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Abstract

2 Hearing the desired sound in presence of noise is the biggest challenge faced by a hearing aid
3 user and hence is an area of great concern for the hearing aid designers as well as
4 professionals. This paper reviews the noise reduction strategies currently implemented in
5 hearing aids with an objective to precipitate the concerns and provide a clear direction of the
6 developments further required. Techniques used for noise reduction in hearing aids are
7 classified into two: - noise reduction using microphones and the noise reduction through
8 digital signal processing techniques. Noise reduction using two omni-directional
9 microphones, three omni-directional microphones and adaptive beam-forming techniques
10 were reviewed under the first category. In the second category, signal processing techniques
11 such as Wavelet Based noise reduction, Modulation Based Digital Noise Reduction,
12 Multichannel Wiener Filter, Blind Source Separation and Scene Analysis method were
13 reviewed. On comparison, adaptive beam-forming technique was found to be the most
14 efficient one among the microphone based noise reduction strategies. Wavelet Based noise
15 reduction and Scene Analysis method were found to be the better signal processing
16 techniques.

17

18 **Key words** :- Noise reduction, Wavelet Based, Modulation Based, Multichannel Wiener
19 Filter, Blind Source Separation, Scene Analysis.

Manuscript

Background

20 Number of people living in this world with disabling hearing loss are estimated to be
21 360 million (World Health Organization [WHO], 2017). Hearing aid is the most widely used

1 tool for rehabilitation of these individuals (Dell'Antonia, Ikino, & Filho, 2013). With
2 technological advancements in signal processing, overall satisfaction of a hearing aid user has
3 increased from 74% in 2008 to 81% in 2014 (Abrams & Kihm, 2015). However, lowest
4 satisfaction level of 50% was observed when the hearing aid users were trying to listen to a
5 conversation in the presence of noise (Abrams & Kihm, 2015). Thus the problem of noise
6 reduction in hearing aids continues to be a challenging task, amidst significant technological
7 advancements.

8 Persons with sensorineural hearing loss are found to have more difficulty in
9 comprehending speech in noisy situations, than people with normal hearing (Park, Moon, Jin,
10 Choi, Cho, & Hong, 2015). Background noise is thus a problem of great concern for people
11 with hearing impairment and thus noise reduction algorithms have significant role in hearing
12 aid signal processing (Ngo, 2011). Hearing aid designers have been trying hard to develop
13 techniques that present enhanced speech in comparison to noise to hearing aid users.

14 Levitt (2001) has identified three types of noise encountered by hearing aid users;
15 those can affect speech intelligibility - Random noise, other interfering voices and
16 reverberation. Random noise and interfering voices have the same spectrum as that of speech
17 which makes it difficult to differentiate between speech and noise.

18 Several techniques have been employed in hearing aids, over the years, to enhance the
19 speech signal in comparison to noise. The focus of all these techniques was to make the
20 speech intelligible to the user amidst background noise, and to enhance the quality of sound.
21 Levitt (2001) reviewed the noise reduction techniques implemented in hearing aids till 2001
22 and tried to give an outline of future developments required in this area. Chung (2004)
23 reviewed the microphone technologies and noise reduction algorithms while trying to figure
24 out the challenges in hearing aids. Bentler (2005) also reviewed the effectiveness of the noise

1 reduction techniques which were implemented in hearing aids of those times. Other than
2 these three reviews (Bentler, 2005; Chung, 2004; Levitt, 2001) no other reviews have been
3 reported in the literature on this topic. Several techniques were implemented in the following
4 years ³⁷ to improve the signal to noise ratio. Beck and Behrens (2016) opined that Digital Noise
5 Reduction techniques which are currently ³⁶ implemented in the state of the art hearing aids
6 improve listening in noise. The ⁹ purpose of this paper is to review the ³⁵ noise reduction
7 techniques currently implemented in hearing aids to precipitate the concerns and provide a
8 clear direction of the developments further required.

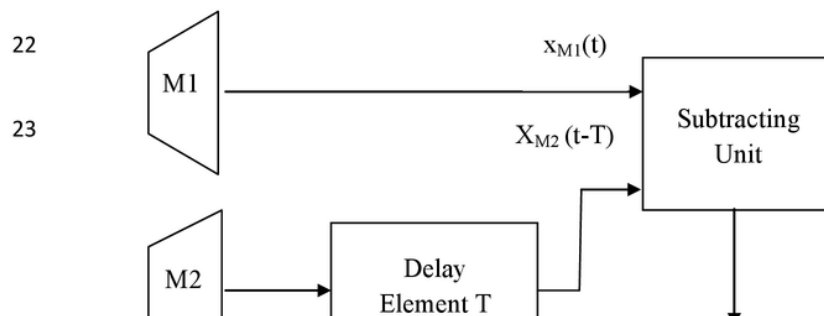
9 Noise Reduction Techniques

10 Ramirez, Jons, and Powers, (2013) while reviewing old and new strategies for
11 hearing in noise classified the techniques ²⁷ used for noise reduction in hearing aids into two: -
12 noise reduction using microphones and the noise reduction through digital signal processing
13 techniques.

14 1. Noise reduction techniques using multiple microphones

15 1.1 Using two Omnidirectional Microphones

16 One technique implemented in hearing aids to reduce noise is to achieve a direction-
17 dependent sensitivity by taking the difference signal of two omni-directional microphones
18 placed side by side, one directed to the front side of the user and the other one directed to the
19 back. Figure 1 illustrates the technique where the output signal of the microphone M2
20 directed to the back ²⁶ is delayed and subtracted from the output signal of the front microphone
21 M1.



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Figure 1: Noise reduction using two microphones

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1.2 Using three Omnidirectional Microphones

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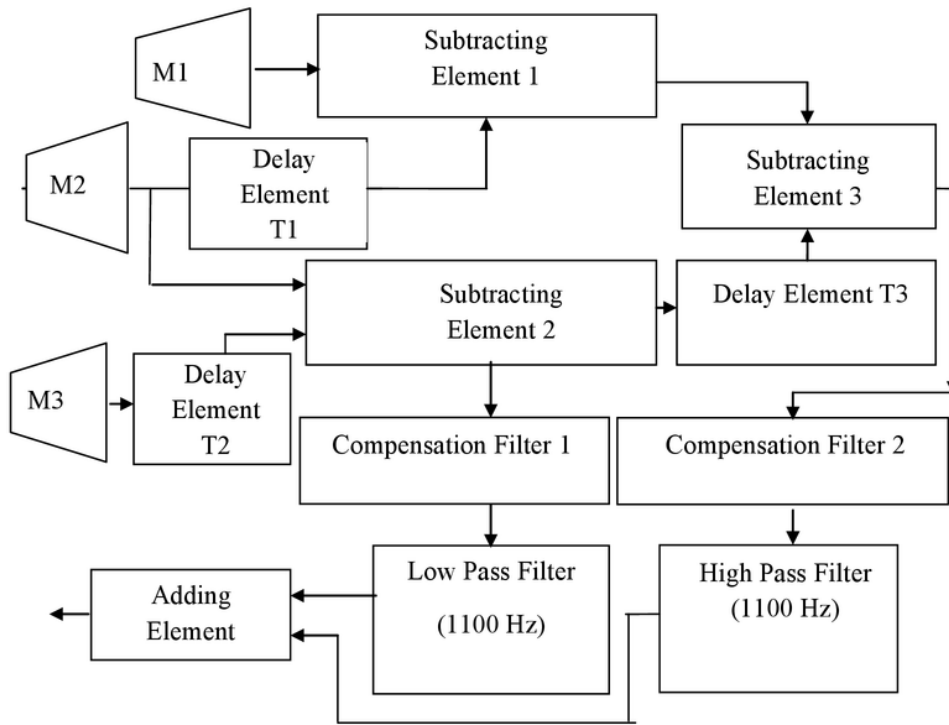
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Figure 2: Noise reduction using three microphones

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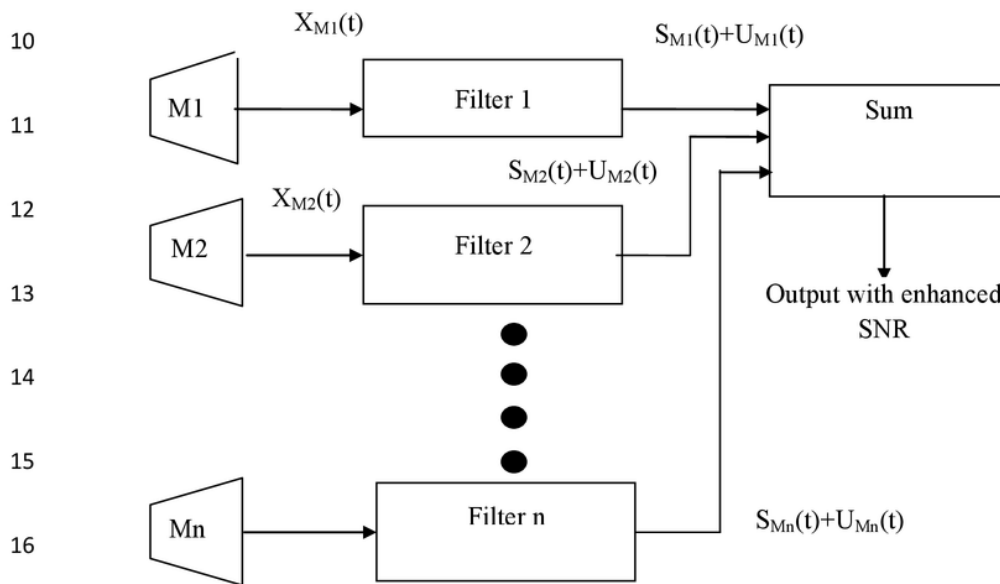
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Subtracting element 2 compares the delayed signal of Microphone M3 and the direct signal of Microphone M2, as shown in Figure 2. Output of this comparison is fed to Subtracting element 3 along with the difference signal of M1 and M2. Outputs of Subtracting element 2 and 3 are added to get the final output where the noises picked up by M1, M2 and M3 would have cancelled out. The SNR of a three microphone system is found to be much higher than that of a two microphone system.

1 **1.3 Adaptive Beam-forming**

2 Beam-forming is another technique for noise reduction where output signals of several
3 microphones are processed to reduce noise. This is achieved by creating a constructive
4 interference in a selected direction and destructive interference in non-selected directions.
5 Processing may be a fixed or adaptive one resulting in fixed or adaptive beam-forming
6 respectively. In fixed beam-forming time dependant transfer functions are used to process the
7 output signals of each microphone as shown in Figure 3. In adaptive beam-forming, at least
8 one of the transfer functions is continuously modified for better noise reduction in the given
9 environment (Kompis & Dillier, 2001).



17 *Figure 3: Noise reduction using fixed beam-forming*

18 Adaptive beamformers can adapt to different noise situations and they can offer better
19 noise reduction in most scenarios. The idea of beam-forming as ²⁵ Generalized Sidelobe
20 Canceller (GSC) algorithm was first implemented by Griffiths and Jim (1982). Their
21 algorithm ² transforms the constrained optimization problem into an equivalent but simpler
22 unconstrained optimization problem. Nishimura, Suzuki, Tsukui, and Asano (2004) proposed

1 a method of beam-forming which showed constraints to preserve perceptual cues at both ears
2 in wide band Minimum-Variance Distortion less Response (MVDR) MVDR beam-former.
3 First beam-former for binaural hearing aids was developed by Lotter and Vary (2006) called
4 super-directive beam-forming, which was based on a MVDR beam-former. But this method
5 was not continuously tracking the target speech. Rohdenburg, Hohmann, and Kollmeier
6 (2007) found that these methods showed good performance to reduce diffused or ambient
7 noise but low performance to reduce babble noise (Rohdenburg, Goetze, Hohmann,
8 Kammeyer, & Kollmeier, 2008). The approach of Rohdenburg et al. (2007) was not referred
9 to wide band applications. Although it was advantageous, the computational cost of this
10 technique was high.

11 **2. Noise reduction using digital signal processing strategies**

12 **2.1 Wavelet Based noise reduction**

13 The speech signal consists of many frequency components; generally it is complex in nature
14 to understand and to process. For analyzing them, we go for mathematical approach. A
15 complicated signal can be broken down into simple waves using Fourier Transforms (FT).
16 Fourier Transform analyses only stationary signals, but speech is not a stationary signal. Thus
17 for analyzing non-stationary signals Wavelet Transform (WT) is advantageous because they
18 are fast. Another advantage is that wavelets are represented both in time and frequency
19 domain whereas Fourier transform is a frequency domain representation.

20 Wavelet-based ³⁴Maximum likelihood estimation of signals in autoregressive noise method by
21 the approach of Kay and Nagesha (1994) is used ³³for reducing correlated noise in noisy
22 speech signals. Many of the Digital Signal Processing methods have the difficulty in
23 distinguishing between noise and consonants so that most of the consonants are removed
24 along with noise, which reduces the intelligibility of speech. Wavelet-based noise reduction

1 and compression approach by Trenas, Rutledge, and Whitmal (1999) gives solution to this
2 issue. It is flexible as the parameters can be modified to fit the hearing loss of the individual
3 and also the noise characteristics. The response time is less and the accuracy is good.
4 Therefore in general, Wavelet-based noise reduction approach is highly reliable compared to
5 Fourier Transform technique.

6 ³² 2.2 Modulation Based Digital Noise Reduction (MB-DNR)

7 This technique ² is widely implemented in hearing aid signal processing to reduce noise. The
8 envelope of the speech signal is amplitude modulated as a result of the vocal tract movements
9 associated with speaking. Modulation rates of 4 Hz to 16 Hz are important for sentence
10 recognition (Drullman, Festen, & Plomp, 1994). Environmental noises are either
11 unmodulated or have a rate of modulation beyond the 4 Hz to 16 Hz range. The signals which
12 are steady are nothing but noise and signal which is modulated more are speech like (Van,
13 Festen, & Plomp, 1991). The detection and analysis unit recognizes the presence of noise if
14 rate of modulation is beyond the 4 Hz to 16Hz range. In MB-DNR, signal is passed through
15 different frequency filters. During filtering, if any output is less modulated that contains more
16 noise component, the gain of that channel is compressed and channel with fair modulation is
17 passed. This may reduce speech of particular frequency speech component but increases the
18 overall Speech to Noise Ratio (SNR).

19

20 2.3 Multichannel Wiener Filter (MWF)

21 This is basically a modified form of wiener filter introduced by Doclo and Moonen (2002) to
22 enhance the desired signal reaching from any arbitrary direction. MWF is widely used in
23 binaural hearing aid technology for noise reduction. This method gives ¹³ a Minimum Mean

1 Square Error (MMSE) estimate of the speech component in the signal received from one of
2 the microphones (Spriet, Moonen, & Wouters, 2001). In binaural hearing aids, the noise
3 reduction can be done by MWF and MWF with partial noise estimate (MWF-N). These
4 methods preserve Inter-aural Time Difference (ITD) and Inter-aural Level Difference (ILD)
5 which enhances the S/N ratio. A famous technique in MWF is Speech Distortion Weighted
6 MWF (SDW-MWF) which can adapt to various noise scenarios. SDW-MWF does not
7 require any assumptions about the location of target speech and characteristics of
8 microphone, unlike the Generalized Sidelobe Canceller (GSC). Doclo, Spriet, Wouters, and
9 Moonen (2007) proposed implementation of the SDW-MWF in frequency domain. It is
10 advantageous that every frequency bin can be processed separately. When compared with
11 Adaptive Directional Microphones (ADM), MWF offers good speech localization (Bogaert,
12 Doclo, Wouters, & Moonen, 2008). Perpetual tests on SDW-MWF and MWF-N showed that
13 transmitting only one channel to the contralateral hearing aid is enough to ensure good noise
14 removal (Bogaert, Doclo, Wouters, and Moonen, 2009) and the noise localization cue is lost
15 in SDW-MWF and preserved in MWF-N. Li, Sakamoto, Hongo, Akagi, and Suzuki (2011)
16 proposed a two-stage processing and Wiener filter. The two stages were designed for
17 equalization and cancellation of the target signal to get a noise estimate, this noise estimate
18 gives the parameters of the Wiener filter. This technique dominates other spectral subtraction
19 and beam-forming approaches and retains the localization cues of the target signal.

20

21 ²⁴ 2.4 Blind Source Separation (BSS)

22 Blind Source Separation method takes the combination of audio signals, filters them and
23 gives separate output for those signals which are present in that combination. It uses multiple
24 microphones for its operation and assumes that there would be at least as many microphones

1 as the number of sources. One source must be Gaussian and then Independent Component
2 Analysis (ICA) ³¹ can be used to retrieve the original source signal (Buchner, Aichner, &
3 Kellermann, 2005). The output of the BSS will not have the localization cues. (Kocinski,
4 2008; Parikh, & Anderson, 2011). BSS algorithm with Wiener filter or Adaptive filters used
5 for retaining localization cues. BSS combined with adaptive interference cancellation was
6 introduced by Aichner, Buchner, Zourub, and Kellermann (2007) for preserving spatial
7 information. This scheme is powerful if it was for a known scenario. For the unknown
8 scenario, ⁶ directional BSS and Wiener filtering were combined. Advantage of Directional BSS
9 over BSS is that a ⁶ good interference estimate can be achieved quickly even with a small
10 number of microphones (Reindl, Zheng, & Kellermann, 2010).

11 The new BSS algorithm developed by Hild, Erdogmus, and Principe (2002) has been
12 reported to be very efficient. Computationally this method is more complex than the rest.
13 Techniques for separating out the acoustic signal in BSS can be divided into two classes, i.e.
14 frequency and time-domain techniques. The time-domain BSS technique (Buchner, Aichner,
15 & Kellermann, 2005) is applied for sound source localization. The BSS-based method
16 outperforms other known techniques, most notably in high reverberant environments.
17 Interestingly, the approach remains applicable ²³ in the underdetermined case, where there are
18 more sources than microphones. As computational power of hearing aids continue to
19 increase, BSS techniques will become applicable for noise reduction or sound localization in
20 all future hearing aids (Cornelis, 2011).

21 **2.5 Scene Analysis method**

22 Scene analysis approaches are characterized by the use of measurements taken from the
23 input signal to compute a set of frequency responses that are used to filter out the noise.
24 Individuals with hearing impairment finds difficulty in separating out speech content of

1 interest amidst interfering ⁵ sounds, background noise and reverberation. Speech segregation
2 algorithms seek to improve the intelligibility of a desired speech source by attenuating
3 unwanted sounds (Wang & Brown, 2006). It is necessary for human listeners to differentiate
4 and isolate the desired speech through auditory scene analysis. But, it is not enough to
5 recognize the target signal coming from front for scene analysis. A Direction of Arrival
6 (DoA) strategy was proposed by Chisaki, Matsuo, Hagiwara, Nakashima, and Usagawa
7 (2007). In Chisaki's method DoA of target is estimated by the ¹⁷ Interaural Time Difference
8 (ITD) and the Interaural Level Difference (ILD) of the input signals. ³⁰ The head related
9 transfer function (HRTF) corresponding to the estimated DoA is used as frequency response
10 to perform the filtering. But there is ambiguity in ILD and DoA relation, this leads to
11 ambiguous DoA estimation. Li, Akagi, and Suzuki (2008) proposed a method which
12 estimates the noise signal by left and right microphones, and then noise ²² is subtracted from the
13 input signal to get the enhanced speech signal for both ears. But this needs target signal to be
14 in phase for both ears. Later Kamkar and Bouchard (2009) proposed a method which uses
15 coherence function to estimate a frequency response to cancel interference and noise. Shao,
16 Srinivasan, Jin and Wang (2010) ⁸ proposed a method based on Computational Auditory Scene
17 Analysis (CASA) system for differentiating desired speech from noise. They estimated input
18 speech signal, in two stages. In the first stage, ¹ voiced portions of individual sources in each
19 time frame were separated out using harmonicity. ³ Onset/offset analysis was used to segment
20 the unvoiced regions. Speaker characteristics were used to group the Time-Frequency units
21 across time frames, in the second stage. The resulting masks were used for automatic speech
22 recognition. The proposed system was found to be ¹ capable of separating and identifying the
23 contents of a target utterance in the presence of another speech utterance or speech-shaped
24 noise.

25 Discussion

Table 1: Performance comparison of multiple microphone noise reduction techniques

Parameter	Two microphones	Three microphones	Adaptive Beam-forming
SNR for subjects with hearing loss	a. 3 dB improvement in moving noise b. 4 dB improvement in stationary noise (Bentler et al., 2004)	a. 4 dB improvement in moving noise b. 5 dB improvement in stationary noise (Bentler et al., 2004)	Two stage adaptive beam-former always performed with better SNR than the adaptive directional microphone (Maj, Royackers, Wouters, & Moonen, 2005)
Average Directivity Index	4.5 to 6.0 dB (Bentler et al., 2004)	6.5 to 7.8 dB (Bentler et al., 2004)	Significantly higher sequential Directivity Index (sDI) values compared to dual microphone system (Herbig & Froehlich, 2015)

In Table 1, all the three multiple microphone techniques which are currently used for noise reduction in hearing aids are compared on the basis of two parameters - SNR and Directivity Index. As evident from the Table 1, adaptive beam-forming technique is found to be superior as it showed better SNR and higher sDI. The limitation of adaptive beam-forming is that, for its efficient performance, the noise sources need to be restricted in number as well as the noise sources need to be directional. A system which has M microphones can control only noises from M-1 noise sources and if the number exceeds M, then the efficiency of noise reduction reduces. Generally, as only one sound source will be dominating in a band of frequency, this drawback will not have much impact in practical situations. Another major deficit of adaptive beam-forming arises out of the reverberation in the environment. Greenberg and Zurek (1992) showed empirically that with increase in reverberation time of

the listening environment, the efficiency of adaptive beam-forming reduces. This issue needs to be addressed by increasing the temporal length of the adaptive filter.

Table 2: Digital noise reduction techniques - Comparison of merits and limitations

Wavelet Based noise reduction	Modulation Based Digital Noise Reduction	Multichannel Wiener Filter	Blind Source Separation	Scene Analysis method
2 MERITS				
Flexible as the parameters can be modified to fit the hearing loss of the individual and also the noise characteristics. The response time is less and the accuracy is good. (Trenas, Rutledge, & Whitmal, 1999)	Several Hearing Impaired listeners ²¹ proved intelligibility from scores near zero to values above 70%. (Healy, Yoho, Wang, & Wang, 2013)	Binaural cues of both the speech and the residual noise should also be better preserved. (Cornelis, 2011)	Optimization of outcome leading to better SNR (Shanmugapriya, & Chandra, 2014)	¹ Capable of separating and identifying the contents of a target utterance in the presence of another speech utterance or speech-shaped noise. (Shao, Srinivasan, Jin, & Wang, 2008)
LIMITATIONS				
	Interfering signals with speech-like modulation properties could not be reduced.	Assumes that the speech and noise are uncorrelated	Highly complicated to solve and computationally more expensive. ¹¹	

Table 2 compares the merits and limitations of various digital signal processing strategies employed for noise control in hearing aids. Wavelet based noise reduction and scene analysis methods do not have any major limitations. The parameters of Wavelet based noise reduction can be modified to suit the hearing loss of the user. Moreover the technique is accurate and

the system works very fast. Scene analysis method is capable of separating out a desired speech by isolating the undesired speech, through which a major limitation of the modulation based strategy is overcome. These two methods are easy to implement and are not computationally expensive and thus figure out to be the two best strategies. Scene analysis method assumes that the target signal is in-phase at both sides, which is not true for all frequencies and direction of arrivals due to the head shadow effect. Wavelet based method doesn't have any such limitation and hence is considered to be a better option compared to the scene analysis method.

Conclusion

Among the three multiple microphone techniques which are currently used ¹⁰ for noise reduction in hearing aids, adaptive beam-forming technique is found to be superior as it showed better SNR and higher sDI. The limitation of adaptive beam-forming is that, for its efficient performance, the noise sources need to be restricted in number as well as the noise sources need to be directional. Also, with increase in ¹² reverberation time of the listening environment, the efficiency of adaptive beam-forming reduces. This issue needs to be addressed by the designers.

Among the various digital ¹¹ signal processing strategies employed for noise control in hearing aids, Wavelet based noise reduction and scene analysis methods were found to be the two best strategies. Wavelet based noise reduction is considered to be a better option compared to the scene analysis method.

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