A FIRST PRINCIPLES METHOD TO RAPIDLY OPTIMISE THE ACOUSTIC GAIN OF A SOUND SYSTEM WITH MULTIPLE LIVE MICROPHONES

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1 INTRODUCTION

The concept of feedback in audio amplifiers is well known by audio practitioners and amplifier designers who are well versed in ways to optimize the electronic loop gain parameters that determine the performance of the overall system.

In the domain of sound reinforcement systems, the term “feedback” is colloquially used to describe the howling sound that occurs when a system’s gain is excessive. Although the audio community understands that a sound system provides gain, the concept of acoustic loop gain is not commonly considered.

As with any system that has a closed feedback loop, the ratio of the output sound level to the input sound level depends on how well the acoustic loop gain has been optimized. When sound reinforcement systems incorporate numerous microphones, such as in courtrooms and parliaments, optimizing the loop gain for each microphone can be a time-consuming process.

This paper describes a method to optimize the acoustic gain of a sound system, which is both quick to implement and reliable.

2 FEEDBACK SYSTEMS

2.1 Classical Structure

Figure 1 shows the classic simplified feedback structure of a typical amplifier, be it a large or small-signal amplifier. Also shown is a switch which breaks the closed feedback loop and allows the open-loop transfer function to be measured. Note the use of a negative summing junction after the input; this is why it is called negative feedback.

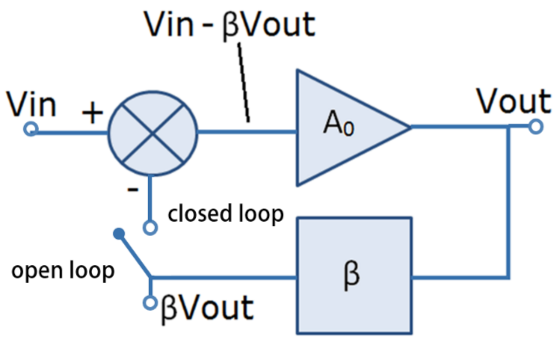

Figure 1 Classical feedback system

Some relevant relationships are:

- Open loop gain $G_{ol} = A_0\beta$
- Closed Loop gain $G_{cl} = \frac{A}{1+A\beta}$
- If $G_{ol} = 1$, i.e. a gain of 0 dB and phase of 180º, then:

$$G_{cl} = \frac{A}{1+(-1)}$$ which is a very big number.

2.2 Regeneration and Degeneration

If part of the amplified sound arrives back at the microphone substantially in phase with the original sound picked up by the microphone, then the amplified sound will regenerate or amplify the original sound. If this becomes audible as a tone, it is commonly known as “feedback”. If another part of the amplified sound arriving at the microphone is substantially out of phase with the original sound, then the feedback will degenerate or cancel the original sound. This loop will become regenerative if the feedback loop provides the Nyquist conditions for instability.

Figure 2 illustrates the four stability phases of an electronic loop with an open-loop phase response of 180º while Figure 3 shows the regeneration and degeneration phases. However, this situation is not identical to the acoustic phases, as we shall see.

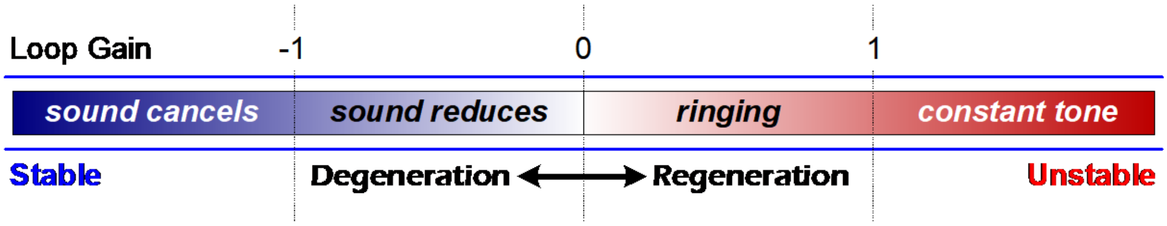

Figure 2 Range of outcomes according to loop gain

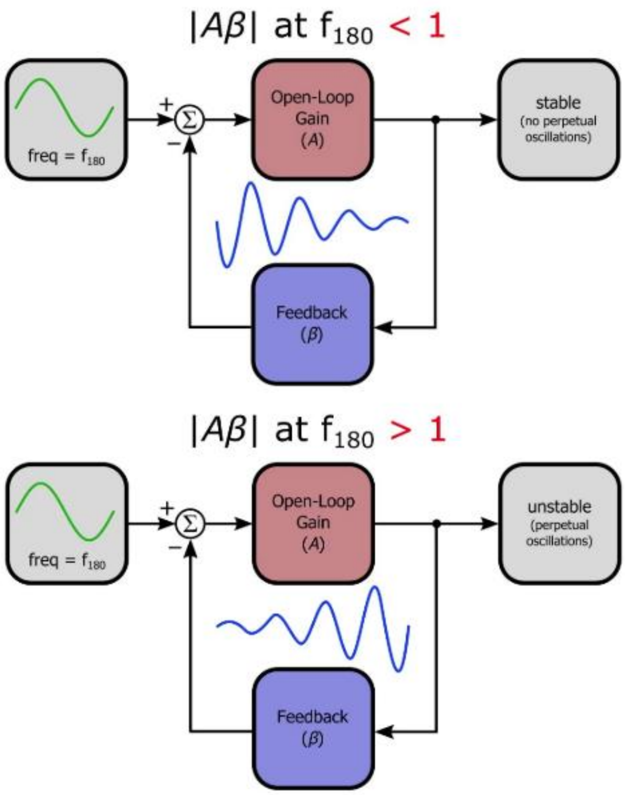

Figure 3 Effect of gain on the degree of regeneration or degeneration (from 1)

2.3 Acoustic Structure of Feedback Loop

A sound system with microphones and loudspeakers sharing a common acoustic space constitutes a closed feedback loop. While ever the microphones can ‘hear’ the loudspeakers, a feedback loop is active. Figure 4 shows the components of the system.

Unlike their electronic counterparts, two of the key components in a sound reinforcement system are contained in both the forward and feedback sections of the system. These components are of course microphones and loudspeakers, which have characteristics in the feedback loop that may be different to the characteristics present in the forward part of the chain. The directional characteristics that microphones and speakers at each frequency can be designed to reduce the open loop gain. However, for the forward gain to be optimised, the talker and listeners should be located on the axis of maximum pickup or sound radiation.

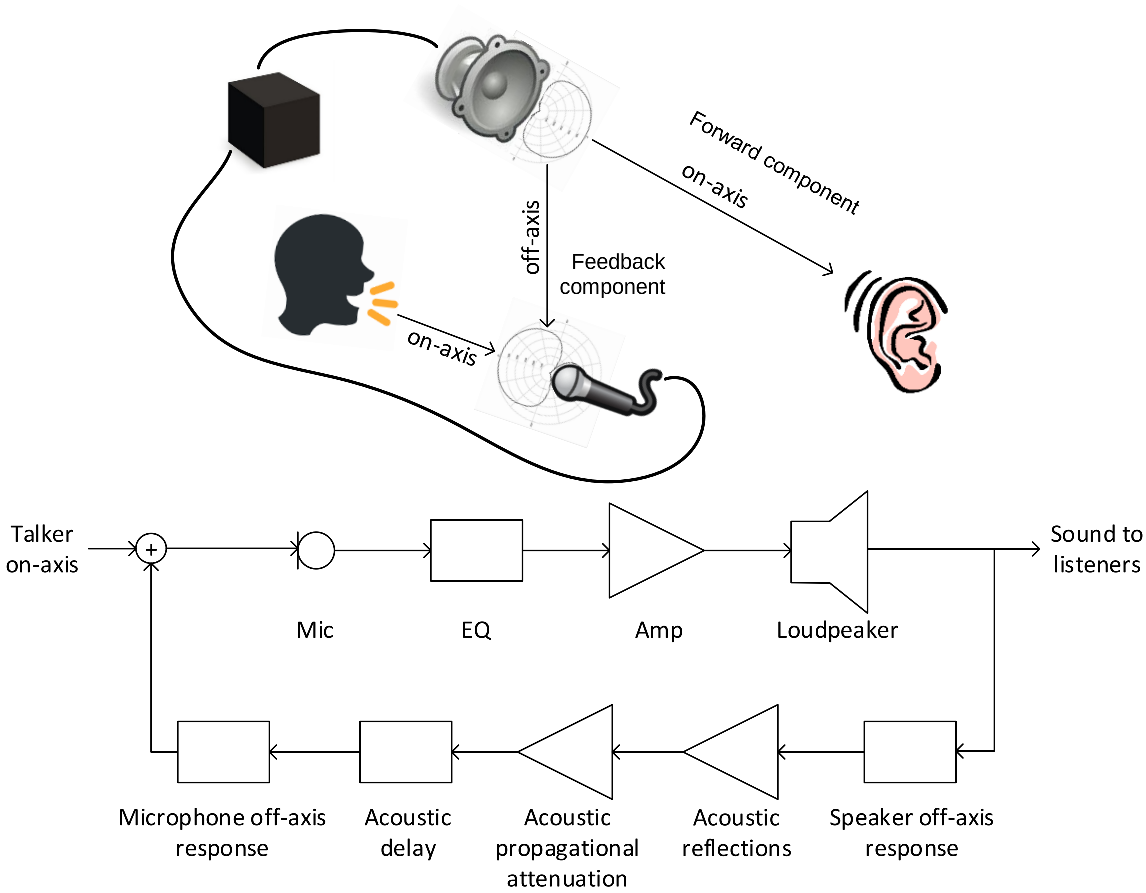

Figure 4 Acoustic feedback loop system.

In contrast to the subtractive summation of the feedback signal that occurs in amplifiers, an additive summation of the feedback signal occurs after the input in acoustic loops. This positive summation produces a closed loop gain: $G_{cl} = \frac{\mathrm{A}}{1-\mathrm{A}\beta}$

When the loop gain $A_0\beta$ is close to 1 and the phase is 0º, the output regenerates and becomes very large.

2.4 Open Loop Responses

It is instructive to compare the open-loop responses of an electronic system with those of a sound reinforcement system. Figure 5 shows the typical open-loop amplitude and phase response of an amplifier's feedback loop. The amplitude roll-off and phase response result from the low-pass filter action of capacitances in the forward block of the amplifier.

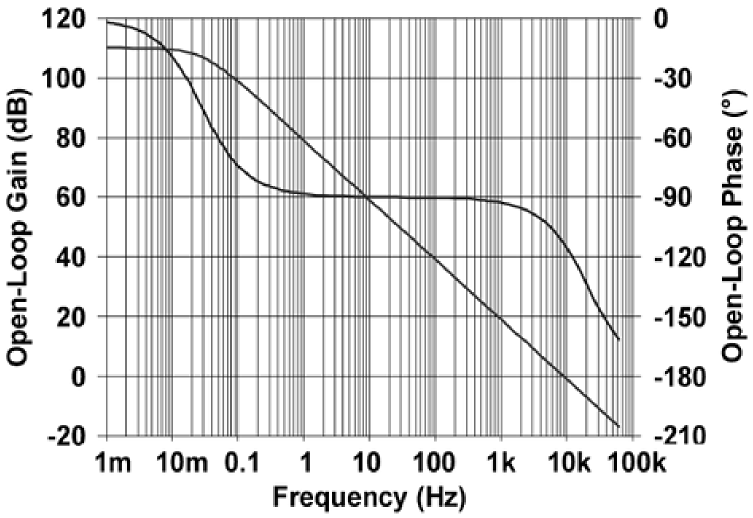

Figure 5 Typical open-loop gain and phase response of an op-amp

Figure 6 shows the schematic representation of a method to measure the open loop response of a sound system. As injecting the test signal into the microphone is time consuming and requires a loudspeaker with an ultra-flat frequency response, the authors inject the test signal electronically. In this situation, the feedback- summation block of this test system is electronic, (rather than acoustic as would occur in actual system operation) and the microphone can be regarded as being wholly within the feedback loop.

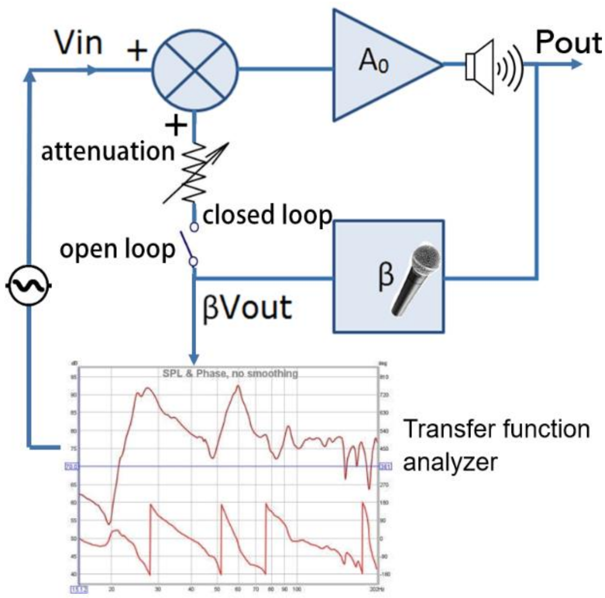

Figure 6 Schematic arrangement for measurement of open and closed loop responses of a sound system.

The shapes of the acoustic open-loop amplitude and phase responses of a sound system are much different to those of amplifiers. Figure 7 shows an example of the frequency response of the open-loop amplitude (gain) of a sound system. In contrast to the decreasing response of the electronic system with increasing frequency, the open-loop amplitude response of this sound system is relatively constant above 400 Hz. The reason why the response below 400 Hz is reduced in this example was not investigated by the authors.

Of equal importance to the relatively flat amplitude response is the cyclical form of the phase response, which varies between 0 and 180°. This cyclical form is quite different to the phase response of an electronic feedback loop and results from the propagation of sound in space and time within the feedback loop. The cyclical form also implies that both regeneration and degeneration are permanent hallmarks of a sound-reinforcement system when mics and loudspeakers share a common acoustic space.

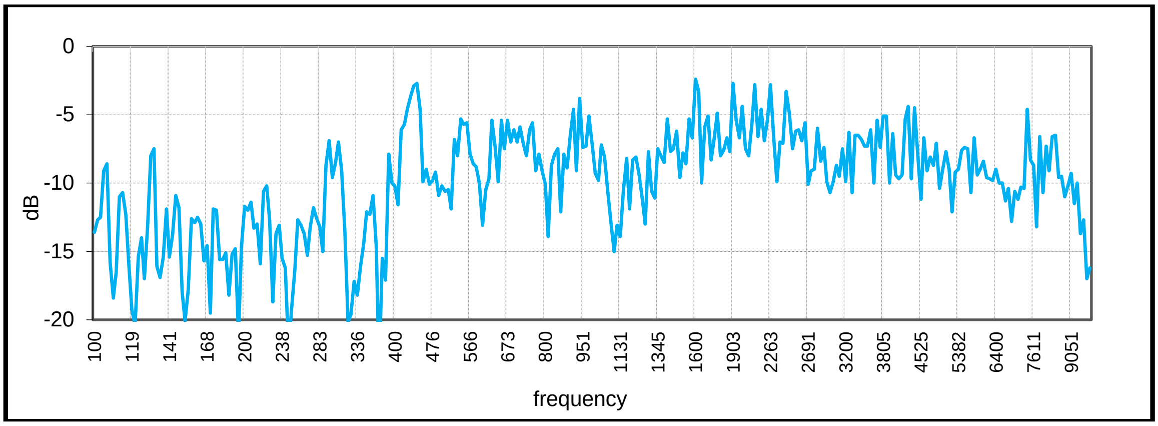

Figure 7 Example of the open-loop gain of a sound system.

Figure 8 shows the open and closed-loop amplitude responses and the open-loop phase of a microphone of a courtroom sound system. The responses show that regeneration and degeneration continually occur. Green lines indicate examples of frequencies at which regeneration occurs, which correspond to an open-loop phase response of 0 degrees.

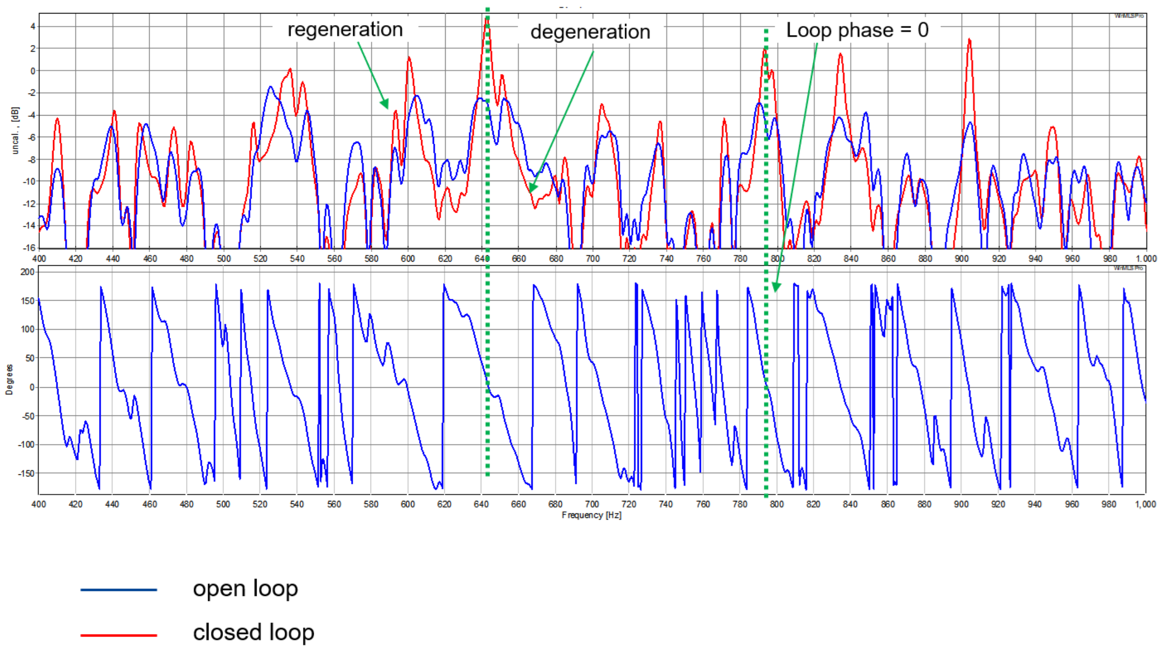

Figure 8 Open and closed responses of a speech reinforcement system.

3 STABILITY AND EQUALISATION

3.1 Gain and Phase Margins

General rule of thumb to provide a satisfactory stability of electronic feedback systems are that the open-loop response should have gain and phase margins as follows:

- Gain margin: gain should be less than -6 dB when the phase shift is 0º.
- Phase margin: phase should be greater than 60º when the gain is unity.

Figure 9 shows a generic open-loop amplitude and phase response illustrating gain and phase margins.

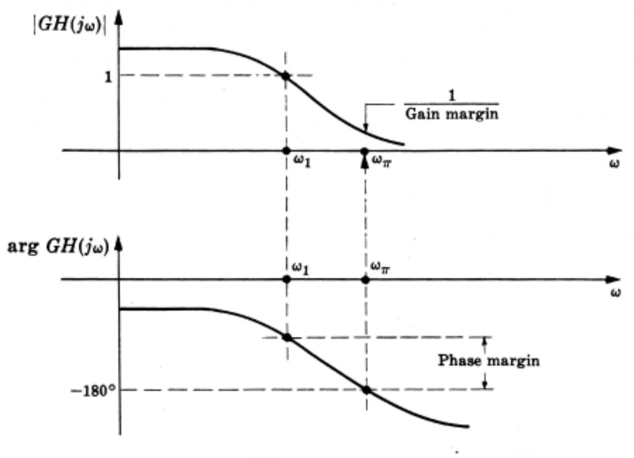

Figure 9 Generic open-loop gain showing amplitude and phase.

3.2 Equalisation

3.2.1 Amplitude

Equalisation of the amplitude of the open loop gain is easily executed using narrow-band notch filters with targeted shapes and depths. These filters are generally implanted by biquadratic IIR filters.
There are two primary aspects to achieving appropriate equalisation:

- A suitable target for the overall open-loop gain must be chosen, to which frequency response peaks in the open-loop gain must be reduced. This is discussed further in Section 3.3 below.
- The equalisation must carefully match the inverse shape of the peak in the frequency response in the open-loop gain.

3.2.2 Phase Equalisation

Phase equalisation that is both robust and targeted is not readily implemented; this difficulty is described in the following rationale with reference to Figure 10, measured in a parliamentary sound system. That figure shows the amplitude responses of the open and closed loops and the open-loop phase response over the 1/3 octave spanning 1.6 kHz to 2 kHz. There are approximately 27 phase rotations in this range. The brown dotted line indicates two peaks in the open loop response in which both 0º and 180º phases occur. In these narrow frequency regions, simply flipping the phase will result in new frequencies with a phase of 0 degs occurring within the amplitude peaks.

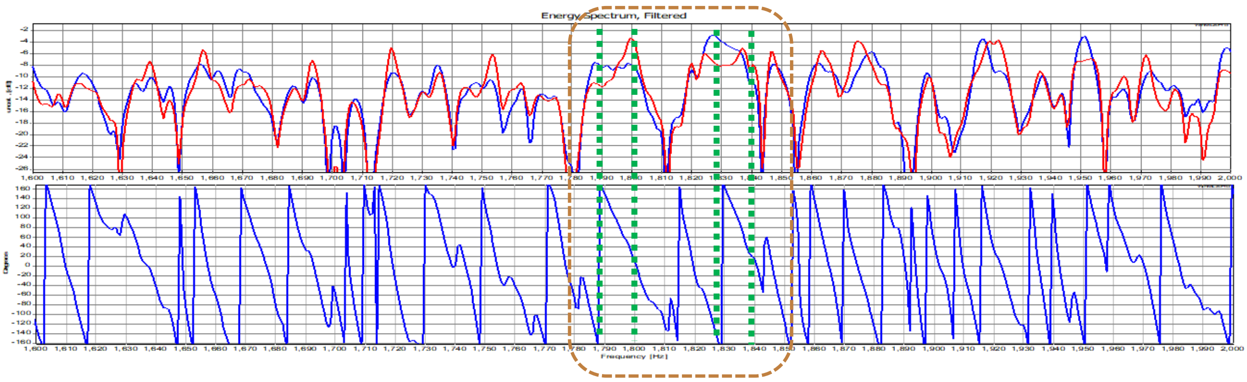

Figure 10 Open and closed loop amplitude and open loop phase responses showing multiple frequencies at which loop phase is 0º and 180º and amplitude response is high.

Simple static phase equalisation is not feasible for the following reasons:

- To substantially change the phase at frequencies with 0° phase whilst not affecting the nearest neighbours with high amplitude responses requires phase equalisation that has extremely narrow bandwidth. All-pass filters based on IIR prototypes are incapable of the ultra-narrow bandwidths without highly-audible ringing.
- Within a very small frequency range of a given amplitude peak in the open loop response, the phase response could include both 0º and 180º situations, so changing the phase at one frequency is likely to cause undesirable phase response at another, both of which share a peak in the amplitude response.
- FIR filters are not necessary at cure as they introduce additional delay, which will change the phase response.
- Small changes in the phase due to flight distance or reflection could easily occur during normal operation, rendering any static phase equalisation inaccurate.

Dynamic phase changing devices such as pitch changing devices can introduce serious degradation in the sound quality.

3.3 Selecting a Suitable Stability Target

The target that the authors use deals only with the gain margin and ignores the phase margin for the reasons described above. We have found that an open-loop gain of between -6 dB and -4 dB provides a good balance between maximizing acoustic gain and minimising colouration from regeneration.

The extent of equalisation that is required to achieve the target depends on the overall flatness of the open-loop frequency response:

- if only a few frequencies are above the target, they can be reduced by equalisation.
- if many frequencies lie above the target, the overall gain should be reduced to prevent damage to the tonality by excessive equalisation.

We have also found that applying a low-frequency shelf filter to the target to reduce the open-loop gain at low frequencies helps to minimise the amount of ringing and colouration due to regeneration. That shelf often commences its roll off at 800 Hz and plateaus out at 150 Hz. Figure 11 shows an example of closed and open loop magnitude responses and the equalisation target.

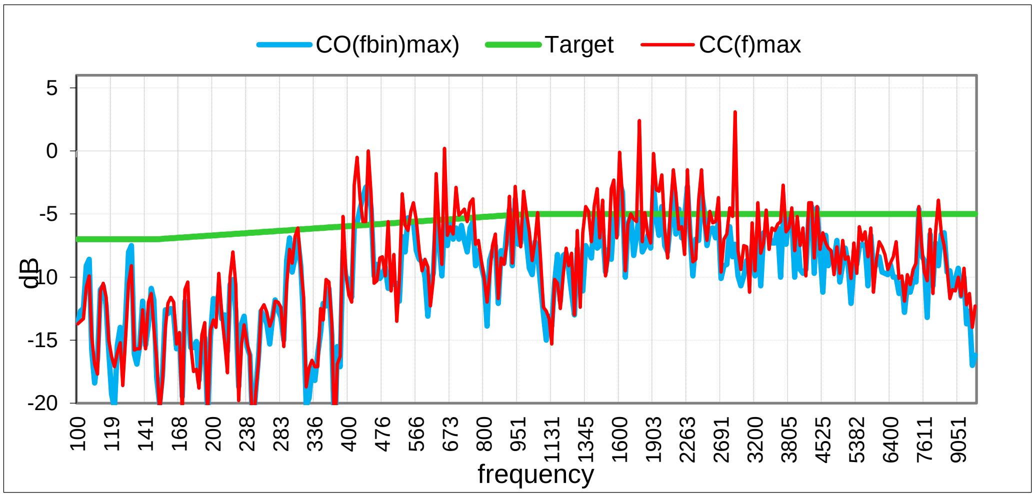

Figure 11 Example of closed and open loop magnitude responses and the equalisation target.

We do not fully understand why the loop gain need to be lower at low frequencies for optimum sound quality. However, we consider that that the following factors will be associated with this issue:

- Sound reinforcement systems are always regenerating or degenerating, although we may not perceive it as such.
- The components of speech utterances often have a longer duration at low frequencies than at high frequencies. The regenerative components of speech at higher frequencies will come and go quickly and possibly be masked by lower frequencies.
- The spectrum of speech is generally highest in the 200 Hz to 500 Hz band, and this means that the loudest regeneration will often be in this frequency range.
- Sound reinforcement systems have longer regenerative build-up and decay times at low frequencies compared to high frequencies, resulting from the longer period required for the same number of cycles compared to a similar number of cycles at high frequencies.
- This method was developed in rooms that had relatively constant reverberation times (RT) between 100 Hz and 5 kHz, and therefore we conclude that higher RTs at low frequencies are not associated with this issue.
- It is possible that the combination of the longer duration of regeneration and the time constant of loudness-growth in human hearing could lead to a higher perceived sound level at low frequencies.

4 OPTIMIZATION

4.1 DSP Implementation

Figure 12 shows the block diagram of typical configuration of a DSP unit for measuring loop gain. The following elements form the conceptual blocks shown in Figure 6:

- The test signal is injected into the system before the microphone signal is distributed to the various zones.
- The output of the microphone pre-amplifier represents the output of the loop and is fed to the analyser input.
- For closed-loop measurements, summation of the test signal input and the feedback signal from a microphone is implemented in a simple mixer block.
- The relative gains of the microphone channel to each loudspeaker are retained and utilised for these measurements.

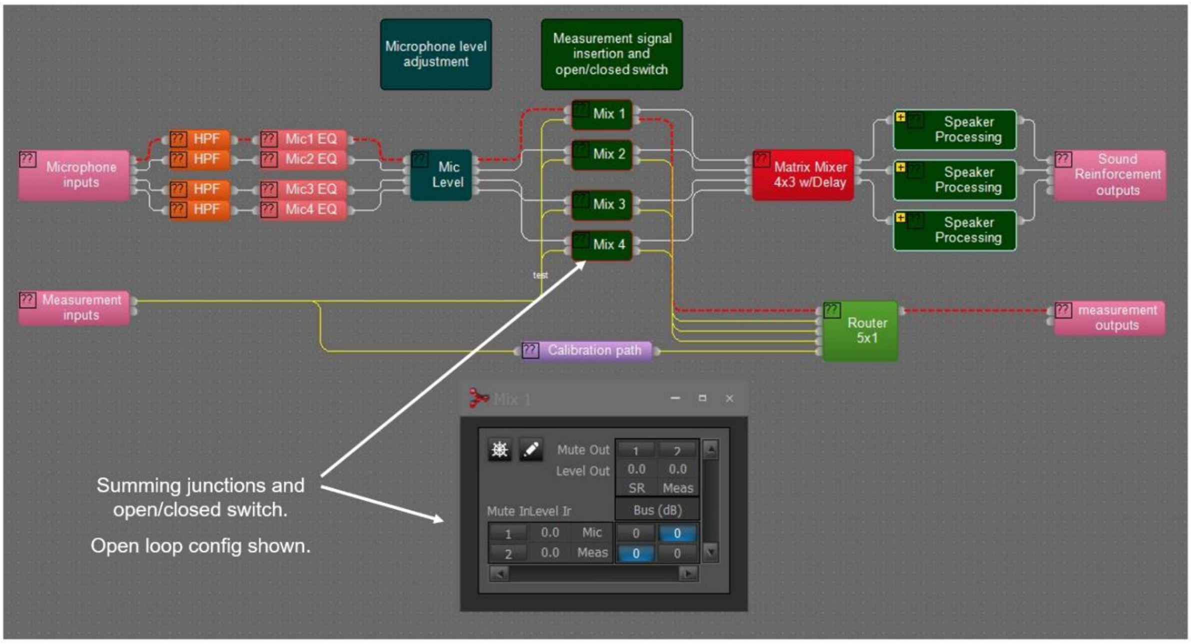

Figure 12 Block diagram of typical configuration of DSP unit for measuring loop gain.

4.2 Steps to Measure the Open and Closed Loop Responses

Before optimising the loop gain responses, the sound system must be fully optimised. This optimisation usually requires the following parameters to be addressed:

- Loudspeakers are optimally equalised.
- Delays between loudspeakers are optimally set.
- Any equalisation applied to microphones to optimise tonality should be implemented.
- The relative gains applied to a microphone feed to different zones are set.
- The relative delays applied to a microphone feed to different zones are set.
- All latencies in the DSP system that will occur in normal operation of the system must be in place.

The following steps are used to measure the open and closed-loop frequency responses:

1. Calibrate the measurement system with a unity gain return path so that the analyser reads 0 dB.
2. An assistant sits or stands at the microphone under test at the normal speaking position.
3. Turn on the microphone to be optimised, activate the closed-loop switch and force the system into controlled audible regeneration by slowly increasing the gain.

   This process allows frequencies with peaks in the response and a phase of approx. 0° to regenerate. This process eliminates the need to analyse the phase response separately, since the differences between closed and open-loop responses show occurrences of regeneration.
4. Define CO (Combined Open) as the open-loop magnitude response and CC (Combined Closed) as the closed-loop magnitude response.
5. Measure CO and CC impulse responses using very slow (20 second) logarithmic sine sweeps as the test signal. This sine-sweep duration provides sufficient time for low frequencies to regenerate.
6. Obtain frequency domain responses with high resolution FFT (2^17).

4.3 Determining the Loop Gain Equalisations

The following steps are used to determine the equalisations:

1. Define the level of the CC response above the CO response at which the closed loop is considered to be in regeneration – set to 0.5 dB by trial and error (named “regen_margin”).
2. Create a “target” open-loop gain consisting of a line in the frequency domain (trial and error suggests -6 dB above 800 Hz, to -10 dB at 80 Hz).
3. Find frequencies with CO>target AND CC-CO>regen_margin.
4. For frequencies where the above is TRUE: equalisation E(f) = target-CO
5. If there are many frequencies above the target line over the entire frequency range, then the overall level should be reduced or the target line increased, as equalisation will effectively produce the same effect, but with poorer sound quality.
6. Using a peak-detector within each band, allocate the spectra into 1/48th octave bands for convenience.
7. Apply automated filter-response matching procedure to produce inverse parametric filters.
8. Export filters to the DSP unit.

Typical filters that authors have found necessary range from 1 to 4 dB deep with bandwidths of between 0.1 and 0.3 octaves. It is noted however, that the loudspeakers and microphones in the sound systems in which this method has been used have off-axis frequency responses that are smooth and relatively flat. Figure 13 shows an example of an equalisation developed for a courtroom system.

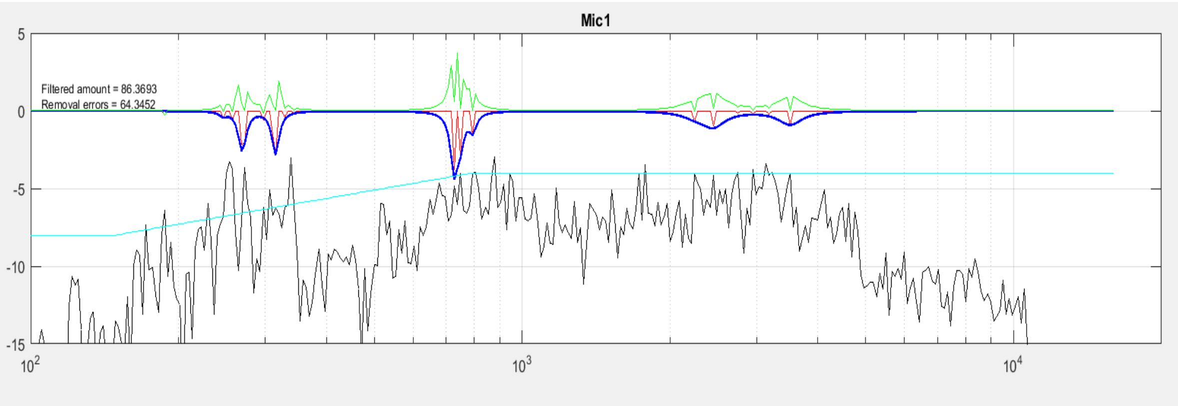

Figure 13 Example of an equalisation developed for a courtroom system.

In Figure 13, the black trace is the equalised open loop gain and the light blue is the Target line. The red trace represents the inverse of the response peaks to be equalised, the dark blue trace represents the actual equalisation IIR biquad filter responses and the green trace shows the equalisation error resulting from the biquad implementations.

5 DISCUSSION

a) With loudspeakers that have both constant directivity and a smooth frequency response and microphones that are optimally orientated with a smooth off axis response, we have generally found an increase in overall gain of approx. 3 dB after equalization. This translates into a decrease in the equivalent acoustic distance[2] of 40%, which is highly worthwhile. With systems that have only a few high-level peaks in in the open-loop frequency response and the bulk of the open-loop frequency response is well below the target, the increase in gain that this method provides would be much more than 3 dB.

b) In systems that use multiple loudspeakers to cover different areas with only one speaker effectively feeding sound back to the microphone, this method will apply unnecessary filtering to the other loudspeakers, which may be audible. To address this, the method can be expanded by determining the relative contributions of each loudspeaker to the overall open-loop gain and equalizing each loudspeaker's feed proportionately. This method is described by Leembruggen[3].

c) The method does not account for room acoustic effects imposed onto the talker's sound that are picked up by the microphone which add vectorially with the talker's voice. Important reflections that are excluded from the input chain are room modes and image sources. The talker's ability to excite these reflections is different to the loudspeakers, due to the different radiation pattern and source locations.

d) The authors always undertake a final listening check with transient utterances to check for colourations and lingering sounds, which are likely to result from the room-acoustic effects described above. Filtering is then applied to soften or remove these effects.

6 CONCLUSION

This paper describes a measurement method to reliably increase the gain of a sound reinforcement system. The method identifies those peaks in the amplitude frequency response of open-loop gain that exceed the nominated gain-margin and whose associated phase response is close to 0º that would produce strong regeneration in normal operation.

Using a calibrated measurement of the open-loop gain, an equalisation is automatically calculated to reduce the level at these frequencies to a nominated open-loop gain target.

7 REFERENCES

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