FIELD EXPERIENCE ON MEASUREMENTS WITH THE ACOUSTIC CAMERA

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1. INTRODUCTION

This paper describes some field experience on the use and measurement results and procedures of the Acoustic Camera manufactured by GFaI in Germany. By the use of Acoustic Camera in measurements it is possible to differentiate and localize different sources. Main measurements have been done on a Wind Power plant close to a road but also some other measurements will be discussed. Finally there will be given some real-time hands-on tips as well as a discussion about sound power level calculations.

2. ACOUSTIC CAMERA

The Acoustic Camera from GFaI is based on beamforming of a conventional delay-and-sum beamforming (DAS) in the time domain.

$$P(t, \vec{r}) = \frac{1}{N} \sum_{i=1}^{N} p(t - \Delta_i(\vec{r})) \tag{1}$$

DAS beamforming can be performed in either the time or the frequency domain, whereby time domain DAS is done by separately delaying each microphone signal, making them align before summation and normalization. DAS in the frequency domain is different by that each microphone signal is first transferred into the frequency domain by FFT and then shifted, after that the signals are summarized and normalized. The GFaI Acoustic Camera currently uses the time domain DAS mainly because of the faster processing speed and new signal processing algorithms.

Figure 1. Typical setup up of an Acoustic Camera system: Microphone Arrays (Ring, Star and Sphere) (Left),
Data Acquisition unit (Middle), Software (right)

2.1. Specification

The performance of the complete system is typically:

- Sampling speed: up to 192kS/s simultaneously for all channels
- Dynamic range: approx. 35 – 130 dBA (24 bit AD)
- Frequency range: approx. 100Hz to 20kHz (depending on microphone array)
- Measurement distance: approx. 40cm to 600m (depending on microphone array)
- Calibration: TEDS and clicker
- Filtering: Lp, Hp, Bp and Bs-filters with up to 120dB/octave slope

The functionality of the complete system covers:

- Recording to disk: including playback, wav-file export
- FFT and/or 1/3-octave spectra
- Spectrogram with filtering and playback of filtered signal
- Ordergram
- Acoustic Photo and playback
- Spectral Photo and playback
- Acoustic Movie and playback
- 3D Acoustic Photo

2.2. Outdoor considerations

Outdoor measurements have been done in 10 different positions, as well as complementary sound level meter measurements. Typically 32s of data have been sampled at 48 kHz and stored onto a hard-disc, and then the signal has been post analyzed. In the field this system have been run on battery using a car battery and a 12 to 220VAC converter, it is here important to use a professional converter with true sine conversion, instead of cheaper “pulse converters”. For outdoor use we have selected the “Star-array” due to its low frequency specification: from 100Hz to 7kHz (diameter 3.4m = wavelength of 100Hz), and its integrated windscreens. By this we can optimize the setup time which was typically found to be approx. 10-15min first time and <5 min for each successive time. It is also easier to move around from each measurement position.

During the measurement on the wind power plant, it was a rather flat field making it possible to have the equipment in the trunk of the car, while driving from position to position. First it is good to check the “auto-enable microphone” box so that one makes sure the system recognizes the array and its sensitivity by the means of TEDS. Secondly it is good to check the overall performance by use of the clicker and to check the microphone performance and to set the range by studying the level plot.

The wind speed was noted to be around 8m/s with higher gust winds likely 10-12 m/s. This could cause spikes in the microphones, so it is necessary to look at the time function after each recording. There is an aid in the playback function so that it is easy to directly listen to the recording. By checking the level plot one will also get information about the measurement range and eventual “overload” – which is also available on-line.
The software calculates the Acoustic Photos or Acoustic Movies referenced to a plane, so it is necessary to measure the distance to the objects. A separate distance laser meter helps in this case. Even so it is possible to change the distance after the measurement, this also makes it possible to move around quickly. Some other preparations to consider is to prepare measurement directories, disk space, back-up disk and that your batteries are fully charged. A spectral photo will easily be 1 GB in size.

3. MEASUREMENTS

3.1. Noise Levels

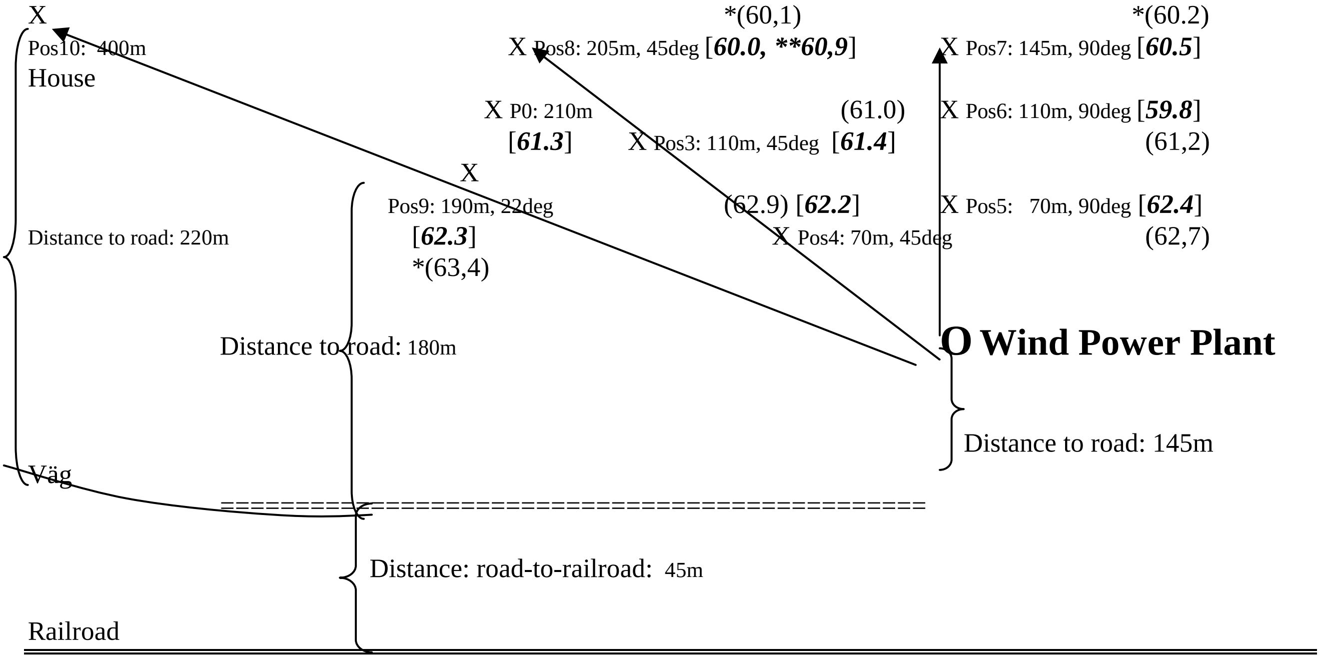

Figure 3.1. Overview of location of Wind Power Plant and levels in dBA, Tvååker
[Square Brackets] = SLM values
(Parenthesis)…....= Acoustic Camera measurements, corrected, see Ref.[1]
Acoustic Camera values as averages out of parts of 32s time function.
* Measurement values calculated as average of two 32s measurements.
** Including approximately 30s long freight train.

3.2. Acoustic Photos

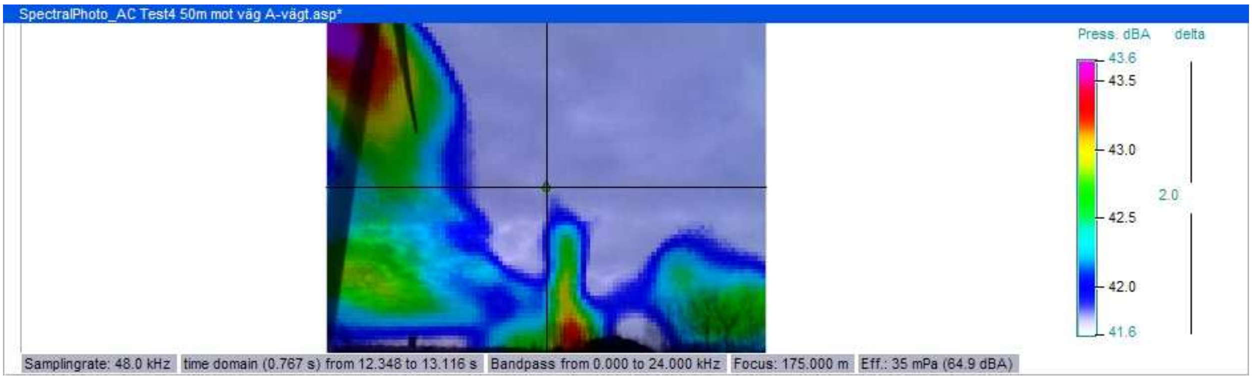

Figure 3.2. Sample of Acoustic Photo on both Wind Power Plant and Road, filtered 71-345Hz

An acoustic photo is an overlay of the acoustic sound pressure “on top” of the image from the camera. The temperature scale on the right side is called the “acoustic contrast”, and typically the “noise floor” of this acoustic contrast in an acoustic camera systems is somewhere between 10 – 15 dB.

The acoustic photo is generated from a part of the time signal which can be filtered in dBA or by butterworth filters: Lp, Hp, Bs or Bp. It is also possible to playback the marked time function for a selected microphone

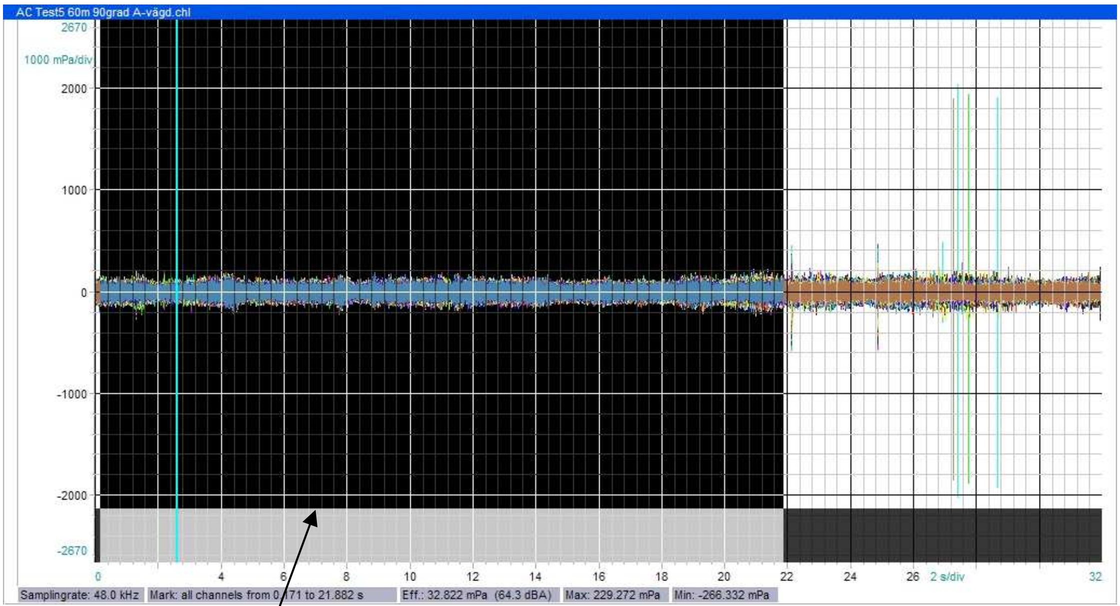

Figure 3.3. Marked segment of 32s time function for 48 channels, incl. wind noise at the end.

The spectrogram is used to generate acoustic photos by studying tonal components and to easily do filtering including playback of selected area, so that the display/generation of the acoustic photo is optimized.

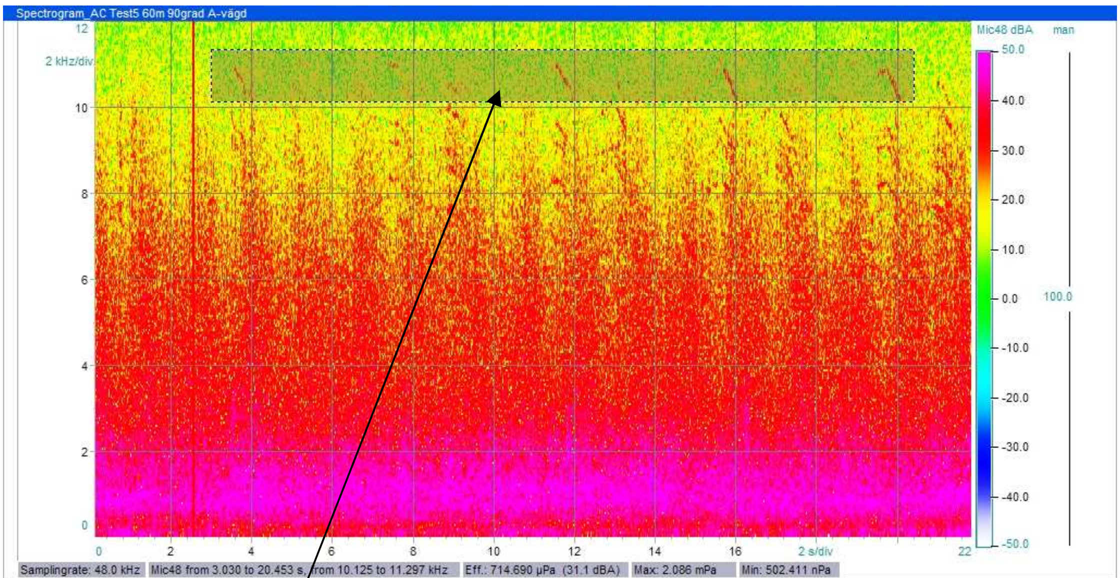

Figure 3.4: Marked segment from 10.125 to 11.267Hz, from 3.030 to 20.453s time function

The filtered signal from the spectrogram will generate an acoustic photo which will show the localization of the dominating source.

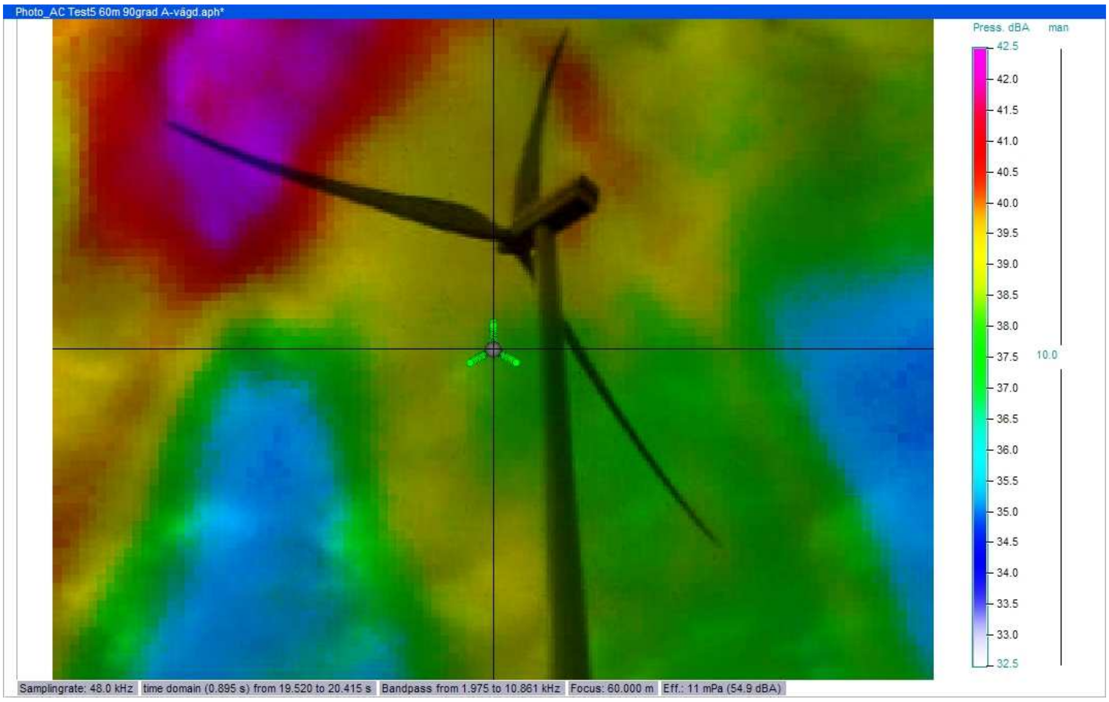

Figure 3.5: Acoustic Photo from marked segment 1975 to 10861Hz, from 19.520 to 20.415s time function

From the spectrogram it was clear that there was one wing which had significant extra contribution in the high frequencies (10-11kHz). By the use of acoustic movies it is now possible to determine which of the wing in the time signal that was giving this contribution. It is also possible to playback the filtered noise in the movies.

Figure 3.6: Acoustic Photo from marked segment 10.608 to 11.0381Hz, from 11.426 to 20.228s time function
The left side wing with a high frequency contribution.

In the Acoustic Photo each of the pixels have a corresponding spectra, it is therefore possible to display the spectra for every pixel in the photo. Vice versa another useful post-processing algorithm is the “Spectral Frames”, where it is possible to do the opposite: For every part in the frequency spectra it is possible to mark an area and then to display the corresponding location for this part. This is very useful to determine from where tonal components come from. A measurement on the wind power plant gearbox shows this:

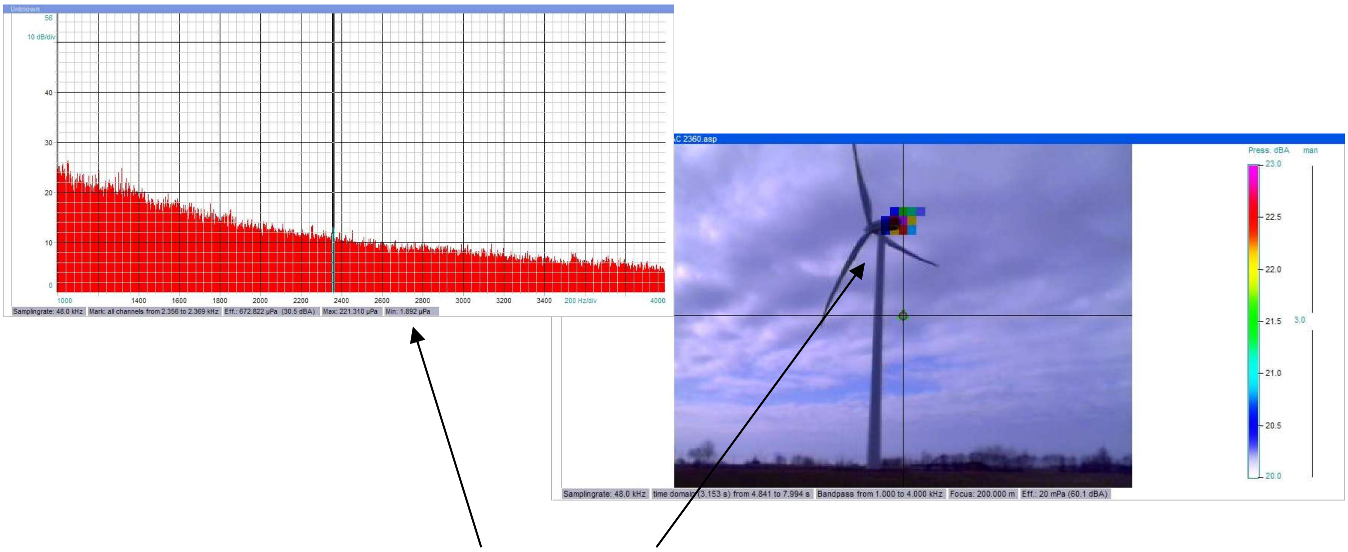

Figure 3.7: Spectral frames and location of tonal component 2359Hz

Next phase of development is to calculate the sound power level Lw, an ongoing study (graduation work at LTH, Sweden) will soon provide results on how accurate this is.

Figure 3.8: Sound Power estimation Lw=72.5 dBA from region of interest (ROI)
(note: Filtered tone at 2359Hz)

4. SOUND POWER CALCULATIONS

The acoustic camera software NoiseImage calculates the sound power from the SPL data and the measurement distance. Measurements were done in an anechoic chamber with a reference sound source (90.7dBA) and a disturbing source. The influence of microphone distance compensation was studied, as well as how the sources influenced each other. The original distance is the fixed measurement distance. Four cases:

1. The reference sound source is placed at a fixed position (p0) while the disturbing sound source is moved along the focus plane in four 0.65m steps to positions p1, p2, p3 & p4. Measurement distance = 5m.
2. Disturbing sound source is held at fixed position (p0) while reference sound source is moved along the focus plane in four 0,65m steps to positions p1, p2, p3 and p4. Measurements are made at 5m distance.
3. The reference and the disturbing sound source are placed 0.32 m apart in the focus plane. Measurements performed at 1, 2, 3 and 4m distance.
4. The reference- and disturbing sound source was placed close together far off to the left in the acoustic photo. The distance from the array to the reference and disturbing sound sources were 5.36m and 5.44m.

For tests with measurement setup 1 and 2 the measurement distance varies in regards to the measurement positions. The sound power calculations are presented in fig. 4.1 and 4.2, where in figure 4.1, the distance has been modified so that it more accurately resembles the true distance and fig. 4.2 is the original.
s1 = the reference source, s2 = disturbing source, s1,2@s2 = Both s1,s2 “ON” measured at s2.

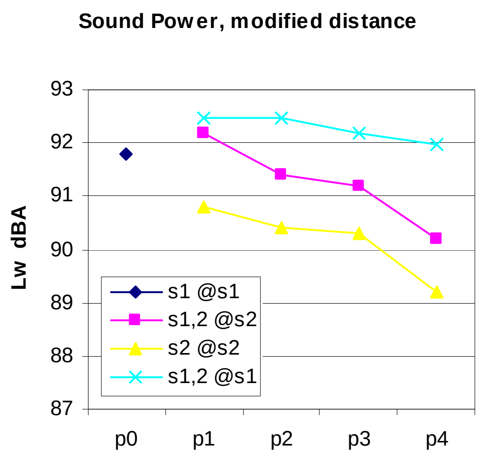

Figure 4.1

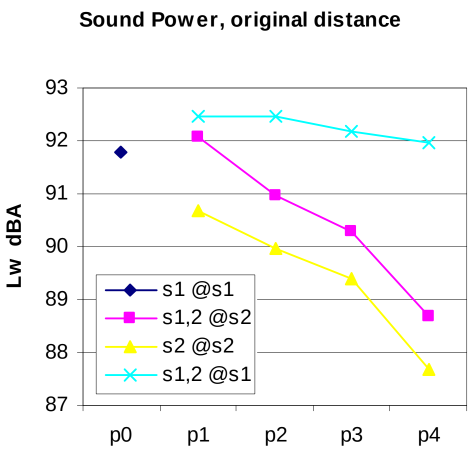

Figure 4.2

Measurements done with setup 2 are presented in Figures 4.3 and 4.4.

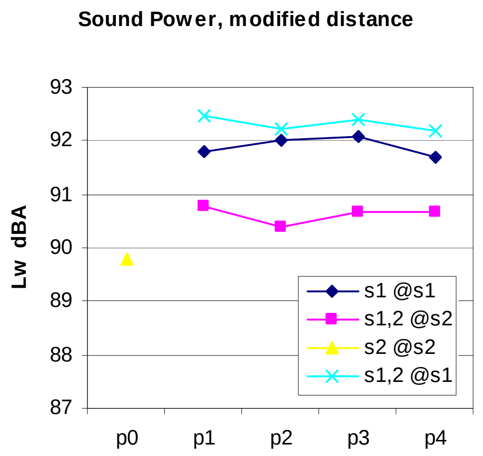

Figure 4.3

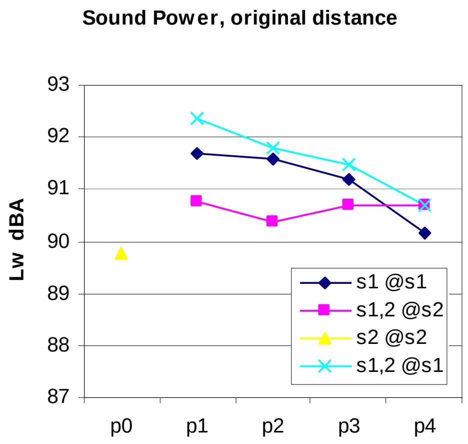

Figure 4.4

Sound power calculations for measurement set-up 3 (1 to 4m) are presented in figures 4.5. Figure 4.6 is for measurement set-up 4.

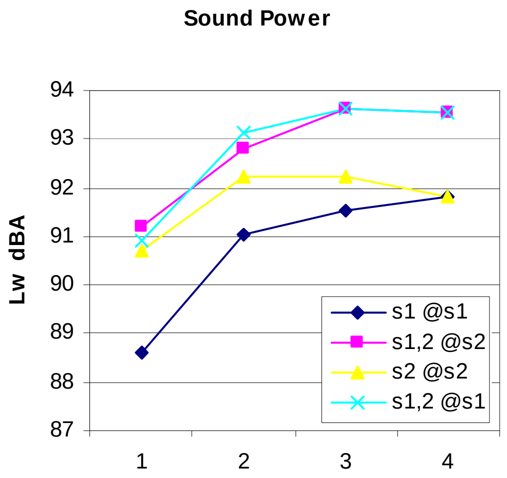

Figure 4.5

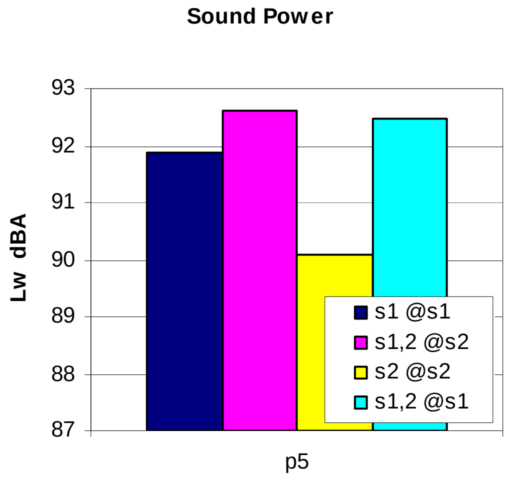

Figure 4.6

Within limits of spatial resolution, the SPL readings for the individual sound sources are given. It could be expected that the influence of a nearby sound source would decrease as the distance between them increase. This is overall true considering exceptions are explainable. The sound field produced by the disturbing sound source, the compressor, is unknown. As the microphone array is exposed to different angles of the compressor in the positions (p0, p1, p2, p3 and p4) it is possible that variations in the compressors radiation pattern cause deviation from the expected. This ought to be the reason for the higher values SPL recorded from the disturbing sound source in measurement set-up 3 as well.

To evaluate sound power calculation capabilities, values on SPL are proportional to the acoustic cameras L_w approximation. One dB SPL scales to one dB L_w, given the same focus field distance.

5. CONCLUSION

By the use of Acoustic Camera in field measurements it is possible to localize different sources, even with other dominating sources present. It is possible to cover a large number of measurements per day if one makes proper preparations. The measurements results from the Acoustic Camera shows good correlation with sound level meter measurements, after applying correction. By the use of the various new evaluation possibilities such as Acoustic Photo, Acoustic Movie and Spectral Frames it is quite possible to localize noise sources, also when these do not really dominate the overall levels. With new algorithms it is possible to get the sound power levels Lw directly from one single measurement even with disturbing sources close.

6. REFERENCES

[1] Döbler, D., *Time-Domain beamforming using zero-padding,* Berlin Beamforming Conference (BeBeC), 2008.

[2] Baumann, P., *Total effective sound pressure level of acoustic camera (Ring 32-4211) compared with four free-field ½" microphones*, Berlin Beamforming Conference (BeBeC), 2006.

[3] Kern, M., Opfer, H., *Enhancment of the dynamic range in acoustic photos by modified time domain beamformning,* Berlin Beamforming Conference (BeBeC), 2008.