Present folder contains implementations of ideal channel selection (ICS) algorithms (also known as binary mask algorithms) described in Chapter 13.

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Usage: ics(noisefile, clfile, outfile, nsnr, thrd)

- % noisefile name of masker file
- % clfile name of clean stimulus file
- % outfile name of output file
- % nsnr is the overall input SNR (in dB) for noisy file
- % thrd is the SNR threshold (in dB)

Examples:

In MATLAB, type:

>> ics('babble1.wav','S_01_01.wav','out.wav',-10,-5)

For the above example, input SNR=-10 dB and SNR threshold=-5 dB. The noisy (mixture) file is contained in 'S_01_01-noisy.wav' file.

Segregated file is in 'out.wav'

The wav files can be played via a Media Player, Cool Edit, Audition, etc.

It can also be viewed and played through our toolbox 'Colea': http://www.utdallas.edu/~loizou/speech/software.htm

Reference:

Li, N. and Loizou, P. (2008). "Factors influencing intelligibility of ideal binary-masked speech: Implications for noise reduction," Journal of Acoustical Society of America, 123 (3), 1673-1682

Demo of separating two talkers speaking simultaneously:

Usage: ics_competing_talker(filename, clfile, t_outfile,
m_outfile,thrd)

- % filename mixture filename
- % clfile clean target filename
- % t_outfile output file: Target talker

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% m_outfile - output file: Competing talker
% thrd - SNR threshold in dB
Example:
>> ics_competing_talker('talker_mixture.wav','S_01_
10.wav','target.wav','masker.wav',-5)
In 'talker_mixture.wav' mixture file, the competing talker was
added at SNR=-5 dB
 (target and competing talkers were same for this example).
Files 'target.wav' and 'masker.wav' contain processed sentences
of the segregated target and competing-talker talkers
respectively. The SNR threshold was set to -5 dB.
%----- SNResi rule (constraint rule) -------
Usage: ics_constr_rule(filename, clfile, outfile, GAIN)
% filename - noisy speech filename (mixture)
% clfile - clean speech filename
% outilfe - name of output file
% GAIN='Wiener'; 'MMSE', 'logMMSE', 'MMSE-SPU'; 'pMMSE';
'SpecSub'
Example:
>> ics_constr_rule('S_01_02-babble_m10dB.wav', 'S_01_
02.wav','out_constr.wav','Wiener')
Target was corrupted with babble at -10 dB SNR. The Wiener gain
function was used.
Other possible gain functions: 'MMSE', 'logMMSE', 'MMSE-SPU';
'pMMSE'; 'SpecSub'
Example with competing-talker:
>> ics_constr_rule('talker_mixture.wav','S_01_
10.wav','out.wav','Wiener')
Another example in babble at input SNR=-5 dB
>> ics_constr_rule('S_02_02-babble_m5dB.wav', 'S_02_
02.wav','out_constr.wav','Wiener')
References
Kim, G. and Loizou, P. (2011). "Gain-induced speech distortions
and the absence of intelligibility benefit with existing noise-
reduction algorithms, "J. Acoust. Soc. Am. 130(3), 1581-1596.
```

Loizou, P. and Kim, G. (2011). "Reasons why Current Speech-Enhancement Algorithms do not Improve Speech Intelligibility and Suggested Solutions," IEEE Trans. Audio, Speech, Language Processing, 19(1), 47-56.

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Usage: ics_reverb(reverbfile, clfile, outfile, thrd)
% reverbfile - name of file containing reverberated stimulus
% clfile - name of clean sentence file
% outfile - name of output (processed) file
% thrd is the threshold (in dB) for signal-to-reverberant ratio
criterion
Example:
>> ics_reverb('rev800_2.wav','clean_2.wav','outrev.wav',-8)
File was corrupted with RT60=0.8 sec reverberation.
The signal-to-reverberant ratio (SRR) threshold was set to -8 dB.
Reference
Kokkinakis, K., Hazrati, O. and Loizou, P. (2011). "A channel-
selection criterion for suppressing reverberation in cochlear
implants, "Journal of the Acoustical Society of America, 129(5),
3221-3232.
%----- masker-based rule -----
Usage: ics_masker_rule(filename, clfile, outfile)
% filename - noisy speech filename (mixture)
% clfile - clean speech filename
% outilfe - name of output file
Example file corrupted at -10 dB SNR with babble:
>> ics_masker_rule('S_01_02-babble_m10dB.wav','S_01_
02.wav','out masker rule.wav')
Reference:
```

Kim, G. and Loizou, P. (2010). "A new binary mask based on noise constraints for improved speech intelligibility," Proc. INTERSPEECH, Makuhari, Japan, pp. 1632-1635.

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Publications (from our lab) assuming ideal conditions (see

Chapter 13):

Hu, Y. and Loizou, P. (2008). "A new sound coding strategy for suppressing noise in cochlear implants," Journal of Acoustical Society of America, 124(1), 498-509.

- Kim, G. and Loizou, P. (2011). "Gain-induced speech distortions and the absence of intelligibility benefit with existing noise-reduction algorithms," J. Acoust. Soc. Am. 130(3), 1581-1596.
- Kim, G. and Loizou, P. (2010). "A new binary mask based on noise constraints for improved speech intelligibility," Proc. INTERSPEECH, Makuhari, Japan, pp. 1632-1635.
- Kokkinakis, K., Hazrati, O. and Loizou, P. (2011). "A channel-selection criterion for suppressing reverberation in cochlear implants," Journal of the Acoustical Society of America, 129(5), 3221-3232.
- Li, N. and Loizou, P. (2008). "Factors influencing intelligibility of ideal binary-masked speech: Implications for noise reduction," Journal of Acoustical Society of America, 123 (3), 1673-1682
- Li, N. and Loizou, P. (2008). "Effect of spectral resolution on the intelligibility of ideal binary masked speech," Journal of Acoustical Society of America, 123(4), EL59- EL64
- Loizou, P. and Kim, G. (2011). "Reasons why Current Speech-Enhancement Algorithms do not Improve Speech Intelligibility and Suggested Solutions," IEEE Trans. Audio, Speech, Language Processing, 19(1), 47-56.

Publications (from our lab) assuming realistic conditions (see Chapter 13):

- Hu, Y. and Loizou, P. (2008). "Techniques for estimating the ideal binary mask,&" Proc. of 11th International Workshop on Acoustic Echo and Noise Control, September 14th-17th, Seattle, Washington.
- Hu, Y. and Loizou, P. (2010). "Environment-specific noise suppression for improved speech intelligibility by cochlear implant users," Journal of the Acoustical Society of America, 127 (6), 3689-3695.
- Kim, G., Lu, Y., Hu, Y. and Loizou, P. (2009). "An algorithm that improves speech intelligibility in noise for normal-hearing

listeners," Journal of the Acoustical Society of America, 126(3), 1486-1494

Kim, G. and Loizou, P. (2010). "Improving Speech Intelligibility in Noise Using Environment-Optimized Algorithms," IEEE Trans. Audio, Speech, Language Processing, 18(8), 2080-2090.

Kim, G. and Loizou, P. (2010. "Improving Speech Intelligibility in Noise Using a Binary Mask that is Based on Magnitude Spectrum Constraints, " IEEE Signal Processing Letters, 17(2), 1010-1013

Kim, G. and Loizou, P. (2009). "A data-driven approach for estimating the time-frequency binary mask," Proc. Interspeech, Brighton, UK, Sept 6-9, 2009