Current and future developments in loudspeaker management systems

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Current trends

- Embedded devices (Self powered speakers)
- AV-Networks

- Digital I/O
  - AD / DA conversion will not be the bottleneck for the dynamic range anymore

- Increasing MIPS allow
  - More algorithms simultaneously
  - More sophisticated algorithms

- Complex loudspeaker concepts

- More information on loudspeakers
  - Consider directivity measurements
Signal processing

- Crossovers, Filters, EQs

- Strengthen the weak link in the reproduction chain

- Must not introduce distortion, delay or noise
LTI System / Transfer function / Impulse response

- System $g$ introduces unwanted behaviour

\[ x(t) \quad g(t) \quad y(t) \]
Introduce Inverse Filter

- Inverse filter h corrects for unwanted behaviour of g

\[ x(t) \xrightarrow{h(t)} y(t) \]
### Convolution

#### Time domain

\[ x(t) \rightarrow h(t) \rightarrow y(t) = h(t) \ast x(t) \]

\[ \mathcal{L} \text{aplace} \]

\[ X(s) \rightarrow H(s) \rightarrow Y(s) = H(s) \cdot X(s) \]

\[ \mathcal{L} \text{aplace} \]

#### Frequency domain

- Straight-forward implementation in the frequency domain is more efficient but introduces latency
FIR Filter and Impulse Response

- FIR filter in the time domain => Imp. Resp. = filter coefficients
Myth: FIR filters have a high group delay

- 3 FIR filters
- Same magnitude response
- Different phase responses
With lower resonance frequencies, the impulse responses get longer => more coefficients necessary
IIR Filters to the Rescue

- Besides all well-known advantages and disadvantages:
  - Infinite impulse response ⇒ constant calculation power over frequency
  - No direct access to impulse response / phase
  - More sensitive for quantization noise
Polyphase Filter Bank Approach

- Division of the signal in M frequency bands
- Synthesize filter by setting gain and phase per band
- Reconstruct signal
- Constant calculation power regardless of number of filters
MESA / Raised Cosine Filters

Gain (dB)  

Boost=4  

F_{lo}=-2  

W_{lo}=2  

W_{hi}=3  

F_{hi}=2.5  

Frequency (octave relative to 1KHz)

31.25  62.5  125  250  500  1K  2K  4K  8K  16K  32K  Frequency (Hz)

Freq Resp. Lake Contour GEQ 500-2k +12dB (einzel und alle)

Freq Resp. dbx 4800 graphischer Terz EQ 500 2k +15dB
Equalizing with FIR filters / Consider phase of speakers

Derive filters from measurement of speakers (amp. and phase)
The FIR filters itself are not necessarily linear phase but the whole system of crossovers and loudspeaker is
Example / Response to Double Rect Pulse Input

- **louder speaker responses**
- **overall system response**
- **FIR filter responses**
- **rect pulse response**
- **min. phase equalization**
- **Complex equalization**
Phase Responses

- Linear phase of total system to very low frequencies possible
- The lower the frequency the higher the latency
- Compromise for non-playback situations: Mixed equalization with minimum and linear phase of total system

Identical mag. responses for all three options
Multirate Processing

- FIR filtering at low frequencies requires many coefficients
  - Example: Filter with 1024 taps at 48 kHz
  - \( \frac{48000 \text{ Hz}}{1024} = 46.9 \text{ Hz} \) resolution at 49 MIPS
    - At low frequencies, spectral line density is too low to define filter
  - Downsampling with factor 16:
    \( \frac{48000 \text{ Hz}}{16} / 1024 = 2.9 \text{ Hz} \) resolution at 4 MIPS
Extended Capabilities

- Overlapping frequency bands easily realisable
- Influencing directivity
- Phase of filters calculated from speaker measurements
- No struggling at overlapping regions to match phases
Example: Directivity with passive and FIR crossovers

Improvement of system with passive crossovers (top) by using FIR filters with overlapping frequency bands (bottom)
Conclusion / Outlook

- More and more vendors implement FIR filters in LMS
- At low frequencies only useful with multi-rate processing
- Useful for directivity control
  - Plays a more and more important role
- Future: Fast convolution algorithm?
  - Combination of time- and frequency-domain convolution with no algorithm-inherent latency