The Yamaha VP1 Editor

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Yamaha Software from 1994

Tested on a Power Mac G3 from 1998

On Mac OS 9.2.2

INDEX

The Yamaha VP1 ____

The Yamaha VP1 was produced as a prototype polyphonic Virtual Acoustic synthesizer, in parallel with the Yamaha VL1. The two look very alike visually, like they came out of the same mold. The similarities end there, however. The VP1 is VERY different inside from the VL1. Basically, the VL1 is based on parametric equations developed at Stanford University for emulating digital waveguides. It has a very complex, calculated Driver with multiple modifiers on the front end and a simple resonator on the back end. It's great for creating expressive wind or bowed instruments. The VP1 is the opposite. It has a simple recorded set of Drivers on the front end, and goes into great depth calculating the Resonator on the back end. It primarily emulates plucked or struck instruments, in particular instruments that are inharmonic.

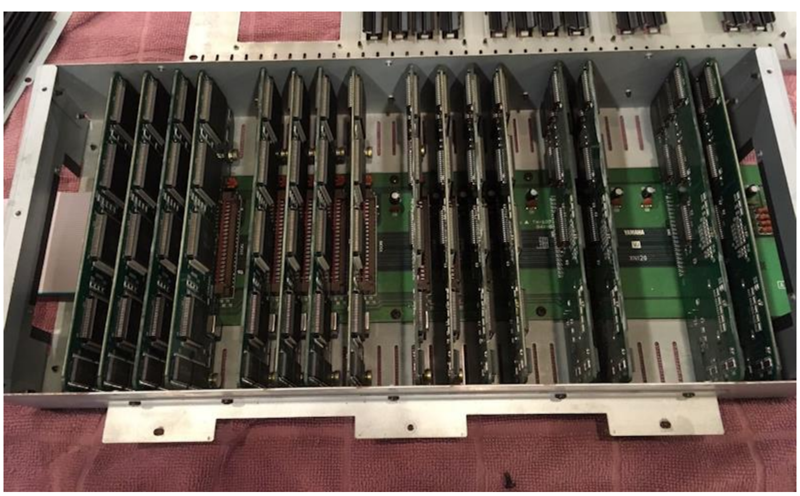
After listening to Manny Fernandez' discussion on SoundCloud about Yamaha VL, I can see why the VP1 used recorded Drivers. Apparently the VL model is very unstable and requires a lot of effort to keep the parameters in the sweet spot, where the sound resonates and stays alive and is in tune. Adjustments are required for every VL note up the scale. The selections you were allowed in the VL Visual Editor actually link into a lookup table for each combination, to ensure the parameters in the model are stable for every possible combination of Driver and Resonator. The problems in the VL1 are all due to the 11 or so wind modifiers added onto the model on the front end. Each modifier affects all of the others. Change one or two, and you have to go back and adjust the others to knock it back into stability and back into tune. Manny calls it "chasing your tail"!

To clarify, the VP1 Drivers are just an input pulse, it's not a complete note recording like a sample-based synthesizer uses. A sample based synthesizer like the Yamaha Motif uses a long, 5 second note sample as a short cut to reduce the computing power required, the VP1 does not. The VP1 uses a transient input pulse to start the vibration in the model, which sounds nothing like the instrument it's named after. The pulse goes into the Resonator string or tube where the sound wave forms, creating the sound wave you hear coming out of the instrument.

Despite using simple, recorded input pulses, the timbre of the VP1 is very complex. The simplest VP1 patch includes a two-part Driver module, with a voiced instrument oscillator and an unvoiced noise oscillator in each part. These four oscillators feed into two pairs of Delay Loop processors (with cross-connections) and then into the Effects module. This is called an Element. You can use up to FOUR Elements in one patch, for a total of sixteen complex oscillators per note. Every parameter in each section of the patch is accessible in the Editor and most are mappable to a controller or envelope. A large number of parameters are also graphed to behave differently over different sections of the keyboard. One very inauspicious parameter in the Editor is called “Stiffness” which we’ll discuss in depth due to it’s effect on the harmonics. It’s one key to the VP1’s signature sound. I finally got to read the official Yamaha Editor manual, but the Yamaha Engineers really didn’t say much of anything about this parameter. We’ll cover it in the Appendix A. Where programming an FM synth is a real challenge just to get the timbre right, the VP1 makes it a bit easier. It still takes a spectrum analyzer to get the timbre right on the money, though. You spend more of your programming time on the VP1 setting up controllers and behaviors rather than building the timbre up from scratch.

The VL1 and VP1 are very similar in one main aspect, controller behaviors. The VL1 uses controller inputs to mold and shape the sound of the instrument in ways that emulate human performance on an acoustic instrument. In particular, the Saxophone is at the heart of the mathematical model, most of the controllers modify the sound driver, the mouthpiece of the instrument. The VP1 does the same, but in a different way. It takes in a massive number of hardware controller inputs and maps them to every aspect of the Resonator model. The VP1 natively has a huge collection of controllers, including Velocity, Aftertouch, three mod wheels, two sliders, a 2-axis trackball, three foot pedals, two foot switches, three scene control buttons with a slider/foot pedal, and a Breath controller input. That’s an insane number of hardware controllers. A live musician can handle about five or six competently, but over eighteen at once? Almost every parameter in the model can be mapped to a physical controller in the Editor. The sheer number of behavior controls is staggering, altering the timbre in ways you can’t easily get your head around. Overall, the VP1 is a very complex instrument to program.

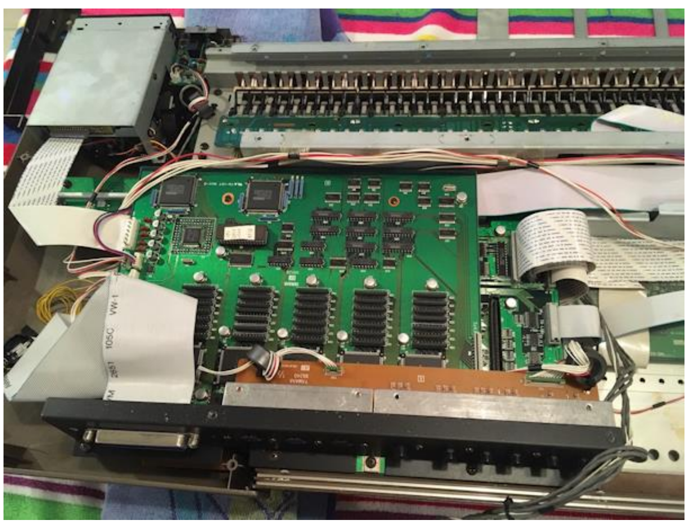
To accomplish all of the required calculations the VP1 requires 32 custom Yamaha DSP processing chips and 32 VLSI computer processors. That’s 64 processors! Imagine, Yamaha built a 64-core computer into the VP1 back in 1994 when all of the personal computers of the day only had ONE. The number of MIPS of computing power in the VP1 is somewhere in between a Cray X-MP and Cray Y-MP supercomputer from just 8 years earlier. Today an intel 10-core i9 processor is about 400 times faster, but to assemble a 64-core computer back in 1994 was unbelievable! We just barely hit the 64-core level in the Ryzen Threadripper in 2020, some 26 years later. Yamaha was truly way ahead in their thinking on this one. The amount of computer hardware also dictates the size of the shell. Looking at the insides, it’s packed. There isn’t any spare room in there, just a bit on one end where the mod wheels are located. It makes sense for it to have a 76 key keyboard, the shell had to be that long anyway. It’s also really heavy, 63 lbs.

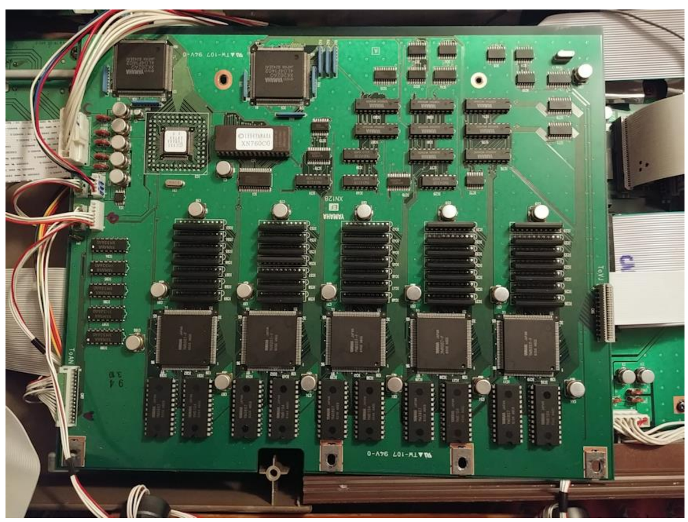


Each of the 16 voice circuit boards (in the central enclosure above and pictured individually below) has 4 VLSI Processors on it. The VL1 uses 3 of the same processors per voice, it’s less intensive to model than what’s in the VP1. The original pricing of the VP1 voice boards was $624 apiece (which Manny and Reinhold determined during their rebuild) so the cost of producing the VP1 added up quickly just from the cost of the processor boards. That adds up to $9,984 just in the 16 voice boards. The heat from the enclosure caused problems in the first few prototypes, which was corrected soon afterwards. Only Reinhold Heil's VP1 prototype is in the original configuration. The others, the pre-production models, were all retrofitted with cooling vents all along the back edge of the synthesizer, and slots in the bottom. One prototype without the vents was destroyed during shipping, which we'll talk about in a minute.

There are also 2 more Processors on the main control board (below left in the top left corner) along with a large number of stacked RAM chips, and another 7 Processors on the effects circuit board which mounts underneath it (below right) with another set of stacked RAM chips.

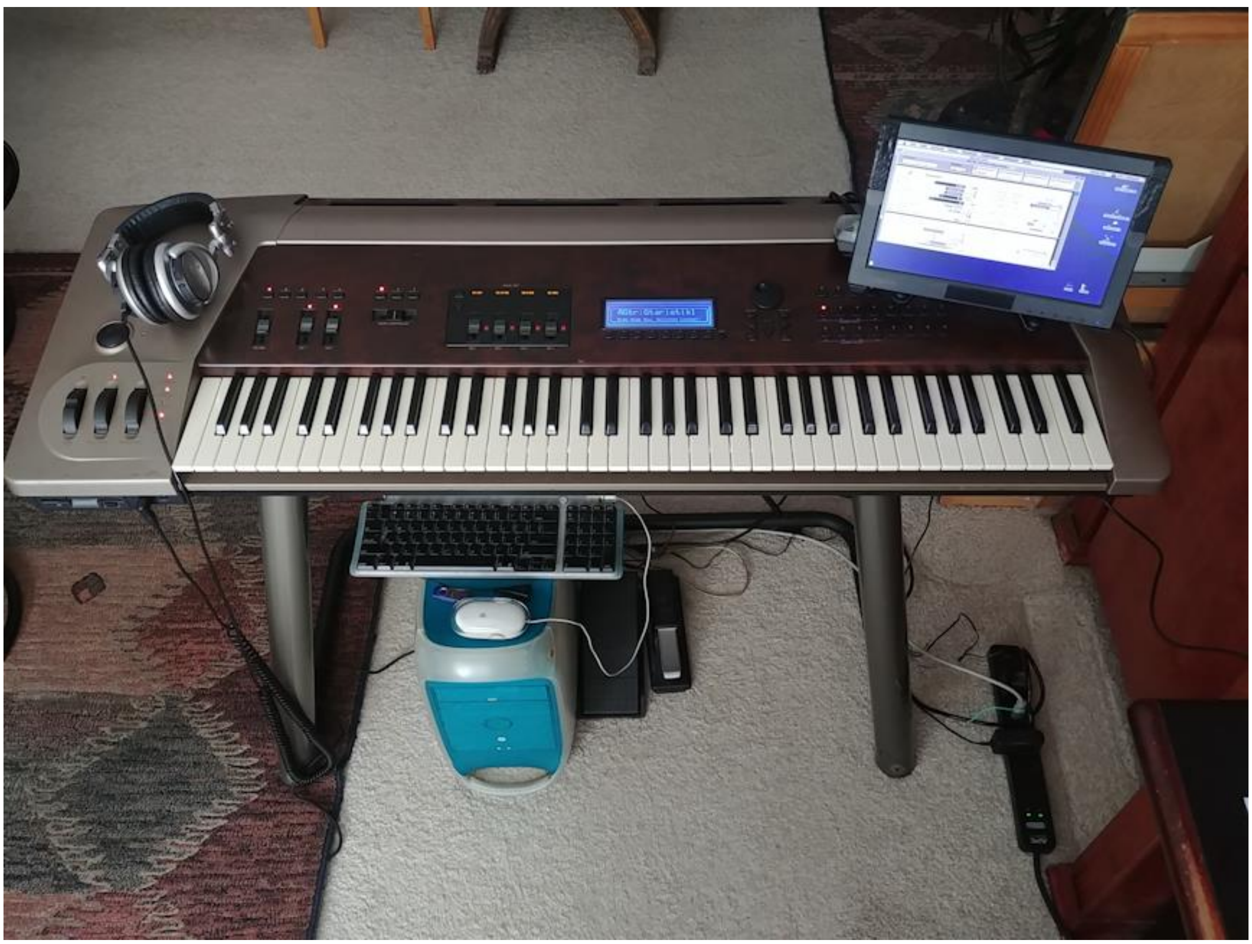
This is also the reason Yamaha initially wanted to charge about $20,000 for the VP1 prototypes (using the 1991 exchange rate) they were thinking of the cost of the 64+9 VLSI processors inside the synth. They cut the price significantly two months later when they realized nobody was going to spend that much money on one synthesizer. They were only selling off the prototypes anyway, not trying to recoup all of the development costs of the project. Daniel was offered one, and so was Reinhold, but the price was still 'real money' and more than most musicians wanted to spend on one keyboard at the time. Times have changed, I see CS-80's and DX1's listed at exorbitant prices on Reverb. These prices aren't real transaction prices, but they do suggest a trend for collectable pieces like the VP1.





It took ten years for computer horsepower to catch up to Yamaha's vision. When it did, Yamaha built an 8-voice (not 16) prototype version of the VP1 in a rack with a CS1X keyboard chassis (only one mod wheel) using fewer, faster processors. Scott Kinsey called it the VPX in the ProSynth live-stream #120 (on YouTube now). It was slated to be released at the $2000 price point, Scott thought "it played and sounded awesome!" When it was demoed to a group of studio musicians, the reception fell fatally flat. Some felt that without all the physical controllers of the original, the sound was just too simple to match expectations. Other musicians just didn't care for it. The morphing sound capability of the new prototype was well below the original VP1 and the new machine never got past the initial phase of development. (Personally, I think the fault was probably in the patch programming, or lack of it.) In contrast, the VL70m was a very expressive, single voice synth with far fewer physical controllers and was doing modestly well in the marketplace. The VL70m also only contained two of the VLSI processors and was a more reasonable $699.

Sadly, very few Yamaha VP1's were ever produced, Yamaha internally thinks there may have only been 8 or 10 in total. Looking at serial numbers, there might be 12 to 15. One is in the Yamaha museum in Hamamatsu, Japan, another is at Yamaha Europe in Germany, and a private owner has another pair in Germany (one which was sold by the Music Store in Cologne). Reinhold Heil, a music producer in the US, owns one in California (no cooling vents and he refinished his Black recently) which he purchased in Germany. It had a massive crack in the left corner which was miraculously fixed when they located the one and only spare shell end section in existence. Reinhold moved to Hawaii, but Manny still has his VP1 for a few more months. One other famous Japanese musician Yasunari Takahashi purchased one a few decades ago, but has sold it in the last few years (we have no idea who has this one) and Kurt Ader at KAPro has one without a working programming interface. At the moment of writing this, his VP1 is in the shop with some major problems they've been trying to fix for the last year. He commented "It's a nightmare" on the forum. I've also found another reference to a possible VP1 owner online, Ryuichi Sakamoto of the Yellow Magic Orchestra who had one listed in his touring gear from back in 1995, and I found a Japanese auction for a VP1 online from 2010. I have no idea which one that was.



There is also the VP1 prototype which Yamaha lent out for voicing in the US (no cooling vents) which was used by Scott Kinsey and by Michael Jackson's keyboard player, Michael Boddicker, on the Bad album. It was ordered crushed by US customs during shipping back to Germany for not having the proper electrical labeling. A sad event. Another VP1 has also surfaced at an IT university in Japan, where a grad student there brought it back to life, the one I now have access to. Thanks Peter! It was hiding in the corner of a music room with a dead battery, along with a VL1. No one is sure which ones have the programming interface and which ones don't, but most of the prototypes do seem to have the hardware installed in the synth. The serial interface and software were apparently an option upon purchase. Mine came without them, I engineered an interface with Manny Fernandez's help, and I got the software from Daniel Forro a year before this VP1 was even found. You really meet great people in this industry! In addition, another unit has just shown up on YouTube, owned by Yoshiki Hoshi of the group Himekami in Japan. That makes ten we can account for: eight alive, one with severe problems, and one destroyed. The few that are still in hiding are most likely in Japan.

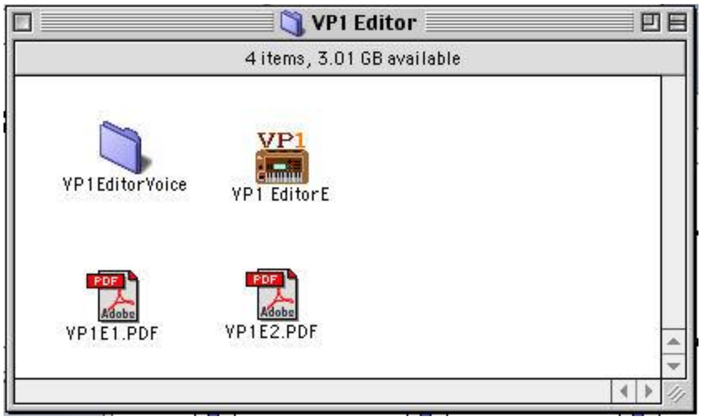
Most of the patch programming at the factory was done using a custom hardware interface called OTOMI. It was on a Sun Sparc Station micro-computer set up in-house with a touch screen to program patches. Looking at dates, it was probably a SPARCstation 10.

To allow outside specialists to program patches to create a commercial patch set, Yamaha produced a software editor written for Macintosh computers from the same era. A friend of mine in Germany had a photo of an original, a Power Mac 6100 series with all of the accessories, manuals and disks displayed below.

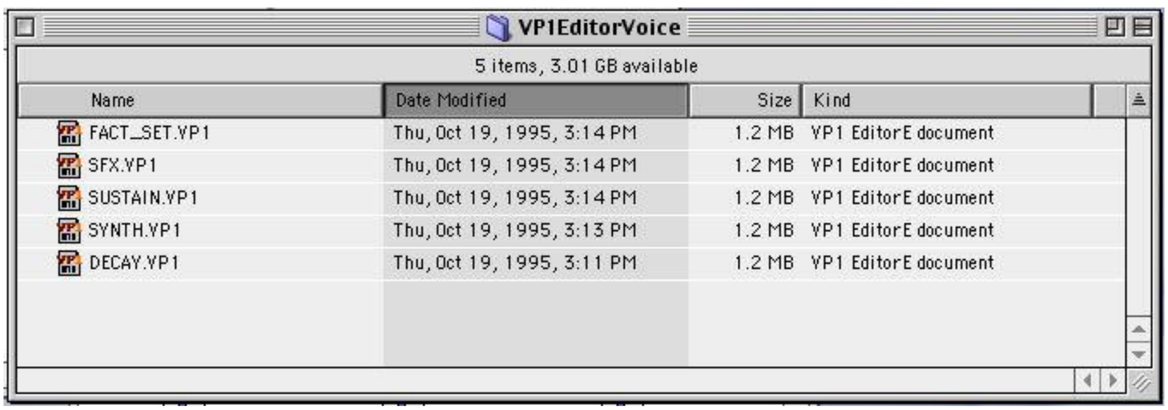
I have a working copy of the Macintosh OS7/8/9 editor (thanks to Daniel!) and recordings from Daniel Forro who worked with Yamaha as a product demonstrator and who produced the only recorded music we have played on the VP1 back in the 1990's. To use the editor, you connect the synthesizer up to a Macintosh via one of the serial ports, connected to a serial to parallel converter and a custom ribbon cable. The VP1 has a 50-pin Centronics parallel connector on the back, like printers had back in the 1980's. Details of the programming interface are covered in the back of this manual. If you need to build one, go there. The standard Mac interface is known to be somewhat unstable and unreliable, which I've fixed. The details are in Appendix B.

Due to the sheer complexity of the internal models, the full potential of the synth was never really tapped into. People have commented over the years that very few individuals had a chance to work with the VP1 in the early days, so very few patches actually exist for the synth. Manny mentioned he and the other programmers had the instruments for over a year, getting to know the machine it was so complicated. After the program was cancelled, the VP1 never really got the attention required to properly program a more complete set of patches which hurt the development of the synth. In contrast, the VL1 received extensive programming time which really shows in the quality of the patches it was released with. Despite this aspect of the development, the cost of manufacturing the VP1 was ultimately it's downfall. It was listed at 2,700,000 yen in the sales literature, which was equal to about $23,000 US back in 1994. Before this brochure the price was closer to $30,000 talking to Reinhold.

You can see the editor icon below in the folder on my PowerMac G3, running on Mac OS 9.2.2. The VP1 Owners Manual and the Performance Notes were included with the Editor, the two pdf's in the next row. The VP1 Editor Voice folder contains the five factory voice banks. The Editor also came with a printed manual, which contains a few more details on how the editing interface works, but I was surprised at how little programming direction there was in there. I've had to figure out everything empirically using the Editor interface, and by studying physical modeling on my own. Manny Fernandez mentioned that when Dave Polich received one of the prototypes for voicing, Yamaha Engineers told him something like ***"…this is the VP1, we do not know what it can do".***



The Editor comes with a folder labeled 'VP1EditorVoice' with five patch banks in it. Up until now, I only knew about the main Factory patch bank which is detailed in the Performance Notes. Most of the demos you've heard are from the Factory-Set patch bank. (See below.) Manny has recorded the 10 factory demos and posted them up on YouTube, and I've expanded the number of patch demo songs on my website to about 30. Most are good, a couple aren't.



Besides the FACT_SET.VP1 patch bank, the other four patch banks are mostly unknown, only a few of those patches have ever surfaced. A small set have shown up in demo recordings by Yasunari Takahashi on SoundCloud, including BigBlower, BigMyPiano, Guitarra, HyperRock, HydroStorm, CosMoom, MiamiOcean. There is also a 24-bit sample library you can purchase online on ebay and elsewhere. A few of the patch names in there also match names of the unknown patches. And possibly, Kurt Ader at KAPro might have used some of these unknown patches in his sound banks for the Motif and Montage synthesizers.

It has been great fun to finally hear what the other four banks of patches sound like! Manny Fernandez, one of the original patch programmers, described that the additional patches sounded very similar to the Factory set. After hearing the VP1 now, it's hard to really define what a single patch sounds like, even with recordings to refer to. There are SO MANY controller options in this synth, the timbre bends in so many ways, you can hear the same patch played with different settings and not recognize it.

Patch Lists ____

You can see lists of all of the patches in the five patch banks below. Each bank came on one Floppy Disk, the five color coded disks in the photograph below, which load directly into the VP1. The other two disks include one with the Mac Editor on it, and a compressed backup of saved patches off the VP1. The two manuals include the Operator Manual for the VP1, the Performance Notes detailing specifics on using each of the Factory-Set of patches, and on the table is a printed copy of the Guide Book for the Editor. The box on the right holds the serial-to-parallel interface for connecting the VP1 to an older Macintosh computer. I have the programming interface figured out, which is in Appendix B.

First up is the Factory Set, the patches everyone is familiar with from the Performance Notes. None of the other patch sets have a Performance Notes pdf.

FACT_SET.VP1

Output Null Port

1 : SpacePorta	17 : Viologue	33 : Soodosynth	49 : ElpBass
2 : WaterBell	18 : Jody	34 : BigSlowPad	50 : ElePiano
3 : StrSqueak	19 : RoughStr	35 : BigDeepPad	51 : Bravinova
4 : OrcheStr	20 : SlowStrs	36 : DeepBelPad	52 : EG-Go!
5 : VowBra	21 : Eleanor	37 : DistModPad	53 : Classical
6 : NoisyKing	22 : FoggyAtk	38 : Aquarimba	54 : Harpy
7 : Fantasy	23 : StrOrgan	39 : Vpepad	55 : Asiakoto
8 : Sweepy	24 : TearDrop	40 : MadVoxLead	56 : Miyabi
9 : HarpBell	25 : PsycheStr	41 : Ellegant	57 : StarDust
10 : HyperDrive	26 : OctBrs	42 : Celluloid	58 : WindChimes
11 : SteelSpeak	27 : RudeBrass	43 : VP Atk	59 : GlassWhine
12 : FeedBackB	28 : PressToFly	44 : FantaBell	60 : VP Seq
13 : Oriental	29 : WoodFamily	45 : ResoMetal	61 : AsianDance
14 : Clox	30 : WindMorph	46 : HardSeq	62 : SteamOrgan
15 : TalkDrum	31 : SyacScream	47 : Stormy	63 : Abyss
16 : Harmagedon	32 : MetalWind	48 : DigiCompin	64 : CosmoFctry

The SFX.VP1 patch set is predominantly a set of Special Effects patches. A few can carry a tune, but most are more percussive or are wild and crazy movie sound effects.

SFX.VP1

Output: Null Port

1 : Gong	17 : InSink	33 : BigBlower	49 : Waterfalls
2 : PercusivePC	18 : Hall-Of-Oz	34 : CmprestAir	50 : BigFly
3 : Sledgehammer	19 : Poltrgeist	35 : Lawnmow-me	51 : AlumiRings
4 : Ra	20 : LoneBeach	36 : V-Cycle	52 : MetaJungle
5 : GarakTaiko	21 : WatrWhistl	37 : SubHull	53 : HellHouse
6 : TalkTom	22 : WaterBell2	38 : MenAtWork	54 : KrazeeCykl
7 : TSUGARU	23 : ModPad	39 : BigMyTruck	55 : BellsInSeq
8 : HAWAII	24 : NoiseOrgan	40 : Shinkansen	56 : CrazyBow
9 : PerkSwirlz	25 : NoisePad	41 : SpaceMtn	57 : SpaceSymph
10 : FractalFan	26 : Hydrospher	42 : Bowscraper	58 : RadioConf.
11 : Ethnikola	27 : Akubi	43 : AirprtAlrt	59 : AnalogFX!
12 : Zepitar	28 : StrSqueak2	44 : Chase	60 : Belltazaar
13 : Caramba	29 : MarinGhost	45 : Hatsu	61 : CosmoAlert
14 : Slither	30 : Typhoon	46 : Shoumyow	62 : DarkCrystl
15 : TimpRoll	31 : Jura	47 : Umeki	63 : ZON
16 : I Scream	32 : VacuumPipe	48 : MetalFrog	64 : CosmoFleet

Even though the VP1 is mainly designed for plucked or struck sounds, the Oscillators have mode settings with selections for 1-shot, Cycle, and Free. 1-shot is a single strike and pluck method, but Cycle and Free have the ability to continuously loop part of the sample for sustained instrument sounds.

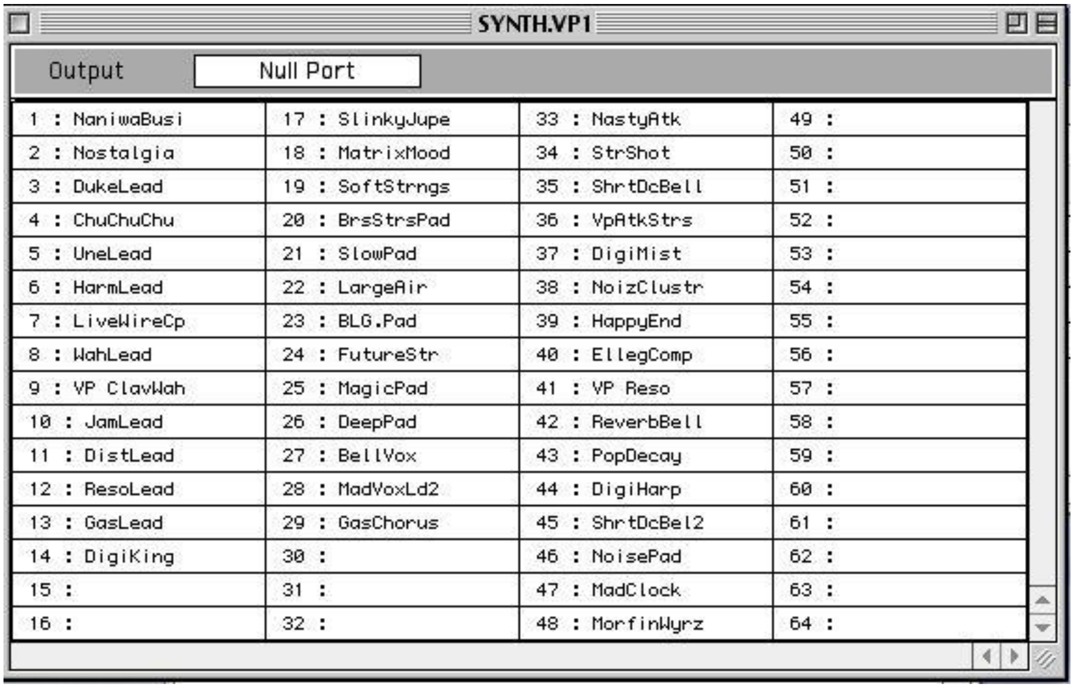
The SUSTAIN.VP1 patch set are mainly voices where the sound continues until the key is released, like string, brass, or wind instruments. The VP1 is not limited to just the F/VA Free Oscillation model mentioned in the literature, it can also access an S/VA Self Oscillation mode, though vastly different from the VL1 model.

SUSTAIN.VP1

Output: Null Port

1 : BowedMorph	17 : Psychestra	33 : PowBra	49 : VP Organ
2 : BowedAirs	18 : SteelHarp	34 : HyperWind	50 : Vocalise
3 : UnisonBows	19 : Harpy2	35 : Hrmownd	51 : FemaleChor
4 : VP Cello	20 : PizzaRoll	36 : SS.Sax	52 : MadChorus
5 : Vio.Fue	21 : L.A.Brass	37 : SynSax	53 : FrogChorus
6 : MelloRetro	22 : PolyBrass	38 : Saxoid	54 : Syakuhachi
7 : ReverseStr	23 : DawnBrass	39 : DigiFunnel	55 : Syou/suzu
8 : Jody2	24 : LiteBrs	40 : SynthWind1	56 : Bagpipe
9 : Jody3	25 : BrasFamily	41 : Shakatube	57 :
10 : SweetStrs	26 : OldBrsSect	42 : BcShaka	58 :
11 : S/H Strngs	27 : Bendicon	43 : NzFlt	59 :
12 : FastStrs	28 : Tromphon	44 : PanFlt	60 :
13 : DynaSyStr	29 : FacetiousB	45 : ClickPipe	61 :
14 : PowStr	30 : SynBraSec	46 : SynthWind2	62 :
15 : CowSinging	31 : VP Brass	47 : WoodPiper	63 :
16 : StrOrgan2	32 : AnalogBrs	48 : PipeOrg	64 :

The SYNTH.VP1 set are synthetic voices, more like an Analog Synth or FM Synth voices. Quite a number of the slots are empty in these patch banks, suggesting the unfinished nature of the VP1.

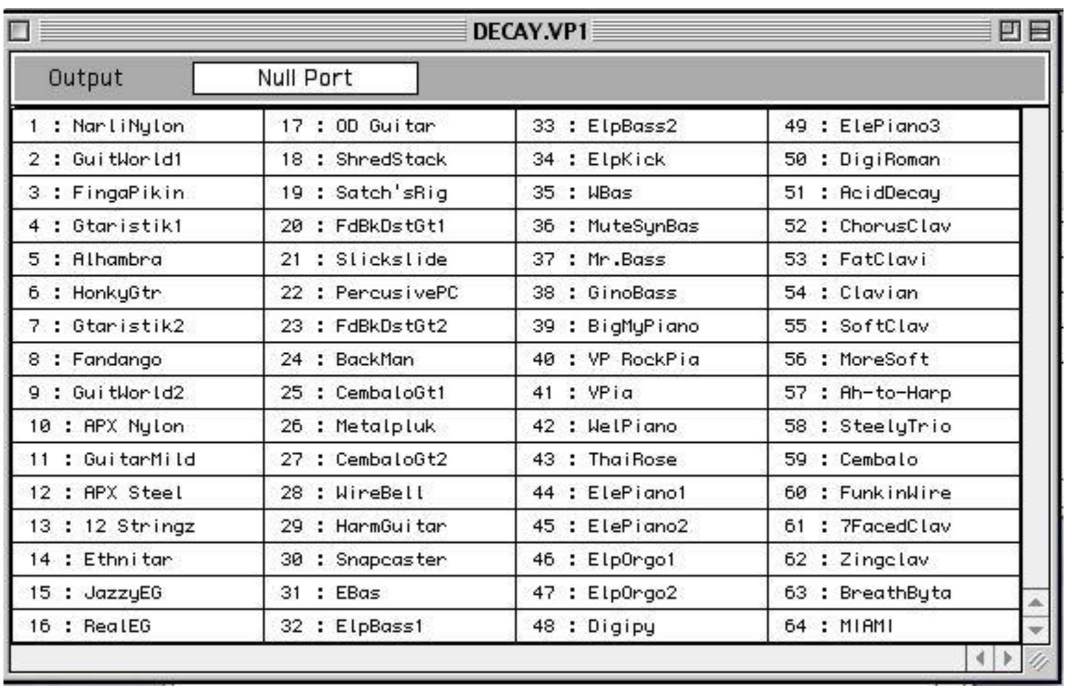


SYNTH.VP1

Output Null Port

1 : NaniwaBusi	17 : SlinkyJupe	33 : NastyAtk	49 :
2 : Nostalgia	18 : MatrixMood	34 : StrShot	50 :
3 : DukeLead	19 : SoftStrngs	35 : ShrtDcBell	51 :
4 : ChuChuChu	20 : BrsStrsPad	36 : VpAtkStrs	52 :
5 : UneLead	21 : SlowPad	37 : DigiMist	53 :
6 : HarmLead	22 : LargeAir	38 : NoizClustr	54 :
7 : LiveWireCp	23 : BLG.Pad	39 : HappyEnd	55 :
8 : WahLead	24 : FutureStr	40 : EllegComp	56 :
9 : VP ClavWah	25 : MagicPad	41 : VP Reso	57 :
10 : JamLead	26 : DeepPad	42 : ReverbBell	58 :
11 : DistLead	27 : BellVox	43 : PopDecay	59 :
12 : ResoLead	28 : MadVoxLd2	44 : DigiHarp	60 :
13 : GasLead	29 : GasChorus	45 : ShrtDcBel2	61 :
14 : DigiKing	30 :	46 : NoisePad	62 :
15 :	31 :	47 : MadClock	63 :
16 :	32 :	48 : MorfinWyrz	64 :

The DECAY.VP1 set below are plucked or struck instrument voices, the kind you expect from initial Karplus-Strong physical acoustic models. There are lots of Guitars in this set, as well as several pianos.



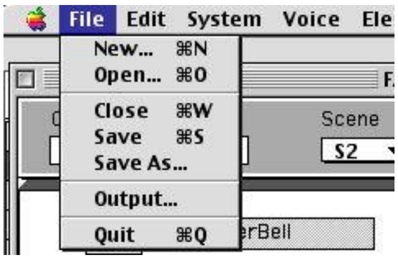
DECAY.VP1

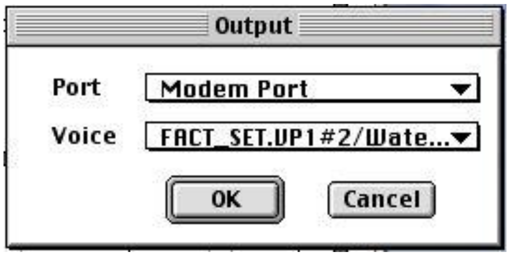
Output Null Port

1 : NarliNylon	17 : OD Guitar	33 : ElpBass2	49 : ElePiano3
2 : GuitWorld1	18 : ShredStack	34 : ElpKick	50 : DigiRoman
3 : FingaPikin	19 : Satch'sRig	35 : WBas	51 : AcidDecay
4 : Gtaristik1	20 : FdBkDstGt1	36 : MuteSynBas	52 : ChorusClav
5 : Alhambra	21 : Slickslide	37 : Mr.Bass	53 : FatClavi
6 : HonkyGtr	22 : PercusivePC	38 : GinoBass	54 : Clavian
7 : Gtaristik2	23 : FdBkDstGt2	39 : BigMyPiano	55 : SoftClav
8 : Fandango	24 : BackMan	40 : VP RockPia	56 : MoreSoft
9 : GuitWorld2	25 : CembaloGt1	41 : VPia	57 : Ah-to-Harp
10 : APX Nylon	26 : Metalpluk	42 : WelPiano	58 : SteelyTrio
11 : GuitarMild	27 : CembaloGt2	43 : ThaiRose	59 : Cembalo
12 : APX Steel	28 : WireBell	44 : ElePiano1	60 : FunkinWire
13 : 12 Stringz	29 : HarmGuitar	45 : ElePiano2	61 : 7FacedClav
14 : Ethnitar	30 : Snapcaster	46 : ElpOrgo1	62 : Zingclav
15 : JazzyEG	31 : EBas	47 : ElpOrgo2	63 : BreathByta
16 : RealEG	32 : ElpBass1	48 : Digipy	64 : MIAMI

The Main Menus ____

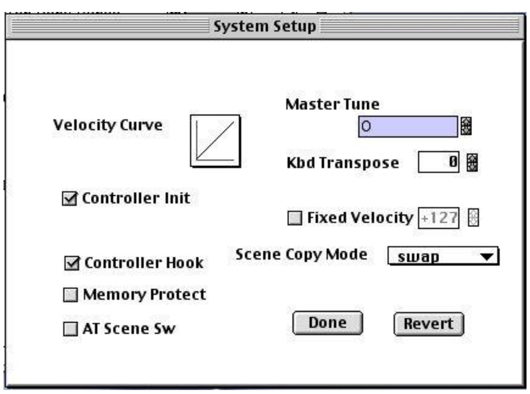
The main menus run across the top of the Mac screen and are used to access different sections of the Editor.
All but one of the **File Menu** selections deal with Patch Libraries (see below). 'Output' sets the output destination for the synth. Modem comms are at 38,400 dps and the Printer Port runs at 9,600 dps. Both used a Mac serial port. My new interface uses a USB port instead, and runs at 19,200 dps thanks to Manny's help.

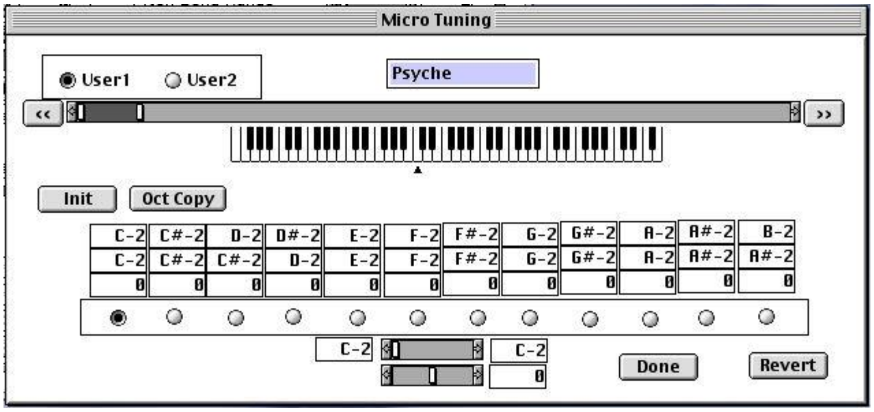




The **Edit Menu** has all of the standard selections, Undo, Cut, Copy, Paste, Clear, and Select All. It's used to assemble new patch libraries from the existing ones.

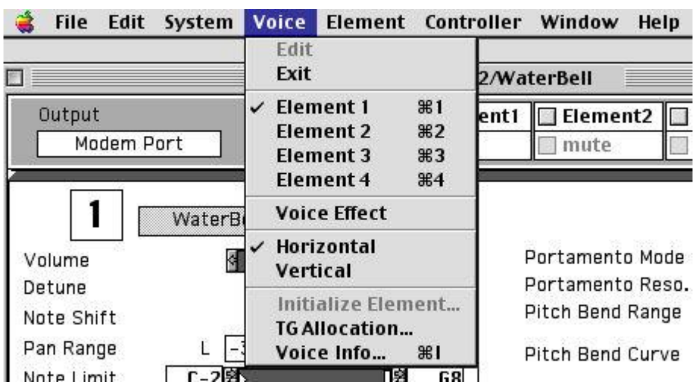
The **System Menu** contains items for globally setting up the VP1, and also for Micro-Tuning the keyboard note scale. The System Setup page contains the Memory Protect option, which is probably the only one you'll typically use, unchecked to edit settings. Controller Init and Controller Hook are explained in the manual, pages 3-14 and 3-15. The AT Scene Switch turns on Aftertouch control of the Scene Slider, manual page 3-13. On the Midi Setup page the Kbd Transmit Channel is really the only one you'll change, leaving the Basic Receive Channel on Omni. The Local checkbox disconnects the keyboard from the Tone Generator for when you use the VP1 as a controller for another synth, as described in the manual. I've heard Reinhold uses his VP1 as a controller quite often.





With Micro Tuning, it's not something you typically dig into. The standard micro tuning options (selected in the Elements Window) are usually more than adequate. This page allows you to create custom tunings, which are saved for loading later.

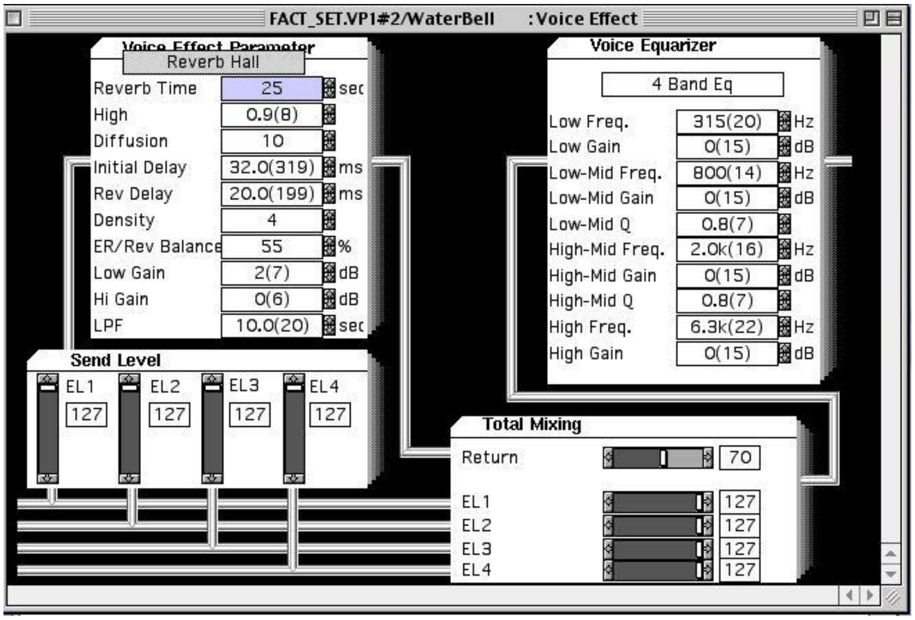
The **Voice Menu** opens the Element Edit window and allows you to turn On/Off the four different Elements in a patch. Most patches only use one Element. You get two waveform layers in each Element so using just one is not restrictive in any way. Complex layering of sounds is a big part of the VP1 patches.

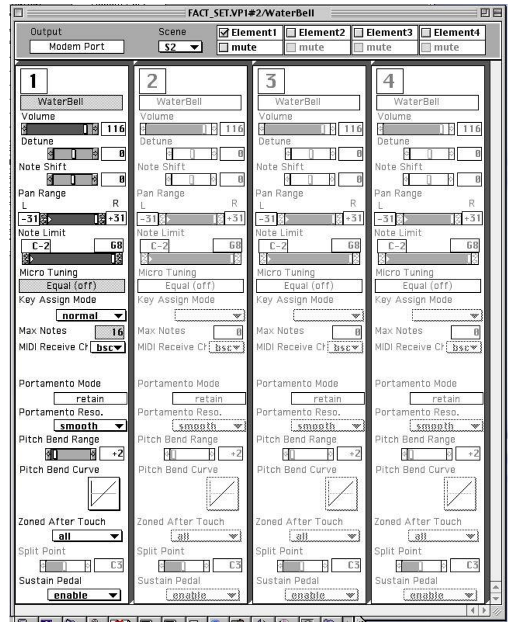


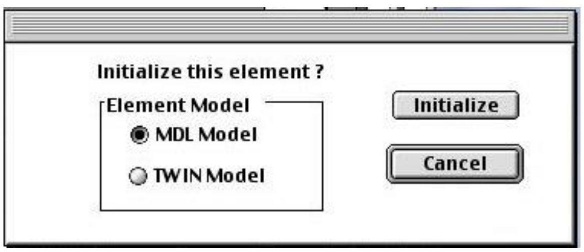
The Voice Menu includes a link to the Voice Effect window (shown below) and it also adds three items at the bottom when you create a New patch library. The Voice Effect menu item is the ***only*** link to the overall Voice Effects for the Editor. The Voice Effect pictured below is applied to all four Elements of a patch, feeds into a mixer, and is paired with an output Equalizer. There are also individual Element level Effects, giving you up to a maximum of three different Effects acting on each Element. Each Effect has a large set of parameters which can be used to alter the sound. Most people forget you can alter the Effect settings when you set up a patch on any synthesizer. A list of all of the selectable Effects is included in the Effects section of this guide, towards the back.

In the middle of the menu you can select how the Elements window displays the Element parameters. Horizontal is the default, since most voices only use one Element. For more complex patches, you can select Vertical and see all four Elements side by side. We'll go over all of the Element settings later on in the guide.

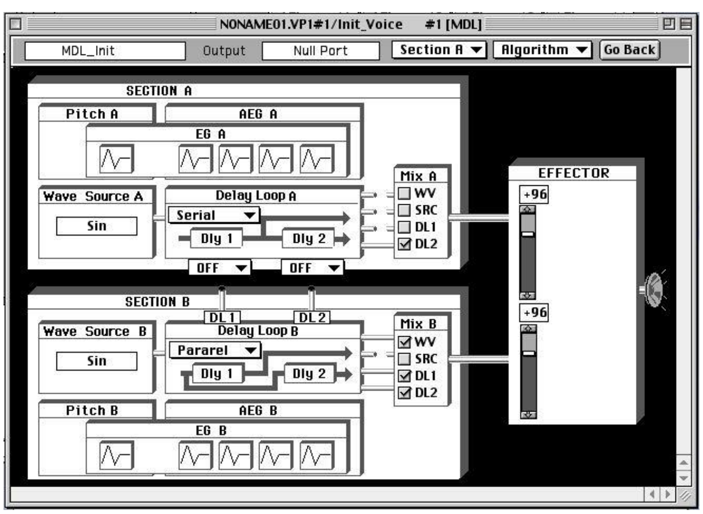
The VP1 console has a QuickEdit section in the middle with four sliders which allow you to alter system level parameters while you play the synth. One item on the second row of settings is EF DEPTH, which is the depth of the system level Effect you've selected. It's the ***only*** Effect setting you can access in real time. See pages 3-16, 3-18 in the manual. There is also a way to alter this setting with MIDI commands using the Switch settings, if you're creative.

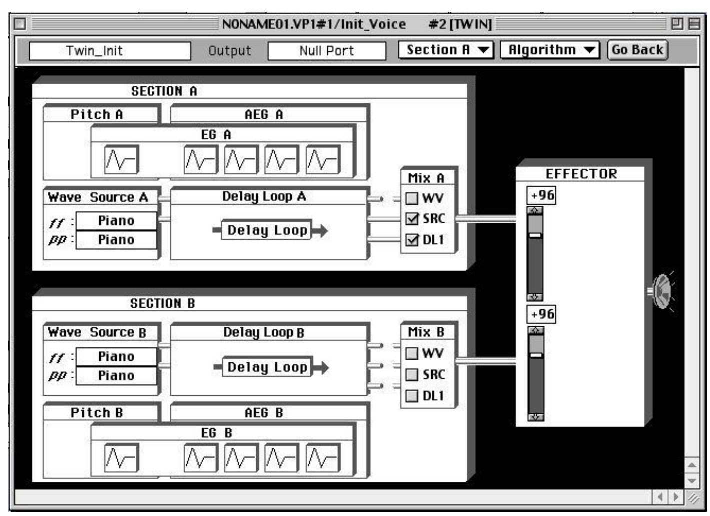






You can Initialize an Element using one of two different models, the MDL Model and the Twin Model. TWIN targets plucked or struck instruments, MDL includes sustained instrument sounds. MDL is the complex one, with more controls in the Resonator section. The MDL Model allows you to use one Driver with dual Delays, with cross connects between Sections A and B of the patch. The Twin Model is a bit simpler, it uses two blended wave Drivers per Element, one Delay, and no cross connects. You can see the block diagrams for each below. MDL is on the left, TWIN is on the right.

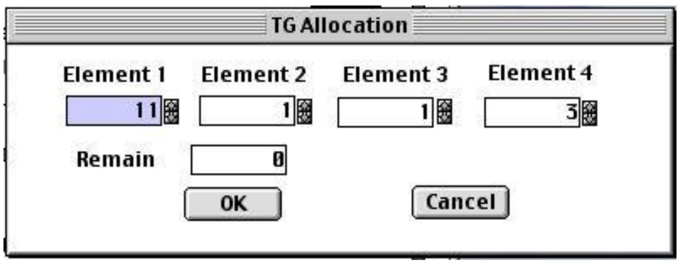


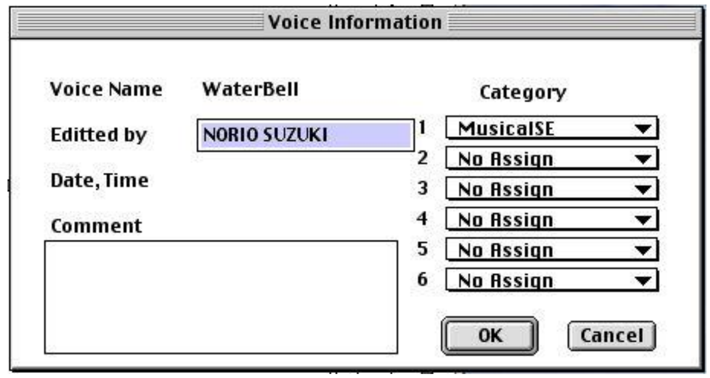


MDL Model

TWIN Model

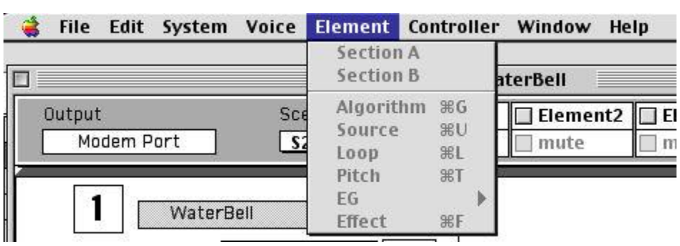
TG Allocation (Tone Generator, see below) on the Voice Menu allows you to select how many internal voice cards are allocated to each Element. There are 16 cards, so you can assign up to 16 notes to one voice Element. When using two or more Elements, you have to split the 16 cards between them. You can assign different midi channels to each Element, so you might assign 12 cards to one and 4 to the other. Controlled from a DAW or sequencer, you may need more notes for one Element than the other. In a sense the VP1 is 4-Part multi-timbral, but only if you use a DAW (Digital Audio Workstation) and only if you set up a patch with the four Elements you want to use first. This is currently difficult to do. The demo songs use this feature extensively. The drum sounds, harmony, and melody are all played simultaneously on the VP1. That fact implies there is a very crude sequencer in the firmware and ten songs stored in there, but you have no access to any of it. All you can do is play back the demo songs. Manny has made these available on YouTube if you'd like to hear them.



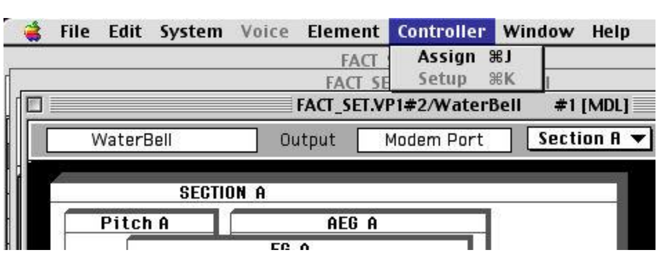


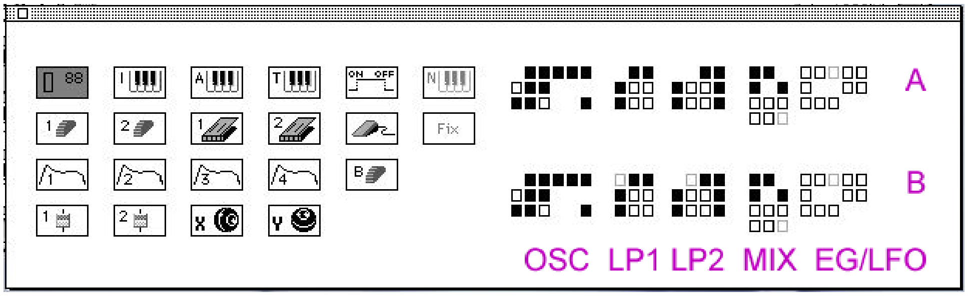
Voice Info is the next item on the Voice Menu. Interestingly, the name of the patch programmer is listed here. You can assign the patch to up to six instrument categories for sorting, and even add in a Comment. A number of the patches I checked do have comments, mostly instructing how to use the controllers on that patch. Part of the Comment shows up on the VP1's display when you load the patch.

The **Element Menu** gives you access to each of the Editor parameter windows detailed in this guide. These panes are mainly selected by double clicking on locations in other windows, like the Element window or Algorithm window. We'll go over each of these menu sections in separate chapters of this guide. The top two items allow you to switch the editor window between part A and part B of the patch for each of the menu sections listed below them. It would be really useful if there was a more convenient On/Off switch for each part, I need it all the time while I'm editing. The Algorithm window is really the only easy place to see what's on and what's turned off.



The **Controller Menu** allows you to see the assignments for all of the different physical controllers to parameters in each of the parameter windows.





MDL

The first item in the Controller window is highlighted (the greyed square with the left rectangular gauge and the number 88 on the right). It assigns a single value to a parameter and is the default controller. All of the black squares on the right correspond to this controller being selected for a large number of parameters in the synth. The top set are Section A, the lower set are for Section B of the patch. Layering sounds and being able to morph smoothly between the three scenes is a big part of the VP1.

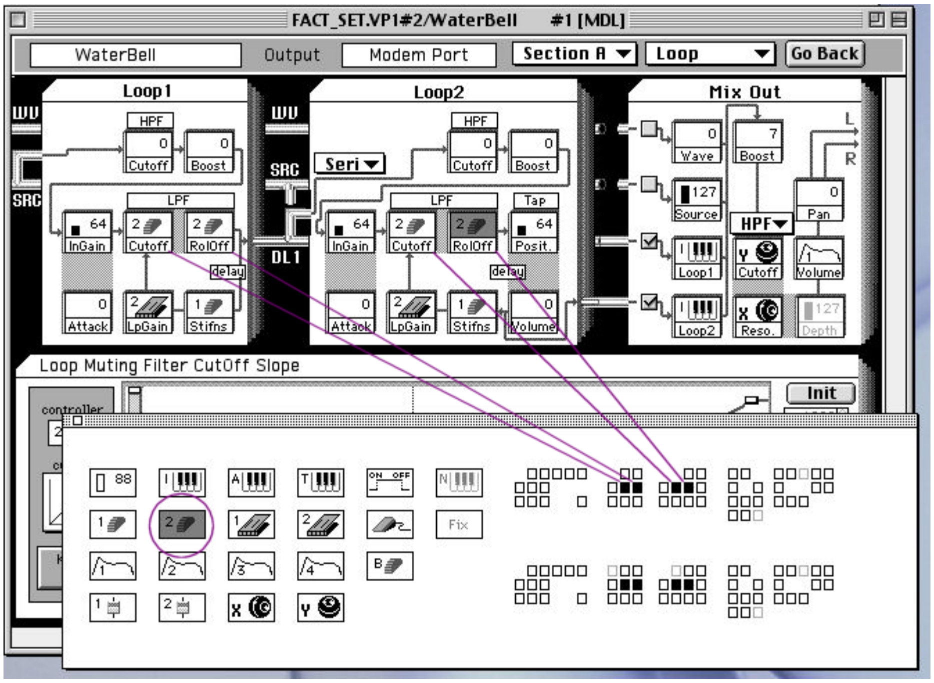
The little squares on the right correspond to the arrangement of parameters in the different Editing windows. The first grouping matches the arrangement of the OSC/Noise blocks in the Source Window. The second group matches Loop1 and the third matches Loop2 in the Loop window. The fourth group matches the Mixer, and the fifth group matches up to the Envelope windows. For the Envelope Generator windows, the first column of squares in the fifth group is the A-PEG block, the second column is the A-EG/CC1 block, and so on. The three lowest squares in group five match the Pitch controls (that took some trial and error to figure out!). None of this is explained in the original Yamaha editor guide.

You can see there are five different keyboard controllers in the first row of icons after the default setting control; ‘I’ is the Initial Velocity which uses a graph for setup, ‘A’ is Aftertouch, ‘T’ is Touch EG and ‘N’ is Note based pitch control. You can see an ON/OFF button icon in row one, which is a Key On/Off Switch. Touch EG is a combination of Velocity and Channel Aftertouch, an Attack-Decay shaped controller envelope. The ‘N’ Note based controller is only used in the Pitch window.

You click on one of these controller icons, then click in an editing window on an item in the block diagrams to assign the controller to a parameter. Controllers are greyed out when they can’t be assigned to any parameters in the visible window, and the little squares grey out when the parameter isn’t accessible because of radio-button selections. Row two includes the two Mod Wheels, two Foot Pedals, a Breath Controller, and a ‘Fix’ EG control icon. Fix is only available in the Envelope Generator windows, it bypasses the envelope and uses a fixed value. Row three includes the four Envelope Generators and the Pitch Bend Wheel. Not shown are the CC’s (Continuous Controllers) and LFO’s that go with the Envelope Generators.

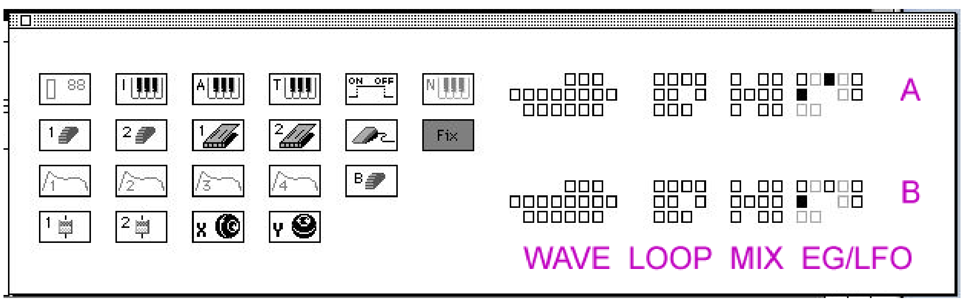
Sliders one and two are on the next row, along with the Track Ball X and Y controllers. If you count them, there are 21 icons, for 20 different physical controllers and one fixed value option. There are also two Foot Switches and the Scene Control Slider which are not displayed here since they can’t be reassigned to a different function, and the four CC’s and LFO’s are not shown either. There are also 16 Quick Edit parameters on the synth, accessible when you play live. That adds up to a LOT of controllers.

I haven’t seen anywhere near this level of controller flexibility or the sheer number of controllers on any other synth, except for one other synthesizer: the Oberheim Matrix 12. The name spells out it’s specialty, it’s controller matrix. It’s a modular Analog synth without all of the patch chords, everything is patched internally. I’m working on finding a Matrix 12 to fill in the gap in my synthesizers, I hope I’m successful. It and the VP1 share a lot of capabilities.



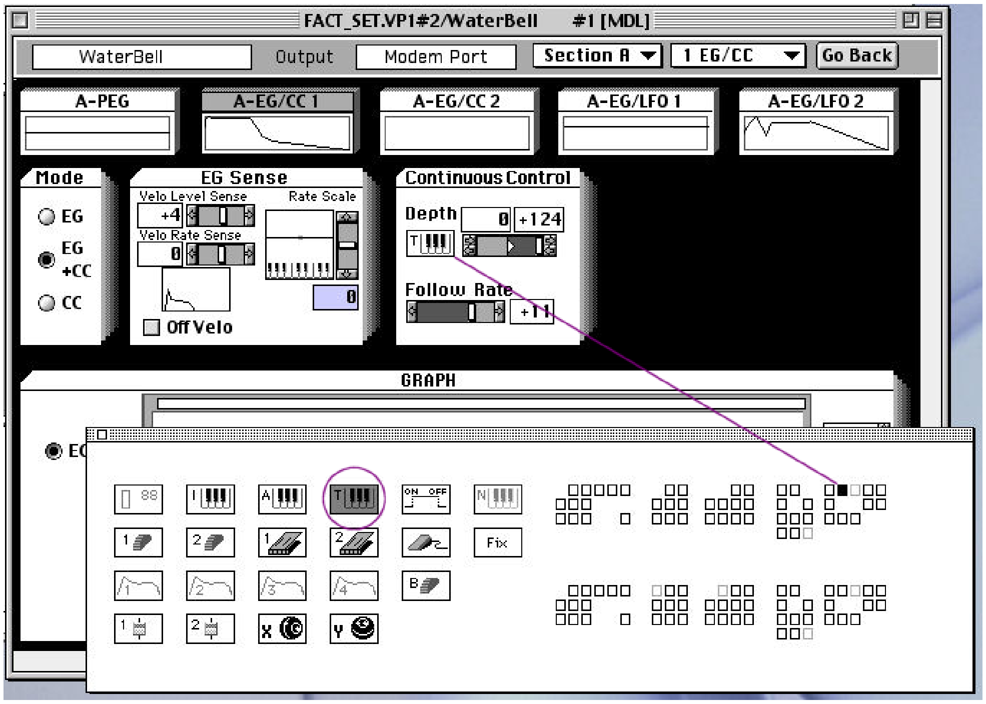
MDL

In the example above, the second Mod Wheel is selected (and circled). If you look in the Loop1 and loop2 blocks, you can see where the four icons are located for the #2 Mod Wheel for Cutoff and RollOff settings. Now look at the little squares representation and can you see how the four little black squares correspond to the locations of the #2 Mod Wheel icons? The squares are arranged in the same pattern as the controllable parameters are in the window. It doesn't work quite this way for the EG's, but it's close. It's actually a pretty compact way to see all of your controller mappings at once. The groupings also change shape when you select TWIN rather than MDL for the patch type. See the TWIN model below.

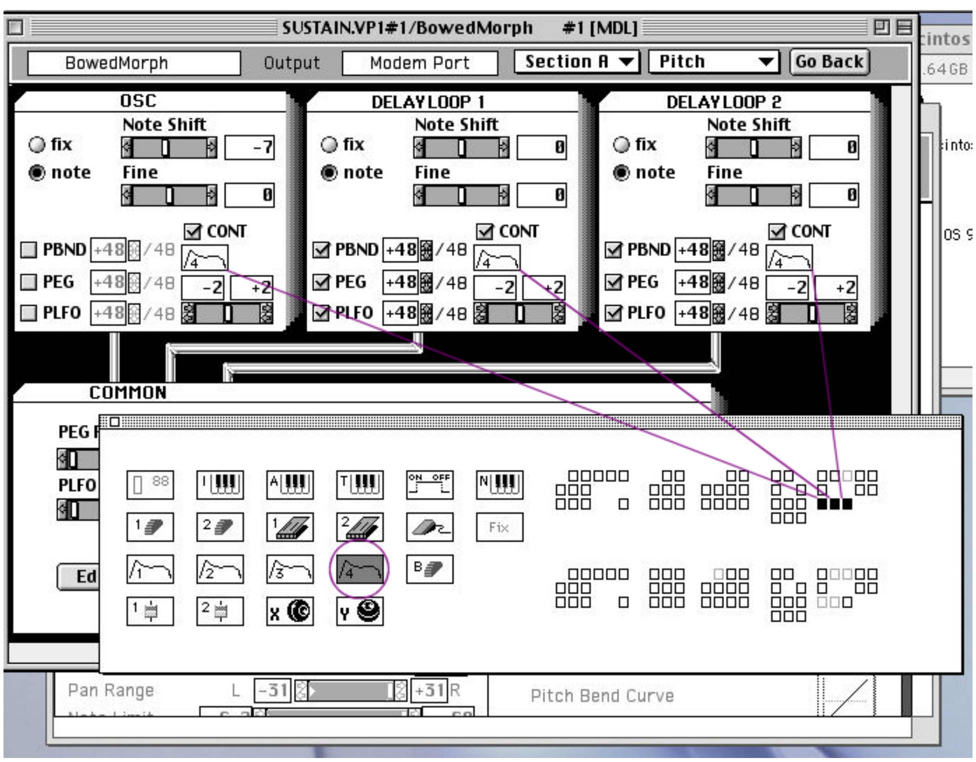


TWIN

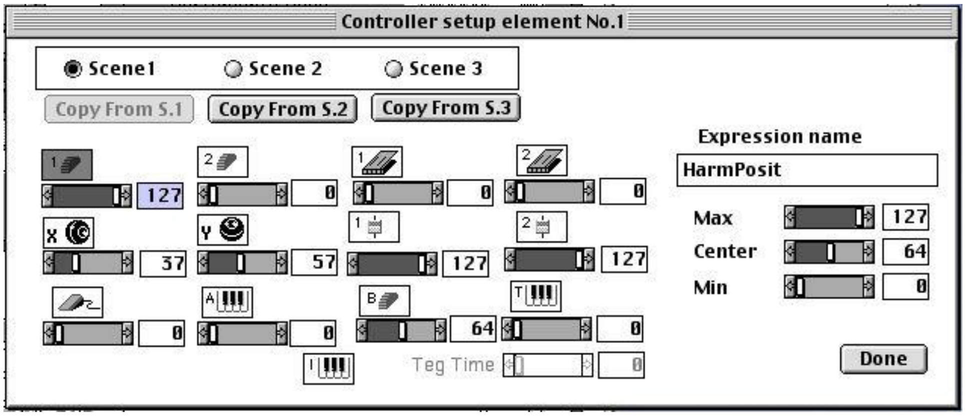
Below is an example of an Envelope Generator window. The black square for the circled Touch EG controller corresponds to the Depth setting for the Continuous Controller block. See the little keyboard icon with the letter 'T' in the box under the Depth label? The A-EG/CC1 window corresponds to the second column of small squares in the Controller pop-up window. Also notice that the "I" Velocity controller is pre-assigned to the Envelope Generators in the EG Sense block for Level and Rate Sensitivity.



The Pitch window below also has three assignable controllers in MDL mode, shown to the right above (which drops to two in TWIN). In the lower half of the window, Envelope #4 has been assigned to all three pitch controllers. Can you see the square with an ADSR envelope in it with the number 4 beneath the CONT checkboxes? In the Pitch OSC, Delay Loop 1, and Delay Loop 2 windows, EG-4 is used as the CC (Continuous Controller). These are also indicated in the little squares by the three black ones highlighted on the right. They are a part of the Envelope Generator grouping. It took me a while to figure out what controllers these were connected to. The Note radio button selects the type of pitch control for a Pitch item, the Fix radio button works on the Envelopes as a bypass, turning on the normally greyed-out “N” Note and “Fix” controller icons when this window is selected.



Getting back to the menus, the second item on the Controller menu is **Setup**, for the three Scenes (below). It allows you to set the preset value and output value ranges for all of the physical controllers. Each of the three Scenes can be set up differently here. A Scene is a specific patch setup, where the controllers are positioned to give you a specific type of sound out of the patch. There are so many controller settings combinations it would be impossible to hit a specific setup manually, especially during a live performance. The Scene controls allow you to select one of three different Scenes, and to blend between them smoothly and easily using the Scene Slider or a Foot Pedal.



The two Foot Switches (not listed) unfortunately can't be assigned to control anything different, they're limited to just Sustain and Portamento functions. 'Note' and 'Fix' controllers don't require any settings either and are not listed here.

The manual Default control values are set on the page where the controller is applied. We'll go over that later. None of those items show up here, since they aren't connected to a physical controller.

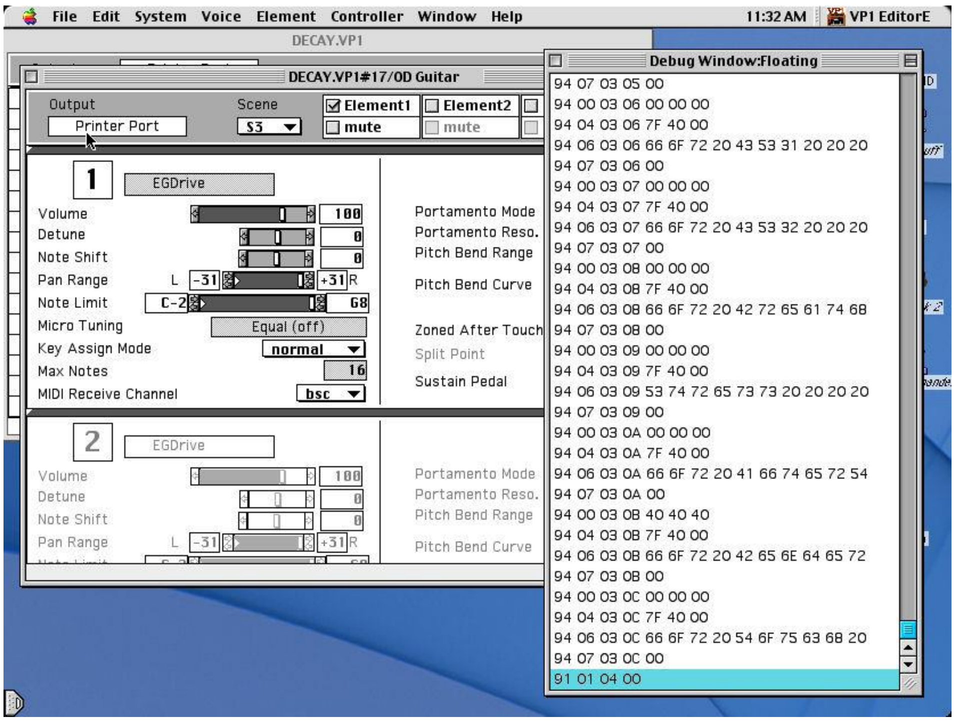
Several of the standard MIDI controllers are pre-assigned, and more information is available in the VP1 manual in tabular form on page 3-10, and in the Reference section page 1-6 and 2-10. Expression (11) one of the main midi standard controllers, is completely absent which baffles me. I'm using Breath Control as an alternative to Expression in my sequencer.

Standard MIDI Controllers:

1	Modulation	MW1
2	Breath Control	BC
7	Volume	VOL
11	Expression	
64	Sustain	SUST SW
65	Portamento On/Off	PORT SW

One more comment on Midi, you can run a midi song into the VP1 while the Editor is running. It even ignores channel change messages and keeps all of the settings intact when you do. Those can be annoying normally outside the Editor on the VP1, midi songs have a bad habit of changing the patch on you. And yes, the Sustain Pedal works while you're editing. I tested it to make sure.

The **Window Menu** contains three items, Show/Hide Debug Window, Clear Debug Window, and Select Window. The Debug Window displays the commands sent to the VP1 through the programming Interface. This could be used to determine all of the hex commands used to set up the synth, allowing you to program a new VP1 Editor for a different operating system like Windows 10 or OSX. For now, we just have the Mac OS 7-9 editor. The hex commands are somewhat similar to Midi sysex commands, but have their own rules and format. From observation, I think the commands are delimited by a carriage return or line feed, since the command strings are all different lengths and have no standard prefix or terminators like Midi sysex uses. I have a PowerMac G3 set up to run the editor, so I don't feel any pressure to program a new one.



There is one command you can't get with this window, the one that tells the VP1 to enter MIGS editing mode. That command you would have to intercept by putting a serial port sniffer on the RS-232 line.

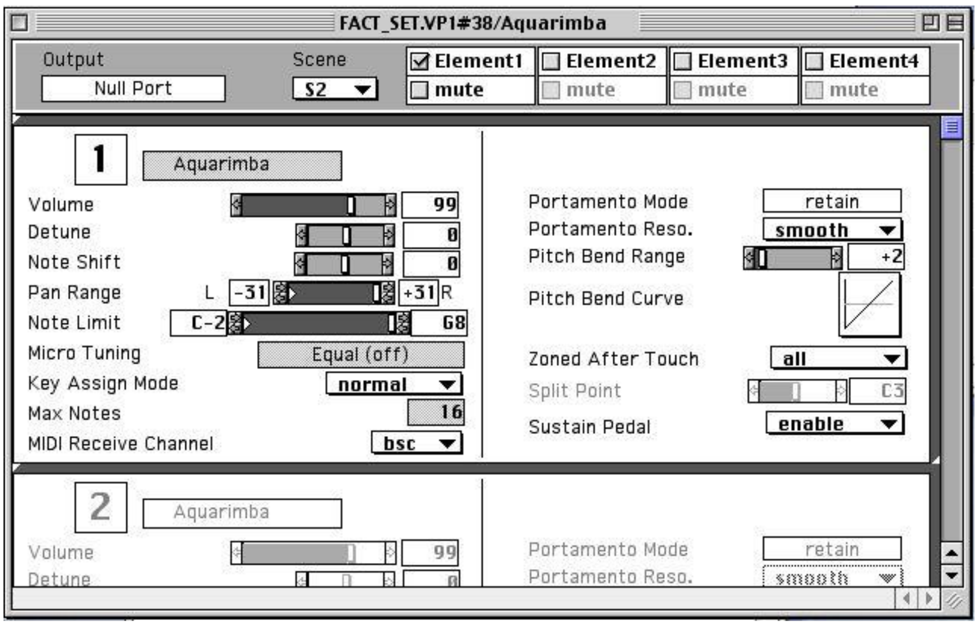
There is also a **Help Menu**, but it's only for the Mac OS, not for the VP1 editor.

Elements Window

Double clicking on one of the patch names in the patch bank window (way up on page 7) brings up the Element window. There are up to four Elements per patch, but you sacrifice polyphony when you use more than one. A one Element patch allows for sixteen notes, two Elements drops you to eight (if distributed evenly) three drops you to five, and four notes is the bottom level. You can control how note Elements are routed to the 16 internal processing cards in the synth using the TG Allocate menu item.

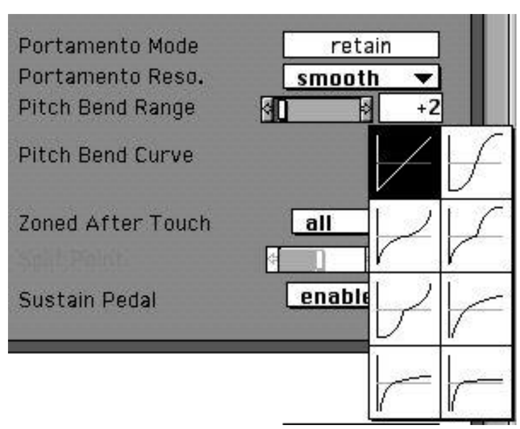
At the top of the window you select which Elements are active in the patch, which ones are muted for editing, and which Scene you want active. Most of the individual Element settings are straight forward. Notice there is a MIDI Receive Channel for each Element. The VP1 can be set up to be multi timbral, providing up to four different instrument timbres at a time, each on a different channel. You can access this from an external midi sequencer after setting it up.

I believe you can also copy and paste from one Element to another using this window using the menu selections at the top of the screen (not shown).



Left Section

Volume, Detune, Note Shift	Single Value
Pan Range, Note Limit	Two Values
Micro Tuning:	There are a number of tuning scales to select from.
Equal (off)	Equal Temperament
Pure Major C, C#,D…B	All Keys
Pure Minor (same)	All Keys
Mean Tone (same)	All Keys
Pythagorean	
Werckmeister	
Kirnberger	
Valotti & Young	
1/4 Shifter Equal	



1/4 Tone
1/8 Tone
Psyche
Reverse

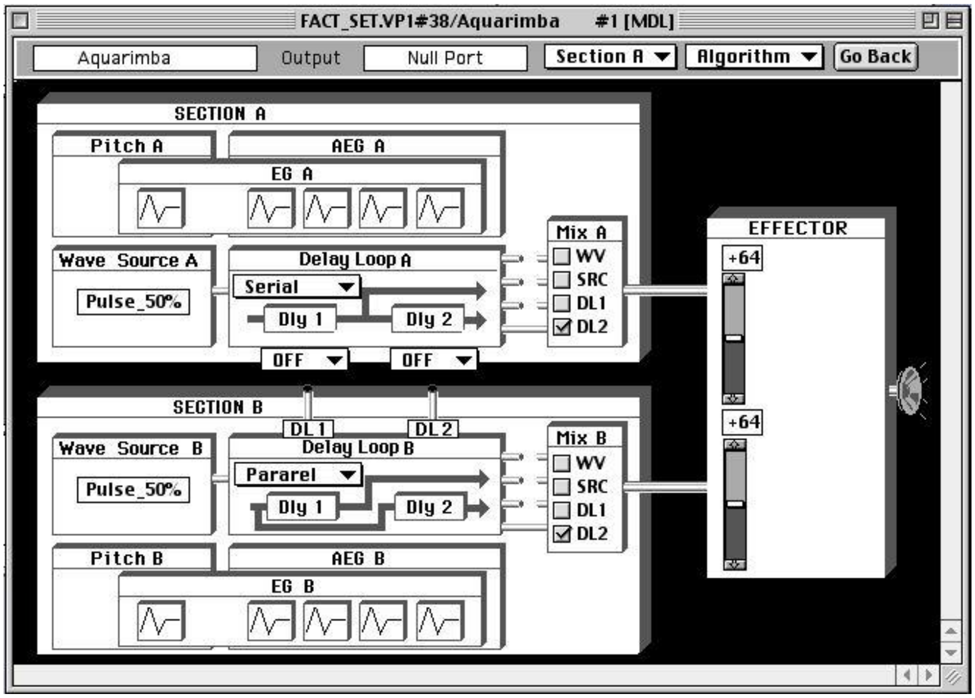
Key Assign Mode:	Normal, lo on, hi on, Yamaha, lo rem, hi rem	
Max Notes	number display	
Midi Receive Channel	bsc, 1 – 16	(you can designate a Basic Receive Channel, or assign a separate midi channel to each Element = 4 Timbres!)

Right Section

Portamento Mode	if Max Notes =1: Fingered or Fulltime, if >1: retain
Portamento Reso.	Smooth, 50 cent, 100 cent, 200 cent, 400 cent (smooth or stepped)
Pitch Bend Range	single value
Pitch Bend Curve	8 choices of graphs (see them above)
Zoned After Touch	All, Top, Bottom, Mid, spl_hi, spl_lo (split)
Split Point	Note
Sustain Pedal	Enable, Disable

Algorithm

Clicking on the Element number 1 (1 of 4) in the Elements window brings up the Algorithm window which displays the overall signal routing. You can see in the MDL Algorithm window below that there are two Sections to each patch, A & B, each with it's own Waveform and Delay Loops.

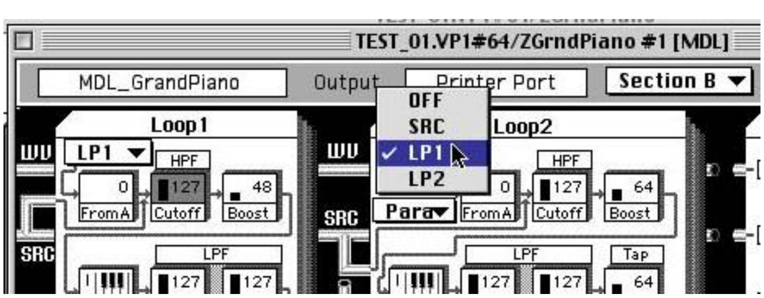


MDL

Most of the items in the window take you to separate settings windows if you double-click, there are only a few actual settings accessible from this window. Use this window to copy and paste different sections of the patch around, such as copying Section A to Section B. Click on the Section A top bar, press Copy in the menu, then click on Section B and press Paste. This works for all of the other subsections of the patch as well.

The Wave Source lists the names of the waves used in sections A & B of the patch. In the Delay Loop block you can set up the two delay loops to be in Serial or Parallel (Pararel! Grin) as shown above. Delay Loop A is in Serial mode, and Delay Loop B is in Parallel mode in the example above. The Mix A & B checkboxes are duplicates of the ones in the Mixer, which we'll get to later on. These turn on and off the different inputs into the Mixer. Last, the Section A & B mixing is done here using the two Effector volumes. These controls are not duplicated anywhere else, so remember to come back here to balance the volumes.

Selecting the pop-down box just below the Dly 1 or Dly 2 label (OFF/ON) for Section A opens the cross-routing connection into Section B, which also show up in the Section B Loop Window. You can select Section A: OFF, SRC, LP1, or LP2 to route into either Delay Loop in Section B, see below. A volume control is also provided when you cross-route.

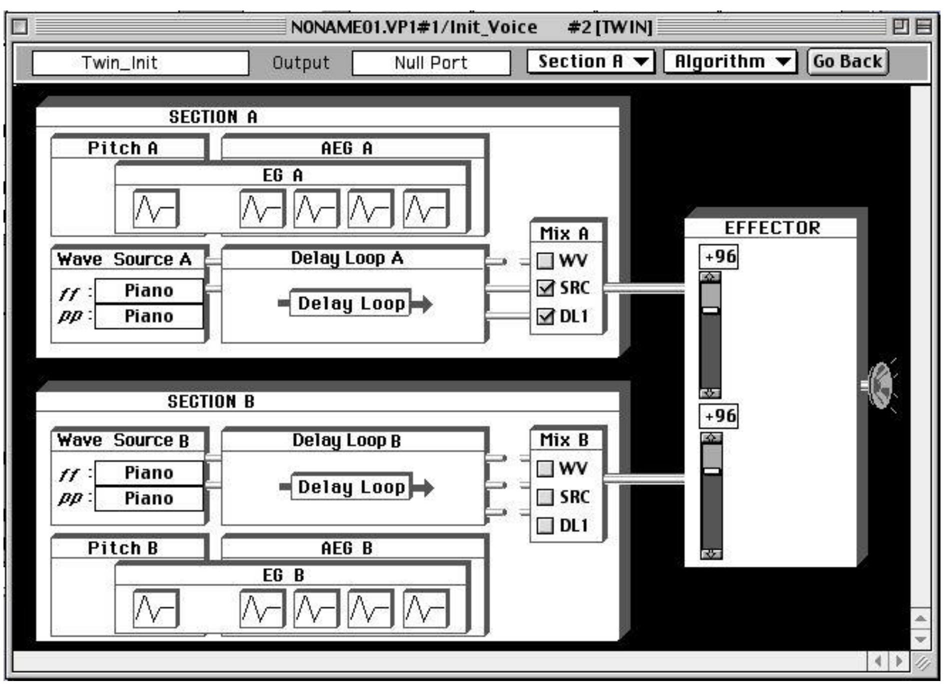


MDL is shown above. TWIN (below) is slightly different, with two Wave Sources and only one Delay Loop per section. According to the original editor manual, TWIN is suitable for sounds which are used with "feedback" performance techniques, and MDL is suitable for string and pad sounds. Looking into the modes closer, TWIN only does plucked or struck instruments, MDL adds sustained instrument sounds. For some reason, our TWIN example below uses the same Piano wave for both. The TWIN model includes a section in the Wave Source to create guitar pick sounds, mallet strikes and to hit the first part of the notes like a piano which the MDL model doesn't have.

The next step up in instrument sound quality comes from layering. The VP1 includes two physical models in each Voice Element, a Part A and a Part B, which are blended to produce a very complex instrument sound. On top of this, the VP1 also allows you to layer FOUR Elements in one patch, running up to eight physical models simultaneously per note. This chews up polyphony though, reducing the output to just 4 notes at a time. The majority of the VP1 patches use only one Element, using just two layered physical models per note, and providing the player with 16 notes of polyphony.

The VP1 allows you to map the controllers to just about any parameter in the acoustic models. The Oberheim Matrix 12 did this too, it was the main feature of the 1985 synth, hence the word 'Matrix' in the name. To create dramatic timbre shifting patches, use different Wave-forms for Sections A & B, then set up a controller to allow you to shift the timbre manually. Another angle is to set up the keyscaling to allow one waveform to dominate in one part of the keyboard, and the other waveform to dominate over the rest of the keyboard. This is how some of the factory patches create a patch with one instrument down low and an entirely different instrument up higher on the keyboard. Yet another technique is to use the Inharmonicity parameter to alter different sections of the keyboard, changing the timbre in and out of metallic tones as you play up the keyboard. Messing with the Filters is the standard method on most synths. To really create something wild, double up on controllers, use an envelope to control the shape and frequency of an LFO!

As a note, the Mix A/Mix B checkboxes are the same ones you see in the Mixer block in the Source and Delay Loop Windows. Also notice there are no cross-connects between Section A and Section B like MDL provides.



TWIN

Section A

Delay Loop A	Serial, Parallel	(in MDL mode)
Dly 1: OFF, SRC, LP1, LP2	routing to Section B	(in MDL mode)
Dly 2: OFF, SRC, LP1, LP2	routing to Section B	(in MDL mode)
Mix A: WV, SRC, DL1, DL2	routing On, Off to the EFFECTOR (Effects Module)	

Section B

Delay Loop B (in MDL)	Serial, Parallel
Mix B: WV, SRC, DL1, DL2	routing On, Off to the EFFECTOR

Effector (the Effects module)

Output Volume Levels	Balance the relative volumes of Section A and B here. It would have been nice to have a controller assignment here, but there isn't.

Internal Physical Models

Let's take a look at acoustic instrument physical models for a minute. The first major model was the **Karplus-Strong** algorithm, for a plucked guitar string. It modeled a triangular pulse moving up and down the string, bouncing off the frets and bridge of the instrument, using one delay loop. The delay loop had a filter and gain reducer in it to lower the amplitude of the wave as it passes through each time, and to reduce the higher frequency harmonics faster than the lower ones. The VP1 TWIN algorithm covers this model. The original model sometimes used a noise pulse to trigger the sound. The VP1 provides this option along with a large number of recorded instrument Drivers, including Analog waveforms like Sine, Sawtooth, Triangle, and Pulse waves.

Later, the **Wave Guide** model was developed at Stanford University. It adds a second traveling wave in the string from the plucked point, one moves up towards the frets, the other travels down to the bridge and back. One delay loop is short, the other is long. The VP1 MDL algorithm models this method. This method is particularly useful in modeling wind instruments like the Saxophone as well as stringed instruments and is at the heart of the Yamaha VL1.

To further improve the quality of the sound, a third model was developed, the **Commuted Physical** Model. Researchers inserted a wave shaper after the two delay loops, to mimic the timbre changes from the body of the instrument, the resonant body. To reduce the number of calculations, they decided they could skip the wave shaper and inject a digitized waveform into the model at the beginning, one which already had the timbre of the instrument in it. This technique dramatically improved the quality of the instrument sound without significantly increasing computations. The Commuted model Drivers are included in both the TWIN and MDL algorithms in the VP1. There is a problem with this concept. If you use a waveform recorded at a specific pitch, then you shift the pitch to match the key pressed, you shift the resonating body filter frequencies. The VP1 uses a multi-pitch impulse waveform which reduces this problem.

On a related topic, this is just an observation, but if you use the Cycle mode and bypass the Delay Loops, the components that remain make up a complete Polyphonic **Virtual Analog Synthesizer**… The only item missing from an ARP 2600 is the step sequencer in the ARP keyboard. Everything is also digitally routed on the VP1, you don't need patch chords like you do on an Analog modular synth. A patch is contained in one settings file, it's not a mess of cables like you see in some Analog YouTube videos.

You also have a **Wave Table Synth** in the VP1 as well, with the 75 waveforms injected by the Oscillator section. Selecting the Cycle or Free OSC modes allows the waveform to repeat, giving you a wave table waveform setup if you bypass the Delay Loops. These take on the characteristics of a Sample when ran through the virtual string, i.e. adding the Delay Loops gives you more of a **Sample-Based Synth**.

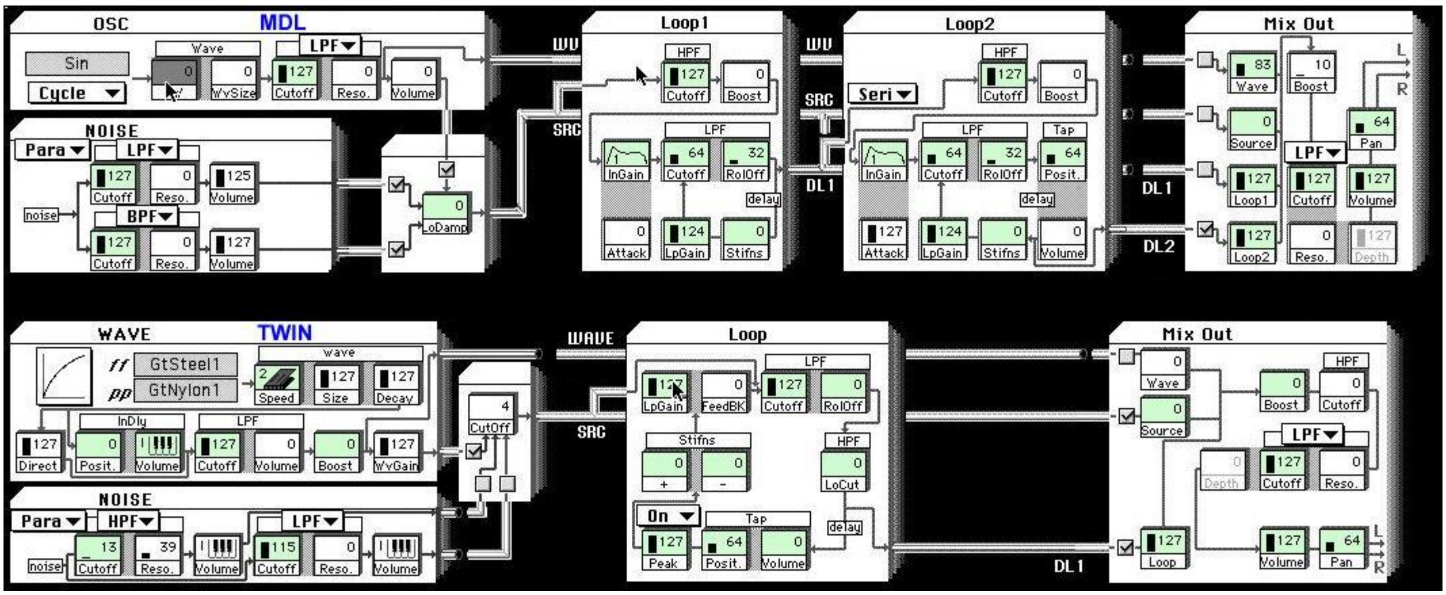
With all of the built in filters, subtractive synthesis slips right into place. Looking deeper into the filters that abound in the VP1, you can also create **Formants** as well. A Formant is just a narrow, tunable, bandpass filter to pass your waveforms through. The Element Equalizer in the Effects module has five tunable bands that can be used to create FIVE parallel Formant filters! Not as many as the eight available in the FS1R, but definitely on par.

So there you have it, a synthesizer with almost all of the available physical acoustic synth models built into it, all in one package. If only it had made it past the design phase and into production! But almost no one knows what might have been.

Now comes the fun part, programming it!

Overview

Before we go into specifics, here is a complete overview of both models. Yamaha broke the block diagrams up to fit more neatly into the settings windows in the Editor, I put them back together here. The two are very different, I've tried to get a patch to work on both internal models using the same settings, but it just hasn't worked out very well. Sometimes it works, sometimes it doesn't.



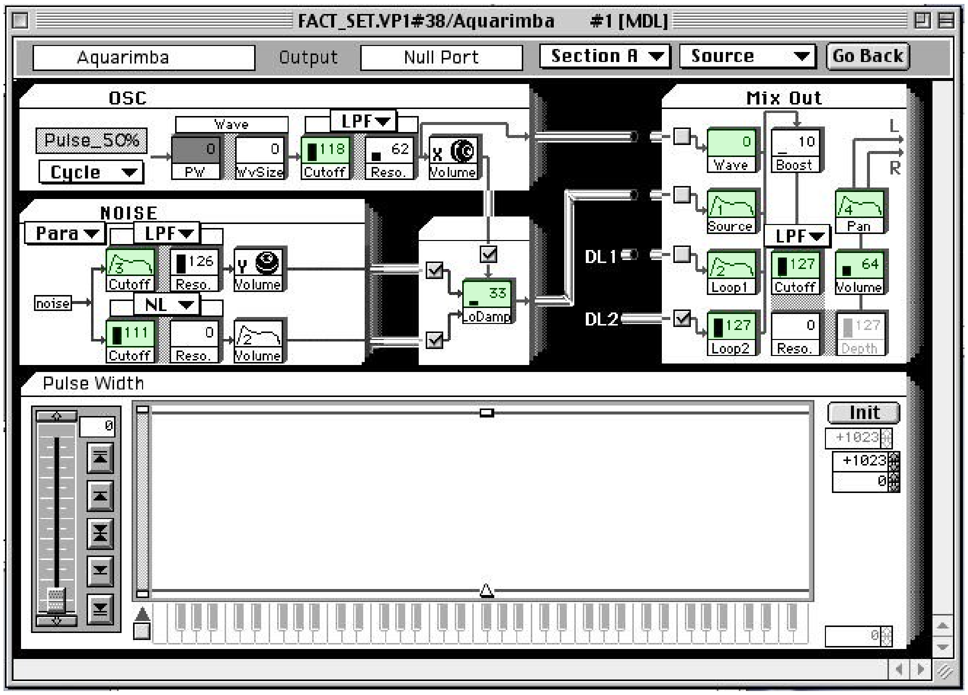
If you need to compare the two algorithms, this is the diagram to use. Notice how there are more keyscalable parameters (green shaded) in the TWIN Oscillator as compared to the MDL Oscillator. And notice that the MDL model includes two Delay Loops where the TWIN model only has one. Each model has it's advantages.

The Noise blocks and Mixer blocks are nearly identical, though they're arranged slightly differently.

It has been proposed in one of the forums that MDL stands for Multiple Delay Loops. That sounds right.
TWIN simply refers to the twin Driver waveforms in the Oscillator section.

Source ____

This section of the synth controls the settings for the Driver Oscillators (one window for Section A and another for B) along with mixing a Noise source into each Section. Filter and controller settings are in here, along with the Mixer. Do you see the mini track balls in the Volume blocks? These icons indicate which controller is mapped to which parameter, the trackball is selected here.



MDL

The Green blocks are all of the Keyscaled parameters, the remaining blocks are single-value parameters (one value over the width of the keyboard). It isn't really explained anywhere, but the Source input line to the Mixer also goes into the Delay Loops as well. It's shown in my Overview diagrams, but not anywhere else.

When you select a block in the diagram, it alters the Graph window below to show you the settings for that controller. Wave Pulse Width (PW) is selected above. The default value controller provides you with a maximum and minimum value on the graph and a slider on the left to select your setting with.

The PW parameter above only has one value you can set across the keyboard. A value of '0' is currently selected as the default value since no Controller is assigned to this parameter. 0 is duller, 127 is brighter. You can set the default value as a proportional value between the minimum and maximum lines to whatever you want using the left slider. In the graph you grab the handles in the center of the lines to drag them up and down on the graph to change the control limits, and you move the control slider on the left upwards to select a default value. The triangle is the minimum grab handle (0) the rectangle is the maximum value grab handle (127). Double-clicking on the graph adds a new breakpoint to the plot, and you delete breakpoints by double-clicking on the handles. This parameter is not shaded green (it's not Keyscaled) and doesn't have that option.

The ADSR curve icons show you which parameters are controlled by Envelope Generators, see the MixOut Source block for an example. Controllers include all of the physical pedals and sliders, envelopes, and LFO's. ANY of the blocks in the model can have a controller assigned… *Think about that for a minute!*

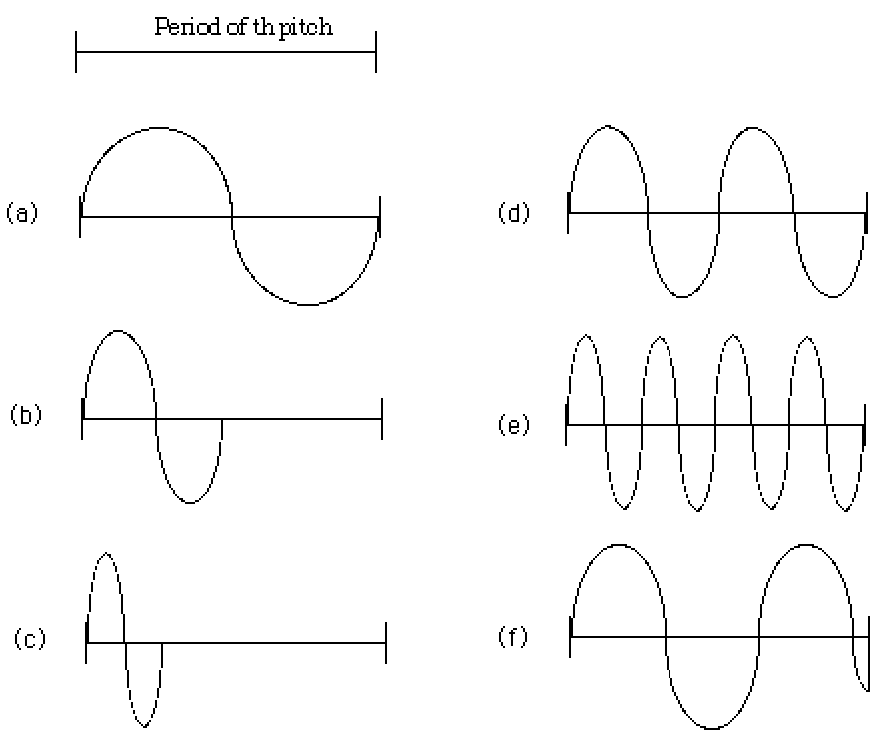
MDL MODE

The MDL Oscillator has three mode settings, 1-Shot, Cycle, and Free. '1-Shot' uses just four cycles of the waveform. 1-Shot creates a plucked or struck note where Cycle and Free are continuously bowed or blown Drivers. 'Cycle' repeats the waveform continuously from the beginning to the end. 'Free' is similar to 'Cycle', but it repeats the waveform from a completely random point (altering the phase) on each keypress. I can't hear any difference between Cycle and Free.

Let's talk about the nature of the Drivers for a second. Played raw in 1-Shot mode, they do NOT sound like someone plucking or playing the instrument they're named after. The Violin sounds more like a person thunking on the body of the instrument with a bit of note mixed in. The Trombone sounds like a person brapping into just the mouthpiece. The Flute sounds like a strong air pulse, like a pronounced 'P' injected into the pipe embouchure hole. The Drivers contain the acoustic response of the instrument body (Commuted Model) as well as the instrument string pluck or mouthpiece sound. It's an input energy pulse. You have to run them into the virtual strings in the model through the Delay Loops in order to get the instrument to resonate under 1-Shot. They also resonate in Cycle or Free, but the harmonic resonance is much simpler this way, as compared to adding in the Delays. Cycle and Free modes sound like a synthetic sustained note on the instrument in question, but without any of the timbre changes we're used to as the note continues.

Adding the Delay Loops (the virtual string without any controllers) it can sound very acoustic in nature, like a sample-based synth with a lot of harmonic depth. The Delay Loops also need to be set up properly in order to match the internals of the instrument you're attempting to model. Therein lies the challenge I'm still working on. I'm learning from the VL1 patches how to set this up for wind instruments. And, adding in physical controllers on top of this takes it to an all new level of variability and expression.

The Wave PW control allows you to turn the sound wave into a variable-width pulse wave, as shown in (a), (b) and (c). 0 is duller since it's just one complete cycle (a), 127 is brighter as a pulse wave (c). The highest value (127) narrows the pulse to ¼ of the total width as in (c) below. You could also assign an envelope controller (or any other controller) to Pulse Width. It's also not limited to just a single value even though we're calling it a single-value parameter, it's just not keyscaled.

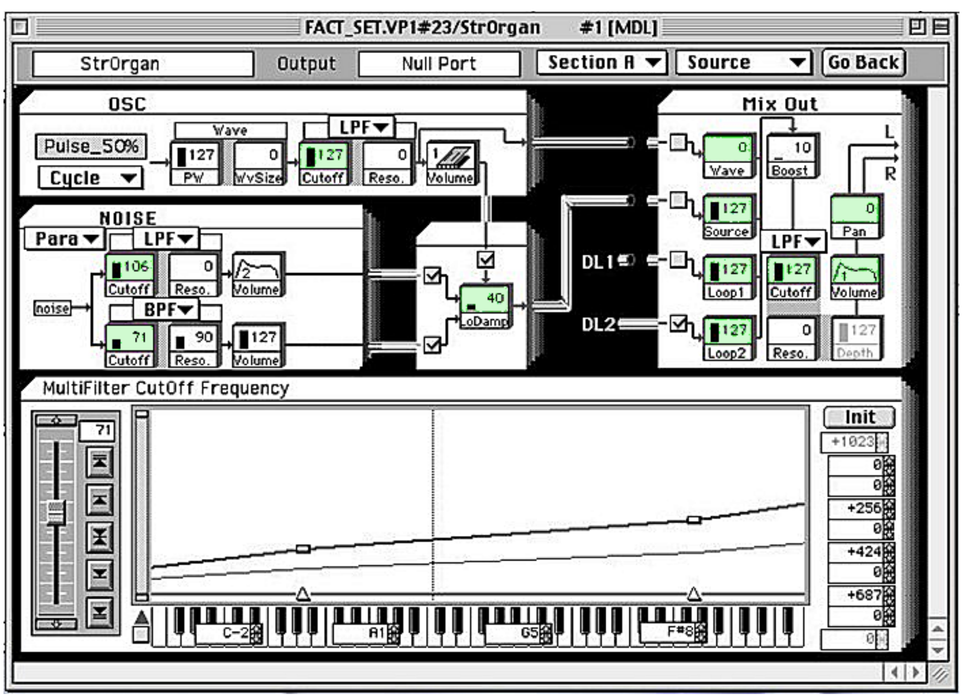


WvSize specifies how many cycles (1 to 4) will be used in one period as shown in (a), (d), (e) and (f). 0 is duller at one cycle (a), 127 is brighter with four cycles (e). Figure (f) is about 1.6 cycles, a setting of about 25/127. To get one complete cycle use 0, two cycles is 42, three cycles is 85, four is 127. Using both PW and WvSize maximized at the same time, you would get all four cycles smashed into just one-fourth of the period, four pulses in the place of the one shown in figure (c). You can also get cut-off fractional waveforms as in (f), or create a cut-off pulse wave more like a combination of (c) and (f). Both PW and WvSize increase the pitch of the pulse sent into the resonator Delay Loops, adding in more higher frequency components. The actual Pitch however, is determined by the length of the Delay Loops (see the Pitch Window) NOT by PW/WvSize (MDL) or Speed/Size (TWIN).

In the next window pictured below, the LPF Filter Cutoff setting is selected. It has two sets of grab handles for moving the breakpoints on the plot lines. You can move a line section up or down, and rotate the end sections to change their angles. You can also drag the handles left and right to change the piano keys where the parameter handles are located. Dragging from different locations on the lines do different things, try a few different things to see the different behaviors. The keys are shown on the bottom of the graph, and the levels are shown on the right side in the boxes.

As you can see in the plot below, the filter cutoff frequency changes across the keyboard. A large number of parameters in the VP1 can be adjusted to vary across the keyboard, basically all of the ones I've highlited in Green. On Yamaha FM synths, only the Operator Frequency and Level are key-scaled, but on the VP1 up to 60 different parameters can be key-scaled! (30 in Section A, 30 in Section B.) I haven't found any other synthesizer with this much variability. And the key-scaling here can be programmed with multiple break points. This allows the patch programmer to set up different instrument sounds across the keyboard. You might have a Bass Viol down low, a Clarinet mid range, and a Flute up on top. A single patch can sound like a three-instrument ensemble all by itself.

The controls on the left of the graph apply to the Controller assigned to this parameter, indicated by the range lines shown in the graph. With no Controller assigned, a value slider is displayed (shown below as 71). It positions the middle grey line showing you the default value that is being used, which you can actually see this time. When a Controller like a foot pedal is assigned, the Default grey line goes away and the left side panel changes to match the controller. The Controller can move the value up and down between the dark black maximum and minimum lines (maximum = rectangles, minimum = triangles).



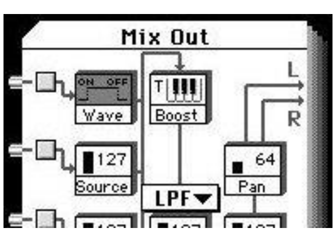
MDL

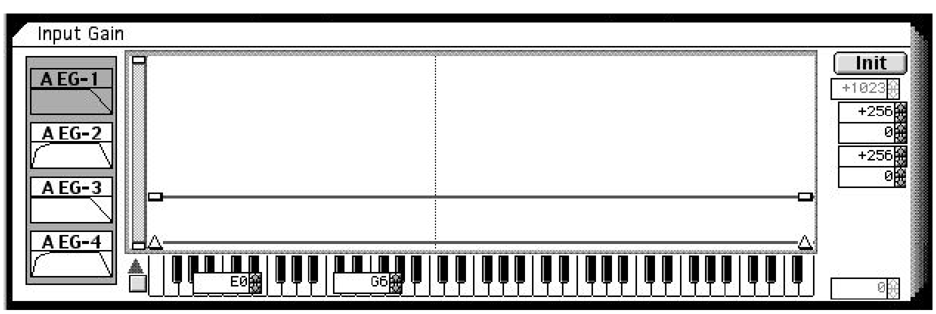
The square just to the left of the keyboard with the arrow above it allows you to zoom in on the graph vertically. Check the box to turn it on, then move the window scroll markers on the left side of the graph. The window range marker turns black when it's active. Uncheck the box to go back to maximum view range.

On the right side of the Graph are the boxes for the breakpoint amplitudes. You can see a maximum and a minimum box at the top and bottom slightly offset, then <u>four</u> pairs of max/min breakpoint boxes in between. The keyboard values for the <u>four</u> breakpoints are listed in the keys section along the bottom of the Graph. (0,0) is off the graph to the left, (256,0) is the break point visible towards the left, (424,0) is visible towards the right, and (687,0) is off the graph to the right. The higher number is for the maximum line, the lesser is for the minimum line in the number pairs. The internal Tone Generator is capable of 128 notes, but the VP1 only has a 76 key keyboard. The 'Init' button resets the entire graph back to the Initialization patch settings.

<u>Inverting Response:</u> Sometimes you want one parameter to go up while another goes down, using the same controller. You can invert the response of a physical controller, but it's not obvious how. Look at the plot above, and you see the black maximum line has little rectangles at the break points? The minimum line has little triangles. You can drag the breakpoints up and down in the editor. If you drag the triangle up a ways, you can then drag the matching rectangle below it… The controller always moves from the *triangle to the rectangle* as you move it in the increase direction, from 0 to 127 as you press the pedal down or raise a slider. (Use the default slider on the left if you want to test this, or move the assigned controller.) Reversing which one is on top lets you invert the response. Got it? If you go down three pages you can see a window where the plot is inverted, the 'Wave Read Speed' plot has the triangle minimum line *above* the rectangle maximum line. You can also see this used in the RudeBrass patch which fades between the Oscillator and the Noise Source as inputs into the Delay Loops. You can even make the max/min lines cross over each other, (+) response on one end of the keyboard, (-) response on the other. I've also seen one patch where it criss-crossed several times, throwing the Pan setting all over the place as you play up the keyboard.

Controller assignments are saved as part of the patch. There are examples of all of the possible icons in the Main Menu section. There is also an On/Off switch controller, which is simply a Key On or Off control (see it below left). In the PipeOrg patch it's used to control the sustain (LpGain) of the Delay Loop, rather than use the Cycle or Free oscillator modes. It cuts the sound off short, rather than allowing it to die off. The figure below shows a typical Envelope Generator control window, where the range of the envelope can be varied across the keyboard, separate from the shape of the envelope. The one below is fixed across the keyboard, but it doesn't have to be. Notice that there are two breakpoint boxes in the keys section and two pairs of max/min breakpoint amplitudes on the right side of the Graph. Keyscaled parameters can have up to 4 breakpoints. You add more breakpoints by double-clicking inside the Graph window, delete breakpoints by double-clicking on the handles. Drag the handles around to change the envelope's response over the width of the keyboard.





Now, let's talk about something more interesting: waveforms. There are a LOT of impulse waveforms to choose from: 75! Most are recognizable instrument names. Strangely, very few of these waveforms are actually used in the Factory_Set. The Sine, Saw, and Square/Pulse waves are used the most. I think most of the patch programmers for

Yamaha were used to Analog synthesizers and weren't briefed on how the VP1 Commuted model was designed to operate. Since the waveforms are used mainly to fine-tune the acoustics of the Commuted Physical Model, they don't have quite the impact on the sound you would expect. This isn't a sampler, it's a physical modeling synthesizer. The drivers are used very differently. I've made notes below on what each of the Drivers sound like isolated by themselves in 1-Shot or Cycle mode, and in some cases when the Delay Loops are applied. The Drivers are just a short input pulse, they are not a complete waveform as used in a sample-based synthesizer. You have to send then into the Resonator to hear what type of sound they create, in a vibrating string or tube.

OSC

75 Different Impulse Waveforms:

Sample1,2	These two are both blank. Sample import slots maybe?
GtNylon1,2,3	Guitar, 3: lowest pitch 1: dominant harmonic is up one octave 2: up two octaves
GtSteel1,2	Guitar, 1: more muted 2: brighter
A.Bass	Nice bowed acoustic bass down low on Cycle
E.Bass1,2,3,4,5	1: smooth bass /reedy organ
	2: higher pitched pluck /bright synth clarinet
	3: more twang than 1
	4: more metallic /higher harmonics
	5: fuller and lower than 3, 5-3-1 lowest to highest dominant harmonics
EG_Mute	Electric Guitar, Muted
BanjoBody	Somewhat like a Banjo twang
Koto	(13 strings) mellowest of the Banjo, Koto, and Shamisen set
Shamisen	(3 strings) higher pitched twang than the Koto or Banjo
Violin	needs noise in the driver, and a softer bowed attack
Cello	fuller with additional higher harmonics than the Violin
Cembalo	early Harpsichord, gut strings
Harpsichord	metal strings
CP80	Yamaha Electric Grand, somewhat similar to a mellow electric guitar sound
Rhodes	tone bars
Piano	good down low, needs more resonance up higher
P-Hummer	Harmonica /resonant pipe organ
HommondB-3	Hammond Drawbar Organ
Itopia	mellower than the Hammond
Clarinet	mellower than E.Bass2
PanFlute	needs noise in the driver
Trumpet	needs shaping to improve the tone
Trombone	higher harmonics than the Trumpet
Whistle	metal whistle
Vocal_Ba	very Clarinet like, needs a formant filter (band pass) to be more voice like
Xylophone	single sine wave + high click
Marimba	secondary tones + mallet strike
Vibe	more resonant than the Marimba, add amplitude vibrato
Celesta	struck Bar Piano, no secondary tones like the Vibe, clearer tone
Agogo	more like a wooden bar tone (should be 2 bells)
BD	Bass Drum if no Delay Loops. With Loops = Electric Bass w/ some steel drum
Conga	needs a little delay for the tone, reed organ on Cycle
Claves	needs a little delay for the tone, less reedy pipe organ on Cycle

Typist	Typewriter Clicks near middle C, on Cycle an organ flute up higher, Harpsichord down low	
DrumKit1	E0 – A0 Snare	Kit1 is the lowest of the three
	A#0 – A1 Cymbal Tap	in MDL (but only on D0# and E0 in TWIN due to blending)
	A#1 – A2 Bass/Toms	(A#1 ~ C2 Bass Drum)
	A#2 – E4 Toms	(~C4 – E4 Practice Pads, Sticks/Rim Shots maybe)
	E#4 – G6 Practice Pad sounds	
	Adding some delay adds an airy note to the sound, more = short Piano Like tone	
	Cycle mode yields a buzzy electric piano sound	
DrumKit2	up a third from DrumKit3	1-3-2 lowest to highest pitch
DrumKit3	up a fourth from DrumKit1	
Noise	chiff sound, plucked string, buzzy sawtooth, or sharp pipe organ depending upon settings	
Noise->LPF	chuff sound, on Cycle there are comb-filter harmonics in all these noise waveforms	
Noise->BPF	toned thunk, strong 2^{nd} harmonic, few upper harmonics	
Noise->HPF	high chiff sound	
Sin	Sine Wave, 'Noise->BPF' is similar and an octave higher	
	Delay Loops = Odd harmonics 1, 3, 5, 7, 9, etc are strong and present (w/ PW=0, WvSize=0)	
	Loops + PW = ALL1 waveform + RES2 rounded waveform adjustable up to the 8^{th} harmonic	
	Loops + WvSize = FM 1:2, or 2:4, or 3:6, or 4:8 waveform adjustable up to the 4^{th} harmonic	
	Loops + WvSize + PW = ALL1 + RES2 waveform adjustable out to the 14^{th} harmonic	
Pulse_10%,25%,50%	10% = clusters of 9 harmonics, every 10^{th} one is missing, string tone	
	25% = clusters of 3 harmonics, every 4^{th} one is missing	
	50% = Odd harmonics only, every other one is missing, wood tone	
Saw, Saw_10%,25%	Buzzy Sawtooth, muted LPF version of the Pulse waveform, plucked string	
	10% = clusters of 9 harmonics, every 10^{th} one is missing	
	25% = clusters of 3 harmonics, every 4^{th} one is missing	
Triangle	Triangle Wave, Odd harmonics 1,3,5,7, wood block tone /Clarinet like	
DigiWave1-8	1: FS1R ALL1 + 16^{th} harmonic (up 4 octaves), fuzzy Sawtooth + a wood block strike	
	2: Buzzy formant sound, with delay = Brassy in the lower octaves, plucked tube	
	3: harmonics 1, 5.2, 9.4, etc, adding delay fills in 3.1, 7.3, etc.	metallic blown pipe sound
	4: 10% pulse wave with inharmonicity	buzzy synth lead
	5: Odd harmonics + strong 13, 15, 37, 52.5 + inharmonicity	metallic tubular lead
	6: harmonic triangles at 3, 7, 10, 16, 23 etc + inharmonicity	buzzy synth harpsichord
	7: metallic, every 4^{th} harmonic, inharmonicity, strong 17^{th} harm.	bell tone lead
	8: triangle + three metallic inharmonics	brass + metal accent
DigiVx1,2,3,4	1: first 6 harmonics, then triangles on the 18^{th} & 30^{th}	buzzy brassy 'ah' sound
	2: slightly brighter than 1, triangles on 2-5, 17-19	
	3: buzzier sound, triangles on 2-6, 11, 17-20	
	4: lower base pitch, triangles on 4-6, 14-17, 24-27	
DigiWild	higher frequency, pitched squidgey sound	
Organ_Wv	clear electronic organ /raspy pipe organ using delays	
PipeOgWv	multi-rank electronic organ /pipe organ using the delays	
A.Sax_Wv	mellow reed organ (Alto Sax)	
T.Sax_Wv	buzzy reed organ (Tenor Sax)	
BassonWv	dominant 3^{rd}, 9^{th}, 14^{th} harmonics (Bassoon), needs formants	
MtReedWv	Oboe/Flute-like mix	
Sin2	Sine Squared, slightly brighter than Sin, a little of the 2^{nd} and 3^{rd} harmonics are added in	

OSC Setting:	1 Shot, Cycle, Free	
Wave: PW, WvSize	Single Values, changes the Pitch & shape of the sound entering the Delay Loops	
Filter Selections:	NL, THR, LPF, BPF, HPF	(NonLinear, Threshold, Low Pass, Band Pass, High Pass)
Cutoff	Keyscaled,	Variable across keyboard
Resonance, Volume	Single Values	

Noise

Noise:	Parallel, Serial routing	
Filter Selections:	NL, THR, LPF, BPF, HPF	(NL = NonLinear Filter)
	(NonLinear, Threshold, Low Pass, Band Pass, High Pass)	
Cutoff	Keyscaled,	Variable across keyboard
Resonance, Volume	Single Values	
LowDamp	Keyscaled,	Removes the DC level in the signal

Mix Out

Filter Type:	THR, LPF, BPF, HPF, BEF	(BEF = Band Eliminate Filter)
Wave, Source, Loop1, Loop2	Keyscaled	
Boost, Resonance, Depth	Single Values,	Boost is used to create distortion, clipping in the waveform
Cutoff, Volume, Pan	Keyscaled	Volume does not affect the waveform, no Distortion

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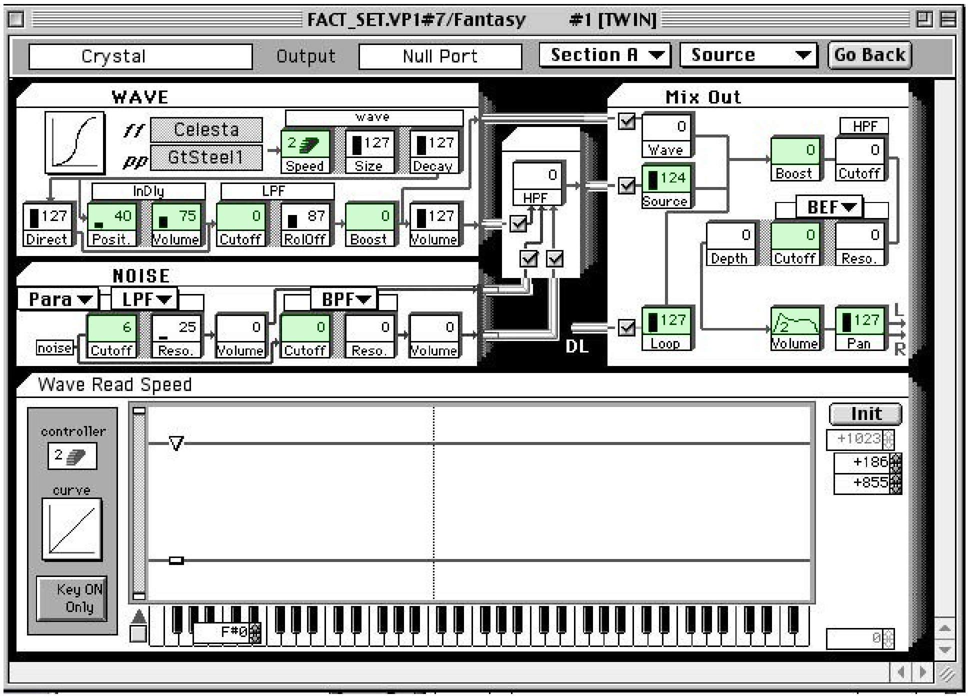
The VP1 Driver waveforms are not just a single, simple instrument pulse. Listening to the Drum2 waveform, there appear to be at least five distinct samples per waveform. ** In MDL there is no blending between samples, but in TWIN it does blend the samples. Curiously, there are no Drum patches in any of the patch banks. You may also have noticed the FS1R waveform references in some of the driver descriptions. The VP1 came first in 1994, these waveforms are the precursors to the FS1R/FMX waveforms.

At the same time the VP1 was under development, other Yamaha synths like the SY77 and RM50 used AWM synthesis using a set of fully resonating instrument samples from one instrument, with up to 61 samples per element for a drum set. Most used around a half dozen samples. The RM50 was also capable of stretching one waveform over the entire length of a 61 key keyboard, the opposite method. Later, Motif models used a system where samples have key-bank information, detailing which keyboard keys the sample is to be used on, with blending used to blur the transitions. The VP1 drivers don't need to be quite this sophisticated since they are extensively modified by the physical acoustics engine before emerging as a sound from the speakers. They're simply a set of impulse sounds to strike the resonator with, a short pulse, not a fully developed sample. It's very difficult to tell if all of the waveforms in the VP1 use just five impulse samples or not. I'm pretty sure they do, I can hear the transitions while I'm editing if I'm listening for it.

TWIN MODE

In the TWIN Window you can see the Fantasy patch has TWO impulse waveforms assigned when using the TWIN Model. See below where Celesta and GtSteel1 have been assigned in WAVE. There is a small label next to each, one is *ff* the other is *pp*, implying one is used for loud key strokes and the other for soft ones. The two are blended using the scheme displayed in the square shape icon just to the left. The blending mode is selectable. Also notice this reduces the number of Delay Loops down to one (next section). Can you see the single DL label in the center of the window feeding into the Mixer? Previously we had two Delay Loops. Again, the Source line (a mix of the Wave and Noise oscillators) also goes into the Delay Loop in the next section as well as into Mixer.



TWIN

WAVE Mode Setting:	(1 Shot only)	
Speed	Keyscaled	(Keyscalable parameters are highlited in Green)
Size, Decay, Direct	Single Values	
InDly: Position, Volume	Keyscaled	
Cutoff, Boost	Keyscaled,	Boost is used to create distortion, clipping in the waveform
RollOff, Volume, HPF	Single Values	

Notice that the WAVE parameters for the Oscillator are also different than those seen in the MDL OSC block, and that it has no mode to select (no 1 Shot, Cycle, Free). TWIN only allows 1-Shot mode, a plucked or struck instrument sound.

<u>Wave Speed</u> (backwards to WvSize in the example above because the response default lines are inverted) specifies how many cycles of the impulse waveform are used, from zero up to a maximum of four cycles in one period. 1 is duller, and 127 is brighter (if not inverted). Zero is no sound. Around 8 adds in a pitched thump below the harmonics 1,3,5, a setting of 15 sounds muted and without the thump, with harmonics 1,3,5. To get one complete cycle (one copy of the waveform) use a setting of 32. Two cycles is 64, three cycles is 96, and four is 127. If Wave Speed and Size match, you get just a sine wave (a single copy of the Speed-selected waveform with no additional harmonics). If they don't match, you get multiple harmonics with the Speed-selected one emphasized. The higher the WvSize, the more harmonics that are present up to about a setting of 7. (See the table below.)

<u>WvSize</u> selects how many of the waveform cycles play back in the pulse before Decay kicks in, (it must be >0 for any sound to come out). The default is 127. See the graph below. <u>Decay</u> controls how fast the waveform decays in amplitude. Using a long Decay value extends the length of the input pulse. Using 0 up to about 10 adds a buzz onto the first part of the note, kind of like the start of bowing a Cello string. From 20 up to 127 there is no real difference. Decay doesn't really buy you much of anything, so just leave it at 127 most of the time (default).

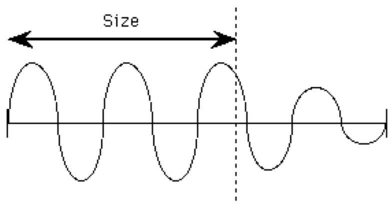
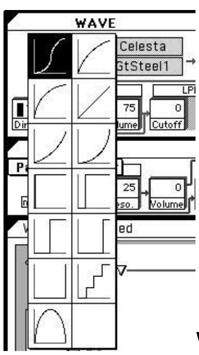


FIG.30

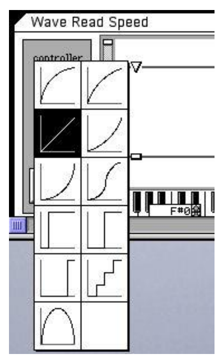
TWIN, Wave Speed = 127 (4 cycles) and Size = 88 (out of 127)

Additional settings are Direct, which is a gain adjustment for the waveform bypassing the filter Input Delay. Position is the length of the time delay before the "Comb Filter" kicks in. Using a high Position value (a long time delay) lengthens the note Attack by delaying filtering on the first part of the sound. Setting it to almost zero sharpens the pluck, and raising the value tunes the harmonics in the note strike. To emphasize the attack, lower the Direct gain and raise the InDly Volume. It can add a string snap or pick snap at the beginning of the note for a guitar, or the sharp Attack on a Piano (see any of the Guitar patches and the BigMyPiano patch). This feature is not included in the MDL model.

Above, the selected block for Wave Speed also shows a Mod Wheel Controller in the graph section on the left. Notice that the minimum line (little triangle) is above the maximum line (little rectangle). When the Mod Wheel is at 0 the value is at it's highest, and when you roll the Mod Wheel forward the value drops. This displays the inverted response I talked about earlier, when you drag the maximum line down and the minimum line up above it on the Graph. The triangle is also pointing down now, where it normally points up. The limits now start at 855 and drop to 186 as the Mod Wheel is moved upwards. The minimum and maximum values are displayed in the boxes on the right side of the Graph.



Wave *pp/ff* Mixing Curves



Controller Response Curves

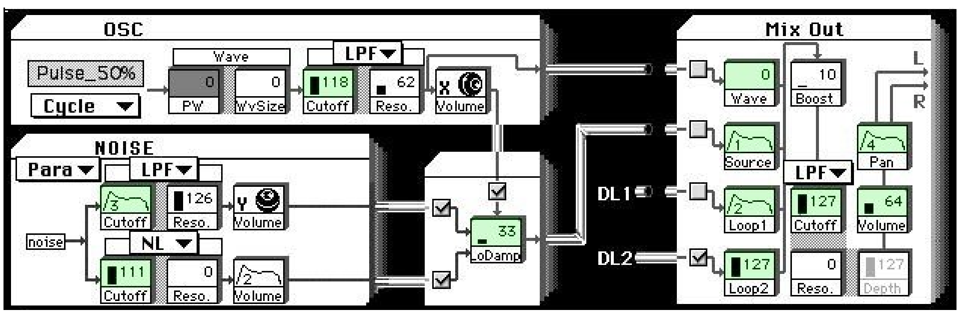
MDL Mode, you may have noticed under the Sine waveform selection for the Oscillator that the waveform creates some very interesting harmonics when you run the sine wave through the Delay Loops. In MDL with the PW and WvSize both set to zero the delays resonate, adding in the Odd harmonics, 1, 3, 5, 7, 9, etc. This sounds close to a raspy Clarinet. By increasing the PW slightly you can modify the resonance and get the Even harmonics as well, shifting it towards a sound more like an Oboe mixed with a brass horn. Further increasing the Pulse Width creates a rounded shape in the harmonics, pushing the rounded peak out to a maximum of two octaves above the pitch, up to the 4^{th} harmonic. This is like the FMX RES2 waveform. If you set the PW to 0 and raise the WvSize you get a triangular peak in the harmonics, which can be adjusted up to the 4^{th} harmonic, the FMX RES1 waveform. Used together, the PW and WvSize can push the peak out to the 16^{th} harmonic. There is also a set of harmonics at the fundamental like the Yamaha FS1R or Montage ALL1 waveform, which stay put as the newly created shaped-peak moves up in frequency. The combinations are similar to the resonant waveforms in the Yamaha FS1R/Montage, mixed with a damped ALL1 waveform. I have another paper on my website describing FM Synth Programming which contains a section in the back discussing how to utilize these waveforms on the FS1R. Basically just remember low PW/Size values are dull, high values are bright.

In TWIN mode it's different harmonically from what we just described. Here is a short table of what you get at different values of WvSpeed.

	0	No sound
Dull	8	Drum thump and soft 1,3,5 harmonics
	15	Muted 1,3,5 harmonics
	32	Strong sound, harmonics **1**,3,5 (Harmonic #1 is the strongest)
	64	Strong sound, harmonics 1,**2**,3
	96	Harmonics 1,2,**3**,4,5
Bright	127	Harmonics 1,2,3,**4**,5,6,7

If WvSpeed = WvSize, just the peak harmonic is present.

The combination of using the correct Driver, PW/WvSize or WvSpeed/WvSize settings, and the Pitch settings for the Delay Loops, and some creative filtering we should be able to mimic the acoustic characteristics of most of the wind instruments on the VP1 much like the VL1 can. That's my theory anyway. We'll see if it works in practice. Instruments that use formants like the Cello, acoustic Guitar, and Bassoon will be much more difficult, but we have a solution for that too in the Effects module using the Equalizer.

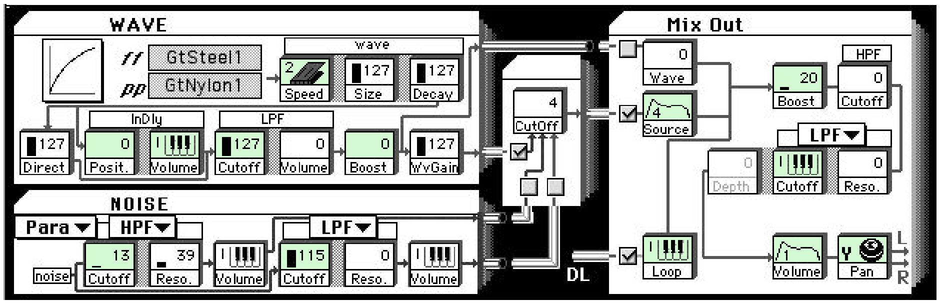


MDL

Noise

Originally the Karplus-Strong model used a noise pulse into the delay loops to initiate the sound. The VP1 has a noise input block into the system, with two parallel filters to shape the noise pulse. The filters can be placed in Series or Parallel, along with standard filter settings. These can be used to add the air hiss into a wind instrument patch, the bow hiss of a string instrument, or the voice hiss you hear in a human voice. The Noise inputs require an envelope on the Volume block if you don't want them to hiss continuously. Envelope Generator #2 is being used above on one of the Noise sources. TWIN is identical in function (below) but the blocks are arranged differently in the window.

The output from the Noise source is mixed with the output of the Oscillator through a Lo Damp Filter before going into the Delay Loops and the main Mixer (Mix Out) in the MDL model. A Lo Damp filter removes the DC level from the signal which can creep in through prior sound manipulation, and centers the signal back on zero. You can't really see this unless you run the signal through an oscilloscope, but it can cause clipping in the signal, taking definition out of the sound and introducing high frequency hiss. In the TWIN model, the output from the noise source and the Oscillator go through a High Pass filter instead. These two types of filters are actually very similar.



TWIN

Mix Out

Mix Out is the block where many of your physical controllers will be mapped. The mixer input blocks allow you to control the volumes of different parts of the sound with different controllers, like velocity or a foot pedal. You can fade one sound source in while fading another out, or hit one part hard on the attack separate from the others with an envelope. The Wave and Source inputs can be used to mimic a string pluck or piano hammer strike. Place a velocity controller (I, T) on an input block to make notes velocity sensitive (see above on the Mixer Loop input).

The rest of the Mixer controls are pretty standard, with added keyscaling on Boost, an extra HPF in TWIN, then a final output Filter, Volume and Pan control. Boost can be used to create distortion, Volume does not. These are where you might introduce a wind vibrato LFO controller on the Boost, add a controller to change the timbre of the sound with Aftertouch on the LPF filter, and add in a volume envelope to shape the sound. I generally like to put Velocity (I or T) control on the LPF Cutoff so strong key strikes are brighter than soft ones. Using an Envelope with a Continuous Controller (CC) on the Volume block would allow you to alter the overall shape of the note, then control the amplitude of the envelope with something like an Aftertouch controller or Touch EG to make the patch more expressive. On Pan, sometimes like on my Piano patch I Pan the lower notes left, and higher notes to the right.

As a note, the filter Depth is only used on the Band Eliminate filter (BEF).

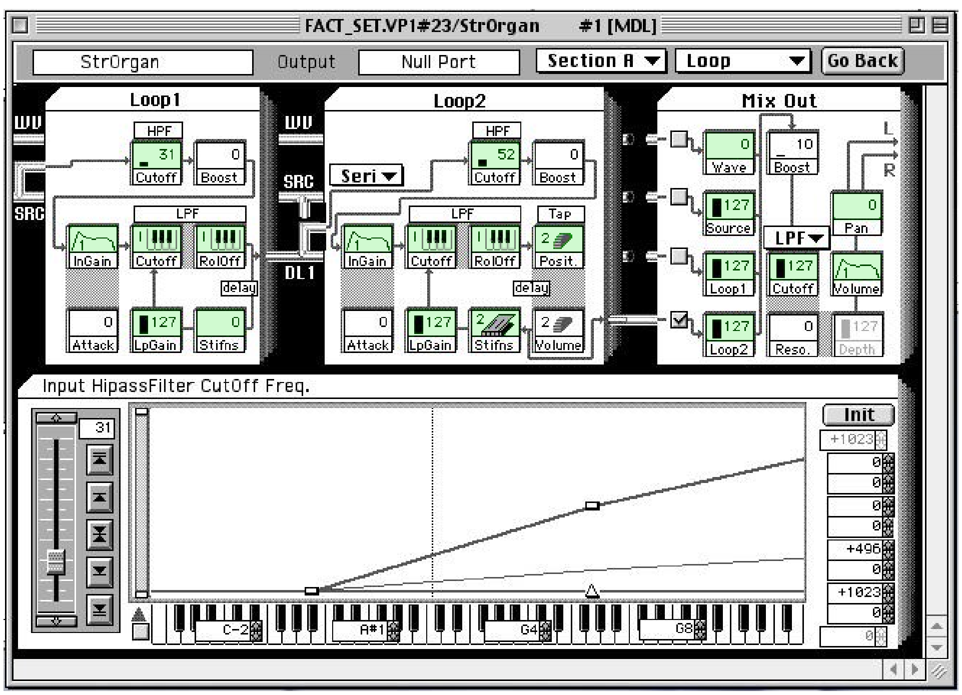
It takes some time to add in all of the controllers after you have the patch working, but it’s worth it since it adds so much expression into the patch. A lot of people skip this step which is why some custom patches are a bit mechanical and lifeless.

Loop ____

The next window for the MDL model contains two processing Delay Loops for altering the characteristics of the sound, and the same Mixer seen in the Source window. These model the instrument's strings or air tube. The Loops contain more filters and envelopes, and add in a special Stiffness block we'll talk about in a minute. Loop2 has a Tap setting which taps into a higher harmonic, changing the resonant pitch of the patch. The two Delay Loops can run in parallel or in series in MDL mode.

You enter Loop1 through a High Pass Filter, through a Boost block which can add distortion, then the <u>Input Gain</u> can be altered with an Attack before entering the loop proper. The <u>Attack</u> here is very subtle, it's nothing like the bold InDly gain parameter in the TWIN Oscillator. Lower InGain to enhance the Attack. The repeating loop uses a Low Pass Filter to reduce high frequency sounds more than low during each pass, then it passes through an inharmonicity <u>Stiffness</u> block which shifts the higher harmonics sharp (see Appendix A) then through an <u>LpGain</u> block which attenuates the signal during each pass through the loop so the volume dies away. If LpGain = 127 there is no decay, it holds the note like an Organ. Loop2 is nearly identical, it adds a harmonic <u>Tap</u> to allow you to tap off higher harmonics from the notes to include in the patch. You can tune this to add a tone like a third, a fifth, an octave, even inharmonic tones. You can also use the two delay loops for different functions, one can be the sustain sound while the second can be a sharp note attack, as used in my Grand Piano patch.

You may have noticed there are a LOT of Filters in the Source windows and the Loop windows. There is one for each voiced oscillator, one for each noise source, a combined output filter, two in each Delay Loop, another in the Mixer, and then there is a 5-Band Equalizer in the Effects section. The VP1 has the characteristics of a Subtractive synthesizer internally, aside from the delay loops. It doesn't have to be used that way, but that is an option. Using excessive filtering can completely squash the timbre right out of the Drivers and turn the sound into something more like an Analog waveform. A number of the factory patches suffer (or benefit) from this.



MDL

Loop1

HPF Cutoff, InputGain, LPF Cutoff, LoopGain	Keyscaled	(Keyscalable parameters are highlited in Green)
LPF RollOff		How fast the filter drops off
Boost, Attack	Single Values,	"Boost the input signal to create distortion."

Attack	"Emphasizes the InGain for the period of one Delay Loop."	
Stiffness (Inharmonicity)	Keyscaled,	Variable across keyboard

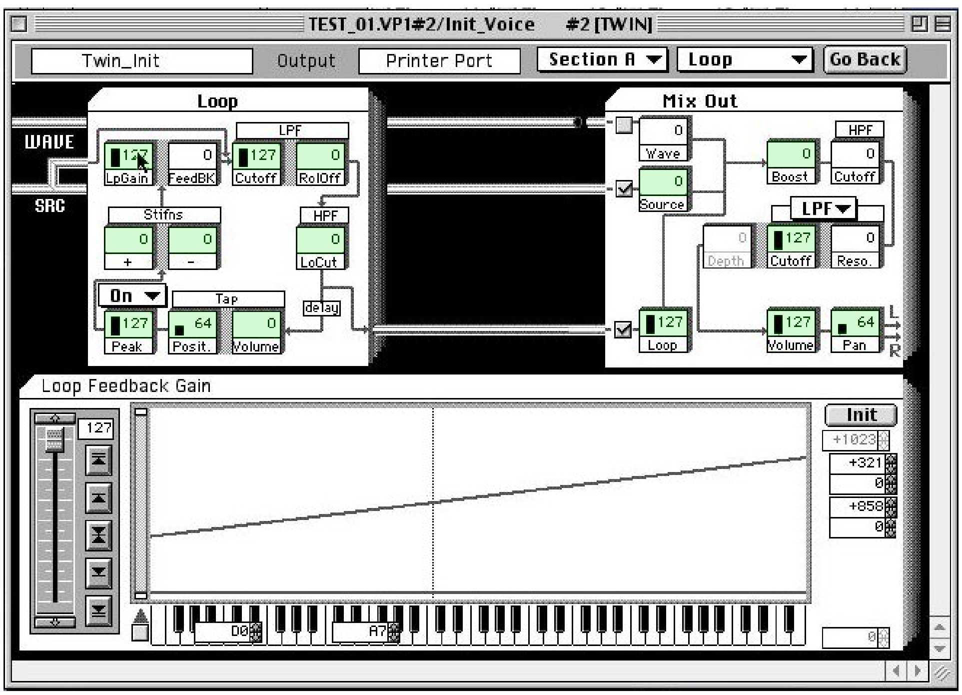
---

Loop2

Series or Parallel	Routing in Loop2	
HPF Cutoff, InputGain, LPF Cutoff, RollOff, LoopGain	Keyscaled	
Boost, Attack, Volume	Single Values,	Boost can cause Distortion, Volume does not
Stiffness (Inharmonicity)	Keyscaled,	Variable across keyboard
Loop Tap Position	Keyscaled,	"Creates harmonics."

Mix Out

Wave, Source, Loop1, Loop2 volumes	Keyscaled	
Filter Type:	THR, LPF, BPF, HPF, BEF	
Cutoff, Resonance, Depth	Single Values	
Boost	Keyscaled this time,	another Boost to create distortion
Cutoff, Pan, Volume	Keyscaled,	Volume does not



TWIN

TWIN mode is a little different, it only has one delay loop. Looking at the Loop section, the loop is more complicated than the MDL model. The filters are in a different order, the Low Pass Filter is first, then the High Pass Filter is inside the loop here where it wasn't in the MDL model, followed by the Tap blocks. There is a <u>Peak</u> limiter next which you can turn on and off which can act as an Attack setting at low values. This can turn a Harpsichord into a Banjo by setting the Peak Limit to 0. The original manual says "smaller numbers for the Peak Limiter result in a stronger attack". <u>Stiffness</u> can be adjusted differently for the – and + parts of the waveform. The manual says "Setting the two + and - parameters to different values will cause dramatic changes in the sound." <u>LpGain</u> is next attenuating the sound as it repeats the loop, setting the note decay time. <u>Feedback</u> can be used to distort the sound or put the loop into a sustained sound mode, more like Cycle or Free OSC settings. Doing this, the TWIN algorithm can also create a sustained waveform, not just a plucked one. See the BowedAirs patch.

Sometimes you just can't seem to get the right setting out of a parameter using the slider, one value is too high, the next notch down is too low. You can move the two limit lines on the graph (which range from 0 to 1023) closer together to get more resolution for the controller (which only ranges from 0 to 127). If you leave the limit lines at the extreme top and bottom of the graph, the controller will jump (1023 / 127 = 8) 8 values per click. Moving the lines closer together you can reduce that to one value per tick on the controller, for increased resolution.

Loop (TWIN)

LoopGain, LPF Cutoff, RollOff, HPF Cutoff	Keyscaled	(Keyscalable parameters are highlited in Green)
LPF RollOff		How fast the filter drops off
Loop Tap Position, Volume, Peak	Keyscaled,	"Creates harmonics."
Peak Limiter	ON/OFF,	"Smaller values result in a stronger attack."
Stiffness (Inharmonicity)	Keyscaled,	+ and - shifts are available simultaneously.
Feedback	Single Value,	Feedback as on an electric guitar.

Modeling in the VP1

While we're here buried hip deep in the modeling parameters, it might be a good idea to explain HOW these model an acoustic instrument. Generally, the model starts with a Driver signal, the pluck of a string or impulse sound of a reed mouthpiece. The Source window injects the sound impulse into the string model along with a Noise source. Both are filtered and have gain adjustments. There are three Driver modes, 1-Shot, Cycle and Free. 1-Shot is a struck or plucked Driver, like a mallet hitting a Marimba. Cycle and Free are continuous Drivers, like bowing a Violin or blowing into a Clarinet mouthpiece. There is also a noise oscillator in parallel with the waveform driver. If you listen to a wind instrument, or listen closely to a bow on a string instrument you hear a lot of hiss along with the harmonic sound. The VP1 has the ability to match this in a patch, though not quite as well as the FS1R does.

The second section of the Model is in the Loop window, where the resonator is located. In a string instrument, the wave travels away from the plucked point on the string in two directions, towards the bridge of the instrument and towards the upper frets. Mathematicians were calculating the wave equations specifically for this back in the 1700's. The wave reflects off the ends of the string, inverts, and travels back the other way. These two paths are modeled with Delays, one short one for the bridge and a long one for the frets. That's why you see two Delays in the Loop window. The Pitch settings (next window) adjust the lengths of the two Delays. The sound wave inverts when it reflects off the bridge and fret, causing a sharp step change in the wave. This introduces aliasing (high frequencies) into the sound wave. We add a low-pass filter into the loop to remove these high frequencies, since a real acoustic instrument smooths out the sharp corners. The sound also attenuates (loses energy) when it reflects off the bridge and fret so we add in an LP Gain block to reduce the energy in the Loop. Due to the Stiffness of the string, the frequencies of the harmonics shift sharper as well, so the VP1 adds in a Stiffness block to model this in the Delay loop. We'll talk more about Stiffness in a minute. The VP1 also has a high-pass filter on the front to give us one more adjustment parameter just for good measure, since really long waveforms can overload the Delay Loops (mentioned in the original editor guide).

This model using two delays also works for wind instruments. The research paper at Stanford University that inspired the VL1 model shows you how to simulate a tapered tube of a Saxophone using two unequal-length cylinders. The sound wave travels down the two tubes, reflecting off the ends of the tubes. The equations are identical to the string model, so two-delay models work quite nicely for wind instruments. It can also be modified for other configurations. For a Trumpet or Flute, the lip reed is at the end of the tube. The short Delay gets set to zero (or left at the same value as the other Delay) and only the longer value is used. For some reason, different delay lengths were never used in any of the factory patches of the VP1 like it's used in the VL1. I intend on correcting that in a new set of patches.

The next part of the model is the resonant body of the instrument, the body of the guitar or the wooden body of a bassoon. The VP1 models this partly by initializing the Driver input pulse with the acoustic characteristics of the instrument body. It uses a digitized recording of a thunk on the instrument body rather than a triangle wave or white noise as input. Earlier physical models used a generic noise pulse as input to excite the system. There are 75 different drivers to select from in the VP1, all of which contain the acoustics of the instrument in them. The Yamaha VL1 in contrast, calculates the resonator body effects on the sound waves in high detail, using add-on modifiers in the processing chain. We can only roughly approximate this in the VP1. It's almost as if the designers of the Commuted Model at Stanford University neglected this part of the model.

I also realized why initializing the Drivers with the instrument body characteristics doesn't work 100%… The instrument body essentially acts like a set of <u>fixed</u> frequency band pass filters (formants). When you prefilter the Driver to match the instrument body, then modify the pitch to match the key you've pressed, you've shifted the locations of all of the prefilters. This destroys the formant characteristics of the instrument body you were trying to emulate. The multi-pitch wave form in the VP1 somewhat corrects for this. A set of bandpass filters applied at the Effects block would work better.

If you dig through all of the VP1 Effects section, you will find Box, Board, and String Effects. These allow you to add additional resonant characteristics to any of the Drivers more like the instrument body we want to model. There should also have been three more, for Cylinder, Cone, and Bell resonators for wind instruments. These are missing. But never fear, there is a way in the VP1 to create the instrument body Formants. In the Effects section is the Element Equalizer which has the settings and ability to create five Formants, similar to the ones in the Yamaha FS1R. If you're interested in learning more about Formant Filters, go get my primer on FM Programming and read the section towards the back on the Yamaha FS1R Formants. We can use these to model the characteristics of the instrument body.

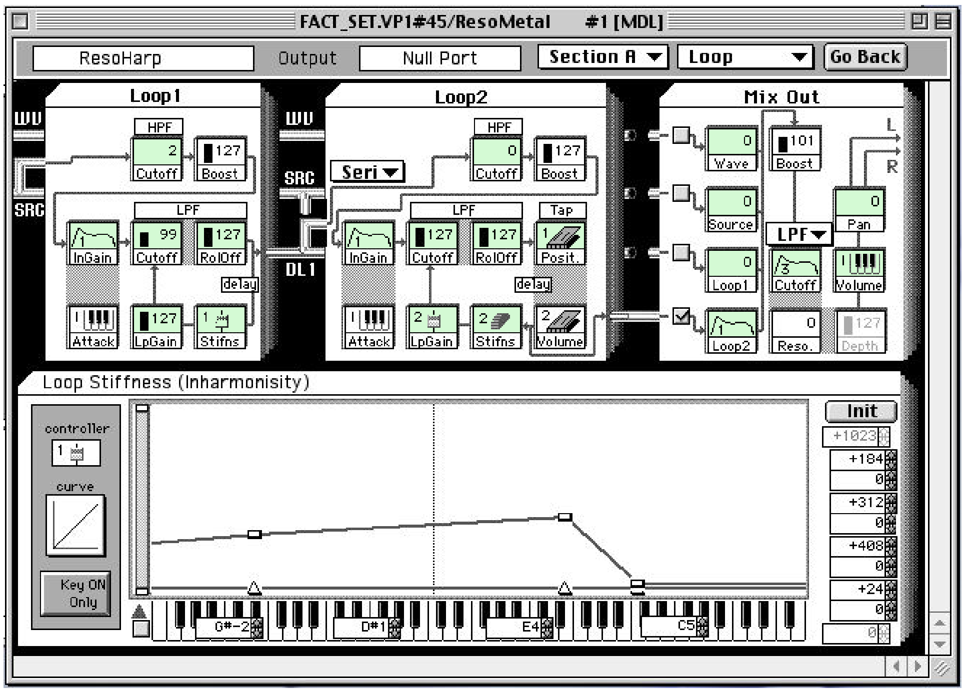
I think you can see from here where most of the parameter blocks in the VP1 come from and how they were meant to be used to model an acoustic instrument. It's now up to the patch programmers to use the tools provided to create expressive instrument patches from the model. Looking through the patches, very few actually use the model the way it was intended. All of the patches I've looked at have kept both Delay Loops at the default length, they didn't use two different delay lengths as in Stanford University's Saxophone and String Instrument mathematical models. Only a couple of the factory patches use the Delay Loop with the Pitch Bend disconnected to model the mode shifts in a Trumpet or Flute. And, very few of the factory patches model actual acoustic instruments which were at the core of it's design. These seem to be significant oversights from my point of view.

Loop Stiffness (Inharmonisity)

OK, lets talk about Loop Stiffness for a minute. Notice that it is labeled 'Inharmonisity' on the graph below? This is one of the key parameters to the VP1's harmonics. This adjusts the spacing of the harmonics to give you an expanding fractal fan of harmonics unique to the VP1, the harmonics get sharper as the frequency increases. The VP1 can alter the Inharmonisity well beyond any acoustic instrument, creating some really bizarre patches. It's discussed more in **Appendix A**, and it's adjustable across the VP1 keyboard. Inharmonisity creates an expanding set of harmonics which are seen in many acoustic instruments (listed and quantified in the Appendix). This parameter gives the VP1 an edge over the VL1. The VL1 doesn't model Inharmonisity at all, which is why it has trouble matching stringed instruments.

In the example below, Inharmonisity is adjusted using Slider #1 on the keyboard console. Inharmonisity can drastically change the harmonic structure of a patch, changing a low smooth String sound into a Gamelan, then into a Water Glass as you go up the keyboard. Using the Slider you can alter the harmonics actively as you play, warping the sound in all kinds of directions. Inharmonicity is why the VP1 can natively model the Piano, Guitars, metallic instruments and the string section so well. In spite of this capability, the Piano driver waveform already contains most of the inharmonicity it needs already. I'm finding that out as I set up a new Piano patch from scratch.

As far as I know, the VP1 is one of only three hardware synthesizers built with this inharmonic-fan capability. I've tried replicating it on the Yamaha FS1R, but I can't replicate the way the fan changes as you go up the keyboard or alter it with a controller like the VP1 can. The Korg Oasys has a dispersion setting which models the gauge of the string, and I also recently found a "stiffness" setting in one of the undocumented String blocks in the Nord Modular G2. The block is only accessible using the open source editor, the factory editor left it out entirely. It looks like Nord were intending on adding it in, but never hooked it up. Both Loop1 and Loop2 in the window below link the Stiffness parameter up to a manual controller, Loop1 to a slider and Loop2 to a mod wheel.



MDL

There is also an anomaly in the Inharmonisity algorithm in the VP1, it gets progressively more pronounced as you go up the keyboard. It models a single string, changing the length of the string for each different pitch, where most stringed instruments use multiple strings, each with a different Stiffness. In order to keep it under control, the patch programmers have decreased Inharmonisity at the top end of the keyboard to calm it down. The graphs for Inharmonisity all reflect this. Just be aware of this, Inharmonisity doesn't behave quite like you expect it to.

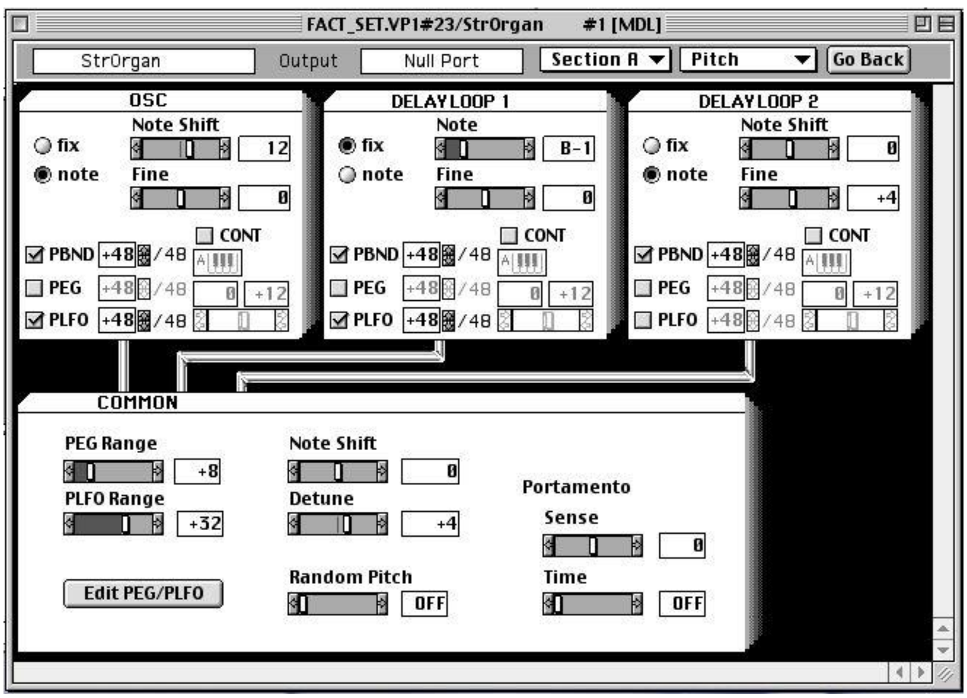
A foot pedal is connected up to the delay Tap above. The Keyboard controller with a letter 'I' in the block uses the Initial Velocity to set the parameter value in between the two lines on the graph. The loop Attack parameters use this to sharpen the Attack slightly if you strike the keys harder. Initial Velocity is also linked in the Envelope Generator Windows to the envelope depth and speed. You can also see a 'Key ON Only' button in the Controller block of the graph. This makes the controller function only at Key-ON, then it turns it off for the rest of the note.

Pitch

The next section of the editor adjusts the Pitch of the different sections of the patch, the Oscillators, Delay Loops, and an overall control for the output signal. OSC Pitch adjusts the overall Pitch of the oscillator, the input pitch into the Delay Loops, not the actual pitch of the note. Higher input pitches increase the higher frequency harmonics of the notes. Lowering the OSC pitch deepens the sound, as seen in the BigMyPiano patch Element 1 where the pitch is lowered over an octave. Loop Pitches are not so obvious, they control the length of the modeled string used to calculate the timing of the Delay Loops and set the actual pitch of the note. These essentially override the oscillator (if you use the Delay Loops). There are separate Pitch settings for Parts A and B. In fact, there are two of everything in the models, with the exception of the Effects block.

The Fix radio buttons allow you to select a fixed length Delay Loop if your instrument model includes this. A Note Shift setting of +12 bumps the Oscillator pitch up one octave increasing the higher frequency harmonics. Shifting it down -12 steps deepens the harmonics, richening the sound. There are also controller mappings for the Pitch controls. Can you see the 'A' icon block just under each CONT radio box? These map the Aftertouch controller to the Pitch Continuous Controller in this patch when the Continuous Controller CONT checkbox is checked. The Continuous Controller is a secondary Pitch controller, which augments the Pitchbend kind of like an offset control. All of the items under the CONT checkbox belong to the Continuous Controller, providing you with a mapped controller, range values and a range slider.

The Common block contains links to the PEG and Pitch LFO window, and the Portamento settings. There is also a Pitch shift/detune setting for this Element here.



If Pitchbend is unchecked in the Delay Loops (keeping the Delay Loop or resonator from changing length) you can get an effect like a brass instrument when you Pitchbend. The instrument jumps upwards through the different harmonics like a Trumpet can, not bend the pitch like you do on a Violin. Flutes do this too when they are overblown, they jump up one harmonic (up one octave) when you push them hard. This is a feature of the Yamaha VL1 we can use here as well. You need to increase the Pitchbend Range in the Elements Window (page 16) to jump harmonics, maybe up to 48 for 3 octaves.

OSC

Fixed or Note:	Note or NoteShift, Fine “A setting of C2 simulates a string length with a pitch of C2.”
PBND, PEG, PLFO	Single values
CONT	Additional Pitch Controller On/Off
CONT Range & Slider	Single values

DELAY LOOP 1

Fixed or Note:	NoteShift, Fine
PBND, PEG, PLFO	Single values
CONT	On/Off
CONT Range & Slider	Single values

DELAY LOOP 2

Fixed or Note:	NoteShift, Fine
PBND, PEG, PLFO	Single values
CONT	On/Off
CONT Range & Slider	Single values

COMMON

PEG Range, PLFO Range, Note Shift, Detune	Single values
Random Pitch:	On, Off
Portamento:	Sense, Time

I wrote Dave Polich about his BigMyPiano patch, to ask him how he came up with the patch. This is what he said:

> “To be honest, the VP piano was just a happy accident. I was moving parameters at random until I had an "a-ha" moment and realized I could turn it into a piano of sorts. In other words, I had no idea what I was doing!”

I disagree, I think he did know what he was doing. Part A is lower in tone, Part B is higher and more tinny in nature. His patch uses the Pitch parameters very creatively, he down-shifted the OSC Pitch on Part A (-13) half steps (over an octave) to lower and richen the timbre, and up-shifted the Delay Loop 2 Pitch slightly in contrast for Part B. Comparing it to the Railsback tuning curve for the piano (in the Appendix) this matches the flat lower register notes and sharp upper register notes on the plot. Very nice! I actually prefer his patch with Element 2 turned off, more Grand Piano than Honky Tonk. My own Grand Piano patch is sounding pretty good now too, a bit thicker than Dave’s and not as sharp on the attack since I used the MDL model rather than TWIN like he did, and I also used some Inharmonisity in mine where he didn’t.

I’ve also gone through the process of inserting all of the controllers to allow warping it in all kinds of directions just for fun. I’ve set up a standardized controller map so I can set up all the controllers similarly on each patch. The factory patches are all over the place, every patch is different. You almost have to open the editor in order to figure out how the controllers are mapped. I’m hoping to make mine more consistent from patch to patch.

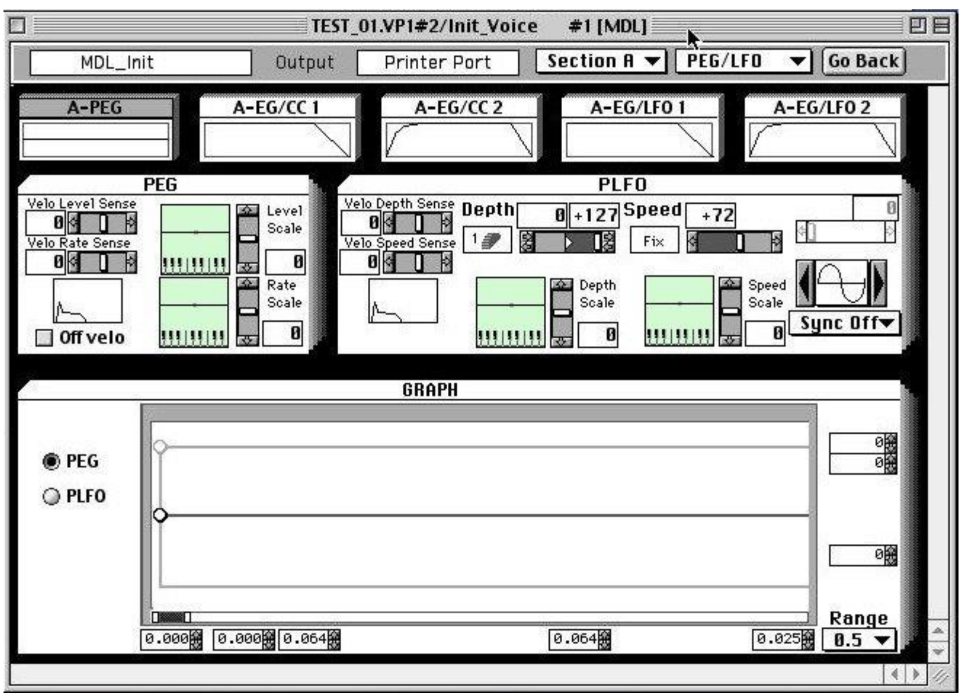
PEG/PLFO

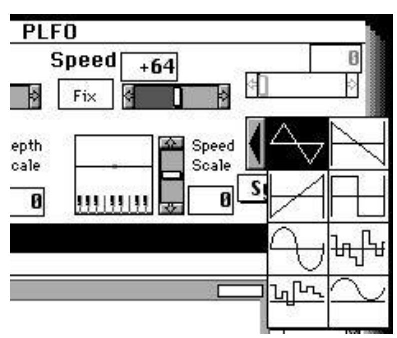
The Pitch Envelope Generator and the Pitch Low Frequency Oscillator are next in the first Envelope window. The Velocity you strike the keys with is an active input into these parameters. You can alter the sound quite a bit just in the way you strike the keys. The PEG envelope depth and the envelope speed are Velocity sensitive, as are the PLFO depth and speed. Setting the envelope speed negative makes harder key presses ring longer. The VP1 also has Aftertouch, and a combination of Velocity and Aftertouch called TouchEG, which can be used to modify parameters. There are two PEGs and two PLFO's, one set for Section A and another for Section B.

All of the Envelopes and LFO's on the VP1 are keyboard following. (Keyscalable parameters are highlited in Green.) For example, on the PEG there is a larger square with a keyboard across the bottom and a plot line across the middle. As you alter the Level Scale or Rate Scale up or down with the slider, the plot line changes angle. You can see how the scaling changes across the keyboard by the slope of the plot line. If it's sloped down and to the right, the PEG envelope is low amplitude at the right end of the keyboard and high amplitude at the left end. The Rate Scale controls the speed the envelope runs at. You can also turn the PEG key-off velocity control on and off (the part of the note after you release the key). Not-checked (lower left corner) means it's turned OFF, release is not velocity sensitive.

There are lots of different ways to control vibrato through the PLFO. The Pitch LFO has Velocity actions just like the PEG, the PLFO Depth and Speed are set the same way with the sliders, and on small plots for KeyScaling. The PLFO depth and speed can also be altered using the assigned physical controllers above the KeyScaling plots. There is also a Sync On/Off/PGSync setting and a set of different LFO waveforms you can select from. Above the waveform is a greyed Sync value and slider which I haven't been able to activate. PGSync synchronizes the LFO across several notes when they're pressed at different times, and you can shift the phase of the LFO to change the start point. SyncOFF and SyncON seem to be the same, no synchronization between notes with the LFO start point.

In the Graph window below you can select either the PEG or the PLFO. The Pitch envelope can have five breakpoints before note hold and three after note release. The Pitch LFO has two set points, a start and a stop point. You can delay the start of the LFO (vibrato) and turn it off at a specific time.





In the PLFO block there is a greyed slider and value box in the upper right corner. I've tried everything I can think of, but I can't get them to turn on. You'd think they would be part of the Sync controls, but they won't turn on for any of the Sync selections. I've written Yamaha, but I'm not really expecting them to answer back. I don't think anyone will remember a detail this small. I've had trouble finding anyone with any real knowledge of the VP1 period, let alone minute details like this. This Guide is probably the best source of information on the VP1 anymore.

PEG

Velo Level Sense, Velo Rate Sense	Single values	
Level Scale	Single value,	KeyScaling, higher pitches can produce greater levels.
Rate Scale	Single Value,	KeyScaling, higher pitches can produce faster envelope speeds.
Off Velo	Checked	Turns the Key-Off Velocity Sensitivity ON/OFF (You can change how the note releases with this.)

PLFO

Velo Depth Sense, Velo Speed Sense	Single values	
Depth, Speed	Single values	
Depth Scale,	Single Value,	KeyScaling, higher pitches produce greater amplitudes.
Speed Scale	Single value,	KeyScaling, higher pitches will produce faster speeds.
Sync:	On, Off, PG	PGSync syncs the LFO start points of sounding notes.
Sync Adjustment	Position,	Sets the LFO starting point

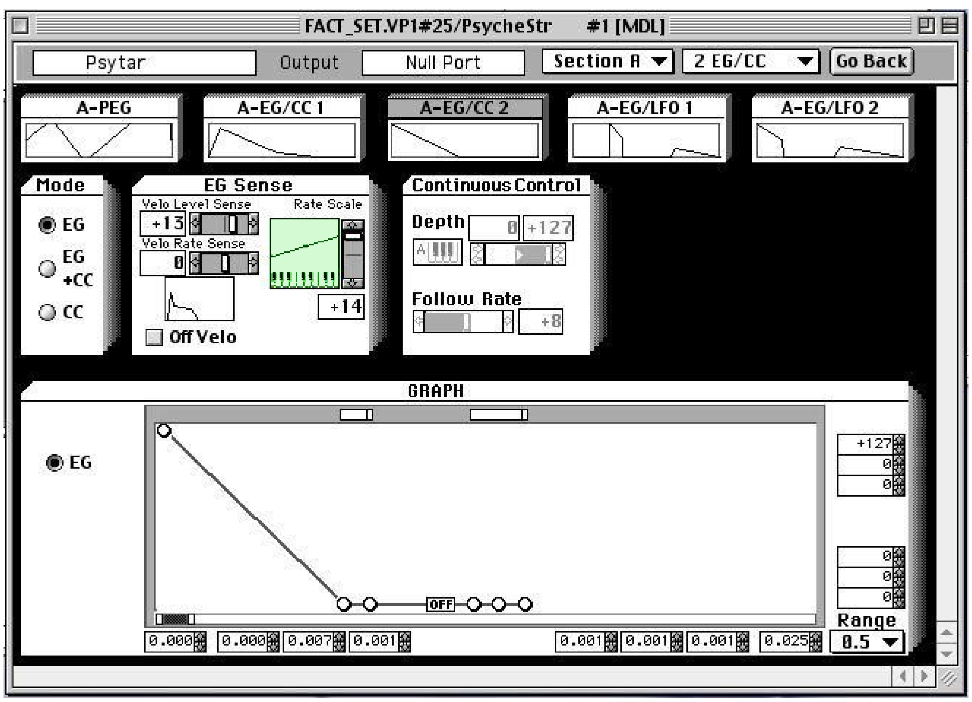
EG/CC 1, 2 ____

Envelope settings are next, along with a Continuous Controller to match. You can select an Envelope Generator (EG), an Envelope Generator plus Continuous Controller, or an Envelope Generator plus Low Frequency Oscillator (next window). Continuous Control (CC) parameters are greyed out below since it isn't Mode selected. The CC Continuous Controller on the EG/CC pages adds an additional controller for the depth of the Envelope Generator. The envelope depth is mainly velocity sensitive, which is controlled by setting the EG Sense values to something other than zero. The Follow Rate allows you to smooth the controllers effect, making it less responsive. EG response can also be Key Scaled by adjusting the slider for the Rate Scale graph, changing how it responds in different sections of the keyboard. (Keyscalable parameters are highlited in Green.) You might want high notes to have a faster envelope than low notes. There are two sets of envelopes and CC's as well, two for section A and two for section B.

The numbers on the right side of the lower Envelope Graph are breakpoint levels, while the numbers across the bottom seem to be rates. The Owners Manual uses Attack Rate, Decay Rate, Sustain Level, and Release Rate for Quick-Edit. Rate slopes don't seem to match the numbers on the bottom of the graphs, though. These may be times in fractions of a second. Note that the black bar on the bottom edge of the window is to the left, allowing all of this very short envelope to be viewed in the window. Use Velo Level Sense or Velo Rate Sense to make notes Velocity sensitive by assigning this envelope to a Mixer volume, and to the Filter Cutoff frequency.

Envelopes on the VP1 are also quite capable. You can designate up to eight break points in time. Do you see the boxes up the right side of the Graph? Each box holds one of the breakpoint amplitudes for the envelope. In this example, there are three break points before Note Off, and three after the key is released. Double click to add a new breakpoint to the graph, to delete breakpoints double-click on the handles. Check the 'Off Velo' checkbox to make the last three breakpoints velocity sensitive, changing the note release.

In the Graph window below the settings you can set up the envelope shape. The envelope can have up to five breakpoints before note hold and three after note release.



Mode

Mode: EG, EG+CC, CC

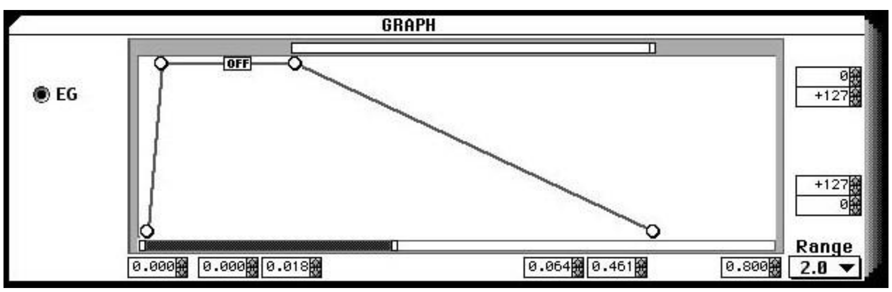
EG Sense

Velo Level Sense, Velo Rate Sense	Single values	
Rate Scale	Single value,	KeyScaling, higher notes can produce faster envelope speeds. Rate Scale (-) slows down the envelope.
(There isn't a Key-Scaled Level Scale)		
Off Velo	Checked	Turns the Key-Off Velocity Sensitivity ON/OFF (You can change how the note releases with this.)

Continuous Control

Depth,	Single value,	Adds an additional offset Controller to the envelope. Depth sets the EG depth using a controller.
Follow Rate	Single value,	Follow Rate can slow down the responsiveness of the controller to smooth the effect.

After playing with the envelope graph windows a bit, I finally figured out how the settings work. The overall envelope shows up in the little window below the label (A-EG/LFO1 in the example above) and changes as you change values on the graph, but the Range button in the lower right corner is a bit confusing. You have to move the sliders on the bottom edge of the window (change the length of the black strip) in order to get the entire envelope visible in the window (see below). Range varies from 0.5 up to 200. I've found that 2.0 is about right most of the time to show the entire envelope. You change the envelope shape by dragging the little circles around, or by changing the numbers on the edges of the graph.

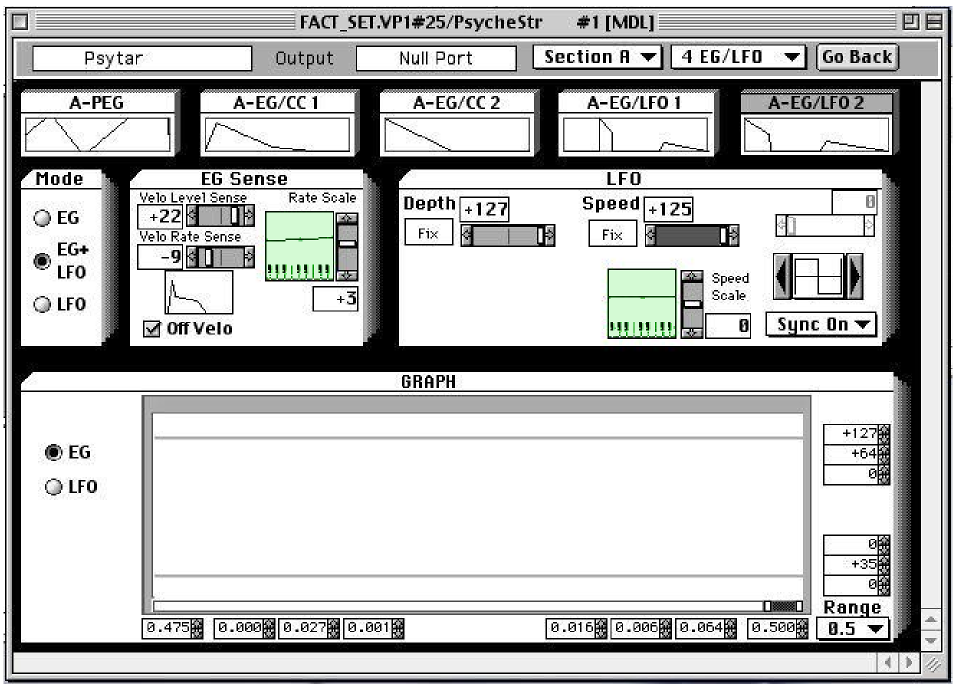


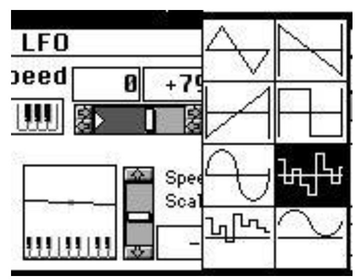
I've found the Envelopes are mostly useful in shaping the note Attack and Release, you don't really need them to control the sound Decay since it's controlled with the Loop Gain in the Delay Loops. The note Sustain is controlled by the 'off' level (ADSR). Place the envelope on the Volume block in the Mixer. The envelope is also useful to clip off the 'click' that occurs in a sound that bypasses the Delay Loops. To remove the 'click' on the first part of a Cycle/Source-only sound, add a 0.015 second rise on the front of the output Volume envelope in the Mixer (or higher than 0.03 to remove the note strike). This one is set at 0.018.

EG/LFO 1, 2 ____

When you select EG all by itself, the LFO window is greyed out, and vice versa. Both EG and the LFO are Mode selected and all parameters are active in the example below. The LFO here also includes the greyed out slider and value box in the upper right corner the PLFO has, the ones I haven't been able to turn on. They may be left over remnants of computer code that were left in by mistake, because I can't get them to work and they don't seem to have a useful function. LFO's behave the similarly to the PLFO, as described in the earlier section, but without the Velocity sensitivity controls. There are two sets of these envelopes and LFO's as well, two for section A and two for section B. Really, there are eight envelopes in the VP1 Element models, not four.

In the Graph window below the settings you can select either the EG or the LFO. The envelope can have up to five breakpoints before note hold and three after note release, for a total of eight. The LFO has two set points, a start and a stop point. You can delay the start of the LFO and turn it off after a number of seconds.





Mode

Mode:	EG, EG+LFO, LFO	

EG Sense

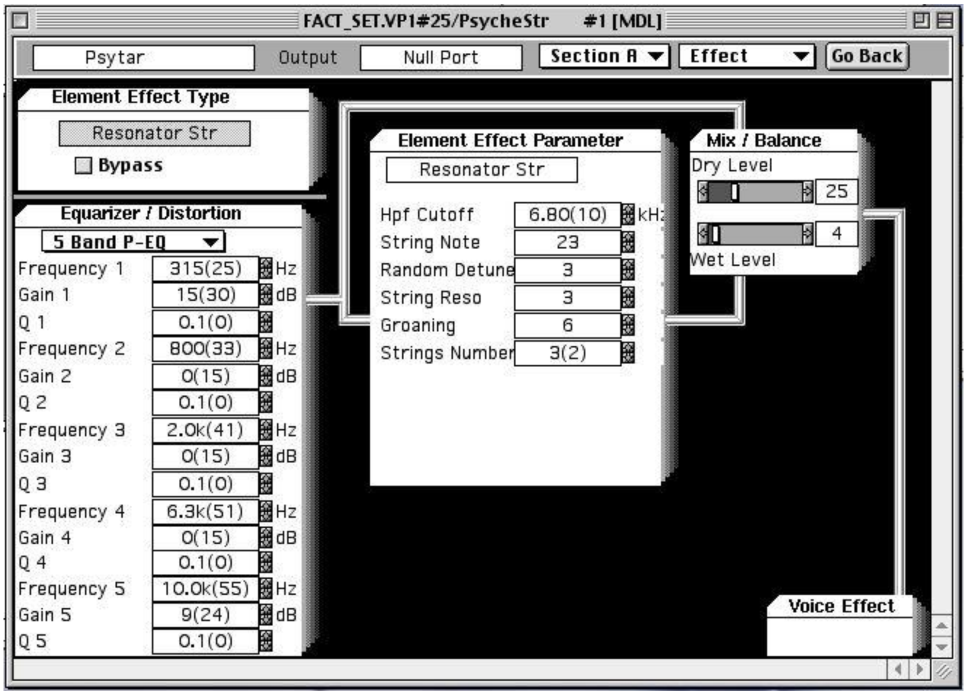
Velo Level Sense, Velo Rate Sense	Single values	
Rate Scale	Single value,	KeyScaling, higher notes can produce faster envelope speeds. Rate Scale (-) slows down the envelope.
(There isn't a Key-Scaled Level Scale)		
Off Velo	Checked	Turns the Key-Off Velocity Sensitivity ON/OFF (You can change how the note releases with this.)

LFO

Depth, Speed	Single values	
Speed Scale	Single value,	KeyScaling, higher notes can produce faster (or slower) rates.
Sync:	On, Off, PG	PGSync syncs the LFO on sounding notes.
Sync Adjustment	Position,	Sets the Sync start point.

Effects ____

Lastly, there is a window for settings for the Element level Effects generator and the output Equalizer, which includes Distortion. We all know how much you guitar players love distortion. The Bypass checkbox is handy for turning the Effects Block on and off during editing. Notice below that these Element Effects are applied before the overall Voice Effects, (in the lower right corner) and are applied to both section A and B of the patch (same Effect on both parts A&B, same settings). The Voice Effect Window is mentioned up on page 11 in this programming guide, which is controlled by the one Quick Edit slider on the console.



Element Effect Type

57 Different Effects:	There are lots of standard Yamaha Effects to choose from.
Reverb Hall	These are similar to the early MU series tone generator XG Effects, but higher quality.
Rev Room 1,2,3	
Rev Stage 1,2	
Rev Plate	
Rev White Room	
Rev Tunnel	
Rev Canyon	
Rev Basement	
Early Ref.1,2	Early Reflection
Gate Reverb	
Reverse Gate	
Delay L,R	Left, then Right output channel
Delay L,C,R	Left, Center, then Right
Stereo Echo	
Pitch Change 1,2,3	This is a new one I haven’t seen before.
Flanger	
Stereo Flanger	

Chorus
Stereo Chorus
Symphonic
Stereo Symphonic
Phaser
Stereo Phaser
Aural Exciter
Rotary Speaker
Ring Modulator

	The next three are interesting, instrument body type resonators.
	I've had trouble with these, applying them after setting up the rest of a patch.
	It may be better to turn these on early before you adjust everything.
Resonator Box	Formants at about 850 & 2400 hz, + some subtle metallic harmonics up high
Resonator Board	A little brighter than Box…
Resonator Str	String Resonator (for a guitar, violin, or cello body) much sharper & brighter than Board
	'Str'ing can create an insane number of harmonics, all the way up the spectrum
	Turn down the 'String Reso'nance, and tune the 'String Note' setting
Echo->Rev	
Flanger->Rev	
Chorus->Rev	The rest of these are combinations of two Effects.
Sympho->Rev	The overall Voice Effect module (vs Element) doesn't include these double Effects

Exciter->Rev
Flanger->Delay L,R
Chorus->Delay L,R
Sympho->Delay L,R
Hall&Plate
Echo&Rev
Delay&Rev
Flange&Rev
Chorus&Rev
Sympho&Rev
Flange&Dly L,R
Chorus&Dly L,R
Sympho&Dly L,R
Flange&Chorus
Flange&Sympho
Sympho&Chorus

Bypass	Checked	Turns OFF the Effect Module.

Equalizer/Distortion

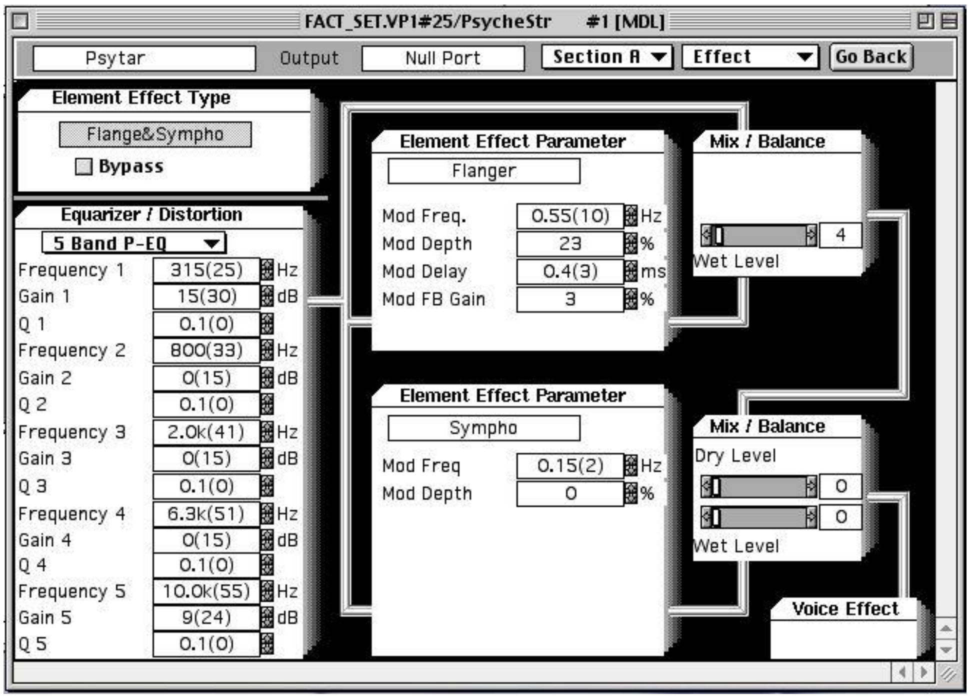
Mode:	5 Band P-EQ, 5 Band EQ, Distortion+EQ	
5 Band P-EQ, Bands 1-5:	Frequency, Gain, Q	These can be used to create 5 FS1R Formant Filters
5 Band EQ, Bands 1-5:	Frequency, Gain	
Distortion+EQ:	Type, Drive, Tone, Dist Mix Offset	
	Low Shelv.Frq, Low Gain	

Mid Frq, Mid Gain
Hi Frq, Hi Gain

Element Effect Parameters (Doubled for two Effects)
Each one has different set of parameters.

Mix/Balance
Dry Level, Wet Level

When you select a double Effect like Flange&Sympho, you get two Element Effect Parameter windows instead of one.

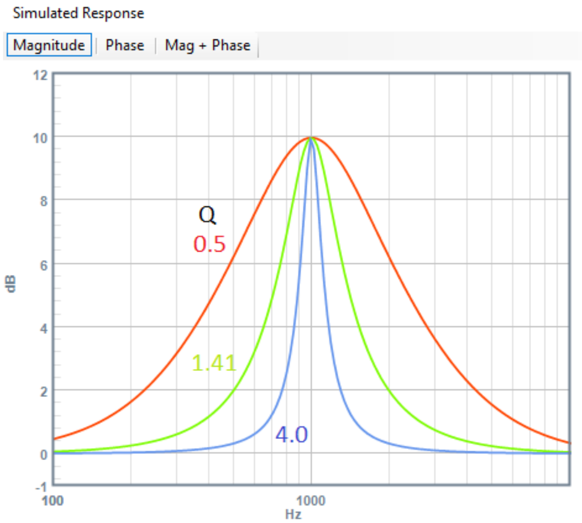


Equalizer/Distortion

Now we get to the final module in the VP1, the Equalizer. The VP1 has one final trick up it's sleeve, one that I just figured out. Notice that the 5-Band P-EQ settings above allow you to set the center Frequency, Gain, and the Q of the filters? (Q is just a bandwidth setting.) With these you can create FIVE **Formant** filters just like you see in the Yamaha FS1R. The FS1R has eight, but you can do a lot with five. Looking back at my FS1R patches, I generally used about six Formants, so with a little bit of creativity I should be able to do just fine with five. This is a phenomenal feature! Before this, the VP1 had a gap in it's capability when it came to modeling the resonant body of an instrument. This fixes that. Look in my FM Programming Guide in the back to find the section on using Formant filters in the FS1R.

Distortion mixed with a 3-Band Equalizer is also available, along with a simpler 5-Band Equalizer with the Q settings set for you, probably set to the default of 0.1.

To get some guidance on how to set the 'Q' of the filters, see the figure below. Setting it to 0.1 is the default shown above, which gives you very wide, overlapping bands with the filters. Use a higher value to narrow the filter to create an instrument body Formant. Coupled with matching Gain settings, you should be able to match the characteristics of most instrument bodies. Acoustic instruments with dominant Formants include the Violin, Cello, Bass, Bassoon, Mandolin, Acoustic Guitar, Viola and English Horn, to name the ones I've tested. Use a spectrum analyzer to set the Gains.



In equalizers, Q is the ratio of center frequency to bandwidth, and if the center frequency is fixed, then bandwidth is inversely proportional to Q—meaning that **as you raise the Q, you narrow the bandwidth**.

Conclusion

I learned a lot by studying the VP1, I can see where the timbre of the sound is being altered by multiple controllers in ways I can't hope to replicate on any other instrument. The VP1 is one of a kind, I feel very privileged to have access to one. You might be wondering why anyone would dig so deeply into a synthesizer that is so obscure. Well, studying the best examples I think is the best way to learn, and Yamaha is definitely one of the best. The structure of the VP1 has taught me how a virtual acoustic synthesizer is constructed from a block diagram level. As an engineer I can appreciate what goes into the design far better now than reading words in a research paper. I will also be able to apply what I've learned to my other synthesizers. I have a Yamaha FS1R, a Technics SX-WSA1R, and a Nord Modular G2 that some of this will apply directly to.

All in all, I really like the Yamaha VP1. Maybe someday Yamaha will produce a VST or Soft Synth modeled after the VP1 so everyone can all play around with the VP1 for real. The VL1 however, stands a better chance of being resurrected, maybe even in the Reface format.

Until then, **Cheers!**

Appendix A

Inharmonicity: Fractal Harmonic Fans

The Yamaha VP1 can create fractal, expanding sets of harmonics. Yamaha had a strong reason for programming the VP1 this way, let's take a look. In Figure 1 below you, can you see how the harmonics spread out wider and wider apart as you go up in frequency? This is the exaggerated Inharmonicity effect the VP1 can create when you apply the Stiffness parameter. This is very unusual, and imparts a very metallic tone to the sound, a lot like an Asian Gamelan.

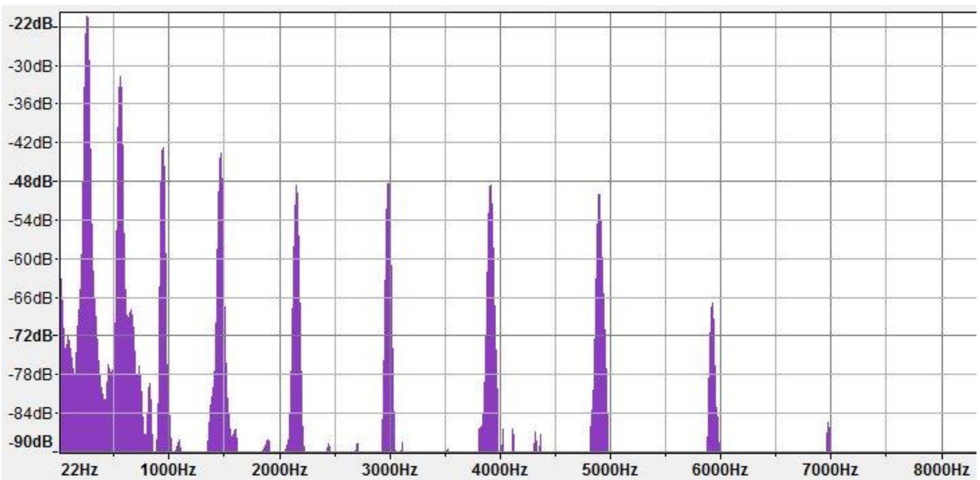


FIGURE 1. Yamaha VP1 HarpBell Patch

Most wind instruments like the Trombone do NOT have expanding harmonics. They create very evenly spaced harmonics when they are played, as shown in Figure 2 below. See how nice and evenly spaced the harmonics are? The two graphs show the same note pitch, but are very different in character.

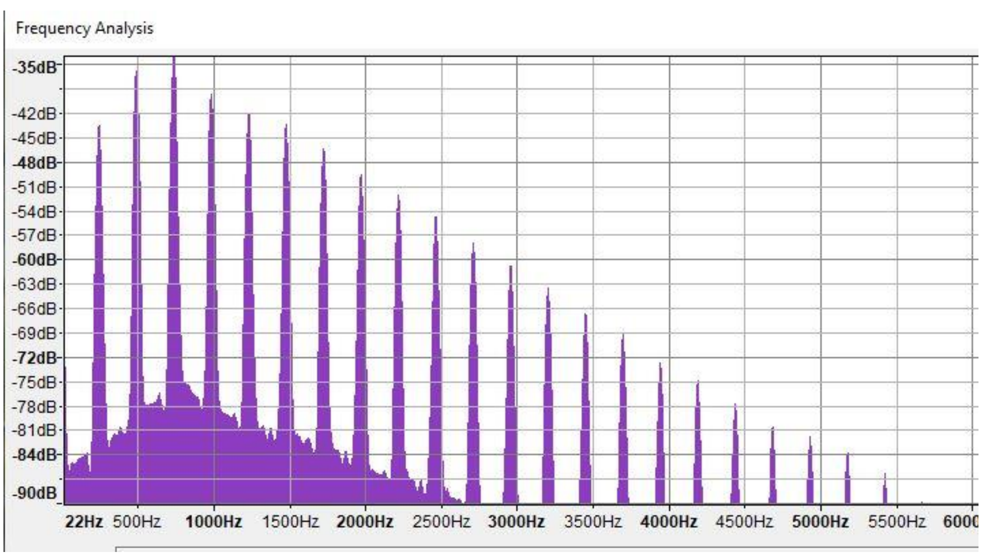


FIGURE 2. Trombone Harmonics

I didn't think any acoustic instruments were capable of Inharmonisity, but I wasn't sure, so I went off to check using a collection of instrument recordings and a Spectrum Analyzer. After analyzing all of the instruments in a standard orchestra I found that a number of the instruments actually DO create harmonics like this.

Let's take a look at a different instrument, the Piano, which is somewhat inharmonic. A plot of a note near middle C is shown below in Figure 3. If you look carefully, you may be able to see that the harmonics on the far right side of the plot are slightly further apart than the ones on the far left side. The spacing increases by about 1% between each pair of harmonics. By the time you get up to the 20th harmonic the spacing is about 20% wider than the spacing between the first two harmonics. The Piano is the most extreme acoustic instrument I analyzed. The harmonics are out of tune sharp all the way up the keyboard. It's easier to see the expansion if you measure the frequencies of the harmonic peaks in a Spectrum Analyzer and compare the harmonic ratios in a spreadsheet.

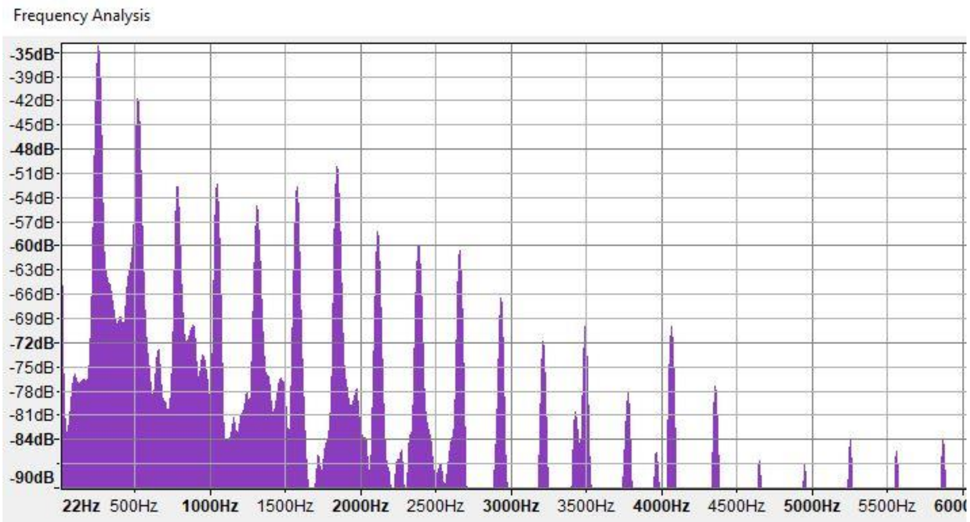


FIGURE 3. The Piano

If you look online about the piano you'll find reference to the Railsback Curve, Figure 4, which shows how a Piano is tuned. It shows graphically the increasing sharpness in the tuning as you go up the keyboard. Varying stiffness in the strings due to mass and tension affect the properties of the sound. This is probably the reason electric piano patches sound so different from a real Piano. They try to use evenly spaced harmonics to mimic a real Piano when they should be using a fractally expanding fan.

The Piano is tuned this way to help all of the notes and harmonics to be slightly more in tune overall. Low notes are tuned flat so the harmonics of the low notes match the pitch of the middle register notes. Upper notes are tuned sharp so they match the harmonics of the middle register notes. All of this works together and sounds more in tune to your ear.

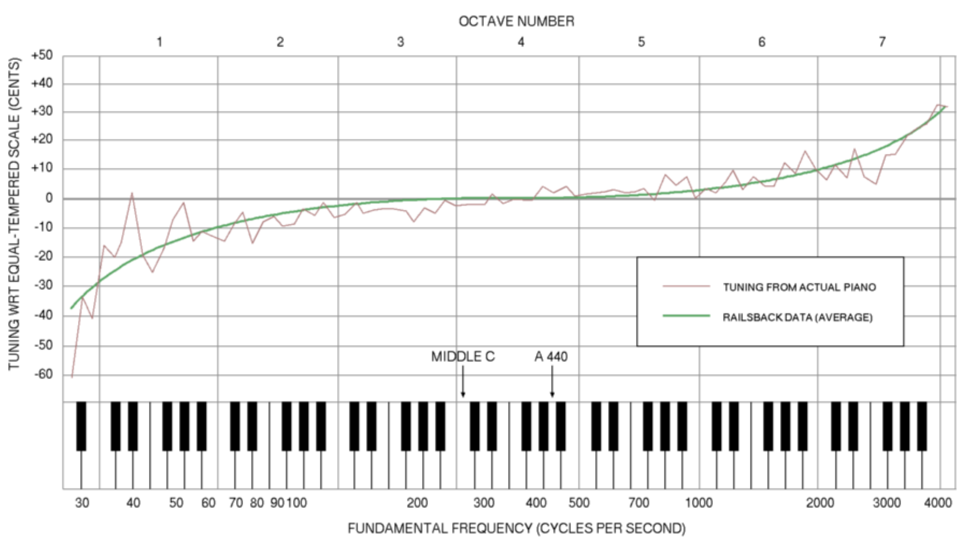


Figure 4. The Railsback Curve

The next set of instruments that exhibit this type of inharmonic behavior are the stringed instruments, the Violin, Cello, Bass, and oddly the Mandolin. These instrument's harmonics are all sharp about 55% the amount of the piano, but only in the lower register. High note harmonics are all in tune, with only about 1% error compared to the Piano.

The next instrument down in sharpness is the Bassoon at about 34% of the piano. You can't really see the expanding harmonics on the plot, you have to go in and measure the frequency peaks using a Spectrum Analyzer.

One instrument that surprised me was the Guitar, which exhibited only about a 6% expansion in the harmonics compared to the piano. I expected it to be similar to the Mandolin, but it was much more in tune than expected.

Two instruments were actually flat (vs. sharp) which was another surprise. The Viola and English Horn harmonics were about 20% flat compared to the Piano in the lower register.

I also analyzed human singing voices in different registers. Singing voices are all over the map, some are sharp in the lower register, some are flat, but all were in tune for the high notes.

Every instrument except for the Piano was in tune in the upper register. The Piano isn't since it has multiple strings for each note, and the ganged strings are generally slightly out of tune to each other.

The instruments which were in tune all the way across the entire spectrum (no Inharmonicity) were the wind instruments, including the Flute, Oboe, Clarinet, Saxophone, Trumpet, French Horn, Trombone, and Tuba.

All of the tubular wind instruments have nice, evenly spaced harmonics like the Trombone, which is what you would expect from a tuned wave guide. These all behave nicely and are fairly straightforward to model using evenly spaced harmonics. Frequency Modulation (FM) works great for this, as do standard delay loops (with no stiffness control).

The Very Odd Yamaha VP1

Getting back to where we started, Yamaha had a good reason to model Inharmonicity in the VP1. The Piano in particular has this characteristic, and has been exceedingly difficult to model over the years without resorting to recorded samples. The best Piano synthesizers today like the Yamaha Motif series and now the Montage line are based on recordings of the best Pianos available. The BigMyPiano patch on the VP1 is a great synthetic piano patch by the way, but it's not the patch people remember.   Nobody seems to care anymore that the VP1 could natively model the piano, the string section, guitars, and the mandolin extremely well. Everyone zeros in on the very inharmonic patches.

Yamaha added expanding harmonic fans into the VP1 using the 'Inharmonisity' parameter. Analyzing the VP1 patches on a Spectrum Analyzer, the VP1 goes out of tune as you go higher up the VP1's keyboard on a lot of factory patches. The internal model is for a single string and doesn't compensate for using a full set of strings in an instrument, with decreasing Stiffness as the strings go higher in pitch. Because of this, it actually behaves backwards to an acoustic instrument, more in tune down low, more out of tune harmonics up high. You can however compensate for this in the settings.

In one extreme VP1 patch, the HarpBell patch, it starts in tune as a nicely plucked string in the lowest register, then transitions into an inharmonic bell-like sound part-way up the keyboard, then it hits a distinctive inharmonic Gamelan sound above middle C, smoothing out as you go higher into a water glass tone as the harmonics slide back almost into tune again. Most of the VP1 patches are not quite this extreme. I do find that most of the inharmonic patches are metallic in nature though, which is what you get when you exaggerate the expansion in the harmonics. Our ears don't really parse Inharmonicity all that well. We interpret everything inharmonic as a metallic sound. I find that very interesting. We can easily label the acoustic instrument a harmonic patch sounds like, but we can't really do that with inharmonic patches.

Personally, I would have tuned the VP1 algorithm in the other direction, so it behaved more like other real acoustic instruments, getting more in tune as you go higher. But, it does make the VP1 very unique. The patch designers left it alone, despite making the VP1 behave unlike any real instrument. It definitely makes the patches sound more interesting than making it behave like an ordinary acoustic instrument, don't you think?

Appendix B

The Programming Interface

The Yamaha VP1 was programmed in the factory through a ribbon cable using a Sun Sparc Station computer, using a hardware interface called OTOMI which had a touch screen. For outside patch programmers, Yamaha created a Macintosh computer-based editor which is the one I've been using. To interface it to the Mac the engineers used a Mac modem cable and connected it to a serial to parallel converter, then connected the converter to the VP1 using a ribbon cable. Not all VP1's have this interface, at least one does not.

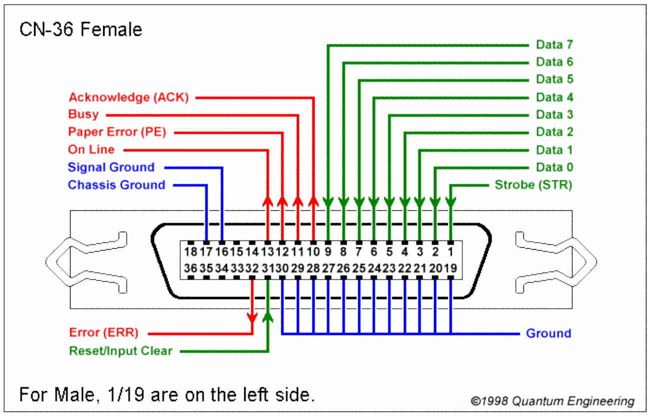
There is a major problem with interfacing it this way on the Mac. The Macintosh computer actually uses an RS-422 serial interface, not an RS-232 interface. RS-422 runs on voltages from 2 to 6 volts, where RS-232 runs on voltages from about 5 to 15 volts. There is some overlap, but different devices run at slightly different voltages so this isn't very reliable. All of the connection problems and losses of connection while editing are caused by this. Yamaha probably did it this way because there really wasn't a better option. The interface was designed before USB ports were introduced, and the early Macs didn't have standard serial ports.

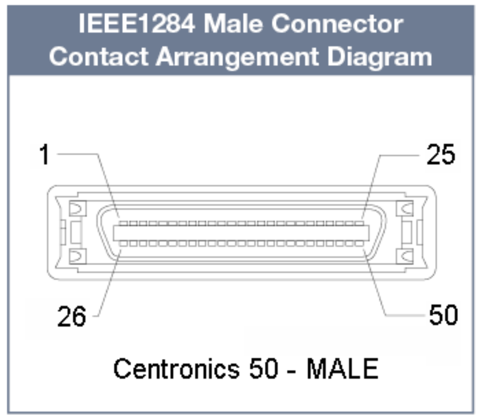
I've found a better way using a **Tripp Lite Keyspan USA-19HS USB** to RS-232 serial port (the black unit on the right below). It emulates the Mac Printer port and outputs full voltage RS-232 for the serial-to-parallel converter. These are readily available from numerous vendors. Next in line is a standard serial DB-9 female to DB-25 male adapter. Then for the serial-to-parallel converter, a somewhat available **Inmac SP-16 Serial-To-Parallel Converter** works just fine. I couldn't get the ATEN 325A in the photo in the first section to work at all, that's the one that came with Reinhold Heil's VP1. I think they changed something internal to the original ATEN serial converters, which is why Manny's works and my after-market one doesn't. Last is the ribbon cable to connect to the VP1 which is detailed below. The wiring is different from the standard parallel setup at the serial to parallel converter. I suspect Yamaha altered the ATEN converter internally and changed the wiring, which is why I can't get it to work at all, and why the ribbon cable is wired oddly. Manny has an interface that works, one that originally came from Yamaha. I think they altered things so the ordinary musician couldn't just go out, buy a few parts, and create his own programming interface. Yamaha wanted you to buy one from them. I'm sure it costed quite a bit back in the day.

Inmac SP-16, DB9 F to DB25 M Adapter, then the Tripp Lite USA-19HS external printer port.

And, I just found and tested another serial-to-parallel converter that works! The **BlackBox PI015A Converter II** is another and there are three available right now on ebay, which is a good sign since there aren't any Inmac SP-16's. It required a male-to-male gender changer to match my Inmac ribbon cable, on the CN-36 end of it. Otherwise, it's a drop-in replacement for the Inmac.

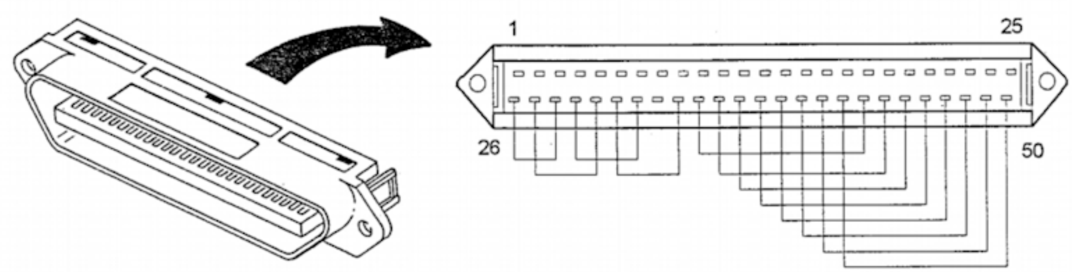
The Ribbon Cable is a bit more complicated, it needs to be wired as outlined in the table two pages down. It goes from a Centronics 36 pin connector (female to match the Inmac SP-16, male to match the BlackBox PI015A) to a male Centronics 50 pin connector to connect to the VP1.





● CONNECTOR PIN ASSIGNMENTS FOR MIGS INTERFACE TEST

(MIGSテスト用コネクターのピン配列表)



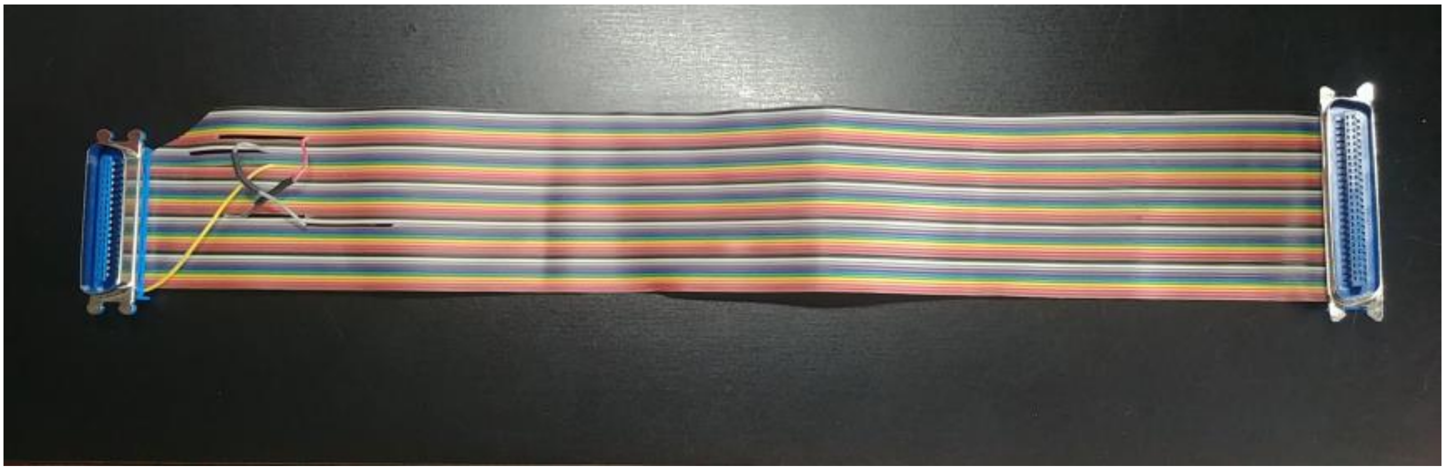
PIN	DESTINATION	SIGNAL	REMARKS
26	PIN 34	MIN0	PARALLEL DATA0
27	PIN 35	MIN1	PARALLEL DATA1
28	PIN 36	MIN2	PARALLEL DATA2
29	PIN 37	MIN3	PARALLEL DATA3
30	PIN 38	MIN4	PARALLEL DATA4
31	PIN 39	MIN5	PARALLEL DATA5
32	PIN 40	MIN6	PARALLEL DATA6
33	PIN 41	MIN7	PARALLEL DATA7
42	PIN 45	STBO	STROBE
43	NOT USED	BUSYO	BUSY
44	PIN 47	DAKO	DATA ACKNOWLEDGE
46	PIN 49	BUSYC	BUSY
48	PIN 50	VPON	POWER ON SIGNAL
1 - 25	NOT USED	NOT USED	–

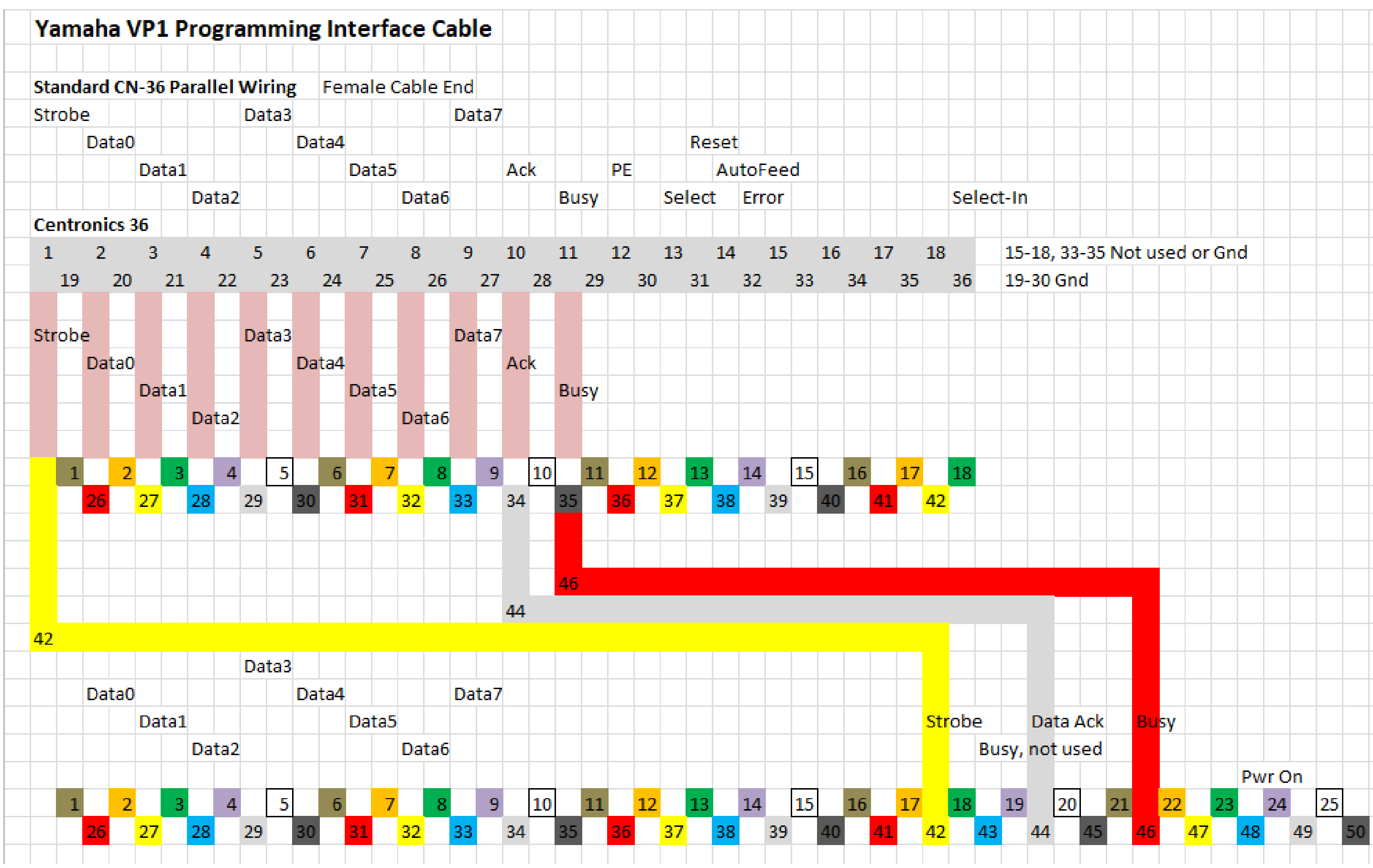
＊ The part number of the connector that is used for this MIGS test is "VA250600."
＊ MIGSテストには、部品番号VA250600のコネクターを使用してください。

Page 86 in the Yamaha VP1 Service Manual (Thanks Manny!)

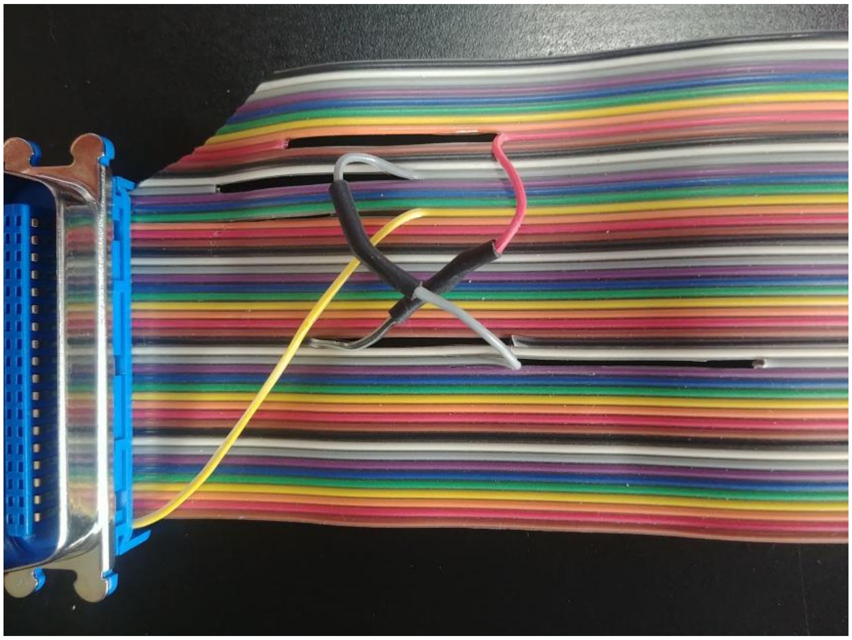
You can see Yamaha changed the wiring at the VP1 from a standard parallel setup to a custom one. Standard can be seen above on the 'CN-36 Female' diagram. The test connector above might be why they changed the wiring. Notice how all of the jumper wiring is all on the same side of the connector.

Below is a photo of one of the new interface cables. It matches the BlackBox converter, but not the Inmac. The Inmac cable has a female CN-36 connector on the end of it, this one is male. The CN-50 connector is wired standard, the CN-36 connector is not. The yellow wire (CN-50 pin #42) crosses over to pin 1 on the CN-36 connector. To correct communication problems, CN-50 pin #44 needs to be rewired to CN-36 pin #10 (Data Acknowledge) and CN-50 Pin #46 goes to CN-36 pin #11 (Busy signal) which the cable below has wired jumpers for (corrections on my part to match the Yamaha Service Manual diagram). I made the corrections in the spreadsheet below to get a very clean ribbon cable, much cleaner than the original. I wrapped the three wire jumpers with black electrical tape to keep them tucked in.





If this spreadsheet is confusing, email me and I'll try to explain it better. This wiring is corrected to match the Yamaha Service Manual diagram and is much cleaner than the original cable. When you crimp the CN-50 connector onto the ribbon cable, make sure it's crimped all the way. If you can see silver between the ribbon and the housing, it's not crimped enough. The cable won't work otherwise. It's a real pain to check the pins and wiring when it doesn't work.


Here's a close-up of the wiring changes.

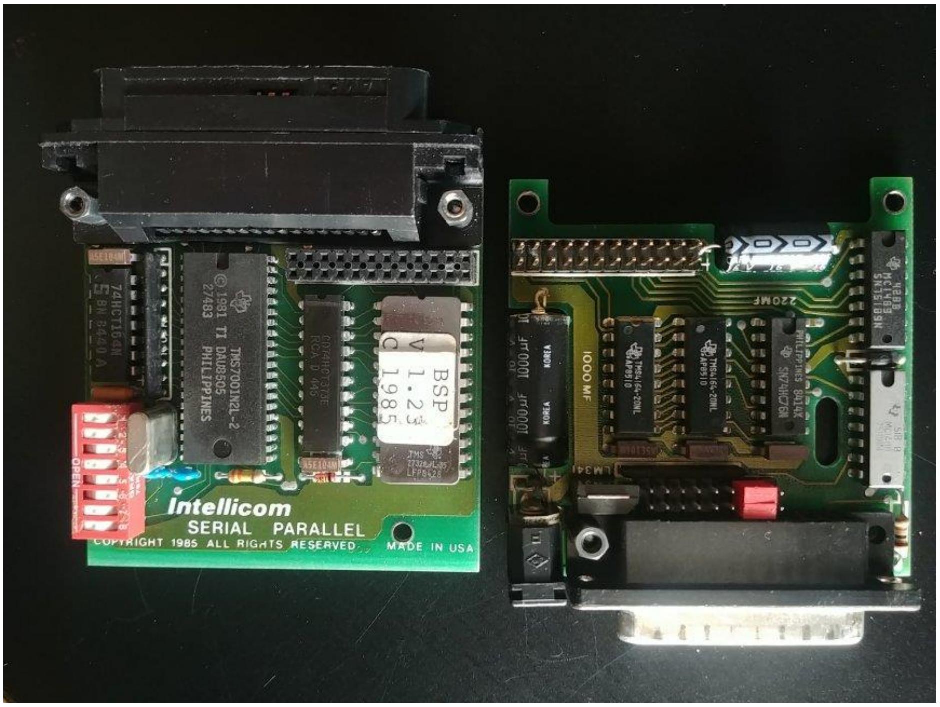
None of my interface is the same as the original. The computer is sending signals out a USB port, not the Mac serial port, I used an external aftermarket serial port, the serial to parallel converter has to be a different brand, and the ribbon cable has been rewired. Basically I had to redesign a completely new interface.

Converter Settings

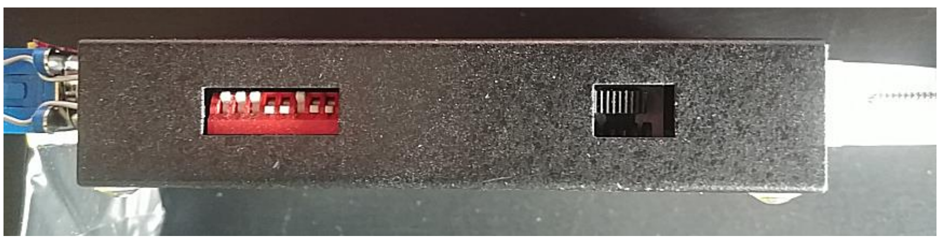
The next step is setting up the serial-to-parallel converter. So far, I've been successful with an **Inmac SP-16** and a **BlackBox PI015A** Serial-to-Parallel Converter. The two converters mentioned in the 'VP1 Editor Connection Kit' document are no longer available and I couldn't get the ATEN SXP-325A or ATEN SXP-320 to work either. Luckily the Inmac SP-16 and the BlackBox PI015A work just fine and are somewhat readily available. I had an SP-16 on hand, which is why I tested it out, a superbly lucky coincidence! The settings are as follows.

Appletalk OFF
Printer Setting in the Editor, 9600 baud. My setup only allows Printer Mode so far.
Serial-to-Parallel Converter, set to 19,200. Select the 'Double Baud Rate' in the Keyspan Control Panel.
(The Modem Setting allows 38400 baud, which is not available with the Tripp Lite. Manny gets 38400 and 19200 on Modem but I can't get the original ATEN converter to work at all.)
Xon/Xoff
8 bits, 1 stop bit
None Parity
Serial to Parallel
DCE

The Inmac SP-16 has to be taken apart to get to the dip switches and jumpers. You can see their settings below. The dip switches are in the lower left corner of the photo in the red block, and the two red jumpers can be seen in the lower right corner of the upper circuit board. The BlackBox converter has external dip switches, much easier to set up.



Inmac SP-16 Settings


BlackBox PI015A Settings

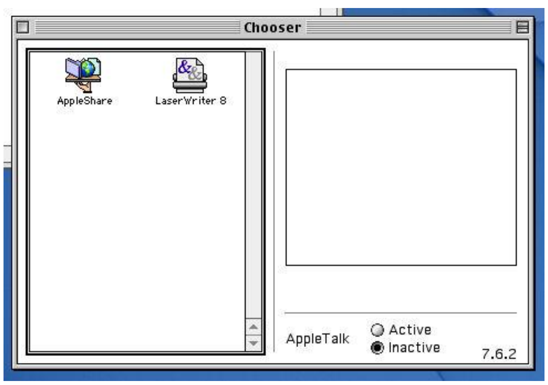
Below is what the setup looks like with the BlackBox PI015A Converter and the Tripp Lite LED side up.

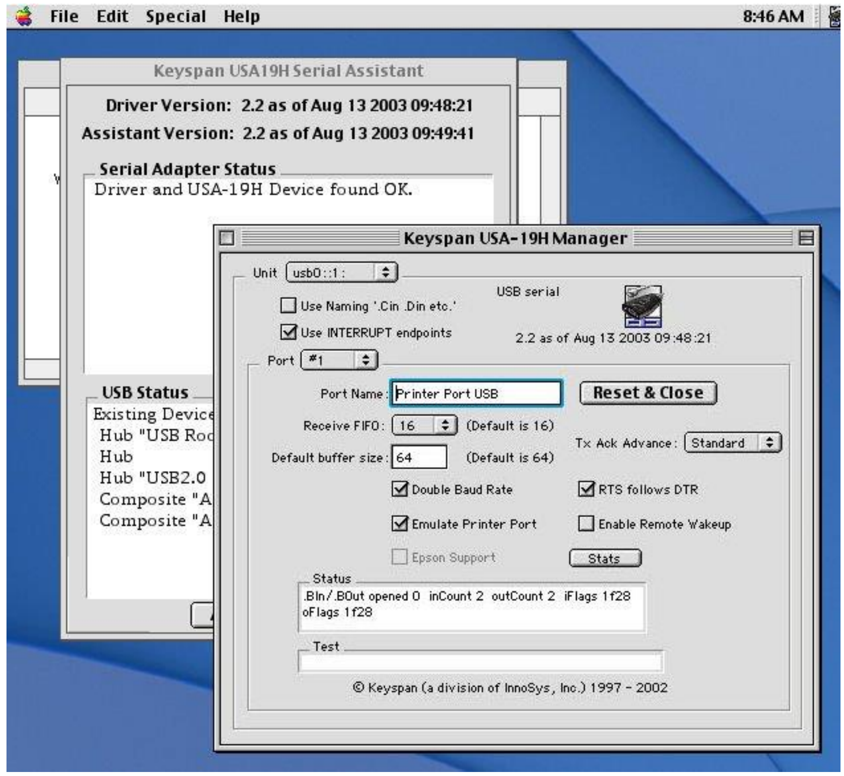
If you can't find either of these serial-to-parallel converters, I would guess that a similar Inmac or BlackBox unit would also work, as well as different brands I haven't tested (I don't recommend ATEN unfortunately).

You need the Mac OS8/9 driver for the Tripp Lite Keyspan USA-19HS. I found it on the WayBackMachine on the archived Keyspan website from back in the 90's, and it's also archived here: http://www.fosh.com.au/article/keyspan-device-drivers I also had trouble unpacking it, I had to get an OS7 version of MacBinary on the www.macintoshrepository.org website to get it to open properly. If you get stuck, send me an email and I'll send you the unpacked driver install file. Thor276@yahoo.com

On the Mac you need to turn Appletalk OFF, which is on the apple menu as 'Chooser'. See the figure below on the left. The Tripp Lite Keyspan driver uses the default settings, as shown below on the right. It shows up on the apple Control Panels menu as 'Keyspan USA19H Serial Assistant'. Select 'Emulate Printer Port', 'Double Baud Rate', and turn off the printer port in any other serial port Control Panels you have running. (9600 x 2 = 19,200!)

All of the parts to make an interface add up to about $130, then it takes several hours to make the ribbon cable and test everything out. An electronics repair shop should be able to make you a new ribbon cable for about $150, then you need to install the driver for the Tripp Lite on your Mac and check a few settings.





OK, I think that's everything you need to get it running. I wanted to document all of this so nobody else has to struggle for a month+ getting it working like I did. I used a couple of serial port testers to troubleshoot the communications, most people don't have those. This new hardware setup is super stable and connects properly every single time with no errors or dropouts. I've also made a second setup for another VP1 owner in Germany, so I know I've got the wiring diagram right.

VL1 Editor: I also attempted recently to install a USB midi port and driver on the Mac so I could run the VL1 Editor on the same machine. The MidiSport driver I have interfered with the Tripp Lite printer port setup, killing my VP1 interface. I uninstalled the files manually and restored my VP1 interface. Oh well. It was worth a try. I have other ways to edit VL voices anyway.

Macs That Work

The round Apple printer/serial ports are not the ideal voltages. I tried it first until I realized it was RS-422 and the wrong voltage range. I have no idea how the original factory interface can even operate like this. It takes a USB port to correct it. I use a Blue & White PowerMac G3 with OS 9.2.2 installed, which has the necessary USB ports. I've also tested a Mac Mini G4 in dual boot mode. On the G4's you will need to snip out a 10 to 20 GB partition out of the OS X partition, which can be done using the Disk Utility app in OSX 10.4.6 or later. You can also get to it by booting off an install CD and selecting Disk Utility off the top menu at the beginning of the install. Re-partition the drive in OSX, then install OS 9.2.2 into the new, smaller partition. Dual boot by holding down the 'Option' key during startup. I've tried using the Mac Mini running OS X 10.4 with OS 9 Classic Apps capability, but I can't get the Classic side to use the Tripp Lite as a Printer. The Tripp Lite driver doesn't run properly. Dual boot is the working option, but it's very difficult to set up. Just use OS 9.2.2 if you can't get OSX to work.

The Mac's that work need three things: 1) G3 or G4 Power PC Processor 2) Run OS 9 3) USB Ports

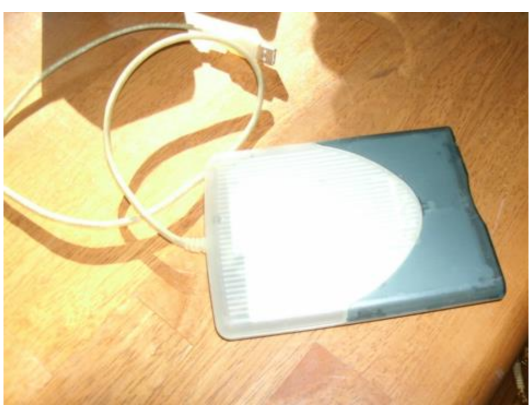
PowerMac G3, only the Blue & White case have USB ports
PowerMac G4's, up to the 933's (not the 1.0's or higher)
eMac G4, 700 or 800
iMac G3's
iMac G4, 700
iBook G3's
PowerBook G3, 400 or 500
PowerBook G4, 400 Ti - 1.0 Ti
Mac Mini G4 With a special copy of OS 9.2.2 (see below)

There may be additional Macs that work too.

To check the specifications yourself, I've found this site works really well. Check the Macs with G3 and G4 processors, these are the ones that have USB ports and run OS 9 natively.
https://everymac.com/systems/by_processor/

If you want a reliable machine with a warranty, try here. (I have no connection to these guys.) They do ship internationally. They also have all the other Macs including the laptops on the list too, you just have to decide what you like best. https://www.usedmac.com/products/apple-powermac-g4/powermac-g4-733-digital-audio-512mb-40gb-cdrw-pre-owned

To get the special Mac Mini G4 OS 9.2.2 Install Disk .iso, go here. They also support other unsupported G4's.
http://macos9lives.com/smforum/index.php?topic=4365.0



USB Floppy Disk Drives

It takes a special floppy drive to work correctly with the older Macs, some work, some don't. None of the newer G3 and G4 Macs came with a floppy drive, which you need to transfer patches to the VP1.

Mine is an external **USB SmartDisk VST Floppy Drive** model number **FDUSB-M**.

Running the Editor:

When you double-click on the Editor icon it launches MIGS Mode which shows up immediately on the VP1 Display, then you select a patch bank. (No, I don't know what MIGS stands for, sorry.) The Keyspan serial-to-parallel converter's green LED will go from flashing once per second to a steady light when the connection is made after a patch bank is selected.

Troubleshooting: If it doesn't immediately go into MIGS mode, close the Editor down all the way, unplug the Inmac or BlackBox converter, plug it back in, wait a few seconds, then launch the Editor again. Sometimes the serial device needs to be reset by cycling the power. If that doesn't work try cycling the VP1 off, then back on. The VP1 display should change to MIGS *immediately* when you relaunch the Editor. If it still doesn't work, check all your connections and make sure Appletalk is OFF. Check your Control Panels/Keyspan_USA-19H_Serial_Assistant/Advanced settings against the picture in the previous section.

Next you select a patch by double-clicking on one, which immediately loads the patch name onto the Display, while it continues to load the patch into memory. The Keyspan flashes quickly for five seconds as commands are sent to the synth. It is essentially executing a Bulk Dump to the synth, for those of you familiar with sysex commands. However, Yamaha never got around to making this available through the midi interface.

You can now play and edit the patch you've selected. Each time you make a parameter change the Keyspan light will flash once as the command is sent to the synth. You can also monitor the diagnostic window (Window Menu, first item) and see the hex commands as they are sent to the synth. This will enable someone to create a new editor on a modern computer someday.

Saving a patch from the Editor saves it onto the computer, not to the synth. Saving it to the computer, you can load it into the VP1 later using the disk drive.

When you Quit the Editor, it closes MIGS mode and puts you back into Play mode. After closing the Editor, the Keyspan goes back to flashing once per second. The patch you were editing (some of the time) is still in the edit buffer on the synth to play, and load into a memory slot. On mine it works about 50% of the time.

I think this is what you have to do for it to remain loaded on the synth…

With MIGS mode off, load the patch bank you want to edit onto the synth using the floppy disk drive.
Select the patch you want to modify using the buttons on the synth.
Start the Editor on your Mac to enter MIGS mode and select the <u>same patch</u> from the computer.
Both patch numbers need to be the same so it doesn't restore a different patch when you exit MIGS mode.
Edit away.
When finished, save it to the computer.
Exit MIGS mode by closing down the Editor.
Check to see if your edits are still loaded in the synth.
If it worked, you can save the patch to a memory slot in the synth using the store buttons.

Best of Luck!

Afterword

If your display looks like the photo below, you can buy a replacement display online. Below is the VP1 in Hamamatsu Japan, on display at Innovation Road. It really needs a new display…

The sole VP1 display vendor in the United Kingdom on ebay and Reverb modifies the display so it fits in properly, and adds a long ribbon cable just for the VP1. When you replace the display, you can also remove the back-light and it’s separate voltage supply. Replacing the display is a real chore, you have to practically disassemble the entire synth to get to it. When you flip the VP1 upside down to work on it, the display is under everything else… It took me several days to accomplish since I was being very careful to get the synth back together correctly. It looks great afterwards though! See the photo below.

Several different color combinations are available, mine is white text on a blue background.

It looks great!

One final note… My VP1 came with an original, matching stand. It's the Yamaha L7S or L7B (Tyros) stand with the front two columns painted a gold color to match the finish on the VP1. They sell for about $250 on eBay, occasionally for less. It's very stable and handles the weight of the VP1 quite well.

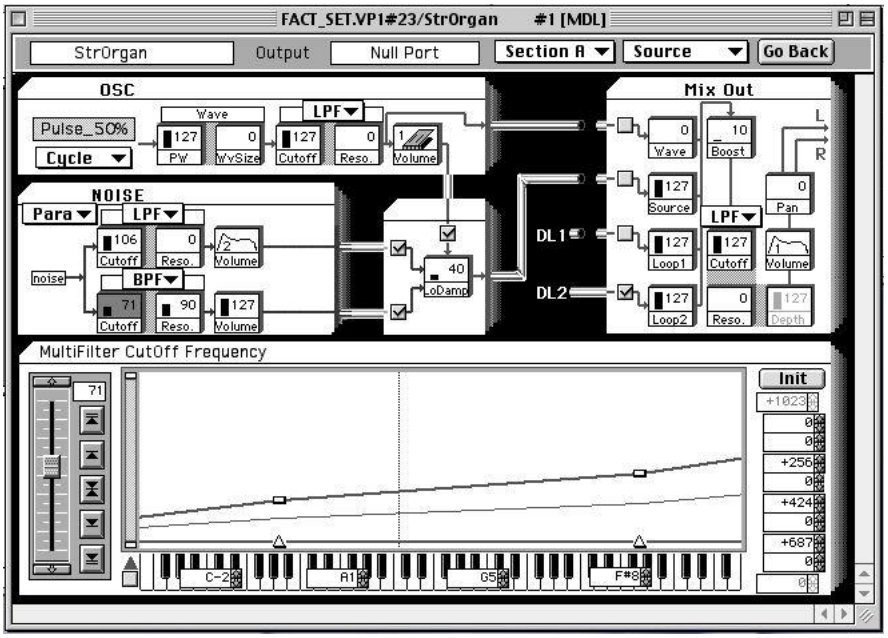

Appendix C

Contains Data Sheets for detailing patches.

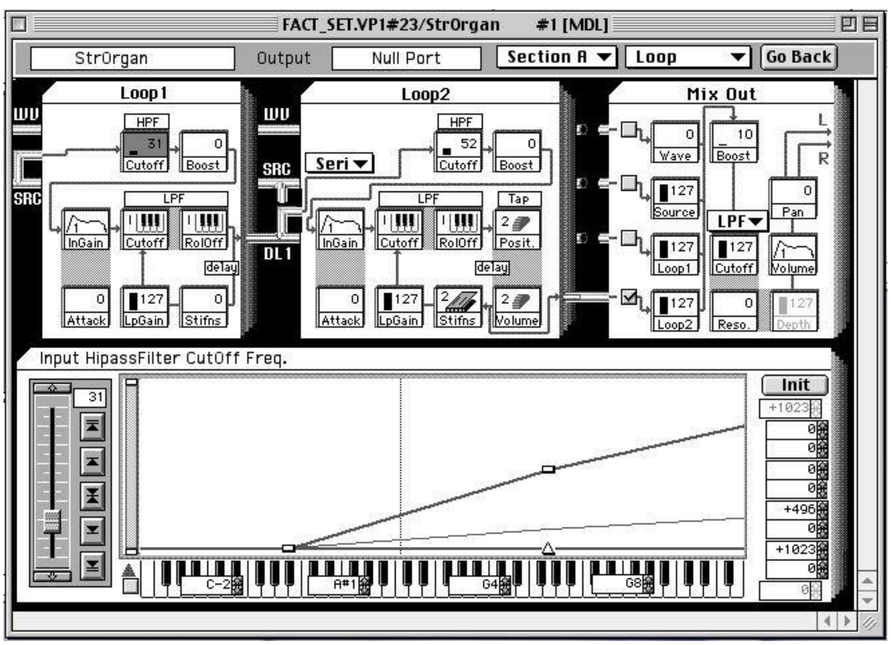
MDL is on the first two pages, TWIN is on the second two pages.

These don't capture every parameter in the synth, just the main ones in the models.

Wave A		**Name**	
Wave B			

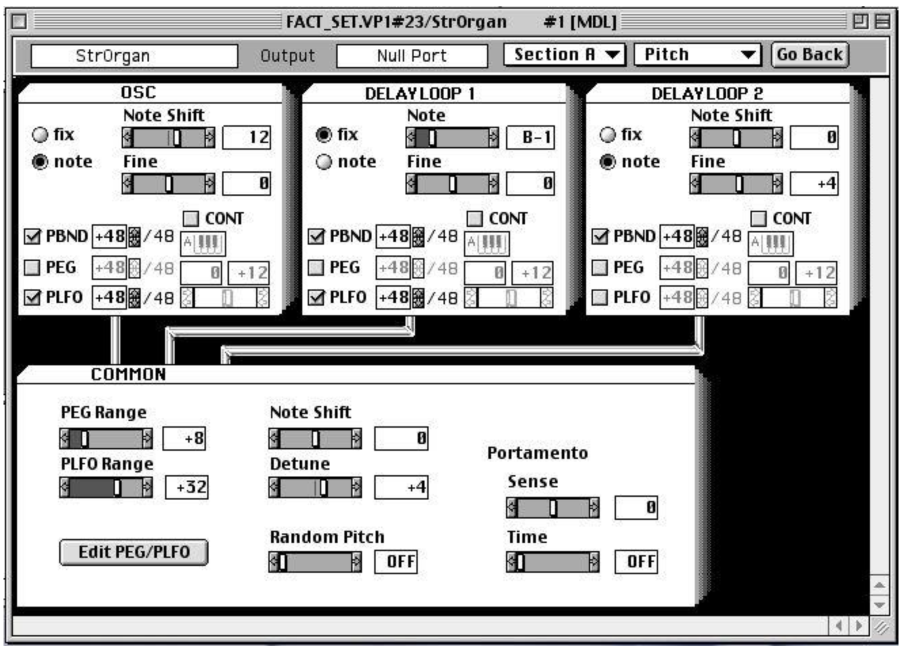


OSC	Mode		
	PW		
	WVSize		
LPF	Type		
	Cutoff		
	Reso		
	Volume		
	Para/Ser		
Noise1	Type		
	Cutoff		
	Reso		
	Volume		
Noise2	Type		
	Cutoff		
	Reso		
	Volume		
	LoDamp		



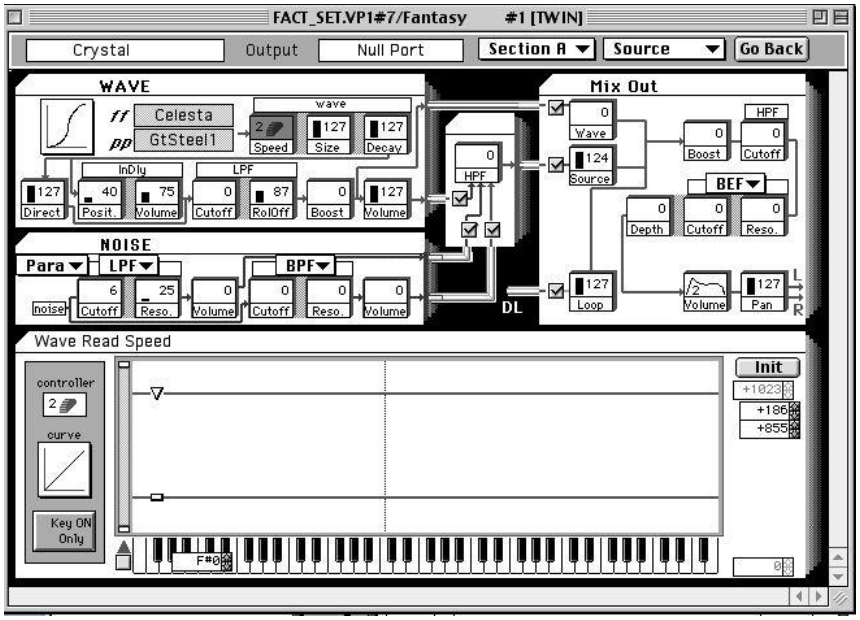
Loop1	Cutoff		
	Boost		
	Attack		
	InGain		
LPF	Cutoff		
	Rolloff		
Stifns	Stifns		
	LPGain		
	Ser/Para		
Loop2	Cutoff		
	Boost		
	Attack		
	InGain		
LPF	Cutoff		
	Rolloff		
Tap	Posit		
	Volume		
Stifns	Stifns		
	LPGain		

Mixer	Wave		
	Source		
	Loop1		
	Loop2		
	Boost		
LPF	Type		
	Cutoff		
	Reso		
	Depth		
Out	Volume		
	Pan		

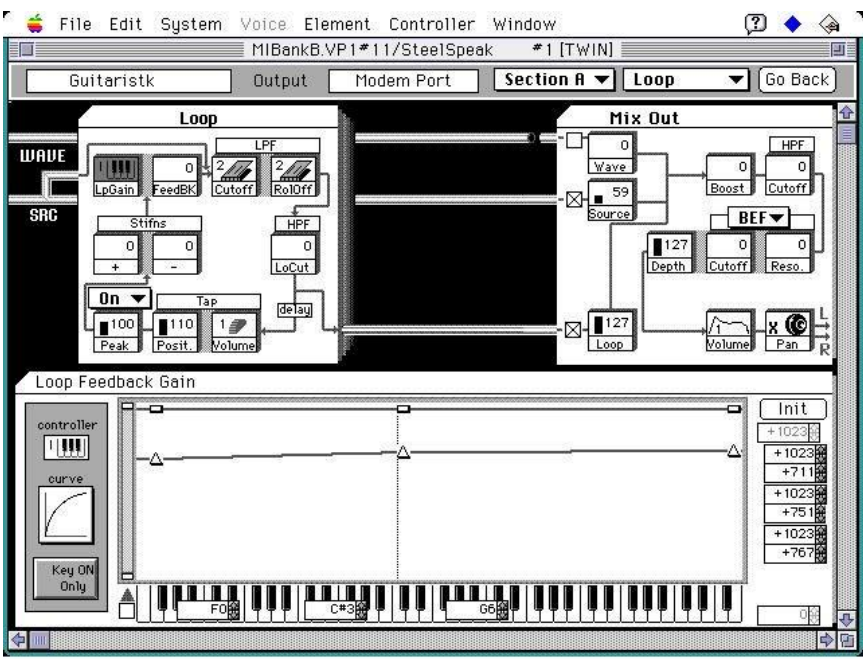


OSC	Fix/Note		
	Shift		
	Fine		
LOOP1	Fix/Note		
	Shift		
	Fine		
LOOP2	Fix/Note		
	Shift		
	Fine		
EFFECT	Type		
	Wet		
	Dry		

Waves A			**Name**	
Waves B				

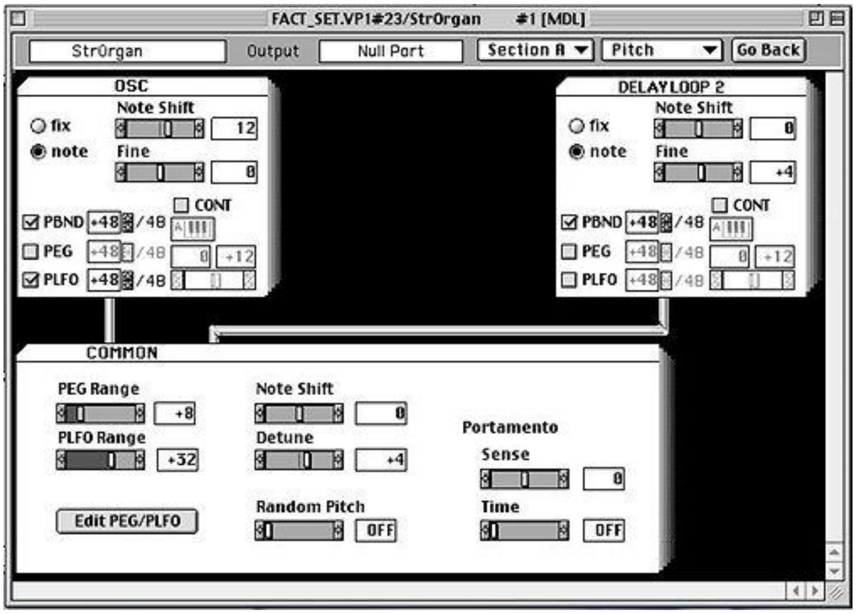


Wave	Speed		
	Size		
	Decay		
	Direct		
InDly	Posit		
	Volume		
LPF	Cutoff		
	Rolloff		
	Boost		
	Volume		
	Ser/Para		
Noise1	Type		
	Cutoff		
	Reso		
	Volume		
Noise2	Type		
	Cutoff		
	Reso		
	Volume		
	HPF		



Loop	LPGain		
	FeedBK		
LPF	Cutoff		
	Rolloff		
	LoCut		
Tap	Volume		
	Posit		
Peak	On/Off		
	Peak		
Stifns	Stifns +		
	Stifns -		
Mixer	Wave		
	Source		
	Loop		
	Boost		
HPF	Cutoff		
LPF	Type		

	Reso		
	Cutoff		
	Depth		
Out	Volume		
	Pan		



OSC	Fix/Note		
	Shift		
	Fine		
LOOP	Fix/Note		
	Shift		
	Fine		
EFFECT	Type		
	Wet		
	Dry		
