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D E S I G N

Sound Library For The Korg DW/EX 8000

An Introductory Level Manual

May 1988

By James Fellows

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T A B L E   O F   C O N T E N T S

CREATING AND EDITING SOUNDS ON THE DW 8000

Introduction :

This manual is the distillation of my first five newsletters, the original programming manual which I wrote in 1985, and the programming notes that I shipped with the original versions of volumes 1-7. In this present form, this information is arranged in a more logical and complete way. This manual does not cover the advanced programming techniques and examples which will be found in the 10 page manual that comes with Volume 8. Nor does it include the data tables for my sounds, which are available for volumes 1,2,4,7, and 8.

The first part of this manual deals with the programming and editing skills you will need to use the DW 8000 (and our sounds) to advantage. The second part contains information relating to the DW 8000, such as listings of available software, ROM upgrades for the DW 8000, service information, and other tips. The third section lists the sounds in each volume and includes the notes for those sounds. So, even if you do not own those volumes, you can find out what sounds are included in them.

Learning to edit and program sounds:

If you are an advanced user, please be patient or skip ahead. This first paragraph is for those who are entirely new to synthesizers. First, the DW 8000 is a programmable synthesizer, which means that you and I can create a very large (almost infinite if you count minor variations in sound as different programs) number of sounds by using only the front panel controls to access the 53 parameters which are used to create sounds on the DW 8000 synthesizer. These sounds, once created can be stored in any of the 64 memory slots which are numbered 11 through 88. Since there are no keys on the DW 8000 for either the numeral 9 or Ø, you only have 64 different locations, even though a sound may have a number up to 88. This is because there is no such location as 19, 29, 5, etc. Each of these memory locations are capable of storing just one program at a time. All memory locations are identicle except for the programming information which you store in them. When you write a new program to a memory slot, you erase previous programming stored there. So, you will need to learn how to use the cassette interface to store sounds you have created on tape before loading new sounds into the memory locations where you have written your favorite sounds. There is nothing stopping you from creating sounds with your synthesizer just as I have done, you don't need a computer or any knowledge of electronics. What you do need is skill, patience, imagination and lots of time. To begin, you will probably limit yourself to editing existing sounds slightly in order to make them fit into the particular composition you are working on. There is no clear point at which your tinkering is no longer an edit and becomes a "new" sound. I usually consider a sound to be new when I have changed the settings for several parameters, and the result is a sound which is easily distinguishable from the original and has some new features the original did not. Otherwise, I consider it to be an edit of the original sound intended for a particular use. You will find your own guidelines depending on how you work, the way in which you organize and store your sounds, and your musical needs.

As mentioned in the last paragraph, the DW 8000 has 64 locations in which you can store sounds until you need them. However, there is only one active location in which you can play a sound. This location is called the "buffer". When you call up a sound by typing in its number, you are loading it into the buffer. When you write the sound to memory, you will transfer the information presently in the buffer to whatever location you specify. Your owner's manual covers this procedure adaquetly. When you turn the 8000 on, it loads program 11 into the buffer. This will remain unchanged until you load a new program into the buffer, or enter edit mode and begin to change some or all of the parameters. If you change every parameter value to be that listed in a data table for some other "program", you will have manually loaded that program into the buffer and can now write it to any location you choose. Since this is a very slow operation and relies upon your accuracy, it is not a good way to enter an existing program to the synthesizer. We will come back to this in the "tips" section of this manual when we discuss creative ways to use the tape (cassette) interface.

The parameter shown in the illuminated display:

When you call up a program you will notice that the center portion of the display shows the number of a parameter and the right (green) portion shows the current value of that parameter. This is a very nice feature of the DW 8000 whereby the parameter which was selected at the time the program was written to memory (from the buffer) is the one which is selected when the program is called back into the buffer. My programs frequently use this feature to point you to a particular parameter which I think you will want to experiment with in that particular sound, or which I think you may want to adjust via the edit slider while performing the sound.

The most common edit procedures:

Here are the edits that you will make most often, and which every player must be able to perform. We will next go into detailed explanation of these parameters, but first, to get you started, we will give you this "quick and dirty" approach to adjusting sounds:

Parameter 31 (all parameters are hereafter listed as p31,p32, etc.) controls the overall timbre (brightness) of a sound. You should experiment with this parameter on a few programs until you have a clear idea how it effects the sound.

Other parameters which are commonly adjusted are p76 to raise or lower the level of the DDL, p32 to introduce or remove resonance, p64/p65 to increase or decrease the amount of modulation, and p81/82/83 to adjust after-touch. You may also want to adjust p47 to change the way velocity effects the sound and p35 to restore the sound to its original timbre after adjusting velocity sensitivity. It is quite often necessary to raise or lower the sound by an octave and this is done via p11/21. They must be raised and lowered the same number of steps. If they are both at "8", you can raise them to 4 or lower them to 16 together. If they are at different settings, you must preserve that relationship: if you find them set to 16 and 8 respectively, you will raise them to 8 and 4. If you find them at 16 and 4, you cannot adjust them further without changing the nature of the sound. We will now go into an introductory level explanation of DW 8000 programming, having touched on some of these preliminary explanations.

A guide to the parameters of the DW 8000:

Although your DW 8000 is a completely "digital" instrument, it derives its programming scheme from earlier "analog" instruments. Any experience you have with any analog synthesizer is directly applicable to the DW 8000. Any of the many books and articless on analog instruments are equally useful.

The DW and EX 8000 are identicle instruments and the following information is applicable to either. You will find the number of the parameters being discussed listed in the left hand margin for easy reference.

The DW 8000 has 53 parameters with which to create and edit sounds. Each parameter gives you control over one aspect of the sound. An experienced programmer prefers to have as many parameters (controls) as possible. The novice may find these options confusing. I suggest a systematic investigation of each parameter, one at a time. Don't be timid about getting started: you can't do any damage, you will learn quickly and...it's fun.

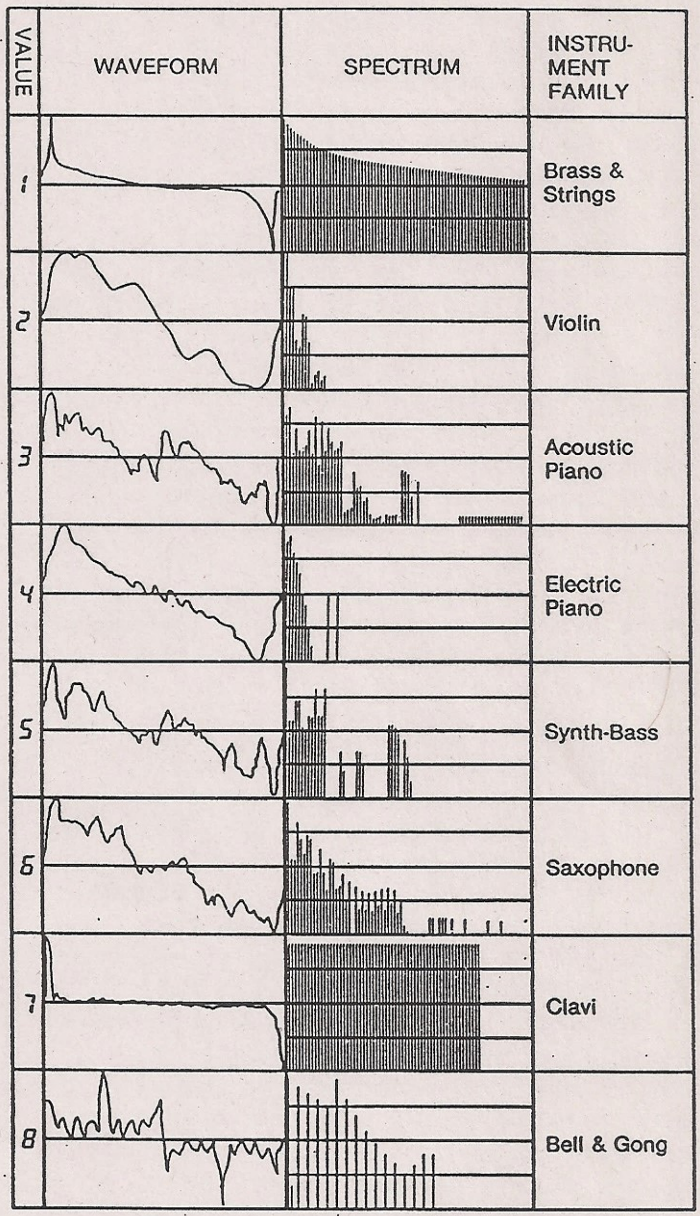
P31

Call up any program and enter parameter (edit) mode. Call up p31. Adjust it's level with the slider or edit +/- buttons. Try this on several programs. At the low end of p31's range you hear silence (or near silence) when you try to play a note, at the upper end (63) you hear a buzzy sound that is very similar regardless of which program you start with. You are opening and closing a filter, one called a VCF (voltage controlled filter) which cuts out all frequencies above the level you set it at.

In order to use this filter you need to understand a little bit about acoustic "theory." Every sound is considered to contain a spectrum of components called by various names such as "harmonics","overtones" or "partials." The volume of each individual harmonic, or its absence contributes to the sound we hear. These component parts change their individual volume (amplitude) during the course of the sound. Sometimes this happens extremely quickly. A lot happens in those few miliseconds, more than we can sometimes comprehend easily. Nevertheless, we hear a noise which we identify as a musical tone or sound under certain circumstances. It is

Waveforms

not necessary to analyze a sound to be able to hear it and this is worth some reflection. You do not need oscilliscopes, or a solid understanding of acoustics to create sounds. However, you do need to train your ears and your mind to hear and listen for changing harmonic spectra." This applies as much to creating sounds with FM instruments or samplers as it does the DW 8000. All of the difficult work has been done for you by the engineers who designed the DW 8000. They have given you 16 different patterns, which they call waveforms, containing examples of harmonic spectrums that will be particularly useful to you. The oscilliscope diagrams on the front panel are next to useless, even to an expert, in understanding the content of these waveforms. We will come back to these waveforms later, at the moment we will simply lament the absence of spectrum diagrams in the manual



and reproduce a crude chart of spectrum representations from the DW 6000 manual. Each verticle line represents the volume (by its height) of a particular harmonic component of the sound. You will note that some harmonics are missing entirely from some waveforms, such as the bell/gong represented at #8. Thus, missing harmonics can have just as much to do with a sound as the relative volume of those harmonics which are present. All of the waveforms represented here, as well as those included with the DW 8000, do not change over time. If you listened to them alone, they would be steady drones with slightly different qualities. You should also note that the waveforms represented here do not necessarily relate to those of the same name or number in the DW 8000.

In musical sounds, these harmonic components, whatever their amplitudes, have a definite relationship to each other known as frequency. You know that the middle A on a piano is usually tuned to a frequency of A=44Ø. When you actually play that A on the piano however, you create many other frequencies besides 44Ø cycles per second. We hope that your piano is not so bad that you can't even recognize the pitch of 44Ø...but beyond the 44Ø cycles per second which allows you to identify the pitch, there are hundreds of other pitched frequencies that contribute to the quality of the sound in various ways. Some of these are not audible to the human ear (a healthy ear can hear up to about 2Ø,ØØØ cycles per second). Others are heard but not noticed, such as the thud of the hammer hitting the string, and some very quickly passing harmonic material at the beginning of the tone. All of this material contributes to the sound you do hear and recognize. The component of the sound which is most important(usually) is what we call the fundamental. This is the same as the frequency of the pitch we are playing, such as A=44Ø. Most harmonics proceed in a regular mathematical progression from the fundamental. The first and most prominent of these are the first and second octave frequencies above the fundamental, and some harmonic intervals such as 13th, etc. The further up the overtone series you go, the more frequencies you come to that are not harmonically "correct".

So, to summerize: the 16 DW 8000 waveforms contain static (unchanging) examples of harmonic spectra containing different combinations and amounts of the various component frequencies of the "overtone series." While no "sub-harmonic" components are included (ie: frequencies below the fundamental pitch) they can be added by tuning the second oscillator to a lower octave and adjusting it's amplitude. We might also note that KORG has several times intimated that some of the waveforms are composed of multi-samples: ie, that certain waveforms are actually several waveforms assigned to different zones of the keyboard, so that the sound you get when you play C4 might be derived from a different wavetable than the sound you get when you play C2. If this is true, the purpose would have been to provide more useful waveforms, since real instruments have hundreds of dynamic waveforms which change with pitch, intensity of the note played, articulation, etc. I have never been able to make much use of this information, but I was able to verify that waveform #8 does seem to change at the F#3 to G#3 split when set to an octave setting of 16. You may wish to repeat this experiment to familiarize yourself with the waveforms: set p31 to 63, set p35,47,57,64,65,76,26,23,32,33,14,41 and 51 to Ø. Set p11 to 16. Set p12 to 1. Set p43,45,53,55 and 77 to 31. Now, using portamento, create a glide from the highest note to the lowest note of the keyboard. Use p12 to listen to the different waveforms. You are now listening to the pure waveforms in a continous glide from the highest to the lowest pitch. When you listen to #8 you should hear a definite change at G3. To make this experiment more useful you should listen to all of the waveforms and make notes about their harmonic content: bright, nasal, dull, rich, etc. You should select the terms that make the most descriptive sense to you.

Waveforms

"Key" Program

Using this same experiment, we will now return to p31, which controls the level at which the filter begins to remove harmonic components from the waveforms. Select waveform #1 with p12. Now select p31 and set it to about 35 using the slider. Play a note. Play the note again, and simultaneously push the slider to maximum quickly. A trombone, eh? Practice with this, using the slider to create shifting timbres as the note sustains. Introduce a rhythmic modulation. Try playing a note where you immediately push from 35 to 63 and back to Ø. Practice until you can do this as quickly as possible. You have just learned the single two most important lessons you will ever learn about the DW 8000. #1 : use that edit slider as a performance controller. #2:(even more important) it is the shifting level of p31 which creates useful musical qualities in a sound. When we get into programming, you will learn how to use p35, p41-47 and p65 to do complex tricks with the filter level automatically, allowing sounds too difficult to achieve with the edit slider and freeing our hands to do more on the keyboard. For the moment, you can save yourself months by spending an hour using the edit slider to create interesting sounds with all of the waveforms. You will find quickly that what works for one waveform may not work for the next. You will also discover that waveform #16 is a sinewave, and only contains the fundamental pitch, without any harmonics, so that raising and lowering p31 does nothing but change the amplitude (volume.) You might note at this point that FM synthesizers provide sine waves, which the programmer uses to distort each other into more complex waveforms such as other waveforms in the DW 8000. Finally, you should note that one reason the DW 8000 is so popular with some pretty heavy-duty players is that it allows this edit slider access to all parameters for some unique possibilities in performance.

set p77 to 0

edit slider

P31

Using the parameter set-up given in the top paragraph of the last page we will begin to explore parameters 32-47. Make sure p77 is set to Ø if you do not have a control pedal plugged into the portamento jack. Pick any waveform at p12 and a setting of about 35 for p31. Now raise and lower p32. Try this with several waveforms and then with several values
P32 for p31. You are adjusting resonance, which emphasizes the volume of the harmonics closest to the point where the filter cut-off level is set to remove them. So, you get a peak in harmonic content just at the top of the overtone series. When set near maximum, this upper harmonic will be so strong that it's pitch becomes more important than the fundamental, and if you adjust p31 in such a situation, you will hear this pitch rise and fall with the filter cut-off level. This is lesson #3.

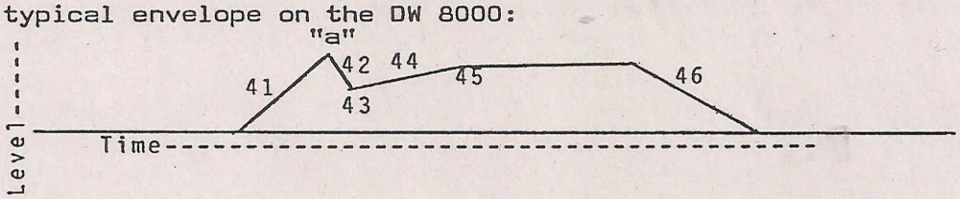
Now, let's listen to some of this harmonic content I've been talking about. Set p32 to 31, set p12 to 1. Now hold middle C (C3) and begin to raise p31 starting at Ø. When you get to about 1Ø you begin to hear some of the subharmonic content that I said wasn't included (I lied). Around 31 to 32, depending on the temperature, you will begin to hear the fundamental pitch. Then you begin to hear the individual overtones until you get up to about 62, at which point they become too high to be audible, and at which point (what a coincidence!) the engineers decided to cut you off. As you go up the overtone series you will find that they become softer and softer, since our ears are not able to hear these tones as well. You will begin to hear more and more of the lower harmonic content, and the effects of the resonance will become less obvious as the harmonic which is resonating becomes too high to hear. This particular experiment is not really that useful in teaching you how to program, but it will give you a clear idea of waveform content. Try the experiment on other waveforms.

Parameter 33 changes the way the setting of p31 effects the keyboard in different areas. At its normal setting (p33=Ø) the same filter cut-off level is active across the entire keyboard. As you play up the keyboard
P33 you will find that the sound becomes less brilliant, especially if p31 is set to a very low value. This is because you are getting closer and closer to the cut-off frequency (or past it) and so your notes are getting less and less harmonic material past the filter. Setting the value of p33 to a higher level causes it to track the harmonic content of the notes as you go up the keyboard, so that they will all have the same number of harmonics. None of this really works quite as simply as it is stated here, there are factors which the engineers took into account when they designed the waveforms, the filter and the other parameters so that the DW 8000 would work as well as it does and be as simple to operate. You will begin to appreciate this if you explore the Ensoniq ESQ-1, which has great features, but which doesn't compare to the DW 8000's sound quality and noise free operation. In order to get the DW 8000 to work as well as it does, they did a lot of adjusting of "theory."

We are going to ignore p34 since it has not proven terribly useful to me in most programming, and because it would be better discussed later. Before we get to p35 you will need to understand that p41-46 allow you
P35 to automatically shape the harmonic content of a note without having to adjust the edit slider and p31 while playing. The shape that you create and assign to the note via p41-46 is called it's "filter envelope." P35 simply controls the intensity with which the filter envelope effects the filter. At low settings, the envelope barely changes the filter

cut-off frequency at all. At maximum level (p35=31) the filter is directly controlled by the envelope's shape, just as if you had your finger on the edit slider. The key to succesful programming lies in the skilled use of the filter envelope, it's intensity and the way this is controlled by the velocity of the key strike (via p47.)

The filter envelope (and the amplitude envelope) are comprised of "six" stages, according to the manual. Actually, there are five segments
P41-46 to the envelope which you program using six parameters. Here is a diagram of a typical envelope on the DW 8000:



All DW 8000 envelopes begin at Ø. Your first programmable option is how fast they rise to their maximum level. This time is controlled by p41. You will always end up at the maximum amount of the envelope at the end of this segment. This point corresponds to the filter setting which you have determined at p31 as modified by the effects of p35,p47 and the velocity with which you hit the key. If this is confusing, don't worry, just read on. If you want to start at the maximum-open part of the envelope, you set p41 to Ø causing the synthesizer to open the filter to its maximum setting so fast that your ear thinks it was open right at the beginning of the note.

Parameters 43 and 45 control levels. In our example the filter opens from Ø to the point indicated at "a" at a speed controlled by p41. The filter then closes to the level controlled by p43 at the speed determined by the value of p42. P44 then controls the speed at which the filter opens to the amount specified by p45 (which can be greater or less than p43, but not greater than "a" or less than Ø.) At this point the note and the envelope will remain unchanging as long as you hold down the key. Once you let it go, the filter will return to Ø at the speed which you program with p46. Surprisingly, these envelopes are about as sophisticated as you will find on most synthesizers, which is another reason the DW 8000 is so popular. Even more surprising: the instruments which have more powerful envlopes are the even less expensive Casio CZ syntheizers. Already newer synthesizers are beginning to come out which better reflect the importance of the envelope generator. How about 4Ø stage envelope generators? The more envelope generators and the more segments per generator the better!

Parameters 51-56 control identicle speeds and level parameters on a second envelope which works independantly to control the overall volume of the
P51-56 note. Of course, the filter envelope may often have a very "rëal" effect on volume as well, since the harmonic content of a sound has a lot to do with its perceived amplitude.

P47 works in conjunction with p35 which controls the intensity of the
P47 effect the envelope has on the filter's level. If p47 is set to 0, velocity has no effect on the envelope or the filter level. As you raise the value of p47, you will find that you must hit the key harder in order to get the same sound. So, raising p47 gives you access to lower settings of p35 than the one you have programmed. You will only achieve the full value of the p35 setting at maximum velocity (Midi value 127.)

P47 This is important because it gives you a whole range of timbres depending on how much velocity you use to strike the key. Good programming usually tries to give the player a sense of control over a fluid and expressive sound, and the key to this is control over the filter and filter envelopes via velocity. The KORG DSS-1 synthesizer is a more sophisticated version of the DW 8000 (and also a sampler) which allows control of numerous other parameters via velocity such as the speed of envelope segments, autobend, etc.

Although it might seem that we should try to get maximum expressive capability by trying to use the DW's options to the fullest, this is not always the case. It is important not to "overprogram." There are many situations where the ideal sound is quite simple. The individual player should concentrate on creating a range of sounds which work well with their playing style. Some of these sounds might be quite plain, and yet they may be the strongest part of that player's repetoire. Plain sounds are not necessarily easier to program, either. Getting a sound that fits perfectly into a song is never as straight forward a task as it might seem. An easy way to convince yourself of this is to pick out a particularly useful, but simple,synthesizer sound from a record and try to duplicate it exactly. This is a particularly good exercise for all programers, no matter how advanced, to do regularly. It will keep you from stagnating and will force you to continually expand the range of sounds you can program and play.

Parameters 51-57 do not normally play as large a part in the sound shaping process as their identicle counterparts which act on the filter
P51-56 (parameters 41-47). They will be more important in those sounds where the filter envelope is not as active (lower settings of p35.) P51 is probably the most often used of these par.s, and has a special role to play. The overall purpose of these amplitude envelope segments is to control the volume "shape" of the sound. If that shape is quite simple, such as in this example, the percieved shape of the sound will be determined by what the filter envelope is doing.

A "plain" amplitude (VCA) envelope: p51=Ø p52=Ø p53,55=31 p56=x, where x=p46

P51, which controls the attack portion of the sound, can have a very useful role which is best understood by experimenting. Choose a sound which has p51 set to Ø. Use the edit slider to raise the value of p51 while you play. Try this on at least a dozen different types of sounds. You will find that p 51 can do a number of things. In some string-like sounds, where p41 is set to Ø, p51 is the only parameter which controls the speed of the attack. In other sounds, where the filter envelope has a more complex function, p51 actively interacts with p41, creating all types of interesting and complex attack transients. Slight changes in the value of p51 and p41, when they are both in the range of Ø to 1Ø, can have significant effects on the resulting sound. As one grows longer than the other, even slightly, a string sound becomes a horn sound, or a piano sound becomes a trombone. There is a lot to be learned by fine tuning the relationship between these two attack speed parameters.

Let's summerize the three typical relationships between p41 and p51:

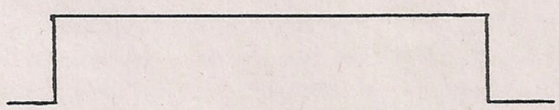
P51-56

1) p51 is not active, is set to Ø, and the percieved attack speed is controlled by the setting of p41.
2) Both p41 and p51 are active, and the relationship between them provides a more complex attack than would be possible with either one acting alone.
3) p41 is not active, is set to Ø, and the percieved attack speed of the sound is controlled exclusively by the value of p51.

In actual situations, these distinctions are not so clear. Many sounds fall somewhere between our three catagories. For instance, in a sound of the "third" type, the value of p41 might be increased until it just begins to have an audible effect on the sound.

Envelopes

There is much more to be said about envelopes. An entire book could be written about the possible relationships between these parameters. The 1Ø page manual which comes with Volume 8 focuses on some of the more advanced techniques with which you can trick the DW into producing very sophisticated sounds. My advice for learning to work with envelopes is as follows. First : ignore most of what you read about envelopes in magazine article and books. For reasons I don't quite understand, most people who write about programming have a hopelessly naive understanding of envelopes. Be particularly suspicious about such terms as "organ" envelopes, "brass" envelopes, etc., especially when they are given as simple formulas with which to produce imitative sounds. The case of the organ envelope is a good one to explore. These writers have confused the operation of the organ keyboard with the harmonic content of the organ sound, and naively assume that the sound is a simple on/off thing. Their formula for the "organ" sound is usually exactly as follows:



I invite you to examine the envelopes of any of the organ sounds in my sound library volumes. You will not find anything nearly as simple as this. Organ sounds are <u>extremely</u> complex. They usually contain attack transients every bit as complicated as a piano, a flute, an oboe, or any other sound. So, beware of simplistic formulas. If you often wonder why factory presets are so horrible, it's because they are created by people who are enthralled with these very limited and formula-bound concepts of sounds. My advice is : open your ears, listen to the world around you. Create a psychological microscope for examining audio phenomenon in detail. Imagine the miliseconds at the beginning of a note as a four hour symphony.

My second piece of advice is to carefully study parameter charts of sounds which you like (or don't like!) and isolate these envelopes for experimentation. My third piece of advice is simply to roll up your sleeves and get to work. There are no short cuts to good envelopes. Every time you create a sound, or edit an existing sound you will have the opportunity to explore the possibilities of the two complex envelopes and their interaction. Or, you can follow a magazine formula and create dumb sounds.

We will come back to the two envelopes and their relationship to key velocity, filter level and to the sensitivity of the filter to the filter envelope at a later point. Now we will turn to an aspect of programming that is much simpler to grasp.

Let's recall our test program from our experiment with the edit slider earlier in this text: set p11=16 p12=1 p13,43,45,53 & 55=31, p14,23,26, 32,33,35,41,42,44,47,51,52,54,57,64,65,76,77,81,82 & 83=Ø p34=1 p31=26. Now set the following: p71=2, p72=Ø, p73=13, p74=12, p75=26.

P71-76 (DDL)

Play and sustain a chord, either with your hand or a sustain pedal. Gradually raise and lower the edit slider for p76. You are probably going to hear some clicking noises as you adjust the slider, just ignore them, they are a normal result of the changes in level you are requesting of the digital processing. The particular effect you are introducing is called "chorusing" and is created by the Digital Delay (DDL) which is included with the DW 8000. You probably already know how to use a digital delay to create other types of effects. You will find quite a few unorthodox applications for the DDL in the text for volume 8. For the moment we will consider a few examples of the type effect the DDL can contribute to our "bland" test program. Try the following setting: p71=Ø p72=Ø p73=15 p74=7 p75=31 p76=15 Then adjust the value of p74 from Ø to 31. This is controlling the speed of "modulation." Modulation speed is just one of the four ways you can adjust the DDL. Modulation intensity is the second. Set p74=31 and adjust p75 from Ø to 31. You are now adjusting the intensity of the modulation effect. Now set p75=31 p74=12 p71/72=Ø now adjust p73 from Ø to 15. You will hear a very slight thickening of the sound as you increase the level, until it begins to aquire a "swirling" quality at level 14. At level 15 you hear true feedback, although in a mild form, due to the other DDL settings. Feedback is the third aspect of the DDL. It is very sensitive to the settings of the other DDL parameters. It does different things in each context. It is not something you will probably use a lot unless you have a big place in your heart for this effect. The important thing to remember about p73 is that the true feedback only comes in when set to a level of 15. From Ø-14 you get a relatively mild effect which is probably more useful in the majority of cases.

The last of the DDL parameters have to do with time. P71 gives you coarse adjustment and p72 gives you fine adjustment. For most DW 8000 sounds the effects of fine tuning (p72) are not terribly significant. The value of p72 is for adjusting the speed of echo and slapback effects to match the pulse of the music. Since there is no sync connection allowing you to hook up an external time controller you will be limited to matching the pulse of a sequencer or drum machine by the adjustments to p72. P71 controls coarse time shifting selection. You will find it selects between the major types of DDL effects, such as flange, chorus, slapback, short echo and long echo. In echo mode, feedback controls the number of echoes you hear before the sound dies away. Here is a quick guide to some effects:

Effect Name	Value of P71
Distortion	Ø-1
Flange	1-2
Chorus	1-3
Double	2-4
Slapback	3-5
Echo	5-7

This is a rough guide. The actual effect will depend on the sound, and the other DDL settings. You can copy favorite settings from parameter charts, but each sound will respond according to its timbre and envelopes.

We will now briefly explore the modulation parameters in the LFO section, which are entirely independant of those in the DDL section. P64 controls the intensity of pitch modulation in exactly the same way that p75 did. P62 controls modulation speed in exactly the same way as p74 did. Since these two modulation sources are independant they can be used in combination to produce a variety of complex effects. Before

P61-65 (LFO) we get to that let's take a quick look at the other LFO parameters. P65 controls the intensity of LFO modulation to the filter. If you remember our first experiment with edit slider control over the filter, you will remember that we were able to introduce filter modulation by wiggling the slider rhythmically. P65 allows you to determine how much the modulation in the LFO section can be routed to control this wiggling of the filter level. P62 allows you to introduce a delay from the time you strike a note until the time the automatic modulation begins. The other LFO parameter is 61, which selects different types of modulation. The setting Ø gives the type used in 99% of all sounds.

Note that LFO effects all notes being played on the DW 8000 simultaneously. That is, there is only one modulation speed and delay within the LFO section and it effects all notes being played in the same way. As long as you are holding a note which has begun to modulate, new notes will begin to modulate as soon as they begin, even though there might be a very long delay setting in the program. If you want more control over modulation than this, you can trigger modulation via aftertouch and be able to specify exactly when, where, and how much you want while you are playing. Set p64 and p65 ("automatic pilot") off. Then use p81-83 to adjust after-touch control of modulation. P81 controls the

P81-83 amount of modulation to both the pitch and filter of the notes being played (combining the function of p64 and p65.) P82 allows you to control the filter independantly by pressure. Remember, if you set p64,65 and 81 to Ø you can now control the filter by rhythmic pressure on the keys and it will have the same effect as wiggling the edit slider in our first experiment. This is probably the most musically expressive way to use modulation, since it exactly duplicates the type of control you would have on a violin or similar "real" instrument. You will not be bound to a predetermined speed and amount of modulation which is stiff and unmusical. Use p83, which controls volume, with p82 to get the right intensity of effect. The actual settings will depend on the filter and filter EG bias (p35) for that sound, as well as a number of related key velocity and envelope settings.

Although the automatic pilot type of LFO controlled by p64 and p65 can be rigid, it can also be quite useful in real musical situations. Subtle amounts, at slow speeds can add a shimmering quality of movement similar to that added by chorusing. Better, you can adjust the speed and

LFO+DDL intensity to play with the speed and intensity of the DDL modulation. This gives you richer sounding modulation. It is particularly useful for imitating rotating speaker systems which create the "Leslie" effect that is such an important part of some organ sounds.

Let's go back and take a closer look at the ways in which we can generate complex harmonic waveforms. Earlier we looked at the sixteen waveforms when used by themselves. Combining waveforms gives us 172

P11,12,21,22 different timbral combinations if we limit ourselves to merely using them all at the 16' octave settings. However, we can triple this figure by assigning different octave settings to each oscillator, such as: p11=16 & p21=8 or p11=16 and p21=4. We won't bother with settings of 8 or 4 for both oscillators, since this merely shifts the range of the keyboard, without introducing new harmonic relationships. Combining waveforms at different octave settings can create many new and interesting timbres. However, you will frequently find that the resulting harmonic material is more characteristic of the particular octave relationship than of the component waveforms. That is: the shifting of waveform relationships by octaves is so important an effect, that it tends to overide the individual content of each waveform. To make this scewing less obvious, you can bring down the volume of one of the two waveforms so that is merely adds a color to the other waveform rather than setting up an obvious octave relationship. There are several types of sounds which are easily created by waveforms tuned to octave (or other) relationships. Organs are the most noticeable instruments which use this type of harmonic material. However, you can invent a whole family of instruments, not heard before, which use this approach. You can also create bell or tuned-percussion harmonics by having one waveform tuned a fifth (or fourth) + an octave or two higher than the other. You do this by using the combined power of p21 and p24. You will probably find that you can get much better bell waveforms that way than by using the rather one-dimensional waveform #15. Using one of either waveform #4 or #5 in such a combination is a good place to start. remember to adjust the relative volumes of the components for better results.

It is also well to keep in mind the importance of relative volumes of the two waveforms when creating normal sounds which have both oscillators tuned to the same octave. Detuning should be used sparingly. Many people seem to use detuning as the only way to adjust a sound. It <u>can</u> be
P13,23 very important, but it is so overused (and other parameters so under utilized) that I tend to be very suspicious when I see a parameter chart
P25 with a setting of 4, 5 or 6 for p25. You can do more with chorusing and other parameters. Besides, "out-of-tune" is not a very good ideal to strive for. You will need to use higher settings of the detune function in sounds where oscillator 2 or 1 is set to very low volume. By the way, it is not wise to have oscillator 2 turned up to full volume and Oscillator 1 turned down, since all detune and pitch shift parameters effect Oscillator 2. Better to have Osc. 1 be your main sound source and use Osc. 2 for subtle shadings and tunings.

There are many functions for the Auto Bend parameters, one of which is subtle, time controlled detunings. The manual for volume 8 goes into many of the more tricky ways to use Auto Bend. If you don't have that manual to refer to, remember that you can do more with Auto Bend than
P14-17 emulate a joystick automatically. In fact, that's the one thing you
(Auto Bend) won't want to do with Auto Bend. Auto Bend allows you to shift the pitch of either oscillator independantly of the other. By allowing you to assign a time and amount to this shift, as well as a direction, you can create many very flexible and much more subtle varieties of detuning. A slightly detuned oscillator which gradually resolves into perfect intonation can make subtle harmonic transients coming from a second oscillator create very "realistic" instrumental qualities.

Other uses of Auto Bend are discussed in Volume 8. You may also look through the parameter charts of individual sounds to spot ways to use this effect.

Developing programming strategies:

By now it should be clear that it is the relationship between all 53 parameters that contributes to the quality of a sound. I rarely start "from scratch" when creating a sound. Instead, I pick the sound I'm most interested in at the moment, or one which I know has qualities which will be useful for the type of sound I want to create. It is much easier for me to program the DW now than it was 3 years ago, because I have developed a large library of sounds to start from, and have a well stocked "bag of tricks." I also find it useful to review all similar sounds before starting on a new programming session. This saves me from wasting time duplicating past efforts. This is the best, and only legitimate use for data-tables, as far as I am concerned.

Having data-tables handy, when programming, allows you to quickly recover successful LFO,DDL, and other parameter settings from past sounds. It is also important to learn how to quickly cut-out distracting elements when trying to focus on a sound. If you find a quality you like in a sound,and want to determine where it is coming from , you can switch certain groups of parameters on and off and thereby narrow down the search until you find the one parameter which controls your particular effect. This requires you to understand exactly how each parameter works and how they are related. However, this chart may provide some helpful shortcuts:

P14 turns off all Auto Bend functions when set to Ø.
P76 turns off all DDL functions when set to Ø.
P23 turns off all functions of p21-25 when set to Ø.
P32 turns off resonance when set to Ø.
P35 can reduce or eliminate the effect of p41-45, but you may need to compensate by increasing p31.
P32 can accent the shape of the filter envelope when raised to maximum, causing the envelope to sing it's "shape."
P62 does <u>not</u> turn of LFO.
P64 and P65 turn off LFO.
P74 does <u>not</u> turn off DDL modulation.
P75 turn off DDL modulation.
P47 and 57 turn off Key Velocity.
P81-83 turn off after pressure when all are set to Ø.
P13 turns off waveform #1.
P23 turns off waveform #2.
P25 turns off detuning when set to Ø.
P77 turns off portamento. Inserting a plug in the portamento jack does the same.
P73 turns off DDL feedback.
P26 turns off noise.
P31 adjusts the filter.

Harmonic Movement

Remember that the waveforms and their relative volumes and tunings provide only part of the story about harmonic content. The attack portion of the sound, which contributes the most to the concept of the sound that our mind forms when listening to a performance, can be much more important than the actual waveforms used. The filter envelope, the setting of the EG intensity (p35) , and Auto bend are all factors which can contribute to the harmonic content and movement within the attack portion of the sound. This concept of movement is important. The DW 8000 waveforms are static. Our ears tend to identify sounds by the way their harmonic content changes during the initial few miliseconds of each note,rather than by freezing one particular waveform and analyzing it. So, to this extent, the filter envelope will contribute more to the sound than the waveform.

P35 vs. P31

P35 and p31 have a peculiar relationship. Although their functions are quite different, the effects of both parameters may be identicle or appear to be quite similar to our ears. So, there are sometimes two ways to get to a certain sound. In one case you can lower the value of p35 and raise the value of p31. In the other case you can do exactly the opposite and still come to the same sound. It becomes important, therefore to decide where you are headed, because the type of sound which you are after will require you, eventually, to choose a particular setting for p35. Since much of the work you will do elsewhere in the sound will depend on this setting, you will want to be correct from the start in your choice of values for p35. The way to decide is to think about how important the envelope shape will be to the type of sound you are going to create. How dramatic is it's attack portion? Is there a lot of movement or just a little? The more complex your sound,the higher the contrast between dynamic levels within the sound, the more percussive the attack, or the greater the range of filter velocity, then the higher you will need to set p35. Look through the parameter tables and you will develop an idea of what values go with certain sounds. Once you select the value of p35 you develop a filter envelope and setting for p31 and p47 to get the sound you want. All of this will change if you then decide to change the value of p35. In practice, it is rarely possible to get the value of p35 right on the first shot. It is common to have to revise all of the related parameters several times in the process of creating a sound.

P41-47

Even with all of this advice in mind and the best working habits, you will find that patience and time will provide your best results. Every time I think I have found the definitive version of a sound I am wrong. A month or a year later I stumble upon a better way to do the same thing, or a way to do something even better. Your standards for judging sounds will increase along with your expectations as you become more intimitely involved with acoustic phenomenon. Your playing skill and breadth should also develop as a result of the increasing power of your hearing and imagination.

TAPE INTERFACE INSTRUCTIONS

·Loading sounds from our tapes.
·"Partial-loading" techniques for combining sounds from several tapes.

Loading sounds (data) to the DW 8000 from tapes:

If you are familiar with the tape interface please skip to the next section. If you are not,please study the manual and the following instructions until you are. Tape instructions are on pages 43-5Ø of the Korg manual. We can supply a reprint ($1.00) if you don't have these instructions. Our reprint is taken from the DW 6000 manual, which is more clearly written. The procedures are identicle.

A word about the tape interface: Contrary to some people's impression, this type of data interface is extremely reliable and unquestionably the most suitable for this type of application. You should select an inexpensive recorder (the $19.95 variety is fine) and leave it permanently connected to both the to/from tape connectors on the DW. Tape down the level controls once they are correctly adjusted. If this doesn't turn you on, you can pretend it's some kind of exotic disk drive. This interface will take much more abuse than a disk drive and still operate correctly. It runs on batteries. It takes exactly 8 seconds to transfer data to and from tape. It works, nearly 100% of the time. It probably won't malfunction when you most need it. Cassettes are cheap. We can supply 20 five-minute cassettes for $12.50. You can buy packs of 100 from our supplier for about $35.00. Any questions? Read the manual or check below.

Tape interface problems are due to user error most of the time. Turn off noise reduction, bass boost, stereo "enhancement" MPX, or any other controls that would alter the sound of the data being "played". Listen to the tape as audio material first to become familiar with it's format. Pilot tone...lower tone...data....pilot tone. That't it. When making tapes the pilot tone should be set to = Ø VU. I record on both left and right channels, so you have two sources for each data dump, should one be damaged. The second dump is provided in case any imperfection in the tape makes loading from the first dump impossible.

Tape Loading should not cause any problems if you follow the directions in the manual. But, if you do have problems check these items : If the load signal will not show in the display enable the "write" switch on the back. If the load message appears but does not change when the data tape is played you are not getting the signal into the DW (or it is so low it does not register.) Check cables, connections, etc. Make sure high/low switch is set to high if you are using tape headphone jack, or set to low if you are using line out (RCA connectors). Make sure you are using a mono mini plug to go into the DW (one plastic ring on tip, not two.) Make sure you have the proper adaptors and connection at the tape unit. If the headphone jack is stereo you may need to get a "Y" splitter to get a mono signal from either the left or right channel of the tape. Don't mix these signals; they are identicle and may cause phase cancellations. If the load signal begins to count off the data being received but switches to "err" before getting to "8" then the signal is arriving and being read but is not entirely intelligible. The error signal will appear if even the tiniest error is detected. Adjust volume carefully, since signals which are either too high or too low will both give the same error signal. There is a narrow range within which the signal can be correctly read. If you get erratic results (some good, some error) then try to hone in more closely on the center of the "good" zone. If you still have problems make sure heads are clean and de-magnatized. Make sure tape transport is even and steady (music will "wobble" when played on unsteady tape player.)

If all else fails (including using alternate channels or second dump) and you cannot get any tape to load into the DW, you may need to get a different tape recorder. I have yet to hear of a DW with a defective tape interface, so you can nearly rule that out.

"Partial-loading" techniques for combining sounds from several tapes:

It is possible to load less than 64 sounds from a data dump stored on tape. The DW 8000 loads sounds from tape one at a time. Every time it begins to load a new set of eight sounds it sends a new number to the display. It sends "1" when it begins to load sound 11, "2" when it starts to load sound 21, etc. You may press the "cancel" button at any time. Although the "error" message will appear, you will have loaded all sounds up to the point where you stopped the transfer. The remaining (higher numbered) sounds will be the ones that were already loaded in memory. By using your musical skills you can count eighth notes against the "whole" notes of the "1...2...3...4...5...6...7...8" message. With even a little practice you should be able to stop the data transfer quite easily at 42,71 or whatever. I usually leave at least one or two programs as a buffer zone in case my timing is less than perfect. Now, re-arrange the combined set, moving all of the sounds that you intend to keep towards the higher numbered locations.
Be sure to remove any sounds you want to keep from the lower numbered slots, because you will use these spaces to transfer-in sounds from other cassettes. By loading, re-arranging, and re-loading, you can obtain any combination of sounds into the synthesizer memory. Then re-arrange them to your own design and make a tape copy of the arrangement so that next time you can re-load your combination easily. You will need to make prepared cassetes for this operation which have all the sounds which you wish to incorporate from a particular set arranged in the lower numbered slots. You will need to make one of these temporary dumps for each group of sounds you are going to extract one or more sounds from for inclusion into your master set.

We can make arrangements of your sounds,(or ours)for you and send them to you on cassette. Call for details.

SUPPLIES: Radio Shak stores are the best source of cables, adaptors, and other small audio accessories at low cost. Connectors are of the following types : Phone (not telephone!) jacks come in quarter-inch and mini (eighth-inch) varieties in both stereo and mono. The quarter-inch is the standard for your audio output to mixer or amp. Mini is standard for interface on Korg synths and common for headhone/mic inputs on portable cassette players. Decks and professional models usually use quarter-inch connectors. Some computers as well as Yamaha synths sometimes have a special cassette interface connector which looks like a midi jack/plug. The other type of audio connectors are the pin-type phono plug/jacks known sometimes as RCA connectors. These are the standard for home-audio line-in/out connections. Cannon,low-impedance or XLR connectors are three pronged plug/jacks for microphone and other audio lines using low impedance signals.

MISCELLANEOUS

ROM UPGRADE : The most current ROM upgrade for the DW 8000 is number 12. This is the first which solves the MIDI glitch (stuck notes, and uncontrolled burst of note data sent from DW MIDI out port to slave modules and sequencers.) It also makes EX 8000's compatible with DW 8000 computer software. The first of these improvements has been tested at Angel City and proven to be 100% effective. I have not tested the EX 8000 compatibility claim, but assume it works, since I have not heard otherwise from clients who use it for running DW software. Diagnostic Routine: Turn power on while holding down 1 and 2 buttons. (This will not work for the EX 8000) The green portion of the display should show the number 12. If it does not, you do not have this upgrade installed. Earlier ROM's start at Ø6. ROM 12 is part # AC-4827128A and bears the paper label marked 85Ø712. It is available free (or inexpensively) from Authorized Korg Service centers. It can be installed in less than an hour (plug-in EPROM chip,no soldering.) ROM 12 was introduced in the spring of 1987. If you intend to install the part yourself, please check newsletter #5.

KEY CONTACT PROBLEMS in early DW 8000's (serial numbers ØØØØ2ØØ and below) are a well documented problem. Korg replaced the key contact assemlies of these units while under warrenty. I have also heard of cases where the work was done free even after the warrenty expired. The problem involves notes which don't sound as loudly as their neighbors when hit with equal velocity, don't sound at all, or perform erratically. Normal key contact problems result from dirt,dust,etc. interfering and can be solved by simple cleaning (or just by blowing forcefully under the key assembly.)

EX/DW 8000 are identicle except for the following features: The EX does not have either a keyboard or the arpeggiator found on the DW. The EX has two additional features : a "window" allows the user to set a upper and lower midi note within-which the module will respond to data on it's assigned MIDI channel. It also has an extra parameter (78) which allows midi transfer of program data: send one program from buffer, send entire bank (64 sounds),receive one program (to buffer), receive 64 sounds. The same operations are available on the DW 8000 but cannot be activated from the front panel. They must be initiated from a computer using the correct commands.

MEX 8000 /SEQUENCER is Korg's memory storage unit for 4 banks of DW/EX 8000 programs and is connected to the MIDI in/out ports on the synthesizer. It can also be used as a sequencer with very limited abilities by setting its rear panel switches as follows : all "protect" and "Midi channel" switches in up position. First "device" switch in down position. Last three "device" switches in up position. Buttons are assigned as follows: save="write", cancel="play" load="stop". When used as a sequencer, banks A,B and C are overwritten. D retains program data. Records about 1ØØØ notes. Records up to 16 notes at one time. Records velocity. No overdubs or edits. If you play while in mode "3" the sequence loops indefinately.

KEY TRANSPOSITION ON THE DW 8000: This can be achieved with a set-up that calls for an extremely simple "home made" modification. Use p66 to determine how many semi-tones up or down you want to transpose your performance. You have the capability of transposing up to an octave in either direction. Then, you must find a way to keep the joystick in the full left (transpose down) or right (transpose up) position. The easiest way to do this is with a piece of tape about 4 inches long. The only type which works for me is the synthetic "cloth" tape used in medicine for attaching bandages, etc. to skin. The brand I use is called Dermicel and is made by Johnson and Johnson. It is sticky enough to hold the joystick in place, won't slip, and doesn't leave a mess behind when you remove it. It is a light, synthetic material that comes in 1" rolls and is available at almost any drug store.

A permanent and more attractive solution can be arranged with only slightly more ingenuity. Glue a piece of material,(preferably something which looks nice attached to the DW,) on either side of the joystick. These attachments should be such that they have the proper strength, shape and position to be able to hold a third part between themselves and the joystick such that the joystick is firmly wedged into the full left or right position. If you choose the right shape and position, this arrangement will work in either position and will be extremely easy to use and as reliable as the glue you use. I suggest 5 minute epoxy (clear) as the choice for the glue. You can use anything from a pencil to a chopstick for the third part that wedges the joystick over to the side. I have seen several successful choices of materials for the two blocks. These range from plastic parts used to hange "decorator" venetian blinds to little scraps of ebony left over from fingerboard replacements in guitar repair shops. I recommend plastic, or some permanent material that won't wear out, but wood is ok. Try to find a way to make this attachment improve, rather than detract from, the appearence of the synthesizer. There are undoubtedly dozens of ways to mechanically hold the joystick in the full left/right position, so think about it a bit before you plunge ahead...and send me your solution if you stumble upon something brilliant. I'm still using tape, because it works.

You can assign some of the normal joystick functions to after-touch, and this may be necessary since your joystick will be disabled while engaged in transposition. I saw a clever solution using rubber-bands (the heavy kind) which held the joystick either left or right but still allowed it to be used for modulation and pitch bend in the opposite direction to which it was pulled by the rubber band. That stretchy feel is pretty cool in performance, especially when you let go and it snaps back to transposed pitch.

USER GROUP : It has been nearly a year since I have received contributions for the user sound exchange. So, it does not seem likely that there will be any further user volumes. If you have any information that should be shared with other users, or have sounds to contribute, please call.

ALTERNATE WAVEFORMS : additional waveforms could be added to the DW 8000 through a fairly simple hardware / software upgrade. This would significantly expand the DW's power. I would very much like to see a poly-timbral version of the DW with new waveforms and am willing to help any company who wishes to get involved. If you would like to see such a modification, write to Grey Matter Response, 15916 Haven Ave. Tinley Park, IL, 60477 and tell them that you are interested in such an upgrade. They won't "respond" unless they get lots of requests, so get all of your interested friends to write too.

VOLUME 5 is no longer available. It was a compilation of the sounds in one of the earlier versions of volumes 1,2 and 4, arranged by catagory. Since these sounds have been updated several times, and since there are now two newer volumes, this "index" volume has been discontinued. It will be replaced, in the future, with a different approach to the same task.

VOLUME 9 may or may not be available in late 1988 or early 1999 depending on interest and my ability to find new sounds for the DW 8000. It will probably contain more sounds characteristic of the DW 8000, especially lead, and accent sounds. Suggestions? (please don't ask for strings, brass, the Mormon Tabernacle choir or more pianos!)

SOFTWARE FOR THE DW/EX 8000 : If you have the EX you will probably need ROM 12 to run most of this software.
IBM : Synthetix released a very complete and professional editing/librarian program for the DW/EX in late 1987. It's many features are too numerous to describe here. Some examples: on screen numeric editing of all parameters. Supports Mouse (MicroSoft or compatible) operation at all levels. Graphic editing of envelopes with mouse. Overlay mode compares VCF and VCA envelopes. Adjacent numeric listing of envelope values updates continuosly as you re-shape envelope graphs using mouse. Librarian features: 16 character names for each patch as well as 80 character text entry for notes. You can search all files for text entries such as "strings" or other words and pull all sounds with those entries. This is great for re-locating all sounds used in a certain album, mix, or gig. The possibilities are endless. Patch generation is equally sophisticated. You can design sets of parameters to be randomized, name those sets, and save them to disk. Banks of sounds can be generated using these set-ups. This is the method of choice for professional results. You can have two banks of 64 patches on screen and move patches between banks with the mouse. This is the ideal way to re-arrange sounds. Unlimited storage of patches (of course) on disks.Prints patches and patch parameters in a variety of formats including 1, 8 or a complete bank of patches and parameters per page(!) The program can emulate a DW 8000 so that other computers with DW 8000 libraries can download directly into LIB-8000 (!) 40 page manual.

COMMODORE 64/128 Synthetix also has a scaled down version of LIB-8000 for C-64/128 computers. On-screen editing of all parameters, compatible with all Midi interfaces. Excellent manual. Store up to 40 banks of patches per disk. Patches can be swapped between banks for re-organization of sets. 9-character names can be given to each patch. The program includes a programmable random patch generator, like the IBM version, but you can't save your randomizing schemes to disk, just the sounds which result. The default mode randomizes OSC 1/2 waveforms and octaves as well as filter cutoff, resonance, tracking and EG intensity. The manual is concise, clearly written and well laid out. There are some extra features not discussed in the manual. You can use the program to filter midi data in real-time and assign splits, layers, and midi channels to your DW 8000 midi output before sending it on to your other Midi modules. Free telephone and Electronic mail support for both Synthetix programs.

COMMODORE 64/128 KORG distributes an editor librarian (list $125) with the following features. You can have four banks of patches in memory at one time and swap sounds between these banks. You can send and retrieve sounds from the synth and disk. On screen editing of all parameters. So far, very much like the above program. A mini-sequencer is included to help with editing EX 8000 modules (as far as I'm concerned you need a Midi merger to edit sound modules using computers.) The patch generator is quite different than any other I've seen. It does not involve any randomization (I'll buy that!) but substitutes choices of 8 types of paramter settings (called "Macros") for each of the following parameter groups: VCF EG, VCA EG, FILTER, DDL, LFO, OSC (waveforms, etc) Keyboard (velocity, etc.) and Miscellaneous (mono/poly, noise, auto bend, etc.) So far so good. However, neither the macros nor the parameter groups to which they are applied are user programmable so you are at the mercy of the creators concept (or lack of concept) about sound creation. With names like: Percussive,Sustaining,Double Attack,Brass,Wind,Slow Rise, etc. I am frankly dubious that the resulting sounds would be much more interesting than the infamous Korg presets. However, from here you can go on to edit the sounds to your tastes, so this might be a useful feature.

MAC A version of the same program is available for the Mac. It's features and price are similar. However, it supports mouse operations such as click, drag, etc. It does not have (thankfully) the Macro section. It does have something stupendous in its place: cut-and-paste editing of parameter settings between patches. If that feature is well implemented it would make me consider buying a MacIntosh. A less inspired feature is the graphic display of the same crude drawings of the "waveforms" that already deface the DW 8000's front panel. No, of course you can't do anything with them, just look at them some more and wonder why. There is an interface for loading and dumping programs to the MEX. Both programs were developed for KORG by PJM Associates of Greenwich ,CT. Since I have not seen these programs (and haven't heard of anyone outside Korg spokespeople who have) I can only take it on faith that they actually exist.