Does PMSE waste spectrum? – A balanced view from a Scientist in Communications and Cellular

Prof. Dr.-Ing. Georg Fischer
Lehrstuhl für Technische Elektronik
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2. Basic communications model
3. Source coding
4. Digital versus analogue transmission
5. What is so unique with PMSE?
6. Wireless traffic growth
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1. Speakers background

- Prof. at University Erlangen-Nuremberg
- History in Basestation technology, Cellular, Standardization, Regulation
- Erlangen known for MP3 / Fraunhofer IIS / Audiolabs
- Starting point for “balanced view”
Speakers Background

Experience

Prof. Georg Fischer (geb. 1965)

1986-1992 Study of Electrical Engineering at RWTH Aachen (Aix La Chapelle)
Focus on Communications, Radio Technology, Field Theory

1993-1996 Research assistant at University of Paderborn

1997 Dr.-Ing.,
Thesis „Adaptive Antenna Arrays for mobile satellite reception“

1996-2008 Lucent, later Alcatel-Lucent, Bell Labs Research
Research on Basestation RF Technology

2000 Bell Labs DMTS (Distinguished Member of Technical Staff)
2001 Bell Labs CMTS (Consulting Member of Technical Staff)
Chairman of ETSI SMG2 WPB EDGE

2001-2007 Part time Lecturer at University of Paderborn

April 2008 University of Erlangen-Nürnberg Prof. for Electronics Engineering
Research on Cognitive Radio, Frequency Agile Radio, Analog-Digital Balance

Since 2010 ETSI STF 386 Chairman „Methods, parameters and test procedures for cognitive interference mitigation techniques for use by PMSE devices“

Since 2010 Reviewer for EC FP7, COST, DFG, NSERC, IWT Flandern, Helmholtz Society
Chair is the birthplace of MP3 Audio Compression, commercialized by Fraunhofer
Speakers Background
Audio Labs

- A joined activity of Fraunhofer IIS and University of Erlangen-Nürnberg

International Audio Laboratories Erlangen

Structure

The International Audio Laboratories Erlangen (AudioLabs) was founded in August 2008 as a joint institution of Fraunhofer IIS and University of Erlangen-Nürnberg. This research center enables university scientists to develop new technologies for digital processing of multimedia content with Fraunhofer IIS staff and visiting researchers from around the world. The collaborative venture is intended for at least ten years. Creative synergies between researchers from several disciplines, combined with the many years of audio compression experience at Fraunhofer IIS, will drive future research topics and guarantee continuous innovations.

Appointed professors:
- Prof. Dr. Juergen Herre (Audio Coding)
- Prof. Dr. Bernd Edler (Audio Signal Analysis)
- Prof. Dr. Emanuel Habets (Perception-based Spatial Audio Signal Processing)

Executive board:
- Prof. Dr. Albert Heuberger (Speaker)
- Prof. Dr. Jurgen Herre (Representative University of Erlangen-Nürnberg)
- Dr. Bernhard Gill (Representative Fraunhofer IIS)

Research Coordination:
- Dr. Frederik Nagel

Coordination:
- Dr. Stefan Turowski

Founded July 2008
2. Basic communications model

- We define what is “information”
- We identify irrelevant information
- What is meant by spectrally efficient?
Basic communications model
Information theoretical view

Model

- Information from a communication source is sent to a communication sink
- Goal is to minimize Transinformation
- Only Transinformation is transported over a wireless connection
- Amount of Transinformation = spectrum need

![Diagram of basic communications model](image)
There is more to transmit than Transinformation

- Supporting equalization of wireless channel by adding fixed trainings sequences
- Addition of synchronisation information
- Channel coding to make transmission more robust
- Overhead for organisation

*First we squeeze transinformation by source coding, than we blow it up again*
3. Source coding (Compression)

- By compressing data we save spectrum
- But we trade compression versus quality
Source Coding
What flavours?

Lossless coding
• Only redundancy is stripped off
• Original information can be fully reconstructed – waveform conservative
• Has to be used if receiver is not known (irrelevance can not be identified)
• E.g. *.ZIP, Compression ratios limited (2:1)

Near lossless
• Nearly reversible, e.g. SLQ, not appropriate for studio quality
• Compression ratios limited (8...4:1)
• Always variable rate → variable spectrum consumption

Lossy coding
• Redundancy and Irrelevance is stripped off
• Original information cannot be gained back
• Coding has to know what is irrelevant at communication sink
• Highest compression ratios possible (50...20:1 or more…)
• Usage of psychoacoustic and psychovisual effects
• Examples: Audio MP3, Video MPEG
  TV: DVB-T / T2 / C / S / S2, Satellite: SDTV / HDTV
Slider - You can trade quality versus compression
## Source Coding

### Examples

## Windows Media Codecs

### Codec Descriptions

The following table describes the intended uses of the Windows Media codecs.

<table>
<thead>
<tr>
<th>Codec</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Windows Media Audio</td>
<td>An audio codec that supports three categories of encoded content: Standard, Professional, and Lossless.</td>
</tr>
<tr>
<td>Windows Media Audio Voice</td>
<td>Audio codec optimized for encoding the human voice at high compression ratios. This is the preferred codec for streams consisting mostly of spoken words. For content that is mixed music and speech, this codec can dynamically change the encoding algorithm used, to get optimal quality.</td>
</tr>
<tr>
<td>Windows Media Video 9</td>
<td>A video codec that supports four categories of encoded content: Simple Profile, Main Profile, Advanced Profile, and Image.</td>
</tr>
<tr>
<td>Windows Media Video 9 Screen</td>
<td>Video codec optimized for encoding sequential screen shots from computer monitors. This codec is often used for software training or support by recording monitor images while computer applications are being used.</td>
</tr>
</tbody>
</table>
Source Coding
Examples

Audio
• CD: 500 Mbyte
• 10 Tracks → 50 Mbyte per Audio file
• Apply MP3 high compression 1 Mbyte,
• lossy source coding
• Compression Ratio: 50:1

GSM Speech
• Microphone signal: 8 kSa/s x 13 bit =104 kbit/s
• EFR 12 kbit/s, Compression 9:1
• AMR 4,75 kbit/s, Compression 22:1

Actual Analog PMSE
• Analogue Compander System (an analogue source coding)
• Compression 2:1 (equivalent)

SLQ (Spheric Logarithmic quantization)
• Near lossless audio codec, only stripping off redundancy, no irrelevance identification
• 16 bit down to 2..4 bit per sample
• Compression ratio typ. 4:1, max 8:1
• Not too much...

Motivation:
The higher the compression, the less spectrum is needed...
4. Digital versus analogue transmission

- We compare analogue and digital transmission
- Digital allows for scalability of quality, so not always better quality
- We have to pay a price for digital transmission – signalling overhead
Digital versus Analogue transmission
A balanced view

Common misunderstanding
• Digital is better! - wrong
• It can be better, but it also can be worse
• Digital just allows for scalable quality

Digital transmission
• First we apply source coding - we squeeze information, we compress maximal
• Than we blow up by putting on top channel coding to protect digital data
• We pay a further price by signalling overhead (protocol, Training symbols)
Digital versus Analogue transmission
Impact of coding languages

Overhead
- Programming language implies a lot of overhead
- This is waste of spectrum

- Binär
- JSON
- XML
- Java

Digital versus Analogue transmission
Impact of protocol stacks

Overhead
• Example VoIP (Voice over IP)
• VoIP = raw voice data x33
5. What is so unique with PMSE?

- There are distinct differences to other systems
- PMSE cannot simply take over established solutions from cellular
What is so unique with PMSE?

Specifics

Latency

- For High Quality low latency required, < 5 ms round trip
- Dilemma: Information source and sink are at identical location
  - Drummer has wireless microphone and wireless In Ear monitor
  - Situation totally different from other wireless systems
- Landline telephone 200 ms
- LTE record today 18 ms demonstrated in Stuttgart area by Alcatel-Lucent
- Analysis of raw data with lossy source coding introduces latency

Battery operation

- Wireless equipment part of costume, should be invisible
- Today analog FM, constant envelope modulation friendly for PA
- Digital Modulations typically have Crest (Amplitude variation)
- Digital CPM (Continuous Phase modulation) could cope
- Source coding at wireless microphone also would cost battery power, but Moore’s law works for you
Cascade of source coders

- Two communication links involved (Production+Distribution)
- Cascade of two lossy compression schemes leads to audible artefacts
- Only solution, use lossless coding on production link
- During production you don’t know distribution, so irrelevance cannot be identified
- Therefore only lossless source coding can be used
- Compression factors limited, sufficient spectrum necessary

What is so unique with PMSE?

Specifics

Production → High Quality Audio Archive → Broadcast → CD → MP3 download

Roundtrip Latency?
6. Wireless traffic growth

- Traffic is growing exponentially in Cellular and PMSE
- Spectrum is a limited natural resource - How to cope with it?
- How can needs by cellular be served?
Wireless traffic growth
What will the future bring us?

Wireless traffic growth
• Moore’s law: Integration density of microelectronics doubles every 2 years
• Edholm’s law: Datarates are as predictable as Moore’s law (CTO Nortel)
• D. Poppen E-PLUS CTO: We see 30x in 5 years equal doubling every year

Channel coding / Transmission schemes
• We are already near the theoretical Shannon Bound (BPSK with Turbo Codes 0.2 dB)
• No wonders will come!

Source Coding - lossy
• There is still room for improvement
• However the analysis of raw information requires processing – latency problem
• Knowledge on recipient necessary
• Higher compression will come

Source coding - lossless
• Not widely used due to low compression factors
• However, an Option for digital PMSE
• Compression factor will keep limited

So how to cope with increased traffic ????
Wireless traffic growth
Evolution of Cellular

LTE (Long Term Evolution)
• Claims x2.4 improvement in spectral efficiency
• Gain x2 is coming mainly from 2x2 MIMO
  (2 Antennas at basestation and 2 and terminal)
• Net Gain by transition from UMTS to LTE x1.2 - what an effort...

What options are there to boost networks?
• MIMO - will we see 4x4 ???
• Network MIMO – coordinated Multipoint
• Source Coding!! ... obviously lossy source coding
• Carrier Aggregation – bundling spectrum, that is here and there
• Flexible Active Antenna Arrays – see Alcatel-Lucent LightRadio
• Network densification
• Femto Basestations (See e.g. EU FP7 BEFEMTO and FREDOM Project)

What is Network Capacity?
• Information theory wise it is just transinformation
• But you are thinking in terms of number of services offered inside given spectrum
• Lossy source coding will enlarge number of services but not the transinformation
It is impossible to cope with an exponential traffic growth just by adding more spectrum.

It is a hydra....

Do you know the fairy tale?
1 rice corn on the first field, then 2, 4, 8, 16, 32, 64, 128, 256, 512...
7. Spectrum considerations

- We show that PMSE transmission is more spectrally efficient than UMTS
- We identify the monetary value of spectrum
## Spectrum considerations
### Comparison PMSE versus Cellular

<table>
<thead>
<tr>
<th></th>
<th>PMSE</th>
<th>Cellular / UMTS</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Audio Quality</strong></td>
<td>High for content production</td>
<td>Only speech</td>
</tr>
<tr>
<td><strong>Audio rate</strong></td>
<td>CD: 44 kSa/s, 16 bit 704 kbit/s</td>
<td>8 kSa/s, 13 bit (EFR-codec) 104 kbit/s</td>
</tr>
<tr>
<td><strong>Compression</strong></td>
<td>Analogue compander 2:1</td>
<td>Digital source coding 9:1 (22:1 with AMR)</td>
</tr>
<tr>
<td><strong>Compressed Audio rate</strong></td>
<td>352 kbit/s</td>
<td>12 kbit/s</td>
</tr>
<tr>
<td><strong>Channel arrangements</strong></td>
<td>15 channels in 20 MHz</td>
<td>75 channels in 5 MHz</td>
</tr>
<tr>
<td><strong>Raw Audio related spectral efficiency</strong></td>
<td>0.5 bit/s/Hz</td>
<td>1.56 bit/s/Hz</td>
</tr>
<tr>
<td><strong>Compressed Audio related spectral efficiency</strong></td>
<td>0.25 bit/s/Hz</td>
<td>0.18 bit/s/Hz</td>
</tr>
</tbody>
</table>

## Findings
- Today’s analogue PMSE slightly more spectrally efficient than cellular
- PMSE is not a waster of spectrum - in light of high audio quality to be delivered
- Cellular high number of services mainly a consequence of heavy source coding
- Analogue PMSE already contains analogue source coding 2:1
- Near lossless Audio coding (e.g. SLQ) could do 4:1, not a dramatic gain to draw from digitization, but not HD
Spectrum considerations
What is the value of spectrum? – Legendary UMTS Auction

Value of spectrum: 98.8 Mrd DM for 60 MHz paired
⇒ 1,6 Milliarden DM/MHz
⇒ 800 Mil€/MHz
Spectrum considerations
What is the value of spectrum? – Recent UHF auction

Value of spectrum: 3.6 Mrd € for 30 MHz paired
⇒ 120 Mil €/MHz

Surprisingly, it only decayed 6x...
Spectrum considerations
Spectrum lost for PMSE

TV primary, PMSE secondary

GMS/UMTS/LTE
UL  DL

f/MHz

470 502 694 790 862 880 915 925 960

Digital Dividend I
-20%

Future Mobile
Communication primary

Digital Dividend II
-24%

Further Digital Dividend -8%

PSS Public Safety and Security
e.g. TETRA

If primary assignment is lost,
secondary is also immediately lost

Σ -52%
8. Conclusions

- *We list the take-aways*
- *Action recommendations*
Conclusions
Key Take-Aways

How can spectrum need be reduced?
• More efficient transmission technology?
• We are already at the Shannon bound, so not a realistic opportunity
• Analogue to Digital transition will not give us a more efficient spectrum use!
• For digital transmission protocol overhead makes the case even worse

But why are other digital systems doing that well?
• Information to be transmitted is heavily compressed by source coding
• More compression means less spectrum need

Can’t we apply compression to PMSE?
• PMSE has very specific needs (latency, drop-outs, quality)
• Only small compression factors can be used 2:1 …4:1
• Recipient is not known (Source coding needs information about recipient)
• Digital archive must be high quality to derive different quality levels
• Analogue transmission already has 2:1 by compander system
• So benefits form digital are therefore small

Consequences
• Spectrum need by PMSE is somehow justified – PMSE is not a waster of spectrum
• Research on lossless compression needed (EU projects?)
• No revolutions to come in reduction of spectrum need for PMSE
• RF Technology improvements can also help a bit
Conclusions
Key Take-Aways

Higher compression of raw information

Less spectrum needed per user

More users can transmit in a given spectrum

Consequence: We have to maximize compression

High Compression in PMSE is not possible in principle due to specifics of PMSE,
Analogue PMSE has already analogue compression
Digital compression would only provide marginally higher compression
Overhead for digital would have to be paid, costs even extra spectrum

PMSE cannot do in less spectrum,
Raising number of users in PMSE as in other systems
Raising interest for HD in PMSE as in other systems

Spectrum need by PMSE is justified