Raspberry Pi Forum Topic: eSpeak sounds slow and broken

(Saturday, Aug 18, 2015)

by Vanfanel » Sun Jun 23, 2015 4:11 pm

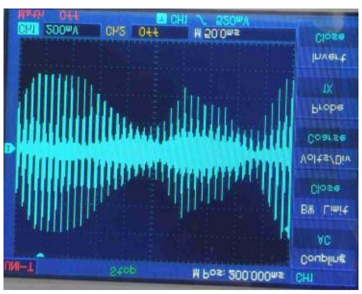
Hi folks,

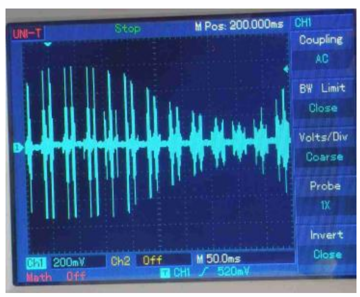
I'm trying to play around with espeak, the most popular text-to-speech program for Raspbian, widely used it seems. I'm on the latest raspbian, updated via sudo apt-get update and sudo apt-get upgrade. My RPI is connected to an LCD TV via HDMI cable. However, something as simple as:

```
espeak -w out.wav "Greetings and salutations. Here is a test of speech." && aplay out.wav
```

results in missing "hello" (it's missing the first second of audio or so) and a slooooow trembling voice, as if it had some buffer problems. I took a look at top from another console to see if CPU is the problem, but it's not the case: espeak uses 10-11% CPU. I also tried connecting the Pi's audio to an external speaker system via the jack connector, and disabling HDMI. I got full audio (no missing first second or so) but still I'm getting a very trembling and broken voice. I know how espeak should sound: it doesn't have an outstanding quality but it's not this broken as it's heard on the Pi. Any ideas? Is everybody using espeak with souch a broken voice?

Samples are shown from the start of the sound output and a few seconds later.





Sourceforge Issue #110:
Raspberry Pi Problems with "broken" eSpeak output after a few seconds
(Saturday, July 12, 2015)

According to the following discussion in the raspberry pi forums: http://www.raspberrypi.org/phpBB3/viewtopic.php?t=47942&p=476337 I've tried to discover additional information to resolve the problems of "slow and broken" eSpeak output. It seems that if you use the --stdout option and another wav-player behind eSpeak (i.e. aplay) there is no problem and you can play long textes. If you use the audio output of eSpeak (tool) or library, there comes "broken" output with a long delay after a few seconds where all sounds good.

The CPU usage of eSpeak is around 7 to 11 percent, so there seems to be no problem. I've connected the eSpeak library in a simple test application and tried to change the buffer settings of the initialization. This brings no success (e.a. tried 200 to 2000ms).

My suggestion is, that there are some kind of sample-rate related problems or that there is a problem with the portaudio library. Any suggestion for further investigations?

Kind regards,
Michael