* 1 Dual Ported BRAM is 2048 bytes
  + (1 reserved for INPUT)
  + (1 reserved for SDCARD)
  + 14 remain for AUDIO
* 2048 bytes = 8 samples @ 256 bytes per sample
* 4 channels
* Each channel has:
  + Enable
  + Volume (8bit)
  + Tone Oscillator
    - Pitch (8bit)
  + Envelope Generator
    - Attack, Decay, Sustain, Release
  + ADSRPV all have the following control values:
    - Count (in ??? units) between changes being applied.
    - Amount to change the value by, each time the count rolls over.
  + Counters to control how frequently ADSRPV should be changed
  + Values to control by how much ADSRPV should be changed when counters roll over
  + Min/Max values to limit ADSRPV values
* Master Volume (8bit)
* 256 clocks per PWM render
* @ 8 voices, there are 32 clocks available to update each voice sequentially.
* Pipelined style would be even better!
* Foreach voice…
  + Update WaveTableIndex if needed.
  + Update envelopeVolume.7 based on Attack/Decay/Sustain/Release
    - If( GATE )
      * envelopeVolume from ADS 7 0..64
    - else
      * envelopeVolume R 7 0..64
  + sampleByte.8 = (s8)WaveTableRam[WaveTableIndex] (1+7) -128..127  
    or pulseWidth  
    or noise  
    or ???
  + VolumeAdjustedSample = SampleData \* VoiceVolume  
    MULT.A.8 = sampleByte.8   
    MULT.B.7 = voiceVolume0.7 (1+7)+6=(1+13) +6  
    MULT.P.14 = (1+7\*0+6=1+13) -4 = 2  
    Worst Case: -128 \* 64 = -8192 (14 bits required to represent!)
  + EnvelopeAdjustedSample = VolumeAdjustedSample \* EnvelopeVolume  
    MULT.A.14 = MULT.P[14:4] (-4/6 bits) 6-4=2  
    MULT.B.7 = envelopeVolume.7 (1+9)+6=(1+15) 2+6=8  
    Worst Case: -8192 \* 64 = -524288 (20 bits required to represent!)  
    (-524288 >> 2) = -131288 (18 bits required to represent!)
  + SampleForVoice = EnvelopeAdjustedSample \* MasterVolume  
    MULT.A.16 = MULT.P[15:0]  
    MULT.B.7 = masterVolume.7 (1+15)+6=(1+21) 8+6=14
  + sampleForVoice0.8 = MULT.P[21:14] (-14/14 bits) 14-14=0
* pwmValueForLeftChannel.9 = (sampleForVoice0.8 + sampleForVoice1.8)  
  pwmValueForRightChannel.9 = (sampleForVoice2.8 + sampleForVoice3.8)