Investigation of an AGC for Audio Applications

Haerizadeh, Seyediman; Jørgensen, Ivan Harald Holger; Marker-Villumsen, Niels; Bruun, Erik

Published in:
Proceedings of IEEE PRIME 2015

Link to article, DOI:
10.1109/PRIME.2015.7251355

Publication date:
2015

Document Version
Peer reviewed version

Link back to DTU Orbit

Citation (APA):
https://doi.org/10.1109/PRIME.2015.7251355
Investigation of an AGC for Audio Applications

Seyediman Haerizadeh, Ivan H. H. Jørgensen, Niels Marker-Villumsen and Erik Bruun
Technical University of Denmark, Dept. of Electrical Engineering, DK-2800 Kgs. Lyngby, Denmark
seyha@elektro.dtu.dk, ihhj@elektro.dtu.dk, nbvi@elektro.dtu.dk, eb@elektro.dtu.dk

Abstract—An investigation of an amplifier with discrete time Automatic Gain Control (AGC) which is intended for implementation in hearing aid is performed. The aim of this investigation is to find the AGC’s minimum gain step size for which the glitches become inaudible. Such AGCs produce undesirable glitches at the output turning into audible sound effects. In order to find this minimum gain step size both objective and subjective evaluation methods have been used. The investigations show that the objective measures indicate a lower limit for the step size where the sound artefacts are no longer audible. This is in contrast with the subjective method where several test persons can hear the sound artefacts for all step sizes. Thus, the investigated AGC is not suitable for IC implementation therefore an alternative AGC system is proposed.

Keywords— Hearing aid, Microphone Channel, Automatic Gain Control, PEAQ, Objective evaluation, Subjective evaluation

I. INTRODUCTION

Since using a hearing aid is associated with being old and senile, the persons wearing them have the tendency to hide them. Therefore the need for designing hearing aids which are so small that they are partially invisible is paramount. Small hearing aids restrict the designer in terms of power consumption and in the majority of the smallest hearing aids power consumption is limited to 1 mW. In addition to the power limitation, the required dynamic range is also very high (more than 100dB). In order to increase the dynamic range one can increase the supply voltage but the supply is limited to the voltage of a single zinc-air battery which is approximately 1.0-1.2V. Therefore making a microphone amplifier and a ADC (called a Microphone Channel) with a Signal to Noise Ratio (SNR) of more than 100dB will be too power hungry. Thus an AGC can be used in hearing aid to increase the DR and hence designing the circuitry for more moderate SNR[1]. A microphone channel for hearing aid applications is shown in Fig. 1.a. When the signal at the output of a variable gain amplifier (VGA) exceeds a certain level the AGC reduces the gain to avoid clipping. Simultaneously the digital gain will be increased by the same level to keep the total gain of the channel constant and will do vice versa when the signal level is reduced to a specific level to prevent the degradation of the SNR. Effectively the AGC increases the DR (Dynamic Range) by the amount, the gain can be adjusted in the AGC. However it introduces glitches into the system as a result of its gain switching. These glitches sound like clicks and are undesirable. The generation of a glitch is illustrated in Fig. 1.b.

Therefore the AGC should be designed in such a way that these glitches be so small that are not audible. One way of doing this is reducing the AGC’s gain step size while increasing the number of the gain switching occurrences. On the other hand, decreasing the gain step size increases the AGC complexity in the sense of analog electronics design. Therefore it is needed to investigate that what is the minimum gain step size for which the glitch is not audible anymore and at the same time minimizing the complexity of the VGA. To do this an audio quality evaluation of the microphone channel output is required to find this minimum gain step size.

The modeled version of the microphone channel is shown in Fig 1.b. The model is implemented in MATLAB, it includes an up-sampling stage, an AGC (Automatic Gain Control) which acts as the mentioned gain control system for increasing the dynamic range of the system, a third order Butterworth low pass filter with the cut-off frequency of 30 kHz which is emulating the transfer function of the ADC, a digital gain compensation stage and finally a down-sampling stage, it should be mentioned that in this model and investigation, the circuit noise is not considered.

For evaluation of the sound quality two methods are available. One is the subjective method which is performed by running an alternative forced choice listening test [7] and the other solution is using an objective method. The subjective method is time consuming in comparison with the objective one, however it is more precise. The objective method which is used, is PEAQ (Perceptual Evaluation of Audio Quality) [10]. This method estimates the audio quality of the signal by incorporating the human auditory system properties. For our specific application (evaluation of the glitches audibility) the two recommended MOVs (Model Output Variables) of PEAQ [5,6] that are ADB (Average Distorted Block) which returns the logarithm of the ratio of the total distortion to the total number of severely distorted frames and MFPD (Maximum Filtered Probability of Detection) which measures the maximum of the probability of detection after low pass filtering, are utilized. These two objective metrics are used to
assess the transient error level of the signals. It is also important that the reliability of these metrics in defining the AGC’s minimum gain step size for which the glitches cannot be audible anymore, be investigated. [2,3,4]

II. AUTOMATIC GAIN CONTROL

Three principal parameters are considered to be the foundation for an AGC implementation, these parameters which control the behavior (function) of an AGC are attack time, release time and the gain step size. Different combinations of them can produce glitches with different patterns and audibility level. The attack time is the time that we control between two attack events and the release time is the time that we control between two release events. Three different zones are defined for the AGC which are called attack zone, release zone and dead zone, shown in Fig. 2. The AGC’s attack zone is set to be above 0.8 (relative to the supply voltage) and below -0.8. If the output of the VGA is within the attack zone the gain is reduced by one gain step for an attack time. The AGC’s release zone is set to be between 0.6 to -0.6 and the zone between the attack and release zones is called dead zone which basically no attack or release happens in this zone. AGC makes decision based on in which zone, the detected output of the VGA (Variable Gain Amplifier) is lying and then it attacks or releases or does neither.

The AGC can be designed with different gain step sizes, reducing this gain step size makes the RMS error value of a glitch smaller, therefore the glitch becomes less audible. The RMS error value is obtained by subtracting the microphone channel output (Fig. 1.b) from the filtered value of the input (passed through the same filter as mentioned for the microphone channel) and taking RMS from it as it is shown in Fig. 3. The reason for filtering the input and then subtracting it, is to compensate the phase shift which is applied to the signal passing through the microphone channel. It is critical that the AGC’s attack and release time effect on the audibility of the glitches be eliminated so that the RMS error value becomes only step dependent and only the effect of the step size choice on the audibility of the glitch is being investigated. Thus a total attack and total release time have been considered for the system which will be the same for all the step sizes as will be proved further on. By assuming the RMS error value for 1dB step size to be \( e_{\text{rms,1}} \) and assuming that \( N_1 \) is the number of the occurrences for the 1dB glitches, the total RMS error value will be:

\[
e_{\text{total RMS error,1}} = e_{\text{rms,1}} \sqrt{N_1}
\]

(4)

Now if the step size is reduced by a factor of \( K \), each RMS error value of a glitch with the gain step size of \( K \) must be approximately:

\[
e_{\text{rms,K}} = e_{\text{rms,1}} \frac{1}{K}
\]

(5)

However as \( T_{\text{At,step}} \) and \( T_{\text{Re,step}} \) is also reduced by a factor of \( K \), the number of the occurrences must be increased approximately by a factor of \( K \). Thus the total RMS error value for \( K \) is:

\[
e_{\text{total RMS error,K}} = e_{\text{rms,1}} \frac{1}{K} \sqrt{N_1} = e_{\text{rms,1}} \sqrt{N_1}
\]

(6)

This simplified analysis shows that there is no obvious choice for the step size, as the total RMS error value regardless of the step size is almost the same. Hence the need for investigation for finding the minimum gain step size arises which is fulfilled with audio quality evaluation methods.

Clearly using large gain steps in the AGC will result in large glitches and thereby a system of no practical use as the glitches will be very audible. On the other hand using very small gain steps introduces another problem. Consider a low frequency sinusoidal input. The main idea of the AGC system is to adjust the gain in the VGA such that the peak of the signal at the VGA output is located in the dead zone and thus the AGC enters a steady state. However, as the gain steps are very small the release time per step will also be very small and thus the AGC will increase the gain as the signal passes through the release zone. If this increase in the gain is sufficiently large the AGC then have to decrease the gain

![Fig. 2. Defining the AGC’s Attack, Release and Dead zones.](image)

![Fig. 3. RMS error value calculation.](image)
again as the signal enters the attack zone. As the step size and thereby the release time is reduced this phenomenon occurs at higher and higher frequencies for the input signal. Again no obvious choice on the gain step size appears.

III. EVALUATION OF THE AUDIO QUALITY

A. Production of the test signals

The test signals for both subjective and objective methods were generated by modeling the microphone channel using MATLAB (Fig. 1.b). A Tuba music sample from the EBU Sound Quality Assessment Material CD [9], which lasts for 2.5s was chosen as the input of the channel (as it was reported to be the worst case scenario with regards to the level of the transient errors (glitches) based on [6]). For generating the sound files, initially each input signal was up-sampled by a factor of 8 from 44.1 kHz into 352.8 kHz to avoid the aliasing of the harmonic distortion which is the result of the AGC activity. At the end, the produced samples were down-sampled to 44.1 kHz and saved as WAVE files. The depth of the AGC was chosen to be -18dB. The test signals were generated by different AGC’s gain step sizes ranging from 0.01dB to 1dB. They were generated with a variety of 100 different step sizes in this range. The reason that 0.01 dB was chosen as the smallest step size in these tests is that by reducing the steps size further the complexity of the circuit increases which makes it very difficult to be implemented in analog electronics. In total two groups of test signals were produced based on two different combinations of total attack time and total release time. One group was generated by choosing a shorter total attack and total release time (total attack time of 1ms and total release time of 100ms) and the other group by choosing a longer total attack and total release time (total Attack time of 4ms and total Release time of 400ms).

B. Objective Method

As the subjective method is time consuming, for finding the AGC’s minimum gain step size for which the glitch is not audible anymore, an objective method can be used. However the selected objective method should prove itself as a reliable tool for the specific application. ADB and MFDB from the PEAQ’s MOVs were used to define the minimum gain step size. The test signals for both test 1 and test 2 were fed into the PEAQ algorithm[8]. The results shown in Fig. 4 and Fig. 5 were obtained for ADB and MFDP, as it can be seen, ADB and MFDP metrics showed that for test 1, for a step size of 0.01dB , both metrics are zero (although for MFDP, it is very close to zero and negligible) and for test 2, for step sizes lower than 0.37dB, the values of ADB and MFDP will be zero, which means that based on these metrics by choosing these thresholds with the mentioned $T_{ADB, total}$ and $T_{MFPD, total}$ one can make sure that the glitches become inaudible or in the worst scenario, the chance of the glitch detection will be intensively minimized ( almost no glitch is audible). Consequently for assuring the accuracy of these metrics, listening tests as the subjective method were executed.

![Fig. 4. ADB and MFDP MOVs for Test1’s test signals.](image1)

![Fig. 5. ADB and MFDP MOVs for Test2’s test signals.](image2)

C. Subjective Method

For investigating the reliability of the objective method in defining the gain step size, we tried to verify these results with a subjective one. Thus a listening test was executed at the Technical University of Denmark (DTU) double-wall sound attenuating listening booth. The test was carried out with the total number of 15 participants in the age range of 25 to 35 years old. All the subjects were interviewed to assure that they are having normal hearing ability. The whole test procedure was approved by the Science-Ethics Committee for the Capital region of Denmark (reference H-3-2013-004).

The listening test was implemented based on three intervals, three alternative forced choice (313AFC) with 1-up 1-down method (which determines the fifty percent detection probability) [7]. The listening test consists of trials, each trial includes three windows, and windows are separated by a short pause. In each trial two of the windows play the reference signal (Tuba music without any error (glitch)), and the one remaining is the one which contains the error for that specific step size, the order of the windows is set randomly. If the subject recognizes the window containing the erroneous signal in a trial correctly then the test will be run again with the same step size. If the second response will be correct as well, the step size will be decreased. However if the answer is wrong then the step size will be increased. The test starts from 1 dB step size, initially the attenuation starts with big jumps (step difference between the two consecutive steps) for finding the subject’s threshold faster which the next following steps will be 0.5 dB, 0.1 dB, 0.05 dB and finally 0.01 dB. The test starts with large step differences, as it goes forward the step size difference will be reduced and at the final part ends into 0.01 dB.
dB step difference. The test continues till the minimum step size for which the test subject can no longer hear any glitches be detected, the mechanism is that, after getting into the minimum step difference (0.01dB) the test will continue for seven more trials and then stops. The mean value of these last 7 trials is the subject’s detected minimum gain step size. Prior to the test execution, the test subjects were trained to increase the possibility of the correct detection of the errors during the test process (according to ITU standards), each subject carried out the two mentioned tests (test 1 and test 2). Fig. 6 shows the obtained listening test results for both test 1 and test 2. The results show the detected minimum gain step size by each test subject for which the glitch is no longer audible. The circle is representative for test 1 and the star is representative for test 2. The mean of the obtained minimum step size detected by the subjects for test 1 is 0.06dB while for test 2 this value is 0.12dB. The minimum step size which is detected in test 2 is pushed up in compare to test 1, the reason is that the total attack and total release time for test 2 is higher than test 1 therefore the quantity of the glitches is lower for test 2 in compare to 1 which makes it more difficult for the test subject to detect the error and shifts the detected minimum gain step size upward. This can indicate that the $T_{\text{A, total}}$ and $T_{\text{R, total}}$ should be made very large but then the AGC will become so slow that clipping at the output of the VGA will start to occur. However some of the test subjects have been able to reach into the minimum gain step size implemented in both test (0.01dB). In [6] it was claimed that for making the transient errors (glitches) to be inaudible, one can target ADB and MFPD for a specific range, which is mapped to step sizes larger than 0.01 dB step size, the reason for this claim is that in [6] the number of steps has been maintained while the step size has been reduced, although we have followed a different logic, which is reducing the step size while having more steps, therefore our results are different with [6]. Consequently we cannot use ADB and MFPD as the objective measures to make any conclusion as the glitch is audible where the ADB and MFPD are not able to evaluate the audibility.

FIG. 6. Listening Test Results (Detected Minimum Gain Step Sizes).

IV. FUTURE WORK

The future work will be to implement a system with a duplicated microphone channel (Fig. 7) which avoids glitches by running the two microphone channels in parallel and then only switch between the two channels when the signal in the channel where the gain is changed has completely settled. This will provide a microphone channel without glitches.

V. CONCLUSION

In this paper an automatic gain control system is investigated to find the gain step size for circuit implementation. The system is evaluated using a MATLAB model for producing output sound. Initially the objective measures Average Distorted Block (ADB) and Maximum Filtered Probability of Detection (MFPD) indicate that a lower limit for the gain step in the AGC exists where the sound artefact are not audible. However, a subjective sound test show that many test subjects can hear the sound artefacts even at gain steps of 0.01dB. Thus, it is not practically possible to implement such an AGC system in circuitry with the sound artefacts being audible. A proposed solution is to implement two AGC channels in parallel using one when the gain in the other one is changed, thereby avoiding the glitches.

VI. REFERENCES