From: SCOTT 3-FEB-1984 15:54
To: ROY,LARRY

MiniGumby Sound Synthesis System

Analog Output Section Specification

v1.0 2/3/84 sJg

<FF>

MiniGumby Sound Synthesis System

Analog Output Section Specification

MINIGUMBY OVERVIEW

The MiniGumby was conceived as an extremely low-cost sound synthesis IC intended for use in plug-in game cartridges for the 3600 home video game system, augmenting the sound generation capabilities of the TIA chip present on the 3600 motherboard. As such, it faces severe constraints on package size, die size (and thus cost), external component count (preferably zero), and general producibility. The overriding concern has been to produce a synthesis subsystem whose total cost, installed on the cartridge, would be less than $1.00. Thus, an "add 'til full" design approach has been used. Secondary concerns were to produce the highest quality sounds possible within the first constraint, with particular attention being paid to:

1. Versatility - many sounds are available, with ROM-based waveforms redefinable on a per-game basis
2. Musicality - musically useful features, such as fine frequency resolution, wide ranges of timbre variation, and careful choice of noise generation techniques
3. Listenability - a wide dynamic range generation technique and output section assure low distortion and high quality previously unattainable at this price point.

Pursuant to these goals, a generation technique that results in the production of 10-bit, two's complement sound samples at a 28 KHz sample rate has been adopted. Current plans are to implement the chip in VTI's 3u HMOS1A process, and the cost constraints dictate that the die size be 120 mils square or less.

The most severe difficulty faced in the design of this chip is that of the output section, consisting of a 10-bit two's complement DAC and its associated output buffer. We lack the expertise in designing HMOS analog functions, and must now try and subcontract this highly critical design outside.

ANALOG OUTPUT SECTION SPECIFICATION

The analog output section is responsible for converting the 10-bit output sample into an analog current, capable of driving the external audio line on the 3600 cart connector. It is essentially a 10-bit monolithic DAC, followed by a current buffer (incorporating, if possible, a first or second order lowpass filter). The current floorplan for the rest of the device calls for this section and its associated output pad to reside in an area 20 mils square in the extreme corner of the die under the (minimum-spaced) Vdd and Vss rings.

The load seen by the analog output pin can be modelled as 1uf in series with 6.8Kohms to a badly behaved virtual ground summing node. This node is in fact the input to the TV sound subcarrier FM circuit used to convey the audio signal to the user's television (see attached schematics). At audio frequencies, it behaves reasonably like a normal virtual ground, but at higher frequencies, its impedance rises rather erratically, and it will source about 500 mV at 4.5MHz at the output pin. This behavior may cause some difficulty in the design.

It is necessary for the output pin to be able to swing 2 volts at full scale into this 6.8Kohm load, thus sourcing about 225 uA. The 3600 system capacitively couples the incoming signal presented on the external audio line, allowing the output pin to be biased up to any level desired.

In specifiying the required linearity and monotonicity for this device, it must be kept in mind that the final application is to produce music and/or sound effects for a consumer videogame system. Clearly, having +/- 1/32 LSB linearity would be nice, but it is not likely to happen in this technology, with this resolution, in this amount of space. Thus, the relavent specification is to simply keep it monotonic. If monotonicity is acheivable, then the various flavors of nonlinearity will be well enough controlled for this application.

The lowpass filter mentioned above can be a simple first or second order type incorporated into the output buffer. It is used for the normal anti-aliasing functions, when combined with the rolloff inherent in the RF modulator and TV Audio stages (thus the relatively gentle rolloff spec). Its corner frequency should be approximately 4 KHz.

The input to the DAC will be taken from a 10-bit wide, active-low data bus. Logic external to this module will provide an active-high signal which should be used to latch the value of the data bus for conversion.

Any questions or comments should be addressed to:

Scott Griffith
General Computer Company
215 First Street
Cambridge, Ma. 02142
(617)492-5500