

The WFS system at La Casa del Suono, Parma

Fons Adriaensen
Casa della Musica, Parma

Linux Audio Conference 2010
Utrecht, Holland



- A museum dedicated to the history of audio technology.
 - * Created by the City of Parma.
 - * A collection of vintage audio equipment.
 - * A view on current and future technology.
- Two audio installations using Linux:
 - * The 'Lampadario Acoustico'
 - * The 'Sala Bianca'

- A room of approx. 7.5. by 4.5m, entirely white.
- 189 small speakers around the full inner perimeter, including the doors.
- All speakers driven individually, together they form a Wave Field Synthesis system.



- One speaker every 12cm.
 - * Constructed in blocks of 10, 15, or 17.
 - * Two-way bass-reflex with mixed order crossover.
 - * Designed by Audio Link, using components from Ciare.
 - * Very good performance for its size.
 - * Frequency range is 50Hz to over 20kHz.





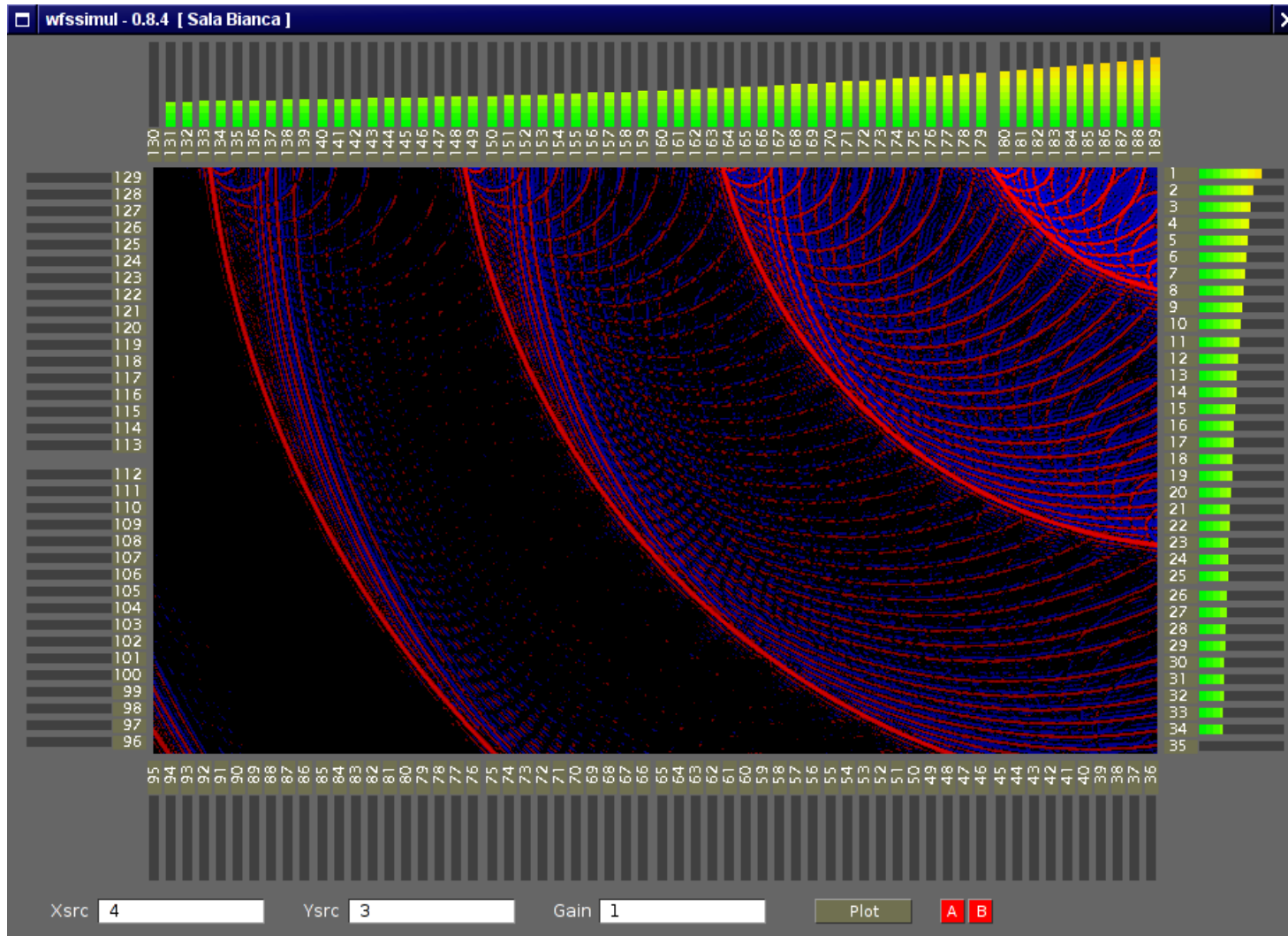
- Based on the Kirchoff-Helmholtz integral:

The wave field inside a source-free volume V delimited by a surface S is completely determined by either the pressure or the volume velocity at all points of the surface S

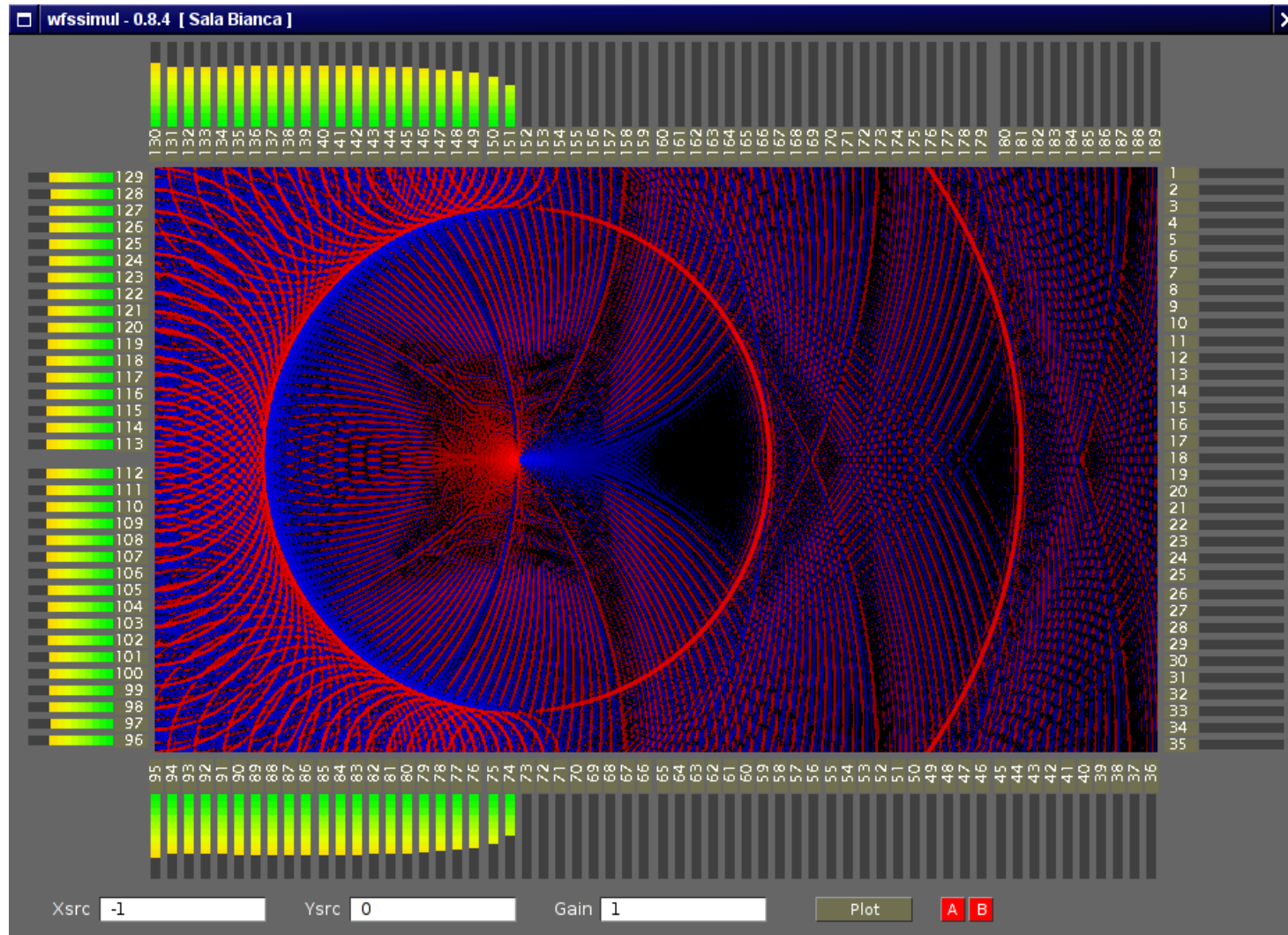
- Practical realisation requires some compromises:
 - * Replacing the infinite number of points by a grid of discrete sources.
 - * Reduction from 3D to 2D.
- WFS recreates the sound field of any number of *primary* (virtual) sound sources using an array of *secondary* (real) sources.
- A variation of the technique can be used to create sources *inside* the volume or perimeter.



- The Sala Bianca system will support up to 48 moving primary sources, rendered in real time.
- Source movement is implemented by sample-rate update of rendering parameters, not by crossfading between static source positions. This means that e.g. the Doppler effect is rendered as well.
- The system is used
 - As part of the museum, for public demonstrations of the technology.
 - For scientific research: as a listening room allowing virtual speaker setups, and for further development of WFS algorithms.
 - As an *instrument* for electro-acoustic music concerts.
- Planning started in 2007, and the system is in use since February 2009.



A primary source at $X = 4$, $Y = 3$.



A primary source inside the perimeter.

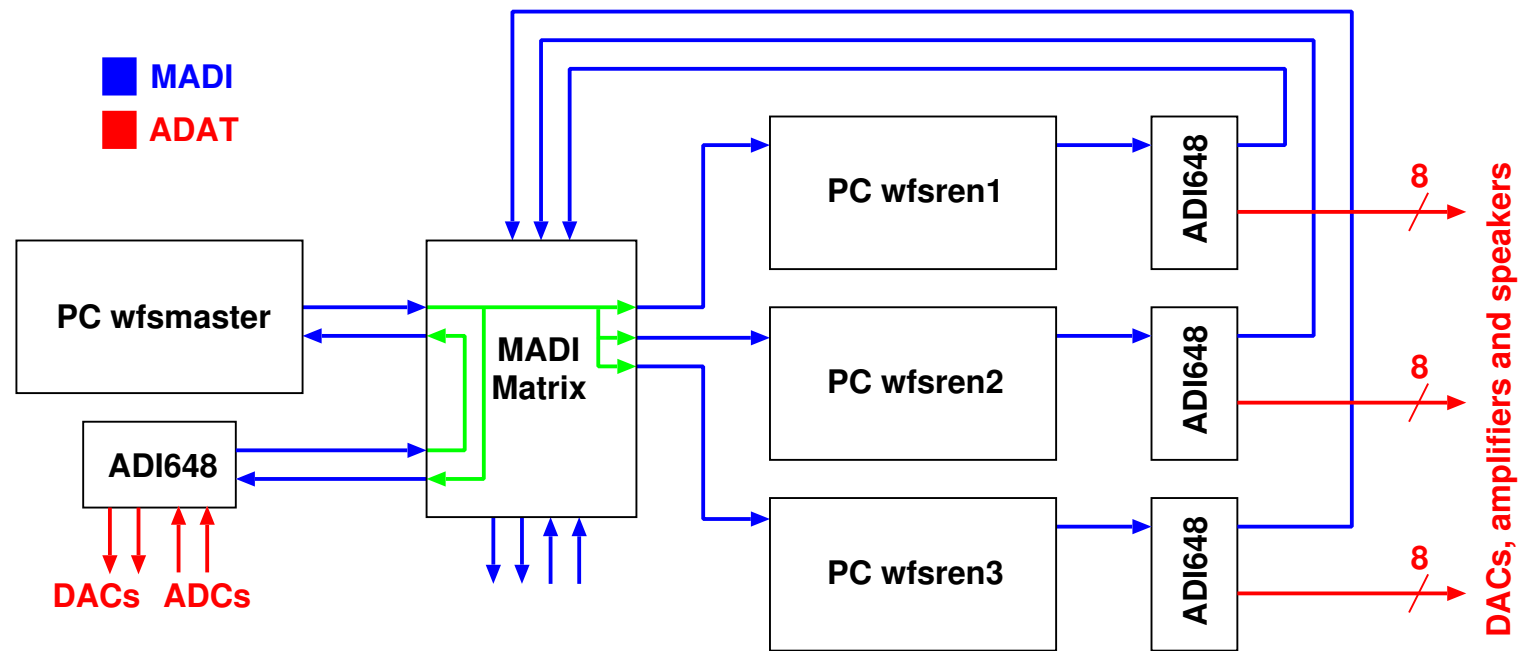


- 4 Siemens/Fujitsu PCs running ArchLinux
- RME MADI interfaces, convertors and matrix.
- 192 channels of Aphex DA converters.
- 24 8-channel QSC amplifiers.
- Lots of cables.

- Originally meant to be an Ambisonics room at double the size.
- Contains all PCs, (also for the Lampadario), network HW,...
- Mic preamps and analog audio lines to Sala Bianca, for recording, concerts, etc.



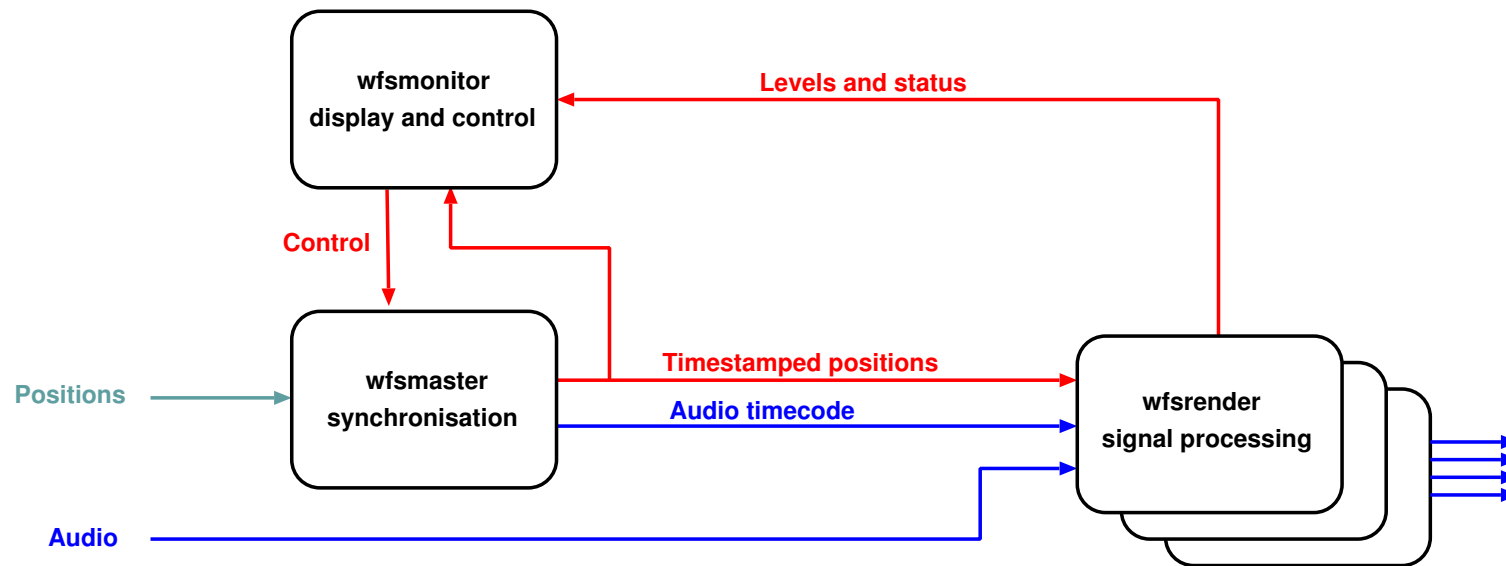
The audio system – Structure





Using a Gigabit network for audio connections instead of MADI was considered but rejected.

- The synchronisation issues could be solved easily.
- Solutions were emerging, but untested for e.g. 48 channels.
- Performance of the hardware and drivers was not at all guaranteed.
- It would probably result in larger system latency.
- It would not permit major savings for the audio hardware.

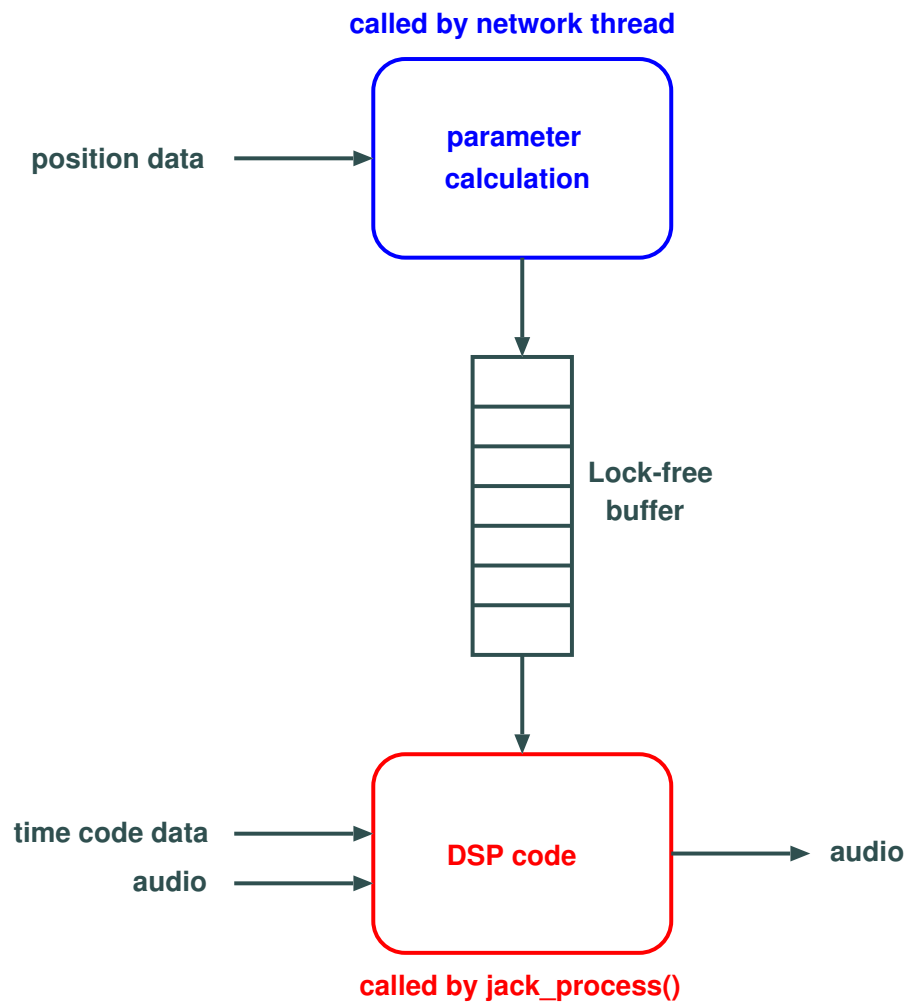


- Command line application, controlled via WFSmonitor or Python supervisor.
- Provides a single access point for source position and movement commands.
- Translates these commands into periodic updates sent to the render applications.
- Ensures **audio and data synchronisation** in the entire system:
 - * Source movement data must be applied with sample accuracy on all machines, but
 - * jack periods on the render machines are not synchronised, and
 - * position updates arrive asynchronously via the network.
- * An audio time code generated by WFSmaster is used as a time reference by the render machines, and
- * position updates are timestamped using this code.

- Command line application running on all rendering machines
- Controlled and monitored via WFSmonitor.
- Receives mono source signals and audio time code via the audio interface.
- Receives source position commands via the network
- Computes signals for up to 64 speakers.
- All instances are equal and read hostname to determine the set of speakers to use.
- Major parts are implemented as plugins:
 - * The [layout](#) plugin defines system geometry.
 - * The [engine](#) plugin defines the DSP algorithms.



- Defines the geometry of the speaker array.
- Provides functions to read or calculate:
 - * Speaker coordinates,
 - * Vectors orthogonal to the line of speakers,
 - * Distance of a point to the line of speakers,
 - * Inside or outside determination of source locations,
 - * Calibration data and parameters.
- Also allows linear (non-closed) arrays.

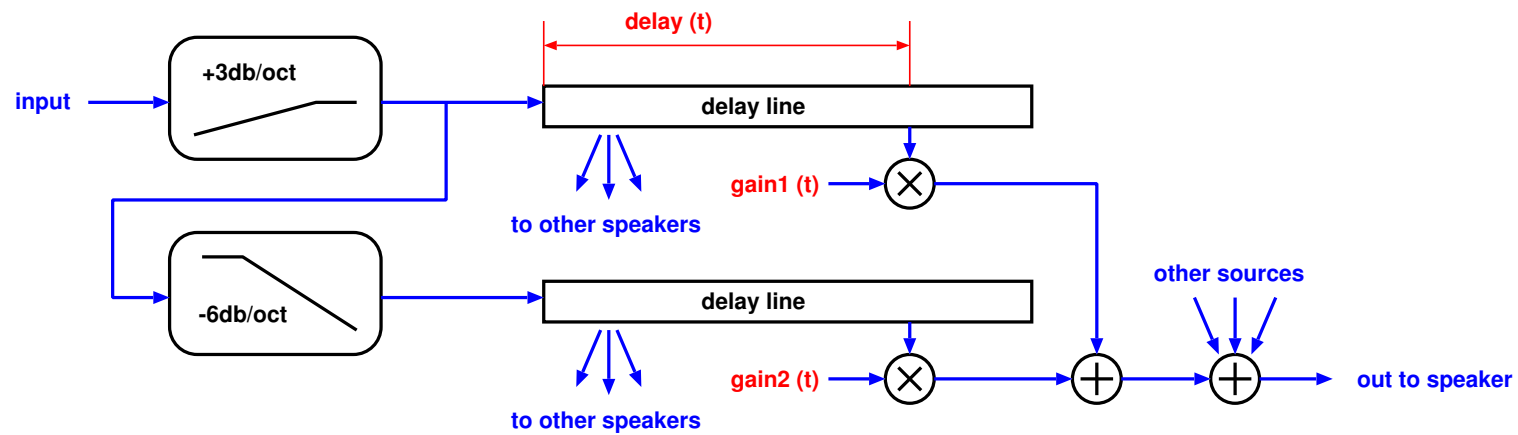


- Contains two main routines:
 - **Parameter calculation** called by the network receiver thread.
 - **DSP code** called by `jack_process()`.
- A lock-free buffer is required between the two.



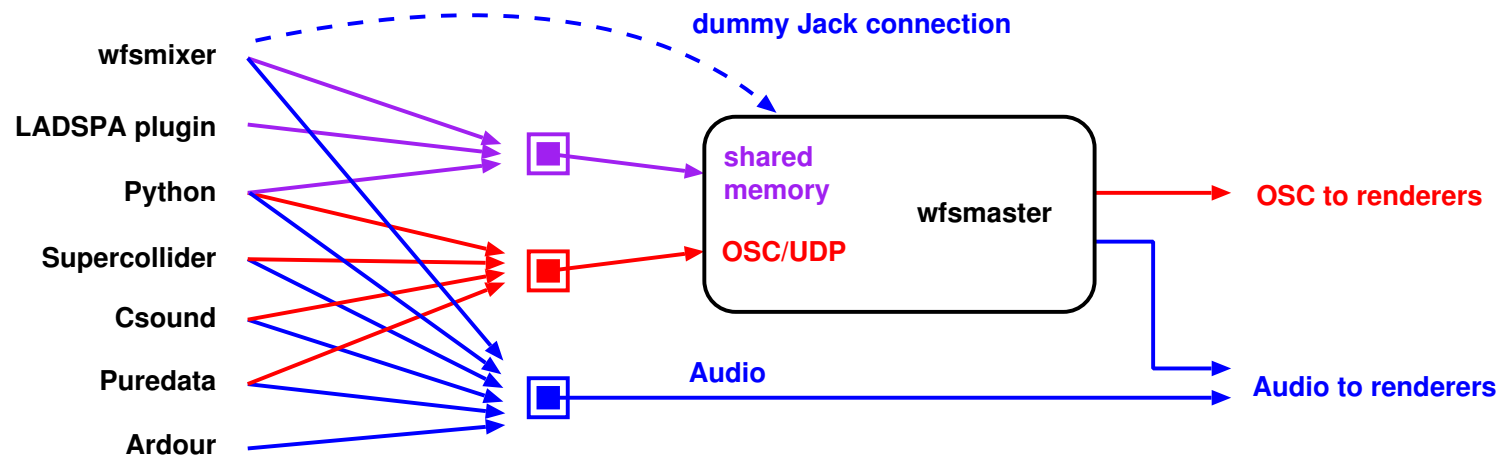
- Control rate calculation of rendering parameters.
 - Each combination of (primary, secondary) source depends on a specific delay and filter.
 - The filter can be decomposed into two fixed filters combined with variable gain factors.
 - This results in a delay value and two gain factors for each combination of (input, output).
 - Control rate is variable, but normally set to 1024 samples (around 25ms or 40 updates per second).
- Audio rate calculation:
 - Two filters for each input.
 - Implementation of the sample-accurate synchronisation.
 - Linear interpolation of the the three rendering parameters.
 - Calculation of the speaker driving signals.
 - An optional correction filter for each output.
 - Calculations can be optimised for stationary sources.

Three parameters for each (input, output) pair, updated at sample rate: *delay*, *gain1*, *gain2*.



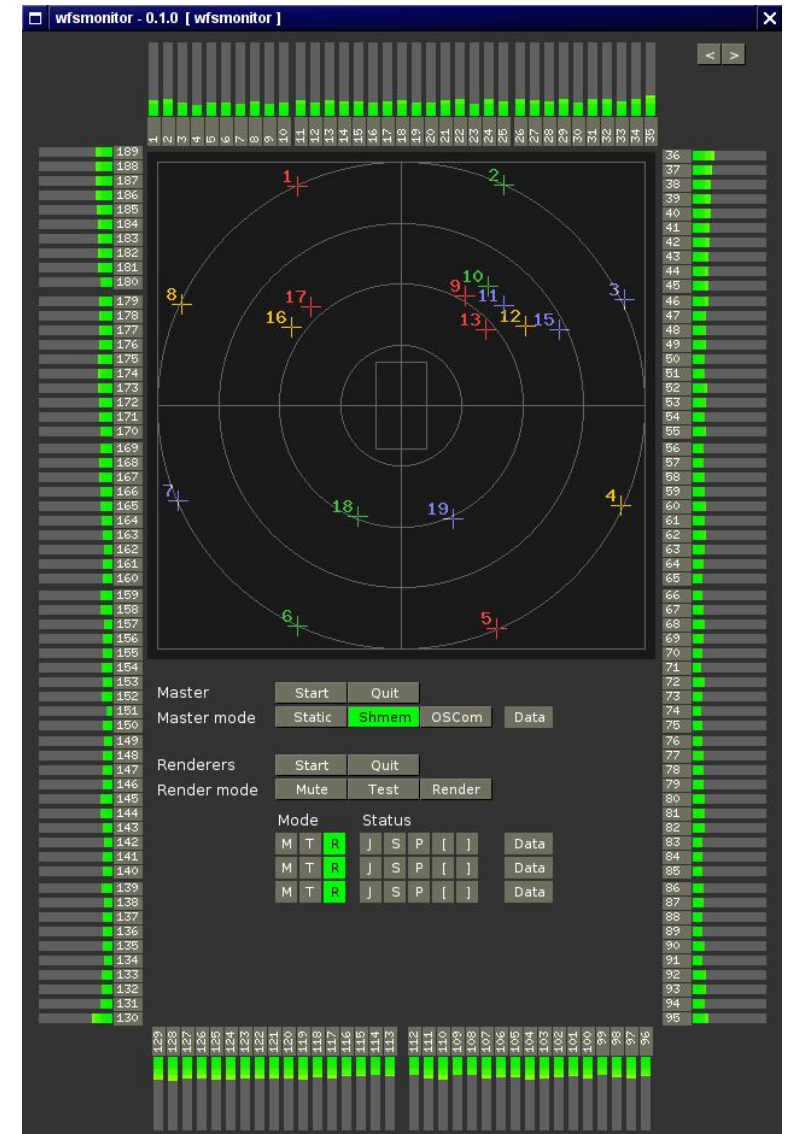


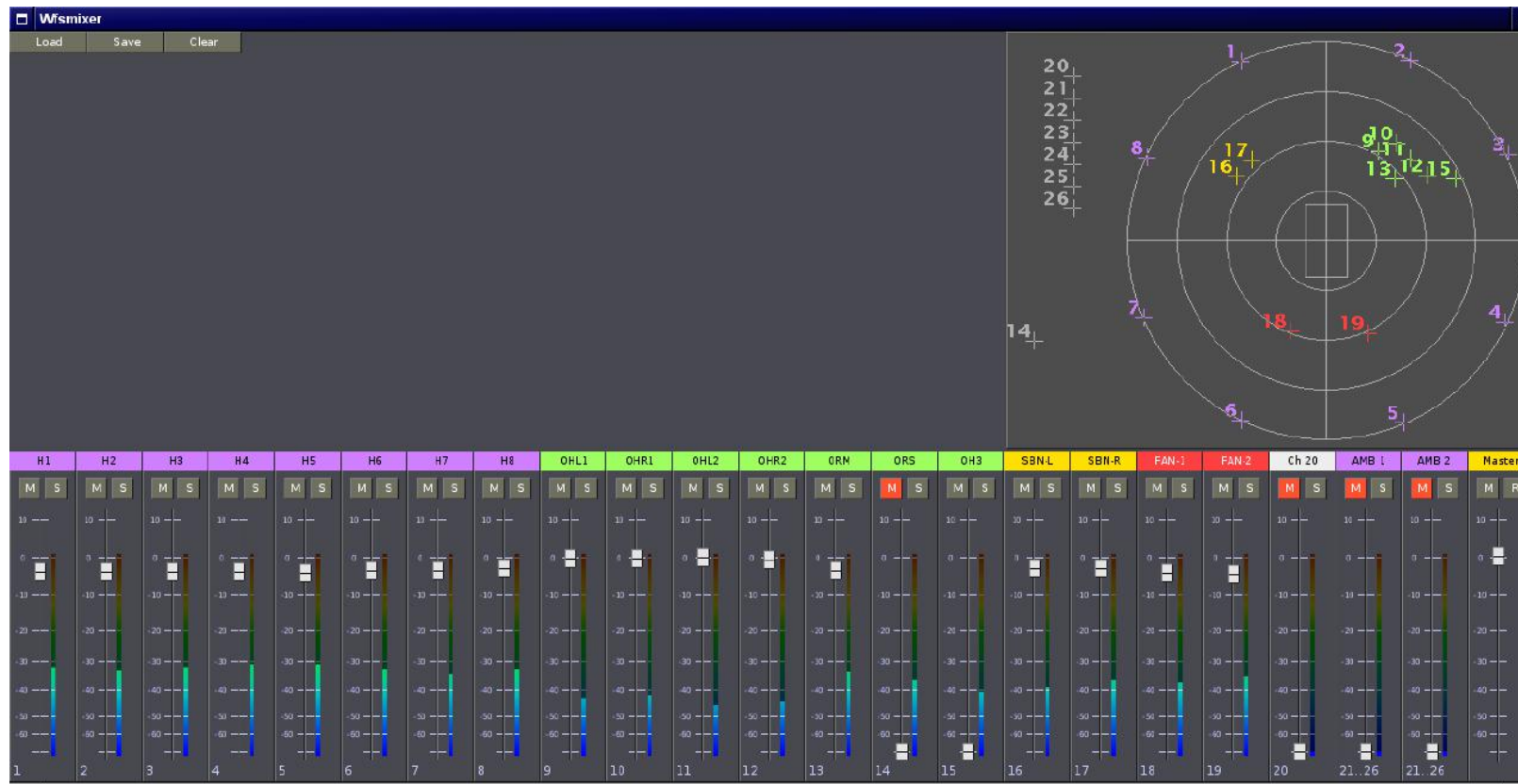
- Primary source position can be controlled via two interfaces:
- Shared memory interface.
 - Available only on the *wfsctl* computer.
 - Allows sample-accurate synchronisation of audio and control data.
 - Used by LAPSPA plugins, WFSmixer and Python code.
- OSC interface.
 - Available anywhere on the network.
 - Used by major composition and synthesis tools.
 - Basic commands allows controlled-velocity movement.
- Available audio interfaces are analog, ADAT and MADI.



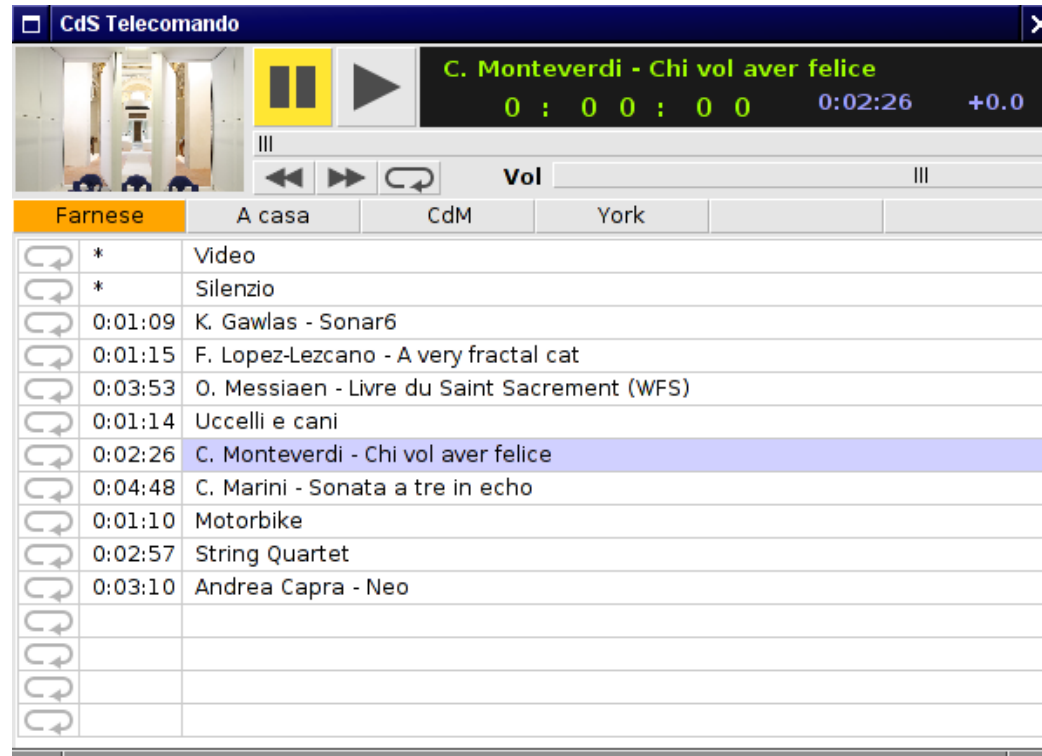


- Start, stop and monitoring of the master and render applications.
- Monitoring includes:
 - Program, network and timecode status.
 - Timing of rendering commands.
 - Primary source positions.
 - Speaker levels.
- Also provides test facilities, solo,...





- Original version (shown) provided level control, panning and simple movements.
- New version has also in-line panning, EQ, Aux sends and stereo monitor.
- 'Total recall' and OSC remote controlled.



- Fully automatic operation under control of a Python supervisor program.
- Audio playback via PyJackPlayer, a multichannel player app implemented as a Python class.
- Most other components are Python classes as well.
- The supervisor program also acts as a server to one or more remote control clients running on EEE-PCs.



Many thanks to all who made it possible to realise this project:

- Prof. A. Farina, University of Parma.
- The president and direction of La Casa della Musica.
- My former colleagues at Audio Link.
- The authors of the Linux sound system, in particular Jack.
- All members of the LAD list who have provided help and hints.