16th European Signal Processing Conference (EUSIPCO 2008), Lausanne, Switzerland, August 25-29, 2008, copyright by EURASIP

AN INTEGRATED APPROACH FOR NOISE REDUCTION AND DYNAMIC RANGE COMPRESSION IN HEARING AIDS

Kim Ngo[1], *Simon Doclo*[1,2], *Ann Spriet*[1,3], *Marc Moonen*[1], *Jan Wouters*[3] *and Søren Holdt Jensen*[4]

[1]Katholieke Universiteit Leuven
Department of Electrical Engineering, ESAT-SCD
Kasteelpark Arenberg 10, B-3001 Leuven, Belgium
email: kim.ngo@esat.kuleuven.be

[2]NXP Semiconductors
Corporate I&T-Research, Sound and Systems Group
Interleuvenlaan 80, 3001 Leuven, Belgium
email: simon.doclo@nxp.com

[3]Katholieke Universiteit Leuven
Department of Neurosciences, ExpORL, O. & N2
Herestraat 49/721, 3000 Leuven, Belgium
email: Jan.Wouters@med.kuleuven.be

[4]Aalborg University
Department of Electronic Systems, MISP
Niels Jernes Vej 12 A6-3, 9220 Aalborg, Denmark
email: shj@es.aau.dk

ABSTRACT

Hearing aids typically use a serial concatenation of Noise Reduction (NR) and Dynamic Range Compression (DRC). However, the DRC in such a concatenation negatively affects the performance of the NR stage: the residual noise after NR is amplified by the DRC, resulting in a signal-to-noise-ratio (SNR) degradation. In this paper, we present an integrated solution for NR and DRC. The solution is based on an estimate of the amount of speech and noise in each time segment. In case the speech is dominant, the NR is less active and it is desirable to have as much DRC as possible, whereas in a noise dominant segment the NR is more active and the idea is not to compromise this operation by applying DRC. Experimental results confirmed that a serial concatenation of NR and DRC degrades the SNR improvement, and that the proposed solution offers a better SNR improvement compared to a serial concatenation.

1. INTRODUCTION

Reduced audibility and reduced dynamic range between threshold and discomfort level are some of the problems that people with a sensorineural hearing loss are dealing with. Furthermore, background noise (multiple speakers, traffic etc.) is a great problem and is especially damaging to speech intelligibility. It is known that hearing impaired people need a higher signal-to-noise-ratio (SNR) to communicate effectively [1]. Therefore, Noise Reduction (NR) and Dynamic Range Compression (DRC) are basic components in hearing aids nowadays [2], but generally these components are developed and evaluated independently of each other. Although sophisticated algorithms for NR and DRC exist there is still an open question as to how these algorithm should be combined into an integrated approach, which has not received a lot of attention so far.

The interesting issue now is to analyse undesired effects when these algorithms operate together in an integrated scheme. The integration of hearing aid algorithms is a challenging task since each algorithm can counteract and limit the functionality of other algorithms. When NR and DRC are serially concatenated, undesired interaction effects typically occur, since each algorithm serves different purposes. For instance, DRC can counteract NR by amplifying the residual noise after NR, which consequently degrades the SNR and defeats the purpose of using NR. An integration of single-channel NR and DRC was proposed in [3] where a minimum mean square error and a maximum a posteriori optimal estimator are proposed that incorporate DRC in the derivation of the NR algorithm.

This research work was carried out at the ESAT laboratory of the Katholieke Universiteit Leuven, in the frame of the Marie-Curie Fellowship EST-SIGNAL program (http://est-signal.i3s.unice.fr) under contract No. MEST-CT-2005-021175, and the Concerted Research Action GOA-AMBioRICS. Ann Spriet is a postdoctoral researcher funded by F.W.O.-Vlaanderen. The scientific responsibility is assumed by its authors.

Another issue is the evaluation of such integrated schemes where the lack of an overall evaluation criterion indeed makes the integration more difficult. In the evaluation the crucial question will be as to which effects are most damaging to speech intelligibility, e.g. the amount of background noise or the audibility. In this work, objective quality measures are used to evaluate the integrated scheme, such as SNR and signal distortion measures. Subjective evaluation using hearing aid users is not included in this work.

In this paper we will focus on integrating NR and DRC and investigate if any undesired effect occurs when combining NR and DRC. The integrated scheme applies to both single-channel and multi-channel NR. The paper is organised as follows. In Section 2 the standard DRC scheme is introduced. Section 3 discusses the integration of NR and DRC. In Section 4 experimental results are presented. The work is summarized in Section 5.

2. DYNAMIC RANGE COMPRESSION

In this Section, we briefly introduce the basic concept behind DRC. The role of DRC is to map the wide dynamic range of a speech signal into the reduced dynamic range of a hearing impaired listener. The basic concept of DRC is to automatically adjust the gain based on the intensity level of the input signal. Segments with a high intensity level are attenuated while segments with a low intensity are amplified. This makes weak sounds audible while loud sounds are not becoming uncomfortably loud. DRC is typically defined by the following parameters:

- Compression threshold (CT).
- Compression ratio (CR).
- Attack and release time.
- Hearing aid gain G_{dB}.

CT is defined in dB and is the point where DRC becomes active. Below CT the gain is linear and above the CT, DRC is active i.e. the gain is reduced. CR determines the degree of compression. A CR of 2 (i.e. 2:1) means that for every 2 dB increase in the input signal, the output signal increases by 1 dB. The attack and release time is defined in milliseconds and specifies how fast the gain is changed according to changes in the input signal. The attack time is defined as the time it takes for the compressor to react to an increase in input signal level. The release time is the time taken for the compressor to react to a decrease in input level. The hearing aid gain G_{dB} is the maximum amount of amplification in dB which is specified by the linear part i.e. below CT of the DRC curve. A DRC curve with CR=2, CT=30dB and G_{dB}=0dB is shown in figure 1.

We define $P^{DRC}_{in,dB}$ and $P^{DRC}_{out,dB}$ as the input and output power in dB of the DRC, respectively, as

$$P^{DRC}_{in,dB} = 10\log_{10}(|P^{DRC}_{in}(\omega,k)|^2) \quad (1)$$

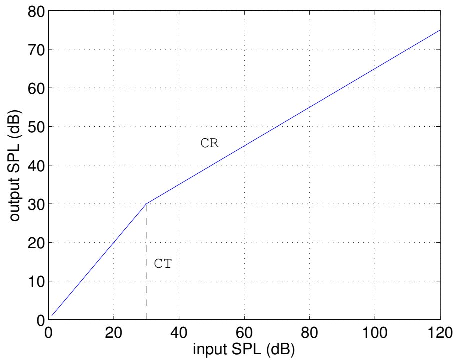


Figure 1: DRC curve (CR defines how the slope is changed and CT is the point at which the slope changes).

and

$$P^{DRC}_{out,dB} = 10\log_{10}(|P^{DRC}_{out}(\omega,k)|^2) \quad (2)$$

at time instant k and frequency $\omega=2\pi f$. The DRC curve is defined based on a linear curve and a compression curve in Eq. 3 and Eq. 4, respectively:

$$P_{linear,dB} = P^{DRC}_{in,dB} + G_{dB} \quad (3)$$

$$P_{compression,dB} = CT + \frac{1}{CR}\cdot(P^{DRC}_{in,dB} - CT) + G_{dB} \quad (4)$$

where $P^{DRC}_{in,dB}$ is the input power in dB. The DRC curve is given by Eq. 5

$$P^{DRC}_{out,dB} = \begin{cases} P_{linear,dB} & if\ P^{DRC}_{in,dB} < CT \\ P_{compression,dB} & if\ P^{DRC}_{in,dB} \geq CT \end{cases} \quad (5)$$

The DRC gain in dB is calculated as the output level minus the input level, i.e.

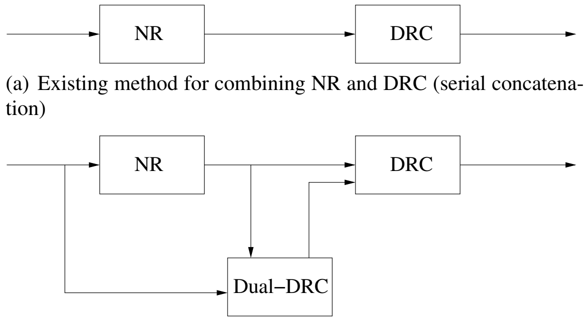
$$G_{DRC,dB} = P^{DRC}_{out,dB} - P^{DRC}_{in,dB} \quad (6)$$

The attack and release time are then applied to the DRC gain $G_{DRC,dB}$ using a first-order recursive averaging filter, before the DRC gain is applied to the input $P^{DRC}_{in,dB}$.

The power estimation in the DRC scheme used in this paper is based on individual FFT bins. If it is desired to have the DRC working on a specific number of frequency bands e.g. critical bands, this can be achieved by combining the FFT bins (e.g. by using individual FFT bins at low frequencies and combining FFT bins at higher frequencies) as is typically done in hearing aid applications [4].

3. INTEGRATION OF NOISE REDUCTION AND DYNAMIC RANGE COMPRESSION

The goal of a NR scheme is to improve speech intelligibility by reducing the effects of any noise source (e.g. multiple speakers, traffic etc). The NR can be a single-channel or a multi-channel algorithm. The DRC on the other hand amplifies signals based on the intensity level and makes no distinction between speech or noise. This means that noise already attenuated by the NR algorithm can be amplified by the DRC. This is an undesired effect leading to a degradation of the SNR, since the residual noise is amplified and the speech is attenuated. This is one of the crucial problems at hand when cascading NR and DRC. An existing method for cascading NR and DRC is a simple serial concatenation, depicted in figure 2(a).



(a) Existing method for combining NR and DRC (serial concatenation)

(b) Novel method for combining NR and DRC (Dual-DRC)

Figure 2: An existing and a novel approach for combining NR and DRC

3.1 Integration Concept

We introduce a dual-DRC concept to integrate NR and DRC. The goal is to control the DRC without counteracting the NR performance. The integrated approach is depicted in figure 2(b). The basic idea behind this, is to identify the amount of speech and noise present in a signal segment. The amount of speech and noise is estimated based on the input and the output of the NR (see also Section 3.3). The applied gain in the dual-DRC depends on the amount of speech and noise. This distinction between speech and noise makes it possible to apply DRC to make the speech audible without amplifying the residual noise. Note that a standard DRC scheme is based only on the input intensity level and does not make any distinction between speech and noise.

The basic concept is to apply a different DRC to the speech and the noise segments. We therefore introduce two DRC curves which are defined similarly as in Eq. 3-5.

- $P^s_{DRC,dB}$ - speech DRC (speech dominant case).
- $P^n_{DRC,dB}$ - noise DRC (noise dominant case).

The superscripts s and n are used to refer to speech and noise, respectively. In the case where speech is dominant we apply the speech DRC based on $P^s_{DRC,dB}$. If noise is dominant it is undesirable to amplify the noise and therefore a lower gain is applied i.e. we apply the noise DRC based on $P^n_{DRC,dB}$. In the case where speech and noise are present at the same time we define a weighted sum of the two DRC curves given in Eq. 7.

$$P_{dual,dB} = (1-\beta)\cdot P^n_{DRC,dB} + \beta\cdot P^s_{DRC,dB} \quad (7)$$

Here β is an estimate of the probability that speech or noise is present. If $\beta = 1$ there is no dual-DRC and the DRC is based on $P^s_{DRC,dB}$. As $\beta < 1$ dual-DRC is active with a trade off between $P^s_{DRC,dB}$ and $P^n_{DRC,dB}$. The dual-DRC gain is,

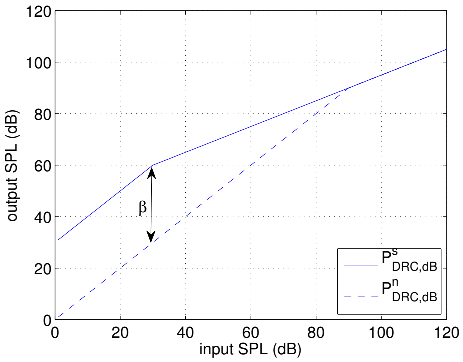
$$G_{dual-DRC,dB} = P_{dual,dB} - P^{DRC}_{in,dB} \quad (8)$$

which is applied to the output of the NR algorithm.

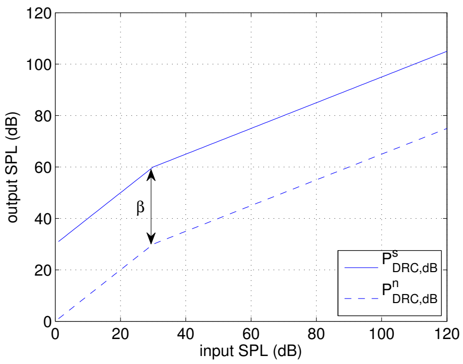
3.2 Speech and Noise DRC curves

The basic extension from a standard DRC scheme to the dual-DRC is that in the dual-DRC we use two DRC curves. The reason that we need two DRC curve is that we wish to trade off between speech and noise segments. $P^s_{DRC,dB}$ and $P^n_{DRC,dB}$ are bascially defined by chosing different values for the CT, CR and G_{dB} defined in Eq. 3 and Eq. 4. The dual-DRC curves are shown in figure 3. The tradeoff parameter β between the two DRC curves are defined in Eq. 7.

The rationale behind the noise DRC curve is that it has a lower gain compared to the speech DRC curve, as we indeed wish to apply



(a) Dual-DRC1



(b) Dual-DRC2

Figure 3: Two different approaches for dual-DRC.

a lower gain to the noise segments. As mentioned we want to apply DRC without compromising the NR but setting the noise DRC curve too low we might compromise the operation of the DRC. The goal of dual-DRC is thus to find a proper trade off between NR and DRC.

For dual-DRC, we introduce two different approaches which differ in the way $P^n_{DRC,dB}$ is defined. In the first approach dual-DRC1, the noise DRC curve is defined by a linear curve with a linear gain G^n_{dB} of 0dB. This is depicted in figure 3(a). The dashed line represents the noise DRC curve $P^n_{DRC,dB}$ and the solid line represents the speech DRC curve $P^s_{DRC,dB}$. As mentioned, the trade off between these two curves is defined with the parameter β. With dual-DRC1 the impact of β is reduced when the CR is increased, since beyond the intersection between $P^s_{DRC,dB}$ and $P^n_{DRC,dB}$ the dual-DRC concept is not active. Dual-DRC1 can have advantages when the noise is dominant or has a low intensity level, which is where dual-DRC1 has the largest trade off, or if a low CR is desired.

The second approach is dual-DRC2. Here $P^n_{DRC,dB}$ has the same CR and CT as $P^s_{DRC,dB}$ but is shifted towards lower gains i.e. $G^s_{dB} > G^n_{dB}$, see figure 3(b). In dual-DRC2 the range for β is kept constant when the CR is increased. If the gain G^n_{dB} is set closer to G^s_{dB} the integration is approaching a serial concatenation of NR and DRC.

3.3 Speech and Noise Detection

The parameter β that is used to trade off between speech and noise DRC is based on the power ratio between the output and the input of the NR algorithm, defined in Eq. 9,

$$\alpha_{NR}(\omega,k) = \frac{P^s_{out,NR}(\omega,k)+P^n_{out,NR}(\omega,k)}{P^s_{in,NR}(\omega,k)+P^n_{in,NR}(\omega,k)} \quad (9)$$

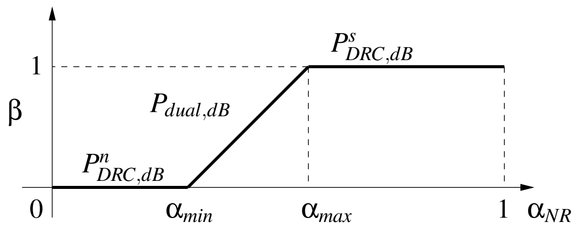


Figure 4: Tradeoff parameters for dual-DRC

where $P^s_{out,NR}(\omega,k)$ and $P^n_{out,NR}(\omega,k)$ are the speech and the noise components at the output of the NR and $P^s_{in,NR}(\omega,k)$ and $P^n_{in,NR}(\omega,k)$ are the speech and the noise components at the input of the NR.

α_{NR} is used to determine whether speech or noise is dominant. The NR algorithm preserves speech while attenuating the noise. Hence, if speech is dominant α_{NR} will approach one. If we assume that the noise is dominant the following statements can be written,

$$P^s_{out,NR}(\omega,k) \approx P^s_{in,NR}(\omega,k) \ll P^n_{in,NR}(\omega,k),\ P^n_{out,NR}(\omega,k) \quad (10)$$

so that,

$$\alpha_{NR}(\omega,k) \approx \frac{P^n_{out,NR}(\omega,k)}{P^n_{in,NR}(\omega,k)} \quad (11)$$

which is considerably smaller than one. This means that we can evaluate the NR performance by observing the noise power before and after NR. If there is noise at lower input SNR the NR algorithm is more active which means that α_{NR} is small. In this case DRC should not counteract the NR and amplify the residual noise. On the other hand, if there is more speech present at higher SNR the NR is less active and α_{NR} will be closer to one. In this case, it is desirable to apply DRC. Basically, we want to trade off between NR and DRC based on the speech and the noise contribution defined by α_{NR}.

The next step in the dual-DRC approach is to map α_{NR} into β based on a threshold function. The threshold function can also be considered as a soft Voice Activity Detector which is illustrated in figure 4.

The threshold function is controlled by parameters α_{min} and α_{max}, and is given in Eq. 12

$$\beta = \begin{cases} \beta = 1 & if\ \alpha_{NR} \geq \alpha_{max} \\ \beta = 0 & if\ \alpha_{NR} \leq \alpha_{min} \\ \beta = \frac{\alpha_{NR}-\alpha_{min}}{\alpha_{max}-\alpha_{min}} & otherwise \end{cases} \quad (12)$$

If α_{NR} is larger than α_{max} DRC according to $P^s_{DRC,dB}$ is applied, and if it is below α_{min} the DRC is based on $P^n_{DRC,dB}$. In between there is a trade off according to the power ratio between the output and input of the NR. When α_{min} and α_{max} are both set to zero the integration corresponds to a serial concatenation, where NR is performed before DRC.

To summarize, we first estimate α_{NR} which reflects the amount of speech and noise in the DRC input. α_{NR} is then used to estimate the tradeoff parameter β. Finally, a dual-DRC gain is computed based on a speech and a noise DRC curve. The objective here is that the speech is amplified/compressed while the noise is attenuated or at least amplified less than the speech.

4. EXPERIMENTAL RESULTS

In this Section, experimental results for the integrated approach for NR and DRC are presented.

4.1 Set-up and performance measures

The multi-microphone NR scheme used in this paper is the well-known Generalized Sidelobe Canceler (GSC) [5] consisting of a fixed spatial pre-processor and a multichannel adaptive noise canceler (ANC). The NR algorithm implemented here is based on a Frequency Domain Adaptive Filter (FDAF) [6] using a Weighted Overlap-Add (WOLA) analysis/synthesis structure. We have performed simulations with a 2-microphone behind-the-ear hearing aid. The speech and the noise sources are located at 0° and 120°, respectively. The speech signals consist of sentences from the HINT-database [7]. The noise signal consist of a multi-talker babble from Auditec [8], at 0 dB input SNR. The input level was set to 65 dB SPL at the hearing aid microphones. The FFT frame size was set to 256 (i.e., 16 ms), at a 16 KHz sampling rate, with 87.5% overlap between sucessive frames. Each frame is weighted with a Hanning window.

To assess the NR performance the intelligibility-weighted signal-to-noise ratio (SNR) [9] is used which is defined as

$$\Delta SNR_{intellig} = \sum_i I_i (SNR_{i,out} - SNR_{i,in}) \qquad (13)$$

where I_i is the band importance function defined in [10] and $\text{SNR}_{i,out}$ and $\text{SNR}_{i,in}$ represents the output SNR and the input SNR (in dB) of the ith band, respectively. For measuring the signal distortion a frequency-weighted log-spectral signal distortion (SD) is used defined as

$$SD = \frac{1}{K}\sum_{k=1}^{K}\sqrt{\int_{f_l}^{f_u} w_{ERB}(f)\left(10log_{10}\frac{P^s_{out,k}(f)}{P^s_{in,k}(f)}\right)^2 df} \qquad (14)$$

where K is the number of frames, $P^s_{out,k}(f)$ is the output power spectrum of the kth frame, $P^s_{in,k}(f)$ is the input power spectrum of the kth frame and f is the frequency index. The SD measure is calculated with a frequency-weighting factor $w_{ERB}(f)$ giving equal weight for each auditory critical band, as defined by the equivalent rectangular bandwidth (ERB) of the auditory filter [11].

The SD is measured between the compressed version of the clean speech reference signal (after the fixed spatial pre-processor) and the speech component of the output signal of the total scheme containing NR and DRC. The clean speech reference signal is compressed without the use of dual-DRC, where the DRC gain is computed based on the clean speech itself. It is assumed that noise dominant frames will have a larger effect on the SD measure in the dual-DRC scheme.

Simulations are performed for different CR and α_{max} for both dual-DRC1 and dual-DRC2 and compared to a serial concatenation of NR and DRC. The first simulation is to verify that a higher CR deteriorates the SNR for a serial concatenation compared to dual-DRC. In the second simulation, the CR is fixed and the effect of α_{max} on dual-DRC1 compared to dual-DRC2 is evaluated. Experiments with dual-DRC1 and dual-DRC2 are basically to show that the performance of the integrated scheme is affected by the way the DRC curves are defined.

For all simulations the attack and release time are fixed to at=5ms and rt=70ms. The hearing aid gain G^s_{dB} is set to 30dB and the CT^s=30dB.

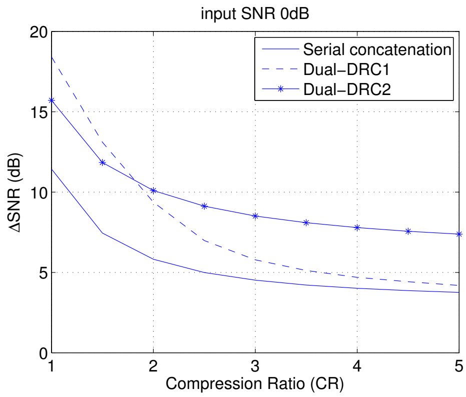
4.2 Effect of CR on SNR improvement and SD

In the first simulation the following setttings are used,

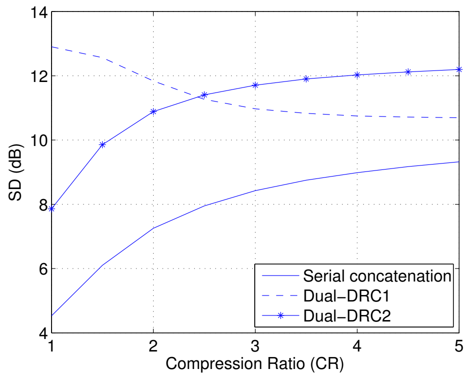
- Serial concatenation of NR and DRC (α_{min}=0, α_{max}=0)
- Dual-DRC1 (α_{min}=0.20,α_{max}=0.70, G^n_{dB}=0dB)
- Dual-DRC2 (α_{min}=0.20, α_{max}=0.70, G^n_{dB}=20dB)

CR is varied from one to five. Figure 5(a) shows the SNR improvement for the case where the CR is increased. The solid line represents the SNR improvement when the NR and DRC are serially concatenated, where a CR=1 corresponds to the SNR improvement for the NR without DRC. The dashed line and the (∗) marker line represents the SNR improvement for dual-DRC1 and dual-DRC2, respectively. Notice that with dual-DRC there is an SNR improvement since DRC is applied to the speech segments, but the residual noise is not amplified. At low CR the dual-DRC1 has a better SNR improvement but at higher CR dual-DRC2 is better. This happens for two reasons: first the impact of β is reduced in dual-DRC1 when CR is increased. Secondly, G^n_{dB} for dual-DRC1 is 0dB and for dual-DRC2 G^n_{dB} is set to 20dB, which means that β for dual-DRC1 has a greater impact at low CR. Setting G^n_{dB} to 0 dB in dual-DRC2 will result in a better SNR improvement for all values of CR, but this comes at the cost of less DRC. For dual-DRC2 the SNR improvement can be controlled by changing G^n_{dB}, if this value is closer to G^s_{dB} the integration is approaching a serial concatenation.

Figure 5(b) shows the SD for the three cases and here the SD is lowest when no dual-DRC is applied. For dual-DRC1 the SD is initially higher which corresponds to the better SNR improvement, and then the SD is decreasing as CR is increased, which corresponds to smaller SNR improvement. For dual-DRC2 the SD is higher when CR is increased which is also the case for the serial concatenation.



(a) SNR improvement for dual-DRC compared to a serial concatenation



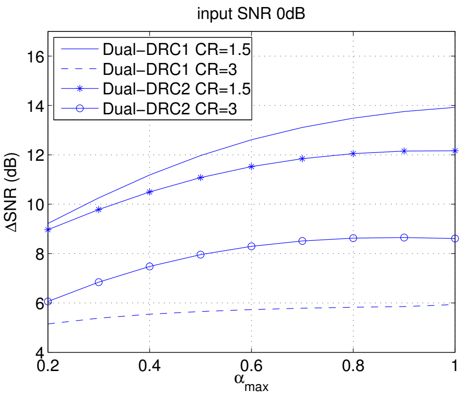
(b) SD for dual-DRC compared to a serial concatenation

Figure 5: Results for dual-DRC compared to a serial concatenation as a function of CR.

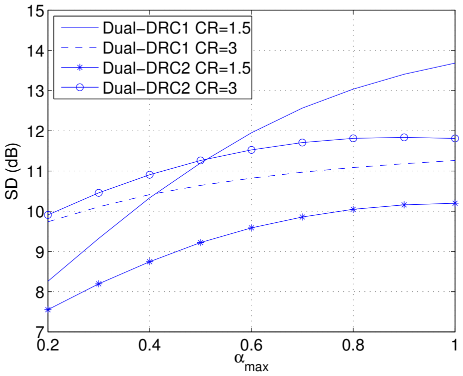
4.3 Effect of α_{max} on SNR improvement and SD

In the second simulation the following settings are used,

- Dual-DRC1 (CR=1.5, G^n_{dB}=0dB)
- Dual-DRC1 (CR=3, G^n_{dB}=0dB)
- Dual-DRC2 (CR=1.5, G^n_{dB}=20dB)
- Dual-DRC2 (CR=3, G^n_{dB}=20dB)



(a) SNR improvement for dual-DRC for CR=1.5 and 3



(b) SD for dual-DRC for CR=1.5 and 3

Figure 6: Results for dual-DRC1 and Dual-DRC2 as a function of α_{max}.

α_{min}=0.20 and α_{max} is varied from α_{min} to one. Figure 6(a) shows the SNR improvement for the case where α_{max} is increased. The solid line represents the dual-DRC1 for CR=1.5 and this approach outperforms dual-DRC2 for CR=1.5 represented by ($*$) marker line. When CR=3 the dual-DRC2 represented by the ($\circ$) marker line outperforms dual-DRC1 represented by the dashed line, and it is clear that for CR=3 dual-DRC1 can not improve the SNR much which is again due to the fact that the impact of β is reduced with higher CR.

Figure 6(b) shows the SD where the dashed and ($\circ$) marker line show almost similar distortion, but here it is worth noting that dual-DRC2 still has a significant SNR improvement compared to dual-DRC1 with CR=3. For the case with CR=1.5 the dual-DRC1 shows a significant SNR improvement, but this comes at the cost of higher SD (shown with the solid line which should be compared to the ($*$) marker line).

Overall, the SNR improvement comes at the cost of greater distortion. The SD basically represents how far away the dual-DRC is from the original DRC curve $P^s_{DRC,dB}$, which means that the SD is higher when the impact of β is larger resulting in more active dual-DRC. In other words, there is a trade off between SNR improvement and how close the dual-DRC is to the original DRC curve. The dual-DRC plays an important role when the noise is dominant i.e. the DRC is approaching $P^n_{DRC,dB}$. This is the case where the SD is highest, which means that especially the noise dominant segments contribute to the SD. We therefore assume that the speech dominant part is less distorted.

5. CONCLUSIONS

In this paper, we have presented a novel approach for integrating NR and DRC based on a dual-DRC concept. The dual-DRC uses a measure of the amount of speech and noise in the DRC input. This measure, defined by α_{NR}, is estimated based on the power ratio of the output and the input of the NR. α_{NR} is used to trade off between the amount of DRC that is applied without counteracting the NR performance. We introduce tradeoff parameters α_{min} and α_{max} to control the integration. When these parameters are set to zero, the integration corresponds to a serial concatenation of NR and DRC. Two dual-DRC approaches have been proposed, dual-DRC1 has a better performance at lower CR whereas dual-DRC2 shows more flexibility and works over a wide range of CR settings.

We have shown that increasing the CR leads to a reduced SNR improvement, when NR and DRC are serially concatenated. Dual-DRC resulted in an improvement in SNR compared to a serial concatenation when CR is increased. With the CR fixed we have shown that by increasing α_{max} it is possible to improve the SNR as a result of the dual-DRC becoming more active.

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