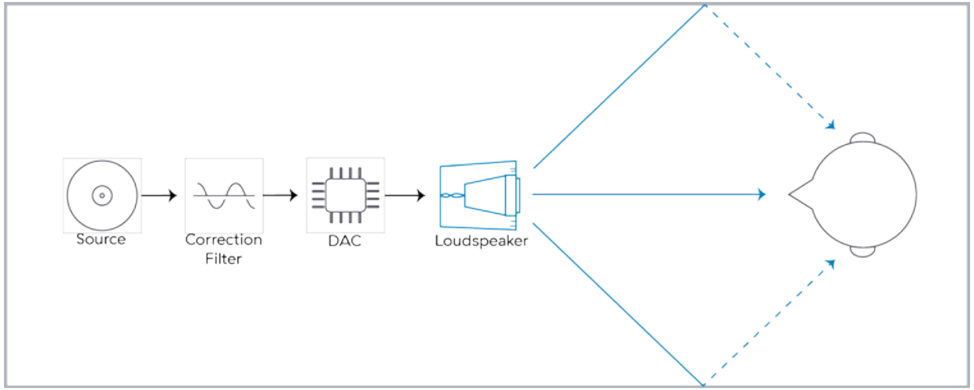

Dirac

Digital Room Correction

Benefits, Common Pitfalls and the State of the Art

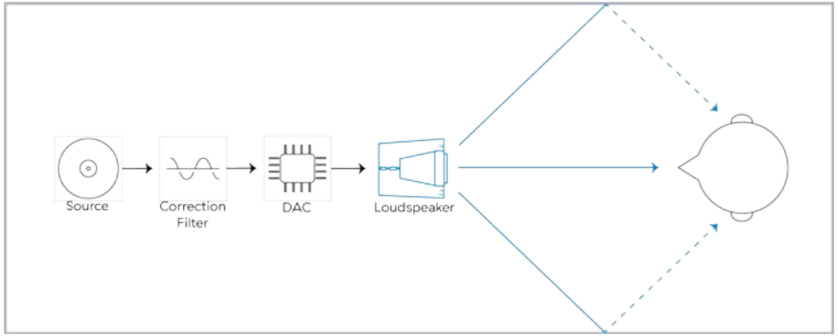
Lars-Johan Brännmark, PhD
Dirac Research AB, Uppsala, Sweden

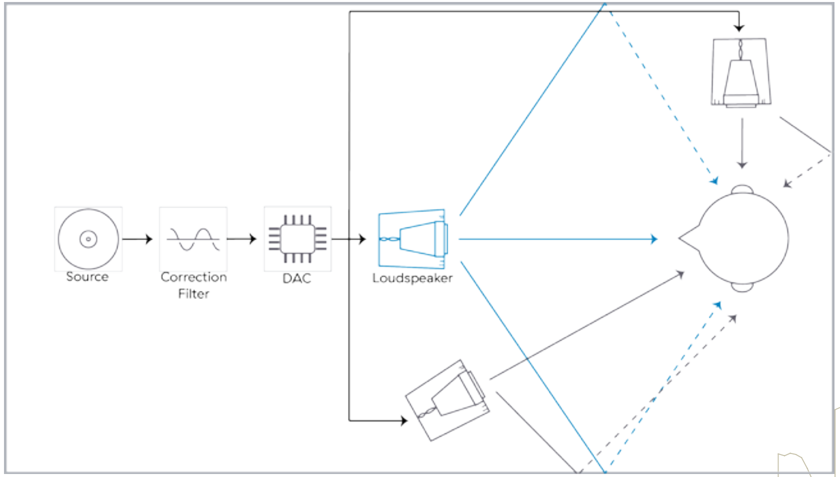
Introduction



What is loudspeaker/room correction?

- Loudspeaker signals are pre-processed, to compensate for time- and frequency distortions in the loudspeaker/room signal path.
- The correction filter constitutes, in some sense, an inverse of the loudspeaker/room signal path.
- Filter design is based on measurements of the loudspeakers, either anechoically or in the listening environment.

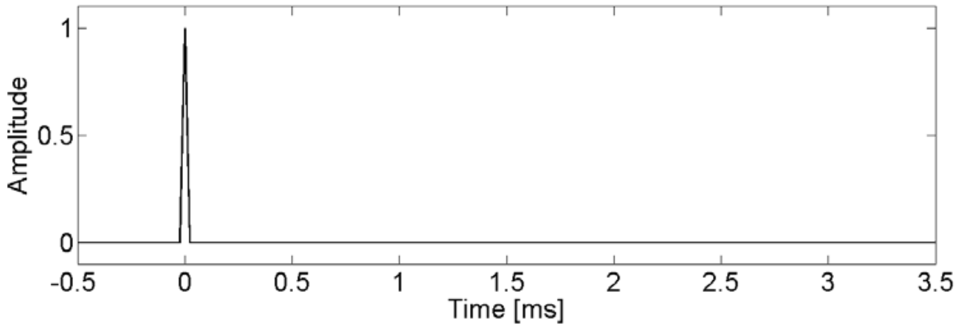




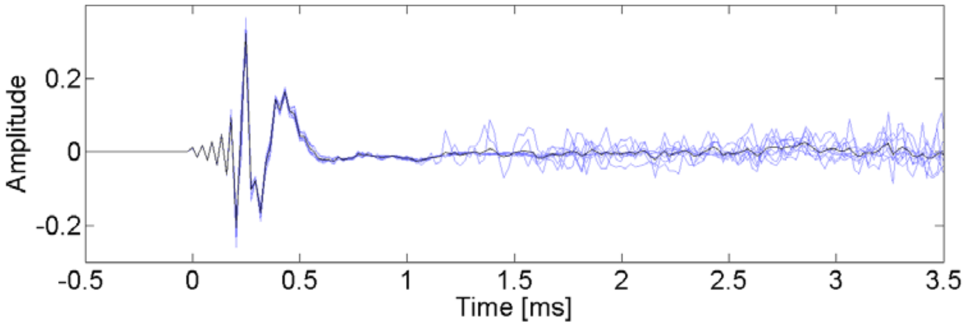
Room correction may employ either single- or multichannel filtering techniques.

Theoretical Basics

Loudspeaker input signal: A unit impulse



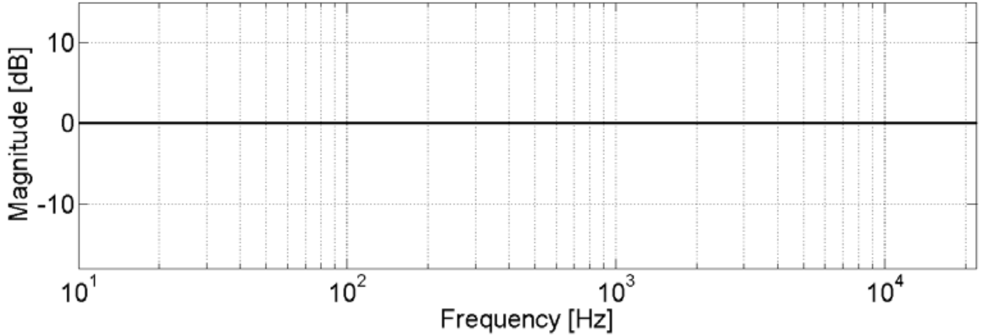
Measured output in 9 positions: Acoustic impulse responses



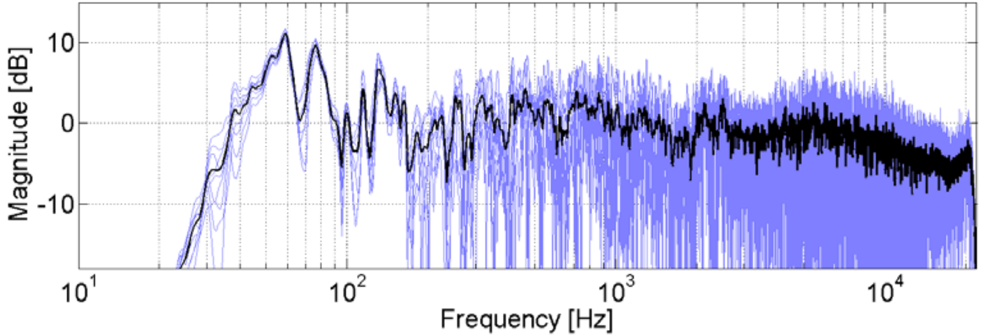
Impulse Response:

Output of the system, when the input is a unit impulse.

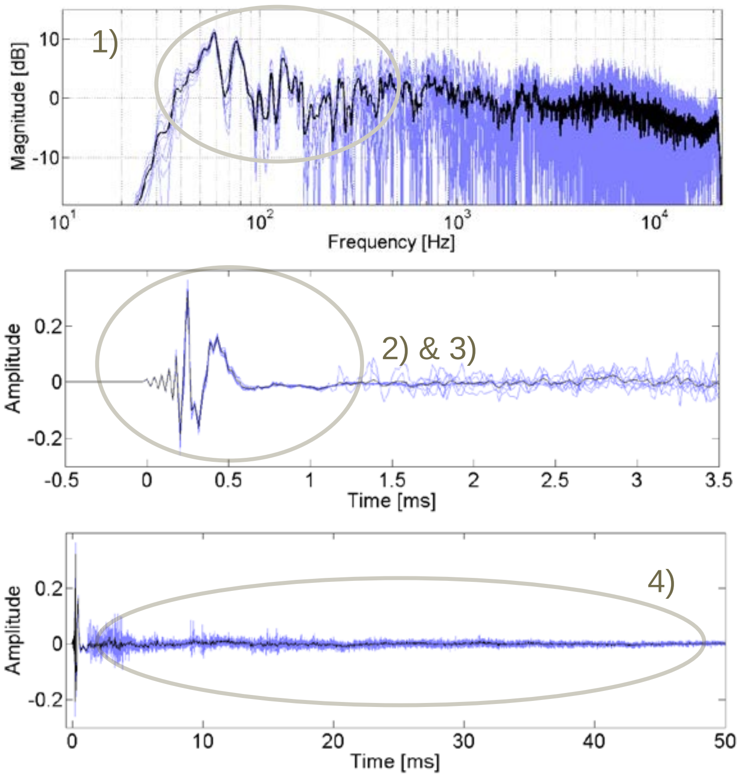
Loudspeaker input signal: A white spectrum



Measured output in 9 positions: Frequency responses



Frequency response:
Amplification of the system, as a function of frequency.



Loudspeaker – Room Distortions

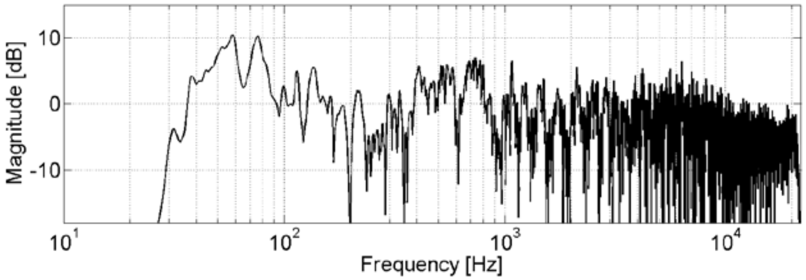
1. Room resonances/standing waves.
   Low frequencies, large wavelength, late sound, all directions.
2. Misalignment of drivers.
   Mid to high frequencies, direct sound, same direction as speaker.
3. Early reflections/cabinet diffraction. Mid to high frequencies, early sound, same direction as speaker.
4. Late reflections/room reverberation.
   Mid to high frequencies, late sound, all directions.

About Phase: Minimum vs. Mixed Phase

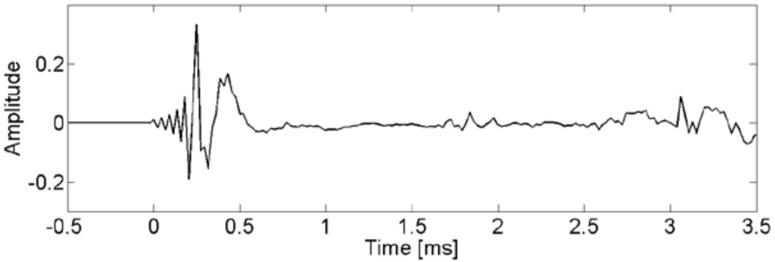
- The frequency magnitude response alone does not convey all information contained in an impulse response.
- The magnitude response together with the *phase response* (phase shift as a function of frequency) fully describes the system at one point in the room.
- Infinitely many phase curves can be combined with a given magnitude curve. Each phase/magnitude combination has a corresponding unique impulse response.
- For a given magnitude curve there is a unique phase curve that results in an impulse whose energy delay is minimum. A system having this phase characteristic is a *minimum phase system*.

Comparison: Minimum Phase vs. Mixed Phase

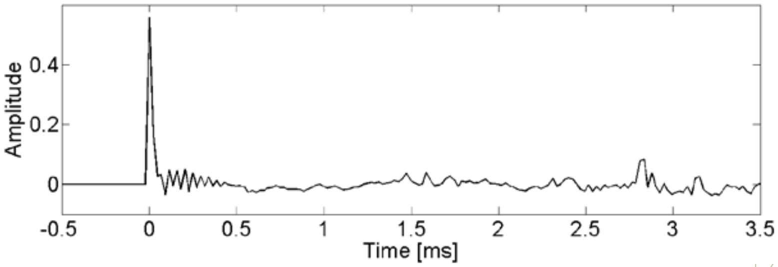
Measurement in one position, frequency response



Impulse response, original version (mixed phase)

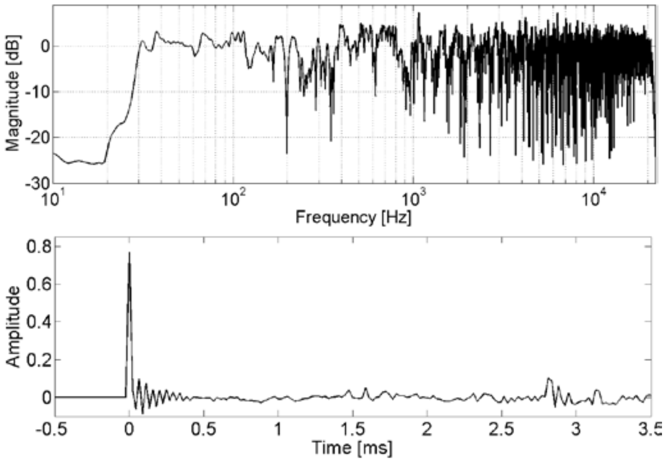


Impulse response, minimum phase version

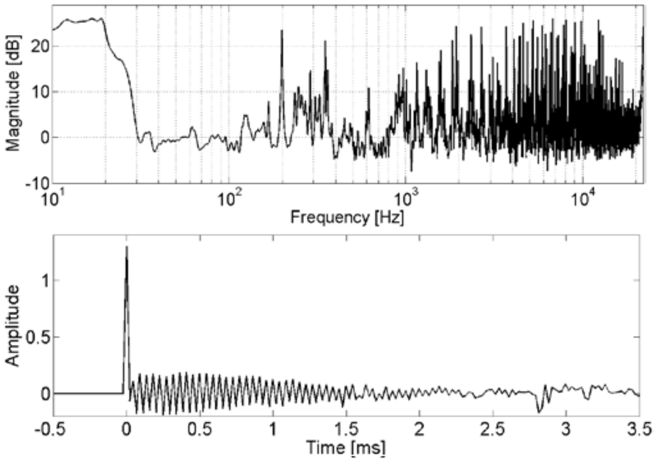


About Phase: Minimum vs. Mixed Phase

- In general a minimum phase system has a good impulse response if it has a reasonably flat frequency response.
- Below is a minimum phase system whose frequency response is similar to a loudspeaker in a room:



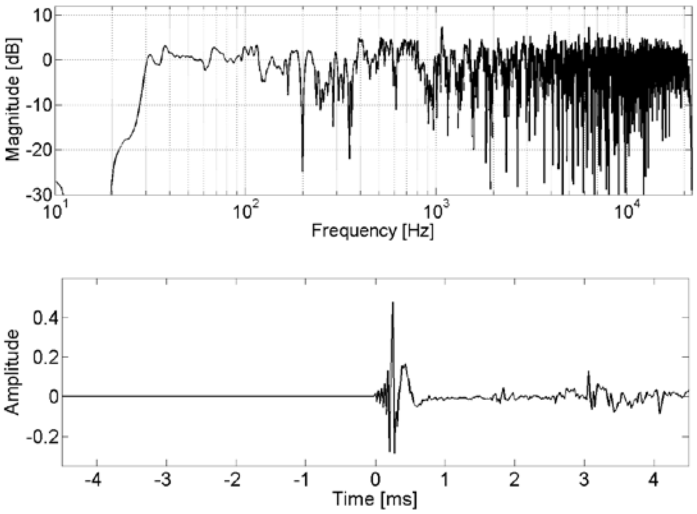
- The inverse of the minimum phase system is also causal and minimum phase. Thus, a minimum phase system can be corrected (inverted) by causal minimum phase filter.
- However...



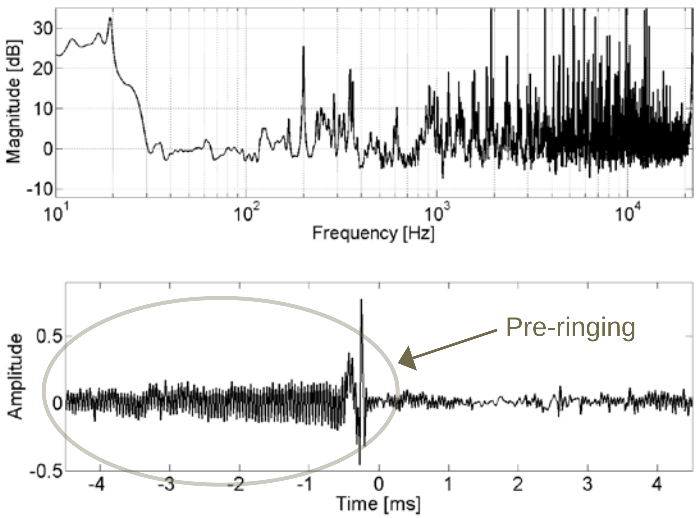
About Phase: Minimum vs. Mixed Phase

However, loudspeakers and rooms are mostly not minimum phase, i.e., they are mixed phase.
Non-causal filters are required for correction.

Mixed phase system: Loudspeaker in a room

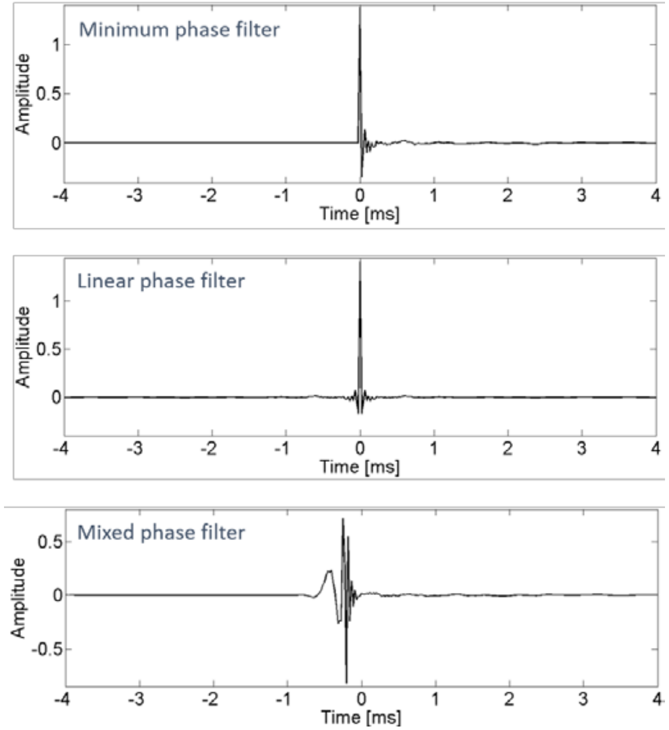


Inverse of mixed phase system



Inverse Filter Design Issues

- Solution to non-causality: A time shift, i.e., introduce an extra delay through the system. However, the resulting filter contains pre-ringings that may be audible.
- Additional problems: System variations and power handling capacity
  - Spatial variability: The loudspeaker–room system varies with listener position.
  - Non-stationarity: The fine details of the impulse/frequency responses vary slightly with time.
  - Power capacity: Risk of loudspeaker saturation/signal clipping if weak parts of the frequency response are inverted.
- “Perfect” corrected system response in one single point most likely implies degradation everywhere else.
- “Perfect” correction in one point is infeasible anyway, due to weakly nonstationary room dynamics and limited power capacity of speakers.



Inverse Filter Design Issues

- If inverse filter is minimum phase:
  - Pre-ringings are avoided, but correction is magnitude-only; phase is not corrected.

- If inverse filter is linear phase:
  - Pre-ringings are introduced that may be audible. Phase is not corrected.

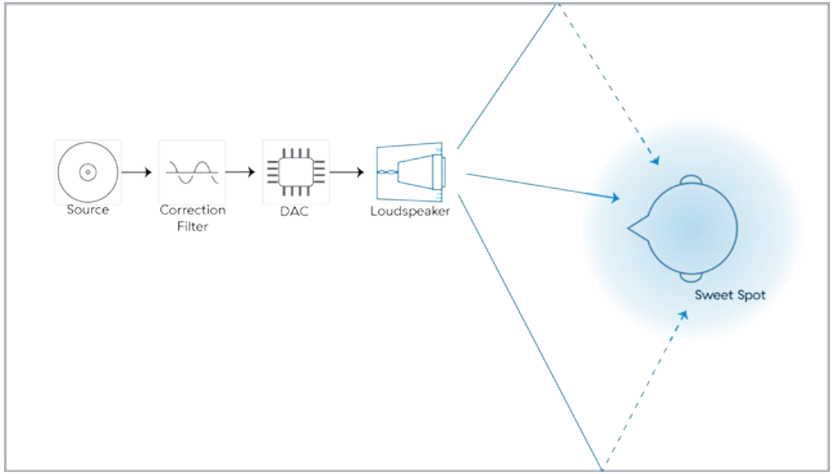
Dirac's solution: ”Cautiously” designed mixed-phase filters. Pre-ringing of filter is matched to the spatially common part of speaker's response.

Dirac's approach: Robust inverse filter design

- Improve the impulse and frequency responses, under constraints:
  - A limit on the maximum allowable gain of the filter.
  - Filter must not introduce any residual audible pre-ringings after correction.
  - Filter must be robust to listener movements and perform well over time.
- Single-channel (Dirac Live) and Multichannel (Dirac Unison) approaches
- Theoretical basis: Discrete-time automatic control
- Foundations developed in 1960s – 1980s by Kučera, Youla and the "Swedish school of control theory": Åström, Ljung, Söderström et al.
- Key mathematical tool: Polynomial matrix equations.
- Contributions in 1980s – 1990s by Ahlen, Sternad (professors at UU and co-founders of Dirac).
- Application to audio/acoustics by Brännmark (PhD 2011), Bahne (PhD 2014), Barkefors (Tekn. Lic. 2014) Current PhD students: Widmark, Gunnarsson.

Dirac Live®

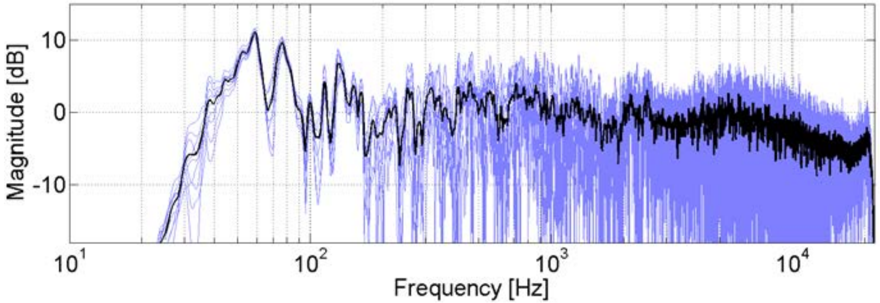
Single-Channel Correction: Dirac Live



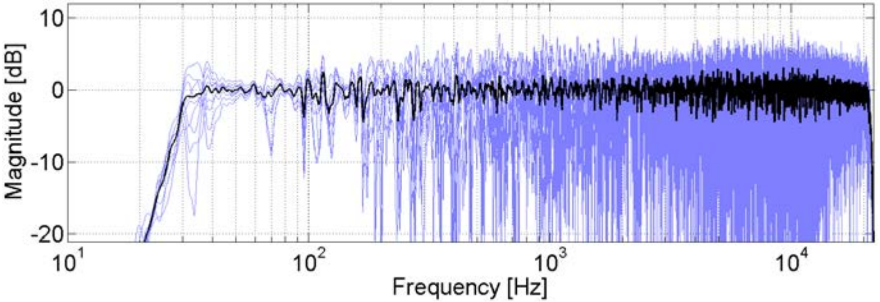
Dirac Live: Our most widely used technology

- A precompensation filter is designed for each speaker in a system.
- Based on measurements in a sweetspot (typically ≈ 9 points).
- Corrects impulse and frequency response behavior in the sweet spot, under gain- and pre-ringing constraints.
- Single-channel filter implies one spatial degree of freedom:
  - Behavior that is common over the sweetspot is corrected.
  - Variations within sweetspot cannot be eliminated.

Original frequency responses in sweet spot, 9 positions



Dirac Live: Frequency responses in sweet spot, 9 positions

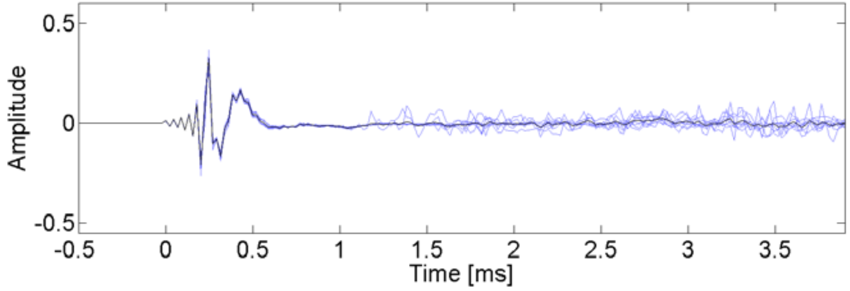


Dirac Live Performance: Frequency Domain

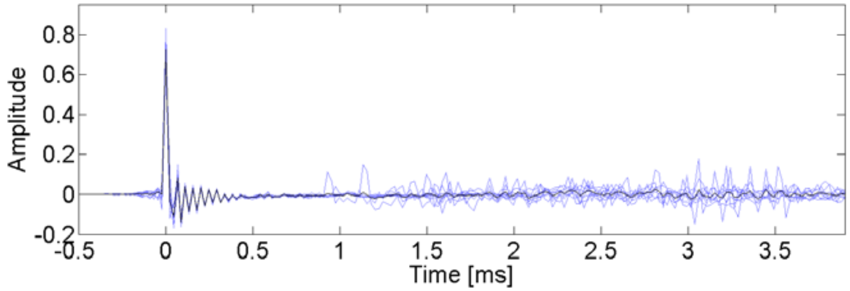
Note 1: Common behavior is significantly improved. Variations between spatial positions remain.

Note 2: In this example, the target frequency response was flat. In general other shapes of the target are desired.

Original impulse responses in sweet spot, 9 positions and average



Dirac Live: Impulse responses in sweet spot, 9 positions and average

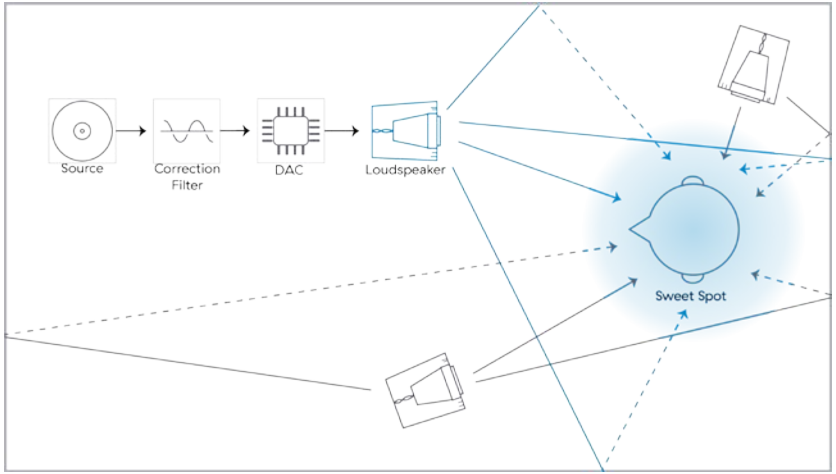


Dirac Live Performance: Frequency Domain

Note: Common behavior is significantly improved. Variations between spatial positions remain.

Dirac Unison®

Multichannel Correction: Dirac Unison®



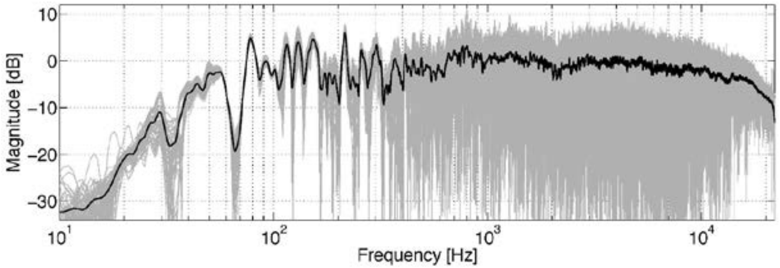
- Speaker cooperation: Each main speaker is helped by one or several support speakers, to attain a target response.
- Superposition of several speaker responses implies several spatial degrees of freedom:
  - Optimal alignment of speakers, e.g., fullrange channels and subwoofers.
  - Spatial variations are reduced.
  - Full dereverberation possible at low frequencies.
- Based on measurements (≈ 9 points) of all speakers in a sweet spot.

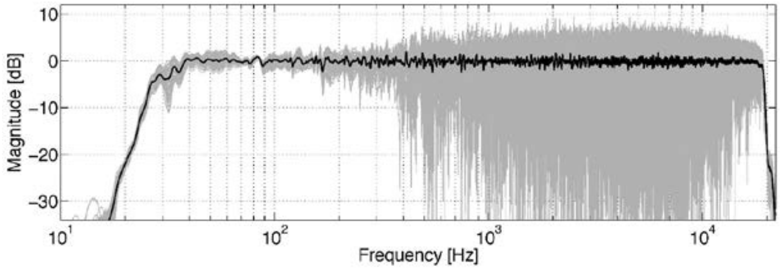
Characteristics of Dirac Unison

- A set of main speakers are selected by the user.
  - For example the main speakers can be the 5 main channels of a home cinema system: L,C,R,Ls,Rs.
- To each main speaker, a set of support speakers are selected.
  - If a main speaker has no support, then Unison = Live for that speaker.
  - A support speaker can be one of the other main speakers, or it can be a dedicated support speaker, for example a subwoofer.
  - The frequency range where speakers are allowed to operate as support is typically limited to below 500 Hz.
- Goal: Improve the time- and frequency response of the main speaker, using all of the assigned loudspeaker resources.
- The sweetspot size, the number of required support speakers, the upper support cutoff frequency, and T60 of the room are related.

Dirac Unison Performance: Frequency Domain

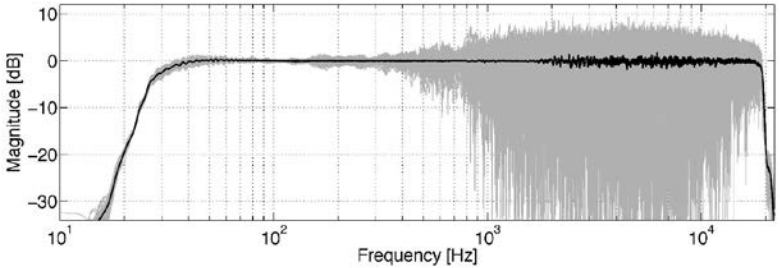
Left front speaker, original frequency responses measured in 64 points.





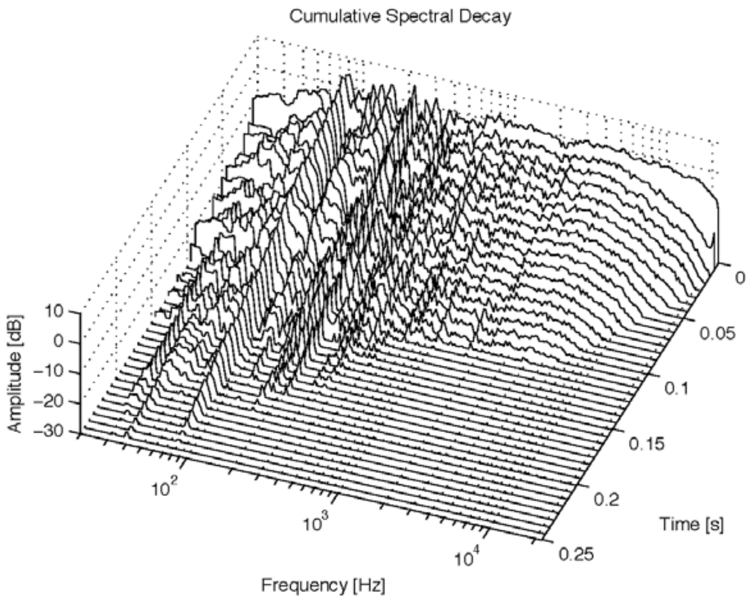
Left front speaker, compensated with Dirac Unison, 5.1 setup (4 speakers and 1 subwoofer are used as support.

Left front speaker, compensated with Dirac Unison, 14.2 setup (13 speakers and 2 subwoofers are used as support.



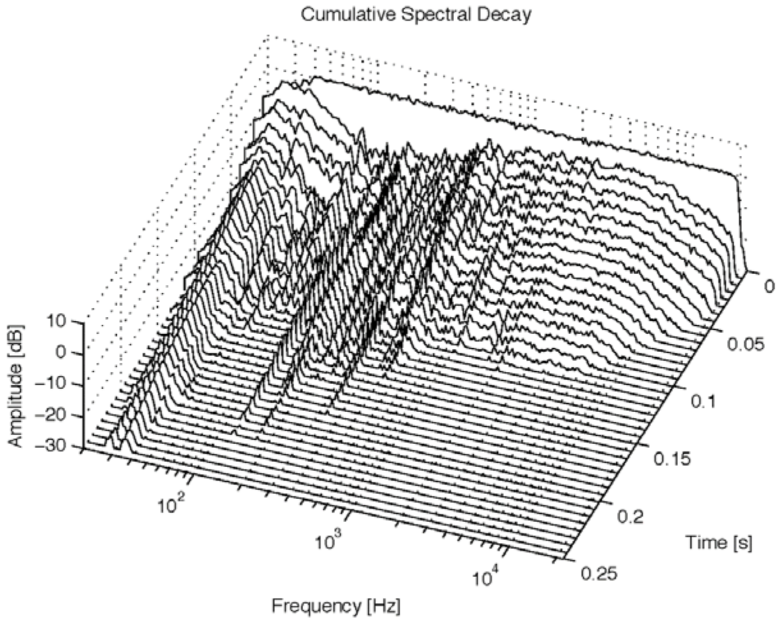
Dirac Unison Performance: Time Domain (Waterfall)

Left front speaker,
original measurements.



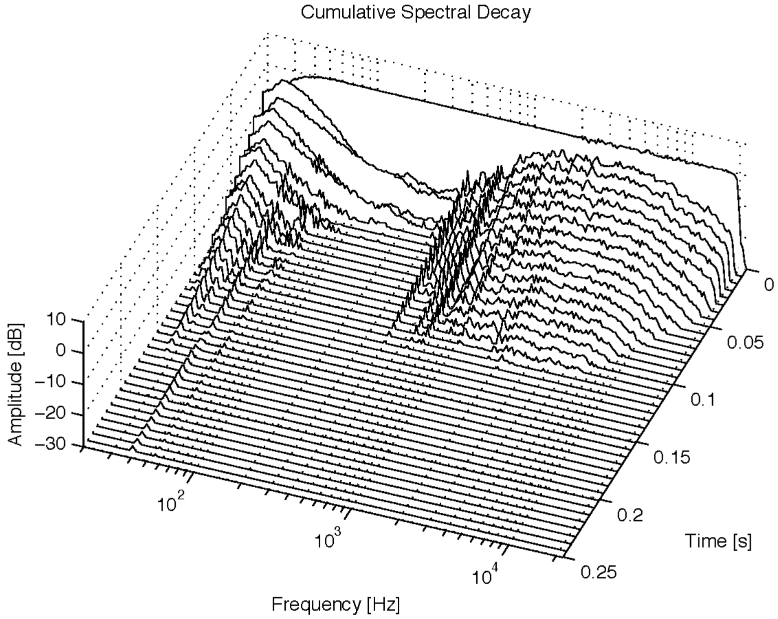
Dirac Unison Performance: Time Domain (Waterfall)

Left front speaker, compensated with Dirac Unison, 5.1 setup (4 speakers and 1 subwoofer are used as support.



Dirac Unison Performance: Time Domain (Waterfall)

Left front speaker, compensated
with Dirac Unison, 14.2 setup (13 speakers
and 2 subwoofers are used as support.



Conclusion

- Standard room correction can improve the average response, but not variations across space.
    - Time-domain problems can be reduced only by careful mixed-phase design.
    - Most systems use minimum phase filters, thus neglecting the time-domain behavior.
- With Dirac Unison, additional speakers are used in synchronization with the main speaker to also remove the variations and further reduce the time-domain problems.
- Common pitfalls and challenges that haunt many attempts at room correction include
    - Spatial variability: Single-point measurements are insufficient for characterizing the room.
    - Target curve: A proper in-room target needs to consider the direct-to-reverberant ratio.
    - Time domain correction:
        - ✓Equal time-domain responses from all speakers are crucial for good imaging and staging (inter-aural correlations, etc, are key to human perception of localization).
        - ✓Mixed-phase correction is required, but causes audible pre-ringings unless carefully designed.
        - ✓Good time-domain correction focuses on the direct wave, very early parts of the impulse reponse and low frequencies. Late high-frequency parts can typically only be improved by passive room treatments.

Thank you!

Dirac's room correction software is available at:
www.dirac.com/online-store