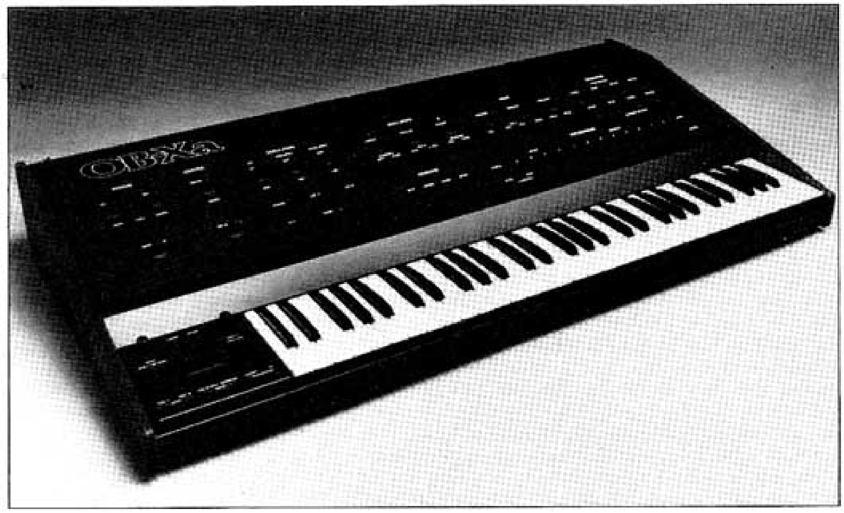
DOMINIC MILANO
KEYBOARD REPORT
Oberheim OB-Xa



OBERHEIM ELECTRONICS introduced their first polyphonic synthesizer in 1975 at the Chicago NAMM show. This instrument featured Oberheim Expander Modules (two VCOs, a multi-mode VCF, envelope generators, and an LFO) coupled with a keyboard. It was called the first 'true' polyphonic synthesizer on the market by marketing people who were into making the distinction between instruments with separate tone-generating circuitry for each note played and those that used organ-type divide-down circuitry for their oscillators. These modular Oberheim polyphonic instruments were available in 4-, 6-, and 8-voice versions. Later, when complaints started coming in from musicians who were finding it hard to adjust the settings on all the modules in live performance situations, Oberheim put out a programming module for these machines. But it wasn't quite enough to quiet complaints, since you couldn't store values for all the front-panel controls in computer-based memory.

Enter the 'fully programmable' polyphonic synthesizer. Oberheim's entry into the market of fully programmable instruments was the well-known OB-X, which came out in 1979. Like its predecessor, it was available in 4-, 6-, and 8-note versions. However, the market place is filled with competition in this particular field, and changes and updates are becoming quite commonplace. In early 1981, the updated OB-X, called the OB-Xa, was introduced. The OB-Xa features a few extras not found on the OB-X, but Oberheim was able to offer it for only a few hundred dollars more than the OB-X because a lot of the changes involved software (computer programming) rather than added hardware. There are also a few extra switches on the front panel, and other switches have been reassigned new tasks. More noticeably, the front panel graphics have been completely redesigned, much to the betterment of the instrument.

As with all of the instruments in this class, the separate voices are under command of a single set of panel controls, so the sounds are mostly homogeneous. However, the instrument does feature a programmable keyboard split so a different sound can be played on the upper portion of the keyboard than on the lower, and a double mode of operation which couples two sounds onto one note. We'll talk about these features below. Each synthesizer voice is made up of the standard two VCOs, a VCF, a VCA, two ADSR envelope generators, and an LFO modulation oscillator. The instrument is available in 8-and 6-voice configurations, but the folks at Oberheim tell us that they're selling mainly 8-voice instruments, since in split mode a 6-voice instrument would have only three voices on each half of the keyboard, while in double mode it would only sound three notes overall. As you may have guessed, we got an 8-voice for review.

The Keyboard. The keyboard has 61 notes, C to C. It has three modes of operation — split, double, and normal. Split divides the keyboard into two separate parts, and the location of the split is programmable. Double mode couples two sounds onto one note. Normal mode distributes the same sound across the entire keyboard. Two switches in the left-hand controller section transpose the keyboard up or down one octave. This transposition isn't programmable. There is another transposition that is effective with the split and double modes. When in split or double mode, you can transpose either the upper four voices or the lower four voices (or both) by hitting the panel controls in a particular sequence. The OB-Xa will then remember this transposition as long as you stay in the same split or double location.

The Programs. You get a total of 32 user-writable programs. These come preset at the factory, but you can always rewrite them with sounds of your own. There is also a cassette interface for storing your patches on standard cassette tapes, and the instrument comes with a cassette with the factory presets on it, so even if you accidentally destroy a factory preset that you really like, you'll still have that sound around on a tape. The 32 programs are divided into four groups, with eight programs in each group. There is a separate switch for each of these groups (A, B, C, D) so you don't have to step through them with a single switch when you're in group A and you want to get to group D.

In addition to the 32 patches, there are eight program locations for storing split and double program combinations. These eight slots contain the following information: whether the mode is split or double, where the split location is, and which of the 32 patches are to be called up. Once you're in split or double mode, you can still call up a different patch to replace either of the two temporarily if you'd like. The way Oberheim has designed the logic of the split and double mode controls seems somewhat awkward at first. For example, you can't go straight from split to double mode or vice-versa, you have to exit back into normal mode first. But we're sure you could get used to this without much trouble. The location of the split is factory-programmed as the third C from the bottom of the keyboard. If you want to change the location of the split while you're playing, all you have to do is hold down the split switch and press a key. The key that you press becomes the lowest note of the upper keyboard. Storing a new split location in memory is more complicated; the owner's manual explains how to do this.

Since there is only one set of panel controls, you might ask how the instrument 'knows' which patch you're adjusting when you're in split or double mode. Next to the Split and Double switches are two more switches labelled Upper and Lower. When you've got one of these pressed (there's an LED inside all of the switches on the instrument to let you know that that function is on) then the panel is active for either the Upper or the Lower program accordingly, and any changes you make in the panel controls will affect only the sound indicated by the Upper or Lower switch.

The Edit Mode. This brings us to the edit function. The unit is always in the edit mode —turning a pot or hitting a switch has an immediate effect on whatever sound you have called up. That is, turning a knob to the right adds to the value in memory, and turning it to the left subtracts from the value, even though the position of the knob may not reflect the value. To get the position to reflect the value, it's necessary to rotate the knob to both extremes to clear it. To get back to the preset value, all you have to do is hit the program switch again. If you want to record whatever edited changes you've made, you push the record button, wait for the LED to come on (it takes a few seconds), and then hit the group button and program button for the position you want the program in, just as if you were writing a new patch.

The Left-hand Controls. In this section are 12 non-programmable controls. The fact that they aren't programmed is actually kind of nice, but in some instances it can also be a drawback. In the middle of the left-hand controller section are two spring-loaded, return-to-center levers. The outer lever is used for controlling modulation, and the inner one is for pitch-bending. The mod lever only works when you pull it towards yourself. The pitch lever works in both directions — towards you bends the pitch up and away from you bends it downward. As you may notice, this is the opposite direction from the way a standard pitch-bending wheel works, but pulling the lever toward you to raise the pitch doesn't feel at all unnatural when you get used to it. There are four switches that control the routing of the pitch and mod levers. For the modulation, you can route it to either OSC 1, OSC 2, or both. For the pitch-bend, you can select either both oscillators or only OSC 2, and you can adjust the range of the bend to either narrow (a whole step) or wide (an octave). These ranges are the same in both directions, and are exactly a whole-step and an octave — nice.

In addition to the controls we've just talked about are the octave switches for transposing the keyboard and two switches for routing the levers to the upper or lower program only. There is also a separate, non-programmable LFO in this section. It has sine and positive-going sawtooth waveshapes. The rate is adjustable from .1 to 20 Hz. The rate pot acts as a pull-on switch for the sawtooth waveform. There is a pot for depth (controls the initial amount of LFO modulation) which can be pulled up to turn it on. The lever adds to the value the depth pot is set at. In effect, this lets you preset a modulation amount, which can be turned on at any time by pulling the knob up.

A note about the levers: Many people have gotten used to Minimoog-like pitch and mod wheels. The levers have their uses. It's easier to learn a natural-sounding vibrato technique on the pitch lever than it is on a pitch wheel. The levers are comfortable to play as long as you keep your thumb anchored just under the transpose switches and play the levers with the index and middle fingers of your left hand. However, bending pitch downward by pushing the lever away from you feels somewhat awkward in this position.

The Panel Controls. From left to right on the front panel of the instrument are six sections: the Manual, Control, Modulation, Oscillator, Filter, and Envelope sections.

The Manual Section. In this group of non-programmable controls are the master volume, balance, auto (tune), hold, chord, and master tuning controls. The master volume and tuning pots are fairly self-explanatory (there is a dead zone at 12 o'clock on the tuning control so that the unit will be at A-440 without any minute adjustments being necessary). The balance control lets you adjust the relative volumes of the upper and lower programs when the unit is in either split or double mode. The automatic tuning is incredibly fast (it seems to take less than one second). The hold button lets you sustain indefinitely any note or notes you are holding down when you hit the button. The chord switch is used to pile up notes to be played in a sort of unison mode. You can put up to seven notes (in an 8-voice) onto one key, so you can play any chord you like with one finger. This makes for a really massive sound. Unfortunately, this feature isn't programmable, so every time you want to use this function, you have to program it in real time. If you don't pile all the available voices up, you'll have some left over for playing a solo above the chord. The chord function operates on a low-note priority, which leaves you with some tricky playing to do, since if you let up on the note that's controlling the chord, the chord will jump up to the lead line note that you're playing above it. Also, if you play too many notes with your right hand, there is a built-in rob function which will take notes away from the held chord as needed (the first notes entered into the chord will go away first). If you hit the chord button while you're only holding down one key, the machine will make a major triad using that note as the root.

The Control Section. Within this group of controls are two pots and a switch. One pot is used to control the amount of glide or portamento (which works both monophonically and polyphonically), the switch puts the machine into unison mode (single-trigger, low-note priority), and the other pot lets you detune OSC 2. This control might seem to be out of place so far from the other controls for OSC 2, but if indeed it is one that you use often, you may find it nice to have in a place where your left hand can get to it in a hurry. The poly glide is a great effect (we're sure that you could get tired of it quickly if you used it too much, but it's lots of fun and sounds great all the same). When you have a long release time set on the envelope generators, have the glide on, and play a note but let up on it before the glide reaches that note, the pitch will continue to glide until it reaches the note, which is a great feature not found on a lot of synthesizers.

The Modulation Section. There are three rotary pots in this section: an LFO rate control (continuously variable from .1 to 20 Hz), an LFO depth (amount) control, and a pulse width modulation depth (amount) control. Six switches let you choose the routing of the modulation (to OSC 1, OSC 2, and the VCF, in any combination) the sources of the modulation (a sine wave, a square wave, or sample-and-hold, only one of which is available at a time), and the destination of the pulse width mod (OSC 1, OSC 2, or both). It's unfortunate that the modulation waveforms aren't a bit more varied or combinable. Also, there is no provision for modulating OSC 1 with OSC 2.

The Oscillator Section. Two VCOs for each voice are in this section. An octave selector pot is provided for adjusting the initial frequency of OSC 1 in increments of one octave, over a four-octave range. Below this pot are two switches for determining whether the output of the first oscillator is a sawtooth wave or a pulse wave (you can't get both simultaneously). The middle pot adjusts the pulse width of both OSC 1 and OSC 2 simultaneously. Fully counter-clockwise is a square wave; fully clockwise is a pulse wave with a 5% duty cycle. You can't control the pulse width of either oscillator independently of the other by any means other than modulating one but not the other. The frequency control for OSC 2 is a rotary pot that's quantized in half-step increments over a five-octave range. A switch lets you sync OSC 2 to OSC 1 while another switch lets you route the filter's ADSR envelope generator to control the frequency of OSC 2. The amount of modulation from the ADSR is controlled by the modulation amount pot in the filter section. With this control set at maximum, OSC 2 will change in pitch by one octave. The waveshapes available from OSC 2 are the same as OSC 1, i.e. sawtooth or pulse, but not both.

The Filter Section. The filter is a lowpass filter. There are four switches which act as a mixer, selecting fixed amounts of output signal from the oscillators to the filter. The first of these switches selects all or none of the signal from OSC 1. The next two switches select either all, half, or none of the signal from OSC 2. An interesting characteristic of these three switches is that when they are turned off, the LEDs in the Oscillator section under the frequency pots for OSC 1 and 2 don't light to tell you what waveform you've got called up on the corresponding VCO. That is a big inconvenience when you aren't all that familiar with what you've got programmed as the waveform for the two VCOs, and you've got one or both of the VCOs turned off in the patch but want to turn them on later within the same patch. The fourth switch in the VCF section is for routing none or all of the noise source's signal into the filter. It's too bad you can't adjust the level of the noise generator in relation to the VCOs for getting pipe chiff sounds and whatnot. In fact, the lack of a fully adjustable mixer is a significant limitation in the instrument. Another switch in the VCF section controls whether the filter has a 12dB/octave (two-pole) slope or a 24dB/octave (four-pole) slope. This is nice for added textural possibilities. And the last switch in the VCF section lets you route all or none of the keyboard voltage to control the filter cutoff frequency. The filter will also track the transpose and master controls.

The three pots in the filter section control the cutoff frequency, the amount of resonance (regeneration, feedback, etc.), and the amount of modulation from the ADSR envelope generator. This last pot also acts on the amount of modulation sent to OSC 2 from the filter's ADSR. Having these two functions linked rather than independent is another minor limitation.

The Envelopes. Those familiar with ADSR envelope generators really don't need them explained. There are two ADSRs in this section — one for the filter and one for the VCA.

The Rear Panel. On the rear panel, left to right as you're facing the back of the instrument, are: three 1/4" output jacks (stereo left and right and a mono output), cassette interface jacks and switches, various footpedal and footswitch input jacks, a multipin computer interface, and a power cord input jack. The cassette interface has become somewhat standard on programmable machines, and is used for storing and loading patches on cassette tapes. The various footpedal and switch input jacks are 1/4" type jacks. One of these is for connecting up an on-off sustain pedal switch for piano-type sustain effects. The problem here is that when a jack is hooked to this input, the release times on the ADSRs immediately go to full when the switch is closed. This definitely wasn't a good idea, as it doesn't allow you to adjust the release time for certain effects that you can get with a sustain pedal. The decay time on a piano, for example, is much shorter than the OB-Xa's full release time value. This limitation means that the sustain pedal option is of very limited usefulness. The program advance pedal input lets you advance programs with a footpedal — a much more useful feature. The hold footswitch is for externally controlling the hold function with a footswitch, so you can create a drone while both hands are busy. This is another useful pedal input.

The footpedal inputs are set up to be used with any voltage pedal with a 50kΩ linear pot in it. These footpedal inputs let you control the amount of vibrato from the LFO in the left-hand controller section, and the cutoff frequency of the filter. Then there's the multipin connector for interfacing the OB-Xa with a computer. This jack, the owner's manual explains, is for interfacing the instrument with future Oberheim pro-

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