An adaptive subtraction filter for feedback cancellation in public address sound systems

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Introduction

Apart from the well known procedures of employing narrow focusing loudspeaker or microphones, it is also possible to make use of various analog and digital filters for feedback suppression in public address sound systems [1][2]. This paper will report on a specific type of digital filter known as the subtraction filter. It must be stipulated that the best possible understanding of the "Loop transfer function" is required when implementing this type of filter, the measurement of which can be acquired by various means [2] [3]. Alternatively the filter function can be determined by incorporating an adaptive algorithm.
The "Loop transfer function" is determined by the various elements of the loop. Fig.1 shows a block diagram of a typical sound system consisting of microphone with pre-amplifier, filter $\underline{X}$ which is to be specified, a power amplifier, a loudspeaker and the room transfer function $ß_{LM}$.
Eq.1 shows the frequency dependant loop gain (see Fig.4) as a product of the single transfer functions. The predominant element for the distinct structure of the formula is the room transfer function. An exception to the rule can be found in the case of certain monitor systems where the distance between microphone and loudspeaker is only very short. In this case the direct sound field dominates over the diffuse sound field in the loop.

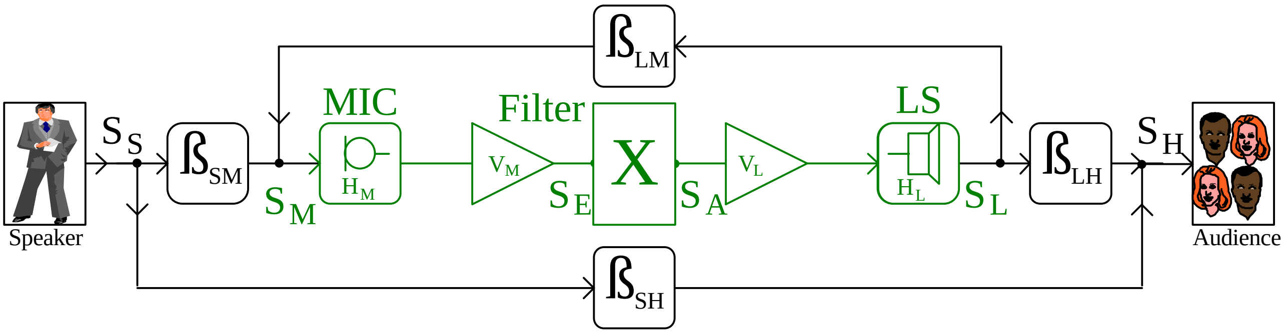

Fig. 1 Block diagram of a simple sound system including filter X

The sound transmission between the loudspeaker and the audience is a combination of the direct room transfer function $ß_{SH}$ and the sound transmission via the sound system including the feedback loop.

$$S_H = S_S \underline{ß}_{SH} + S_S \underline{ß}_{SM} \frac{\underline{H}_M V_M \underline{X} V_L \underline{H}_L}{1 - V_{Loop}} \underline{ß}_{LH} \quad \text{with} \quad V_{Loop} = \underline{ß}_{LM} \underline{H}_M V_M \underline{X} V_L \underline{H}_L \qquad \text{(Eq.1.)}$$

As shown in Eq.1, the loop will become unstable whenever the open loop gain reaches a value of 1 for any frequency. This problem is unavoidable in situations where microphone(speaker) and audience are situated in the same room and must be accepted as a restriction for all such applications.

The principle of a subtraction filter

As its name implies this filter subtracts parts of the signal which have passed once through the sound system. Ideally the filter $\underline{X}$ represents an imitation of the loop components microphone, pre-amplifier, power amplifier, loudspeaker and the room transfer function implemented by an FIR filter $\underline{Y}$ and a subtractor. Unlike the common smoothing filter [6] there are no signal correction functions incorporated into the FIR filter. It consists solely of the impulse response of the loop directly obtained through measurement or the adaptive algorithm of the "Loop transfer function". A more or less precise imitation of the loop can be obtained depending on the number of coefficients in the filter and the length of the impulse response of the loop (Filter function $\underline{Y}$ in Fig.2). If now the filter output signal is subtracted from the filter input signal, a partial cancellation of the signal which has passed once through the loop and returned to the microphone will be obtained. Eq's.2 and 3 describe the effect of the subtraction filter on the sound system. Here, the stability of the system is no longer dependant on the loop transfer function (Eq.1) but on the difference function between the actual loop and its imitation with the FIR filter. Subject to the filter resolution, the influence of the feedback loop to the system can be reduced thus increasing stability.

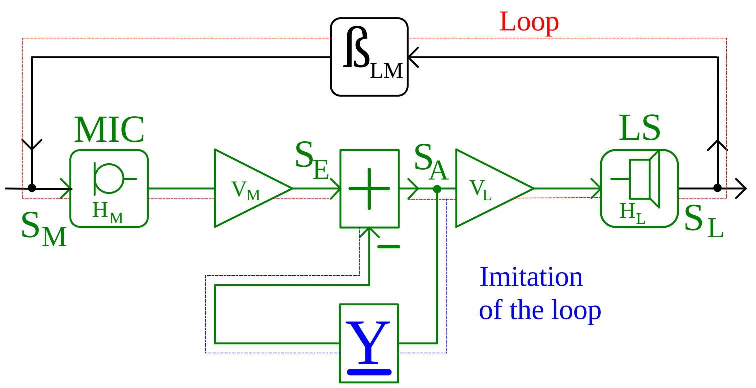

Fig. 2 Loop with subtraction filter

Also obvious from Eq.3 is the fact that the filter function Y no longer appears as a factor in the transfer function of the entire system. It now becomes part of the denominator in the difference function. The chance of falsification of the useful signal passing through the system for the first time and not affected by the filter is thus eliminated. The following idealised example shows a filter with 3392 taps and a sampling rate of 44,1 kHz in a room with an approximate reverberation time of 1,2 seconds. The filter subtracts the first 76 ms from the loop impulse response of the system (Fig.3) and as a result of this the maximum of the loop gain can be reduced from an initial 0 dB (Fig.4a) to -5 dB (Fig.4b).

$$S_H = S_S \underline{\text{ß}}_{SH} + S_S \underline{\text{ß}}_{SM} \frac{\underline{H}_M V_M \frac{1}{1+\underline{Y}} V_L \underline{H}_L}{1 - V_{Loop} \frac{1}{1+\underline{Y}}} \underline{\text{ß}}_{LH} \quad \text{with} \quad S_a = S_e \frac{1}{1+\underline{Y}} \qquad \text{(Eq. 2)}$$

$$S_H = S_S \underline{\text{ß}}_{SH} + S_S \underline{\text{ß}}_{SM} \frac{\underline{H}_M V_M V_L \underline{H}_L}{1 + \underline{Y} - V_{Loop}} \underline{\text{ß}}_{LH} \qquad \text{(Eq. 3)}$$

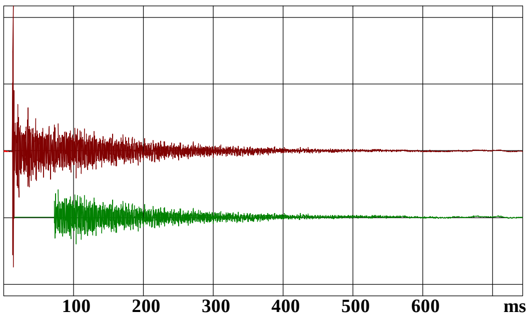

Fig. 3 Loop impulse response without filter respectively with filter (see below)

Influence of microphone and loudspeaker arrangement on the loop transfer function

There are certain statistical statements which can be made concerning room transfer functions [9]. Values above the lowest cut-off frequency of the room which are usually, depending on the size of the room, at the lower end of the frequency spectrum important to sound reproduction systems, can be described as a succession of peaks and dips with an average interval of 2-4 Hz.

The highest peaks are approx. 10-12 dB above the average value (Fig.4a). The function is effected by a great number of interfering wall reflections and is strongly related to the position of both sender and receiver within the room. Even minimal changes to the position of a microphone may result in significant change to the peaks and dips. Fig.6 shows sections of two completely different loop transfer functions brought about by changing a microphones position by approx. 20 cm. On the other hand a smoothed curve shows hardly any differences. Decisive for the maximum gain in a sound system however, is always the greatest peak in the function and not it's average slope.

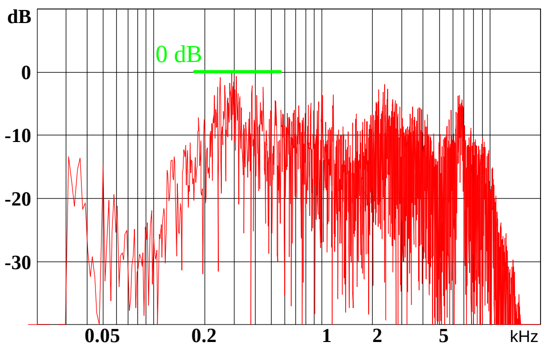

Fig. 4a Loop gain without filter

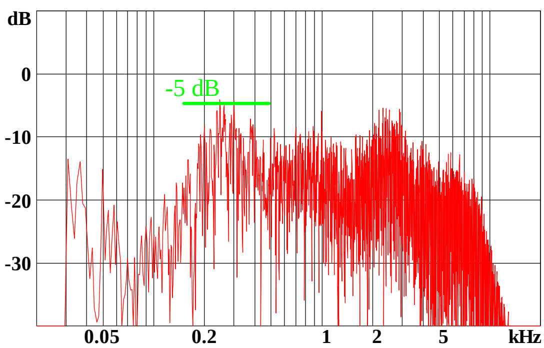

Fig. 4b Loop gain with subtraction filter

Adaptive FIR filter

Nowadays adaptive FIR filters can be found in many areas of the communications electronics. One of the most well-known applications is in the echo elimination in telephone networks and handset-free telephones [7][8]. They are also incorporated in the modulations analysis, most typically where a constant adjustment of filter characteristics is required. Digital filtering accomplishes this with a FIR filter which is equipped with variable coefficients. The imitation of a certain transmission line, in this case the sound system loop, is carried out with the aim to creating the most precise duplication possible. The most popular and simple algorithm for optimum adaptation of an unknown loop function is the LMS (Least-Mean-Square) algorithm. Fig.5 shows the configuration of the adaptive filter and the unknown loop function with the subtractor and the error signal e(k). The speed of the adaptation is directly related to the correlation characteristic of the input signal x(k) and the possible existence of disturbance signals. For the implementation of adaptive filters in sound reproduction systems as shown in fig.2, the signal to the microphone cannot be used to support the adaptation process. This must be regarded solely as a disturbance signal. In comparison to the echo elimination in telephone networks the signal to the microphone s(t) corresponds to the close-up speaker; a distant speaker does not exist. To enable the adaptation of the loop an external auxiliary signal m(k) is introduced to the system. Here, white-noise as opposed to speech or music signals is more suitable. An even better result may be obtained in the convergence of the original signal with a perfect sequence auxiliary signal [8][10]. It must be noted that the use of such auxiliary signals in the sound system can only be implied in the set-up phase of the system before the actual transmission takes place or, in the case of necessary continual operation, the auxiliary signal must be sufficiently masked by the original signal. As to be expected, an adaptation on the basis of the A-weighted curve has proved to be the best solution for the frequency-weighting of the auxiliary signal [3].

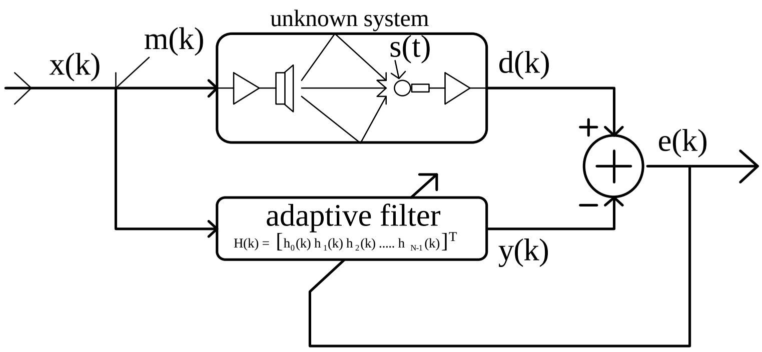

Fig. 5 Block diagram of an adaptive FIR filter

It is possible to do without any kind of auxiliary signal if the object of the application is not to intercept the feedback or resonance of the system in advance. Here, the filter will immediately adapt, once a resonance or feedback at the frequency with the highest open loop gain appears. This is due to the high signal energy at this frequency. In this situation the loop gain can be increased during the set-up phase to the point where the disturbance occurs

and the filter is introduced. Here, a change in the positioning of the system-components leads only to temporary disturbances which are immediately recognised and cancelled out by the filter.
As well as the swift adaptation, the length of the filter impulse response is also crucial to an effective improvement in stability of the subtraction filter. The more energy introduced into the part of the loop impulse response being subtracted by the filter, the greater the improvement to stability. With reference to the frequency domain, this goes hand in hand with the ability of the filter function to reproduce the exerted peaks of the loop transfer function as accurately as possible. Fig.7 shows a section of the measured loop transfer function and the corresponding section of the filter function.
It is evident that the insufficient filter resolution becomes critical when there are exerted peaks surrounded by wide dips. The peak at 160 Hz does not appear in the filter function and this leads to an error of nearly 15 dB. Due to this problem a test hardware, set up in a room with 1,2 seconds reverberation time, could only improve the stability of the system by 5 dB in spite of a calculation power of 340 MIPS. The impulse response and the loop transfer function are shown in figs.3 and 4.

Perspectives and problems

As in all fields of digital signal processing, better results can be expected with the introduction of improved DSP's with greater calculation capacities. Particularly the imitation of the complex loop impulse response would be more exact. In practical use, the implementation of a subtraction filter in the sound system enables an increase of amplification without the introduction of disturbing resonances by 3-10 dB depending on filter length and reverberation times. Sound systems incorporating subtraction filters are often more "good natured" and non-volatile with regards to feedback tendencies. Despite the swift adaptation and filtering leading to the capture of howling feedback, this function, as well as all other possible functions, requires no external operation.

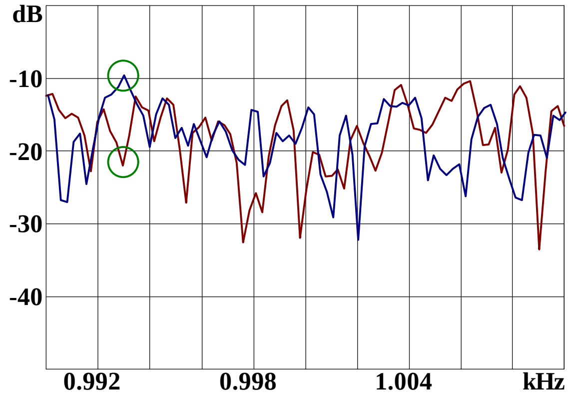

Fig. 6 Increased part of the transfer function for two different microphone positions

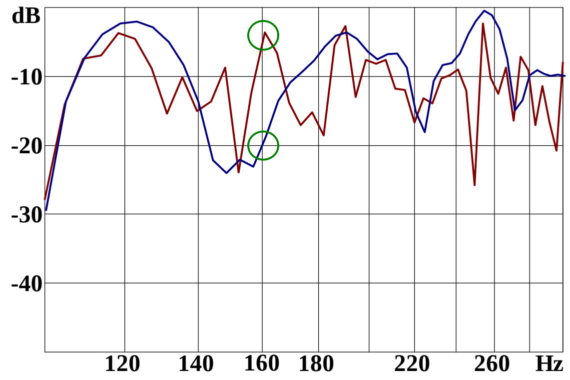

Fig. 7 Increased part of a transfer function and the accompanying adaptive filter function (60 ms filter length)

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