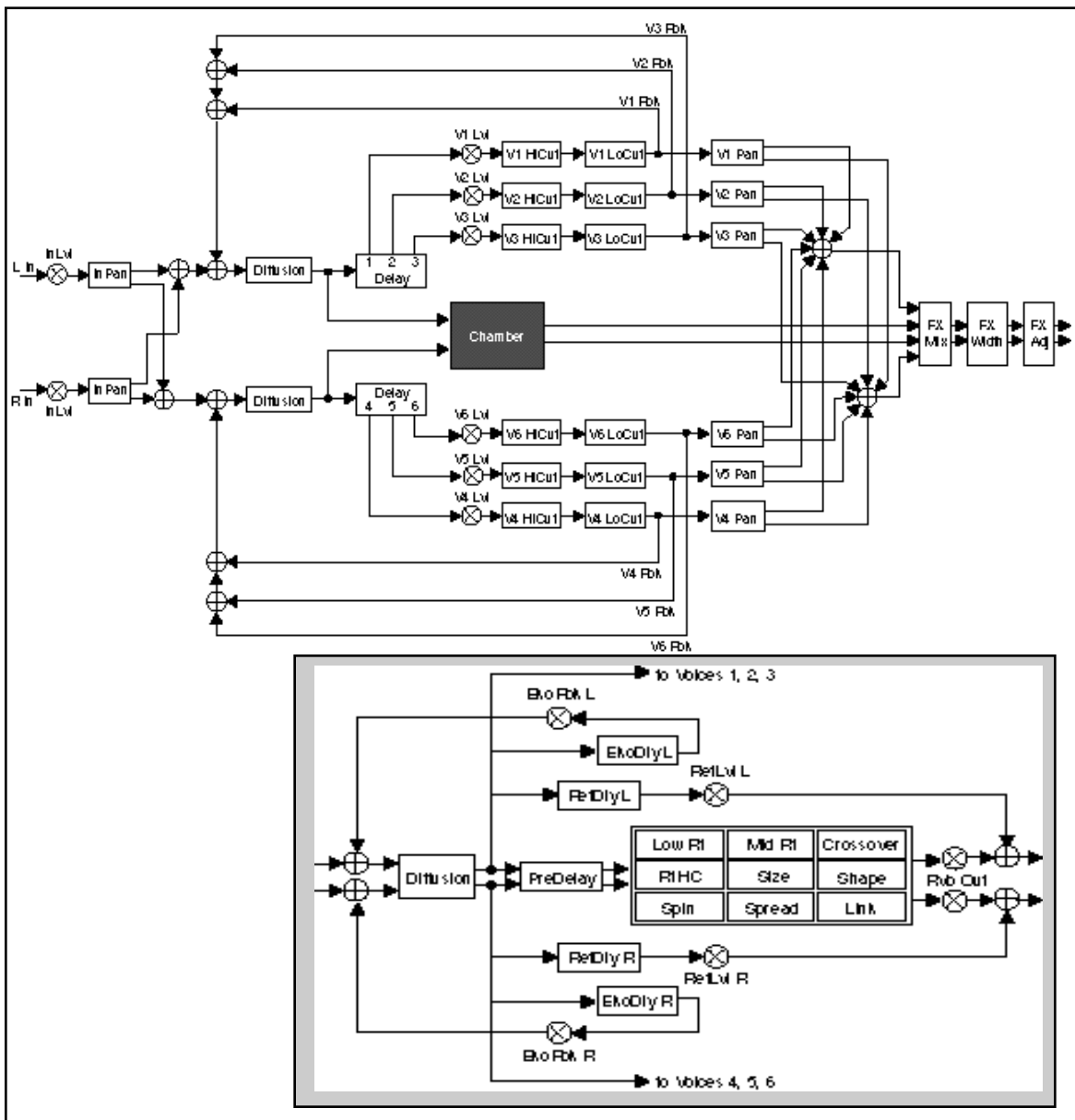
Note that the Diffusion parameter (Rvb Design 2.1) is shared by both the reverb and the chorus effect.



**Chorus+Rvb cont'd.**

0	Control+	0.0 Mn	0.1 FX AdjUn	0.2 RelM Pan	0.3 High Cu	0.4 FX Mn	0.5 FX Mdn	0.6 FX DUST			
1	Env Time	1.0 LowRt	1.1 MidRt	1.2 Crossover	1.3 Rt HC	1.4 Pre Dela.	1.5 RelM DL	1.6 EnvRt DL			
2	Env Design	2.0 6Hz	2.1 Diffusion	2.2 Peak	2.3 Sptn	2.4 Link	2.5 Rvb Out				
3	Level+	3.0 Master	3.1 Voice 1	3.2 Voice 2	3.3 Voice 3	3.4 Voice 4	3.5 Voice 5	3.6 Voice 6			
4	Delay Time	4.0 Master	4.1 Voice 1	4.2 Voice 2	4.3 Voice 3	4.4 Voice 4	4.5 Voice 5	4.6 Voice 6			
5	Chorus+	5.0 Master	5.1 Voice 1	5.2 Voice 2	5.3 Voice 3	5.4 Voice 4	5.5 Voice 5	5.6 Voice 6			
6	Feedback	6.0 Master	6.1 Voice 1	6.2 Voice 2	6.3 Voice 3	6.4 Voice 4	6.5 Voice 5	6.6 Voice 6			
7	Panning	7.0 Master	7.1 Voice 1	7.2 Voice 2	7.3 Voice 3	7.4 Voice 4	7.5 Voice 5	7.6 Voice 6			
8	Modulation	8.0 Mod L FD	8.1 Mod RR Env	8.2 Mod EnvL	8.3 Mod EnvR	8.4 Mod Lrch	8.5 Mod Sml	8.6 Mod Smc			
	Patch++	Patch 0	Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6	Patch 7	Patch 8	Patch 9

**M-Band+Rvb** This effect features six separately adjustable voices, each with its own level control, delay time, low and high frequency filters, feedback and pan controls. The multi-band effect is in parallel with a Chamber effect, providing independent stereo effects. Note that the Diffusion parameter (Rvb Design 2.1) is shared by both the multi-band and reverb effects. Note also that, in this particular algorithm, the diffuser is within the feedback paths of the multi-band voices. This allows you to create filtered echoes that grow more diffuse with each repeat, or to create effects with filtered echoes passing through the reverberator.



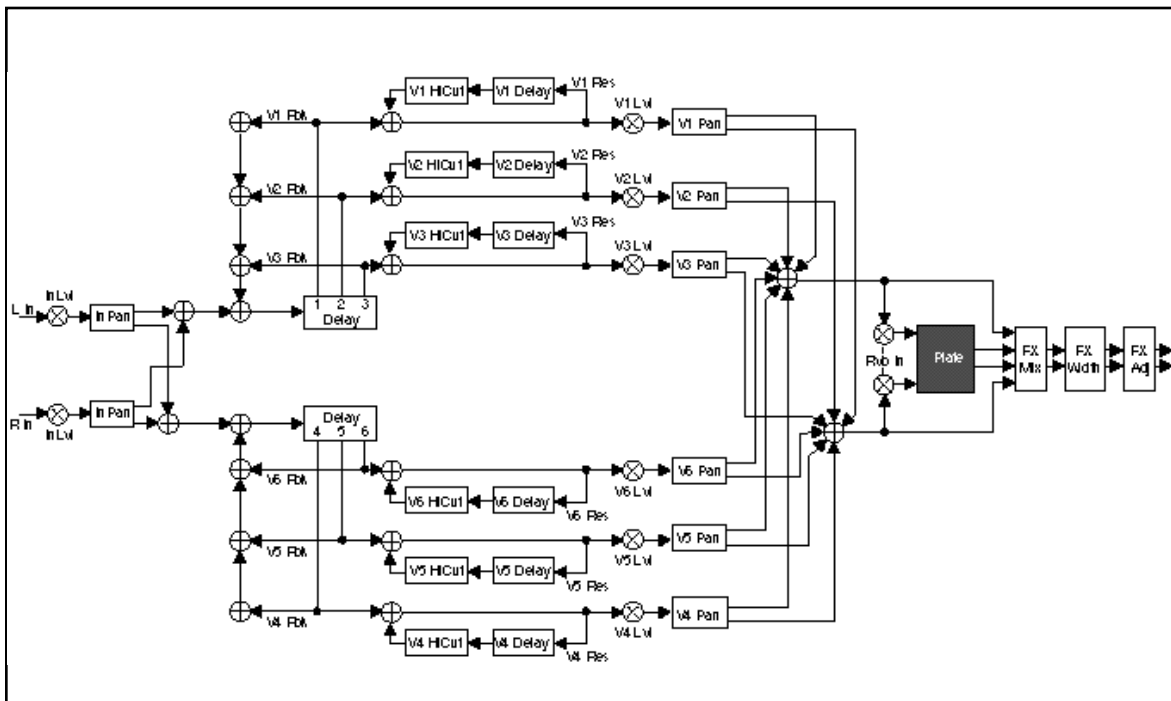
**M-Band+Rvb cont'd.**

0	Control+	0.0 Min	0.1 FXAdjW	0.2 *RelLWIPan	0.3 High Cut	0.4 Voice DI	0.5 FXMix	0.6 FX Width	0.7 *FDJUST
1	Verb Time	1.0 LowR	1.1 MidR	1.2 Crossover	1.3 R HC	1.4 Pre Del.	1.5 *RelLWIDL	1.6 *BioRWDL	1.7 *PnIGid
2	Verb Design	2.0 Size	2.1 Diffusion	2.2 Shape	2.3 Spread	2.4 Spin	2.5 Link	2.6 Verb Width	2.7 *Verb InOut
3	Level+	3.0 Master	3.1 Voice 1	3.2 Voice 2	3.3 Voice 3	3.4 Voice 4			
4	Delay Time	4.0 *Master	4.1 Voice 1	4.2 Voice 2	4.3 Voice 3	4.4 Voice 4			
5	Feedback	5.0 Master	5.1 Voice 1	5.2 Voice 2	5.3 Voice 3	5.4 Voice 4			
6	Panning	6.0 Master	6.1 Voice 1	6.2 Voice 2	6.3 Voice 3	6.4 Voice 4			
7	Modulation	7.0 Mod LFD	7.1 Mod RR Env	7.2 Mod EnvL	7.3 Mod EnvR	7.4 Mod Lch	7.5 Mod Sm 1	7.6 Mod Sm 2	
	Reverb+	Reverb 0	Reverb 1	Reverb 2	Reverb 3	Reverb 4	Reverb 5	Reverb 6	Reverb 7
		Reverb 8	Reverb 9						

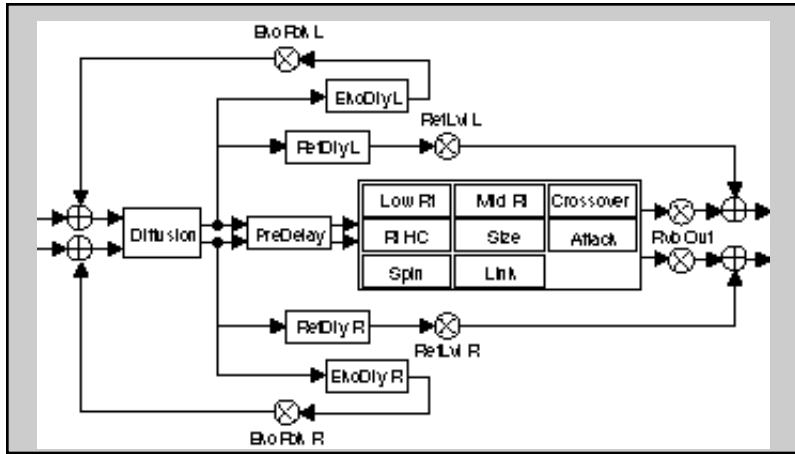
**The Resonant Chord**  
**Algorithms:**  
**Res1>Plate and**  
**Res2>Plate**

The Resonant Chord effects use impulsive energy at the inputs to excite six resonant voices (notes). The level, pitch, duration, and high-frequency cutoff of the overtones for each voice are separately controllable. Each voice can be panned independently. The voices resonate to some degree with any input, but the most effective excitation contains all frequencies, like percussion. Other instruments may give a quality of tonal ambience in which certain notes rise ethereally from the background. The output of the resonator is then fed into a stereo plate reverb effect.

The two algorithms differ in the way pitches are assigned to the resonators. In Res1>Plate, pitches are assigned to the six voices chromatically, in a round-robin. If, for example, MIDI note numbers are used to assign pitch, the resonators will constantly be re-tuned to the pitches of the last six MIDI notes received. (This can produce an effect similar to playing a piano with the sustain pedal depressed.) In Res2>Plate, pitches are assigned to the six resonators diatonically — harmonized with the key, scale, and root of your choice. If MIDI note numbers are used to assign pitch, the resonators will constantly be re-tuned to harmonize with the incoming notes.



**The Resonant Chord Algorithms:  
Res1>Plate and  
Res2>Plate cont'd.**



**Res1>Plate**

0	Control+	0.0	Mix	0.1	FX Rq/Ln	0.2	*RelLvl Pan	0.3	FX Mix	0.4	FX Width	0.5	*RelLvl 6T						
1	Reb Time	1.0	Low Rf	1.1	Mid Rf	1.2	Crossover	1.3	Rf HC	1.4	Pre Dela.	1.5	*RelLvl DL	1.6	EnoRbk DL				
2	Reb Design	2.0	Size	2.1	Diffusion	2.2	Reack	2.3	Spln	2.4	Link	2.5	*Reb h Du						
3	Levle	3.0	*Master	3.1	*Voice 1	3.2	*Voice 2	3.3	*Voice 3	3.4	*Voice 4	3.5	*Voice 5	3.6	*Voice 6				
4	Delay Time	4.0	*Master	4.1	*Voice 1	4.2	*Voice 2	4.3	*Voice 3	4.4	*Voice 4	4.5	*Voice 5	4.6	*Voice 6				
5	Re-assign	5.0	*Master HC	5.1	*Voice 1	5.2	*Voice 2	5.3	*Voice 3	5.4	*Voice 4	5.5	*Voice 5	5.6	*Voice 6				
6	Pitch	6.0	*Assign	6.1	*Tuning	6.2	*Arise	6.3	*Linkon										
7	Panning	7.0	*Master	7.1	*Voice 1	7.2	*Voice 2	7.3	*Voice 3	7.4	*Voice 4	7.5	*Voice 5	7.6	*Voice 6				
8	Modulation	8.0	Mod L FO	8.1	Mod RfR Env	8.2	Mod EnvL	8.3	Mod EnvR	8.4	Mod Lach	8.5	Mod 5m 1	8.6	Mod 5m 2				
	Patch++	Patch 0	Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6	Patch 7	Patch 8	Patch 9								

**Res2>Plate**

0	Control+	0.0	Mix	0.1	FX Rq/Ln	0.2	*RelLvl Pan	0.3	FX Mix	0.4	FX Width	0.5	*RelLvl 6T						
1	Reb Time	1.0	Low Rf	1.1	Mid Rf	1.2	Crossover	1.3	Rf HC	1.4	Pre Dela.	1.5	*RelLvl DL	1.6	EnoRbk DL				
2	Reb Design	2.0	Size	2.1	Diffusion	2.2	Reack	2.3	Spln	2.4	Link	2.5	*Reb h Du						
3	Levle	3.0	*Master	3.1	*Voice 1	3.2	*Voice 2	3.3	*Voice 3	3.4	*Voice 4	3.5	*Voice 5	3.6	*Voice 6				
4	Delay Time	4.0	*Master	4.1	*Voice 1	4.2	*Voice 2	4.3	*Voice 3	4.4	*Voice 4	4.5	*Voice 5	4.6	*Voice 6				
5	Re-assign	5.0	*Master HC	5.1	*Voice 1	5.2	*Voice 2	5.3	*Voice 3	5.4	*Voice 4	5.5	*Voice 5	5.6	*Voice 6				
6	Pitch	6.0	*Master	6.1	*Voice 1	6.2	*Voice 2	6.3	*Voice 3	6.4	*Voice 4	6.5	*Voice 5	6.6	*Voice 6				
7	Panning	7.0	*Master	7.1	*Voice 1	7.2	*Voice 2	7.3	*Voice 3	7.4	*Voice 4	7.5	*Voice 5	7.6	*Voice 6				
8	Modulation	8.0	Mod L FO	8.1	Mod RfR Env	8.2	Mod EnvL	8.3	Mod EnvR	8.4	Mod Lach	8.5	Mod 5m 1	8.6	Mod 5m 2				
	Patch++	Patch 0	Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6	Patch 7	Patch 8	Patch 9								

**The Parameters** PCM 81 parameters are organized into labeled rows within each edit matrix. Although there are similarities among all matrixes, such as having a row of Controls first, and Modulation and Patching rows last, some of the parameters within each row, and some entire rows are unique to specific algorithms.

This section contains descriptions for all PCM 81 parameters, organized alphabetically by row label as follows:

**Chorus**  
**Controls**  
**Delay Time**  
**Feedback/Cross-Feedback**  
**Filters**  
**Glide FX**  
**Levels**  
**Modulation**  
**Panning**  
**Patching**  
**Pitch**  
**Resonance**  
**Rvb Design**  
**Rvb Time**

Individual parameter descriptions within each row are presented, as far as possible, in the order in which they appear from left to right in the edit matrix.

**Chorus** In the Chorus+Rvb algorithm, Row 5 provides parameters which provide rate and depth controls for each of the six chorus voices, as well as master rate and depth controls.

**MstDepth and MstRate**

Two Master Chorus parameters in the first row position allow Chorus, Depth and Rate for all voices to be simultaneously scaled. Each control has a range of 0-200%. Press **Load/★** to toggle between MstDepth and MstRate.

**Voices (Depth and Rate)**

Each voice position has Depth and Rate controls to set the time range (delay) and rate of change across that time range (pitch) of the chorusing effect. (Press **Load/★** to toggle between them.) Depth provides settings of 0-500ms in single millisecond increments. Rate parameter can be set to 0Hz (Off), or to one of one-hundred selections from 0.01-3.50Hz. Depths of 10-30ms combined with Rates as high as 0.50Hz provide subtle chorusing and multivoicing effects. Depths of hundreds of milliseconds combined with higher Rates provide a wide range of pitch shifting effects.

Row 0 of every algorithm contains parameters that provide overall control of both the reverb and voice effects.

## Controls

### Mix

Mix controls the ratio of dry and wet signal present at the PCM 81 outputs. When the PCM 81 is patched into a console or an instrument amplifier through an auxiliary or effects loop, this control should always be set to 100% wet. (Control Mode 1.1 allows you to select a global Mix setting.) When an instrument is plugged directly into the PCM 81, a setting between 45 and 65% is a good starting point.

### FX Adjust

FX Adjust controls the level of the processed (wet) signal before it is mixed with the dry signal (if any) and sent to the audio outputs. The range of this parameter is +12dB to -73dB, and Off (wet output muted.) FX Adjust is inactive when the PCM 81 is in Bypass mode, so it is a good idea to compare the levels with Bypass on and off while editing this parameter. Adjust it until the audio level sounds the same with bypass on and off.

### InLvl L&R, InPan L&R

InLvl L and InLvl R control the level and phase of the unprocessed dry signal into the effects. The range of these parameters is from Full (0dB) Phase Normal, to -85dB Phase Normal, Off, -85dB Phase Inverted to Full (0dB) Phase Inverted.

InPan L and InPan R control the location of the left and right input audio as a stereo image to the effect's audio input. Unmodified stereo imaging to the effects can be achieved by setting InPan L hard left (50L) and InPan R hard right (50R). Changing InPanL to any other value (49L to 50R), takes the left audio input and presents it as a panned source to the effect's left and right input. Changing InPan R to any value other than 50R, does the same for the right audio input.

### High Cut

High Cut sets the high frequency cutoff of a low-pass filter. This parameter affects both channels.

### Voice Dif

This parameter is available in all of the 4-Voice Algorithms, and in Glide>Hall. It controls the density of echoes in the delay voices, independently of Reverb diffusion. High levels of diffusion thicken, or smear, the echoes. This is most noticeable on material with sharp transients.

### FX Mix

FX Mix controls the balance of reverb and non-reverb effects. (Non-reverb effects are those processed as "voices" controlled by level, delay and feedback parameters generally found in rows 3-7.) The range of this control is 0 to 100%.

**Controls cont'd. FX Width**

FX Width can be thought of as an extension of typical mono to stereo imaging controls. The range of this parameter is -360 to +360, in single digit increments. Values of -360, 0, or +360 cause the effect's audio output to be mono. Values of -315 and +45 cause the output to be normal left/right stereo. Values of -45 and +315 cause "swapped", or right/left stereo.

Value	Display Label	Description
360	MONO	Phase Normal Mono
315	R, L	Phase Normal R/L stereo
270	R-L, L-R	R-L, L-R surround
225	STEREO INV	Phase Inverted L/R Stereo
180	MONO INV	Phase Inverted Mono
135	R, L INV	Phase Inverted R/L Stereo
90	L-R, R-L	L-R, R-L surround
45	STEREO	Phase Normal L/R stereo
0	MONO	Phase Normal Mono
-45	R,L	Phase Normal R/L Stereo
-90	R-L, L-R	Phase Inverted R-L, L-R surround
-135	STEREO INV	Phase Inverted L/R Stereo
-180	MONO INV	Phase Inverted Mono
-225	R, L INV	Phase Inverted R/L Stereo
-270	L-R, R-L	L-R, R-L surround
-315	STEREO	Phase Normal L/R Stereo
-360	MONO	Phase Normal Mono

The large and duplicated range of this parameter allows smooth glides from any Left/Right Mix, Phase, Mono/Stereo image point to any other. These are on the output of the FX Mix. Of particular interest are: 0 MONO, 45 STEREO, and 90 L-R, R-L (surround channel). This parameter can be changed in real-time for fascinating spatial effects.

**ADJUST**

This control allows you to specify the range of control of the ADJUST knob when it is patched to a parameter. Low and High settings are available, allowing you to select full range (0-127) for those parameters where it is appropriate and to limit the range in other cases. You can, for example set the Low value to 0, and the High value to 1, to have ADJUST behave as an on/off control. Many of the presets take advantage of this, and have ADJUST range limits which are appropriate to the specific parameter.

In each algorithm, the Delay Time row contains parameters for delay settings of each voice, as well as master delay parameters for all voices.

## Delay Time

### Master, GldResp, GldRange, Clear

Press **Load/★** to cycle through selections: **Master**, **GldResp**, **GldRange** and **Clear**.

#### Master

This control allows you to simultaneously change the delay times of all voices in the effect. The available range is from 0-200%. This provides a simple way to expand or close in all of the voice delay times. If a voice's delay time is set to 500ms, changing the setting of the Master parameter to 200% will change the delay time to 1000ms. Setting this parameter to 0% is an easy way to set all the delays to 0 from a single control.

#### GldResp

In all of the 4-Voice algorithms, and in the Chorus+Rvb algorithm, this parameter controls the response of the glide. The range is 0-100, with a typical default setting of 50. This setting provides a good, real-time glide. Changing a delay time from 0 to 100ms with a glide response of 0, causes the glide delays to take a minute or more, allowing subtle changes in delay times and echoes with no noticeable pitch shifting. A glide response of 100 is ultrafast, causing high speed pitch shift, and even chirping effects with large delay time changes.

#### GldRange

In all of the 4-Voice algorithms, and in the Chorus+Rvb algorithm, this parameter controls the range over which changes to the voice delay time will be performed as a glide. For example, if GldRange is set to 100ms, and the PstDly L and R times are varied over a range of 100ms, those delays will glide smoothly from one delay time to another. Suddenly changing the delay time to a greater value, such as 200ms, will cause the audio delay to change instantly to the new time without gliding. The available range is 0-1.365sec, with 0 causing all delay time changes to be "instantaneous", and 1.365 causing all delay time changes to glide.

#### Clear

This is an On/Off control which allows you to instantly clear all of the audio in the delay voices. A typical use for this control might be to patch it to a footswitch to allow one tap removal of all old audio from the delays to start fresh. While Clear is on, no new audio will pass through any delay voice set above 1ms.

**Delay Time cont'd. Voices**

These parameters set the delay time, in seconds, for each individual voice. The range for each algorithm is shown below.

You can set and display delay values in units of time, or with tempo values. Press **Up** and **Tempo** simultaneously to toggle between these two options. When time units are selected, delay times are set and displayed in milliseconds (From 0ms to the maximum available delay for that parameter). When tempo values are selected, delay values are set and displayed as a ratio of echoes to beats (from 24:1 to 1:24). This will automatically synchronize the delay to the current tempo (MIDI, Internal, or TAP—see *Tempo Mode* in Chapter 2.) For example, a delay setting of 1:2 (1 echo for every 2 beats) will produce half-note delay rhythms synchronized to the current tempo.

<b>Algorithm</b>	<b>Delay Time</b>
Concert Hall	0-1.365 sec
Plate	0-1.365 sec
Chamber	0-1.365 sec
Inverse	0-1.365 sec
Infinite	0-1.365 sec
Glide>Hall	0-10.581 sec
Chorus+Rvb	0-1.365 sec
M-Band+Rvb	0-10.922 sec
Res 1>Plate	0-9.557 sec
Res 2>Plate	0-9.557 sec

When display of values in BPM has been selected, these are set as fractions of a beat. The smallest fraction is 1/24th beat. Changing these values live can produce a variety of pitch shift, and flange effects. Delay times can also be tempo modulated.

Parameters in this row provide level and phase setting for feedback of individual voices, as well as a master feedback parameter for all voices. In the Glide>Hall algorithm, a duplicate set of parameters is included for cross-feedback.

## **Feedback/ Cross Feedback**

### **Master**

A Master Feedback control is available in effects which have a feedback level control for each voice. It allows the feedback level for all voices to be simultaneously adjusted over a range of 0-100%.

### **Voices**

In the 4-Voice algorithms, four voice controls are provided for adjustment of the level and polarity of signals recirculated back to the input of individual voices. Note that, as Voices 1 and 2 and Voices 3 and 4 are paired, caution should be observed not to create feedback within a pair which exceeds more than 100%.

In the Chorus+Rvb and the M-Band+Rvb algorithms, six voice parameters control the feedback level of the voice delays. Voices 1, 2, and 3 Fbk control the individual voice feedback levels from the left channel voice delay outputs to the left channel delay feedback input. Voices 4, 5, and 6 Fbk control the individual voice feedback levels from the right channel voice delay outputs to the right channel delay feedback input.

The range of these parameters is from +100% down to 1% (Phase Normal feedback), 0% (Off), -1% up to -100% (Phase Inverted feedback).

In the Glide>Hall algorithm, the Feedback row provides both Feedback and Cross Feedback parameters as described below. Press **Load/★** at each row position to toggle between Feedback and Cross Feedback parameters.

### **Mstr Fbk, Mstr X-Fbk**

These parameters allow the feedback and cross feedback levels for all voices to be simultaneously scaled over a range of 0-100%.

### **Voices 1-6 Fbk, X-Fbk**

These parameters control the feedback and cross feedback levels and the polarity of the voice delays.

Voices 1, 2, and 3 Fbk control the individual voice feedback levels from the left channel voice delay outputs to the left channel delay feedback input.

Voices 1, 2, and 3 X-Fbk control the individual voice feedback levels from the left channel voice delay outputs to the right channel delay feedback input.

Voices 4, 5, and 6 Fbk control the individual voice feedback levels from the right channel voice delay outputs to the right channel delay feedback input.

Voices 4, 5, and 6 X-Fbk control the individual voice feedback levels from the right channel voice delay outputs to the left channel delay feedback input. The range of these parameters is from +100% down to 1% (Phase Normal feedback), 0% (off), -1% up to -100% (Phase Inverted feedback).

The sum of all Fbk and X-Fbk values for each channel should be less than 100%.

**Filters** In the M-Band+Rvb algorithm, Row 5 contains parameters for cut-off frequencies of low and high cut filters for each voice, as well as master low and high cut controls for all voices.

#### **Mstr LC/HC**

Two master controls are provided in the first row position. Press **Load/★** to toggle between them. The High Cut Master allows simultaneous adjustment of the high frequency cut-off points for the filters on all voices in single semitone increments up or down from the individual voice settings. The Low Cut Master allows simultaneous adjustment of the low frequency cut off points for the filters on all voices in single semitone increments up or down from the individual voice settings.

#### **V1-V6 LoCut and HiCut**

The low cut and high cut filters operate at 12dB/octave. There are individual low cut and high cut filters for each of six voices. The ranges of the filter cut-off points are from 20-20,000Hz at approximately semitone intervals across the entire range for a total of 121 cut off points. The low cut filters have an OFF position just below 20Hz. The high cut filters have an OFF position just above 20,000Hz.

**Glide FX** In the Glide>Hall algorithm, Row 3 contains level, delay and feedback parameters for a pair of stereo gliding delays that precede the 6-voice delay.

#### **Gld Lvl**

Gld Lvl controls the stereo output level from all gliding delays (A, B, Left and Right). It has a range of Full (0dB) down to -85dB and Off.

#### **A Left (A Lvl L and A Dly L)**

These parameters control the output level and delay time of the left channel A Glide Delay. A Lvl L has a range of Full (+0dB), Phase Normal down to -85dB, Phase Normal, OFF, -85dB Phase Inverted up to Full (+0dB), Phase Inverted. A Dly L has a range of 0.0 milliseconds to 42.0 milliseconds in 0.1 millisecond units.

#### **A Right (A Lvl R and A Dly R)**

These parameters control the output level and delay time of the right channel A Glide Delay. They have the same ranges as A Left.

#### **B Left (B Lvl L and B Dly L)**

These parameters control the output level and delay time of the left channel B Glide Delay. They have the same ranges as A Left.

#### **B Right (B Lvl R and B Dly R)**

These parameters control the output level and delay time of the right channel B Glide Delay. They have the same ranges as A Left.

#### **Fbk L/R**

These parameters control the feedback levels of the corresponding left and right channel Glide Delays. The range of these parameters is from +100% down to 1% (Phase Normal feedback), 0% (off), -1% up to -100% (Phase Inverted feedback).

**X-Fbk L/R**

These parameters control the corresponding cross feedback levels of the left and right channel glide delays. Specifically, X-Fbk L controls the feedback from the left channel A+B glide delay output to the right channel glide delay feedback input. X-Fbk R controls the feedback from the right channel A+B glide delay output to the left channel glide delay feedback input. The range of these parameters is from +100% down to 1% (Phase Normal feedback), 0% (off), -1% up to -100% (Phase Inverted feedback).

**Glide FX cont'd.**

In each algorithm, the Levels row contains parameters for the phase and level of each voice, as well as a master level parameter for all voices.

**Levels****Master**

The Master Level controls the overall level in effects which have more than one voice. This permits simultaneous changes in all voice levels without altering the balance between individual voices.

In the Res1>Plate and Res2>Plate algorithms, the Master parameter contains two master parameters for the six resonator voices: Mstr Lvl and Mstr Fbk. (**Load/★** will toggle between them.) Mstr Lvl is a master level control for all of the resonator voices, as described above. Use care when adjusting this value. The resonators will put out high levels if the input source has any components that are close in pitch to the resonator tunings. In general, this control should be set to 0dB or less.

The Mstr Fbk control allows the feedback levels for all voices to be scaled over a range of 0-100%.

**Voices**

Individual controls are provided for each Voice Level. Voices can be turned completely off, full on, or set to any point between, in 1dB increments. There are as many voice level controls as there are voices in an effect.

In the Res1>Plate and Res2>Plate algorithms, positions 1-6 contain both level and feedback parameters for each of the six resonator voices. (**Load/★** will toggle between them.) Lvl controls the individual level and polarity for each voice. Voices can be turned completely off, full on, or set to any point in between in 1dB increments. Use care when adjusting this value. The resonators will put out high levels if the input source has any components that are close in pitch to the resonator tunings. In general, this control should be set to a value less than Full.

The Fbk controls set the feedback level and polarity of each voice. Voices 1-3 Fbk control the feedback levels from the left channel delay voice outputs to the left channel delay voice inputs. Voices 4-6 Fbk control the feedback levels from the right channel delay voice outputs to the right channel delay voice inputs. The sum of the values for either side should be less than 100%. (Disregard the negative sign when adding values.)

**Modulation** The Modulation row, which is the same for every algorithm, contains the parameters for the PCM 81's internal modulation sources. Use the Patch row to assign these modulators to any PCM 81 effect parameter.

**Mod: LFO**

Four parameters are available: **Shape**, **P Width**, **Depth**, and **Rate**.

**Shape** allows you to select the wave shape which will be used when "LFO" is selected as a patch Source. The choices are:

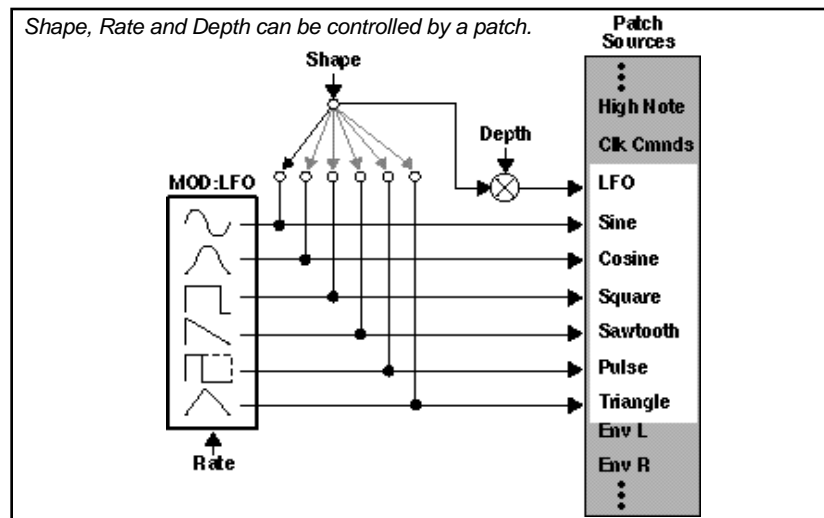
- Sine
- Cosine
- Square
- Sawtooth
- Pulse
- Triangle

**P Width** determines the proportion of each pulse wave cycle for which the LFO is on (1-99%). For example, setting **P Width** to 50% means that the LFO is on for half of its cycle. The effect of this control will only be heard if you are using the Pulse shape.

**Depth** scales the output of the LFO from 0 to 100%.

**Rate** sets the speed (0-25Hz) at which the LFO cycles. It can be set in time values (such as 1.5Hz) or in tempo values (such as 3:2 cycles per beat). Simultaneously pressing **Up** and **Tempo** will toggle these two display options.

Note: The PCM 81 allows six LFO shapes (sine, cosine, sawtooth, triangle, square and pulse) to be selected as patch Sources, as well as the LFO itself. All of these are generated by a single LFO, and are controlled by the single **Rate** control. When "LFO" is selected as a patch Source, the output will be the **Shape** selected here. The amplitude of the LFO output is controlled by **Depth**. Both Shape and Depth are available as patch Destinations and can be controlled externally. Shape, Pulse Width, Depth, and Rate are all available as patch Destination parameters, and can be controlled externally. See *Patching* in Chapter 2.



**Mod: AR Env**

This envelope generator's output, when turned on, will go from 0 to 127. How quickly it goes from 0 to 127 is determined by the setting of **Attack** (0-10 seconds). Once the envelope generator has reached 127, it remains there as long as it is turned on. When it is turned off, it goes from 127 to 0, at the rate determined by **Release** (0 to 10 seconds).

**T Src** allows you to select a Source to turn the envelope generator on and off.

**T Lvl** allows you to select a specific threshold value which the **T Src** must reach to turn the envelope generator on.

The **Mode** parameter allows you to determine the behavior of the envelope controller in relation to the threshold value. Four settings are available: **Repeat**, **One Shot**, **Retrigger**, and **Off**.

- Repeat** As long as the threshold source remains at or above **T Lvl**, the envelope cycles from attack to release. If A=R, the output of the envelope is a triangle wave.
- One Shot** Once **T Lvl** is reached, the envelope will go through its entire attack cycle. Once the attack cycle is completed, if Source value is *below* **T Lvl**, the envelope will immediately fall at the specified **Release** rate. The envelope will go through its entire release cycle, even if the source subsequently rises above **T Lvl**. If the Source value is *at or above* **T Lvl**, the envelope will remain at 127 until the **T Src** falls.
- Retrigger** As long as the level is at or above **T Lvl**, the envelope will go through its attack cycle. If the level falls below **T Lvl** before attack is completed, the envelope immediately begins to release. Likewise, if the **T Lvl** is crossed again before the release is completed, the attack cycle will begin again.
- Off** This control turns the AR envelope off (and frees up processor time). To optimize PCM 81 real-time response, set AR Env to **Off** when it is not being used.

**Mod: Env L and Mod: Env R**

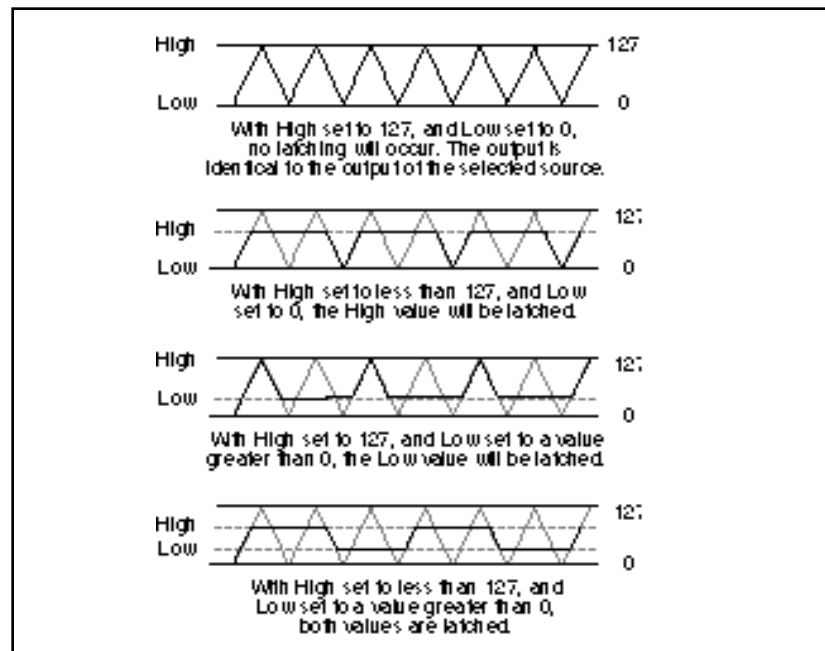
These are left and right input signal envelope followers. The only available parameter is **Release** which is set in milliseconds. This control allows you to specify the release rate (0-10 seconds) when the input level drops.

**Modulation cont'd.**

**Modulation cont'd. Mod: Latch**

The latch is a very flexible modulation source. It can be used to do such things as derive a switch from a continuous "return to zero" source (like MIDI After Touch). It can turn a momentary (on/off) footswitch into a latching footswitch (push on/ push off), and it can divide the switching rates of sources in half or thirds.

The latch has three parameters: **Src**, **High** and **Low**. Any patch source can be the **Src** (See Source listing under Patching.) **High** and **Low** are threshold values. The latch works as follows:



There is no output from the latch until the **Src** value falls within the range defined by the settings of **High** and **Low**. While the source value is between these thresholds, the output of the latch is the same value as the source. When the source value reaches or passes either threshold, the output of the latch holds at the limit value until the source value passes through the threshold twice. The latch can be set to hold at either the low threshold, the high threshold, or both. Setting **Low** to 0 disables latching at the low threshold. Setting **High** to 127 disables latching at the high threshold.

See preset **P0 3.0 FSw2 Rotary** as an example.

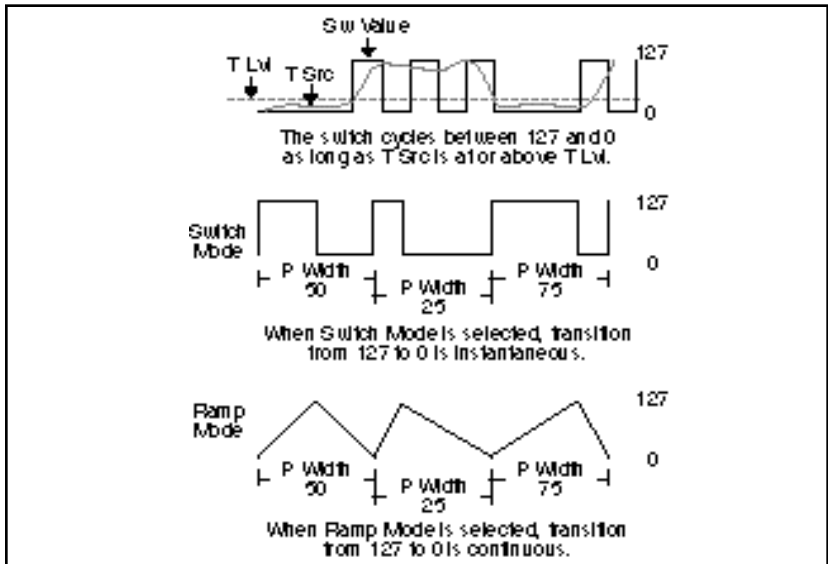
**Mod: Sw 1 and Mod: Sw 2**

These are identical time switches. Each has five parameters: **Rate**, **P Width**, **Mode**, **T Lvl** and **T Src**.

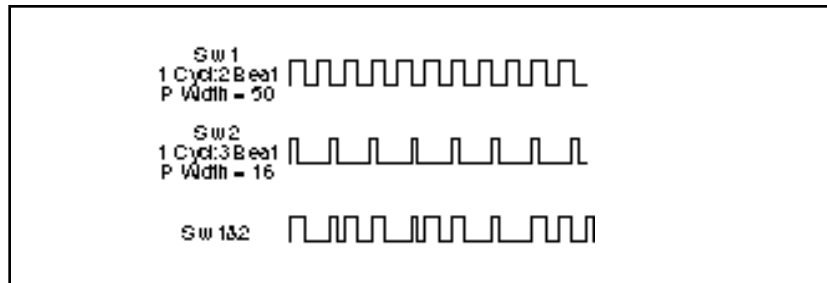
**Modulation cont'd.**

- Rate** sets the speed at which the switch cycles. It can be set in time values (such as 1.5 Hz) or tempo values (such as 3:2 Cycles per Beat). Simultaneously pressing **Up** and **Tempo** will toggle these display options.
- P Width** determines the proportion of each switch cycle during which the switch is on. For example, setting **P Width** to 50% means that the switch is on for one-half of a cycle.
- Mode** determines the "shape" of the switch output. Three settings are available: **Switch**, **Ramp**, and **Off**. When **Switch** is selected, the transition from on to off is instantaneous, i.e. the switch output resembles a pulse wave. When **Ramp** is selected, the transition from on to off is continuous, i.e. the switch output resembles a triangle or sawtooth wave.
- Off** turns the switch off (and frees up processor time). To optimize PCM 81 real-time response, set switch to **Off** when it is not being used.
- T Lvl** sets the threshold value at which the switch will begin to cycle.
- T Src** selects a patch source to drive the switch. The output of the switch is 0 as long as **T Src** is set to a value below **T Lvl**. Once the source value reaches or passes **T Lvl**, the switch will begin to cycle between on (127) and off (0) at the speed set by **Rate**.

Note that both Rate and P Width are available as patch Destinations, allowing them to be dynamically controlled by other patch sources. Switches are reset to the beginning of their cycles whenever **Tap** is pressed.



**Modulation cont'd.** A special, composite output of these switches, called **Sw 1&2** is available as a patch source. The value of **Sw 1&2** alternates between the output of Sw 1 and the output of Sw 2. The alternation occurs on the transition from on to off. Note that both Sw 1 and Sw 2 must be active for the alternation to occur.



**Panning** Parameters in the Panning row provide control of panning of individual effects voices, as well as a master panning parameter for all voices.

**Master**

The Pan Master provides simultaneous control over the panning of all voices in the effect. The range of the Pan Master is 50L (full left) through L1, C (Center), 1R through 50R (full right) for a total of 101 positions. The Master Pan is additive to the individual voice panners such that, if a voice is set to L10, and the Master is set to 20R, the voice will be at 10R.

**Voices**

Individual pan controls are provided to position the output of each voice. The range of each voice is 50L (full left) through 1L, C (Center), 1R through 50 R(full right) for a total of 101 positions.

The last row of each algorithm's edit matrix is the Patch row. This row provides parameters for creating as many as ten patches in each effect. Each row position (Patch 0-9) has three controls available: **Src**, **Dst**, and **Values**. Press **Load/\*** to cycle among these selections.

## Patching

### Src

Use ADJUST to select any of the sources listed below.

### Dst

Use ADJUST to select any effect parameter except those on the Patch row.

### Values

Use ADJUST to assign Destination values to specific Source values. These assignments are made in pairs, each with a value for the Source and a value for the Destination. For example, the default is two pairs mapped as follows:

- minimum Source value (0) = minimum Destination value
- maximum Source value (127) = maximum Destination value

This gives you a linear relationship between the parameter and the controller. Inverse control is accomplished easily by reversing these settings. As many as eight pairs of Destination/ Source values can be assigned here, providing an exciting new level of dynamic control.

See Chapter 2 for a complete description of the Patching System.

PCM 81 Patch Sources		
Internal	MIDI Controller Numbers	
LFO	(PCM 81 interprets 000 as Bank Select)	070 Sound Var
Sine	001 Mod Wheel	071 Timbre
Cosine	002 Breath	072 Release
Square	003 Ctl 3	073 Attack
Sawtooth	004 Foot Ctl	074 Bright
Pulse	005 PortaTime	075 Sound 6
Triangle	006 Data Entry	076 Sound 7
Env L	007 Volume	077 Sound 8
Env R	008 Balance	078 Sound 9
AR Env	009 Ctl 9	079 Sound 10
Latch	010 Pan	080 General 5
Sw 1	011 Xpression	081 General 6
Sw 2	012 Effect 1	082 General 7
Sw 1 & 2	013 Effect 2	083 General 8
Mono Lvl	014 Ctl 14	084 Porta Ctl
Left Lvl	015 Ctl 15	085 Ctl 85
Right Lvl	016 General 1	...
Footpedal	017 General 2	090 Ctl 90
Foot Sw 1	018 General 3	091 FX1 Depth
Foot Sw 2	019 General 4	092 FX2 Depth
ADJUST	020 Ctl 20	093 FX3 Depth
Tempo	...	094 FX4 Depth
On	031 Ctl 31	095 FX5 Depth
Off	(PCM 81 interprets 032 as Bank Select)	096 Data Inc
	033 Ctl 33	097 Data Dec
	...	098 NRPN LSB
	063 Ctl 63	099 NRPN MSB
	064 Sustain	100 RPN LSB
	065 Porta On	101 RPN MSB
	066 Sostenuto	102 Ctl 102
	067 SoftPedal	...
	068 Legato	119 Ctl 119
	069 Hold 2	
MIDI		
P Bend		
A Touch		
Velocity		
Last Note		
Low Note		
High Note		
Clk Comnds		

**Pitch** The Resonant Chord algorithms: Res1>Plate and Res 2>Plate each have a Pitch row that contains the parameters for setting and controlling the tuning of the effect's resonators. Each set of parameters is presented separately here.

#### **Res 1>Plate Pitch parameters**

The voice resonators take the audio impulse from a delay voice and “resonate” it at a desired pitch. Until a resonator is assigned a pitch, it does not resonate. The Res1 effect is a “round robin” pitch assign in that pitch changes are individually assigned to one of (as many as) six voice resonators. For example, assume that all six voice resonators are active in the newly loaded effect, and the first pitch assigned is A4. Voice 1's resonator will be set to the A4 frequency. Assign B4, and Voice 2's resonator will be set to that pitch. In this example, if we assign four more pitches (D5, G4, A#:Bb4, C5, for example), they will be assigned to the Voice 3, 4, 5, and 6 resonators, respectively. If a seventh pitch is assigned, for example, F3, Voice 1's resonator will be changed from A4 to F3 (the other resonators remain at the last assigned pitches).

#### **Assign**

This parameter has a range of C0 to G10 (128 pitches) to be convenient for MIDI patching. As the value of this parameter is changed, each “next value” is assigned to the next active pitch resonator of the six possible active resonators. An example might be driving this parameter with a slow LFO to achieve an effect something like strumming the strings of an open piano.

#### **Tuning**

This parameter allows you to tune the A440 reference of the box anywhere from 430.0-450.0Hz in 0.1Hz increments.

#### **Active**

This parameter allows you to define the number of active voice resonators. You can choose from 0 (no resonators active) to 6. Setting the value to 1 will limit pitch assignments to Voice 1's resonator. Setting the value to 2 will limit the round robin pitch assign to Voices 1 and 2. A value of 6 uses all the voice resonators.

A handy feature of this parameter is the ability to mute resonators. For example, you can patch this parameter to a footswitch or pedal, allowing you to drive the parameter value down to 0 at any time, causing the resonators to stop playing their last assigned pitches. You can then immediately drive it back up to the desired number of active resonators. This “clears” the pitch assign for the resonators, so they don't go back to playing their previous pitches, but rather wait for the next pitch assign to come along before resonating.

#### **Unison**

This is a simple On/Off control. When set to On, pitch assigns will be sent simultaneously to all active resonators. When set to Off, pitch assigns will operate in the round robin method described earlier. This allows you to set multiple delay times for all Voices and allows a single pitch assign (rather than 6 individual assigns) to set all the resonators to the same pitch. The pitch will ring out at 6 different delay times from the input impulse.

**Res 2>Plate Pitch parameters**

The voice resonators take the audio impulse from a delay voice and “resonate” it at a desired pitch. The Res2 effect is a “interval harmonization” pitch assign in that pitch changes generate interval pitches to be assigned to the six voice resonators.

Position 0 contains all of the pitch parameters which affect the voices in the remaining position in the row. Press **Load/★** to cycle through the selections at position 0: **Assign, Tuning, Active, Key, Scale, Root** and **Rule**.

- Assign** This parameter has a range of C0 to G10 (128 pitches to be convenient for MIDI patching). As the value of this parameter is changed, each voice resonator is set to a pitch at the defined voice’s interval (described later).
- Tuning** This parameter allows you to tune the A440 reference of the box anywhere from 430.0-450.0Hz in 0.1Hz increments.
- Active** This parameter allows you to define the number of voice resonators which are active. You can choose from 0 (no resonators active) to 6. Setting the value to 1 will permit only Voice 1’s resonator to ring out. Setting the value to 2 will permit Voices 1 and 2 to ring out. A value of 6 uses all the voice resonators. A handy feature of this parameter is the ability to mute resonators. For example, you can patch this parameter to a footswitch or pedal, allowing you to drive the parameter value down to 0 at any time, causing the resonators to stop playing their last assigned pitches. You can then immediately drive it back up to the desired number of active resonators. This “clears” the note assign for the resonators, so they don’t go back to playing their previous pitches, but rather wait for the next pitch assign to come along before resonating.
- Key** This parameter establishes a key signature for the pitch selection of the voice intervals. The range of this value is the 12 keys from C, C#, etc., up to B.
- Scale** This parameter establishes one of two scales of semitones: a standard Major scale (intervals of 2, 2, 1, 2, 2, 2, 1 half steps for the scale), or a Harmonic scale (intervals of 2, 1, 2, 2, 1, 3, 1).
- Root** This parameter (with values of 1-7) establishes the root of the scale which, when combined with the scale, defines the mode. For example, with a Major scale selected, roots 1-7 define the modes — Ionian (Major), Dorian, Phrygian, Lydian, Mixolydian, Aeolian (minor) and Locrian.

**Pitch cont'd.**

<b>Pitch cont'd.</b>	<b>Rule</b>	<p>This parameter has four values: Round Down, Round Up, Shift Down, and Shift Up. Its exclusive purpose is to tell the interval harmonizer what to do with out-of-key pitch assignments. The values instruct the interval harmonizer as follows:</p> <p><b>Round Down</b> Take the input pitch assign, round it down a half-step to an in key pitch, then calculate the interval pitch.</p> <p><b>Round Up</b> Take the input pitch assign, round it up a half-step to an in key pitch, then calculate the interval pitch.</p> <p>Both of these generate in-key harmonization.</p> <p><b>Shift Down</b> Take the input pitch assign, round it up a half-step to an in-key pitch, calculate the interval pitch then shift the interval pitch down a half-step.</p> <p><b>Shift Up</b> Take the input pitch assign, round it down a half-step to an in-key pitch, calculate the interval pitch then shift the interval pitch up a half-step.</p> <p>Both of these generate out-of-key harmonization.</p>
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Unison and Octave interval pitches strictly follow the input pitch assign. An example of how to use these follows the explanation of Voice Pitch intervals.

#### **V1-V6 Pitch**

The Res2>Plate effect has an individual pitch interval control for each voice resonator. The range of each control is up or down any scale interval through five octaves and Unison, for a total of 81 possible settings. For example: an interval of up one-fifth reads "+5th", and an interval of up one octave and a fifth reads "+1 Oct +5th". Down one-fifth reads "-5th", down one octave and a fifth reads "-1 Oct -5th".

***An example application of Key, Scale, Root, Rule and Voice Pitch Intervals.*** Pitch cont'd.

Active resonators set to 3.

The Key is C.

The Scale is Major.

The Root is 1. (C Major Ionian)

Voice 1 Pitch is assigned to Unison.

Voice 2 Pitch is assigned to +3rd.

Voice 3 Pitch is assigned to +5th. (Basic triad)

In-key pitch assignment examples:

1. Pitch Assign: C4. Voice 1 will resonate at C4, Voice 2 at E4, and Voice 3 at G4. (Standard chord in the key based on the tonic)
2. Pitch Assign: D4. Voice 1 will resonate at D4, Voice 2 at F4, and Voice 3 at A4. (Standard chord in the key based on the supertonic)
3. Pitch Assign: E4. Voice 1 will resonate at E4, Voice 2 at G4, and Voice 3 at B4. (Standard chord in the key based on the median)
4. Pitch Assign: B4. Voice 1 will resonate at B4, Voice 2 at D5, and Voice 3 at F5. (Standard chord in the key based on the leading tone)

Out-of-key pitch assignment — Pitch Assign: C#:Db4.

Rule: Round Down: Voice 1 will resonate at C#/Db4, Voice 2 at E4, and Voice 3 at G4 diminished

Round Up: Voice 1 will resonate at C#/Db4, Voice 2 at F4, and Voice 3 at A4 augmented

Shift Down: Voice 1 will resonate at C#/Db4, Voice 2 at E4, and Voice 3 at G#:Ab4 minor

Shift Up: Voice 1 will resonate at C#/Db4, Voice 2 at F4, and Voice 3 at G#:Ab4 Major

**Resonance** In the Resonant Chord algorithms: Res1>Plate and Res2>Plate, this row contains high cut filter and resonance controls for each voice, as well as master high cut and resonance controls for all voices.

#### **Mstr Res,Mstr HC**

Position 0 of the Resonance row contains two master parameters for the six resonator voices: Mstr Res and Mstr HC. (Press **Load/★** to toggle between them.) Mstr Res is a master resonance control for all of the resonator voices. It allows the resonance settings for all voices to be scaled over a range of 0-100%. Use care when adjusting this value. The resonators will put out high levels if the input source has any components that are close in pitch to the resonator tunings.

The Mstr HC control allows the high cut filter settings for all voices to be scaled over a range of 0-100%.

#### **V1Res,V1 HiCut-V6 Res,V6 HiCut**

The V1-V6 Res controls set the amount and polarity of feedback for each voice. The HiCut controls set the high frequency cutoff frequency of the 6dB per octave low pass filter for each voice.

**RvbDesign** The Reverb Design row, available in every algorithm, contains parameters that affect the structural aspects of the reverb effect.

#### **Size**

Size sets the rate of build-up of diffusion after the initial period (which is controlled by Diffusion). The Size control changes a reverb sound from very large to very small. Generally, you should set this control to approximate the size of the acoustic space you are trying to create, before adjusting anything else. The size in meters is roughly equal to the longest dimension of the space. Audio is temporarily muted when Size is changed.

#### **Duration**

In the Inverse algorithm, Duration determines the length of time, in milliseconds, which passes before the cutoff in Inverse effects.

#### **Attack**

Attack is provided in the Plate algorithm to set the sharpness of the initial response to an input signal. High settings cause an explosive sound, while low settings cause the sound to build up more slowly with time. Attack only affects the level of sound within the first 50 milliseconds.

#### **Diffusion**

A Diffusion control is provided in all algorithms. It controls the degree to which initial echo density increases over time. High settings of Diffusion result in initial build-up of echo density, and low settings cause low initial build-up. Echo density is also affected by Size; smaller spaces will sound denser. To enhance percussion, use high settings of Diffusion. For clearer and more natural vocals, mixes, and piano music, use low or moderate settings of Diffusion.

### **Shape, Spread**

In the Chamber and Infinite algorithms, Shape and Spread work together to control the overall ambience of the reverberation created by the PCM 81. Shape determines the contour of the reverberation envelope. With Shape all the way down, reverberation builds explosively, and decays quickly. As Shape is advanced, reverberation builds up more slowly and sustains for the time set by Spread. With Shape in the middle, the build-up and sustain of the reverberation envelope emulates a large concert hall (assuming that Spread is at least halfway up, and that Size is suitably large — 30 meters or larger). Low Spread settings result in a rapid onset of reverberation at the beginning of the envelope, with little or no sustain. Higher settings spread out both the buildup and sustain. In the Inverse algorithm, Spread is fixed, and only a Shape control is available.

### **Rvb Design cont'd.**

### **Def**

Available in the Concert Hall and Glide>Hall algorithms, Definition affects the echo density buildup rate during the latter part of the decay period. At 0, the rate is determined by the program material. Raising Definition causes the sound to become choppy — the decrease in density of the echoes creates increasingly distinct, repetitive echo trails.

### **Depth**

Available in the Concert Hall and Glide>Hall algorithms, this control sets the output amplitude envelope, changing the listener's perspective from the front to the rear of the hall.

### **Spin**

Spin affects the movement of the reverberation tail. The object of Spin is to continuously alter the timbre of the reverberant sound. This makes the result more natural, without making the position of instruments unstable. Spin should typically be set to values between 10 and 50. Higher values may make the timbre of piano or guitar unstable.

### **Chorus**

In the Concert Hall and the Glide>Hall algorithms Chorus randomizes delay times and introduces modulation to make reverberation sound less metallic. Increasing Chorus increases the rate of modulation. Because Chorusing can cause pitch variation, this parameter should be set with care when using sources with very little pitch wobble (such as guitar or piano). A good practice is to increase the setting until pitch wobble becomes noticeable, then lower it slightly.

### **Link**

This control is available in all algorithms except Inverse. When Link is set to On, the reverb time (Mid Rt) and Spread scale linearly as the Size control is varied. For some special effects, Mid Rt, Spread and Size can be unlinked.

### **Rvb Width**

In all of the 4-Voice algorithms, this control performs the same function as FX Width, but controls the audio of only the reverb and post delay reverb.

### **Rvb In/Out**

Rvb In and Rvb Out control the level of the audio going into and coming out of the reverb effect. Rvb In has a range of Full (0dB) down to -85dB, and Off. Rvb Out has a range of Full (0dB) down to -24dB, and Off. Rvb Out does not affect the level of the pre-echoes set by the Ref Lvl and Eko Fbk parameters.

**Rvb Time** The Reverb Time row, available in every algorithm, contains parameters that affect the time-based aspects of the reverb effect.

#### **Mid Rt and Low Rt**

Mid Rt sets the reverb time for mid-frequency signals. Because low frequency reverb time (Low Rt) is a multiplier of Mid Rt, Mid Rt acts as a master control for the reverb time.

Low Rt sets the reverb time for low-frequency signals, as a multiplier of the Mid Rt parameter. For example, if Low Rt is set to 2X, and Mid Rt is set to two seconds, the low frequency reverb time will be four seconds. For a natural-sounding hall ambience, we recommend values of 1.5X or less.

#### **Low Slope and Mid Slope**

These parameters are only available in the Inverse algorithm. Low Slope determines the shape of the reverb envelope for low frequencies. When set to 0, the level of low reverb remains unchanged over its Duration, then cuts off abruptly (depending on the amount of diffusion in use). Setting Low Slope above 0 causes the level of low-frequency reverb to rise smoothly from soft to loud until the sound is cut off. The greater the slope, the softer the initial reverberation and the more pronounced its rise. With negative values, the low frequency reverb drops from its initial level to a quieter one before cutoff. The lower the slope, the more pronounced the dropoff.

Mid Slope is similar to Low Slope, but applies to middle and high frequencies. The actual frequencies affected are determined by Crossover.

#### **Crossover**

Crossover sets the frequency at which the transition from Mid Rt to Low Rt takes place. This control should be set at least two octaves higher than the low frequency you want to boost. For example, to boost a signal at 100Hz, set Crossover to 400Hz. (This setting works well for classical music.) Crossover works best around 400Hz for boosting low frequencies, and around 1.5 kHz for cutting low frequencies.

#### **Rt HC**

Rt HC sets the frequency above which a 6dB/octave low-pass filter attenuates the reverberated signal. It does not attenuate the reflections. High frequencies are often rolled off with this parameter, resulting in more natural-sounding reverberation.

#### **Pre Delay**

Pre Delay adjusts an additional time delay between the input of signal and the onset of reverberation. The maximum range is 930ms. This control is not intended to mimic the time delays in natural spaces. In real rooms, the build-up of reverberation is gradual, and the initial time gap is usually relatively short. Natural spaces are best emulated by adjusting Spread for the desired effective pre-delay.

#### **Infinite**

In the Infinite algorithm, this control is provided to turn the infinite effect On or Off.

**RefLvl L&R, RefDly L&R**

These controls provide pre-echoes to the left and right channels. The maximum range for Inverse effects is 800ms. In all other effects it is 1.2 seconds. Press **Load/★** to cycle through the selections.

**EkoFbk L&R, EkoDly L&R**

Available in the Plate, Chamber and Infinite algorithms, these controls provide a pre-echo of 1.2 seconds maximum to the left and right channels, with feedback. Press **Load/★** to cycle through the selections.

**PstMix, PstGld**

In the 4-Voice Algorithms, these controls adjust a pair of delays after the reverb. Press **Load/★** to cycle through the selections: **PstMix**, **Pst Dly L&R**, **GldResp** and **GldRange**.

**Pst Mix**

In the 4-Voice Algorithms, this parameter controls the mix of the reverb audio with the post delayed reverb audio (0-100%).

**Pst Dly L&R**

In the 4-Voice Algorithms, these parameters control the time of the post delay. The available range of 0-682ms can produce a variety of pitch shift, flange, or stutter effects on the reverb audio. (Expanding PCM 81 memory will extend this range to 1365ms.)

**GldResp**

In the 4-Voice Algorithms, this parameter controls the response of the glide of the post delay. The range is 0-100, with a typical default setting of 50. This setting provides a good, real-time glide. Changing a delay time from 0 to 100 ms with a glide response of 0, causes the glide delays to take a minute or more, allowing subtle changes in delay times and echoes with no noticeable pitch shifting. A glide response of 100 is ultrafast, causing high speed pitch shift, and even chirping effects with large delay time changes.

**GldRange**

In the 4-Voice Algorithms, this parameter controls the range over which changes to the post delay time will be performed as a glide. For example, if **GldRange** is set to 100ms, and the **PstDly L** and **R** times are varied over a range of 100ms, those delays will glide smoothly from one delay time to another. Suddenly changing the delay time to a greater value, such as 200ms, will cause the audio delay to change instantly to the new time without gliding. The available range is 0-682ms, with 0 causing all delay time changes to be "instantaneous", and 682 causing all delay time changes to glide.

**Rvb Time cont'd.**

You can set and display delay values in units of time, or with tempo values. Press **Up** and **Tempo** simultaneously to toggle between these two options. When time units are selected, delay times are set and displayed in milliseconds (From 0ms to the maximum available delay for that parameter). When tempo values are selected, delay values are set and displayed as a ratio of echoes to beats (from 24:1 to 1:24). This will automatically synchronize the delay to the current tempo (MIDI, Internal, or TAP— see *Tempo Mode* in Chapter 2.) For example, a delay setting of 1:2 (1 echo for every 2 beats) will produce half-note delay rhythms synchronized to the current tempo.

## The Pitch Algorithms

### About the Pitch Algorithms

The PCM 81 contains 7 Pitch algorithms, each combined with uncompromised Lexicon reverb, to provide a wide range of useful pitch shift effects, ranging from pitch correction to unique special effects.

A Quad>Hall algorithm provides a 4-voice pitch shifter, combined with the PCM 81 Concert Hall reverb. The Dual-Chmb, Dual-Plt, Dual-Inv, Stereo-Chmb and VSO-Chmb algorithms use the PCM 81 Dual FX structure for unparalleled flexibility and ease of use. The Pitch Correct algorithm provides tools specifically designed for correction of previously recorded vocal tracks.

The five Dual FX-type algorithms: Dual-Chmb, Dual-Plt, Dual-Inv, Stereo-Chmb and VSO-Chmb, each contain two independent effect blocks — one of three stereo reverbs and one of three different stereo pitch shift blocks. Each algorithm also has the full set of modulation and patch features of the PCM 81 4-Voice and 6-Voice algorithms.

Configuration and routing control is provided by a dedicated Submixer incorporated into each of these five algorithms. Any combination of reverb and effects blocks can be arranged in any configuration simply by changing the Submixer's controls. As the Submixer has continuously variable controls, you can alter the configurations on the fly — changing, for example, from reverb-into-effects to effects-into-reverb. (Use of the Submixer, which appears as a separate row on the PCM 81 parameter matrix, is described fully after the algorithm and parameter descriptions.) Parameters which are common to the 4-Voice and 6-Voice algorithms are not repeated here.

The Pitch Correct algorithm, designed for use with monophonic vocal sources, is described at the end of this section.

## About Pitch Shifting

Digital pitch shifting requires temporal compression or expansion of audio data and the addition or removal of small audio segments to compensate.

Raising pitch is accomplished by compressing the audio data — making it run faster. As this shortens the audio segment, a copy of a section of the audio being processed is added to fill out the remaining time. To lower the pitch, the signal is expanded, requiring the removal of a section of the audio. These operations are performed repeatedly at high speed, at intervals which vary according to the type of audio signal being processed.

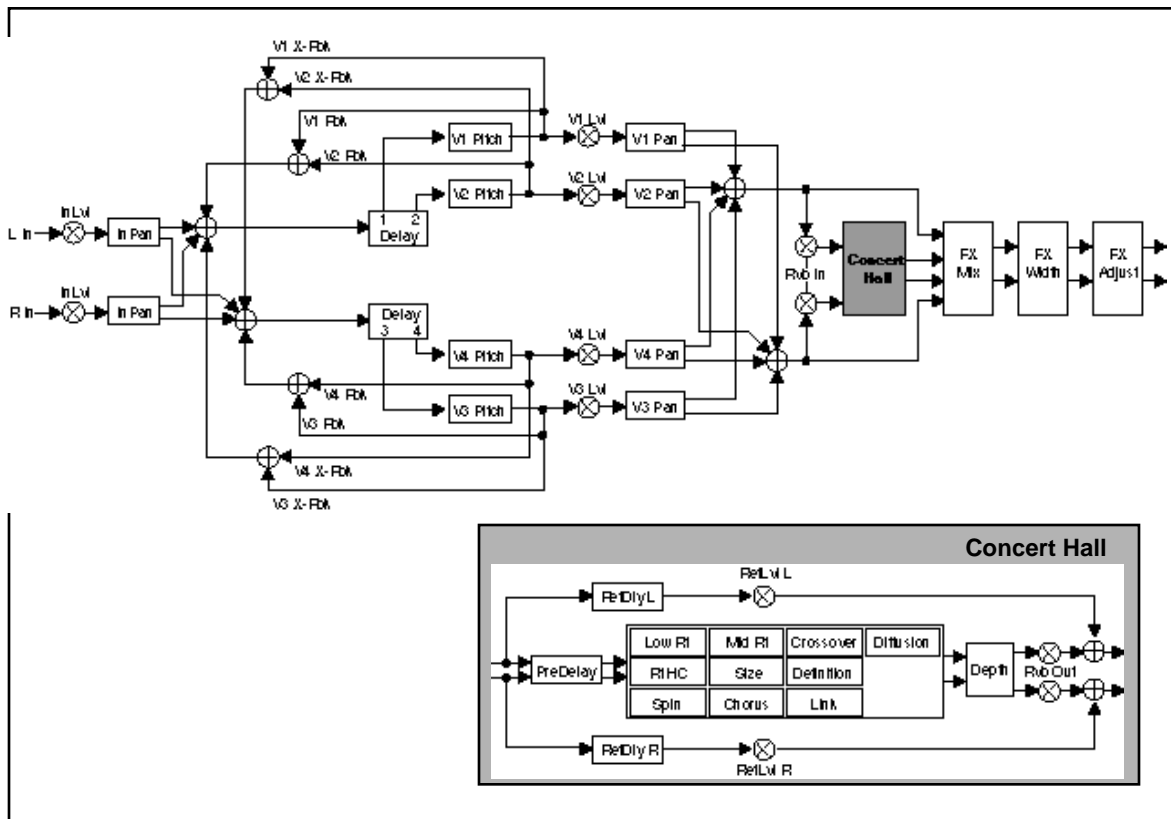
At each point where audio is added or removed, the audio signal must be rejoined, or spliced. If added or removed segments were perfectly correlated with the remaining audio, splicing would be seamless. In reality, these splice points are the source of audible artifacts which can be minimized, but not completely eliminated.

Much of the art of pitch shifting, therefore, is in determining the appropriate intervals for sampling and the optimal points for splicing. As the PCM 81 Pitch algorithms gives you significant control over these operations, you may find it helpful to keep in mind some of the other factors which can affect pitch shift quality:

- **Amount of pitch shifting**  
Large shift intervals require more splicing. As each splice can be the source of an artifact, increasing their number increases the likelihood of audible artifacts.
- **Pitch-complexity of the input material**  
Pure tones have simple waveforms which can be correlated easily for smooth splicing. Pitch shifting of simple tones such as those generated by a solo voice or a monophonic instrument generally produces fewer artifacts than pitch shifting of wide bandwidth material such as piano chords, full orchestra, etc.
- **Low frequency content of the input material**  
Pitch shifting low frequency material requires the addition or removal of larger segments of audio than high frequency material. The increased size of these segments can introduce delays which are heard as artifacts.
- **Transient content of the input material**  
When large amounts of pitch shifting are applied to complex input material with dense percussive transients (drums, strummed mandolin, complex speech, etc.) some of the transients may lose definition or be shifted slightly in time.

**Quad>Hall** The Quad>Hall algorithm, is a 4-voice pitch shifter in series with the Concert Hall reverb. As in the multi-voice algorithms, each voice has its own delay, level, feedback, cross-feedback and panning controls. The maximum delay time for each voice is 1.250 seconds.

In this algorithm, the reverb effect is fixed in position following the pitch shifters, with a final Mix control allowing control over the amount of reverb in the processed sound. Quad>Hall is designed for creating pitch effects, and is an obvious choice for creating sequences of notes or chords, melody fragments, or small amounts of pitch shift for doubling (or quintupling) effects.

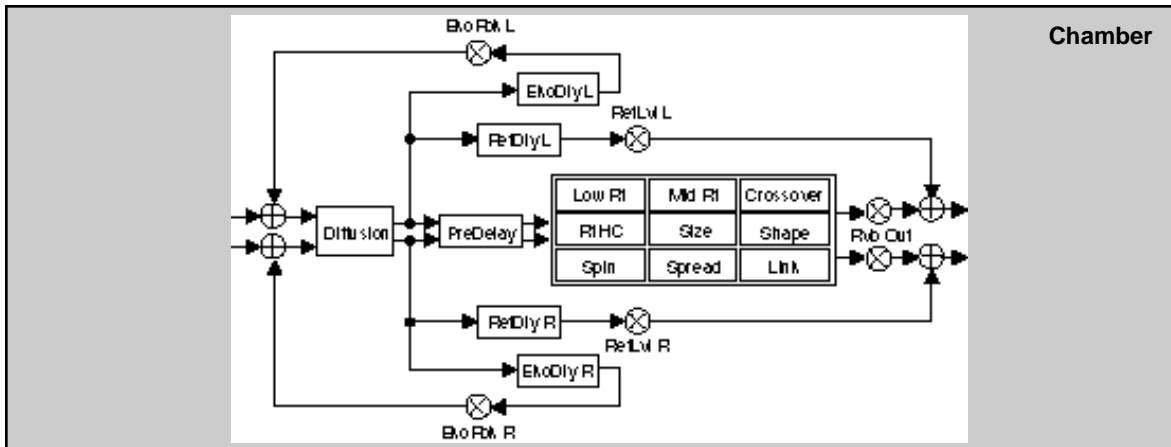
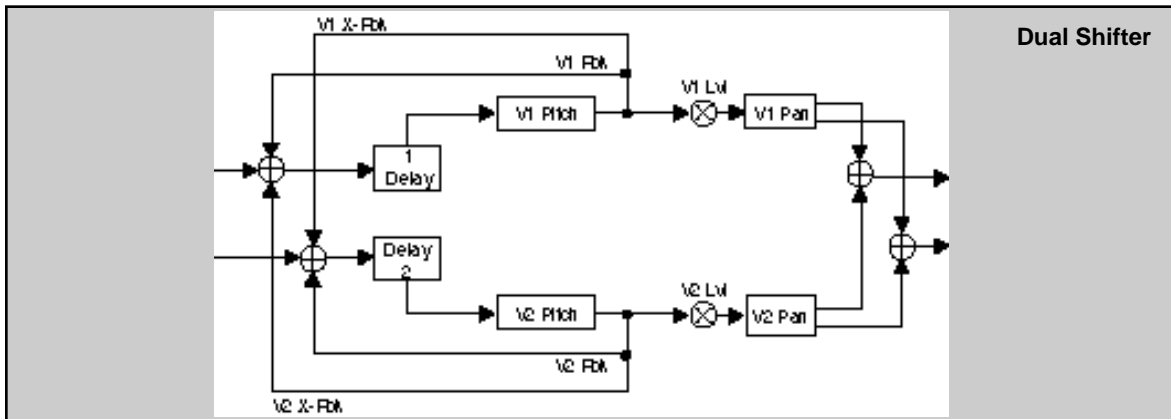


The PCM 81 Concert Hall provides very clean reverberation. It is designed to remain behind the direct sound — adding ambience, but leaving the source unchanged. This effect has a relatively low initial echo density which builds up gradually over time.

0 Control+	0.0 Mix	0.1 L/RAdj	0.2 *L/RPan	0.3 RC Mix	0.4 RC Midh	0.5 *RCU6T				
1 Pvb Time	1.0 LowRi	1.1 Mid Ri	1.2 Crossover	1.3 Ri HC	1.4 Pre Dela.	1.5 *RiLMDL				
2 Pvb Design	2.0 6db	2.1 Diffusion	2.2 Del	2.3 Depth	2.4 6ph	2.5 Chorus	2.6 Link	2.7 *Rub h Dur		
3 Level+	3.0 Master	3.1 Voice1	3.2 Voice2	3.3 Voice3	3.4 Voice4					
4 Delay Time	4.0 *Master	4.1 Voice1	4.2 Voice2	4.3 Voice3	4.4 Voice4					
5 Pitch	5.0 *MasterCen+	5.1 Voice1	5.2 Voice2	5.3 Voice3	5.4 Voice4					
6 Feedback	6.0 *Mtr Rbk	6.1 *V1 Rbk	6.2 *V2 Rbk	6.3 *V3 Rbk	6.4 *V4 Rbk					
7 Panning	7.0 *MtrMidh	7.1 Voice1	7.2 Voice2	7.3 Voice3	7.4 Voice4					
8 Modulation	8.0 *ModLFO	8.1 *ModRR Env	8.2 Mod EnvL	8.3 Mod EnvR	8.4 *ModLach	8.5 *Mod6m1	8.6 *Mod6m2			
9 Patch++	Patch 0	Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6	Patch 7	Patch 8	Patch 9

The following five Pitch FX algorithms: **Dual-Chmb**, **Dual-Pit**, **Dual-Inv**, **Stereo-Chmb** and **VSO-Chmb** are created from 3 Reverb blocks and 3 Pitch Shift Effect blocks. The individual reverb and effect blocks are described below, followed by the diagrams and matrixes and parameter descriptions for the algorithms. Descriptions of the controls common to all of these algorithms is presented in following section *Using the Submixer*.

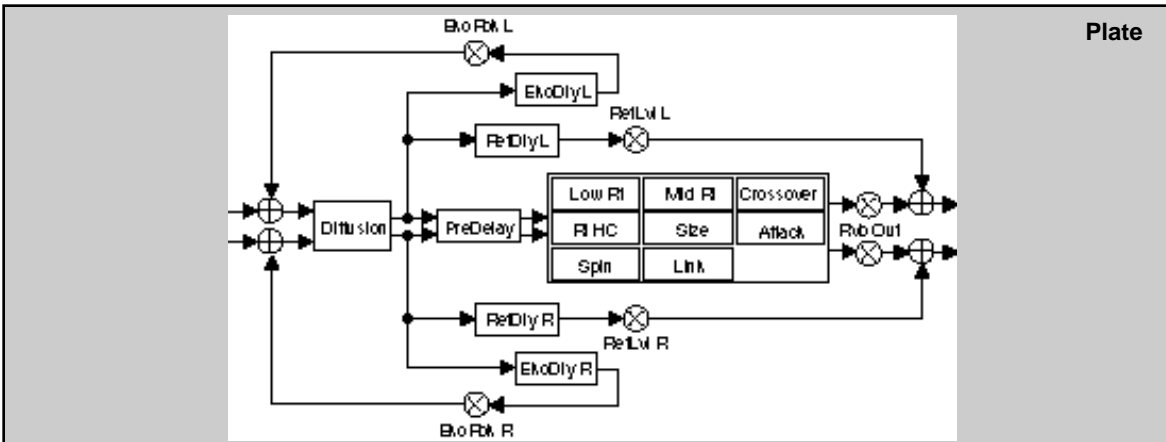
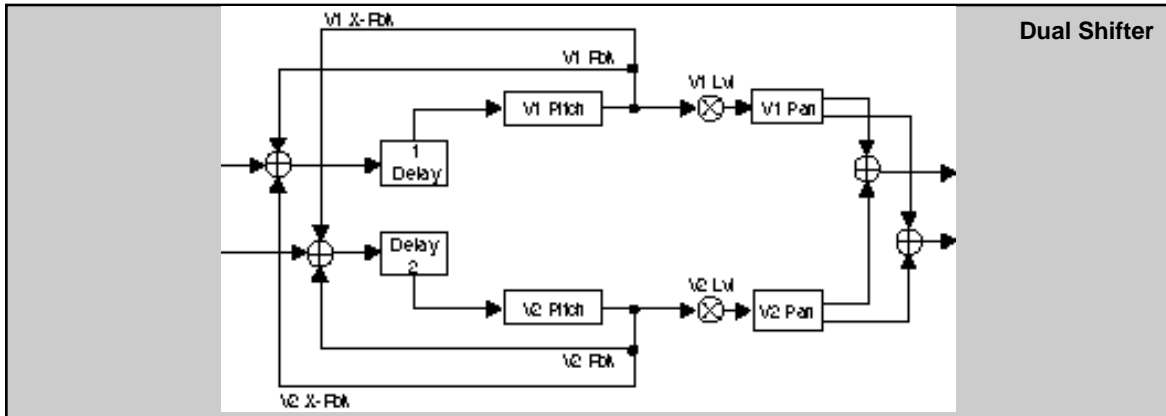
**Dual-Chmb** This algorithm includes a dual pitch shifter combined with the Chamber reverb. The pitch shifter has two voices. Each voice has independent controls for pitch, level, delay, pan, feedback and cross-feedback. The pitch of each voice is adjustable up or down over a 3 octave range with a resolution of 1 cent. Maximum delay time for each voice is 1.250 seconds.



The Chamber reverb produces an even, relatively dimensionless reverberation, with little change in color as the sound decays. The initial diffusion is similar to the Concert Hall algorithm, but the sense of space and size is much less obvious. This characteristic, along with the low color in the decay tail, makes Chamber useful on a wide range of material. It is especially useful on spoken voice, giving a noticeable increase in loudness with very low coloration.

0 Control+	0.0 Mix	0.1 LoudAdj	0.2 *InLvlPan	0.3 *Rc Width	0.4 *ADJUST					
1 Submixer	1.0 *ends	1.1 *Reverb	1.2 *Routing	1.3 *Mix	1.4 *InLvl	1.5 *In W	1.6 *HiCu	1.7 *LoCu	1.8 *OuLvl	1.9 *Ou W
2 *Pre-Time	2.0 *LowR	2.1 *Mid R	2.2 *Crossover	2.3 *Rc HC	2.4 *Pre Dela.	2.5 *RelLWDL	2.6 *RelRkDL			
3 *Pre-D+Ign	3.0 *Size	3.1 *Diffusion	3.2 *Shape	3.3 *Spread	3.4 *Spin	3.5 *Link	3.6 *Pre Out			
4 *Level+	4.0 *Master	4.1 *VoiceL	4.2 *VoiceC							
5 *DelayTime+	5.0 *Master	5.1 *VoiceL	5.2 *VoiceC							
6 *Pitch	6.0 *MasterCent+	6.1 *VoiceL	6.2 *VoiceC							
7 *Feedback	7.0 *Master FBk	7.1 *VoiceL FBk	7.2 *VoiceC FBk							
8 *Panning	8.0 *MasterWidth	8.1 *VoiceL	8.2 *VoiceC							
9 *Modulation	9.0 *ModLFO	9.1 *ModLFR Env	9.2 *ModEnvL	9.3 *ModEnvR	9.4 *ModLch	9.5 *ModSumL	9.6 *ModSumC			
Envelope+	Pach 0	Pach 1	Pach 2	Pach 3	Pach 4	Pach 5	Pach 6	Pach 7	Pach 8	Pach 9

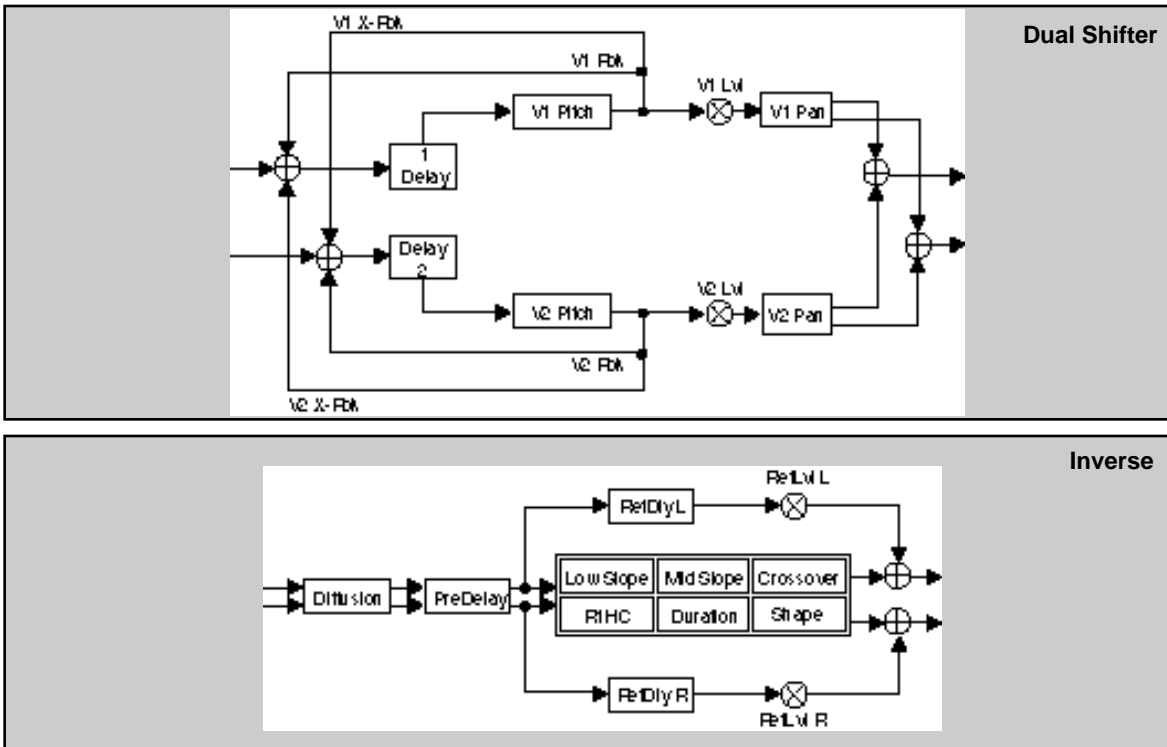
**Dual-Pit** This algorithm includes a dual pitch shifter combined with the Plate reverb. The pitch shifter has two voices. Each voice has independent controls for pitch, level, delay, pan, feedback and cross-feedback. The pitch of each voice is adjustable up or down over a 3 octave range with a resolution of 1 cent. Maximum delay time for each voice is 1.250 seconds.



The Plate reverb mimics the sound of metal plates, with high initial diffusion and a relatively bright sound, making it a good choice for enhancing any type of percussion.

0	Control+	0.0 Min	0.1 LutRdJsh	0.2 *InLutPan	0.3 Fc Width	0.4 *RDJdBT					
1	Subrouter	1.0 Send	1.1 Rsum	1.2 Routing	1.3 *Min	1.4 *InLut	1.5 *In M	1.6 *HiCu	1.7 *LoCu	1.8 *OutLut	1.9 *Out M
2	PubTime	2.0 LowR	2.1 MidR	2.2 Crossover	2.3 Ri HC	2.4 Pre Dela.	2.5 *RiLutDL	2.6 *Ro RmDL			
3	PubDesign	3.0 Size	3.1 Diffusion	3.2 *Pack	3.3 *pin	3.4 Link	3.5 *Pub Out				
4	Level+	4.0 Master	4.1 *Voice	4.2 *VoiceC							
5	DelayTime	5.0 *Master	5.1 *Voice	5.2 *VoiceC							
6	Blush	6.0 *MinCells+	6.1 *Voice	6.2 *VoiceC							
7	Feedback	7.0 *Min Rk	7.1 *Vi Rk	7.2 *Vc Rk							
8	Painting	8.0 *MinWidth	8.1 *Voice 1	8.2 *Voice 2							
9	Modulation	9.0 *ModLFO	9.1 *Mod:PR Env	9.2 *Mod EnvL	9.3 *Mod EnvR	9.4 *ModLch	9.5 *Mod:sm1	9.6 *Mod:sm2			
Envelope+		Envch 0	Envch 1	Envch 2	Envch 3	Envch 4	Envch 5	Envch 6	Envch 7	Envch 8	Envch 9

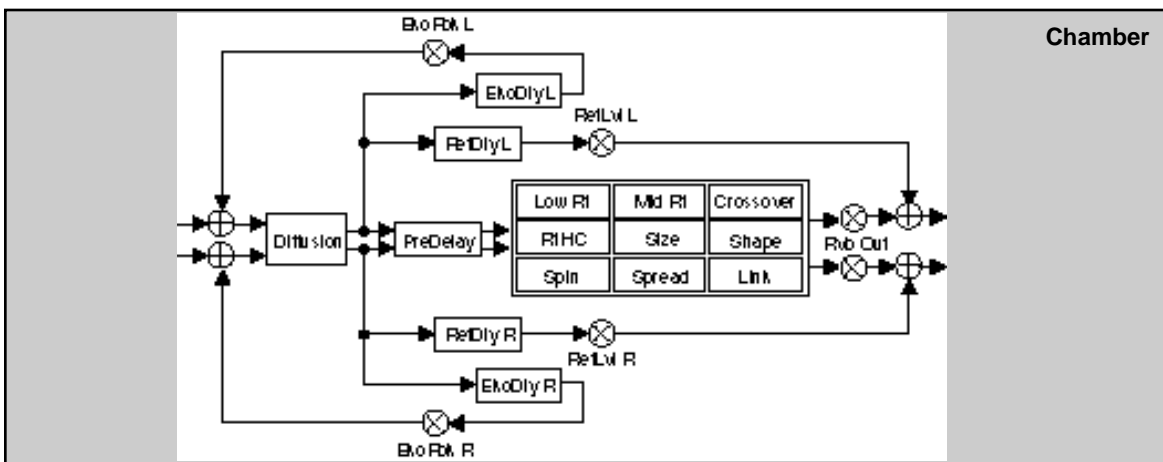
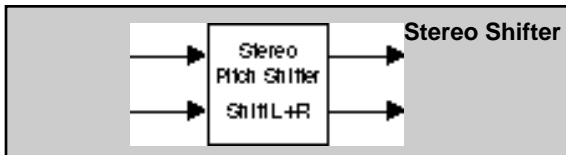
**Dual-Inv** This algorithm includes a dual pitch shifter combined with the Inverse reverb. The pitch shifter has two voices. Each voice has independent controls for pitch, level, delay, pan, feedback and cross-feedback. The pitch of each voice is adjustable up or down over a 3 octave range with a resolution of 1 cent. Maximum delay time for each voice is 1.250 seconds.



Inverse reverb allows you to vary the slope of the initial portion of the reverb envelope. The slope can decay, remain level, or rise over a variable time interval. When the time interval is up, the reverberation cuts off abruptly. The resulting effect is similar to a gate, but is not at all dependent on the level or complexity of the input signal. Slopes are adjustable over a negative, even, or positive slope. Positive slopes create inverse effects, while more even slopes create gated effects. Negative slope values have rather natural reverb tails.

0	Control+	0.0 Min	0.1 LutAdjW	0.2 *InLutPan	0.3 Fc Midh	0.4 *RDJUST					
1	Subtizer	1.0 Sends	1.1 Returns	1.2 Routing	1.3 *Min	1.4 *InLut	1.5 *In M	1.6 *HiCu	1.7 *LoCu	1.8 *OutLut	1.9 *Out M
2	PubTime	2.0 LomSlope	2.1 Mid Slope	2.2 Crossover	2.3 Fc HC	2.4 Pre Dela.	2.5 *PeLutDL				
3	PubDesign	3.0 Duration	3.1 Diffusion	3.2 Shape							
4	Level+	4.0 Master	4.1 VoiceL	4.2 VoiceC							
5	DelayTime	5.0 *Master	5.1 VoiceL	5.2 VoiceC							
6	Pitch	6.0 *MinCents	6.1 VoiceL	6.2 VoiceC							
7	Feedback	7.0 *Min FB	7.1 *Vl FB	7.2 *Vc FB							
8	Routing	8.0 *MinMidh	8.1 VoiceL	8.2 VoiceC							
9	Modulation	9.0 *ModLFO	9.1 *Mod:PR Env	9.2 Mod EnvL	9.3 Mod EnvR	9.4 *ModLach	9.5 *ModSml	9.6 *ModSmC			
	Ensembles	Patch 0	Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6	Patch 7	Patch 8	Patch 9

**Stereo-Chmb** The Stereo-Chmb algorithm is optimized for the best possible shifted audio quality while maintaining the stereo imagery of the source material. Pitch is adjustable up or down over a 3 octave range with a resolution of 1 cent. This effect is combined with the Chamber reverb.



The Chamber reverb produces an even, relatively dimensionless reverberation, with little change in color as the sound decays. The initial diffusion is similar to the Concert Hall algorithm, but the sense of space and size is much less obvious. This characteristic, along with the low color in the decay tail, makes Chamber useful on a wide range of material. It is especially useful on spoken voice, giving a noticeable increase in loudness with very low coloration.

0	Control+	0.0	Mix	0.1	LowAdj	0.2	ReLvlPan	0.3	ReModh	0.4	ReDlyT										
1	Submixer	1.0	ends	1.1	Reums	1.2	Routing	1.3	ReLvl	1.4	ReLvl	1.5	ReLvl	1.6	ReLvl	1.7	ReLvl	1.8	ReLvl	1.9	ReLvl
2	Pre-Time	2.0	LowRf	2.1	Mid Rf	2.2	Crossover	2.3	Rf HC	2.4	Pre Dela.	2.5	ReLvl DL	2.6	ReLvl DL						
3	Pre-Delay	3.0	Size	3.1	Diffusion	3.2	Shape	3.3	Spread	3.4	Spin	3.5	Link	3.6	Reb Out						
4	Pitch	4.0	Low Pitch	4.1	Shift L+R																
5	Modulation	5.0	ModLFO	5.1	ModRFR Env	5.2	ModEnvL	5.3	ModEnvR	5.4	ModLach	5.5	ModSml	5.6	ModSml2						
6	Preset+	Patch 0	Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6	Patch 7	Patch 8	Patch 9										

## **VSO-Chmb**

The VSO-Chmb algorithm provides the same high quality shifter as Stereo-Chmb, designed for on the fly pitch correction of varispeed playback sources. A varispeed parameter allows you to match the varispeed setting of the playback source. The algorithm can match speeds ranging from +55.00% through -35.48% with a resolution of .01%.

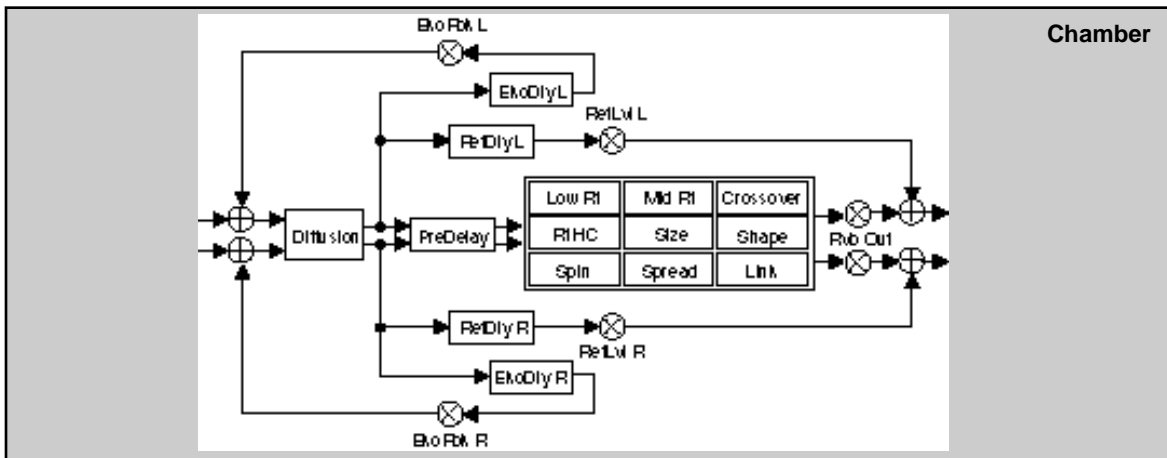
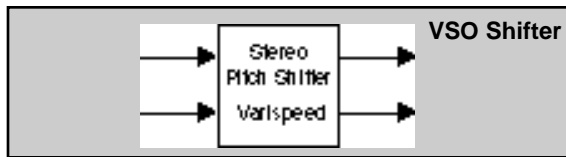
This algorithm is a utility program designed to provide pitch correction of varispeed material. This is a true stereo algorithm which maintains the stereo image of source material. It can be used in applications that require on the fly compression/expansion. Simply match the value of the Varispeed parameter to the varispeed setting of the playback source.

For example, to compress a 30 second spot to 24 seconds, the playback speed must be increased by 20%. The spot now has the correct run length, but increasing the speed created an upward pitch shift of 386 cents. Run the shifted audio through the VSO-Chmb algorithm. Set the Varispeed parameter to +20%. This shifts the pitch down 386 cents. The spot now has a run length of 24 seconds and is at its original pitch.

To expand a 28 second spot to 30 seconds, the playback speed must be decreased by 7.14%. Set both the playback source and the VSO Shifter to -7.14%. The spot will remain at its original pitch and have a run length of 30 seconds.

Like the Stereo-Chmb algorithm, VSO-Chmb is combined with a stereo chamber reverb. This adds considerable flexibility. You can for example, add reverb to the material as it is being pitch corrected (great for voiceovers, etc.).

Note that there is no direct synchronization of the pitch shift with the playback source. The accuracy of the pitch correction is dependent upon the accuracy of the playback source's varispeed mechanism and display.



The Chamber reverb produces an even, relatively dimensionless reverberation, with little change in color as the sound decays. The initial diffusion is similar to the Concert Hall algorithm, but the sense of space and size is much less obvious. This characteristic, along with the low color in the decay tail, makes Chamber useful on a wide range of material. It is especially useful on spoken voice, giving a noticeable increase in loudness with very low coloration.

0 Control+	0.0 Mix	0.1 LowAdj	0.2 *ReLvlPan	0.3 ReLvl	0.4 *ReLvlT					
1 Submixer	1.0 *ends	1.1 *ReLvl	1.2 *ReLvlPan	1.3 *ReLvl	1.4 *ReLvl	1.5 *ReLvl	1.6 *ReLvl	1.7 *ReLvl	1.8 *ReLvl	1.9 *ReLvl
2 Rvb Time	2.0 LowRI	2.1 MidRI	2.2 Crossover	2.3 RIHC	2.4 Pre Delay	2.5 *ReLvlDL	2.6 *ReLvlDL			
3 Rvb Design	3.0 Size	3.1 Diffusion	3.2 Shape	3.3 Spread	3.4 Spin	3.5 Link	3.6 Rvb Out			
4 Pitch	4.0 Low Pitch	4.1 Varispeed								
5 Modulation	5.0 *ModLFO	5.1 *ModLFO Env	5.2 Mod EnvL	5.3 Mod EnvR	5.4 *ModLch	5.5 *ModLch1	5.6 *ModLch2			
Patch++	Patch 0	Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6	Patch 7	Patch 8	Patch 9

## The Pitch Parameters

### MstrCents

This is a master pitch offset control for all voices. The value dialed in here will be added to all voices. This control allows you to transpose pitch voices while keeping the relative interval(s) between them constant.

### MstrScale

This is a master pitch scaling control for all voices. The value of each pitch voice is multiplied by this percentage. This control allows you to shrink or enlarge the relative interval(s) between all voices.

### Low Pitch

This parameter allows you to optimize the pitch shifter for the low frequency content of the input material. Use low values when shifting single note material with low frequencies, polyphonic sources or full bandwidth material. Note that the lower this value, the longer the pitch shifted material is delayed. At the lowest setting, the amount of delay is about 42ms. At the highest setting it is about 12ms.

Use this parameter to minimize the amount of delay inherent in the pitch shift process. If the maximum delay of about 40ms is not a problem, you can simply set this value to its lowest setting. To minimize the delay for single note pitch shifting, set it to a value that matches the pitch of the lowest note in the input material. If you are pitch shifting polyphonic input material you may get better results by setting the value lower than the lowest note in the input material.

Lowest note of some typical sources:

Piano	A0
Bass guitar	E1
Guitar	E2
Trombone	F2
Tenor Sax	Bb2
Vocal: bass	Eb2
Vocal: baritone	Ab2
Vocal: tenor	Db3

### Splice

This parameter allows you to change the number of milliseconds used to crossfade the splices that are constantly occurring whenever pitch is shifted. For most material a value of 4ms is appropriate. Higher values will produce longer/smooth crossfades, but the intelligibility of transients and high frequency content may be affected. Lower values will reproduce transients more accurately and have more high frequency content, but the splices may be more audible.

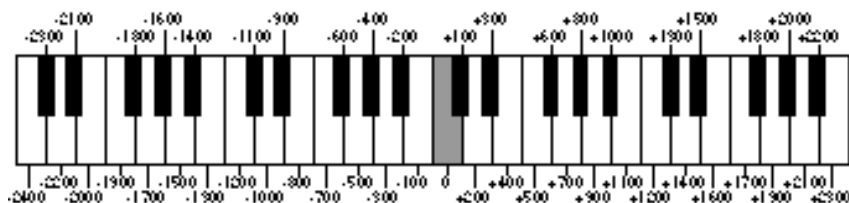
### GldResp

This parameter is similar to the GldResp parameter in the Master Delay parameter of the delay and reverb algorithms. It sets the pitch shifters responsiveness to modulation by internal controllers (LFO, AR, etc.) and MIDI controllers (AfterTouch, Velocity, etc.). It has no effect on the quality of the pitch shifted sound. At the highest setting (100) the pitch will track the modulation source very closely. At lower settings, the pitch will lag behind the modulation source. Use lower settings to smooth out modulation effects.

**Parameter Descriptions  
cont'd.**

**Voice 1-4 (or 1-2)**

These parameters determine the amount of pitch shift for the left and right input channels. In the Quad Shift algorithm, Voices 1 and 2 are left shifts and 3 and 4 are right shifts. In the 2-Voice algorithms, Voice 1 is left shift and Voice 2 is right shift. The value is given in cents (one cent is equal to 1/100th of a semitone — there are 1200 cents per octave, as shown below). The full range of pitch shift is up or down three octaves ( $\pm 3600$  cents). The range is adjustable in increments of a single cent. These values can be modulated smoothly with patch sources to create vibrato, Doppler, whammy bar and other pitch-related special effects.



**Shift L+R**

In the Stereo Shift algorithm, this parameter determines the amount of pitch shift for both the left and right channels. This is a true stereo pitch shifter. The left and right channels are sample synchronous to maintain stereo imaging.

**Tips for Setting  
Pitch Parameters**

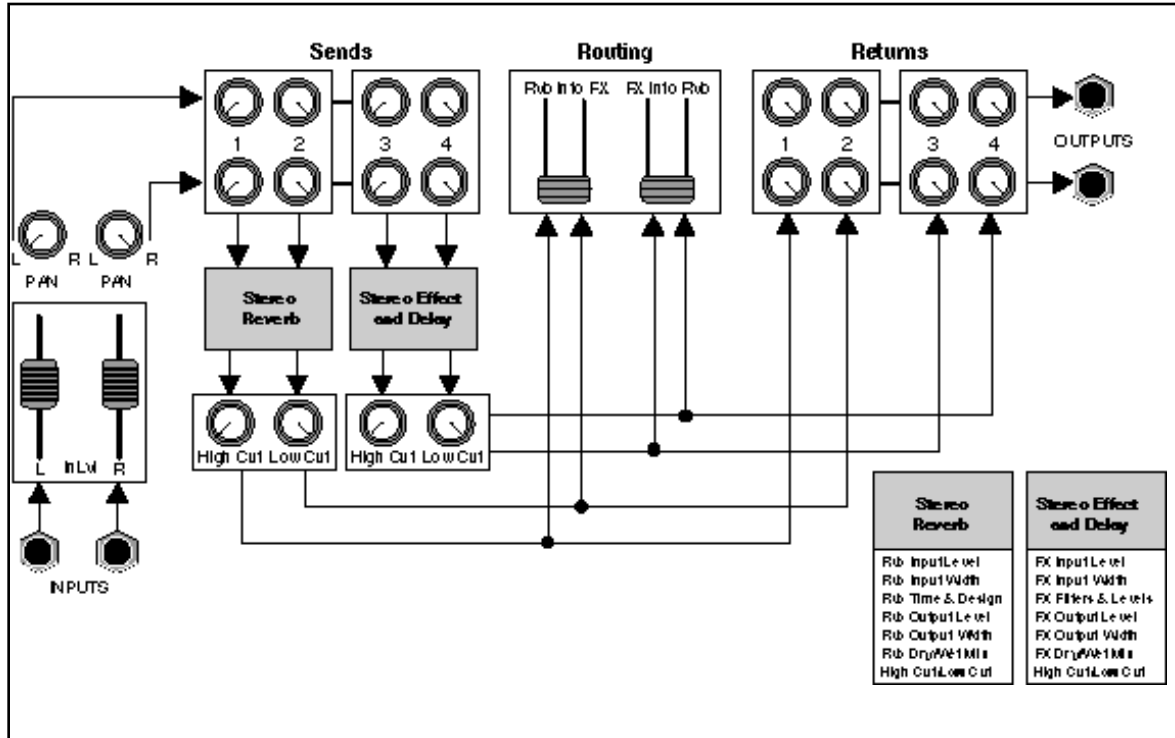
All pitch shifting algorithms will produce some amount of glitches and all pitch shift algorithms have some amount of built in delay. Single note sources shift best. When shifting single notes, Low Pitch should be set at or below the lowest pitch in the input material.

When shifting chords or full bandwidth material, first set the amount of shift. If you need to minimize the delay caused by the shifter, set Low Pitch to the lowest value and slowly increase it to the highest value that doesn't produce unacceptable glitches.

The Splice parameter generally works best for most material when set to 4ms. If the shifted audio sounds a bit harsh, you can smooth it out by raising this value. If there is a loss of intelligibility or rhythmic accuracy in the shifted audio, try setting Splice to a smaller value. As with most effect parameters, you should rely on your ears to find the best settings for a given situation.

Expanded configuration control is available in those Pitch algorithms which have a Submixer incorporated into each algorithm. This Submixer lets you arrange any combination of reverb and effects blocks in any configuration simply by changing the three main Submixer controls: Sends, Returns and Routing. Additional controls are provided for each block: Input Level, Input Width, Output Level, Output Width, High Cut and Low Cut filters, and Wet/Dry Mix.

### Using the Submixer



*The Dual-Chmb, Dual-Plt, Dual-Inv and Stereo-Chmb algorithms contain a complete dual-channel effects mixer.*

The Submixer appears as row 1 on the parameter matrix. The first three parameters in the row are the basic Submixer controls: Sends, Returns, and Routing. Any configuration can be achieved by simply adjusting these three parameters.

1	Submixer	1.0 Sends	1.1 Returns	1.2 Routing	1.3 *Hs	1.4 *hLof	1.5 *h R	1.6 *hCut	1.7 *LoCut	1.8 *Ouf	1.9 *Ou R
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These parameters provide complete configuration and routing control of the effects.

The key to using the Sub Mixer parameters is understanding what the Sends and Returns parameters do, and how they interact with the Routing Parameters.

**Sends** The Sends parameter routes the PCM 81's two panned input signals to the four effect inputs (the *Rvb* and *FX* blocks). This parameter has a range of 0–300. The value of this parameter determines what mix of the left and right input signals is fed to the stereo inputs of the *Rvb* and *FX* blocks. The values 0, 100, 150, 200, 300 correspond to specific configurations and are labeled with text instead of numbers.

Sends Value	Rvb Sends	FX Sends
Stereo (0)	L to left Rvb In R to right Rvb In	L to left FX In R to right FX In
L=Rvb, R=FX (100)	L to left Rvb In L to right Rvb In	R to left FX In R to right FX In
Mono (150)	L+R to left Rvb In L+R to right Rvb In	L+R to left FX In L+R to right FX In
L=FX, R=Rvb (200)	R to left Rvb In R to right Rvb In	L to left FX In L to right FX In
Stereo (300)	L to left Rvb In R to right Rvb In	L to left FX In R to right FX In

**Returns** The Returns parameter routes the four effects outputs to the PCM 81's stereo outputs. This parameter has a range of 0–300. The value of this parameter determines what mix of the stereo *Rvb* and *FX* outputs is fed to the PCM 81's stereo outputs. The values 0, 100, 150, 200, 300 correspond to specific configurations and are labeled with text instead of numbers.

Returns Value	Rvb Returns	FX Returns
Stereo (0)	left Rvb Out to L right Rvb Out to R	left FX Out to L right FX Out to R
Rvb=L, FX=R (100)	left Rvb Out to L right Rvb Out to L	left FX Out to R right FX Out to R
Mono (150)	left + right Rvb Out to L left + right Rvb Out to R	left + right FX Out to L left + right FX Out to R
FX=L, Rvb=R (200)	left Rvb Out to R right Rvb Out to R	left FX Out to L right FX Out to L
Stereo (300)	left Rvb Out to L right Rvb Out to R	left FX Out to L right FX Out to R

## Routing

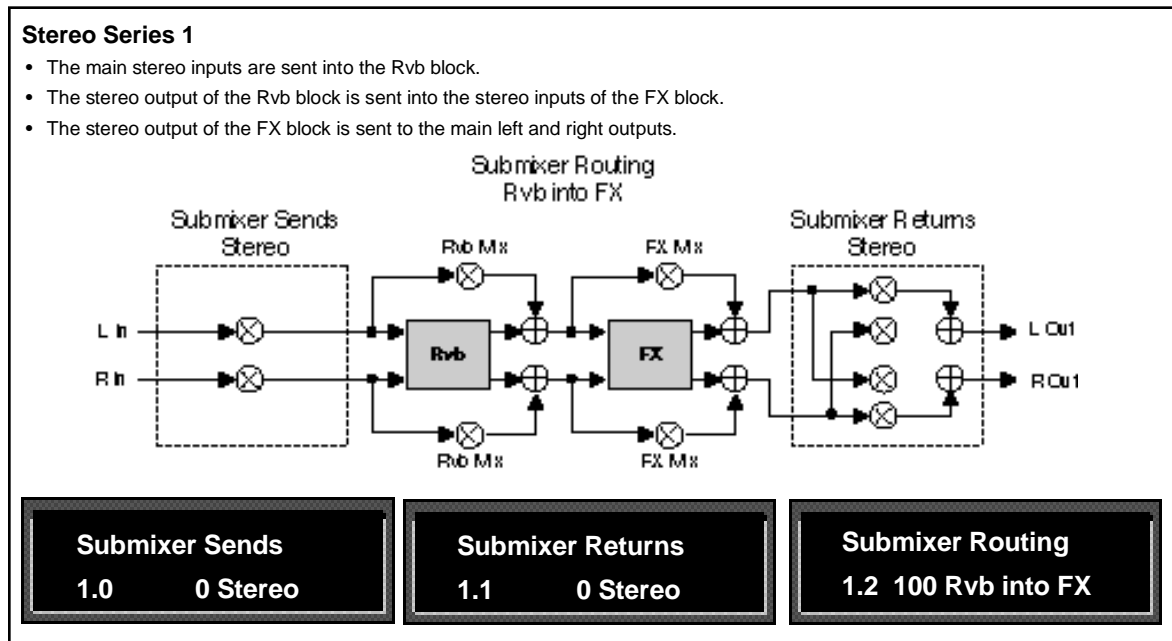
The Routing parameter provides continuously variable control of the FX and Rvb effect routing: Rvb and FX in parallel, Rvb into FX (series) and FX into Rvb (reverse series). This parameter has a range of 0–400. The value of this parameter determines the routing/ordering of the stereo Rvb and FX effects. The values 0, 100, 150, 200, 300, 400 correspond to specific configurations and are labeled with text instead of numbers.

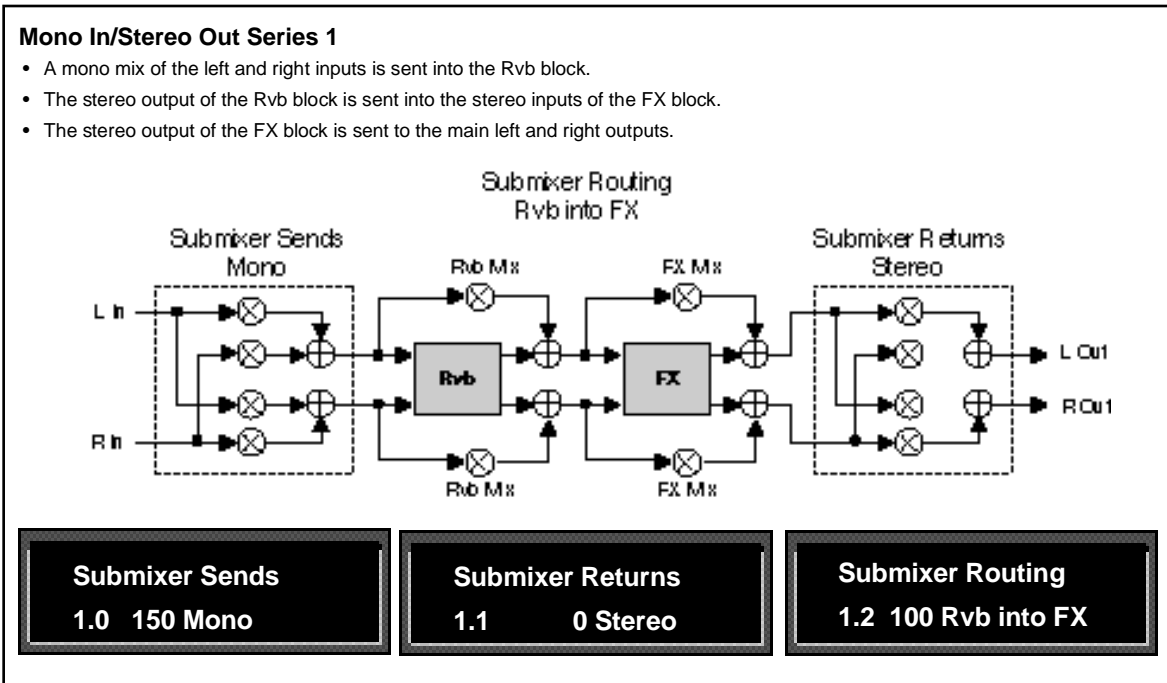
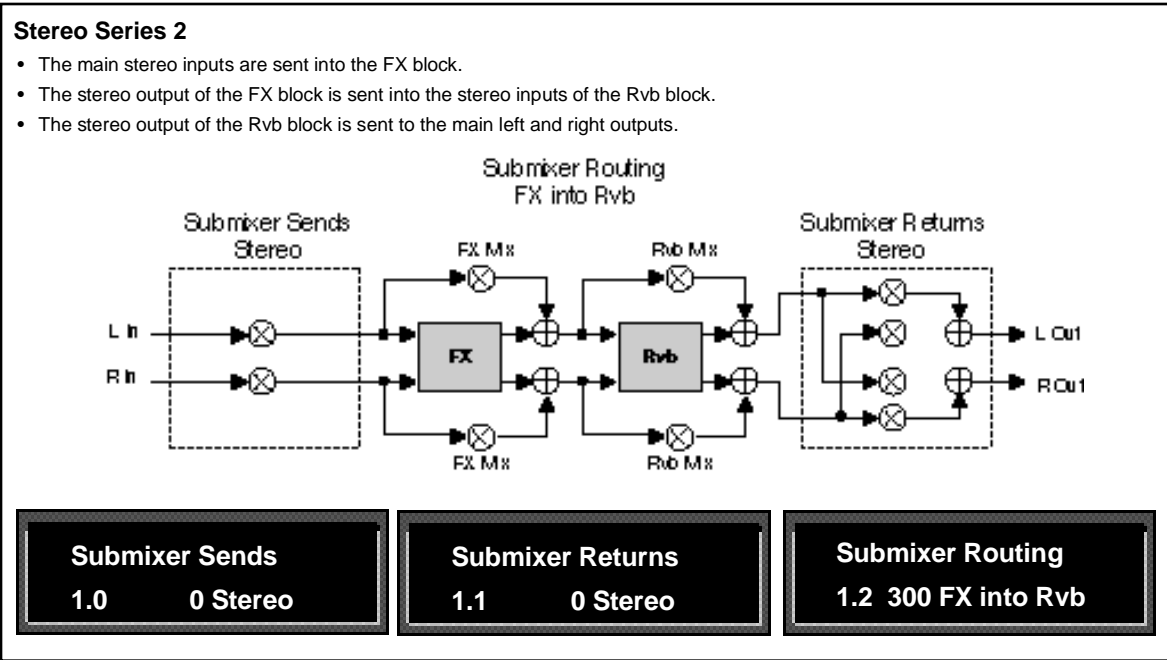
Routing Value	Effects Routing
Parallel (0)	Rvb and FX are parallel
Rvb into FX (100)	Rvb outputs into FX inputs (series 1)
Parallel (200)	Rvb and FX are parallel
FX into Rvb (300)	FX outputs into Rvb inputs (series 2)
Parallel (400)	Rvb and FX are parallel

Note that the routing value takes precedence over the Sends and Returns values. For example, if Routing is set to Rvb into FX (100), no input signal is routed directly to the FX inputs — and the Rvb outputs aren't sent directly to the PCM 81's main outputs.

## Useful Configurations

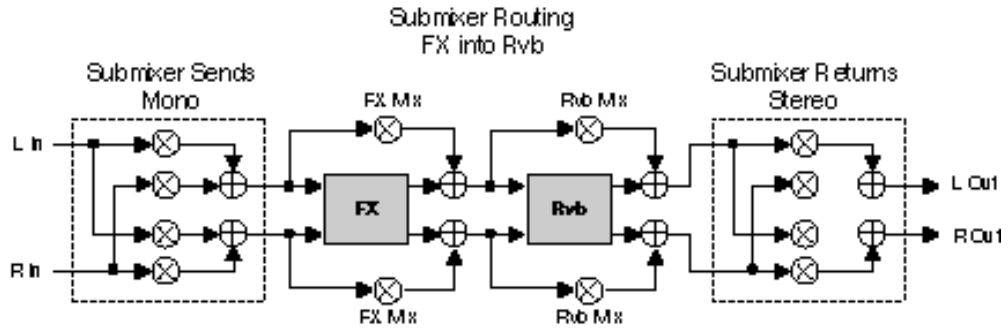
It is possible to arrange the stereo Rvb and FX inputs and outputs into a variety of useful configurations by simply adjusting the Sends, Returns and Routing parameters. The following illustrations show several of these configurations. Each illustration contains a block diagram showing signal flow through the Submixer, as well as the displayed values for the Sends, Returns, and Routing parameters that will produce the configuration.





**Mono In/Stereo Out Series 2**

- A mono mix of the left and right inputs is sent into the FX block.
- The stereo output of the FX block is sent into the stereo inputs of the Rvb block.
- The stereo output of the Rvb block is sent to the main left and right outputs.



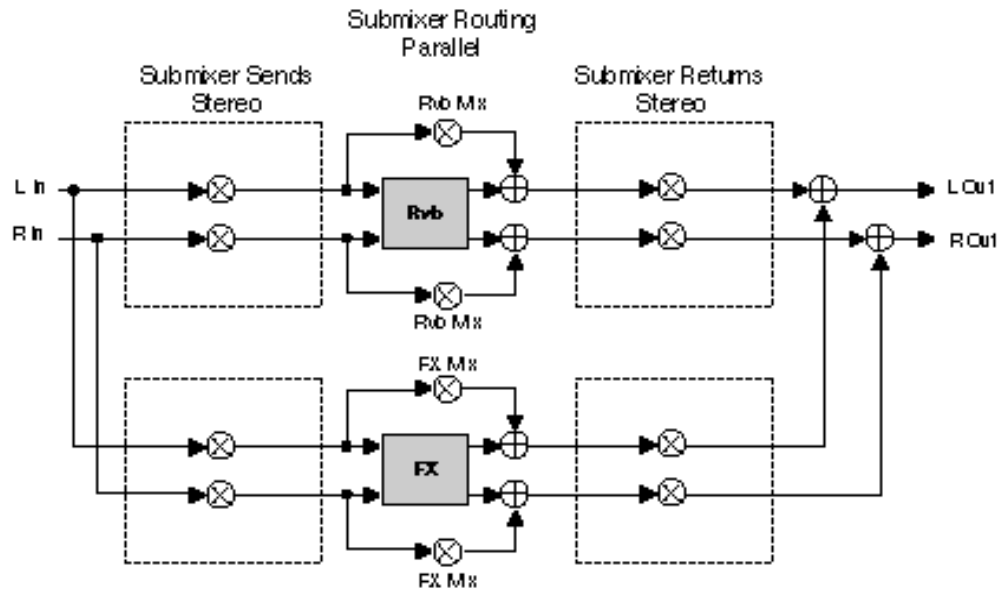
**Submixer Sends**  
1.0 150 Mono

**Submixer Returns**  
1.1 0 Stereo

**Submixer Routing**  
1.2 300 FX into Rvb

**Stereo Parallel**

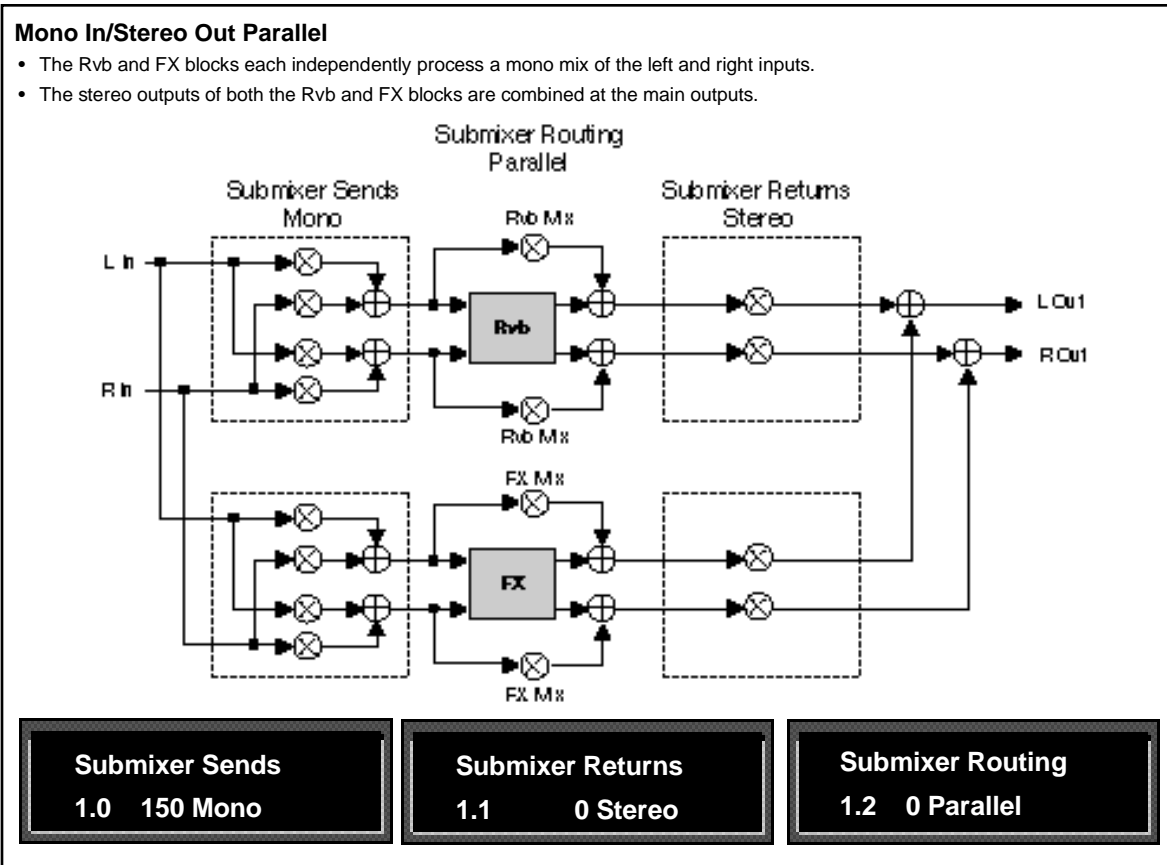
- The Rvb and FX blocks process the left and right inputs independently.
- The stereo outputs of both the Rvb and FX blocks are combined at the main outputs.

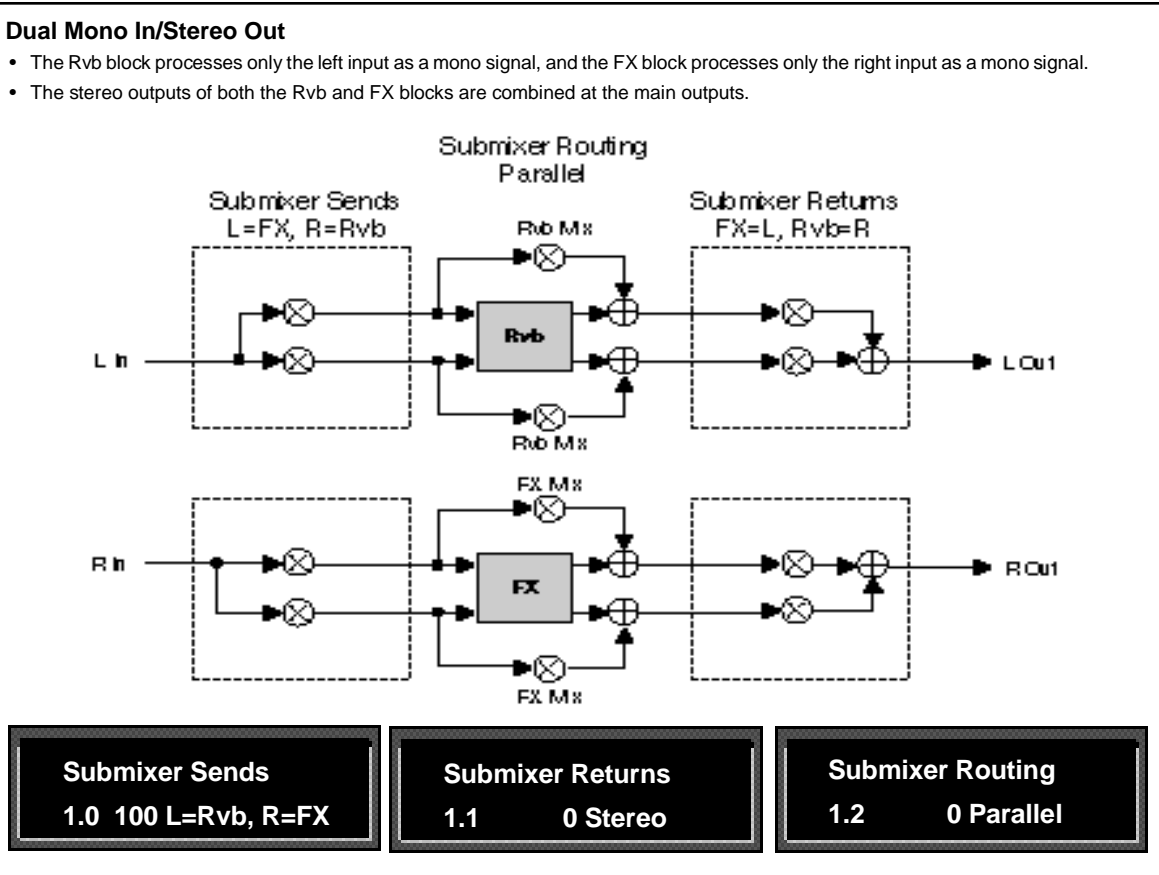


**Submixer Sends**  
1.0 0 Stereo

**Submixer Returns**  
1.1 0 Stereo

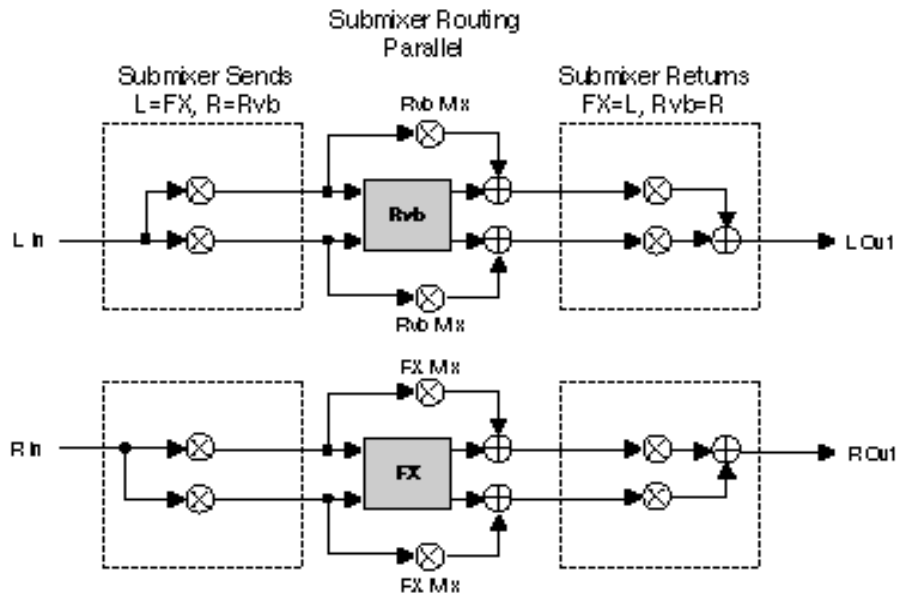
**Submixer Routing**  
1.2 0 Parallel





**Dual Mono In/Mono Out**

- The Rvb block processes only the left input as a mono signal, and the FX block processes only the right input as a mono signal.
- A mono mix of the Rvb block output is sent to the left main output. A mono mix of the FX output is sent to the right main output.

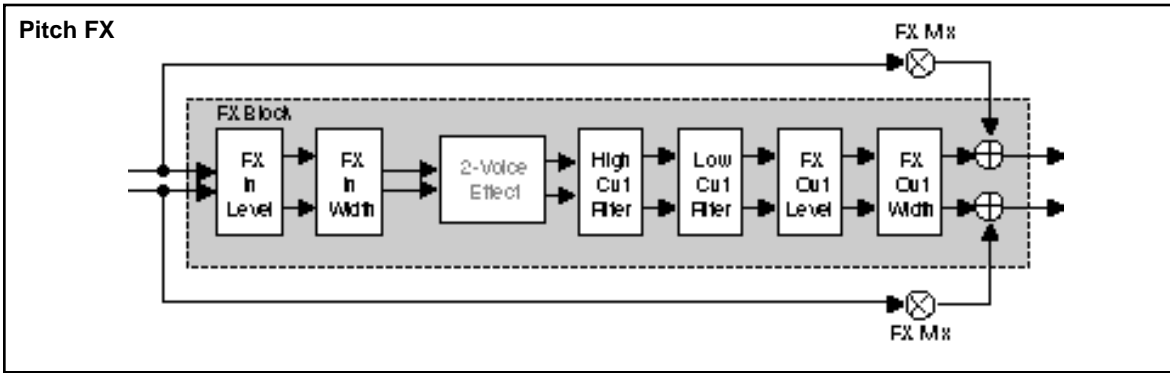
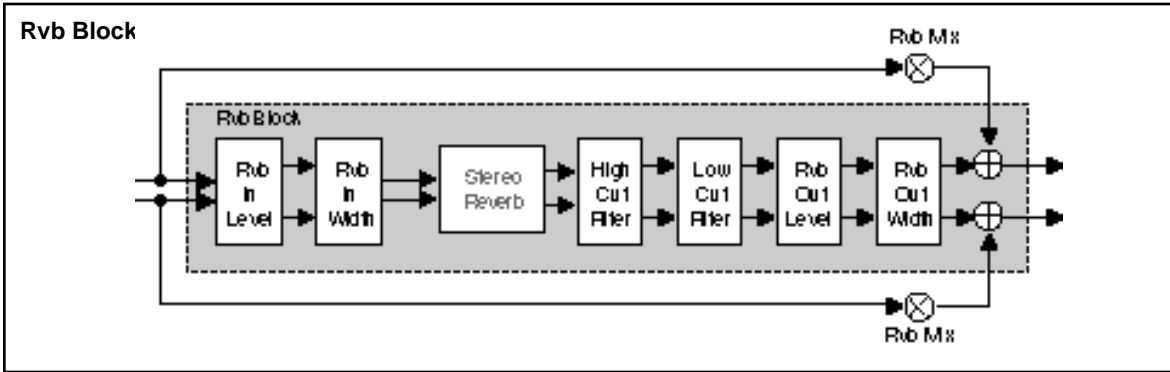


**Submixer Sends**  
1.0 100 L=Rvb, R=FX

**Submixer Returns**  
1.1 100 Rvb=L, FX=R

**Submixer Routing**  
1.2 0 Parallel

The Rvb and FX blocks shown in the previous configurations represent the Stereo Reverb and 2-Voice Pitch Shift effects. Each of these blocks has its own set of controls, shown in the diagrams below. **Rvb and FX Block Controls**



These effect block controls are also found in the Submixer row of each parameter matrix.

1	1.0	1.1	1.2	1.3	1.4	1.5	1.6	1.7	1.8	1.9
Submixer	6 ends	Returns	Routing	Mix	In Lvl	In Wdth	HiCut	LoCut	Out Lvl	Out Wdth

These parameters provide independent control of input and output level and width, as well as high cut and low cut parameters for each FX and Rvb block.

Press **Load/Q** to toggle between Rvb and FX parameters.

These parameters are described below, in the order in which they appear in the Submixer row. (All other effect parameters are identical to those described earlier for the 4-Voice and 6-Voice algorithms.)

**RvbMix, FX Mix**

These are independent wet/dry mix controls for the Rvb and FX effect blocks.

**RvbInLvl, FX InLvl**

These are independent input level controls for the Rvb and FX blocks.

**Rvb In W, FX In W**

These are input width controls for the Rvb and FX blocks. These are essentially the same as the PCM 81 FX Width and Rvb Width parameters. The difference is that the width control in the Pitch FX algorithms is located at the input to the effect. This makes it possible to independently alter the stereo image of material feeding the two effect blocks.

**RvbHiCut, FX HiCut**

These parameters provide independent 6dB per octave, stereo high-cut filters on the outputs of the Rvb and FX blocks.

**RvbLoCut, FX LoCut**

These parameters provide independent 6dB per octave, stereo low-cut filters on the outputs of the Rvb and FX blocks.

**RvbOutLvl, FX OutLvl**

These are independent output level controls for the Rvb and FX blocks.

**Rvb Out W, FX Out W**

These parameters allow independent control of the stereo image of the output of each effect block.

The Vocal Fix Pitch Correct algorithm is designed to work with monophonic (one note at a time) vocal sources. The algorithm contains an intelligent pitch shifter combined with a PCM 81 Chamber reverb. The intelligent pitch shifter detects the pitch of incoming audio and produces effects based on the detected pitch.

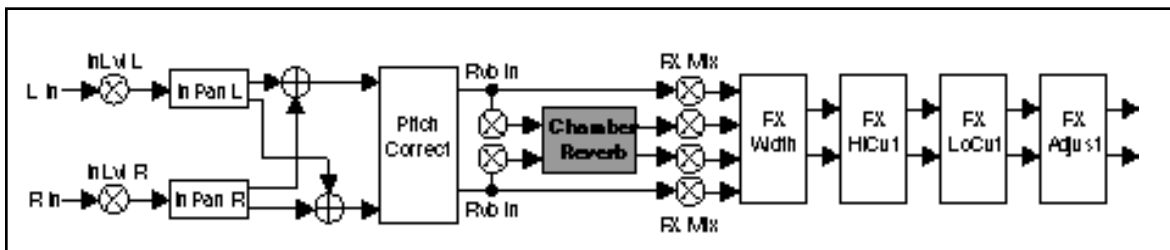
### The Pitch Correct Algorithm and Its Parameters

The reverb follows the pitch shifter in series. The FX Mix parameter is set to 0% reverb as most applications require only pitch processing.

The algorithm provides a simple, yet powerful, set of tools for correction of previously recorded melody tracks which are off pitch. The Pitch Detect display shows the pitch of the input source in real time, displayed as a chromatic note and  $\pm$ cents so that you can see exactly where and how much correction is needed. For most applications correction will only be needed for notes that are audibly sour.

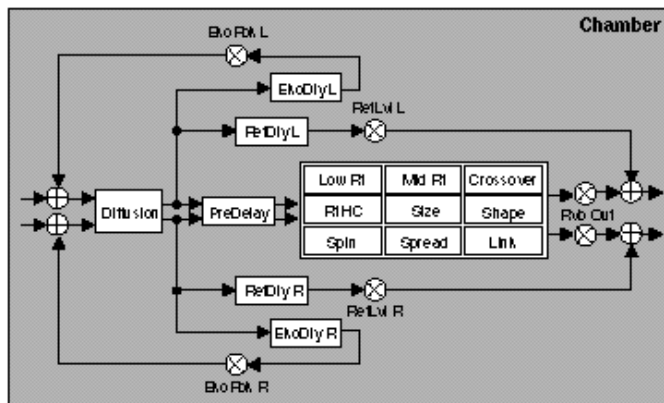
There are two basic approaches to this type of pitch correction. The first is to patch the Correct parameter to a switch or to the ADJUST knob, the second is to use a MIDI keyboard to control correction. The patch method is preferable for fine tuning a performance, whereas using a MIDI keyboard provides more flexibility when handling more serious pitch errors, or when it is necessary to completely rework the melody.

MIDI control is straightforward. Just connect a MIDI keyboard to the PCM 81 MIDI IN jack, set the keyboard and the PCM 81 to the same MIDI channel, and you're ready to start. Don't try to play the melody note-for-note — just press the keys needed to correct bad notes. When you press a key on the MIDI keyboard, the audio is forced to the keyboard's pitch. When you release the key, the audio is processed according to the settings you have made to Correct and Tracking. Typically, Correct will be set to 0% and Tracking set to Fastest. This results in the MIDI note determining the correct pitch and triggering correction on and off. (By the way, you don't need a score to know which note, just watch the Pitch Detect display.)



0 Controls	0.0 Mix	0.1 LowAdj	0.2 *LowLFR	0.3 *RC Mix	0.4 *FX HILo Cut	0.5 *FX Width	0.6 *Lo H Limit				
1 Reverb Time	1.0 Low Rf	1.1 Mid Rf	1.2 Crossover	1.3 Rf HC	1.4 Pre Delay	1.5 *Rf LowDL	1.6 *Rf HighDL				
2 Reverb Design	2.0 Size	2.1 Diffusion	2.2 Shape	2.3 Spread	2.4 Spin	2.5 Link	2.6 *Rf Out Th				
3 Pitch	3.0 Detect	3.1 Correct	3.2 GRResp	3.3 Tracking	3.4 Low Pitch	3.5 High Pitch	3.6 Tuning	3.7 Spike			
4 Gain	4.0 Gain	4.1 *Gainnes									
5 Modulation	5.0 *ModLFO	5.1 *ModRFBW	5.2 Mod EnvL	5.3 Mod EnvR	5.4 *ModLch	5.5 *ModSumL	5.6 *ModSumC				
Reverb	Patch 0	Patch 1	Patch 2	Patch 3	Patch 4	Patch 5	Patch 6	Patch 7	Patch 8	Patch 9	

The Chamber reverb produces an even, relatively dimensionless reverberation, with little change in color as the sound decays. The initial diffusion is similar to the Concert Hall algorithm, but the sense of space and size is much less obvious. This characteristic, along with the low color in the decay tail, makes Chamber useful on a wide range of material. It is especially useful on spoken voice, giving a noticeable increase in loudness with very low coloration.



**The Pitch Correct Parameters**

**Controls Row**

**FX HiCut, FX LoCut**

These parameters control stereo high cut and low cut filters on the algorithm outputs. Each has a range of 20Hz- 20kHz.

**Detect**

This parameter displays the source used for pitch detection: Input, Fixed or MIDI.

**Pitch Row**

- Input Pitch is detected from the sum of the left and right input signals. The detected pitch and the number of cents sharp or flat is displayed and updated in real time.
- Fixed ADJUST scrolls through the range of available pitches. The number of cents sharp or flat from the selected fixed pitch is displayed and updated in real time.
- MIDI This setting is displayed when you use a MIDI keyboard as the note detect source. Connect a MIDI keyboard to PCM 81 rear panel MIDI IN jack and set both to the same MIDI channel. Whenever a preset that uses the PitchCorrect algorithm is running, the shifter will detect the pitch of any MIDI key you press. The display will show the MIDI pitch and the number of cents sharp or flat in real time as the pitch of the input signal varies. Correct the pitch of any melody by simply playing the appropriate MIDI key when ever the melody pitch drifts from the desired note.

**Correction**

This parameter lets you control the amount of off-pitch correction (0-100%) applied. When set to 0, no correction is applied. When set to 100%, notes are shifted as close as possible to the frequency of the detected pitch (or to the frequency of harmonies of the detected pitch).

**GldResp**

This parameter is similar to the GldResp parameter found in the Master Delay parameter of several PCM 81 delay and reverb algorithms. It sets the pitch shifter's responsiveness to the pitch detect source and also to modulation by internal controllers (LFO, AR etc.) and MIDI controllers (aftertouch, velocity, etc.) At the highest setting (100), the pitch will track the detect source very closely. At lower settings, the pitch will lag behind the modulation source. A setting of about 50 generally yields good results for tracking vocals.

**Tracking**

This parameter controls how quickly (accurately) the pitch detector will follow pitch changes in the detect source. This control can be set to Fastest, Fast, Moderate, Slow or Hold.

Fastest This is generally the most useful setting as it provides accurate tracking as well as good handling of vibrato or pitch bends.

Fast, Moderate and Slow These settings are useful for special effects.

Hold This setting freezes at the last detected pitch, effectively turning any melody into a pedal tone.

### Low Pitch and High Pitch

These controls allow you to set a range of pitches to be detected. This is very useful with source material that has extraneous noise, for example, a vocal track which contains some headphone leakage picked up by the vocal mike. If Low and High Pitch are set to bracket only the vocal frequencies, there will be no glitches caused by the shifter detecting headphone material outside the vocal range.

Note that Low Pitch also affects the amount of time that the pitch shifted material is delayed. At the lowest setting, the amount of delay is about 60ms. At the highest setting it is about 20ms.

For reference, a list of the lowest and highest pitches of some typical sources is shown below.

vocal: bass	Eb2 - -Eb4
vocal: baritone	Ab2 - -Ab4
vocal: tenor	C3 - -C5
vocal: alto	F3 - -F5
vocal: soprano	C4 - -C6

### Tuning

This parameter allows you to set a reference frequency for the pitch, "A". The range is from 410.0-470.0Hz. The standard reference frequency is 440.0Hz.

### Splice

This parameter allows you to change the amount of time used to crossfade the splices that occur whenever pitch is shifted. A value of 8ms is appropriate for most material. Higher values will produce longer/smoothier cross-fades, but the intelligibility of transients and high frequency content may be affected. Lower values will reproduce transients more accurately and have more high frequency content, but the splices may be more audible.

### Shift Row Shift Cents

This parameter determines the amount of fixed shift for the stereo input channels. The value is given in cents (1 cent=1/100th of a semitone, 1200 cents/octave). The range is adjustable in increments of a single cent. This value can be modulated smoothly with patch sources to create vibrato and other pitch-related special effects.

### Shift Semitones

This parameter provides  $\pm 1$  octave of pitch shift in half-step (chromatic) intervals.

Note: As the Cents and Semitones controls are additive, it is possible to shift tones by as much as  $\pm 2$  octaves.

## The Presets

The PCM 81 has 300 factory-designed presets which are organized into six banks of 50 each (labeled **P0-P5**). Each bank is organized in a matrix of 5 rows of 10. Press the front panel **Program Banks** button to display the first bank. Press it again to switch to another bank. Simultaneously press **Program Banks** and either the **Up** or **Down** button to backstep through the banks. The display will show the bank label and the matrix location, the preset name, and the algorithm from which the effect is derived. Turn SELECT to scroll through all of the presets in numerical order. Use **Up** or **Down** to jump forward or backward by 10. Press **Load/★** to load any displayed preset.

In the Program Banks mode (as in the Register Banks mode), ADJUST is a soft knob. Each preset has one or more parameters patched to this knob, providing a quick way to make useful changes to the effect. (A ■ symbol in the upper left corner of the display indicates that the currently running effect has an ADJUST knob patch.) When you turn ADJUST, the display will show the name assigned to ADJUST, as well as the patch value. Continue turning ADJUST to alter the patch value along its available range. The program banks are organized as follows:

Bank	Matrix Location	Preset Type
<b>P0</b>	0.0–1.9	Multi Effects
	2.0–3.5	Modulation Effects
	3.6–4.9	Special Effects
<b>P1</b>	0.0–3.6	Rhythmic Echo and Delay Effects
	3.7–4.9	Ambience Effects
<b>P2</b>	0.0–1.0	EQ Effects
	1.1–2.4	Spatial Effects
	2.5–4.0	Gain Effects
	4.1–4.9	Resonant Chord Effects
<b>P3</b>	0.0–2.8	Reverb Effects
	2.9–3.9	Processed Reverb Effects
	4.0–4.9	Remix Effects
<b>P4</b>	0.0–1.0	Vocal Shift
	1.1–1.9	Vocal Harmony
	2.0–2.9	Guitar Magic
	3.0–3.9	Instrument Shift & Harmony
	4.0–4.9	Percussion & Tempo
<b>P5</b>	0.0–0.9	Sci-Fi FX
	1.0–1.9	Pitch & Delay
	2.0–2.9	Pitch Sequences
	3.0–3.7	Pads & Drones
	3.8–3.9	Utility Programs
	4.0–4.9	Pitch Correct

Each preset is described in this section with a header which indicates the matrix location, the program name, the name assigned to the ADJUST knob, and the range of ADJUST knob control. This header is followed by a brief description of the effect. For additional reference, at the end of this chapter is an alphabetical list of the presets, with references to bank and matrix location.

## Program Bank 0 (P0)

### Multi Effects (P0 0.0 – P0 1.9)

#### 0.0 Prime Blue **ADJUST: Efx/Rvb X 0–127**

A combination of 3 stereo effects: 6 voice chorus, rhythmic echoes, and reverb. You can dial-in the exact proportion of each. As ADJUST is turned from 0 to 127 the effect smoothly changes from chorus only, to chorus with echoes, to chorus with echoes and reverb, to reverb with echoes, and finally to reverb only. Use **Tap** to set the echo rhythms.

#### 0.1 EkoChorus **ADJUST: FX Mix 0–127**

A 6 voice stereo chorus with echoes and stereo reverb. ADJUST controls the mix of chorus/echoes with reverb. The Soft Row provides all the tweaks you'll need to get the most out of this classic multi effect.

#### 0.2 Wet Chorus **ADJUST: FX Mix 0–100**

This is a thick wet chorus combined with a small amount of reverb to produce a large lush sound. It can fatten up any track and is especially good for adding body to guitar. ADJUST controls the combination of delays and chorus with reverb.

#### 0.3 Wet Chorus 2 **ADJUST: FX Mix 0–100**

This is a variation of Wet Chorus with a subtle amount of delays and reverb to produce just the right amount of ambience. ADJUST controls the combination of delays and chorus with reverb.

#### 0.4 Wet Chorus 3 **ADJUST: FX Mix 0–100**

A variation of Wet Chorus 2 with even thicker chorus (bordering on a slight flange) and slightly modified delays. ADJUST controls the combination of delays and chorus with reverb.

#### 0.5 Chorus Plate **ADJUST: KorusMix 0–100**

Combination of smooth rolling delays in the reverb shell combined with a more aggressively modulated reverb tail. ADJUST changes the non-modulated to modulated reverb tail. Twisted for guitar.

#### 0.6 Chorus Room **ADJUST: Decay 0–80**

Slow chorus applied to one side of the reverb. The speed of the modulation follows input level. ADJUST controls the reverb decay.

#### 0.7 ChorusToRvb **ADJUST: Decay 0–127**

Input level controls the relative mix of stereo chorus and reverb. Loud signals are mostly chorus and the reverb level comes up as the signal fades away. ADJUST controls the reverb decay time.

#### 0.8 Funkus Room **ADJUST: LFO Rate 0–100**

This preset works great on electronic pianos / keyboards. Strong modulation applied to both delays and reverb tail. ADJUST controls the speed of the LFO modulation.

#### 0.9 Detune & Room **ADJUST: Detune 0–127**

A very tight room combined with separate pitch modulation on the left and right inputs. ADJUST controls the depth of the detuning.

#### 1.0 Detune&DbI **ADJUST: Detune 0–127**

Similar to Detune & Room, but several delay voices have been added to produce doubling effects. ADJUST controls the amount of detune on each side. Soft Row goodies include a master delay control that allows you to vary the time difference between the delay voices.

#### 1.1 Tight Space **ADJUST: SplitWide 0–100**

A bright, dry ambience combined with left and right detune and delay. ADJUST controls the amount of detune as well as the left/ right delay spread.

- 1.2 Flange >Rvb** **ADJUST: FX Mix 0–100** **Multi Effects (PO 0.0 – PO 1.9) cont'd.**  
This stereo effect feeds the output of a flanger into a concert hall reverb. ADJUST controls the mix of dry and wet flanged audio. The Soft Row includes master delay and feedback parameters for adding echoes, as well as parameters for modulation and image control.
- 1.3 Flange+Rvb** **ADJUST: FX Mix 0–100**  
A rich 6 voice chorus in parallel with reverb. ADJUST controls the mix of the two independent stereo effects. The Soft Row includes the essential parameters for creating many variations of this classic combination of effects.
- 1.4 X Eko Flange** **ADJUST: X-Fbk 0–100**  
A stereo flanger with rhythmic echoes feeding into stereo reverb. ADJUST controls the cross-feedback of the echoes (which causes them to bounce from side to side as they repeat). Press **Tap** to synchronize the echoes with rhythmic sources.
- 1.5 Glide > Verb** **ADJUST: FX Mix 0–100**  
Three stereo effects in series: gliding delays, rhythmic echoes and reverb. ADJUST controls the mix of delay effects and reverb. Essential controls for each effect are included in the Soft Row.
- 1.6 Glide X-Ekos** **ADJUST: Eko Time 0–100**  
Similar to Glide > Verb, but ADJUST lets you dial in echo times from 0 to 2 seconds. Feedback and cross feedback are combined to create echo patterns that change as they repeat.
- 1.7 Detuned Ekos** **ADJUST: Ekos/Beat 1–24**  
Plate reverb combined with independent stereo delays. The delay voices are diffused and modulated. The echoes are rhythmic — set the tempo with **Tap** or MIDI clocks. Use ADJUST to set the number of echoes per beat.
- 1.8 StereoEqEkos** **ADJUST: Ekos/Beat 1–24**  
Six rhythmic echo voices are EQ'd and panned across stereo space. ADJUST sets the number of repeats per beat.
- 1.9 ADJUpMyEchos** **ADJUST: EchosLvl 0–10**  
Concert Hall reverb with 4 panned echo voices in the background. ADJUST controls the level of the echo voices. Press **Tap** to synchronize the echoes with rhythmic material.
- 2.0 FSw2 Elevate** **ADJUST: Feedback 0–99** **Modulation Effects (PO 2.0 – PO 3.5)**  
Similar to Detuned Ekos, multiple echo voices with diffusion, modulation and a touch of plate reverb. In this effect, Foot Switch 2 is patched to turn on the AR Envelope which, in turn, sweeps the master delay. ADJUST controls the feedback of these rhythmic echoes.
- 2.1 Chorus &Pan** **ADJUST: LFO Rate 0–100**  
This moving chorus shifts the output signal from left to right, back and forth. ADJUST sets the speed. Increasing the Delay Master parameter will smear the delay images.
- 2.2 Chorus&Amb** **ADJUST: MstChorus 0–100**  
Similar to Chorus&Pan but with an added ambience. ADJUST controls the speed and depth of the chorus.
- 2.3 6 Vox Chorus** **ADJUST: Less/More 0–50**  
Starting point for all chorus sounds. ADJUST controls both the chorus depth and speed. Use the Delay Master to open or close the spacing between the 6 delay voices.

**Modulation Effects  
(P0 2.0 – P0 3.5) cont'd.**

- 2.4 Split C&E** **ADJUST: InputPan 0–100**  
The left input is processed into a lush 3-voice chorus with the voice panners adjusted from center to left. The right input is processed into a rhythmic 3-voice echo with the output panned from center to right. ADJUST cross-pans the inputs. 0 = left/right stereo, 50 = mono, 100 = right/left stereo.
- 2.5 Env:PanKorus** **ADJUST: Chorus 0–127**  
The AR Envelope drives the speed of the pan based on the presence or absence of input signal. ADJUST varies the chorus amount.
- 2.6 6 Vox Flange** **ADJUST: MstrDepth 0–100**  
A rich stereo flanger with a touch of reverb. ADJUST controls the flange depth.
- 2.7 Mod Max** **ADJUST: Mod Knob 0–50**  
Multiple parameters all being modulated together. With a stereo input the LFO drives the input pans to each delay and reverb processor. A definite "twist your head off" effect. ADJUST controls several aspects of the modulation.
- 2.8 PreciseGlide** **ADJUST: Resonance 0–100**  
A very clean stereo gliding delay and reverb. ADJUST controls the resonance of the glide. Soft Row includes master delay and feedback parameters so you can add stereo echoes to the effect.
- 2.9 Round 147** **ADJUST: RotorRate 0–20**  
Big wooden rotary speaker cabinet miked fairly close. ADJUST sets the speed of the rotors. FX Mix is available in the soft row to control the amount of reverb relative to the total effect output.
- 3.0 FSw2 Rotary** **ADJUST: Width 0–100**  
A dual-rotor speaker cabinet with a very wide stereo spin and a touch of ambience. Use ADJUST to control the width and direction of the spinning rotors. 0= very wide left-to-right, 64 = mono, 127 = very wide right-to-left. Foot Switch 2 is patched to the Latch to toggle between slow and fast speeds. The AR envelope is used to simulate the inertial drag as the rotors speed up or slow down.
- 3.1 RotorCabinet** **ADJUST: Slow/Fast 0–1**  
Similar to FSw2 Rotor, but ADJUST is used to toggle the speed of the rotors between slow and fast. Soft Row parameters include FX Width which sets the width and direction of the spin.
- 3.2 MIDI Rotary** **ADJUST: EkoFbk 0–100**  
Similar to FSw2 Rotor, but with echoes added and the rotors patched up for MIDI control. MIDI After Touch toggles speed. Press hard to spin fast, press hard again to spin slow. ADJUST controls the amount of echo feedback. Use **Tap** to synchronize the echoes with rhythmic material.
- 3.3 Tiled Rotary** **ADJUST: Slow/Fast 0–1**  
The RotorCabinet effect in a tiled room. Try it with background vocals, as well as the usual keyboard and guitar sources. ADJUST toggles the spin rates between slow and fast.
- 3.4 RotoWood** **ADJUST: Speed 0–10**  
ADJUST quickly advances the speed of the rotors. Increase the Feedback Master for more upper rotor "howling".
- 3.5 RandomImages** **ADJUST: ImageKnob 0–127**  
This effect works best with solo instruments or voices. Individual input notes come out at random locations in the stereo image. ADJUST varies delay time, adds chorusing, and turns up the level of tempo related echoes. Delays "creep" slowly out to new time values.

- 3.6 Under Water** **ADJUST: DrownKnob 0–100** **Special Effects (P0 3.6– P0 4.9)**  
This effect really pulls you under! It will submerge any track under water. ADJUST controls the over all rate of the effect.
- 3.7 Thunder FX** **ADJUST: FlashTime 0–127**  
An unusual special effect that produces a rolling clap of thunder from a percussive source (tom toms , etc.) and ethereal sweeps from synth pads. The effect is driven by the AR Env, which is available in the soft row. ADJUST controls the AR release rate.
- 3.8 Thunder FX 2** **ADJUST: FlashTime 0–127**  
A variation of Thunder FX. The zaps are produced by setting a much faster release rate. ADJUST controls the release rate.
- 3.9 ChaosImpuls** **ADJUST: GldResp 0–100**  
This special effect is hard to describe but interesting to listen to. It produces chaotic rhythms and detunings from the input source. ADJUST sets the chaotic limits. Try this with individual percussive hits.
- 4.0 DemonDescent** **ADJUST: GldResp 0–100**  
Dark reverb and modulated detuning are combined to create an eerie special effect. ADJUST controls the rate and depth of the detuning.
- 4.1 RoughIdle FX** **ADJUST: Speed 0–127**  
Use this to mutate any steady, broad band source into a poorly tuned combustion engine. ADJUST controls the speed.
- 4.2 RoxStutter** **ADJUST: FX Mix 0–100**  
Use on those strong rock and roll lead vocals. The trailing echoes on voices 3 and 4 go longer as the signal disappears. ADJUST combines the delay effects with a plate reverb effect. Delays 2 and 5 are available in the soft row to add more echo presence.
- 4.3 Split Pitch?** **ADJUST: SplitWide 0–127**  
Modulated varispeed in stereo. One side goes low as the other side goes high. ADJUST controls the rate and depth of the varispeed. Good for special FX processing of various sources including synths and dialog.
- 4.4 TryTalk'n** **ADJUST: GldResp 0–100**  
A variation of Split Pitch? In this version highly colored reverb and additional delay voices have been added. ADJUST controls the rate and depth of the varispeed.
- 4.5 AutoInfinite** **ADJUST: FX Mix 0–100**  
With signal present the reverb time runs long. With signal absent the infinite process is switched on. The threshold for the event is set high such that after a source is running in the infinite process, you can play softer passages against it which chorus and echo at the same time. AR Envelope parameters are included in the soft row. Adjust the threshold to suit your application.
- 4.6 Remove Cntr** **ADJUST: Cntr Fc 0–127**  
This effect uses filters and crosstalk cancellation to remove mono material from a stereo mix. ADJUST controls a low pass filter for the center channel frequencies. As it is turned from 0-127 mono material is added back into the mix.
- 4.7 V-Eliminate** **ADJUST: Cut Zone 0–127**  
This effect is a vocal eliminator. It is similar to Remove Cntr, in that it will remove mono material from a stereo mix. In this effect, however, only frequencies in the vocal range are removed. This keeps mono low and high frequencies (kick, bass, snare cymbals, etc.) in the mix. ADJUST controls the width of the elimination band. Note that phase cancellation will occur if the two outputs are summed to mono.

**Special Effects  
(P0 3.6– P0 4.9) cont'd.**

**4.8 NoCenter Eko** **ADJUST: Center Fc 0–127**  
This is the same effect as Remove Center with delays added to the processed signal. It allows you to add echoes to the left and right material without affecting the mono material of a stereo mix or sub mix. For example, you can add additional echoes to a mix without affecting the vocals, kick or snare. ADJUST controls a low pass filter for the center channel frequencies.

**4.9 Rvb On L–R** **ADJUST: Decay 0–127**  
Similar to NoCenterEko, this effect adds dense plate reverb to the left and right material without affecting the mono material in a stereo mix. ADJUST controls the mid Rt of the reverberator.

**Program Bank 1 (P1)**

**Rhythmic Echo and  
Delay Effects  
(P1 0.0 – P1 3.6)**

NOTE: The echoes in all of the Rhythmic Echo and Delay effects can be synchronized to tempo. Tempo is set by dialing it in, pressing **Tap**, or selecting MIDI Clock as the Tempo. (Tempo mode 0.2) Be sure to try these effects synchronized with MIDI sequence and drum patterns.

**0.0 [n]Ekos/Beat** **ADJUST: [n] 1–24**  
This preset produces stereo echoes with a touch of reverb. ADJUST controls the number of echoes per beat. Soft Row goodies include controls for master scaling of the delay rhythms and feedback, EQ and panning, as well as reverb parameters.

**0.1 StereoTapDly** **ADJUST: Mstr Fbk 0–100**  
A variation of [n]Ekos/Beat this effect has left and right delay voices. ADJUST controls master feedback for both voices. The Soft Row includes controls for rhythm, feedback and panning for each voice

**0.2 OffBeat Eko** **ADJUST: DarkKnob 0–127**  
This variation of [n]Ekos/Beat produces a syncopated echo rhythm pattern. ADJUST controls high cut filters in the echo feedback paths – great for “tape” echo effects.

**0.3 OffBeat Eko2** **ADJUST: DarkKnob 0–127**  
Additional rhythmic delay voices are used to create a more complicated syncopation pattern than OffBeat Eko. ADJUST controls high cut filters in the echo feedback paths – great for “tape” echo effects.

**0.4 Dotted 8ths** **ADJUST: EchoSlope 0–127**  
Rhythmic delay voices produce a dotted 8th-note pattern. ADJUST controls the slope of the first six repeats. 0 = loud to soft (normal echoes), 64 = six repeats of equal loudness, 127 = soft to loud (reversed echoes).

**0.5 Shuffle Ekos** **ADJUST: ShufI Lvl 0–127**  
This effect creates echoes with a shuffle feel. One repeat on the beat followed by one repeat off the beat. ADJUST controls the relative levels of the on-beat and off-beat echoes. 0 = on-beat echo only, 127 = off-beat echo only.

**0.6 Shuffles** **ADJUST: EchoSlope 0–127**  
Rhythmic delay voices produce a shuffle pattern. ADJUST controls the slope of the first six repeats. 0 = loud to soft (normal echoes), 64 = six repeats of equal loudness, 127 = soft to loud (reversed echoes).

- 0.7 Sliding Eko** **ADJUST: Slide It! 0–100** **Rhythmic Echo and Delay Effects (P1 0.0 – P1 3.6) cont'd.**  
 This stereo delay effect lets you dial in the perfect “feel” to match the moment. Two echoes are produced. One is fixed on the beat. The other can be slid in musical time anywhere in front of or behind the beat by turning ADJUST. 0-49 = in front of the beat, 50 = on the beat, 51–100 = behind the beat. Of course, the Soft Row contains additional controls for fine tuning the effect.
- 0.8 Six Trips** **ADJUST: EchoSlope 0–127**  
 Rhythmic delay voices produce a six note triplet pattern. Each repeat gets progressively louder. ADJUST controls the slope of the first six repeats. 0 = loud to soft (normal echoes), 64 = six repeats of equal loudness, 127 = soft to loud (reversed echoes).
- 0.9 6 StrokeRoll** **ADJUST: Attack 0–127**  
 Turn a single drum hit into a six-stroke roll with this effect. ADJUST controls the attack of the first five repeats. 0 = loud to soft (normal echoes), 64 = five repeats of equal loudness, 127 = soft to loud (reversed echoes).
- 1.0 TapGated Rvb** **ADJUST: Gate dB 0–85**  
 This effect uses a tempo-controlled LFO as a rhythmic gate to the inputs of a large stereo chamber. The gate is opened every other beat for a duration of one eighth-note. Use this to add reverb to selected beats of a stereo source. (Try this one synchronized to MIDI with a drum machine as a source.) ADJUST controls the input levels in dB to the reverb when the gate is closed. 0 = always open (0dB), 85 = fully closed (-85dB). The Soft Row includes the LFO parameters so you can adjust the overall timing of the gate.
- 1.1 TapEkoGate 1** **ADJUST: GateWidth 0–127**  
 This effect uses a tempo-controlled LFO as a rhythmic gate to the inputs of a delay and reverb combination. The gate is opened once every four beats for a duration of one eighth-note. ADJUST determines the portion of the four beats in which the gate is open. 0 = open for all four beats, 127 = open for only the first 16th note of four beats. The Soft Row includes the LFO parameters so you can adjust the overall timing of the gate.
- 1.2 TapEkoGate 2** **ADJUST: GateWidth 0–127**  
 This is a variation of TapEkoGate 1 with different echo rhythms. ADJUST determines the portion of the four beats in which the gate is open. 0 = open four all four beats, 127 = open for only the first 16th note of four beats. The Soft Row includes the LFO parameters so you can adjust the overall timing of the gate.
- 1.3 TapEkoClear** **ADJUST: GateWidth 0–127**  
 This preset is similar to the two TapEkoGate presets, except that the rhythmic gate is also patched to clear the delay voices when it opens. This allows very high delay feedback values to be used without creating long echo trails. ADJUST determines the portion of the four beats in which the gate is open. 0 = open for all four beats, 127 = open for only the first 16th note of four beats. The Soft Row includes the LFO parameters so you can adjust the overall timing of the gate.
- 1.4 Tap Chamber1** **ADJUST: Width 0–90**  
 This preset is a tempo controlled variation of SnareChamber. The effect is a tight chamber with liveness that is constantly changing. The change is tempo controlled. Press **Tap** to synchronize it to the beat. ADJUST controls the width of the reverb within the overall stereo image of the effect (0 = mono, 45 = stereo, 90 = surround). Use this to open up sampled drum mixes.
- 1.5 Tap Chamber2** **ADJUST: Decay 0–127**  
 This is a version of Vox Chamber with a rhythmic twist. The width of the reverb, within the overall stereo image of the effect, is changed rhythmically by Sw 1. This change is tempo controlled. Press **Tap** twice to synchronize to the beat. ADJUST controls reverb decay.

**Rhythmic Echo and  
Delay Effects  
(P1 0.0 – P1 3.6) cont'd.**

- 1.6 LatchedEkos** **ADJUST: EchoWidth 1–99**  
The inputs and outputs of stereo delays are gated on and off by two rhythmic switches. The Latch is used to trigger the AR envelope, which in turn alternates the left and right delay outputs. ADJUST controls how long the inputs to the delays remain open over a period of four beats.
- 1.7 X-PanEQ BPM** **ADJUST: Low Tone 0-60**  
This effect cross-pans the inputs between independent left and right band pass filters and adds some bright, highly diffused reverb. The pan rate is tempo controlled. ADJUST controls the low frequency limits of both bands.
- 1.8 Pan->Eko BPM** **ADJUST: DarkKnob 0–127**  
This effect pans the inputs across the stereo inputs of a reverb and independent left and right delays, each with slightly different EQ. The pan rate is tempo controlled. ADJUST controls the high frequency limits of both bands.
- 1.9 Tempo Verb** **ADJUST: Liveness 0–40**  
In this tempo-based reverb effect, decay changes in tempo. Slow tempos yield longer reverb time; fast tempos result in shorter decay. Set tempo with two presses of **Tap**, or with MIDI.
- 2.0 Tempo Gate** **ADJUST: High Cut 0-50**  
This heavily gated reverb effect has duration set by tempo. **Tap** in two quarter-notes to set the gate duration.
- 2.1 Tape Echo** **ADJUST: DarkKnob 0–127**  
This preset simulates a stereo tape echo. The echo rate is tempo controlled. Press **Tap** to synchronize with rhythmic material. ADJUST controls high frequency damping – the echoes get darker as they repeat. The Soft Row includes parameters for adding and reverb as well additional controls for the delays and filters.
- 2.2 NonLinear 1** **ADJUST: EchoSlope 0–127**  
This effect produces echoes with non linear decay. ADJUST controls the decay slope. 0 = loud to soft (normal echoes), 64 = six repeats of equal loudness, 127 = soft to loud (reversed echoes). In this preset the slope is set to produce a simple reversed echo effect. The repeat rate is tempo controlled. The Soft Row contains parameters for controlling the stereo image, adding reverb, scaling the repeat rate, and more.
- 2.3 NonLinear 2** **ADJUST: EchoSlope 0–127**  
Similar to NonLinear 1, with some additional processing. Chorus and reverb have been added, and the delays bounce from side to side as they repeat. In this preset the slope is set to 64 to produce repeats of equal loudness. The Soft Row includes parameters for controlling the stereo image, chorus depth and rate, etc.
- 2.4 NonLinear 3** **ADJUST: EchoSlope 0–127**  
This is a variation of NonLinear 2 with more radical processing and an inverted slope.
- 2.5 Six Across** **ADJUST: Spacing 0-100**  
This multi-tap delay effect filters each voice to a specific bandpass with each tap positioned successively across the panoramic spectrum. ADJUST controls the overall spacing between the voices.
- 2.6 BandEko Rvb** **ADJUST: Center Fc 0–127**  
In this preset, the outputs of a stereo band pass filter are sent to left and right rhythmic delays. These delays are in the reverb diffusor loop. The effect produces diffuse, highly colored echoes and reverb. Use ADJUST to set the center frequency of the band pass filter.

- 2.7 BandEkoSweep** **ADJUST: Mstr Fbk 0–100** **Rhythmic Echo and Delay Effects (P1 0.0 – P1 3.6) cont'd.**  
 A variation of BankEko Rvb. In this preset, the center frequency of the band pass filter is swept by the LFO producing echoes of shifting colors. Reverb and diffusion are turned off, but you can add them in from the Soft Row. ADJUST controls master feedback for the left and right delays.
- 2.8 LFO EQ Echo** **ADJUST: LC Depth 0–100**  
 A stereo rhythmic effect created by modulating low cut and high cut filters with the LFO sine and cosine waves. Interesting on rhythm tracks and individual instruments. ADJUST controls the depth of the low cut modulation.
- 2.9 Chase Echo** **ADJUST: ChaseRate 0–100**  
 This is LFO EQ Echo with panning added to the filtered delay outputs. ADJUST controls the pan rate.
- 3.0 Panned Dlys** **ADJUST: Mstr Fbk 0–100**  
 In this effect, a pair of delays are panned to produce echoes that drift across stereo space. ADJUST controls the feedback of both delays. More goodies await you in the Soft Row.
- 3.1 X-Pan Delays** **ADJUST: Mstr Fbk 0–100**  
 The outputs of the delay pair are cross-panned in this effect. Mono material will produce echoes that repeat, but remain in the center. Stereo material will slowly swap left and right as it repeats. Use ADJUST to set the feedback for both delays.
- 3.2 Dly>EQ>Pans** **ADJUST: Mstr Fbk 0–100**  
 The input signal is split into two frequency bands by this effect. The two bands are then sent through two different rhythmic delays whose outputs are panned in opposite directions. This effect deconstructs the tone of input material, except when the panning and delay rhythms align in the center of the stereo field. Try it with a drum mix or other broad band material. Use ADJUST to set the feedback for both delay voices.
- 3.3 Ekoz 4 Drums** **ADJUST: FX Mix 0–100**  
 This preset is a space setter for percussive instruments. It combines a short plate reverb effect with four high density echoes which are highly diffused. ADJUST varies the amount of delay effect to reverb effect.
- 3.4 Haas PanKnob** **ADJUST: L=0, R=127 0–127**  
 A panner that uses the slight differences in left and right channel delay to produce panning without changing relative levels. Use ADJUST to pan from left to right.
- 3.5 Dial a Delay** **ADJUST: Delay 0–100**  
 This one is your basic delay. Use ADJUST to dial-in up to 2.5 seconds of stereo delay. The Soft Row will take you beyond the basics, with parameters for reverb design, EQ and feedback.
- 3.6 PrecisionDly** **ADJUST: L ms/100 0–100**  
 Use this preset if you need precise alignment of left and right channel audio. ADJUST allows you to offset the left channel by up to 1 ms in 100 increments. The Soft Row contains two sets of additional delay controls for the left and right channels, each with 1ms resolution, as well as independent level and panning controls.

- Ambience Effects (P1 3.7 – P1 4.9)**
- 3.7 PhoneOrRoom? ADJUST: Pick One 0–1**  
Use ADJUST to choose between a mono telephone filter and a small room with stereo ambience. The Soft Row provides access to the filter controls as well as reverb design parameters.
- 3.8 CheapTV Room ADJUST: The Walls 1–10**  
This stereo preset simulates the sound of a Lo-Fi TV in a small room. Use ADJUST to change the reflectivity of the walls. The Soft Row provides access to the filter controls as well as reverb design parameters.
- 3.9 Empty Stage ADJUST: Liveness 0–100**  
Made for creating live sounding spaces. ADJUST opens the space to be more reflective and airy.
- 4.0 Tomb Room ADJUST: The Walls 1–10**  
Use this ambience preset to place source material within a very reflective tomb. ADJUST moves the source deeper into this scary space.
- 4.1 Comb Room ADJUST: Tone 0–100**  
This effect provides a tunable comb filter and reverb to produce a highly colored ambience. Use ADJUST to change the tuning of the comb filter.
- 4.2 Zoom Over ADJUST: Speed 1–25**  
Run an effect (or even a continuous synthesizer drone) into this preset, and the sound will approach you from the center, spread out to the sides as it passes overhead, and recede into the distance behind you. ADJUST controls the speed of the fly-by. This effect will image properly in either 2-channel or Surround mixes.
- 4.3 OneShotCarBy ADJUST: Speed 1–100**  
Input level triggers this left-to-right drive-by. When signal is detected, it will move from left to right. Doppler pitch shift is simulated as the image moves across the stereo field. Use ADJUST to set the speed of the drive-by. Once audio has been detected, the effect will cycle once. To drive by again, stop and restart the audio track. To reverse the direction of the effect, set the Width parameter to –45 — you'll find it in the Soft Row.
- 4.4 AmbientCarBy ADJUST: Speed 1–100**  
This is a variation of OneShotCarBy. In this preset the effect is enhanced with the addition of some stereo ambience. Also, in this variation the effect repeats automatically. ADJUST controls the speed of the drive-by.
- 4.5 HeadOn CarBy ADJUST: Speed 1–100**  
This preset uses less ambience than ZoomOver, and is more appropriate for things that approach on the road than in the air. ADJUST sets the speed of the effect. This effect will image properly in either 2-channel or Surround mixes.
- 4.6 2WayStreet ADJUST: Speed 1–100**  
Another variation of AmbientCarBy. This preset produces both left-to-right and right-to-left drive-by effects — like listening to the sound of traffic on a two-way street. The speed for each direction is slightly different. ADJUST is a master speed control for both eastbound and westbound traffic.
- 4.7 IntoTunnel ADJUST: Speed 1–25**  
This preset simulates the sound of a source approaching you from the side, passing you and then entering a tunnel. ADJUST controls the speed of the source. To reverse the direction of the effect, set the Width parameter to –45 — you'll find it in the Soft Row.



**EQ Effects**  
**(P2 0.0 – P2 1.0) cont'd.**

- 0.8 BandReject 4** **ADJUST: FX Mix 0–100**  
Three independent modulators are used to sweep filters and pan the outputs of this preset. The result is an effect with constantly changing tonal and spatial characteristics. A touch of reverb adds some overall ambience. Use ADJUST to set the mix of filters and reverb.
- 0.9 WaaPedalEko** **ADJUST: FeedBack 0–10**  
Wah-wah with echoes. Foot pedal is patched to filter cutoff. ADJUST controls echo feedback.
- 1.0 RotorWaa** **ADJUST: Rate 0–10**  
A tempo controlled wah-wah effect. ADJUST controls the rhythm of the modulator.

**Spatial Effects**  
**(P2 1.1 – P2 2.4)**

- 1.1 Movable Echo** **ADJUST: Position 0-127**  
This preset is designed for use with either stereo or Surround mixes. It is a dual delay effect that can be positioned anywhere between the center, side and rear channels. ADJUST sets the position. 0 = center, 64 = side channels, 127 = rear channel.
- 1.2 Movable Hall** **ADJUST: Position 0–127**  
Like Movable Echo, this preset is designed for use with either stereo or Surround mixes. It is a concert hall reverb effect that can be positioned anywhere between the center, side and rear channels. ADJUST sets the position. 0 = center, 64 = side channels, 127 = rear channel.
- 1.3 Go Away** **ADJUST: How Far? 0–127**  
Use this preset to take a stereo source and move it anywhere from in your face to far, far away. Use ADJUST to move the source away from you. 0 = unprocessed stereo, 127 = far away. As ADJUST is increased, EQ, reverb and stereo width are all changed to produce the composite effect. This effect is compatible with stereo and Surround mixes.
- 1.4 Circular Pan** **ADJUST: Rate 0–100**  
Similar to Go Away, this preset sums the inputs to mono and uses the LFO to control EQ, reverb and stereo width to pan the sound through a circular orbit. From right through center, to left through rear, to right. As the sound approaches the center, it grows brighter, louder and dryer. As it recedes towards the rear, it grows darker, softer and more reverberant. In a Surround mix, the center and rear positions of the orbit will feed only the center and rear channels of the mix. ADJUST controls the rate of the pan.
- 1.5 Spatial Hall** **ADJUST: Position 0–35**  
A multi-dimensional preset which changes as you increase or decrease the ADJUST knob value. This preset is not mono compatible.
- 1.6 Rear of Hall** **ADJUST: Decay 0–60**  
This is a really huge space and you're in the back of it. ADJUST changes the reverberation characteristics, making it boomier.
- 1.7 Backstage** **ADJUST: How Far? 0–127**  
Remember what the concert sounded like from the green room? If you need to re-create that ambience, this preset will take you there and let you wander around. Turn up ADJUST to travel further from the stage until, finally, you're out in the parking lot. The Soft Row provides reverb parameters that will let you fine tune the the hall to fit your needs.
- 1.8 Steered Rear** **ADJUST: Decay 0–10**  
The Width parameter steers this plate effect from front to back after the AR releases based on input level. ADJUST controls reverb decay. Due to the strong spatial positioning at the end of the decay, this preset is not mono compatible.

- 1.9 Too Deep!** **ADJUST: Decay 0–40** **Spatial Effects (P2 1.1 – P2 2.4) cont'd.**  
The left and right envelope followers control the post delay glides while the AR generator controls reverb width. ADJUST controls reverb decay.
- 2.0 Dyna-Hall** **ADJUST: Decay 0–130**  
A tamer version of Too Deep! No envelope chorusing. Good, beautiful, straight ahead, and spacious.
- 2.1 RotoRox** **ADJUST: FX Mix 0-100**  
RotoRox crossfades deeper into two delay voices when the envelope follower detects an absence of input. Great vocal effects for thickly-produced rock and roll. ADJUST will add more or less reverb into the effects mix.
- 2.2 RotoRoomVox** **ADJUST: Decay 0–100**  
This medium large sized room continuously changes spatial width according to the speed of the LFO which tracks the input. Faster with less input, slower with more input. ADJUST controls reverb decay. Nice roomy effect for background vocals.
- 2.3 RotoRoom** **ADJUST: Decay 0–60**  
Similar to RotoRoomVox except the speed of the spatial width modulation is stable, with the speed of the LFO constant. Room is smaller, more dense-sounding and brighter.
- 2.4 RotoRoom#2** **ADJUST: Rt & Rate 0–60**  
Speed and reverb decay are tied to ADJUST. Way cool on a stinger sound effect! Dynamically spacious and not mono compatible.
- 2.5 Tremolo Tap1** **ADJUST: Depth 0–127** **Gain Effects (P2 2.5 – P2 4.0)**  
This is a basic tremolo effect with a small amount of ambience added. The effect is true stereo. The left and right channels are processed separately to maintain the image of stereo source material. The tremolo rate is tempo controlled (2 cycles/beat). Press **Tap** to synchronize the effect with rhythmic material. ADJUST sets the tremolo depth. 0 = no tremolo, 127 = maximum tremolo.
- 2.6 X-TremoloTap** **ADJUST: Depth 0-127**  
This a variation of TremoloTap1. In this preset the left and right modulation are 90° out of phase, producing a stereo effect that doesn't collapse in a mono mix. If the input source is mono, the sound will seem to move from side to side. If the input source is stereo (or two different mono sources) the left and right channels will alternately grow loud and soft. The modulation rate is tempo controlled. ADJUST controls the tremolo depth for both channels. 0 = no tremolo, 127 = maximum tremolo.
- 2.7 Panner BPM** **ADJUST: Pan Phase 0–1**  
This effect is a tempo controlled auto panner with the inputs independently panned left and right. ADJUST allows you to set the relative phase of the input panning. 0 = in phase, 1 = 90° out of phase. In the preset, the panning is 90° out of phase. If the input source is mono it will pan from side to side. If the input source is stereo (or two different mono sources), the two sources will chase each other between the speakers. Set ADJUST to 0 for "normal" auto panning.
- 2.8 Nice Pan!** **ADJUST: FX Mix 0-100**  
This preset combines static time based delays which move back and forth in the stereo field with a short, bright chorus plate. ADJUST varies the delay and reverb mixing. Great for acoustic guitars.

**Gain Effects**  
(P2 2.5 – P2 4.0) cont'd.

**2.9 Spin & Duck** **ADJUST: Spin Rate 0–127**

In this preset, panning is combined with 6-voice chorus delays and reverb to produce a rich spacious effect. ADJUST controls the panning rate. 0 = slow, 127 = fast. The delays are tempo controlled and ducked by input level. They won't be heard during active passages, but will fade up in the spaces between phrases.

**3.0 MultiFxFade** **ADJUST: FX Mix 0–100**

In this preset, the AR envelope is used to create a moderate fade-in of chorus, delay and reverb effects. ADJUST controls the mix of chorus/delay and reverb. The Soft Row has all the essentials for tweaking the three basic effects, as well as the AR envelope parameters so you can fine tune the fade in rate and threshold. Try this with guitars and keys.

**3.1 Ghost** **ADJUST: FadeShape 0–100**

Source material is accompanied by a ghostly image of itself. Not quite reverb, not quite backwards audio. ADJUST controls the contour of the fade in of the effect. This preset works well with short percussive sources as well as more sustained ones.

**3.2 Ghost Flange** **ADJUST: FadeShape 0–100**

This is a variation of Ghost with some flanging added. Try this with lead guitar. ADJUST controls the contour of the fade in.

**3.3 GhostVibrato** **ADJUST: FadeShape 0–100**

Another Ghost variation. This one will produce a delayed vibrato on sustained notes. ADJUST controls the contour of the fade-in.

**3.4 AutoFadeln 1** **ADJUST: FX Mix 0–127**

This preset senses input level to produce an automatic volume swell into a chorus/delay reverb effect. Sounds great with guitar and keyboard chords. The fade in rate is moderate. ADJUST controls the mix of chorus/delay and reverb effects. The Soft Row includes parameters for all three effects as well as the envelope parameters which set the characteristics of the fade-in.

**3.5 AutoFadeln 2** **ADJUST: Fade Rate 0–100**

This variation of AutoFadeln allows you to set the fade-in rate with ADJUST.

**3.6 AutoFadeln 3** **ADJUST: Rvb Lvl 0–127**

This variation of AutoFadeln 1 has shorter delay times and a lighter touch on the chorus effect. ADJUST sets the mix of chorus/delay and reverb effects.

**3.7 ChordSwells** **ADJUST: Rvb Lvl 0–127**

A more dramatic version of AutoFadeln. The fade time is quite long, and the chorus and delay effects are fairly strong. Try it with sustained piano or guitar chords. ADJUST controls the mix of chorus/delay and reverb.

**3.8 BowedChords** **ADJUST: FX Mix 0–100**

A more subtle version of AutoFadeln. Chorusing is turned off and delays are more subdued. This is essentially a volume swell into reverb. Use it to pull cello-like tones from sustained notes or block chords. Very nice with grand piano. ADJUST sets the mix of clean echoes and reverb.

**3.9 BowedEchoes** **ADJUST: FX Mix 0–100**

In this preset, input level controls echo feedback as well as triggering the fade-in. Feedback is reduced when the signal is loud and turned up when the signal fades out. This produces echoes with soft attacks that appear when the input signal drops. ADJUST controls the mix of echoes and reverb.



**Resonant Chord Effects  
(P2 4.1 – P2 4.9) cont'd.**

- 4.7 LvlSweeper** **ADJUST: Color 0–24**  
This preset is driven by level. Any input that exceeds the input threshold will cause a little burst of resonators that quickly swirl through the stereo field. ADJUST controls the pitch range of the resonators.
- 4.8 Sweeper** **ADJUST: Pitch 0–24**  
This preset is a resonant arpeggiator in which a tempo-driven LFO controls the resonator pitches. ADJUST controls the pitch range of the arpeggio effect.
- 4.9 MIDISustain** **ADJUST: Bright 0–5**  
This preset requires MIDI note input or it will not pass audio. Resonators are assigned as the notes are played (when playing chords, it's best to spread them a little). Footswitch 2 works like a piano damper pedal. When the audio source is the same as the MIDI source, the effect is a little like playing a piano while holding down the pedal. When the audio source is different, well ... ADJUST controls the brightness of the resonators.

**Program Bank 3 (P3)****Reverb Effects  
(P3 0.0 – P3 2.8)**

- 0.0 Small+Stage** **ADJUST: Stage Lvl 0–50**  
This preset combines a smooth, small reverberant space with several stage reflections. Use ADJUST to set the stage reflection level.
- 0.1 Small Room** **ADJUST: Liveness 0–60**  
Use ADJUST to quickly change the ambient characteristics of this typically tight sounding room. Great for ADR work.
- 0.2 Living Room** **ADJUST: Decay 0–60**  
Another ADR preset, medium spaced but with a short Rt. ADJUST quickly changes if not perfect for your application.
- 0.3 Brick Kick** **ADJUST: Liveness 0–100**  
Kicks \*ss on kick drums or the entire submix. ADJUST increases the liveness of the chamber.
- 0.4 Large Room** **ADJUST: Decay 0–60**  
A perfectly smooth listening room with medium diffusion. Very natural sounding on any sound source. ADJUST lengthens or shortens the reverb decay.
- 0.5 SnareChamber** **ADJUST: Liveness 0–60**  
Like Brick Kick, ADJUST increases the liveness of the space. A classic from the PCM 70.
- 0.6 Tiled Room** **ADJUST: Decay 0–60**  
A future classic... ADJUST controls reverb decay.
- 0.7 Rich Chamber** **ADJUST: Decay 0–60**  
Smooth and full sounding. ADJUST varies reverb decay.
- 0.8 Vox Chamber** **ADJUST: Liveness 0–60**  
Combines recirculating echoes which fall away quickly once signal is absent. Increasing ADJUST lengthens the reverb decay, which will mask the reflection echoes. For vocals.
- 0.9 Locker Room** **ADJUST: Depth 0–127**  
Just like in high school. ADJUST creates a deeper, more reverberant sound.
- 1.0 Wide Chamber** **ADJUST: Decay 0–60**  
Big and wide sounding with a preset short Mid Rt. ADJUST will change that if necessary. Use on synth pads or vocals.



**Reverb Effects**  
(P3 0.0 – P3 2.8) cont'd.

**2.5 Plate4 Horns** **ADJUST: Impact 0–100**  
Bright and percussive for those horn tracks needing that certain edge. Use ADJUST to modify the attack and release characteristics.

**2.6 Alley Slap** **ADJUST: Pre Delay 0–100**  
Great for any punctuated sound source — vocals, guitars, anything. Use ADJUST to match the delay to the music.

**2.7 Drum Gate** **ADJUST: Duration 0–70**  
This inverse gated preset is low on diffusion and high on attitude. Made for drums and other high impact sources. ADJUST controls the duration of the effect.

**2.8 Slope Down** **ADJUST: Duration 0–100**  
Very inverse sounding effect. ADJUST varies the length of the slope. The AR Envelope, triggered by mono level, cranks in a downward spiraling pitch shift. Unearthly on voices.

**Processed Reverb Effects**  
(P3 2.9 – P3 3.9)

**2.9 BigBoomRoom** **ADJUST: BoomKnob 0–36**  
The Envelope followers glide the left and right post delays giving a strong modulation effect to the reverb tail. ADJUST adds more or less boom. Use on punchy low frequency sound sources.

**3.0 Whammy Hall** **ADJUST: Decay 0–30**  
If your guitar doesn't have a whammy bar, it does now. The AR generator is triggered from the mono level source. The AR generator is then attached to the LFO depth which drives the two Post Delays after the reverb. The Release constant is set rather long so that the depth reaches full scale after 2.2 seconds. ADJUST sets the reverb time.

**3.1 JetChamber** **ADJUST: Jet Knob 0–127**  
A big chamber reverb with stereo flanging on the outputs. Use ADJUST to set the amount of "whoosh." The rate of the flange is controlled by the LFO. You'll find it in the Soft Row along with the essential reverb parameters.

**3.2 EnvChamber** **ADJUST: Jet Knob 0–127**  
Similar to JetChamber, but here flanging is controlled by input level. The flange effect is most prominent when the input level drops. This lets you process drums with little or no reverb flanging while they are active, but individual hits and fills will have pronounced flanging during the reverb tail. This also works well with a variety of sources including acoustic guitar and piano.

**3.3 Wizz and Wazz** **ADJUST: Decay 0–127**  
This tempo driven effect has delays that whiz from left to right. FX Mix is also linked to tempo, going slowly from delays to reverb, then abruptly back to delays. ADJUST controls reverb decay.

**3.4 Sci Fi** **ADJUST: Wiggle 0–127**  
This preset uses an LFO modulated reverb for its basic effect. ADJUST controls the depth of the modulation. It will put a big spacious halo around unpitched sources, like drums. Lower settings of ADJUST will work best with pitched sources, although higher settings can yield some interesting sound effects. (Listen to a high piano note with ADJUST at about 100 –definite science fiction material!)

**3.5 Wobble Plate** **ADJUST: Wobble 0–30**  
A different type of chorus plate. ADJUST controls glide response.



**Remix Effects  
(P3 4.0 – P3 4.9) cont'd.**

**4.5 ADJToFreeze** **ADJUST: Freeze It 0–1**  
Click ADJUST to create an infinite stereo loop of the input source. Click it again to stop the loop. A little reverb is added while the loop is on. The loop size is tempo controlled. Change it on the fly by pressing **Tap** while the loop is off. The preset is set to freeze 1 beat, but you can set any rhythm you want by changing the Soft Row delay parameter values.

**4.6 ADJToFreeze2** **ADJUST: Freeze It 0–1**  
The same basic loop effect as ADJToFreeze, but the delays are set to different values for the left and the right side. In this preset a 2 against 3 pattern will be created.

**4.7 FSw2 Freeze** **ADJUST: Feedback 0–100**  
A very useful infinite delay effect. Press **Tap** to establish the tempo. Use ADJUST to set the feedback level. Foot switch 2 is patched to turn infinite repeat on and off.

**4.8 FSw2 Speedup** **ADJUST: How Long? 0–100**  
Footswitch 2 is patched to turn on an infinite loop while it is held down. When the foot switch is released, the loop continues but grows shorter with every repeat until the loop size is 0. Use ADJUST to set the amount of time it takes for the loop to shrink to nothing. *A footswitch must be connected to make this preset work.*

**4.9 LongestLoop** **ADJUST: FX Mix 0–100**  
This preset uses the cross-feedback path in the delay voices to create a 5-second mono loop. Footswitch 2 is patched to turn infinite repeat off and on and, at the same time, to change the mix from 0 % to 100 % wet. Foot switch 1 is patched to clear the delays. ADJUST sets the reverb level for the loop. 0 = no reverb, 127 = all reverb. The Soft Row contains the parameters for setting the master delay for the loop as well as several reverb parameters. *A footswitch must be connected to make this preset work.*

**Program Bank 4 (P4)**

NOTE: Most of the Pitch presets in Banks 4 and 5 are set to 100% wet. To hear the harmony/detune in many presets, dry signal should be added from the mixer. If the PCM 81 is connected in line, adjust the Pitch FX Mix parameter (**Control 0.0**) to add some dry signal.

**Vocal Shift** **0.0 FixYerVox** **ADJUST: Offset 0–100**  
**(P4 0.0 – P4 0.9)** Turns your marginal singer into a rock'n'roll star. Bounce your vocals to another track and fix the pitch on the fly. Stereo In/Stereo Out. ADJUST gives you  $\pm 100$  cents of offset to correct any other intonation.

**0.1 FixYerVoxChm** **ADJUST: Offset 0–100**  
Similar to FixYerVox with an added vocal chamber for occasions when you run out of tracks and need to fix the pitch while you mix. Stereo In/Stereo Out.

**0.2 ThickFixVox** **ADJUST: Thickness 0–50**  
Similar to FixYerVox with stereo detuning for occasions when you run out of tracks and need to fix the pitch while you mix. Stereo In/Stereo Out. ADJUST controls the amount of detuning, in cents.

**0.3 Cocomo** **ADJUST: Thickness 0–100**  
The inputs are detuned. The left is delayed 31ms, the right is delayed 47ms. ADJUST controls the amount of + and – detuning in cents. Stereo In/Stereo Out.

**0.4 QuadDTune** **ADJUST: Tune Knob 0–100**  
Fatten up vocals with four pitch shift voices panned across the stereo field. ADJUST sets the amount of detuning. (Two voices go sharp, two go flat.)

- 0.5 ADTStereoIn** **ADJUST: LFO Rate 0–100** **Vocal Shift (P4 0.0 – P4 0.9) cont'd.**  
The left input is pitch shifted up, the right input is shifted down. An LFO modulates the pitch from 2-12 cents. Stereo In/Stereo Out. ADJUST controls the LFO rate from 0-25Hz.
- 0.6 TripleTrack** **ADJUST: Separate 0–100**  
Adds two vocal tracks. ADJUST controls how closely the pitch shifted voices track the pitch and timing of the input track.
- 0.7 Quad Track** **ADJUST: Separate 0–100**  
Similar to TripleTrack, this preset adds a third doubling to the original track. ADJUST controls how tightly the three shifted voices track the original.
- 0.8 Quint Track** **ADJUST: Separate 0–100**  
Makes one voice sound like five by adding four double tracks to the original. Each one is slightly out of time and out of tune with the others. ADJUST controls how tightly the shifted voices track the original.
- 0.9 Add Voices** **ADJUST: How many? 1–4**  
A variation of QuintTrack with ADJUST adding more doublings to the original. 1=1 voice with slight detune, 2=2 voices, 3=3 voices, 4=4 voices.
- 1.0 Major Thirds** **ADJUST: Inversion 0–4** **Vocal Harmony (P4 1.0 – P4 1.9)**  
This preset allows you to treat the input note as the root or third in major third harmony. ADJUST lets you select the inversion. 0=slight detune, 1=sixth below, 2=third below, 3=third above, 4=sixth above.
- 1.1 Minor Thirds** **ADJUST: Inversion 0–4**  
This preset allows you to treat the input note as the root or third in minor third harmony. ADJUST lets you select the inversion. 0=slight detune, 1=sixth below, 2=third below, 3=third above, 4=sixth above.
- 1.2 Major Triads** **ADJUST: Inversion 0–6**  
The input note can be harmonized as the root, third or fifth of a major chord. ADJUST selects the harmony and inversion. 0=slight detune, 1=input is root (1st inversion), 2=input is root (2nd inversion), 3=input is third (1st inversion), 4=input is third (2nd inversion), 5=input is fifth (1st inversion), 6=input is fifth (2nd inversion).
- 1.3 Minor Triads** **ADJUST: Inversion 0–6**  
The input note can be harmonized as the root, third or fifth of a minor chord. ADJUST selects the harmony and inversion. 0=slight detune, 1=input is root (1st inversion), 2=input is root (2nd inversion), 3=input is third (1st inversion), 4=input is third (2nd inversion), 5=input is fifth (1st inversion), 6=input is fifth (2nd inversion).
- 1.4 Major & Dbls** **ADJUST: Inversion 0–4**  
The input note is doubled and also harmonized as the root of a major triad. ADJUST selects the inversion of the chord.
- 1.5 Minor & Dbls** **ADJUST: Inversion 0–4**  
The input note is doubled and also harmonized as the root of a minor triad. ADJUST selects the inversion of the chord.
- 1.6 4ths & Dbl** **ADJUST: Inversion 0–4**  
The input note is doubled and also harmonized as the root of chord made of four perfect fourths. ADJUST selects the inversion of the chord.

**Vocal Harmony**  
(P4 1.0 – P4 1.9) cont'd.

**1.7 MIDI Pitch** **ADJUST: Rvb Mix 0–100**  
Use this preset to harmonize source material via single notes played on a MIDI keyboard. The amount of pitch shift is determined by the size of the interval played above or below middle C. ADJUST allows you to mix in some reverb. Note that the audio source should not be the same source used to generate the MIDI notes.

**1.8 MIDI Harmony** **ADJUST: Rvb Mix 0–100**  
This is a useful variation of MIDI Pitch. Play two notes at once on a MIDI keyboard to add two voices to the input note to create three part harmony. The harmonies are relative to middle C. ADJUST allows you to mix in some reverb. Note that the audio source should not be the same source used to generate the MIDI notes.

**1.9 MIDI Chord** **ADJUST: V2 Pitch 1–25**  
A variation of MIDI Pitch that allows you to create four part harmonies. You can add two notes to the input note from a MIDI keyboard and dial-in an additional fixed harmony with ADJUST which provides 1 octave up or down in half steps. Note that the audio source should not be the same source used to generate the MIDI notes.

**Guitar Magic**  
(P4 2.0 – P4 2.9)

**2.0 SingleDetune** **ADJUST: 10=unison 0–20**  
A high quality detuner. ADJUST shifts pitch up or down 10 cents. Detune is at unison when ADJUST=10.

**2.1 DualDetune** **ADJUST: cents +/- 0–100**  
A classic guitar effect. Two pitch shifters are used to spread out and thicken up the sound. ADJUST controls the amount of shift in cents. One side shifts up, the other shifts down.

**2.2 Phat Detune** **ADJUST: TuneKnob 0–100**  
A classic detune effect with two additional voices. The four voices are panned across stereo space to really widen up the sound. ADJUST controls the overall amount of detuning.

**2.3 EGtrPitchRm** **ADJUST: Fbk Mix 0–50**  
This preset produces a rich, room-like halo around every note. Sounds like reverb, but it's not. How can you tell? Choke off a note and you'll hear that there's no decay. The effect is created by using pitch shifted echoes with cross feedback. The input level controls the amount of cross feedback. When the input dies out, the X-feedback goes to 0. ADJUST lets you dial some back in.

**2.4 VibroShift1** **ADJUST: Rate 0–100**  
A simple unison vibrato with ADJUST controlling the rate. Mix with the original signal for chorusing or use 100% wet for straight vibrato.

**2.5 VibroShift2** **ADJUST: Rate 0–100**  
Similar to VibroShift1 with a touch of octave down vibrato. Nice on single notes or chords.

**2.6 VibroTrem** **ADJUST: Rate 0–100**  
Pitch shift vibrato and out-of-phase tremolo combined to create a new take on a retro sound. ADJUST controls the rate of the throb. For the fullest effect, don't mix any dry signal with the PCM 81 output.

**2.7 VibroTremRvb** **ADJUST: Rate 0–100**  
Another cool take on a vintage effect with reverb added between the vibrato and the tremolo. Play some chunky chords and listen to the verb pump. ADJUST controls the rate.



<b>Percussion &amp; Tempo</b> (P4 4.0 – P4 4.9)	<b>4.0 SnareTools 1</b>	<b>ADJUST: Crack</b>	<b>0–100</b>
	Natural snare reverb with ADJUST providing “crack” control. Detune is added to the original input before both are sent to the reverb. FX Mix controls the blend of dry vs. pitched signal to achieve optimal “crack”. Stereo In/Stereo Out.		
	<b>4.1 SnareTools 2</b>	<b>ADJUST: Crack</b>	<b>0–100</b>
	Trippy backwards effect. Reverb feeds the pitch shifter. Mono In/Stereo Out.		
	<b>4.2 SnareTools 3</b>	<b>ADJUST: Crack</b>	<b>0–100</b>
	Inverse snare reverb with “crack” control via ADJUST. Detune is added to the original input before both are sent to the reverb. FX Mix controls the blend of dry vs. pitched signal to achieve optimal “crack”. Mono In/Stereo Out.		
	<b>4.3 TomTomTools</b>	<b>ADJUST: Boom!</b>	<b>0–100</b>
	Turns oatmeal boxes into cannons. Stereo In/Stereo Out.		
	<b>4.4 Slap Up BPM</b>	<b>ADJUST: pitch</b>	<b>0–100</b>
A tuned percussion plate with a slap predelay. Great for unpitched percussion. Use ADJUST to tune the pitch of the reverb above the pitch of the percussion.			
<b>4.5 Slap Dn BPM</b>	<b>ADJUST: pitch</b>	<b>0–100</b>	
A variation with ADJUST tuning the reverb below the pitch of the percussion source.			
<b>4.6 Thick Slap</b>	<b>ADJUST: detune</b>	<b>0–100</b>	
This preset detunes the reverb above and below the original source. A great alternative for thickening drum sounds.			
<b>4.7 RepeatSlaps</b>	<b>ADJUST: Fbk</b>	<b>0–100</b>	
Pitched reverb echoes produce arpeggios. Try this with any short percussive track for an interesting effect. ADJUST controls the amount of feedback (decay of the arpeggio).			
<b>4.8 AutoScratch</b>	<b>ADJUST: Rvb Mix</b>	<b>0–100</b>	
Run a rhythm bed or drum machine into this preset and it will “scratch it” in tempo. ADJUST adds reverb. Press <b>Tap</b> twice to lock in the rhythm.			
<b>4.9 Stop n Go</b>	<b>ADJUST: Rvb Mix</b>	<b>0–100</b>	
A more radical version of AutoScratch.			

## Program Bank 5 (P5)

<b>Sci-Fi FX</b> (P5 0.0 – P5 0.9)	<b>0.0 PullThePlug!</b>	<b>ADJUST: Off/On</b>	<b>0–1</b>
	Use this special effect to simulate the sound of a tape machine, juke box (etc.) running down when the power has been cut off in the middle of a tune. Turn ADJUST from 1 to 0 to cut the power, and from 0 to 1 to turn it back on.		
	<b>0.1 Stargate</b>	<b>ADJUST: Velocity</b>	<b>0–100</b>
	A dramatic special effect that transforms a single percussive sound into a spatial wash of ascending and descending pitches. Try this with single drum hits or short sound effect samples. ADJUST controls the rate of the ascent/descent.		
<b>0.2 WhiteHole</b>	<b>ADJUST: Big&gt;Small</b>	<b>0–100</b>	
A variation of Stargate that transforms a single sonic event into a randomized series of pitched echoes. ADJUST controls the pitch and delay deviation.			
<b>0.3 MotherShip</b>	<b>ADJUST: Velocity</b>	<b>0–100</b>	
Need some processing for lift-off or acceleration effects? This effect delivers. Drive it with a single impulse or short sound effect to produce a spatial glide that ascends to the limits of audibility. ADJUST controls the rate of ascent.			



**Pitch & Delay**  
(P5 1.0 – P5 1.9) cont'd.

**1.6 FootPdlEkos** **ADJUST: Detune 0–100**  
Designed to be used with a foot pedal. A detune chorus is routed through stereo delays and reverb. The pedal controls the input level to the delays and reverb. Use the pedal to capture specific phrases in the delays/reverb. Great with guitar and other solo instruments.

**1.7 DualDTuneDly** **ADJUST: cents +/- 0–100**  
This preset combines two detuned voices with delays. ADJUST controls the amount of detune. Controls for the delays (and reverb too) are in the Soft row.

**1.8 QuaDTuneDly1** **ADJUST: TuneKnob 0–100**  
For thicker detune and delay effects, this preset uses four pitch shifters for detuning. ADJUST controls the pitch spread of the four shifters. Delay and reverb controls are in the Soft row.

**1.9 QuaDTuneDly2** **ADJUST: TuneKnob 0–100**  
Similar to QuadTuneDly1 with the delays set to rhythmic values (eighths and triplets). Press **Tap** to synchronize the rhythms to the beat.

**Pitch Sequences**  
(P5 2.0 – P5 2.9)

**2.0 Major Across** **ADJUST: MasterDly 0–100**  
A major triad is arpeggiated across the stereo field. ADJUST controls the overall timing of the arpeggiation.

**2.1 Minor Across** **ADJUST: MasterDly 0–100**  
A minor triad is arpeggiated across the stereo field. ADJUST controls the overall timing of the arpeggiation.

**2.2 Dim hARP** **ADJUST: GlissRate 0–100**  
The inputs rise in a diminished arpeggiated scale. This preset turns one harp note into a glissando, or one quarter note into a harp chord. Individual voices are panned across the stereo field. Stereo In/Stereo Out.

**2.3 Dim hARP Vrb** **ADJUST: GlissRate 0–100**  
Similar to Dim hARP with an added reverb.

**2.4 DimScaleDown** **ADJUST: ScaleRate 0–100**  
The inputs descend into a diminished arpeggiated scale. This preset turns one note into a scale or glissando. Individual voices are panned across the stereo spectrum. A little reverb is added.

**2.5 GlassCascade** **ADJUST: MasterDly 0-100**  
A glassy, chime-like tail makes this preset great for sustained single-note melodies. Spooky and pretty at the same time. Stereo In/Stereo Out.

**2.6 Carnival** **ADJUST: MstrFdbk 0–100**  
Turns guitar or piano into steel drums, especially if you play diatonic thirds, as steel drummers have been known to do. Stereo In/Stereo Out.

**2.7 Sequence 1** **ADJUST: MasterFbk 0–100**  
Play single note lines. Adds M6 and M9. Tap tempo-driven. Stereo In/Stereo Out.

**2.8 Sequence 2** **ADJUST: MasterFbk 0–100**  
Similar to Sequence1 with a very different sound.

**2.9 Pentatonics!** **ADJUST: MasterFbk 0-100**  
A sequence of notes from the minor pentatonic scale are played for each input note. The input note is treated as the root of the scale. ADJUST controls the amount of feedback.



**Pitch Correct**  
**(P5 4.0 – P5 4.9)**

With the exception of Demo Correct (4.9), all of the presets are designed to accept a mono vocal track on either (or both) inputs. Note also that, in each preset, the reverb is turned off and all pitch parameters that are not patched can be found in the Soft Row.

- 4.0 VoxFix E2>C6** **ADJUST: On/Off** **0-1**  
 This preset is set up to be used with a MIDI controller with a keyboard, pitch bender and sustain pedal. It provides three different methods to correct pitch problems. You can: use the bender to change the pitch "on the fly," use the sustain pedal to turn correction on and off, or fix a bad note by playing the correct one on the keyboard. Press **Edit** to see the notes displayed. ADJUST is patched to turn correction on and off. All of the pitch parameters that aren't patched are in the Soft Row. Low Note and High Note are tuned to the full vocal range (E2 through C6).
- 4.1 VoxFix Bass** **ADJUST: On/Off** **0-1**  
 The same as VoxFix E2>C6, except that it is tuned to the bass range (E2-A4).
- 4.2 VoxFix Tenor** **ADJUST: On/Off** **0-1**  
 The same as VoxFix E2>C6, except that it is tuned to the tenor range (C3-C5).
- 4.3 VoxFix Alto** **ADJUST: On/Off** **0-1**  
 The same as VoxFix E2>C6, except that it is tuned to the alto range (F3-F5).
- 4.4 VoxFix Sprno** **ADJUST: On/Off** **0-1**  
 The same as VoxFix E2>C6, except that it is tuned to the soprano range (C4-C6).
- 4.5 KnobCentsUp** **ADJUST: Sharp** **0-100**  
 For tracks that are consistently flat. Use ADJUST to tune the track up, and a MIDI keyboard to fix bad notes.
- 4.6 KnobCentsDN** **ADJUST: Flat** **0-100**  
 For tracks that are consistently sharp. Use ADJUST to tune the track down, and a MIDI keyboard to fix bad notes.
- 4.7 Double Effect** **ADJUST: On/Off** **0-1**  
 This preset produces an alternative double track effect. Mix the output of the PCM 81 with the original vocal track. Use ADJUST to randomize the amount of delay and pitch difference between the original and processed tracks.
- 4.8 Knob Freeze** **ADJUST: On/Off** **0-1**  
 This one is a special effect. Turn ADJUST from 0 to 1 to freeze the pitch of the vocal track to whatever was being sung when the knob was turned. Turn it back down to 0 to release the pitch. Mixed with the original, this creates drone-like harmonies. By itself, this effect is a new take on "robot" voices.
- 4.9 Demo Correct** **ADJUST: unused**  
 This preset is for demonstration use. The left input is turned off and all the pitch parameters are in the Soft Row.

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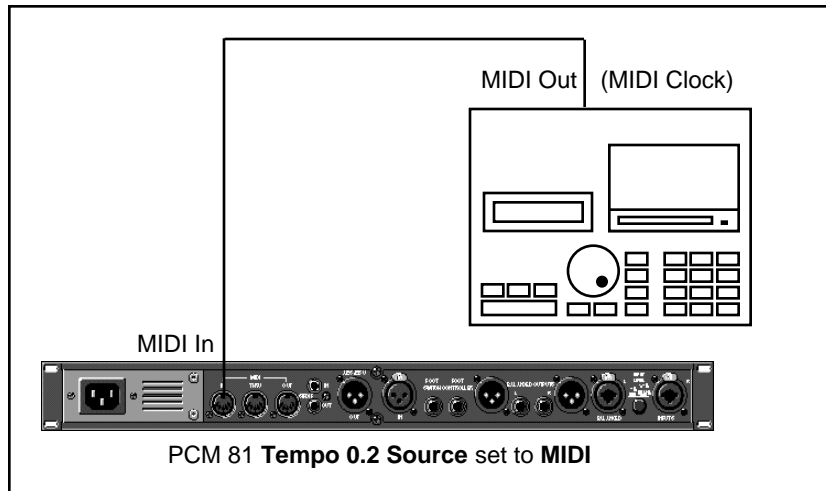
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Alley Slap	P3 2.6	Env Notches	P2 0.7	MIDI Harmony	P4 1.8	RotoRoom#2	P2 2.4	Thick>Ducked	P5 1.5
AmbientCarBy	P1 4.4	Env:PanKorus	P0 2.5	MIDI Pitch	P4 1.7	RotoRoomVox	P2 2.2	Thunder FX	P0 3.7
AR DlyVerb	P5 1.3	EnvChamber	P3 3.2	MIDI Rotary	P0 3.2	RotoRox	P2 2.1	Thunder FX 2	P0 3.8
AutoFadeln 1	P2 3.4	EnveloVerb	P3 3.9	MIDlchords	P2 4.6	RotorWaa	P2 1.0	Tight Space	P0 1.1
AutoFadeln 2	P2 3.5	Evil Echoes	P5 0.9	MIDlSustain	P2 4.9	RotoWood	P0 3.4	Tiled Room	P3 0.6
AutoFadeln 3	P2 3.6	Evil Voice	P5 0.8	Minor & Dbls	P4 1.5	RoughIdle FX	P0 4.1	Tiled Rotary	P0 3.3
AutoInfinite	P0 4.5	FinishLine	P1 4.9	Minor Across	P5 2.1	Round 147	P0 2.9	Tomb Room	P1 4.0
AutoScratch	P4 4.8	FixYerVox	P4 0.0	Minor Thirds	P4 1.1	RoxStutter	P0 4.2	TomTomTools	P4 4.3
Backstage	P2 1.7	FixYerVoxChm	P4 0.1	Minor Triads	P4 1.3	Rvb On L-R	P0 4.9	Too Deep!	P2 1.9
BandEko Rvb	P1 2.6	Flange+Rvb	P0 1.3	MirrorChorus	P5 3.3	Sci Fi	P3 3.4	Toon Voices	P5 0.7
BandEkoSweep	P1 2.7	Flange>Rvb	P0 1.2	Mod Max	P0 2.7	Sequence 1	P5 2.7	Tremolo Tap1	P2 2.5
BandReject 4	P2 0.8	FootPdl Oct	P4 3.8	ModalChords1	P4 3.4	Sequence 2	P5 2.8	TripleTrack	P4 0.6
BigBoomRoom	P3 2.9	FootPdlChord	P4 3.9	ModalChords2	P4 3.5	Shuffle Ekos	P1 0.5	TryTalk'n	P0 4.4
BowedChords	P2 3.8	FootPdlEkos	P5 1.6	Modallmpulse	P2 4.4	Shuffles	P1 0.6	Under Water	P0 3.6
BowedEchoes	P2 3.9	Freez 2 of 8	P3 4.2	Mono VSO	P5 3.9	SingleDetune	P4 2.0	V-Eliminate	P0 4.7
Brass Plate	P3 1.5	Freez 2over3	P3 4.3	MotherShip	P5 0.3	Six Across	P1 2.5	Verbato	P5 3.7
Brick Kick	P3 0.3	Freeze&Speed	P3 4.4	Movable Echo	P2 1.1	Six Trips	P1 0.8	Vibrato BPM	P5 3.6
Carnival	P5 2.6	FSw2 Elevate	P0 2.0	Movable Hall	P2 1.2	Slap Dn BPM	P4 4.5	VibroShift1	P4 2.4
ChaosImpuls	P0 3.9	FSw2 Freeze	P3 4.7	MultiFx Fade	P2 3.0	Slap Plate	P3 1.4	VibroShift2	P4 2.5
ChaseEcho	P1 2.9	FSw2 Rotary	P0 3.0	Nice Pan!	P2 2.8	Slap Up BPM	P4 4.4	VibroTrem	P4 2.6
CheapTV Room	P1 3.8	FSw2 Speedup	P3 4.8	NoCenter Eko	P0 4.8	Sliding Eko	P1 4.7	VibroTremRvb	P4 2.7
Chord Walk	P2 4.2	Funkus Room	P0 0.8	NonLinear 1	P1 2.2	Slope Down	P3 2.8	VibroVerb	P3 3.7
ChordSwells	P2 3.7	Gate Chamber	P3 1.1	NonLinear 2	P1 2.3	Small Room	P3 0.1	Vocal Hall	P3 2.3
Chorus Plate	P0 0.5	Ghost	P2 3.1	NonLinear 3	P1 2.4	Small+Stage	P3 0.0	Vox Chamber	P3 0.8
Chorus Room	P0 0.6	Ghost Flange	P2 3.2	OctDown Verb	P4 2.8	SnareChamber	P3 0.5	Vox Plate	P3 1.2
Chorus&Amb	P0 2.2	GhostVibrato	P2 3.3	OffBeat Eko	P1 0.2	SnareTools 1	P4 4.0	VoxFix Alto	P5 4.3
Chorus&Pan	P0 2.1	GlassCascade	P5 2.5	OffBeatEko2	P1 0.3	SnareTools 2	P4 4.1	VoxFix Bass	P5 4.1
ChorusToRvb	P0 0.7	Glide > Verb	P0 1.5	OneShotCarBy	P1 4.3	SnareTools 3	P4 4.2	VoxFix E2>C6	P5 4.0
Chromatic Dn	P4 3.7	Glide X-Ekos	P0 1.6	Pan>Eko BPM	P1 1.8	Spatial Hall	P2 1.5	VoxFix Sprno	P5 4.4
Chromatic Up	P4 3.6	Go Away	P2 1.3	Panned Dlys	P1 3.0	SpatialDuck	P5 3.5	VoxFix Tenor	P5 4.2
Circular Pan	P2 1.4	Good olPlate	P3 1.3	Panner BPM	P2 2.7	Spin & Duck	P2 2.9	WaaPedalEko	P2 0.9
Cocoma	P4 0.3	Haas PanKnob	P1 3.4	Pedal Swell	P2 4.0	Split C&E	P0 2.4	Wet Chorus	P0 0.2
Comb Room	P1 4.1	HeadOn CarBy	P1 4.5	Pentatonics!	P5 2.9	Split Pitch?	P0 4.3	Wet Chorus 2	P0 0.3
Concert Hall	P3 1.9	HonkyTonkPno	P4 3.1	Phat Detune	P4 2.2	Stargate	P5 0.1	Wet Chorus 3	P0 0.4
Concert Wave	P3 1.8	In The Air	P5 1.4	PhoneOrRoom?	P1 3.7	Steered Rear	P2 1.8	Whammy Hall	P3 3.0
ConcertHall2	P3 2.0	IntoTunnel	P1 4.7	Piano Hall	P3 2.1	Stereo Notch	P2 0.5	WhiteHole	P5 0.2
Deep Space	P3 2.4	JetChamber	P3 3.1	Pitch Across	P5 1.0	Stereo VSO	P5 3.8	Wide Chamber	P3 1.0
Demo Correct	P5 4.9	JurassicSolo	P5 3.0	PitchEkoRvb	P5 1.1	StereoB-Pass	P2 0.4	Wizz & Wazz	P3 3.3
DemonDescent	P0 4.0	Knob CentsDn	P5 4.6	PitchPong	P5 1.2	StereoEqEkos	P0 1.8	Wobble Plate	P3 3.5
Detune&Dbl	P0 1.0	Knob CentsUp	P5 4.5	PitchSlapRvb	P4 2.9	StereoHiPass	P2 0.3	X Eko Flange	P0 1.4
Detune&Room	P0 0.9	Knob Freeze	P5 4.8	Plate4 Horns	P3 2.5	StereoLoPass	P2 0.0	X-Pan Delays	P1 3.1
Detuned Ekos	P0 1.7	Large Room	P3 0.4	PowerNotes	P4 3.2	StereoTapDly	P1 0.1	X-PanEQ BPM	P1 1.7
Dial a Delay	P1 3.5	LatchedEkos	P1 1.6	PreciseGlide	P0 2.8	Stop n Go	P4 4.9	X-TremoloTap	P2 2.6
DialogCloner	P5 0.6	LevlStutter	P3 4.1	PrecisionDly	P1 3.6	Super Ball!	P3 4.0	Xplode Up	P5 0.5
Dim hARP	P5 2.2	LFO Detune	P5 3.4	Prime Blue	P0 0.0	Sweeper	P2 4.8	XplodeDown	P5 0.4
Dim hARP Vrb	P5 2.3	LFO EQ Echo	P1 2.8	PullThePlug!	P5 0.0	SweepVerb	P3 3.8	Zoom Over	P1 4.2
DimScaleDown	P5 2.4	Living Room	P3 0.2	Quad Track	P4 0.7	Sweet Chorus	P5 3.2	[n]Ekos/Beat	P1 0.0



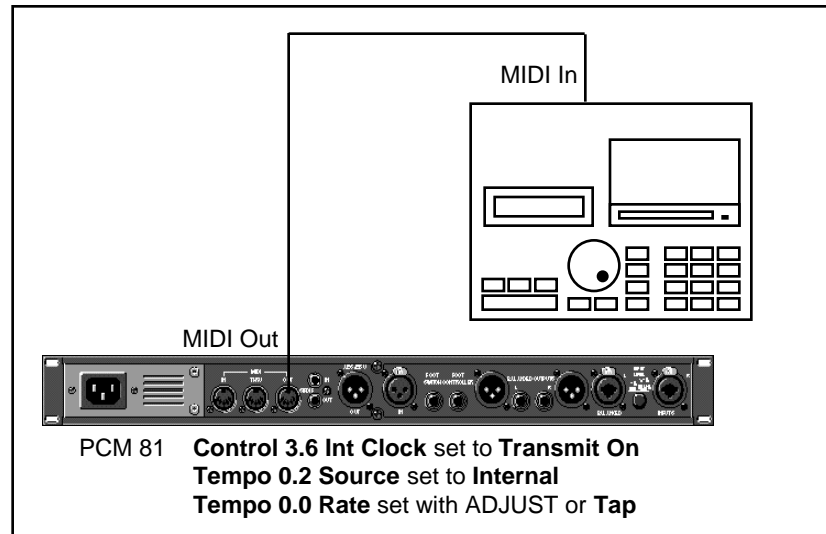


## Controlling PCM 81 Tempo Rate with MIDI Clock

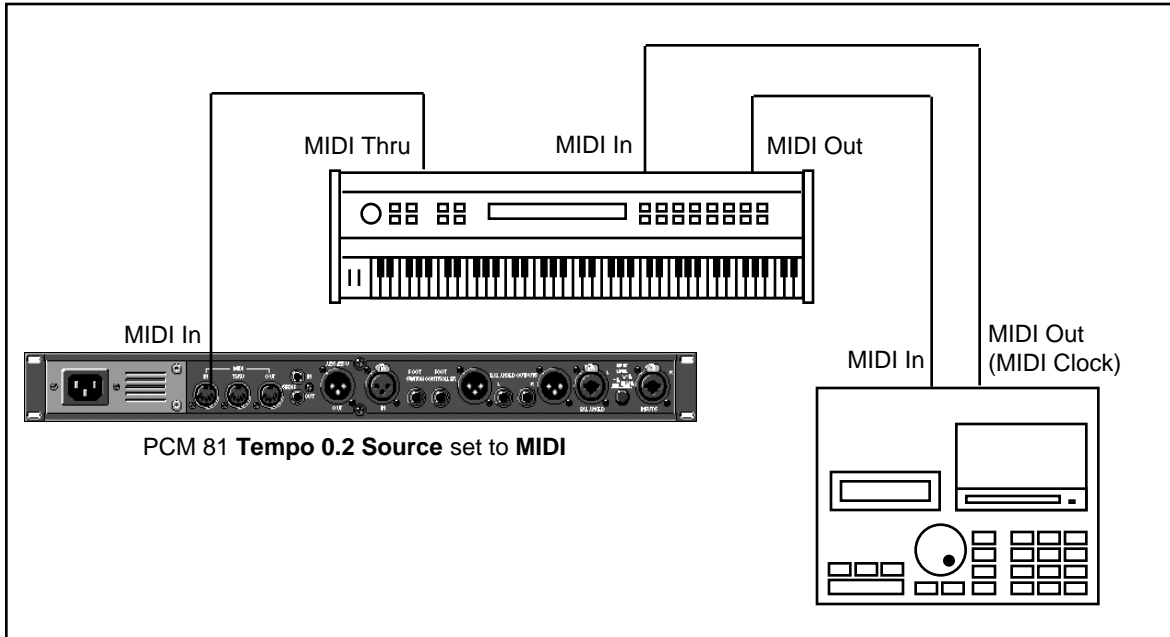
### MIDI Tempo Control



### Using the PCM 81 as a MIDI Clock Source

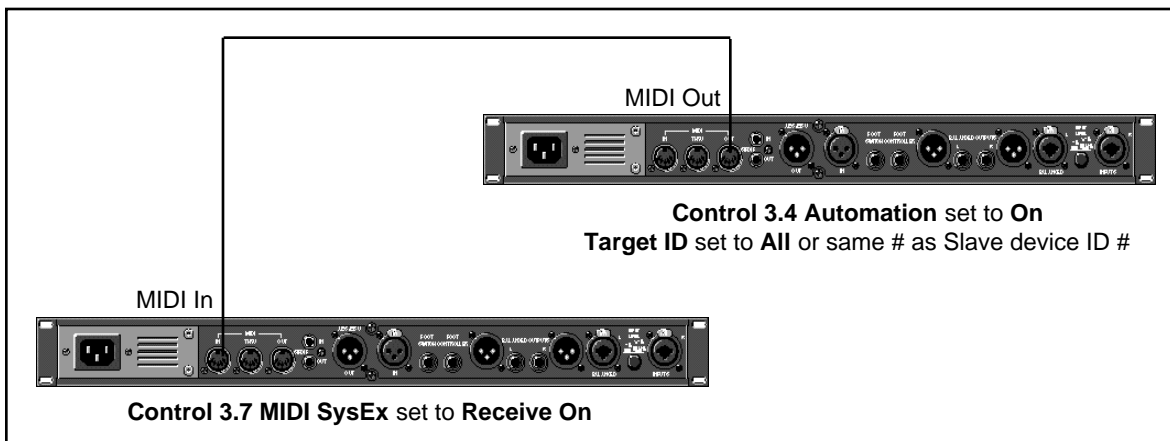


The configuration below shows the MIDI connections for controlling the PCM 81 simultaneously with MIDI Clocks from a sequencer, and messages from another MIDI controller. Note that the controller is set to "local control off" and the sequencer is set to "echo input".



Two PCM 81s can be slaved together by connecting a cable from the MIDI OUT jack of the master to the MIDI IN jack of the slave. Additional PCM 81s can be slaved to the master by connecting a cable from the MIDI THRU port of one slave unit to the MIDI IN port of the next unit. All of the PCM 81s must be set to the same MIDI channel.

**Slaving two or more PCM 81s**



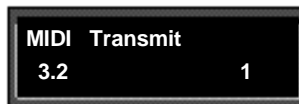
**Controller quirks**

Some synthesizers and controllers cannot send the full range of MIDI program change messages (1-128). Others may appear to be able to send only 32, but actually have a bank mode that *does* let you send all 128 program change messages. Also, be aware that some MIDI devices use a program numbering system that uses 0-127 instead of 1-128. If in doubt, see the manual for your controller.

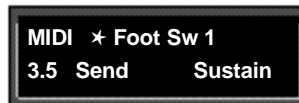
**The ADJUST Knob, Foot Pedal, Foot Sw 1 and Foot Sw 2 as MIDI Controllers**

You can choose to have the PCM 81 send MIDI Controller messages whenever you turn ADJUST (the soft knob in Program Banks or Register Banks modes), or to activate analog controllers connected to the rear panel Footswitch or Foot Controller jacks. This makes it possible to record real-time control of PCM 81 effects with a MIDI sequencer — A simple but quite powerful way to automate effects.

To send MIDI data from these controllers, first set Control mode 3.2 (**Transmit**) to the desired MIDI Channel. (The default is Channel 1.)



Once a transmit channel has been set, go to Control mode 3.5 to assign MIDI Controllers. Press **Load/\*** to display the available controllers: **Foot Pedal, Foot Sw1, Foot Sw2** or **ADJUST**. Turn ADJUST to assign the MIDI Controller data to be sent when the displayed controller is activated. (The default assignment is **None**.)

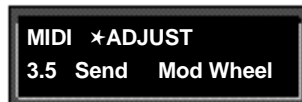


When a PCM 81 controller is assigned to a MIDI Controller, the PCM 81 will respond to incoming controller messages as though its own controller were moved. In other words, if **FootSw1** is assigned to **Sustain**, the PCM 81 will respond to incoming Sustain messages as though FootSw1 had been activated.

Each PCM 81 preset has a unique soft knob patch that allows you to control the effect directly from Program or Register Banks mode with the ADJUST knob. You can also control the soft knob patch remotely from MIDI, or from the Foot Pedal.

### Controlling the Soft Knob with MIDI

To control the soft knob with MIDI, set Control mode 3.1 (**Receive**) to the desired MIDI Channel. Set Control mode 3.5 (**ADJUST**) to the desired MIDI Controller such as **Mod Wheel**.

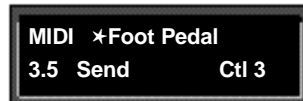
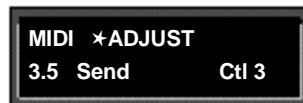


Now, the Mod Wheel on the MIDI instrument will control the soft knob patch of the running effect.

If you have a foot pedal connected to the PCM rear panel Foot Controller jack, you can use it to control the soft knob patch. (Note that no MIDI connections are required to do this.)

### Controlling the Soft Knob with a Foot Pedal

Set both Control mode 3.5 **ADJUST** and Control mode 3.5 **Foot Pedal** to the same MIDI Controller.



Now, the foot pedal will control the soft knob patch of the running effect.

**Program Change Messages**

Reception of MIDI Program Change and Bank Select messages can be selectively enabled/disabled from Control Mode parameter 3.0, MIDI Pgm Change. The manner in which the PCM 81 interprets these messages is determined by the value of this parameter as follows:

**Pgm Change: Off**

All Program Change and Bank select messages are ignored. Pgm+ and Pgm- will load the next higher or lower program in the current bank.

**Pgm Change : On**

Program Change messages 0-49 correspond to PCM 81 Effects 0.0 -4.9 in the current bank. Program Change messages 50—127 are ignored. Pgm+ and Pgm- will load the next higher or lower program in the current bank.

The current bank can be changed with MIDI Bank Select Messages as follows:

- 0-5: Program Banks 0-5
- 6: Internal Register Bank
- 7-11: reserved
- 12-58: Memory Card Banks

The number of banks available on a given card will vary with card size as follows:

Card Size	# Banks
64	2
256	11
512	23
1 Meg	47

**Pgm Change: Map**

Program Change 0-127 can be mapped to any PCM 81 Effect in any internal or card bank. Two 128 element maps are stored internally, additional maps may be stored on RAM cards.

Map 0	Map 1
MIDI 0 = P0 0.0	MIDI 0 = P2 2.8
MIDI 127 = P2 2.7	MIDI 127 = R 4.9

Pgm+ and Pgm- will load the next higher or lower program in the map.

**Pgm Change: Chain**

Any Program Change number can be selected to load any one of ten customized effect "chains." Once a chain is loaded, effects in the chain are accessed by the controller patched to Pgm + and Pgm - (program increment and program decrement).

## Automation

### SysEx Automation

The PCM 81 will transmit SysEx automation messages when Control Mode parameter 3.4, MIDI Automation is set to On. All changes made by front panel operations are transmitted as PCM 80 SysEx messages. This is intended primarily for use by editor programs and in configurations where it is desirable for one or more PCM 81s to be slaved to a single PCM 81 acting as a master. The current mode (Program Banks, Register Banks, Edit, Control or Tempo) of the slave does not follow the master, but the actual parameter values do. (SysEx automation can also be stored on a sequencer and replayed in real-time. As a general rule, automating more than two or three SysEx program changes at once is not recommended.) Because messages are transmitted as PCM 80 messages, either PCM 80s or PCM 81s can be slaved to the master.

The PCM 81 can receive SysEx messages when Control mode 3.7 MIDI SysEx is set to **Receive On**. Note that when using SysEx automation, the device of the receiving PCM 81 must match the Target ID of the transmitting PCM 81.

The Target ID setting can be adjusted from Control 3.4 MIDI Automation. It is selected by pressing **Load/★** after setting Automation to **On**. The default setting for the Target ID is **All**.

The Device ID setting can be adjusted from Control mode 3.7 MIDI SysEx. It is selected by pressing **Load/★** after setting SysEx to **Receive On**. The default setting for the Device ID is **0**.

### Controller Automation

For applications where it is desirable to "automate" changes made to PCM 81 effects with its own controls ( ADJUST knob, Foot Pedal, Footswitch 1 or Footswitch 2), we recommend assigning the controllers to MIDI destinations and recording the changes with a MIDI sequencer (see Control Mode parameter 3.5, MIDI Destinations).

### Reset All Controllers

The PCM 81 recognizes the "Reset All Controllers" message. When received, all patched parameters are reset to their stored values. Patched parameters may also be reset from the PCM 81 front panel —Control Mode parameter 3.0 (the message will be transmitted from the PCM 81 as well).

### MIDI Clock and Clock Commands

The PCM 81 recognizes MIDI clock messages when Tempo Mode parameter 0.2, Tempo Source is set to MIDI. Any Delay or LFO parameter set to display tempo values will be synchronized to the tempo of the incoming MIDI clock.

MIDI Clock and Clock Commands are also available as Dynamic MIDI patch sources. The value of MIDI Clock when used as a patch source is a linear scaling of 0 to 127 (0 = 40 BPM and 127 = 400 BPM). The value of Clock Commands when used as a Dynamic MIDI patch source is 1 for START and CONTINUE and 0 for STOP.

### PCM 80 Compatibility

The PCM 81 can receive MIDI data from either the PCM 80 or another PCM 81. It can also transmit certain messages in PCM 80 format. (See *Bulk Data Dumps* and *SysEx Automation*.) This allows most data to be exchanged between the two products, with any format translations handled automatically by the PCM 81.

**Dynamic MIDI**

The following MIDI messages are available as Dynamic MIDI patch sources:

- MIDI Controllers 1-119
- Pitch Bend
- After Touch (Polyphonic and Channel combined)
- Velocity (Note On)
- Last Note
- Low Note
- High Note
- Tempo (40–400BPM is converted to controller range 0-127)
- Clock Commands

These MIDI messages are also available as threshold sources for several Modulation parameters: AR Env, Latch, Sw 1 and Sw 2. They may also be used as a tap source for controlling Tempo.

MIDI Implementation Details, including System Exclusive documentation, are available to assist experienced programmers in developing software for use with the PCM 81. These can be obtained directly from Lexicon.  
Request: PCM 81 MIDI Implementation Details.

**Bulk Data Dumps**

Control mode 3.8 (**MIDI Dump**) allows selection of the following types of bulk data to be dumped directly from the PCM 81 to another PCM 81, or to editor/librarian software.

Displayed Name	Description
CurrentPgm*	Currently running effect
Bank R*	Internal Register Bank
Bank C0-CK*	Card Banks (card must be inserted)
Map 0, 1*	Internal Program Change Maps
Map 2-33*	Card Program Change Maps (card must be inserted)
Chain 0-9*	Internal Program Chains
Chain 10-19*	Card Program Chains (card must be inserted)
Int Chains*	All Internal Program Chains
Ext Chains*	All Card Program Chains (card must be inserted)
Setup C	Current Setup
Setup 0-4	Internal Setups
Setup 5-9	Card Setups

\* Transmitted in PCM 80 format

Use ADJUST to select the bulk data type. Press **Store** to transmit the data .

## MIDI Implementation Chart

Lexicon PCM 81  
Digital Effects System

Function		Transmitted	Recognized	Remarks
Basic Channel	Default Changed	1 1-16	1 1-16	
Mode	Default Messages Altered	X X	Mode 1, 3 X X	
Note Number		X	0-127	Last Note, Low Note, High Note used as controllers
Velocity	Note ON Note OFF	X	O 9n v = 1-127	used as controller
After Touch	Keys Channel	X X	X O	
Pitch Bend		X	O	
Control Change	1-119	OX	OX	ADJUST, Footpedal, Footswitch 1, and Footswitch 2 can be assigned controllers 1-119 for MIDI transmit
Program Change	True #	X	0-127	See Implementation Details
System Exclusive	Lexicon Real-time non Real-time	OX X X	OX X OX	mfgr ID=6; product ID=7 device ID
System Common	:Song Pos :Song Sel :Tune	X X X	X X X	
System Real Time	:Clock :Commands	OX X	OX OX*	START, STOP and CONTINUE are patchable as a switch: START/CONTINUE=127; STOP=0
Aux Messages	:Local ON/OFF :All Notes OFF :Active Sense :Reset All Controllers	X X X OX	X O X OX	
Notes: PCM 81 transmits and receives in both PCM 80 (product ID 0x07) and PCM 81 (product ID 0x10) formats.				

Mode 1: OMNI ON, POLY  
Mode 3: OMNI OFF, POLY

Mode 2: OMNI ON, MONO  
Mode 4: OMNI OFF, MONO

**O : Yes**    **OX: Selectable**  
**X : No**



# 6

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## Troubleshooting

This chapter is intended primarily to help you recognize some common error states which can be corrected from the PCM 81 front panel, or by simple means such as cable replacement. Any error states which are not covered here should be referred to your local dealer for service by a qualified technician.

In a low-voltage, or "brown-out" condition, the PCM 81 will freeze in its current state. None of the controls will have any effect. When power returns to a normal level, the unit will reset itself as though it had just been powered on. If the unit does not reset itself, turn the power OFF, then ON to resume normal operation.

### Low Voltage

Temperature extremes may cause the PCM 81 to exhibit unpredictable behavior. If the unit has been subjected to temperatures below 32°F (0°C) or above 95°F (35°C), it should be turned off and allowed to return to normal temperature before use. The unit may be damaged by exposure to temperatures below -22°F (-30°C) or above 167°F (75°C), or by exposure to humidity in excess of 95%. If a unit exposed to such conditions fails to operate after it returns to a normal operating temperature, contact your local service representative.

### Overheating

#### **The PCM 81 doesn't respond to MIDI Program Changes.**

Check **Receive** control at Control mode 3.1 and make sure it is set to **On**. Check the MIDI Channel selected as well as the MIDI Channel of the transmitting device. Make sure that **Pgm Change** at Control mode 3.3 is set to **On**. Also check MIDI In/Out connections between the units.

### Common MIDI Problems

#### **MIDI Program Change numbers are off by 1.**

The PCM 81 transmits and recognizes ProgramChange messages 0-127. MIDI devices which transmit 1-128 rather than 0-127, will be off by 1. Simply adjust by 1 when working with such devices.

#### **The PCM 81 doesn't respond to SysEx commands.**

Check the **SysEx** setting (and the Device ID selection) at Control mode 3.7.

#### **The PCM 81 does not transmit SysEx Automation commands.**

Check the **Automation** setting (and the target device ID selection) at Control mod 3.4.

**Operational Problems** **The PCM 81 will not lock onto an incoming digital signal.**  
Check the cables that you are using. **DO NOT USE ANALOG AUDIO CABLE TO CONNECT DIGITAL AUDIO.**

Also check to make sure that your input signal complies with S/PDIF format standards. The PCM 81 will recognize AES professional format signals from an appropriate connector, but will not necessarily read and transmit encoded information accurately.

**No Input**

**Analog:** Check the analog input connection and make sure that the **Analog Lvl** control at Control mode 0.1 is set to 100%.

**Digital AES (XLR):** Check connection to XLR and make sure that the **Digital Lvl** control at Control mode 0.2 is set to 100%. Make sure that **WordClock** (Control mode 0.0) is set to **Ext: XLR 48 or 44.1**.

**Digital S/PDIF (Coax):** Check connection to Coax and make sure that the **Digital Lvl** control at Control mode 0.2 is set to 100%. Make sure that **WordClock** (Control mode 0.0) is set to **Ext: XLR 48 or 44.1**.

**No Digital Audio Output**

Check the **Analog Lvl** and **Digital Lvl** controls at Control mode 0.1 and 0.2.

**No Effects Output**

Check the setting of Mix Mode at Control mode 1.1. Also verify that any controllers patched to **FX Lvl**, **Input Level**, or **Mix** are not turned off.

**Power On Behavior** The PCM 81 performs a series of self tests each time it is powered on, then displays the PCM 81 copyright notice. This should be followed by the display and loading of the last loaded effect. If this sequence does not occur, contact Lexicon Customer Service.

You can restore the PCM 81 to its default state *without* erasing registers by restoring the factory default setup:

### Restoring Factory Default Settings

1. Press **Control**.
2. Use the **Up** and **Down** buttons to locate **Row 4 Setup**.
3. Turn SELECT to **4.1 Load**.
4. Turn ADJUST counterclockwise to select "Factory Settings".
5. Press **Load/★**. The PCM 81 will display the message "Setup restored".

The following table shows the parameters which comprise a setup, along with the factory default setting of each parameter.

<b>Control Mode</b>			
<b>Matrix Location</b>		<b>System Parameter</b>	<b>Default Setting</b>
<b>Audio</b>	0.0	Word Clock	Internal 48kHz
	0.1	Analog Lvl	100%
	0.2	Digital Lvl	0%
	0.3	Word Size	20 bits
	0.4	SCMS	Multi Copy
	0.5	Emphasis Bit	Pass Thru
	0.6	Output Level	+4dBu
<b>System</b>	1.0	Edit Mode	Go
	1.1	Mix Mode	Pgm
		Global Mix Value	100% Wet
	1.2	Tempo Mode	Pgm
		Global Tempo Value	120 BPM
	1.3	Bypass Mode	InputMute
		Bypass Src	Off
	1.4	Pgm Bypass	AllMute
	1.5	Mem Protect	On
1.6	Auto Load	Off	
1.7	Patch Update	Delayed	
<b>MIDI</b>	3.1	Receive	OMNI
	3.2	Transmit	1
	3.3	Pgm Change	On
		Pgm+	Off
		Pgm-	Off
		Map select	0
	3.4	Chain	MIDI
		Automation	Off
	3.5	Footpedal	None
		Sw 1	None
		Sw 2	None
		ADJUST	None
	3.6	Int Clock	Off
	3.7	SysEx	On
		Device ID	0
3.9	Dump Speed	Slow	
<b>Tempo Mode</b>			
<b>Matrix Location</b>		<b>System Parameter</b>	<b>Default Setting</b>
<b>Tempo</b>	0.2	Source	Internal
<b>Tap</b>	1.3	Display	On

**Reinitialization****Reinitializing will erase all registers and setups**

The following procedure will return the PCM 81 to the state it was in when shipped from the factory. This includes erasing all registers and setups, as well as restoring all of the default settings:

1. Press **Control**.
2. Use the **Up** and **Down** buttons to locate **Row 1 System**.
3. Turn SELECT to **1.8 Initialize**.
4. Press **Store**. The PCM 81 will display the message "Are you sure? (Press STORE).

If you don't want to reinitialize your unit, press any button *except Store* to return to matrix position 1.8.

If you press **Store** in response to this message, the display will flash "Restoring original factory settings" and your unit will be reinitialized.

## PCM 81 Specifications

<b>Audio Input</b>	<b>Connectors:</b>	Combined 3 pole XLR and 1/4 inch T/R/S phone jacks (2)
	<b>Impedance:</b>	0 dB/BAL switch position: 100k $\Omega$ , balanced -20 dB/UNBAL switch position: 50k $\Omega$ , unbalanced
	<b>Levels:</b>	0 dB/BAL switch position: -2 dBu min for full scale, +20 dBu max -20 dB/UNBAL switch position: -22 dBu min for full scale, 0 dBu max
	<b>CMRR:</b>	0 dB/BAL switch position: 50 dB minimum, 10 Hz to 20 kHz
<b>Audio Output</b>	<b>Connectors:</b>	1/4 inch T/R/S phone jacks (2); balanced XLRs, pin 2 "high" (2)
	<b>Impedance:</b>	100 $\Omega$ , balanced
	<b>Levels:</b>	+18 dBm, full scale (+4 dBu setting) balanced, unbalanced +4 dBm, full scale (-10 dBu setting)
	<b>Protection:</b>	Relays provided for output muting during power on/off
<b>A/D Performance</b>	<b>Frequency Response:</b>	10 Hz to 20 kHz, $\pm 0.5$ dB
	<b>Crosstalk:</b>	<-65 dB, 10 Hz to 20 kHz
	<b>S/N Ratio:</b>	>102 dB, 20 kHz bandwidth
	<b>THD:</b>	<0.003% max, 10 Hz to 20 kHz
	<b>Dynamic Range:</b>	>102 dB, 20 kHz bandwidth
	<b>Delay:</b>	24 samples (0.54 msec for 44.1 kHz, 0.50 msec for 48 kHz)
	<b>Frequency Response:</b>	10 Hz to 20 kHz, $\pm 0.5$ dB
<b>D/A Performance</b>	<b>Crosstalk:</b>	<-80 dB, 10 Hz to 20 kHz
	<b>S/N Ratio:</b>	>98 dB, 20 kHz bandwidth
	<b>THD:</b>	<0.005%, 10 Hz to 20 kHz
	<b>Dynamic Range:</b>	>98 dB, 20 kHz bandwidth
	<b>Delay:</b>	50 samples (1.13 msec for 44.1 kHz, 1.04 msec for 48 kHz)
	<b>Frequency Response:</b>	10 Hz to 20 kHz, $\pm 0.5$ dB
	<b>A/A Performance</b>	<b>Crosstalk:</b>
<b>S/N Ratio:</b>		>96 dB, 20 kHz bandwidth
<b>THD:</b>		<0.005%, 10 Hz to 20 kHz
<b>Dynamic Range:</b>		>96 dB, 20 kHz bandwidth

<b>Digital Audio Interface</b>	<b>Connectors:</b> Coaxial, RCA type (2); balanced, XLR (2) <b>Format:</b> S/PDIF (IEC-958) and AES/EBU (AES3-1992) professional interface <b>Sample Rates:</b> 44.1 kHz, 48 kHz
<b>Internal Audio Data Paths</b>	<b>Conversion:</b> 20 bits <b>DSP:</b> 20 to 24 bits
<b>External Memory Card</b>	<b>Connector:</b> Accepts PCMCIA Type I cards, 68 pins <b>Standards:</b> Conforms to PCMCIA 2.0 / JEIDA 4.0 <b>Card Format:</b> Supports up to 1MB SRAM (attribute memory not required)
<b>Control Interface</b>	<b>MIDI:</b> 5-pin DIN connectors provided for MIDI IN, THRU, & OUT <b>Footswitch:</b> 1/4 inch T/R/S phone jack provided for 2 independent momentary footswitches System detects normal-open, or normal-closed on power up <b>Foot controller:</b> 1/4 inch T/R/S phone jack provided for footpedal (100 $\Omega$ minimum, 10k $\Omega$ maximum impedance)
<b>General</b>	<b>Dimensions:</b> 19.0"W x 1.75"H x 12.0"D (483 x 45 x 305 mm) 19 inch rack mount standard, 1U high <b>Weight:</b> Net: 6.4 lbs (2.9 kg) Shipping: 9.5 lbs (4.3 kg) <b>Power</b> <b>Requirements:</b> 100-240 VAC, 50-60 Hz, 35 W, 3-pin IEC power connector <b>RFI/ESD:</b> Conforms to FCC Class B, EN55022 Class B (CE), IEC 801-2, IEC 801-3 <b>Environment:</b> Operating temperature: 32° to 104°F (0° to 40°C) Storage temperature: -22° to 167°F (-30° to 70°C) Humidity: maximum 95% without condensation

Unless otherwise noted, all audio specifications assume rear-panel switch set to BAL, input level control is set for unity gain (0dB), and analog I/O connections wired for balanced configuration.

*Specifications subject to change without notice.*

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