Technique for the Derivation of Wide Band Room Impulse Response

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Abstract

Different methods are used today to measure the room impulse response (RIR). However, most of the equipment used is just adequate for the derivation of reverberation times. For the investigation of the RIR's time structure the directivity and the frequency response of the source become more important than for decay curves. Sound sources of well defined directivity and wide band frequency response were developed to allow the measurement of room impulse response that may be used for auralisation.

Introduction

The room impulse response typically is understood as the acoustical answer of a room, when excited with a very short impulse like is generated by handclapping or a pistol shot. Since the room is of three dimensions the impulse response itself depends on the spatial energy distribution of the source and the arrangements of reflecting surfaces in the room. This directivity moreover changes over frequency for both the source and the room. To measure an objective impulse response it is therefore quite important to know at least the directivity of the source precisely. This, however, must not be angle-independent (a sphere) but could be of desired shape.

Nowadays methods like pistol shots are still used in the field, because of the simplicity and the light weight of the equipment. On the other hand non of the above mentioned factors (frequency response, directivity) is met by this kind of source. Another drawback is that the excitation changes with each single shot. To overcome with this, loudspeakers in addition with specially designed excitation signals like MLS or chirps are very often used nowadays. However, loudspeakers are far away from being the solution to the above mentioned problems, only the repeatability of the excitation is rather perfect. With respect to directivity and frequency response the problems are still unchanged, but time-invariant.

To achieve both, directivity and frequency response in the desired way, a little more expense is needed. In this paper two loudspeaker systems (one with omni-directional radiation, the other with the directivity of a human singer) are presented. In addition with these systems a special signal processing is proposed, that enables rather perfect equalisation of the source in both domains, time and frequency. To achieve this the measured complex frequency response of the loudspeakers is inverted and used for the design of a sweep signal that compensates for both errors.

Basic Considerations

Two different types of sound source were designed to do measurement in rooms to derive impulse response signals that would be applicable for convolution purpose. This means that the frequency range should be wide enough as mentioned before, and that the directivity is omni-directional for the one source and comparable to that of a human singer for the other source.

A Three Band Loudspeaker with Point Source Characteristics

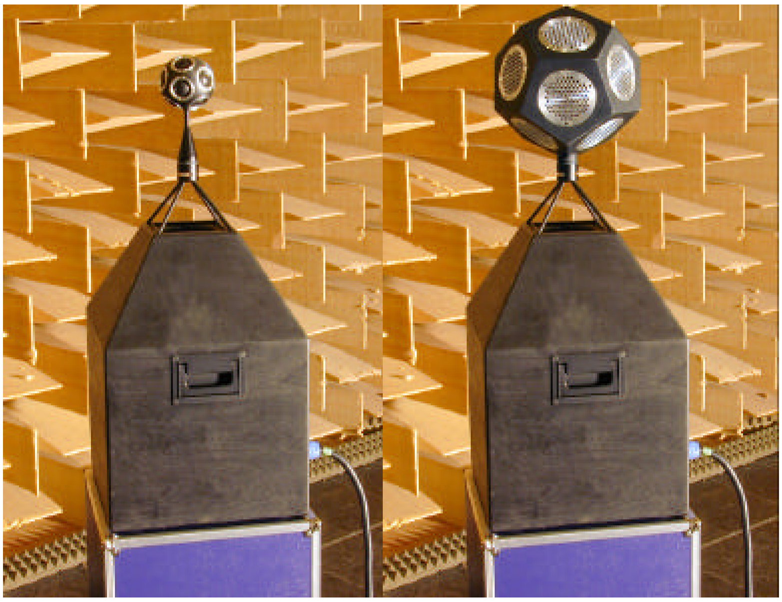
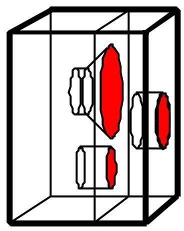


Fig. 1: Three-way measuring loudspeaker with omni-directional radiation characteristics. Note that the mid- and high-frequency loudspeakers are places at the same centre position above the radiating vent of the woofer cabinet. They have to be replaced during measurements, which is easily be handled due to the used Speakon connection

At low frequencies the aim of omni-directional radiation is easily achieved with a cabinet small compared to the wavelength. This can be used only in a restricted frequency band and therefore the common solution is to distribute loudspeakers on the surface of a spherical loudspeaker (dodecahedron). The solution, however, leads to a compromise, that may be applicable in building acoustics, where the detailed sound radiation characteristics is of minor importance. If the directivity has to be equal over a wide frequency band (i.e. 40 Hz to 12.5 kHz) this can't be achieved with only one cabinet. Consequently the full frequency range has to be divided into different bands, that are reproduced by individual loudspeakers. This introduces a new handicap, since the loudspeaker systems for the different frequency bands can not be placed at the same place, which is obviously needed to have a point source.



The proposed solution consists of a low frequency band pass cabinet (see sketch) with an internal driver of 12" and a radiating opening of only 10 cm diameter (re. Fig. 1). The design provides spherical radiation up to 300 Hz and low frequency capability due to the large driver surface. The tuning of the cabinet delivers flat frequency response at a nominal sensitivity of about 92 dB/W/m from 50 Hz to 200 H with –6 dB cut-off frequencies of 40 Hz and 300 Hz.

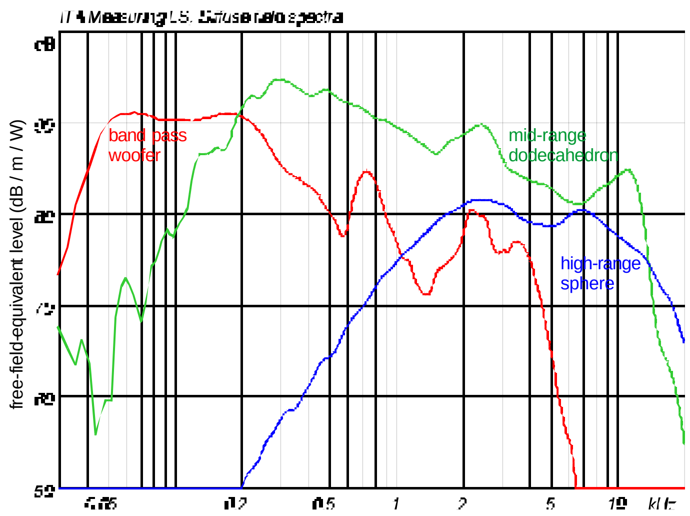


Fig. 2: Frequency response of the three loudspeaker systems. Mid- and high-range unit measured in the reverberation chamber and compensated for the absorption losses. Woofer measured in free field.

The frequency range from about 200 Hz is reproduced by a dodecahedron loudspeaker equipped with divers of 8 cm diameter, which allow an overall diameter of 27 cm for the cabinet. The short distance between neighboured systems is required to achieve the desired directivity even at high frequencies. The midrange unit is placed 25 cm above the radiating opening of the low frequency cabinet (as can be seen in the right picture in Fig. 1). This in addition with the low crossover frequency of only about 220 Hz maintains the desired coincidence of both frequency bands, since the distance is only 0.16 wavelength's at the crossover frequency. The acoustical centre of the speaker therefore is located in the middle of the dodecahedron.

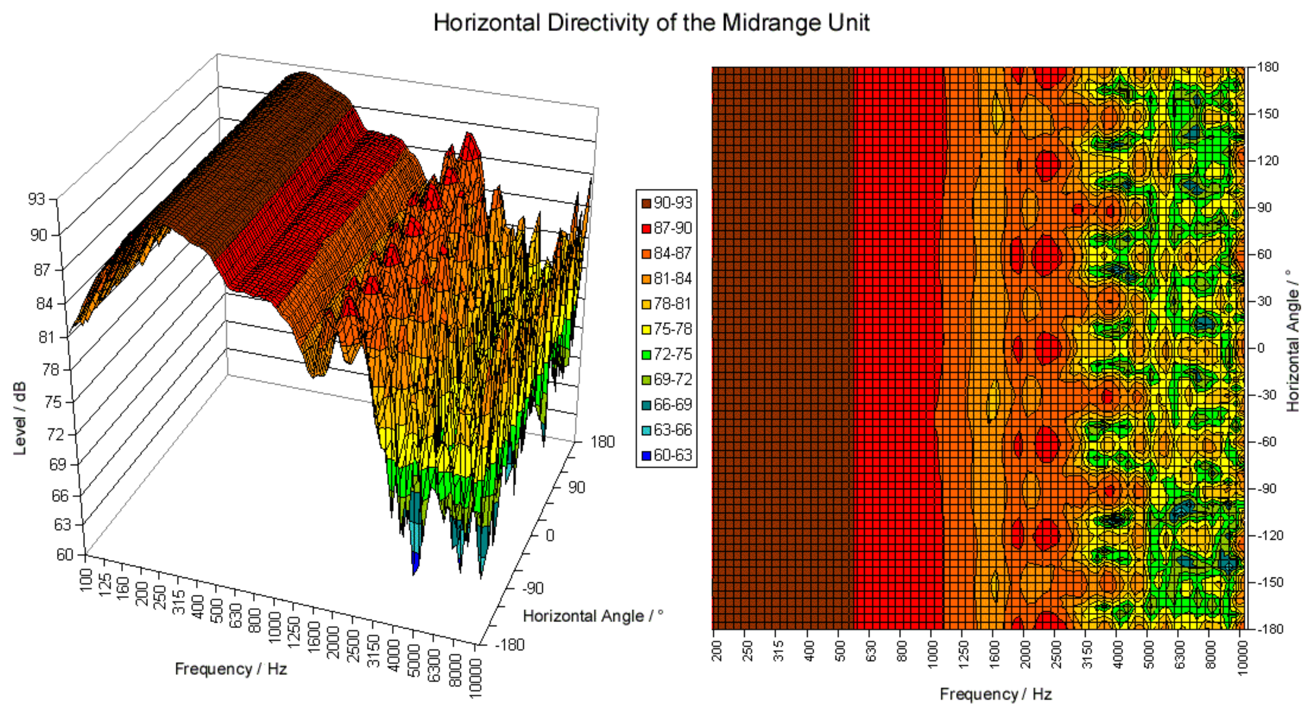


Fig. 4: Directivity plot of the midrange dodecahedron. Left picture shows a three-dimensional plot, right picture shows the map-view of the same properties

The useable frequency range with respect to efficiency ends at 6.3 kHz, which in addition with the woofer makes this loudspeaker useful for all building acoustics applications. The radiation pattern, however is not satisfying above 2.5 kHz. The directivity plots are shown in Fig. 4. The directivity is quite homogenous up to 2000 Hz, whereas above that frequency the alternating ripple shows the speaker arrangement of 6 drivers in the horizontal plane of the loudspeaker. The highest useable frequency (within an error limit of +/- 3 dB) is 2500 Hz.

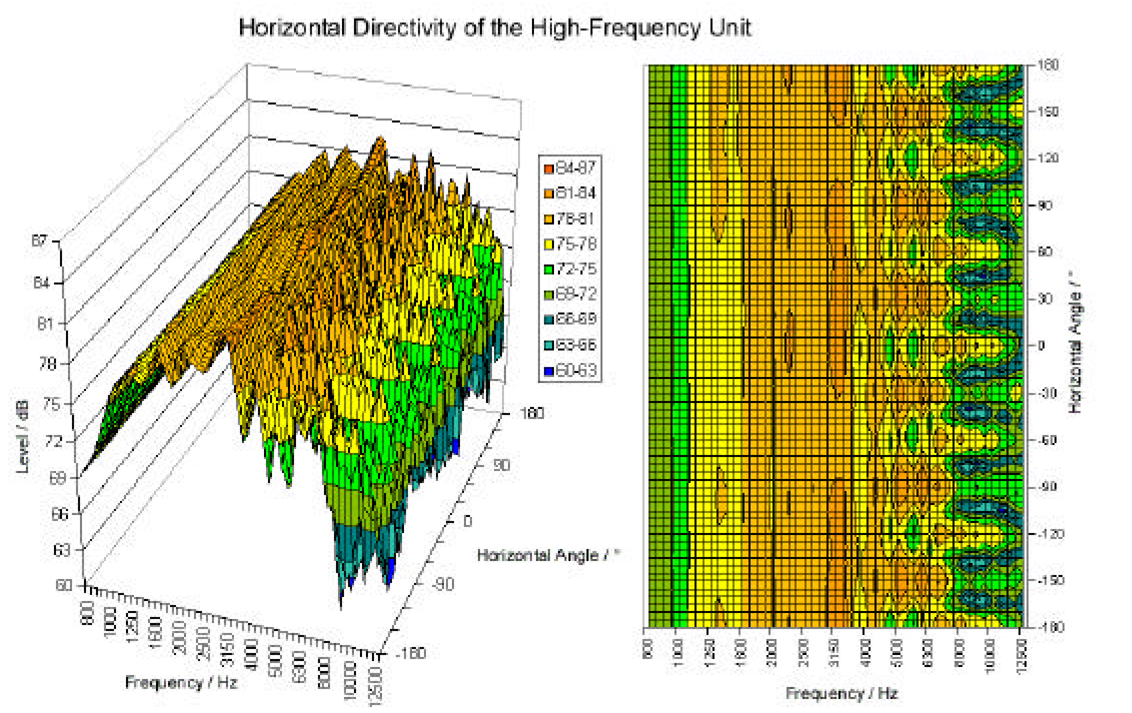


Fig. 3: Directivity plot of the high-range sphere. Left picture shows a three-dimensional plot, right picture shows the map-view of the same properties

The high frequency unit which is designed as a PVC-sphere of only 10 cm diameter equipped with 12 tweeters of 1” diameter (ref. to the left picture in Fig. 1) consequently takes over at about 2500 Hz. The small cabinet was possible to realise with centre magnet drivers using neodymium with an overall diameter of only 36 cm.

Since the high frequency loudspeaker should coincide with the mid-range unit, the loudspeakers must be replaced during measurements (as is shown in Fig. 1), which is an additional expense but the consequent tribute to better measuring results. For this purpose the loudspeakers are mounted on a commercial loudspeaker connector with 4 poles (Speakon), which allows individual connections for mid-range and high-range unit respectively. The high-range unit, however is not as small as it should be for best directivity pattern, which can be seen in Fig. 3. The differences between each individual tweeter are also larger than in the mid-unit, which is the reason for the wavy response up to 5 kHz. Above that frequency again the directivity is dominated by the radiation pattern of the single drivers.

Fig. 5: The construction of the artificial singer (called „Caruso“). The mouth is used for the entire frequency range from 150 to 10 kHz, whereas the low frequencies are reproduced by a specially designed band pass loudspeaker. The radiating opening of the reflex-port is placed just below the larynx in the neck.

The Artificial Singer (“Caruso”)

To have a loudspeaker with human-like directivity, a ‘singer’ was designed, which should be capable of reproducing all kinds of human voice either male or female. The frequency range therefore should start at about 80 Hz and go up to about 10 kHz with substantial acoustical output. This becomes rather impossible if only one small wide-range speaker-system is used inside the head, radiating through a hole in the mouth. However the solution is similar to that of the omni-directional loudspeaker: The range is divided into two bands, a low-frequency band with a band pass loudspeaker cabinet in the torso of the shoulders and the radiating opening in the neck, where usually the larynx is located (re. to Fig. 5). This in addition with a small loudspeaker in the head, behind the mouth, delivers both, good acoustical output power and also a rather flat frequency response without strong ripple as it would have been the case, if the whole frequency band would have been reproduced by one driver in the mouth.

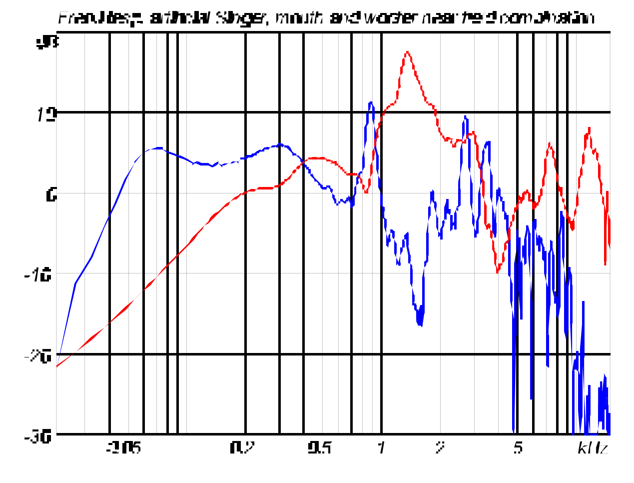


Fig. 6: Frequency response for the artificial Singer. Blue: subwoofer; red: mouth speaker:

Signal Acquisition Techniques

The frequency response curves shown in Fig. 6 are measured with MLS-technique and the flat blue curve was achieved by inverse filtering and adding of both bands at the same level. The equalisation process is required to achieve the desired ideal impulse response, which can be used as a convolution filter for auralisation purpose. This filtering process may be carried out after the measurements, but this requires independent measurements for each particular frequency band since the mix of the two (three) sources in the sound field can not be compensated afterwards.

Measuring with Adapted Signals

Another method to overcome with this problem is to compensate the excitation signal with respect to the source properties. This method may either be applied using an FIR filter in front of the amplifier or to generate signals that directly compensate for the frequency response of the source. The latter method leads directly to the desired impulse response and will be described here.

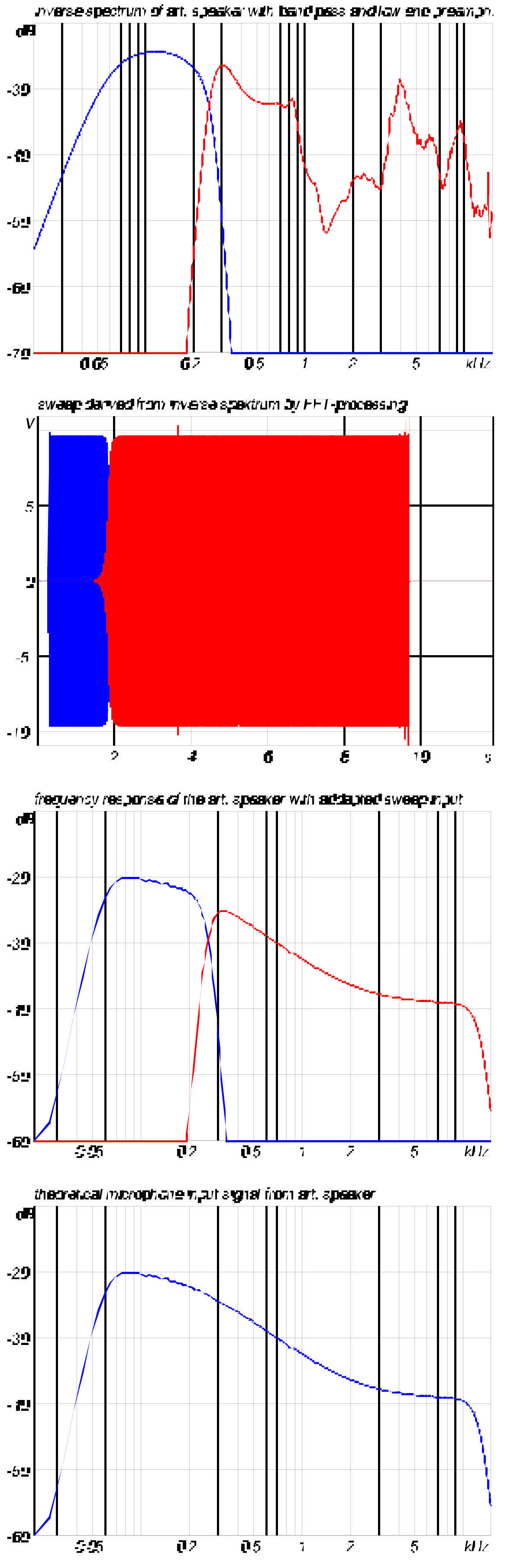


Fig. 7: The steps of the signal generation with adapted sweeps to compensate the source freq. response.

Excitation signals of almost any kind (Noise, MLS, Chirps, Sweeps et.) may be used to maintain this method but, however, it turned out that sweeps are the best choice for almost every measurement situation. They allow feeding the device under test with high power at little more than 3 dB Crest factor, and are pretty tolerant against time variance and distortion. The harmonic distortion products can be separated entirely from the impulse response. When it comes to capturing room impulse responses for auralisation purposes, there is no alternative to sweep measurements: The high dynamic range in turn of 90 dB required for this purpose is unattainable with MLS or noise measurements. The construction of sweeps is somewhat more sophisticated. It will be described here for the example of the artificial singer.

In room acoustics, the signal acquisition has to be at least as long as the RIR itself to avoid errors. This is obvious for the measurement with a single pulse. All its energy is emitted at the very beginning, and the AD converter simply has to collect samples until the RIR has decayed. In case of sweep excitation, the capture period has to be a little bit longer, but in general not much. This is thanks to the sweep's nice property to start with the low frequencies. In room acoustic measurements, the time of silence following the emission of the sweep just has to be as long as the reverberation time at the highest frequencies, which normally is quite short. When using periodic excitation, the total sweep time has to be as long as the reverberation time (with the time of silence added) to avoid time aliasing.

Construction of sweep excitation signals

With the relatively poor frequency response of the singer (especially the mouth output is deteriorated due to tube resonance) the generation of an adapted sweep will be shown (ref. to Fig. 7). First the inverted spectrum of the measured frequency response is calculated which additionally my be band pass filtered to avoid overriding the speaker at the lower and upper end. Moreover an emphasis of low frequencies may be desirable, since all measurements suffer from S/N in the low frequency area (Fig. 7 1st picture.).

After this is accomplished the energy time distribution of a sweep may be calculated in such a way that the amplitude of the sweep remains constant whereas the energy distribution follows directly the desired target response. The result of that generation scheme is shown in the 2nd picture of Fig. 7. Notice that the amplitude of the both sweeps is almost constant. The time dependant sweep rat, which contains all the energy distribution considerations unfortunately can not be seen here.

In the third picture of Fig. 7 the calculated result for the singer is shown. This is the acoustical output of the singer for the two bands as it would be received at the point of the measuring microphone when recording the plot of Fig. 6.

The summation of both signals, however, leads to picture 4 in Fig. 7, which would be the actual recorded signal,

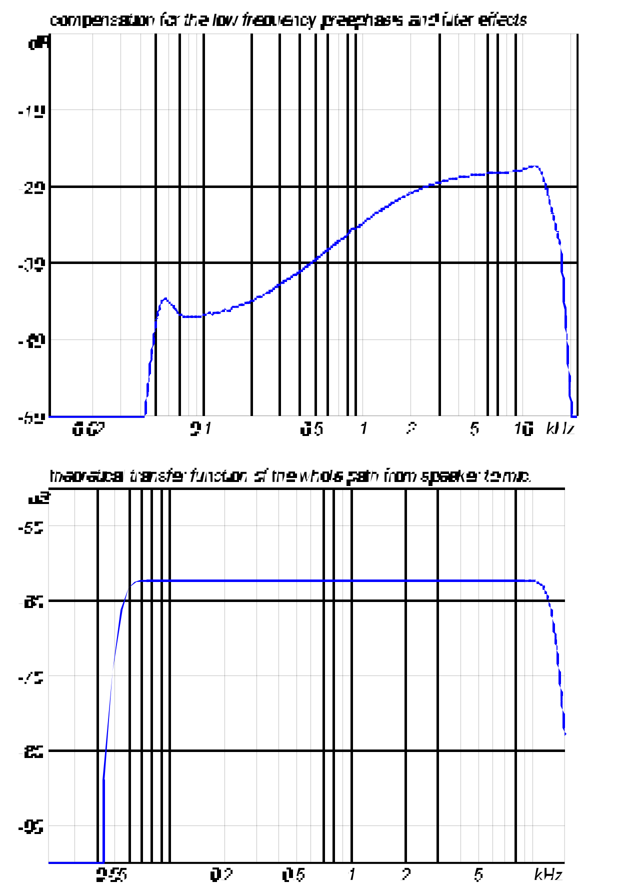


Fig. 8: Further steps for the compensation of the pre-emphasis of low frequencies and the overall frequency response.

when measuring with a two-channel excitation system as it is used in the institute. This power spectrum obviously contains still the preemphasised low frequencies, which has to be compensated to achieve the desired flat frequency response (and after all the impulse response of the measured room).

This compensation spectrum is calculated during the generation of the signal (see 1st picture in Fig. 8) and will now be multiplied with the spectrum of the measurement, which finally leads to the desired frequency response (2nd pict. in Fig. 8).

What is shown here is only a simulated (just calculated) example for the real singer. However, the output signal is flat after these operations and the measurement with the realised sweep signal will lead to rather ideal RIR's.

Removing Distortions and Artefacts

One major advantage of sweeps over other excitation signals is the immunity against distortions, which are always present when measuring with loudspeakers.

With sweep measurement each frequency is reproduced at a different time. The distortion products, however, are signals that are of different frequency (2 times, 3 times etc.) which would be an real excitation signal that comes much later (depending on the sweep rate). On the other hand the microphone records all components at the same time. So we have to separate the distortion products from the wanted signals.

For this purpose the deconvolution of the recorded sweep with the calculated inverse spectrum (containing all information about time and phase of our excitation) helps us, since only frequency components that are correlated to the 'inverse sweep' are back-folded into the desired impulse response. All higher frequencies are correlated to the corresponding frequency and are therefore back-folded to other parts in the impulse response. This depends on the duration of the sweep and hence on the sweep rate itself. As a result of this considerations all distortion products are apparent at the end of the impulse response tail and thereby can be cut away from the interesting part.

Conclusion

For the derivation of RIR's specialised loudspeaker systems with well defined directivity pattern were designed. The two systems are used in room-acoustical measurements to obtain RIR's that may be used at least for time domain analysis like IACC or echo-investigations, where the detailed time structure is of great importance. Moreover the precision of the sound-field generated allows the measurement of RIR's that may be useful for auralisation purposes. Since auralisation is highly sensitive to frequency response deviations and noise, a specialised method for the generation of signals with optimum S/N and distortion suppression was described. The energy distribution of these signals is adapted to the measured output of the loudspeakers using time variable sweep rate, which provides highest possible output of the loudspeakers with low crest factor.

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