

Enhanced Digital LMR Seminar 19th Aug 2016

Wentworth by the Sea
New Castle NH



LMR Codecs

Why codecs?
Which ones?
Why care?

Joseph Rothweiler
Sensicomm LLC
Hudson NH

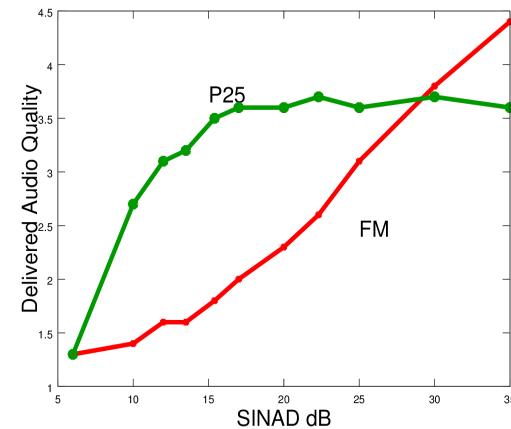
<http://sensicomm.com>

Presentation available at:
<http://rothweiler.us>
Rev. Aug. 22, 2016



Why go Digital?

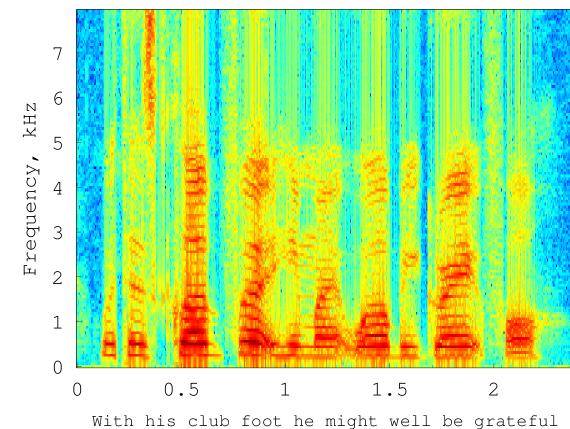
- Clearer voice, easier communication (potentially).
- Range or Power efficiency
 - Longer radio range for the same TX power.
 - Example: P25 > 10dB better than Analog FM.
 - Resistant to RF adjacent-channel or co-channel interference.
- Digital encryption much better than analog scrambling.
- Easy to mux multiple audio channels or other nonvoice data (eg, GPS location tracking).
- Other features: "voicemail", store and forward, archival recording, etc.



Digital Voice requires about 10dB less transmit power here.

Why not use Waveform Coding (PCM)?

- **Waveform coding – Pulse Code Modulation(PCM).**
 - Microphone → Analog-to-Digital converter; Digital-to-Analog converter → speaker.
 - Sample at a fixed rate, with a fixed number of bits per sample.
- **How fast should we sample?**
 - Nyquist's Theorem: Sampling rate must be at least 2x the highest frequency.
 - Speech goes to 6 to 7 kHz, or more.
 - Telephones cut off at 4 kHz.
- **Number of bits must be high enough to minimize quantization noise.**
 - 16 excellent, 12 good, 8 pretty good with nonlinear (μ -law) coding.



Sampling Rate	Bits	BPS	Quality
16000	16	256k	Near Transparent
8000	8 μ	64k	As good as Bell's 1880's carbon microphones.

- Simple compression (eg, predictive coding) can get us to 32 or even 16 kbps.
- LMR needs 8kbps or below to fix existing channel spacings and bandwidth allocations.

→ Vocoders to the rescue!

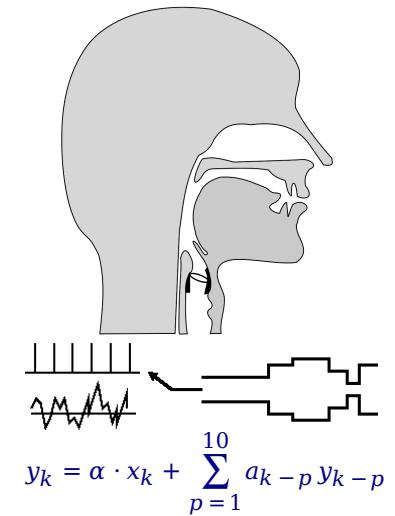
What's a Vocoder?

- **Definitions**

- CODEC: Convert analog signal to digital (enCODer) and vice versa (DECoder).
- Vocoder: a codec built on a model of human speech.
Vocoders transmit a description of the signal rather than the signal itself.
- The decoder synthesizes a waveform that "sounds like" the input.

- **Vocoders use an approximation of the human vocal tract:**

- **Excitation source:**
 - Periodic pulses from the vibrating vocal chords ("voiced"), or
 - noise from friction turbulence ("unvoiced")
- **Mouth and throat form resonant cavities that shape the spectrum (like a pipe organ).**
- **Simplified mechanical model is a pulse or noise signal feeding cascaded tubes.**
 - Math equivalent is Linear Prediction model (or autoregressive or all-pole).
- **Vocoders send**
 - small set of parameters (filter coefficients, pitch, voicing, gain, ...)
 - at a low rate (20-30ms)
 - so the bit rate can be very low



A Somewhat Oversimplified Vocoder Grouping

- **LPC: the original buzz-hiss vocoder model.**
 - SIGSALY(1943) or DoD LPC-10 (1970's).
 - Simple 7-parameter model: amplitude, pitch, voiced/unvoiced bit, 10 coefficients.
 - Highly intelligible, but very synthetic machine-like quality.
 - Problematic in noise, especially the voiced/unvoiced decision.
 - Good approach when you desperately need the lowest possible bit rates.
- ↓
- ***MBE* - Multiband-Excited synthesis.**
 - IMBE, AMBE, AMBE+2, etc.
 - P25, satellite systems, etc.
 - Replace voiced/unvoiced bit with a multiband mix of buzz and hiss excitations.
 - Plus other enhancements.
 - Better than classical LPC, without much increase in bit rate.
-
- ***ELP* - (something)-Excited Linear Prediction.**
 - CELP, ACELP, etc.
 - TETRA, cellphones, ...
 - Use a stored or computed library of excitation functions.
 - Search algorithm to select the optimal excitation.
 - Can reproduce the input waveform if given enough bits.



Some LMR Standards and their codecs

Standard	Past, Present and Future Codecs
P25	IMBE, AMBE+2
TETRA	ACELP, AMR 4.75(opt); Future TBD, MELPe?
NXDN	AMBE+2
dPMR	AMBE+2, Chinese TBD, RALCWI, Manufacturer-Specified
TETRAPOL	RPCELP
DMR	AMBE+2
FirstNet	AMR? AMR-WB? Other LTE?
MPT-1327	Analog FM
LTE(cellular)	AMR-NB, AMR-WB, etc
D-STAR(HAM)	AMBE, CODEC2

Some Codec Characteristics

Codec	Year	Voice Rate	Notes
IMBE FR	1995	4.4	TIA-102.BABA; DVSI
AMBE FR	~1998	4.4	TIA-102.BABA; DVSI
AMBE HR	2007	2.45	TIA-102.BABA-1; DVSI
ACELP	1996	4.567	ETSI EN 300 395-2; Reference and test vectors.
MELPe		0.6,1.2,2.4	NATO STANAG 4591
RALCWI	<2011	2.4,2.74	Proprietary cmlmicro.com
RPCELP	1997?	6.0	TETRAPOL specs
CODEC2	?	0.7-3.2	freedv.org - Ham radio project

- New codecs and improved versions of old codecs appear regularly.

A Note on Evaluation

- Conventional audio measurements (SNR, bandwidth, distortion) don't really apply to vocoders.
 - We don't transmit the waveform, so we can't use waveform measurements.
- Subjective testing: Rating by a panel of trained listeners.
 - Quality: Mean Opinion Score (MOS) - Does it sound good?
 - Rating as: 5:Excellent 4:good 3:fair 2:poor 1:bad
 - Intelligibility: Modified Rhyme Test (MRT)
Can you distinguish consonants?
 - Listener hears: "Please select: red" and selects from: led shed red bed fed wed
- Sounding good is good, but a communication system is for communicating.
 - Intelligibility is what you need.
 - Tests aren't reality: If the users are complaining, pay attention.
- Testing is a really involved process, best left to the experts.
 - Data collection, listening environment and equipment,
 - listener training, habituation* and fatigue.

In English, consonants carry most of the information.
 n _n_g_l_sh, c_ns_n_nts c_rry m_st f th_ nf_rm_t_n.
 I_ E__i_, _o_o_a_ a_ o_ o_ e i_ o_ a io_.



*"Vocoders sound a lot better on Friday than they do on Monday."

Background Noise Problems

- **Effects**

- **Theoretical:** no longer fits the model.
- **Practical:** harder to calculate the filter coefficients, and especially the pitch.
- Particular issue for firefighters.

- **Results**

- Lowered intelligibility.

- **Mitigation**

- **Avoid noise**

- Noise-canceling mic, throat mic, dual-microphone setups, etc.

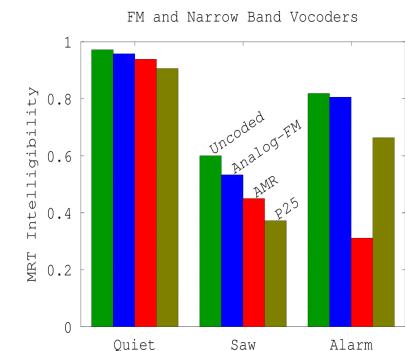
- **Noise reduction algorithms**

- Hard to improve intelligibility this way.

- **Robust codec**

- **Test for it.**

- **Bit rate tradeoff.**



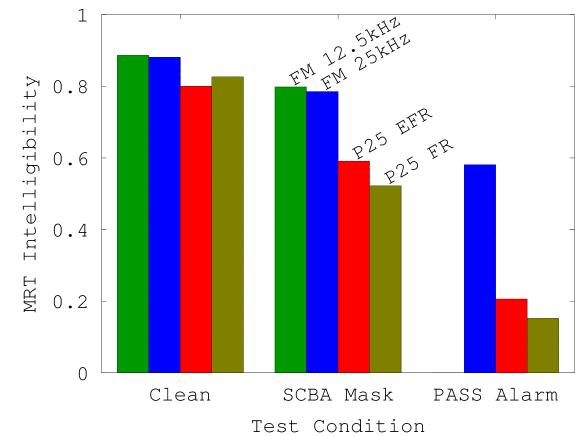
Voice Model Limitations

- **Model approximation**
 - Nasal sounds (/m/, /n/) don't follow the lossless-tube model.
 - Vocal chord vibration isn't really all that regular for many speakers.
 - Breathy voices: simultaneous mix of voiced and unvoiced excitation.
- **Unusual situations**
 - Whispered speech (covert situation?)
 - Distorted speech (Firefighter breathing mask - SCBA).
 - Other talkers.
- **Result: unnatural sound, difficulty recognizing speakers, loss of inflection (stress, urgency, fear, etc).**



Background Sounds

- **Background noises may be important for situation awareness.**
 - Firefighter alarms: man-down, air tank empty, etc.
 - Sirens, etc.
 - Crowd noise.
- **Vocoders are for voice, so nonspeech sounds may be heavily distorted.**
 - Which alarm just went off?
 - Is it a happy crowd or a riot?
- **Test and tell users about any issues.**

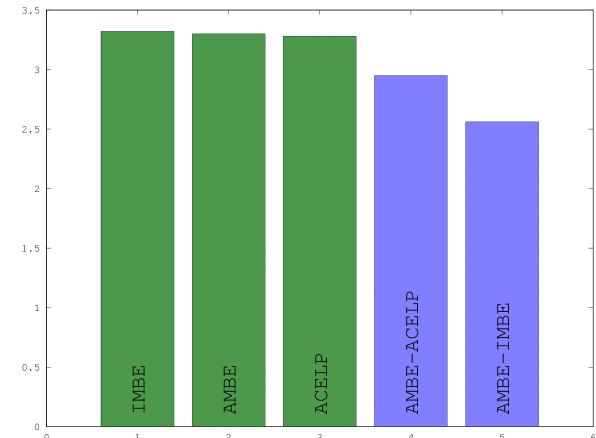


Delay

- Underappreciated issue: Analog FM delay is essentially zero.
- Vocoders analyze 50-100ms segment of audio to extract parameters, so delay is unavoidable.
- Add in processing time, RF synchronization, and error-correction interleaving.
- In full-duplex telephony delays above 500ms become very annoying.
 - Throws off the speak-response rhythm, so perceived as rude behaviour by the other speaker.
 - Can't jump in the gap, so end up interrupting while speaking.
- Issue for precise coordination?

Tandeming

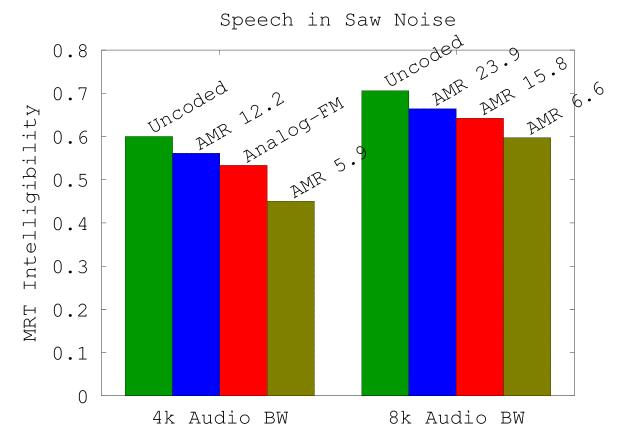
- Connection of 2 different vocoders in series.
 - P25 to TETRA via some sort of gateway.
 - Range extension: radio → satellite link → radio.
 - Old radio to new radio with improved codec, via gateway?
- Avoid tandeming if possible.
 - End-to-end transport of codec bitstream is best.
 - Consider specifying a mandatory codec for interoperation / backward compatibility?



Tandem connection is worse than either alone.

Looking to the Future: Wideband

- Future systems (FirstNet?) may use wider audio bandwidth.
 - AMR-WB codecs support 16kHz audio sampling rate.
 - DVSI advertises a wideband codec.
 - Others exist or are in development.
- Wideband does improve operation in challenging acoustic conditions.
 - Nice to have when you can get it.



Bitrate and Audio Bandwidth improve intelligibility.



Summary Recommendations

- Use the highest bit rate you can get away with.
- Test under realistic conditions.
- Test for usability (intelligibility), not just "quality".
- Think about non-speech signals.
- Consider compatibility:
 - Need to communicate with other organizations' radios.
 - Future buys to expand the system.



References

- **Images on 3, 8, 9 are from Wikipedia.**
- **Charts on slides 1, 8, 10, 12, 13 were generated by the author from data published by NTIA.**
 - **See these NTIA Publications for the full studies:**
 - <http://www.its.blrdoc.gov/publications/download/TR-01-386.pdf>
 - <http://www.its.blrdoc.gov/publications/download/TR-08-453.pdf>
 - <http://www.its.blrdoc.gov/publications/download/TR-09-459.pdf>
 - <http://www.its.blrdoc.gov/publications/download/TR-13-495.pdf>
 - <http://www.its.blrdoc.gov/publications/download/TR-15-520.pdf>