what's inside Yamaha DX7 ?

kjukambe

#1
2nd December 2014
My Recordings/Credits

what's inside Yamaha DX7 ?

I'm interested in voice architecture. Does it have single DSP and DAC, or each voice has it's own DAC? what's actually inside?)

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EvilDragon

#2
2nd December 2014
My Studio

One DAC for all voices.

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Yoozer

#3
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It does not have a DSP but an ASIC.

Start reading all posts from here:
https://www.gearslutz.com/board/elec...l#post10580251

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vinyl_junkie

#4
2nd December 2014

There were no DSP's in 1983 or were very expensive.

It's not a CPU doing the sound generation - it's a custom logic IC handling it. The CPU only does control functions. You could do FM on a CPU, but it wouldn't have been cost-effective at the time - even emulating a lesser Yamaha chip in software requires a good deal more horsepower than the DX7's CPU has to offer.

From what I gather the DX synths are actually phase distortion. A table with a sine being read out for a specific phase.
It's essentially a wavetable. (And don't confuse "phase modulation" with "phase distortion" - PD is just a marketing term, from Casio even.)

Here's something that was posted to the music-dsp list a few years ago. It has to do with the OPL2/OPL3 chips that were used in PC soundcards, but it's probably the same techniques that are involved for all of Yamaha's chips.
http://music.columbia.edu/pipermail/...2008-April.txt

And here Adlib / OPL2 / YM3812

"
Date: Fri Apr 18 02:44:09 2008
Subject: [music-dsp] YM3812 reverse engineering

Me and Matthew Gambrell have an interest in emulation of YM3812, which is the OPL2 sound chip found in the Adlib and 8-bit Sound Blaster cards. A later derivative OPL3 is found in early 16-bit Sound Blasters. We sent one YM3812 and one YMF262 (OPL3) to MEFAS for decapsulation; the cost was around 90 USD each. They indicated that the chips would still be operational after decapsulation, but we had no need to test this. Looking at the revealed YM3812 die surface with a microscope turned out two ROM's.

[snip]

The contents could be read bit-by-bit. The first ROM was a log-sin waveform table, containing one quarter of a sine wave, 256 samples long. The second ROM was an exponential table, 256 samples long. There were no other ROM's larger than 16 samples. This is strong evidence that YM3812 produces the sound without any multiplications, using for frequency modulated (actually phase modulated) synthesis the formula:

out = exp(logsin(phase2 + exp(logsin(phase1) + gain1)) + gain2)

[snip]"

For more professional parts, the kinds used in the synths, you might expect to see variations on this theme, perhaps longer tables or wider data words; apparently the OPL chips have 11-bit resolution per operator, I think most synths after the first DX7s have higher resolution than that.

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kjukambe

#5
2nd December 2014
My Recordings/Credits

Thanks for the info! And what do you think about DX7s difference with NI FM7/8 main engine? I'm thinking of hardware DX7s, cos it is available now in my city, and I had a lot of fun with FM7/8 previously

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vinyl_junkie

#6
2nd December 2014

Quote:

> Originally Posted by **kjukambe**
> *Thanks for the info! And what do you think about DX7s difference with NI FM7/8 main engine? I'm thinking of hardware DX7s, cos it is available now in my city, and I had a lot of fun with FM7/8 previously*

Personally I prefer the sound, hiss etc of the hardware units.

I have 4x different 4op Yamaha FM synths and they all have their own unique sound.

What I prefer on keyboards like the DX-100 is the interface is very easy to use and prefer it to using software.
Regardless of what people say it's very intuitive and every button has it's own function.

Saying that the same doesn't seem to apply to the DX-7 and friends who have it say it's still a bit of a pain to program but it's still more fun than a bunch of mouse clicking imo

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acreil

#7
2nd December 2014

Quote:

> Originally Posted by **EvilDragon**
> *One DAC for all voices.*

...but each voice is converted individually.

Quote:

> Originally Posted by **vinyl_junkie**
> *There were no DSP's in 1983 or were very expensive.*

The main feature of a DSP is a fast hardware Multiplier. 16x16 multipliers were available already in the mid 70s, but were very expensive because they require a lot of logic gates. The Yamaha FM synths instead use logarithms; a multiply is replaced by an addition and exponential table lookup. Many of the implementation details are described in patent 4554857.

Quote:

> From what I gather the DX synths are actually phase distortion. A table with a sine being read out for a specific phase.
> It's essentially a wavetable. (And don't confuse "phase modulation" with "phase distortion" - PD is just a marketing term, from Casio even.)

It's phase modulation; something is added to the carrier's phase accumulator, while phase distortion modifies it with a piecewise linear function.

Quote:

> For more professional parts, the kinds used in the synths, you might expect to see variations on this theme, perhaps longer tables or wider data words; apparently the OPL chips have 11-bit resolution per operator, I think most synths after the first DX7s have higher resolution than that

The GS-1/GS-2, OPL series chips and OPP (DX100, FB01, etc.) have 256 entry tables, meaning that the complete sine wave is 1024 samples long. Everything else (CE20/25, TX81z, DX7, DX7II, SY77, FS1R, etc.) uses 1024 entry tables, so the complete sine wave is 4096 samples long.

Lower table resolution leads to added high harmonics (often aliased), noise and rough sounding envelopes (zipper noise). The additional waveforms that are present in some models are generated by reading out the same quarter log sine table in different ways.

DACs, sample rates, etc. also vary between models. DACs add significant distortion in the GS-1/2, CE20/25, DX7 and 4 operator models, but are relatively insignificant in the others.

Quote:

> Originally Posted by **kjukambe**
> *Thanks for the info! And what do you think about DX7s difference with*

> *NI FM7/8 main engine? I'm thinking of hardware DX7s, cos it is available*

FM7/8 (as well as other recent hardware and software implementations) likely use smooth interpolation to generate the sines, which eliminates the effects of the low resolution tables. DACs aren't emulated. The envelopes and modulation aren't identical, so patch conversion is imperfect (and thus direct A/B comparisons are somewhat flawed). It's FM as it's mathematically supposed to be, but it leaves out the high harmonics, noise and distortion that people have come to associate with the Yamaha synths.

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vinyl_junkie

#8
2nd December 2014

Superb info acreil, thanks!

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kjukambe

#9
22nd October 2015
My Recordings/Credits

thanks for the info! so, the clock of DX7's master DACs runs at a constant speed? and if so, what is the carrier frequency?

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nednerd

#10
22nd October 2015

Quote:

> Originally Posted by **acreil**
> *...but each voice is converted individually.*

With one DAC and 16 voices, how does this work, can you elaborate? Multiplexing?

Last edited by nednerd; 22nd October 2015 at 03:52 PM..

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8golftango

#11
22nd October 2015

Kewl thread

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acreil

#12
23rd October 2015

Quote:

> Originally Posted by **kjukambe**
> *thanks for the info! so, the clock of DX7's master DACs runs at a constant speed? and if so, what is the carrier frequency?*

The sample rate is 49096 Hz, so the DAC is working at 785536 Hz (49096 * 16).

Quote:

> Originally Posted by **nednerd**
> *With one DAC and 16 voices, how does this work, can you elaborate? Multiplexing?*

Yeah, it's time-multiplexed, but unlike the usual polyphonic time multiplexed DAC where each channel gets is own sample and hold, here there are only two sample and holds that are used alternately (and I think two are used so that it's not too fast for the switches, etc.). So one sample and hold gets the even-numbered voices, and the other gets the odd-numbered voices.

Intuitively it might seem like the resulting waveform would be rough or distorted, since the voices are mixed by interleaving rather than summation, but it's smoothed out by the reconstruction filter.

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nednerd

#13
23rd October 2015

Thanks for the great Info acreil!
That DA logic seems to be quite a peculiar construction.
I did not know that it was possible to run a DAC at 785kHz back then. Quite speedy.
...goes to show that restricted means spark inventiveness.

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acreil

#14
23rd October 2015

Quote:

> Originally Posted by **nednerd**
> *Thanks for the great Info acreil!*
> *That DA logic seems to be quite a peculiar construction.*
> *I did not know that it was possible to run a DAC at 785kHz back then*

The rate isn't really remarkable; DACs can go much faster, and ~800 kHz is more or less average for a time multiplexed synth. The main thing is that the DAC needs to have a lot of dynamic range to reproduce a digitally scaled sine; 16 bit DACs were available at the time, but very expensive. So Yamaha's solution was to use a commonly available 12 bit DAC with an external resistor ladder to make a floating point DAC with 15 bits dynamic range. But the resistors aren't very well matched, so this introduces some distortion, and the DX7 II models switched to a 15 bit linear DAC.

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zmix

#15
10th August 2017

Quote:

> Originally Posted by **acreil**

> *The sample rate is 49096 Hz, so the DAC is working at 785536 Hz (49096 * 16).*

If this is true, is it possible to extract each voice after the S/H..?

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acreil

#16
10th August 2017

Quote:

> Originally Posted by **zmix**
> *If this is true, is it possible to extract each voice after the S/H..?*

Yes, that's how the TX802 gets 8 outputs. There are 16 sample and holds. But you'd have to have the proper timing signals (there are some pins that indicate that this might be possible in the DX7) and build demultiplexing and sample and hold circuits.

Ultimately I'm not sure that it would be worthwhile since the DX7 II is already stereo and the TX802 has multiple outputs.

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zmix

#17
10th August 2017

Quote:

> Originally Posted by **acreil**
> *Yes, that's how the TX802 gets 8 outputs. There are 16 sample and holds. But you'd have to have the proper timing signals (there are some pins that indicate that this might be possible in the DX7) and build*

Thanks for the reply..

It's really nothing more than academic interest at this point, to be honest.
I'd love to see the schematics, though. The service manual I've seen online is somewhat general...

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TNC

#18
10th August 2017

There is a battery that you'll eventually need to change. I'm afraid of 80's synthesizers with batteries.

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GeminIAm

#19
10th August 2017
My Studio

What's inside a DX7? Dead flies and traces of cocaine

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acreil

#20
10th August 2017

Quote:

> Originally Posted by **zmix**
> *It's really nothing more than academic interest at this point, to be honest.*
> *I'd love to see the schematics, though. The service manual I've seen*

I think there are service manuals and schematics floating around for the DX7, DX7II, TX802, TX7 and TF1, and also a couple electronic pianos that are similar to the DX7II (PF80, CLP-30), and a fairly thorough patent (4554857). The DX1 and DX5 should use the same exact DAC design as the DX7.

As far as the DAC goes, the first and second generation models operate in pretty much the same way, only the floating point DAC is replaced by a 15 bit linear DAC.

The time multiplexing thing is also done in cheap ICs that use low quality onboard DACs, so that they can get not totally horrible performance from an 8 or 9 bit DAC. These have really considerable DAC distortion, but it sounds kind of nice since it's independent for each voice. Like the DX7, you can't replicate this effect with a bit crusher. You could hypothetically separate the DAC channels in these too, and in some cases it would be more useful because this would permit separate outputs for tone and rhythm sounds, for example. But there aren't any externally available signals that can drive sample and holds.

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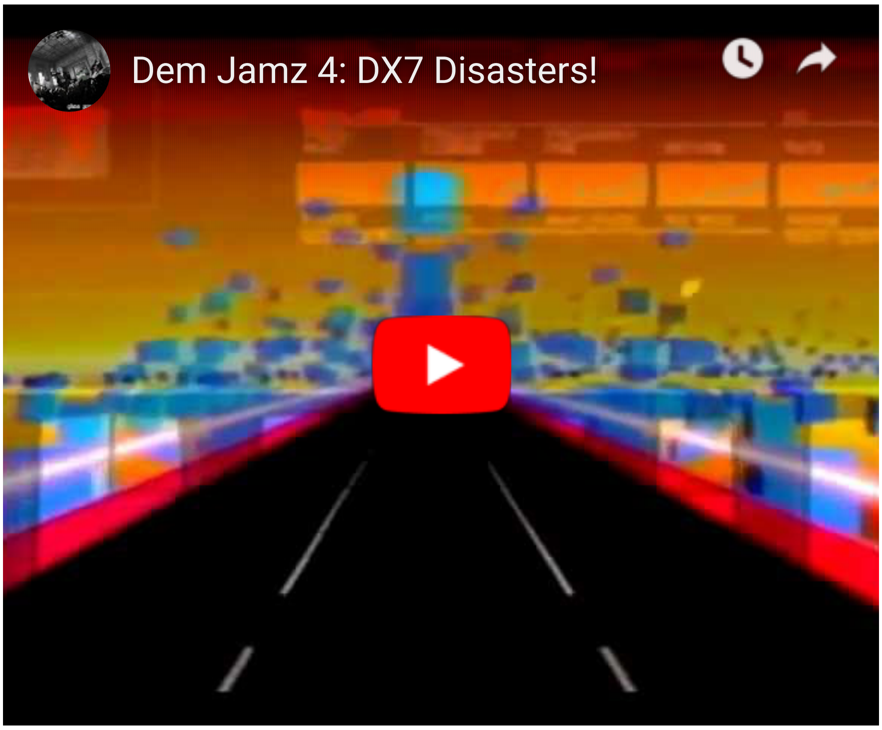
xanderbeanz

#21
10th August 2017
My Studio

A million songs trying to get out.

Here are a couple of them



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#22
20th August 2017

Quote:

> Originally Posted by **acreil**
> *I think there are service manuals and schematics floating around for the DX7, DX7II, TX802, TX7 and TF1, and also a couple electronic pianos*

Casio employed a similar scheme in their VZ series:

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acreil

#23
20th August 2017

Quote:

Originally Posted by **zmix**
Casio employed a similar scheme in their VZ series:

Yeah, oddly the DAC in the VZ appears to individually convert each line (i.e. pair of oscillators) rather than each voice. In the DX7 the advantage of individually converting each voice is that the DAC distortion doesn't cause intermodulation distortion between voices.

In general it's not that uncommon to oversample the DAC in order to increase its dynamic range, although it's not always obvious from the schematic.

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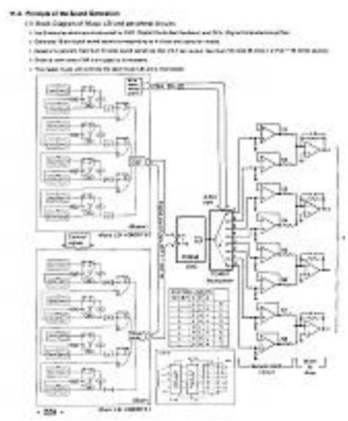
#24
21st August 2017

Quote:

Originally Posted by **acreil**
Yeah, oddly the DAC in the VZ appears to individually convert each line (i.e. pair of oscillators) rather than each voice. In the the advantage of individually converting each voice is that the DAC distortion doesn't cause

I assumed that those are the two 4-voice LSI chips, rather than the internal generator lines, the instrument has 8 voice polyphony.

Attached Thumbnails



Last edited by zmix; 21st August 2017 at 03:36 PM.. Reason: Added image

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acreil

#25
21st August 2017

Quote:

> Originally Posted by **zmix**
> *I assumed that those are the two 4-voice LSI chips, rather than the internal generator lines, the instrument has 8 voice polyphony.*

Actually the LSIs each are 8 note polyphonic for 16 notes total. The VZ-8m has one IC and is thus 8 note polyphonic.

I haven't actually verified yet that each line is converted individually, but it seems to me that it would work that way.

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zmix

#26
21st August 2017

Quote:

> Originally Posted by **acreil**
> *Actually the LSIs each are 8 note polyphonic for 16 notes total. The VZ-8m has one IC and is thus 8 note polyphonic.*

Look at the voice architecture in the block diagram of the 8-voice LSI chip. Each voice in the VZ is comprised of 8 DCO / DCA pairs (M1-M8) each of which can be routed directly to the output, ring modulated with the next, or in series with the next for "phase modulation", and each pair of which (L0-L3) can be routed in series to the next pair or output directly.

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