Studio report: Linux audio for multi-speaker natural speech technology

Dr Charles Fox
Research Associate, Natural Speech Technology
University of Sheffield, UK
www.5m.org.uk
Talk aims

- Tell you about the world of Automated Speech Recognition research – not much overlap yet with Linux audio world. But there may be soon!
- Natural Speech Technology is different from standard ASR (such as DragonDictate), more audio/DSP focussed and needing Linux audio.
- What we are using your tools to do (thanks!)
- What we need your tools to do (please!)
  - (vs computer music requirements in particular)
ASR vs Natural speech

- Automated Speech Recognition (ASR)
  - Historically developed as a separate community from audio research (music, DSP etc)
  - Mature field, 20 years of tools
  - Large-data (eg. 1000 hours) and results driven
  - Standard features (MFCC, PLP) preprocessing
  - Then statistical models of feature data
  - Many ASR researchers don't listen to the speech!
  - Then find models and parameters to optimise word error rate (WER)
  - 10% great, 20% good, 40% normal, depends on corpus
ASR vs Natural speech

- ASR just about a solved problem for single, trained user in noiseless environment
- eg. Dragon Dictate, Siri
  - (both by Nuance Inc)
  - WER 1% obtainable
- Clean but unknown speakers
  - eg. radio news broadcasts
  - Still a research area,
  - WERs eg between 17%-70%
ASR vs natural speech

Recent research area: extending ASR to natural speech (Wolfel & McDonough, 2006)

- Business meetings - minuting
- TV – subtitling
- Interview archives – transcription
- Telephone conversations – transcription + keywords
UK NST project

- Involves many organisations interested in NST:

Running 2011-2015. Sample applications:

- Subtitling TV and radio programmes (BBC)
- Consumer and business meeting transcription (Nuance)
- Security (GCHQ)
- Voice control for disabled patients (NHS)
Natural speech challenges

- Unlike standard ASR tasks, we have:
  - Multiple speakers talking at once
  - Speakers moving around the room
  - Significant room effects (reverb, resonance)
  - Background noise (furniture, traffic, plumbing)
  - Unknown speakers, accents/dialects

- Need to work more with DSP now
  - Begins to look more like computer music type research (eg. Music transcription, CASA)
  - Need nice audio tools
  - but different requirements from musical audio
My background – linux audio

- Score following
  - Raphael Music+1
  - “How to be Lost”, Fox&Quinn ICMC2008
  - Linux, ALSA; Matlab+win.

Particle filter + lostness
Lyon cochlea filter
Chordbank filter HMM
Bayesian musical scenes

- Rhythms
- Structures (Hofstadter's CopyCat)
- Python, Lush Lisp, OSS midi
Music recording

Catch-it Kebabs (Riot Music, 2007)
£11.66 (Amazon/shops) or free direct download
www.catchitkebabs.co.uk
Studio recording still in Cubase/ProTools :-(
Some demos/writing now in Ardour/JACK
*Speech vs music data*

- **Corpus size:**
  - music album ~20 hours. music usually one track at a time, 5 mins.
  - Corpus 1000 hours. Needs fast access to all at once; playback ~1 hour.

- **Quality**
  - Music mixdown 44.1kHz, 16bit;
  - classic ASR usually starts with 16kHz 16bit, then extracts features.
  - NST ASR may need music or better quality, eg. 48kHz 32bit.

- **Channels**
  - Music ~ 20 channels, typically record <10 at once
  - NST ASR, localisation and separation techniques may require large mic arrays, 16 channels common, 64+ would be nice...

- **Compute power:** Music usually on one PC; ASR on 100 core cluster.

- **Real-time**
  - Music album: latency is crucial. Classic ASR: processing done offline.
  - NST: maybe will need realtime interplay with the processing?
Why Linux? HTK setup

- Conventional modern ASR is done offline, on large clusters

- Our set up:
  - 100 Core cluster, running Oracle (Sun) Grid Engine
  - CentOS Linux (Community enterprise version of Fedora)
  - Encode 1000 hours of wav to features ~1 day
  - Train language models, 10M words, ~3 hours.
  - Alignment/decode ~1 day; train audio ~1 weekend.

- Crucial need to hack – NST is a new research area and we don't know in advance which parts of the tool chain will need to be opened up and modified!
Why Linux? ASR Tools.

- Much ASR research uses the HTK (Hidden Markov model ToolKit) tools. Native to Unix/Linux (but win32 ports)
  - Under development for ~20 years, mature.
  - Tools for building speech recognisers, is not a full system
  - Began in Cambridge University; sold to spin-out Entropic; bought by Microsoft; licensed back to Cambridge.
  - Cambridge is then allowed to distribute HTK source on the net under a gratis but non-libre licence.
  - Licence allows users to modify code for own use but cannot re-distribute (eg. In a linux distro or “OpenDragonDictate”).
  - Users are “encouraged” to donate their modifications back to Microsoft for inclusion in future version
  - Microsoft could revoke the licence at any time
  - Many groups have their own mods that they don't donate back
  - A GPL alternative Kaldi is in progress; but HTK is dominant.
Why Linux? HTK.

Small HMMs can be written in a few lines of Python or Octave.

ASR requires very large ones, e.g. 200,000 triphone states /c/ /a/ /t/ vs /b/ /a/ /t/

Transition priors from large language models, e.g. 10M words.

Large models need lots of clever tricks and heuristics to run in reasonable time and memory (e.g. On 100 $gb$-node clusters). Hence HTK.
HTK command line tools

- Preprocessing
  - Listening (eg 1 hour wavs, show speech features)
  - Feature extraction (MFCC, PLP)
- Alignment
  - Line up known transcript with audio, using HMM
- Training
  - EM training of HMM
- Decoding
- Scoring
  - International NIST standard scores, WER
  - Typically controlled from big tcsh or Python scripts
HTK-related ecosystem

- Sox – audio conversion and listening (GPL)
- Perl/Python.regex – transcript processing (GPL)
- HDecode – alternative HMM decoder (?)
- ARPA language modelling tools (govt/BSD?)
- NIST scoring tools (govt/BSD?)
- Padsp – makes HTK work on PulseAudio (GPL)
- OSS – native HTK audio system (GPL)
- Tcsh/Python scripting (GPL)
- Sed/awk – always useful (GPL)
- Audacity, OpenOfficeCalc for inspection.
From ASR to NST

- Unlike standard ASR tasks, we have:
  - Multiple speakers talking at once
  - Speakers moving around the room
  - Significant room effects (reverb, resonance)
  - Background noise (furniture, traffic, plumbing)
  - Unknown speakers, accents/dialects
Lab set up

- 16 channels:
  - Up to 8 ceiling mics
  - Up to 8 table mics
  - + headsets
- Two MOTU Pre8s
- Firewire
- 3D people trackers
- 3 Cameras
- Teleconferencing (H323, sip)
- DAW: 3GHz, 4Gb, Ubuntu Studio 11.10
- Record 16chs, 48kHz, 316chs, 48kHz, 32bit
- For 1 hour meetings
Linux meeting recording

Charles Fox  www.5m.org.uk   LAC2012
Set-up lessons learned

- ffado (firewire) with JACK2 (install Ubuntu packages: jack2d, jack2d-firewire, libffado, jackd, laditools.)
- Unlock memory.
- adduser charles audio.
- edit /etc/security/limits.d/audio.conf
  - @audio - rtprio 95
  - @audio - memlock unlimited
- Qjackctl: 128frms/per, 48kHz, 3per/buf.
- Result: Just 11 xruns in 1hour. Only 25% CPU usage on 16ch/16kHz/32bit 2-core 3GHz machine.
  (So looking good for more mics!)
Multi-mic de-mixing

- Speakers may talk over one another
- And background noises interfere with them
- Use microphone arrays to pull out sources:
  - Assume each speaker and noise is $x_i[t]$
  - So vector of source signals, $x[t]$
  - Assume mics get stationary linear mix, $y=Mx$
  - Search for parameters $M$ to make $x'=M^{-1}y$ look like $x$
    - ICA: assume sources are independent
    - Speech priors: make $x'$ have speechlike harmonics
    - Beamforming: use knowledge of locations and physics...

- More mics $\rightarrow$ better accuracy!
Environment modelling

- Assume each mic channel $j$ picks up a source $x_i[t]$ that has been filtered by the room's reverb and resonance,

$$y_j(t) = \sum_d f_{ij}(d)x(t-d)$$

- Find filter $f$ to “de-verb” the room and retrieve $x$.

- Again, by making prior assumptions about what we would like $x$ to look like

- More complicated when the multiple source problem is there at the same time...

$$y_j(t) = \sum_i \sum_d f_{ij}(d)x_i(t-d)$$
Speaker localisation and beamforming

- Consider the time to reach different mics from the speed of sound.
- **Requires sample-perfect accuracy.**
  
  (1 meter in 1/330s = 3 ms = 145 samples@48 kHz)
- For testing, also sync to our people-tracker and video camera streams (JACK? GStreamer?)
- Time delay between each pair of mics gives a set of possible locations. Intersecting sets from many pairs are probably source locations.
- Speakers may move around over time
  → Kalman like tracking (+video)
- Once we know the speaker locations, then use physics of sound to do source separation – infer back from constructive and destructive interference patterns - beamforming
- Bayesian probability is a useful framework to try solving **all** of these problems simultaneously...
- Bayes is good at chicken-and-egg problems but needs great computing power (eg. Clusters), especially if we want real-time.
Remote NST recognition

Microcones, Android phones, sending speech back to our cluster for analysis.
Linux for natural speech

- Unlike standard ASR tasks, we have:
  - Multiple speakers talking at once
  - Speakers moving around the room
  - Significant room effects (reverb, resonance)
  - Background noise (furniture, traffic, plumbing)
- Linux audio requirements:
  - Large data sets (1000 hours) – mostly (entirely?) for offline work
  - Each recording typically 1 hour
  - Many simultaneous recording channels, 16 good, 64+ better ...
  - Sample-perfect synchronisation between mics for beamforms
  - Sync audio to video and 3D data streams – sample accurate
  - HTK/Microsoft licensing issues (or move to Kaldi?)
  - Linking existing HTK (or Kaldi) offline cluster tools to realtime audio – and over networks for remote users, eg. their Android phones.

- What can we contribute back to GPL Linux Audio from the NST Project?
- Please do get in touch now or after the conference: www.5m.org.uk