

Who Am I?

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Background: speech processing,
acoustic echo cancellation

Maintainer of the Speex codec



Speex: Activity

- Project aim:
 - a free alternative to proprietary speech codecs
 - make it easy to create VoIP apps (do the DSP stuff)
- Features:
 - 8 kHz to 32 kHz sampling rate
 - 2.15 kbps to 44.2 kbps bit-rates
 - floating-point and fixed-point code
 - **acoustic echo canceller in the library**



Speex: Challenges

- Writing a **good** VoIP app is way too difficult
 - libasound is too low-level
 - full-duplex synchronisation is problematic for AEC
 - even experienced engineers get it wrong
- We need a high-level API (PulseAudio?) that
 - abstracts away soundcard details (e.g. period size)
 - gives good latency and synchronisation
 - handles xruns more gracefully



Speex: Dependencies

- Libraries in use:
 - libc, libm
- Projects dependent upon:
 - media frameworks, players
 - VoIP apps (Ekiga, Linphone, Asterisk, ...)
 - N770/Maemo? (yes, Speex runs on TI C55x DSPs)
- Interacts indirectly with:
 - **ALSA**
 - Linux kernel



Speex: Next Steps

- Making acoustic echo cancellation *just work*
 - robust echo cancellation code
 - **simple/usable sound API**
 - good sample code
- Having real-time scheduling when needed
- I offer other projects to:
 - provide use/test cases
 - help with libspeex integration
 - write needed signal processing code

