The Sound Map

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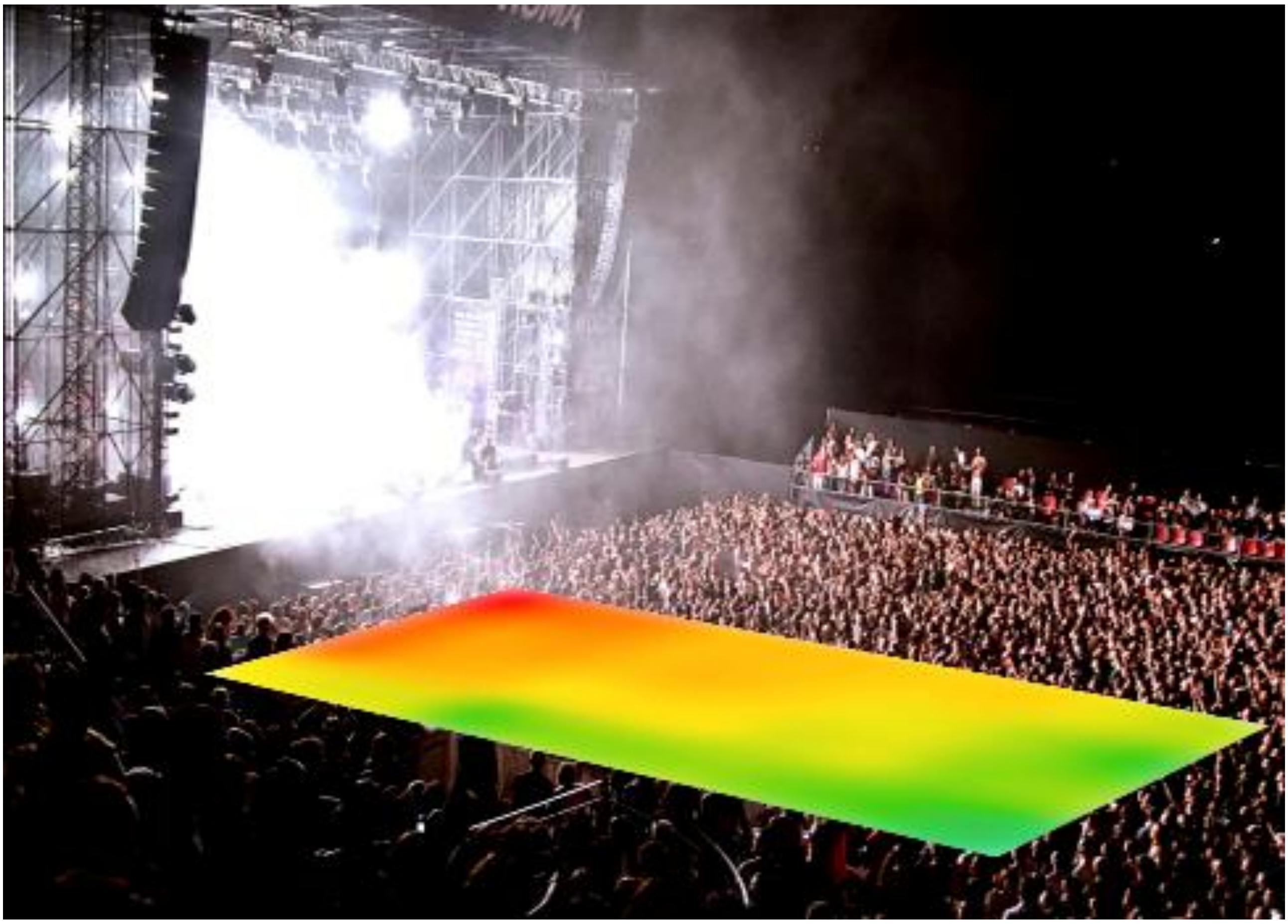
A sound field produced by a subwoofer array was measured for a direct comparison between the resulting sound map and the simulated one. Our conclusions are encouraging. The measurement technique (microphones and transmitters) is worth of particular mention.



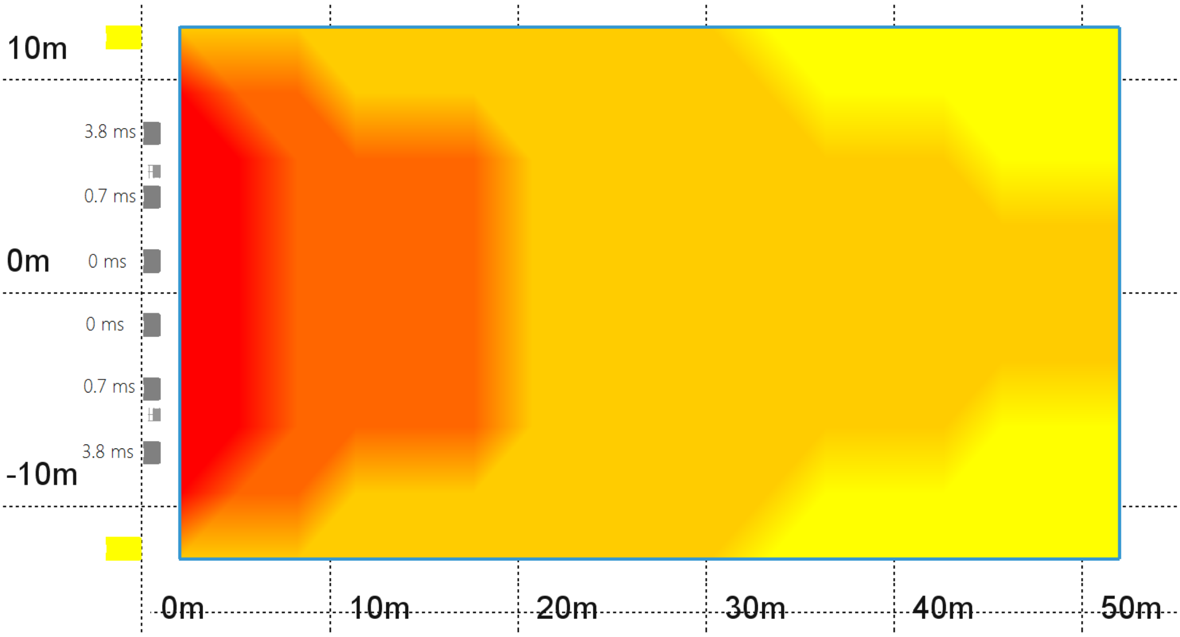
INTRODUCTION

Every summer in Rome there is a long rock festival. All of this huge festival´s concerts take place at the *Ippodromo delle Capannelle* and our Company is proud of being in charge for setting up all technical equipment (sound & lighting systems and logistics).
The rectangular audience area can hold more than 15.000 people and includes two terraces on both sides.
As far as the sound system is concerned, we usually set up two 16-cabinet clusters suspended at 12 meters from the ground, four front fill speakers and 18 cardioid subwoofer-array at the foot of the stage.
In the various editions of the festival we could - most of the time we had to - experiment different system configurations and settings (especially for subwoofers). However, show business time pressure did not allow us to investigate in depth the effects of the changes we carried out.
During the last edition of the festival, we finally had the conditions to make a real sound map of the audience area. Of course, we focused on subwoofers for two reasons: the complexity and variability of the sound field and the divergent opinions of different sound engineers on subwoofer positioning and setting.

The above-mentioned conditions are as follows:
Our know-how;
An affordable but excellent multi-channel measurement system;
An audience area with few sound reflections;
One day off (no concerts scheduled) and free access to the venue;
Time and willingness of all technicians involved in this project;
A sufficient number of measurement microphones (*).

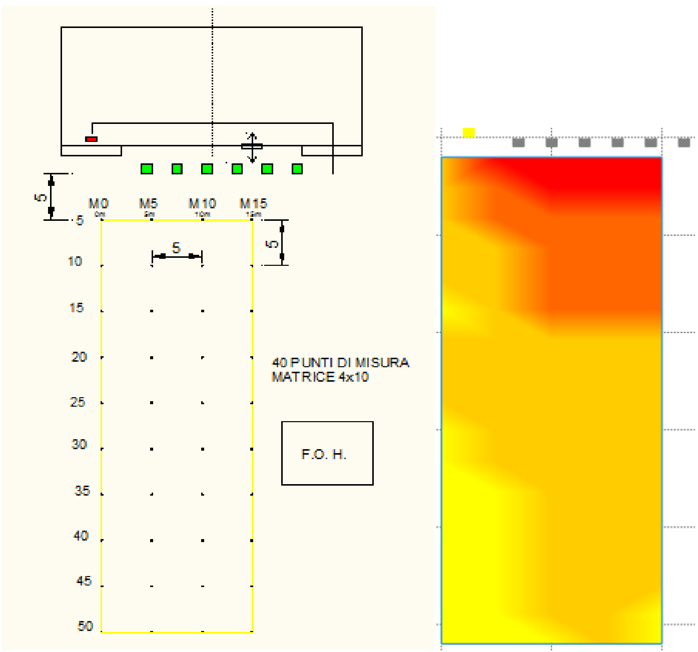
SIMULATION SOFTWARE

Our loudspeaker-dedicated simulator gives us a colour plot of the region of interest from which we can get the sound pressure magnitude and smoothness produced by a particular configuration.
Sound map resolution can be equal to 1,2 m or 5 m. For larger areas, such as the one we are discussing, it is possible to see clearly the trend of our project choices even at the lowest resolution.
For this measurement session, we used the configuration required by the artist's sound engineer the day before (!). We don't usually use this configuration; however, for many reasons, we did it. First of all for the limitations in time and weight of subwoofers.
The following picture shows the simulation graphic output relating to this configuration, with 5m of resolution at 50 Hz. Next to any 3-subwoofer stack the assigned time delay is indicated.



Simulation of the entire audience area at 50 Hz

As previously mentioned, the picture shows the entire audience area. However, due to the symmetry of the area, the number of microphones and measurements required and the fact that it was our first test, we decided to sample only part of the audience area.
If we extract only the area corresponding to the measurement one from the whole map, test validity remains unchanged.
The following pictures show the measurement area on the left and the simulated one on the right.
The dots on the grid represent microphone positions corresponding to the simulation ones on the right. We obtained this map by re-configuring the simulator audience area and keeping all other parameters unchanged, as if it were cut out from the whole map.



Measurement grid and corresponding simulated area

Each measurement point is represented by a square with 5 m per side, therefore the sampled area is 50 x 20 m. Furthermore, the simulated area exactly corresponds to the measured one.
With reference to the above-mentioned grid, we captured 4 simultaneous measurements for each horizontal line. We moved towards the bottom of the audience area 10 times, 5 meters per time, along the 50 m grid and we collected 40 curves. We obtained a sound map directly comparable to the simulated one at its lowest resolution (i.e. 5 m).

The stage and real grid

MEASUREMENT SYSTEM

For years, technicians have been using 2-channel measurement systems:
ch 1 = measurement microphone, ch 2 = reference signal (or vice versa). Today, thanks to new features and better performance of software and PCs, it is finally possible to easily perform simultaneous multi-channel measurements. We can use as many microphones as we want! It only depends on computer processing power. Wow! New horizons are calling us!
For this and other reasons, we began to study and build measurement microphones. In this particular case, we built a special microphone to be connected to a wireless system produced by well-known brand.
Factory technical specifications of transmitters were the following:

Audio Frequency Response

40 – 18,000 Hz, (+1 dB, –3 dB).

NOTE: Overall system frequency response depends on the microphone element

Gain Adjustment Range

UR1: –20 to +35 dB
UR2: –10 to +20 dB

Modulation

FM (45 kHz max. deviation), compander system with pre- and de-emphasis

RF Power Output

See table above.

Dynamic Range

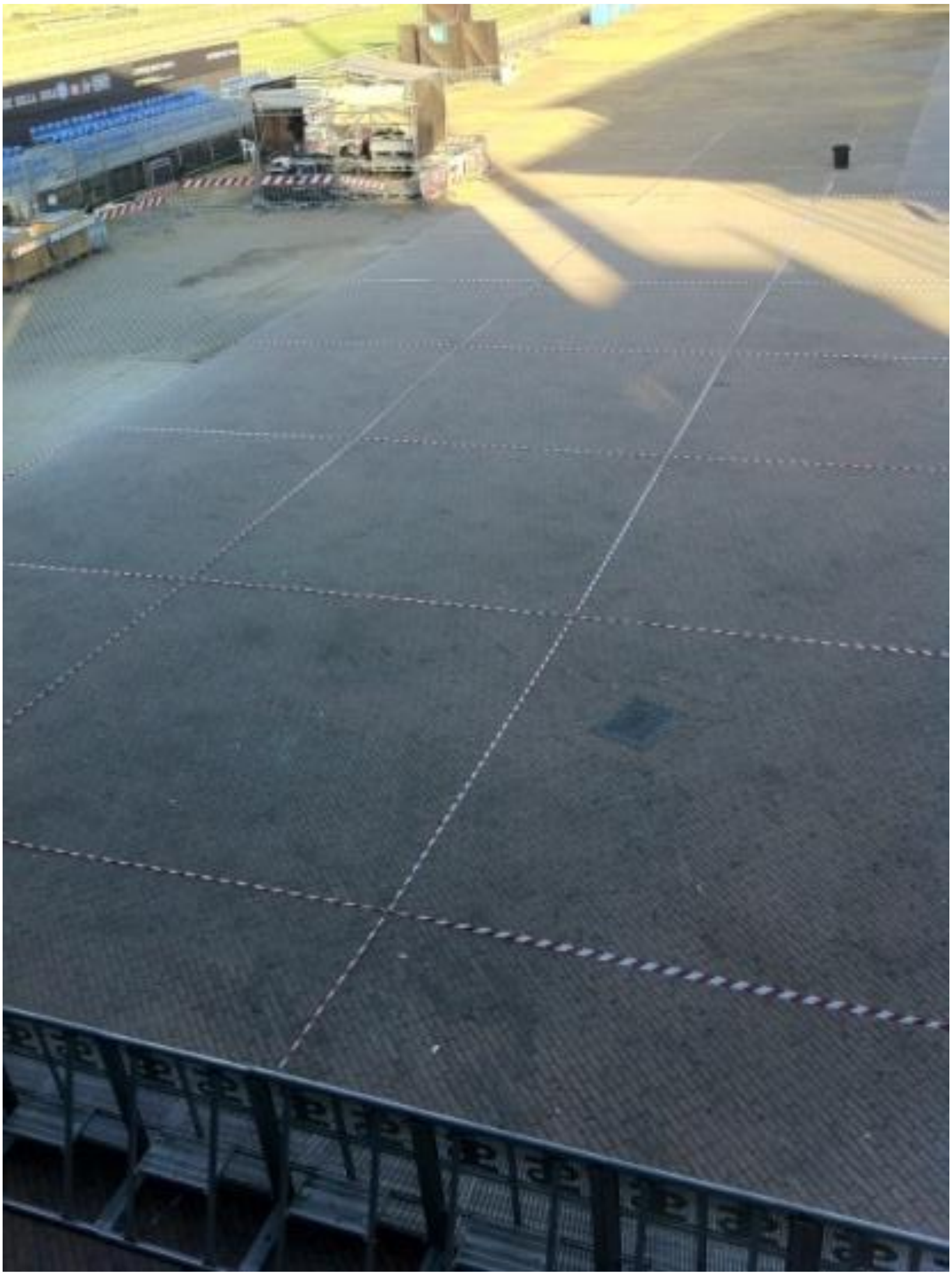
>105 dB, A-weighted

Wireless system main specifications

We regularly use this excellent system with satisfaction, and we compared it with a reference wired microphone. Results confirmed all above-mentioned specifications (see Appendix). The measurement dynamic range is very limited, approximately 20-25 dB, so, by accurately adjusting the preamp gain control, we can allow the transmitter to work in linear range and avoid compression phenomena, which are typical in this kind of wireless systems. The dc voltage at the input connector powers the JFET of the pre-polarized condenser capsule mounted inside our dedicated microphones. The very low frequency cut, below 40 Hz, is confirmed by the comparative test. Despite the 3dB attenuation at that frequency, data processing has also been extended to that third octave (with no corrections), since this test is primarily didactical.
All these considerations and having to capture only spectral curves, with no phase information, led us to use this system with the conviction that we would not have compromised measurement reliability.
Speed of operation, freedom of movement and other conveniences immediately win against the few limitations that the system imposes.

THE TEST

First, we set the measuring points using the red & white tape. In this way, we obtained the grid that is visible in the following picture (what happened to high-brightness lasers?). Then we carefully calibrated the 4 microphone levels. At this point, we started capturing the 40 curves.



The grid seen from left array

Calibration of three wireless microphones (the pink one is "female"!) and one reference wired microphone with the sound level meter

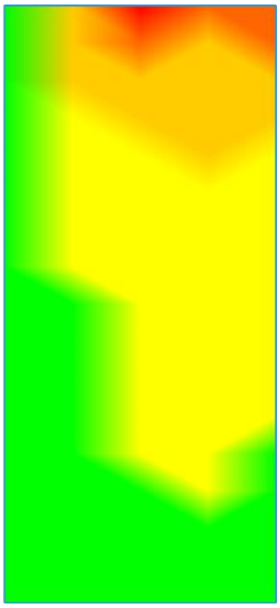
Loudspeakers were excited by pink noise and the curves were captured with an average time of 2 s after a few moments of settling time and microphones on the ground [1].
Average time of 2 s is undoubtedly too short; certainly, the use of a longer time would have provided more accurate data. However, the concern of not being able to complete the session in the available time frame and the relative stability of the curves led us to proceed this way.
Anyway, we recorded on a multitrack all tests for post processing.
Basically, this is our first experience thus an opportunity for improving techniques and procedures. Furthermore, this is a qualitative study rather than quantitative one therefore small-introduced error can be considered relatively significant.
Two of us were sitting at computers, storing the curves on two systems, while other "volunteers" were moving the microphones every time a "shout" was heard when data were successfully stored.
Ultimately, measurement operations have proven to be quick and easy. Thus, once the 50 m measurement was complete, even if it was getting dark, we decided to capture another set of curves.

Going backwards, towards the stage this time, and 10 m per time, we recorded the response of the whole system (excluding front fill speakers) according to the following sequence:
Sub + Left; Sub + Left + Right; Sub + Right; _Right; Right + Left; Left
120 more curves! We hope we shall have the opportunity to examine these data in the future, but now it's time for dinner!

DATA ANALYSYS

Collected curves have been exported in a worksheet; data have been selected and ordered by frequency and measuring point, creating a 4x10 matrix per each frequency.
We assigned a colour code to the cell values and gave them the right dimension in order to obtain a map directly comparable to the simulated one. The following picture shows the first elaboration of raw data for the third octave centred at 40 Hz.

-40.21	-31.58	-31.39	-28.1
-40.93	-35.56	-35.28	-31.99
-41.05	-36.68	-34.85	-33.41
-43.94	-39.71	-37.12	-36.98
-42.84	-39.33	-36.68	-37.16
-42.85	-39.8	-38.37	-37.5
-42.53	-41.3	-40.39	-38.72
-43.41	-40.77	-40.8	-38.48
-44.02	-42.99	-42.37	-40.67
-45.2	-42.68	-44.23	-41.87

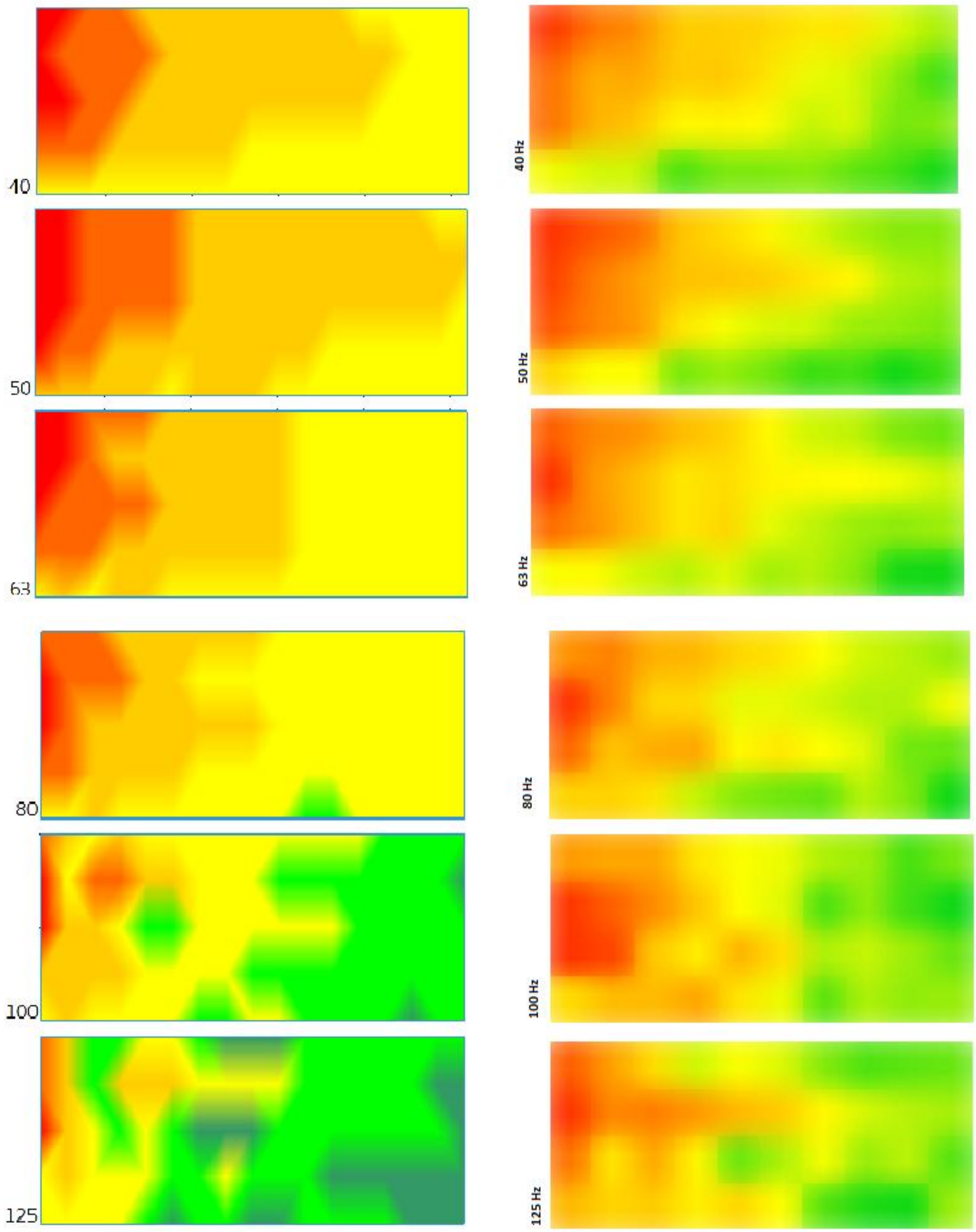


[Numbers shown in the measurement matrix indicate the software/soundcard full-scale dBV level. They will then be normalized according to the calibration carried out on site.]

This first comparison (although with raw data) shows a remarkable correspondence between the two "colour outputs".

Unfortunately, our limited knowledge of spreadsheets did not allow us to use colour scales as desired; therefore, we selected the closest matching colour to the one used by the simulator, shaded a bit the measured maps (quite arbitrary!) and obtained the following table:

SIMULATION MEASUREMENT



The correspondence is remarkable. At higher frequencies there is a greater discrepancy between the simulation and the measurement. In this range of frequencies, the radiation pattern loses uniformity and the effects of the reflections become more pronounced.
The presence of the two terraces, not considered by the software, alters our measurements. In any case, it is still possible to point many correspondences out. However, colours may lead astray. Therefore, it is better to focus on the numerical data.

simulator

scale

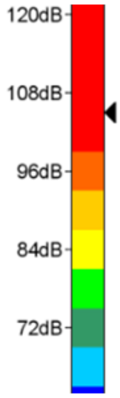


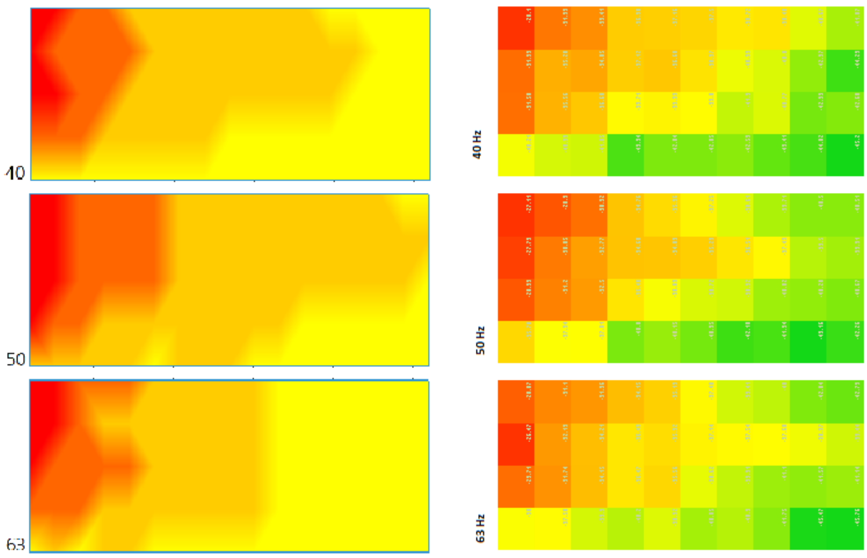
Table 1 - Maximum and minimum SPL of measurement and simulation (-28dB). Dynamic range and deviation [dB SPL].
* Simulated SPLmin estimation presents an approximation of +/- 3dB, being derived from colour code.

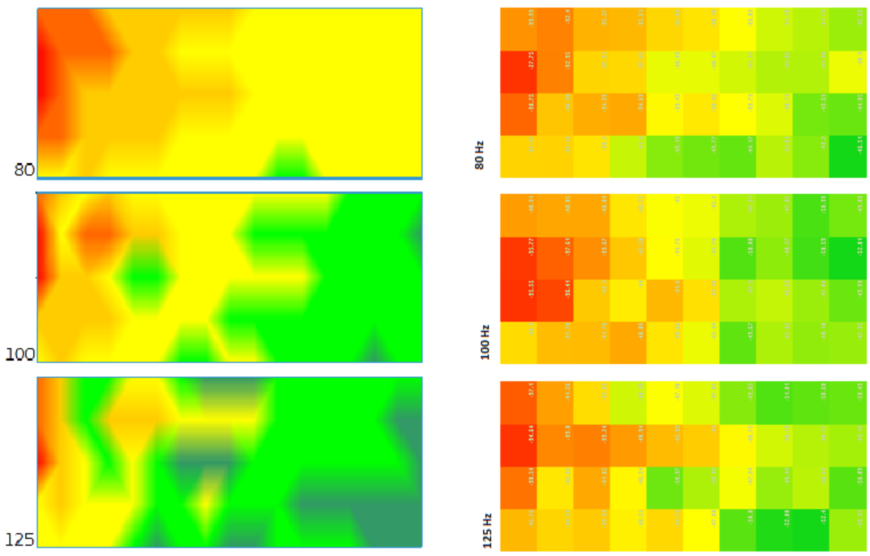
FRQ	MEASUREMENT			SIMULATION			DEVIATION		
Hz	**spl MAX**	**spl MIN**	**DIFF.**	**spl MAX**	**spl MIN***	**DIFF.**	**spl MAX**	**spl MIN**	**dynamic**
40	104.9	87.8	*17.1*	105.1	84	*21.1*	-0.2	3.8	-4
50	105.9	89.8	*16.1*	106.8	84	*22.8*	-0.9	5.8	-6.7
63	106.5	87.2	*19.3*	103.6	84	*19.6*	2.9	3.2	-0.3
80	105.2	86.5	*18.7*	104.6	78	*26.6*	0.6	8.5	-7.9
100	97.4	81	*16.4*	104.4	78	*26.4*	-7	3	-10
125	98.4	80.6	*17.8*	99.7	72	*27.7*	-1.3	8.6	-9.9
mean	103.1	85.5	17.6	104.0	80.0	24.0	-1.0	5.5	-6.5

Table 1 contains the data relating to maximum and minimum sound pressure of both simulation and measurement for each third octave of interest. Simulation software provides only a discrete number, which corresponds to the maximum sound pressure level. By accepting a certain approximation (+/- 3dB) we extrapolated the value of minimum simulated sound pressure based on colours only.
Numbers described above tell us that dynamic range (intended as the sound field max variance [2] or difference between SPLmax and SPLmin) is equal to 17.6 dB for measurement and 24 dB for simulation (arithmetic means). Looking more carefully, a very good correlation between simulation and measurement is observed, except for the 100 Hz band.
The cause of the significant discrepancy in the measured max SPL — 97.4 dB —, may be attributed to an error in the measurement and/or to the destructive effect of reflections not considered by the simulator.
On the contrary, more pronounced discrepancies are obtained by comparing measured and simulated min SPL levels. Simulated levels prove to be, on average, 5-6 dB more "pessimistic" with a maximum of 8 dB at 80 and 125 Hz. Again, major discrepancies are observed at the highest frequencies, main victims of uneven sounds fields because of their wavelengths that mainly "feel" the surrounding architecture. Nevertheless, main uncertainty is related to the estimation of minimum simulated sound pressure level. As previously observed, in the simulator colour scale from which SPLmin value has obtained, each colour represents as much as a 6 dB whereby the value suffers from a high approximation.
As a last consideration, it can be useful to remember that the simulation software lives in an ideal world, free from obstacles: so much fun! It can calculate interferences up to 160Hz,

but it ignores all the complex phenomena a sound field is subjected to in the real world. It also sees the subwoofers as brand new, while our speakers have already stimulated the senses of millions of people. We had no time to verify the performance of each loudspeaker for 18 subwoofers, so we cannot exclude that some “uncertainties” already occurred in sound field generation. All loudspeaker parameters are subject to changes over time and stress, especially moving parts, diaphragm and suspensions, which are exposed to adverse weather conditions and “trekking trips”! I can assure you that these drivers take long walks everyday! Unfortunately, there are no conditions to frequently and thoroughly check each loudspeaker of the entire system. However, the absence of irregularities observed in daily use is a guarantee of proper operation.

Before concluding this fun and useful comparative study, we believe that it is necessary to show the maps without shading off colours.





CONCLUSIONS

The experience was exciting and enormously informative, and the results didn't disappoint our expectations. This simulator, as most of its "colleagues", proved to be very reliable where reflections are limited such as large outdoor areas in which we often install big systems. However, the infinite complexity of the real world should always be considered. Instrumental verification is, therefore, always necessary and crucial to understand the cause of disturbance phenomena and try to "fix" them.
At last, today's multi-channel measurement platforms allow us to simultaneously display the response detected by several microphones. By displacing 5, 6 or *n* microphones, where circumstances permit, we can immediately get an idea of the produced sound field. The effects of possible corrections can be observed from multiple positions and better controlled.
Placing 5 microphones on the main radiation axis of a big cluster and doubling the distance as we move (i.e. 5, 10, 20, 40, 80 m), we have a detailed view of the whole array performance over the whole audience area. Furthermore, we immediately ensure that the sound does not lower by 6 dB as we double the distance. The opportunity to observe more "live" curves simultaneously is as useful as fast. The observation of the transfer function between a radio microphone and a reference microphone is equally and immediately useful. Instead of the mixer output, conventionally used as a reference, we will use the signal from a microphone placed in front of the main system in best conditions. After performing a control/calibration with microphones placed as much coincident as possible, and ensuring of obtaining a flat curve (if microphones are identical), we will ask a kind fellow to walk around the audience area holding the wireless microphone. Transfer function shows the true sound field *variance* that is the extent of the differences between the sound field that surrounds the mobile microphone and the one that beats our "electronic ear", the reference microphone, at best audience position. What we see is our true nightmare. This *fully acoustic* comparison deserves attention. We suggest to try it out.
In short, performing complex – and a bit more comprehensive than usual – measurements is very gratifying and funny. And getting rid of dirty and awkward cables is priceless. For everything else there's... a famous credit card!

ACKNOWLEDGEMENTS

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A very special thanks to Silvana Cinus for her help in translation.

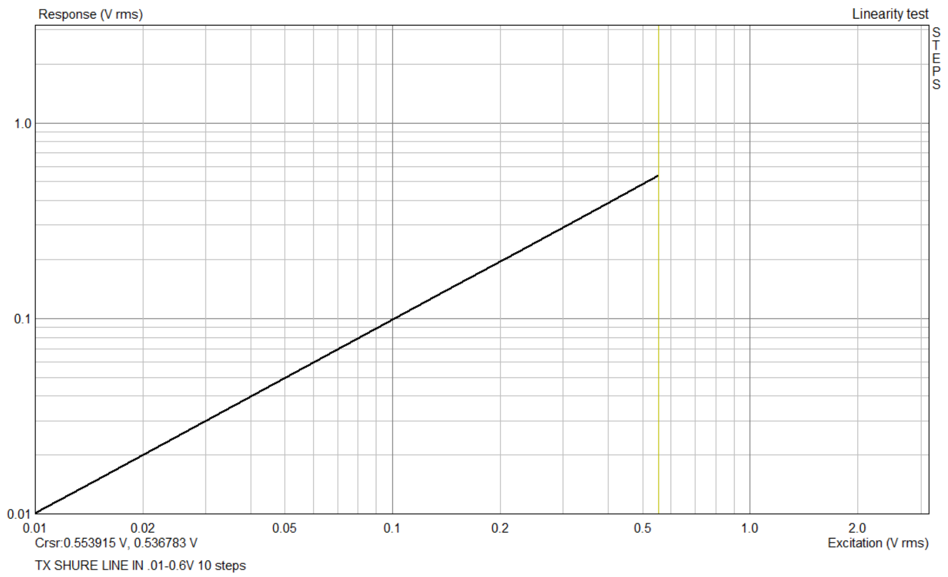
REFERENCES
[1] Gander – Ground Plane Technique – AES
[2] Bob McCarty – Sound System Design & Optimization
Other texts we consider essential:
The Handbook for Sound Engineers, Sound System Engineering, Audio System Design & Installation, Acoustics, Loudspeaker Sound Reproduction in Rooms and many more.

Appendix 1

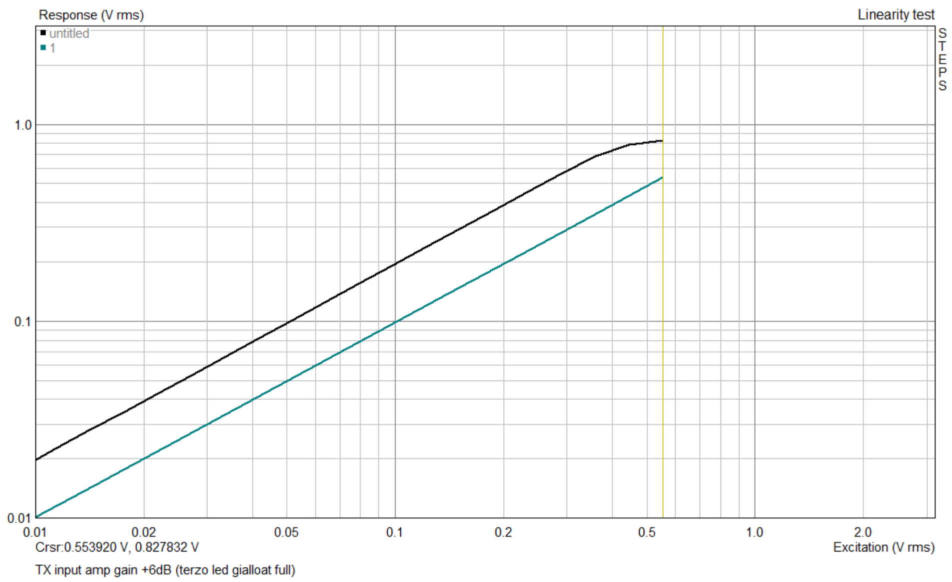
WIRELESS SYSTEM MEASUREMENTS

The following lines include some preliminary measurements performed on the wireless system. These measurements are necessary to determine the appropriateness of using this radio system in the context of interest.

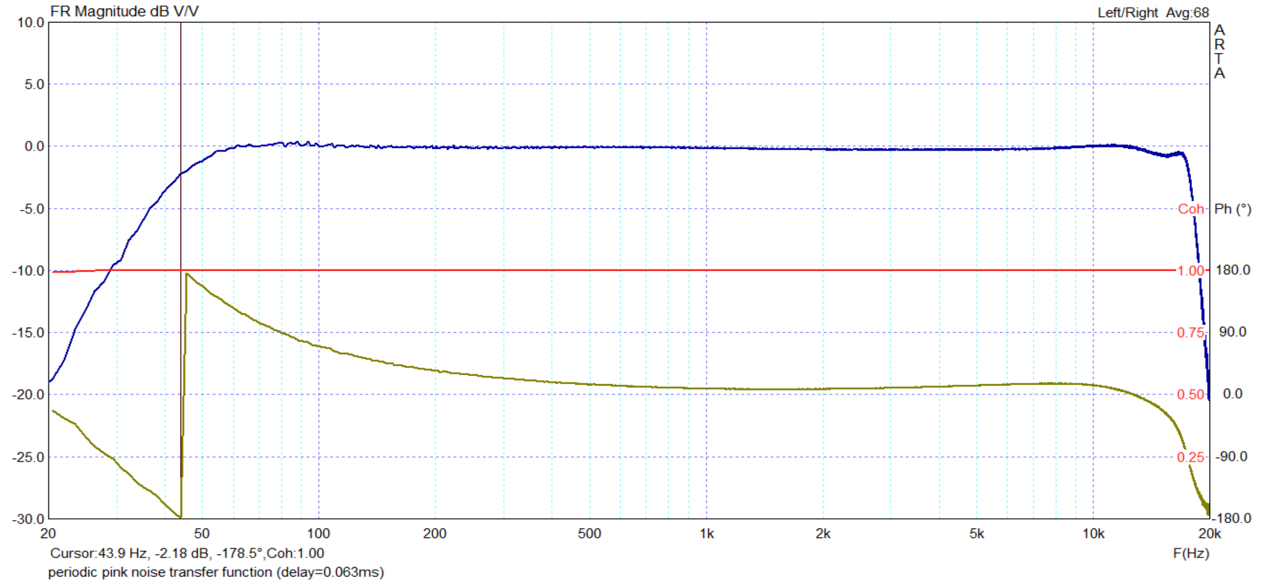
A gradually increasing signal (46 dB of excursion) has been connected to the transmitter with -10 dB of sensitivity and 0 dB of gain in order to turn on the VU-meter LEDs from the first green LED to the second yellow LED, starting from the bottom. The following graph shows a good linearity, excellent for the first 30-35 dB.



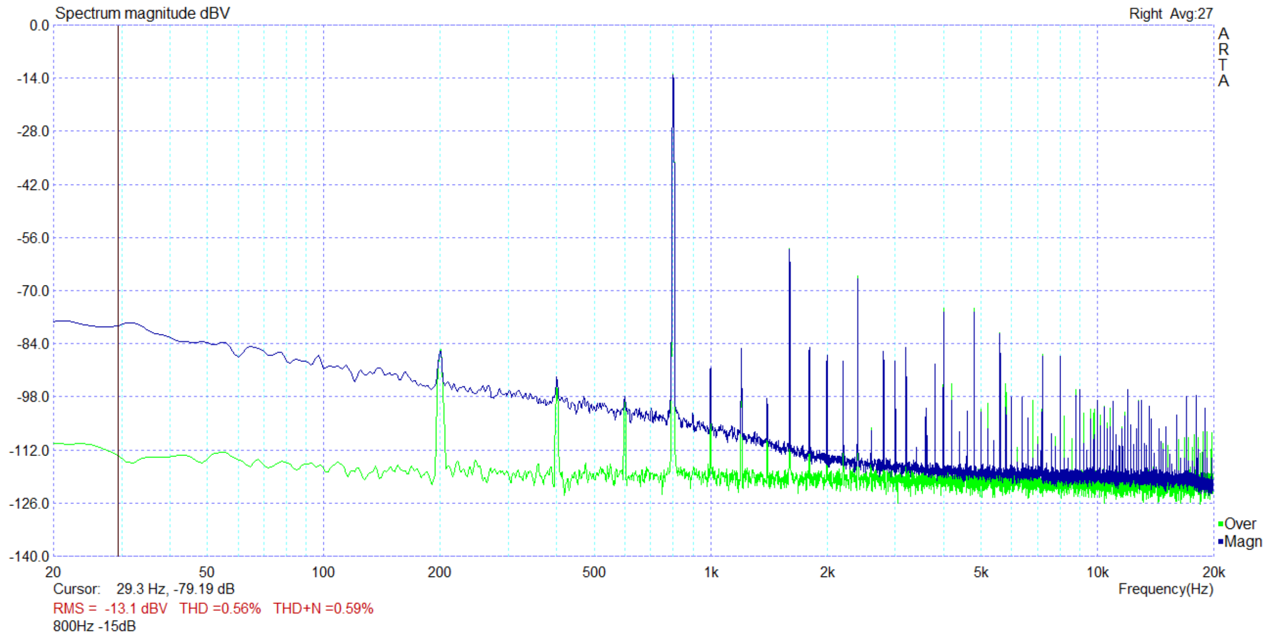
Increasing the tx gain of 6 dB (until the third yellow LED is fully on) obvious compression and/or saturation phenomena are observed.



Amplitude and phase response vs frequency (last green LED fully on; transmitted signal delay = 0.063 ms)



Hasty (tight times!) comparative measurement of distortion and noise (green = loopback; blue = TX, last green LED fully on)



The frequency response "recommends" using the system from 50 Hz to 15 kHz. Distortion and noise do not seem to be a problem and compression phenomena can be avoided by properly adjusting the transmitter gain. In short, the system can be profitably used within certain limits, which are not serious at all.

Evidently, the above-mentioned limits should not be able to invalidate the qualitative/educational purpose of this measurement session in any way.