Signal Processing Techniques To Reduce the Effects of Impaired Frequency Resolution

By RICHARD S. TYLER

The ability to separate different frequency components of speech—i.e., frequency resolution—has been found to be impaired in most listeners with sensorineural hearing loss. This poor frequency resolution probably contributes to poor speech recognition in the hearing impaired in three ways. First, the spectral peaks of speech are less clearly defined. Second, there might be masking of one formant on another. Third, the ability to separate speech from noise will be impaired. This article discusses a variety of procedures that have been used to circumvent impaired frequency resolution. These procedures include 1) reducing the low-frequency gain of a hearing aid, 2) automatically decreasing the low-frequency gain when noise is present, 3) automatically decreasing the gain in any frequency region that contains noise, 4) enhancing spectral peaks, 5) dichotic separation of spectral information, and 6) feature extraction. While none of the procedures have been universally successful, there is some hope that individually fitted algorithms may provide some benefit for some hearing-impaired listeners.

Perhaps one of the most important aspects of auditory perception is the ability to separate sounds according to their frequency. In normal hearing, the basilar membrane and the traveling wave play a critical role in determining frequency selectivity. It is well established that cochlear hearing loss often results in impaired frequency resolution.[1,2] Several investigators have suggested a correspondence between poor word recognition and reduced frequency resolution.[3-6]

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Although they are not the focus of this article, distortions other than impaired frequency resolution also accompany sensorineural hearing loss. For example, abnormalities of the intensity and temporal coding mechanisms also influence speech perception.[7-9] These other distortions will likely require different signal-processing strategies.

Digital processing of speech has opened up new opportunities for modifying the incoming signal. However, the appropriate algorithms for enhancing speech perception are not clear. (Actually, whether the enhancement strategies involve digital or analog processing matters less than what the strategies are.) Digital processing, however, does open up many possibilities in terms of applying more sophisticated algorithms.

One group of strategies attempts to overcome the reduced frequency resolution that accompanies sensorineural hearing loss. This article will review several of these strategies. The review is intended to provide a general understanding of the rationale behind certain types of signal-processing techniques. First I shall discuss normal and impaired frequency resolution and how it is measured.

THE ROLE OF FREQUENCY RESOLUTION

Figure 1-left shows a schematic representation of the spectrum of a speech signal (like a vowel) obtained at one point in time (top). The middle panel represents the auditory system as having a number of bandpass filters. These auditory filters represent the frequency-resolving characteristics. The output of each of these filters sends information along different nerve fibers. The fine tuning or selectivity of these filters allows us to determine where, in the frequency domain, any information is present. This ability is fundamental to understanding speech. The bottom panel shows that the output of the normal auditory filters provide a good reconstruction of the original acoustic spectrum.

Reduced Frequency Resolution

Figure 1-right shows the effects of reduced frequency resolution. The speech (top) is analyzed by the wide auditory filters (middle), found in the hearing-impaired person. The result of coding speech with these broad auditory filters is a less clearly defined spectrum (bottom). If the filters are sufficiently broad,

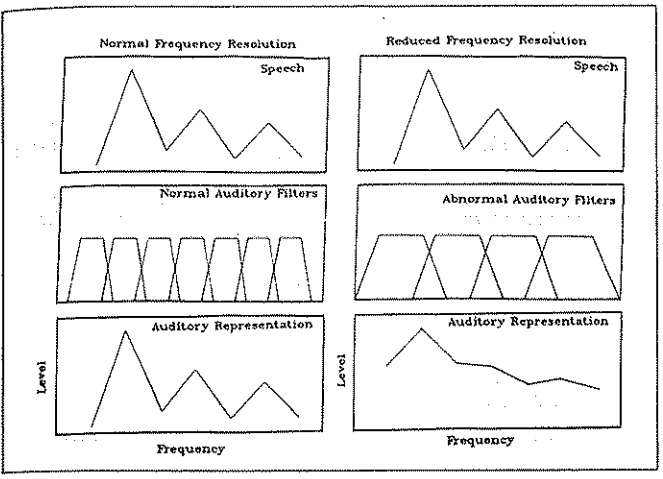


Figure 1. *Left*: Schematic representation of the speech spectrum (top), normal auditory filters (middle) and the auditory representation of the speech (bottom) for a normal ear. *Right*: Schematic representation of the speech spectrum (top), abnormal auditory filters (middle), and the auditory representation of the speech (bottom) for an impaired ear.

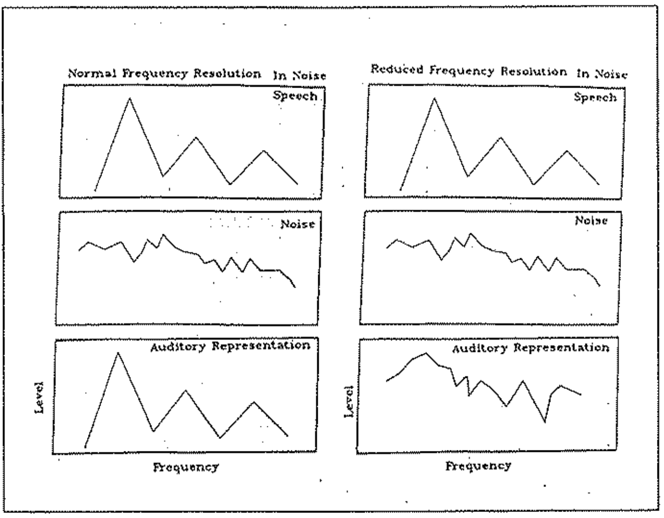


Figure 2. *Left*: Schematic showing that speech (top) and noise (middle) can be separated by normal frequency resolution (bottom). *Right*: Schematic showing that speech (top) and noise (middle) cannot be separated by abnormal frequency resolution (bottom).

then the representation of one formant can 'mask' the representation of another. This will occur when two formants are located within the same filter. Speech perception will be limited in these patients with poor frequency resolution.

Perception in Noise

Frequency resolution is probably fundamental to understanding speech in noise. Figure 2-left shows the speech (top) being presented in the presence of noise (middle). Normal frequency resolution helps us to separate the signal from the noise, because at any point in time, speech will be predominant in some filters, and noise in other filters. The particular filters containing the predominant speech and noise will change as a function of time. The normal auditory system can separate the output of the filters with speech from those with noise, and then fuse the speech signal together (bottom).

However, with reduced frequency resolution, as in Figure 2-right, the larger auditory filters will be unable to separate the speech and noise as effectively. The auditory representation of the speech will be degraded by the noise.

MEASURING FREQUENCY RESOLUTION

There are many ways of measuring frequency resolution (see Moore[10] for a review). These include measuring signal thresholds in the presence of noise, obtaining psychoacoustical tuning curves,[11] and measuring signal thresholds in the presence of a wideband noise masker with a spectral notch "cut out" in the middle.[12]

A variation of this latter method for clinical application was proposed by Tyler and Tye-Murray and is illustrated in Figure 3-left.[13] The signal frequency is chosen in the region of interest. The signal is then presented at a fixed level, clearly audible (top). The notched noise, which can be thought of as a low- and high-frequency band, is also fixed in level. This allows the tester to ensure that the signal and noise are above threshold but are not uncomfortably loud. To measure frequency resolution, the notchwidth between the two noises is narrowed (varied adaptively) until the signal is just at threshold (bottom). The notchwidth at threshold is a measure of the bandwidth of the auditory filter.

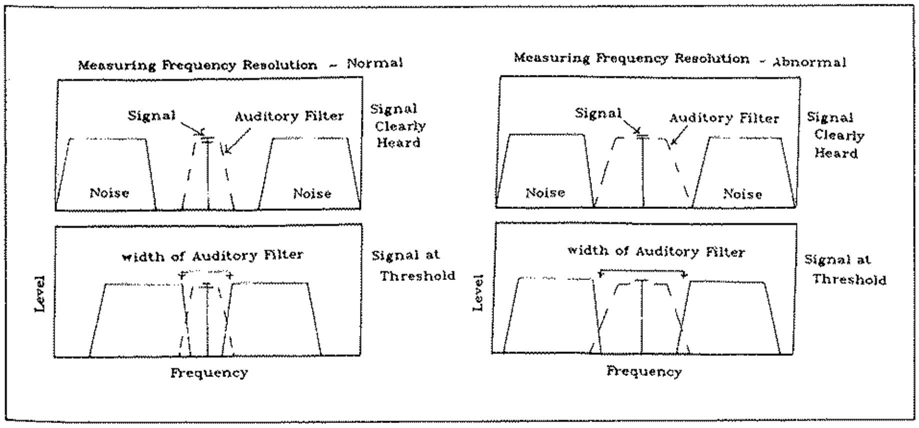


Figure 3. *Left*: Schematic representation of a notchwidth narrowing experiment to measure frequency resolution in normal ears. *Right*: Schematic representation of the notchwidth narrowing experiment to measure frequency resolution in the impaired ear.

In a hearing-impaired person, as shown in Figure 3-right, the auditory filter is typically broader. In measuring frequency resolution in that person, the signal and masker level are chosen to be above threshold and not too loud (top). The notchwidth is narrowed until the signal is at threshold (bottom). Compared to the normal ear, the notchwidth at threshold is typically greater.

It is possible to measure an asymmetry in the filter shape. This is done by holding one noise band constant and varying the other.[12,14] Figure 4 (top) shows the shape of the normal auditory filter when the low- and high-frequency edges of the filter are measured separately. Many hearing-impaired individuals demonstrate a pronounced upward spread of masking, and their filters are represented in the second panel. However, it also is important to appreciate that many other hearing-impaired individuals do *not* show a pronounced upward spread of masking. Some show just the opposite: a pronounced *downward* spread of masking (third panel). Finally, some individuals with sensorineural hearing loss have normal frequency resolution (bottom panel).

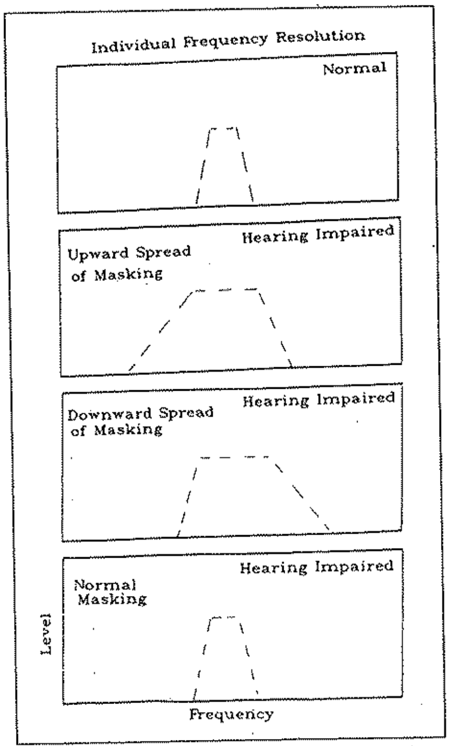


Figure 4. An example of auditory filter shapes in normal (top) and different impaired ears showing pronounced upward (second), downward (third), and normal (bottom) masking.

ATTEMPTS TO COMPENSATE FOR POOR FREQUENCY RESOLUTION

I shall now review a series of attempts to reduce the effects of poor frequency resolution. I do not intend to be exhaustive in the literature survey, only to describe some of the techniques, their rationale, and the general results reported when they have been tested on hearing-impaired persons.

Set the Hearing Aid with Reduced Low-Frequency Gain

There are two reasons why it might be beneficial to reduce the low-frequency gain of a hearing aid. First, this could reduce the upward spread of masking produced by low-frequency noise; and second, it could reduce the upward spread of masking produced by lower frequency formants. In Figure 5-left, we see the speech (top) presented in a low-frequency background noise (second panel). Abnormally wide auditory filters in the mid-frequency regions can respond to this noise. If the low-frequency gain of the hearing aid is reduced, there will be less chance of the noise producing this upward masking.

Punch and Beck tested this premise in 12 subjects with gradually sloping sensorineural hearing loss.[15] In contrast to the previous hypothesis, they noted that the recognition of words in noise was not improved when the amount of low-frequency gain was reduced. Gordon-Salant also reported that 10 individuals with flat hearing loss showed no difference in performance among various amounts of low-frequency amplification in noise.[16]

One difficulty with reducing low-frequency gain is that the low-frequency speech information is also attenuated (bottom panel). Thus any potential gain resulting from decreases in the upward spread of masking produced by the noise must be weighed against the loss of low-frequency speech information.

Figure 5-middle shows that the speech signal can also result in interformant masking, even without background noise. For example, higher-intensity low-frequency formants could mask lower-intensity high-frequency formants. Reducing the low-frequency gain could reduce this masking.

Hannley and Dorman had hearing-impaired subjects identify synthetic /ba/, /da/, or /ga/ in a laboratory experiment.[17] They decreased the amplitude of the first formant relative to the second, and found improved recognition scores for the /ga/ in several patients with noise-induced hearing loss or Meniere's disease (but not presbycusis). While their finding offers some promise, a general reduction of all low-frequency sounds would reduce all low-frequency speech information (Figure 5-bottom). Therefore, a degradation in the reception of some speech sounds would be expected.

In the studies mentioned above, both Punch and Beck and Gordon-Salant found no benefit from reducing the low-frequency gain of a hearing aid when tested in quiet.[15,16] In fact, Punch and Beck reported that word recognition decreased.[15]

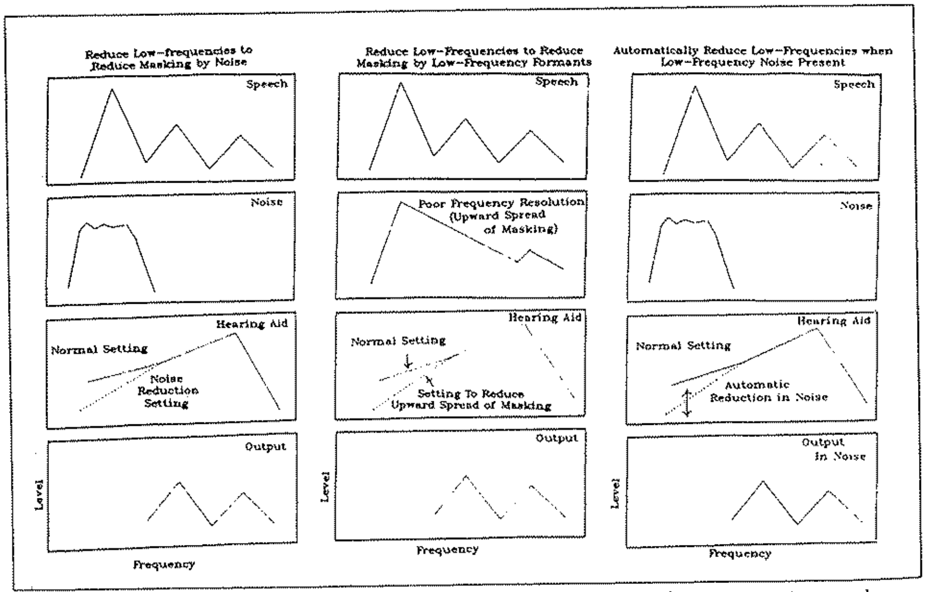


Figure 5. *Left*: Schematic example of setting the hearing aid's low-frequency gain to reduce a low-frequency noise (second). Low-frequency speech is also reduced (bottom). *Middle*: Schematic example of the setting of a hearing aid to reduce the upward spread of masking by a low-frequency formant on a mid-frequency formant. The low frequencies of speech are also reduced. *Right*: Schematic example showing how some hearing aids automatically reduce the low-frequency gain in the presence of noise.

Reduce the Low-Frequency Gain Automatically Whenever Noise Is Present

The system mentioned above has the disadvantage that low-frequency speech information is attenuated even when noise is not present. To circumvent this problem, several hearing aids decrease the low-frequency gain when low-frequency signal levels exceed a preset threshold level (Figure 5-right).

Several of these hearing aids have been evaluated recently. Stach, Speerschneider, and Jerger reported that 12 of 20 patients benefited from the Siemens ASP hearing aid in the presence of a competing single-talker discourse.[18] However, in speech-babble noise, Tyler and Kuk reported that only 4 of 15 patients benefitted with the aid.[19] In a low-frequency narrowband noise, 1 of 6 patients showed improvement. This limited success might be due to the reduction of the low-frequency speech information and the time distortions introduced by switching the filter in and out.

Reduce Any Frequency Region With Noise

The aforementioned strategies are based on the premise that most environmental noises contain low-frequency energy. However, this is not the case in all situations. Therefore, it might be desirable to reduce the noise when the noise is present in any frequency region. A novel approach has been proposed by Intellitech in their Zeta Noise Blocker.[20] Its intent is to reduce noise in a given frequency region in which the circuit determines its presence (Figure 6). The Zeta looks across the frequency spectrum to identify stationary components of the signal, which it identifies as noise. It then adjusts filter corner frequencies to reduce signal energy in the frequency region that contains these stationary characteristics. This presumably reduces the noise in that band and the effects of masking by this band on other adjacent frequency regions. For example, when midfrequency noise is present (second panel), the aid decreases the amplification in that particular frequency region.

In an early examination of prototypes of this device, Stein and Dempesy-Hart reported benefit in 6 of 15 patients in speech-babble noise and 10 of 15 patients in a low-frequency noise.[21] Van Tasell, Larsen, and Fabry, using another prototype version of the device, showed an improvement in 0 of 4 patients in speech-babble noise and 3 of 8 patients in low-frequency noise.[22]

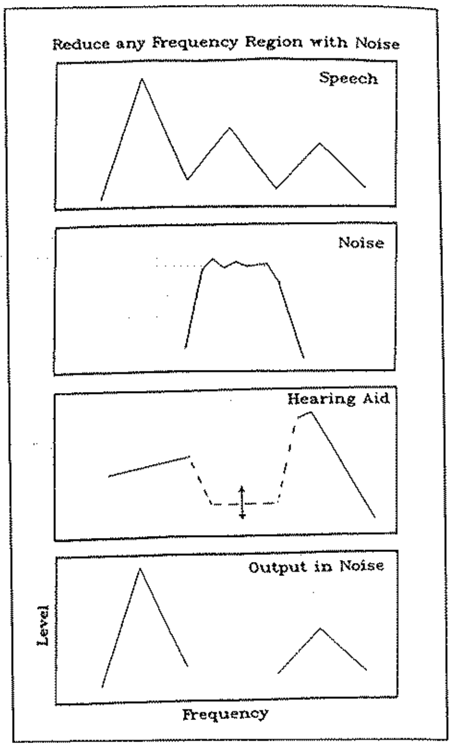


Figure 6. Schematic example showing how some hearing aids automatically reduce the gain in any frequency region, depending on where the noise is located.

In clinical trials of the circuit in a Maico hearing aid, Wolinsky reported that 13 out of 18 subjects benefitted from the noise-suppression hearing aid in the presence of low-frequency noise.[23] In contrast, only 1 out of 18 benefitted in speech-babble noise. Tyler and Kuk found only 1 patient in 12 who showed a significant improvement in speech babble, and only 1 in 6 in low-frequency noise.[19] This approach can reduce noise in a particular frequency region, but it will also reduce speech in that region.

Dichotic Presentation of Different Speech Bands

Poor frequency resolution can result in the formant frequencies masking each other (Figure 7). There can be both upward and downward spread of masking. One way of reducing this interformant masking would be to filter the speech (third panel), and then present different bands of speech into the two different ears (bottom two panels), because most of the masking effects are thought to occur peripherally in basilar membrane mechanics. Presenting different spectral information in two different ears would decrease the opportunity for this peripheral masking.

In a laboratory experiment, Van de Grift Turek, Dorman, Franks, and Sum-

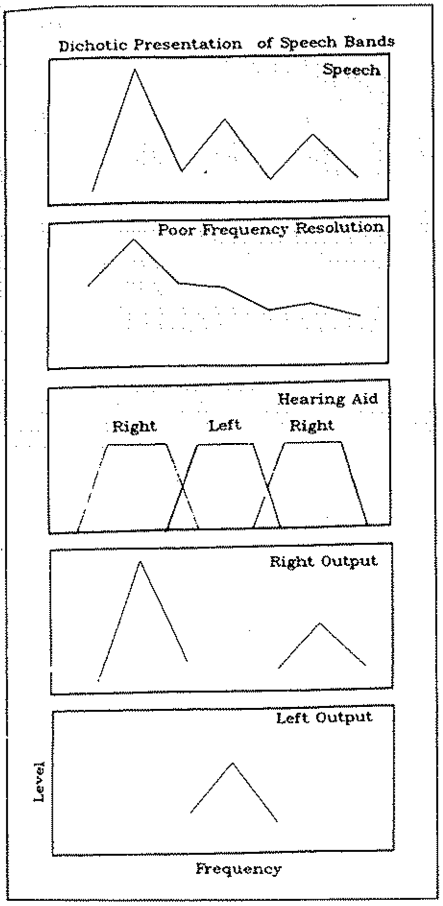


Figure 7. Schematic representation showing how speech can be filtered (third panel) so that different frequency regions are sent to different ears. This might reduce interformant masking.

merfield tested the ability of 10 hearing-impaired individuals to recognize synthetic /ba/, /da/, and /ga/. They presented F1 to one ear and F2 and F3 to the other. In four subjects with flat losses, they found improvement in 3 of 8 conditions evaluated relative to the condition where all the formants were presented to one ear.[24]

A simpler approach, one that can function with natural speech, is to filter speech, and present the output of the filters to two different ears. Franklin filtered speech through two bands (a low band 240 Hz to 480 Hz, and a high band 1020 Hz to 1240 Hz).[25,26] Six of six congenitally hearing-impaired individuals and 8 of 10 patients with an acquired sensorineural hearing loss showed some benefit (relative to the monaural condition with the same bands) in nearly all conditions. (We should note that the monaural condition does not present the entire spectrum; the comparison to a broadband monaural condition was not tested.)

In contrast to Franklin's results, Haas found that none of the 22 subjects he tested showed improvement in speech in quiet when the speech signal was presented to two filters, one to each ear (high and low pass with a crossover of 780 Hz), compared to the broadband monaural conditions.[27]

This approach requires further investigation. It might be difficult for patients to integrate signals presented to the two ears. Doing so must involve some central-processing activity, and particularly for older individuals, this might be very difficult. Furthermore, one might expect difficulties with localization or in spatially separating speech and noise.

Spectral Sharpening

Another way to improve the resolution of the formant frequencies would be to start with a better, more precise speech signal. For example, if the formant-frequency bandwidths were narrower than normal (Figure 8), then even with poor frequency resolution, the formant frequencies might still be resolved.

This approach was evaluated by Summerfield, Foster, Tyler, and Bailey, who synthesized speech with bandwidths that were narrower than normal by 50% and 25%.[28] They synthesized whispered exemplars of "bet," "debt," and "get" for initial consonant recognition and "bib," "bid," and "big" for final consonant recognition. No dramatic changes with the narrower formants were seen for the initial consonant conditions. They did find that a few hearing-impaired individuals improved their scores when the formant bandwidths were set to half their normal values in the final consonant condition. Implementing the enhancement of spectral contrasts is possible with natural speech, but algorithms currently available cannot function in real time.

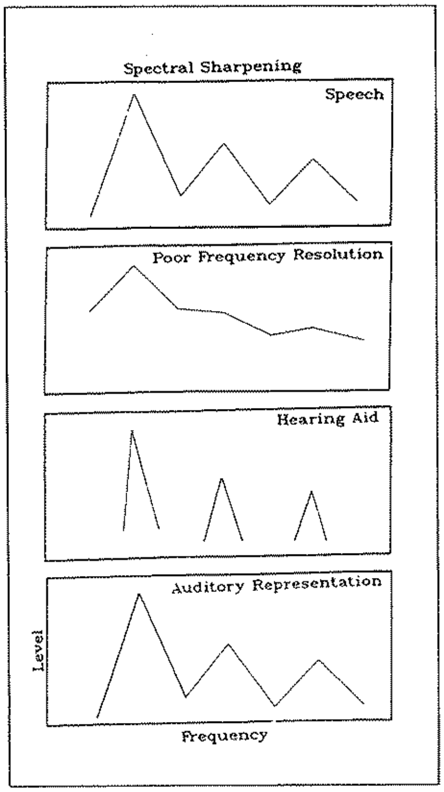


Figure 8. Schematic representation of how speech can be spectrally enhanced by 'sharpening' the formants to overcome poor frequency resolution.

Frequency Transposition

If frequency resolution is impaired in the higher frequencies but is relatively normal in the lower frequencies, then it might be possible to move the speech information contained in the high-frequency regions down to the low-frequency regions (Figure 9).

There are several ways of implementing frequency transposition (see Braida et al. for a review[29]). Unfortunately all the techniques tried have been largely unsuccessful. Patients typically require long periods of learning, and while subjects can improve their performance over time, very rarely does the performance with the frequency-transposed speech exceed that obtained with conventional (untransformed) amplification.

The problems inherent in this procedure are many. First, the sound is very different, and listeners must relearn the important features of speech. Second, frequency resolution in the lower-frequency region might be insufficient to resolve both the high- and the low-frequency information when it is "squeezed"

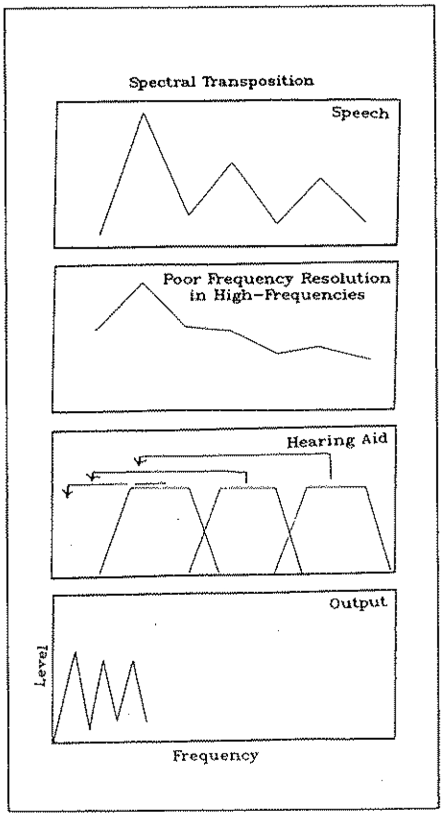
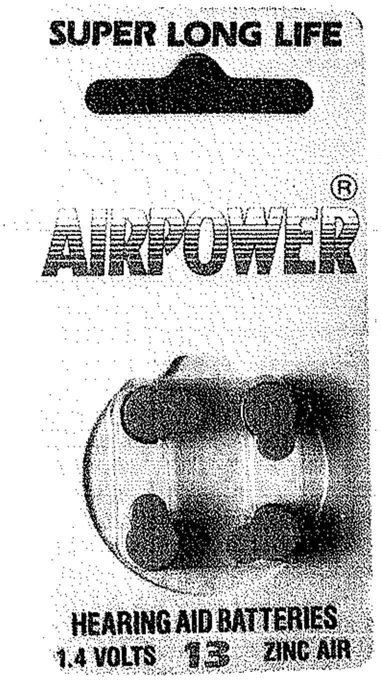
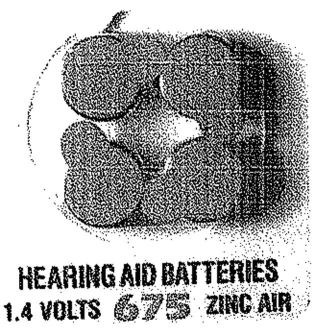


Figure 9. Schematic showing how the frequency components of speech can be transposed to a lower frequency region.





into the low-frequency region. Third, there are temporal distortions. In most implementations, frequency-lowering results in a slowing down of the speech in some way. Newer signal processing techniques might allow individual tailoring of the spectral transposition, and computer-based training procedures could focus and reduce the learning period and improve performance.

Feature Extraction

It is often said that speech is full of redundancies. It is possible that *too* much information is present for some individuals, and a reduction of some nonessential information might result in improved performance.

One example of this is a fundamental-frequency extraction device developed by Rosen and Fourcin.[4(p471)] Although frequency resolution is typically poor in the hearing impaired, the perception of fundamental frequency is relatively unimpaired. Fundamental-frequency information can be extracted (Figure 10-left), although usually through time-domain procedures. While this reduced signal is not intelligible on its own, it has been shown to be a significant improvement in lipreading.[30] Rosen and Fourcin have reported that, at least for a few individuals, speechreading performance with a hearing aid that presents only information about the fundamental-frequency can result in improved audio-visual performance relative to presenting the entire broadband acoustic signal.[4] However, the number of subjects for whom this is appropriate may be small. Furthermore, individuals with a disability this severe are likely to gain more benefit from a cochlear implant.[31,32]

It is possible to extract many features—not just the fundamental frequency—from speech. A major benefit of digital signal processing is that speech can be quantified precisely, and algorithms can be applied to extract these features. An example shown in Figure 10-right indicates that the amplitudes and center frequencies of the formant frequencies have been identified by some multiband feature-extraction procedure.

One implementation of this type has been used in the Nucleus cochlear implant.[33] Figure 11 shows the output of the speech processor for the word 'heard'. The first two formants, their amplitudes and the fundamental frequency have been coded reasonably well.

An implementation of this type in hearing aids has not been attempted. It is not certain what the correct algorithm ought to be for combining the features. However, these parameters can be manipulated freely, and different strategies can be tried easily.

One conceptual difficulty with feature-extraction approaches is that the important "features" of speech are not necessarily known. Another hurdle is to extract the features in natural ongoing speech. This is difficult enough in quiet, and it can be extremely difficult in noise. Nonetheless, this could be a useful approach for further investigation.

CONCLUSIONS

Frequency resolution can result in impaired speech perception in at least three different ways. First, poor frequency resolution results in a poor spectral representation of the speech signal. Second, poor frequency resolution can result in interformant masking. Third,

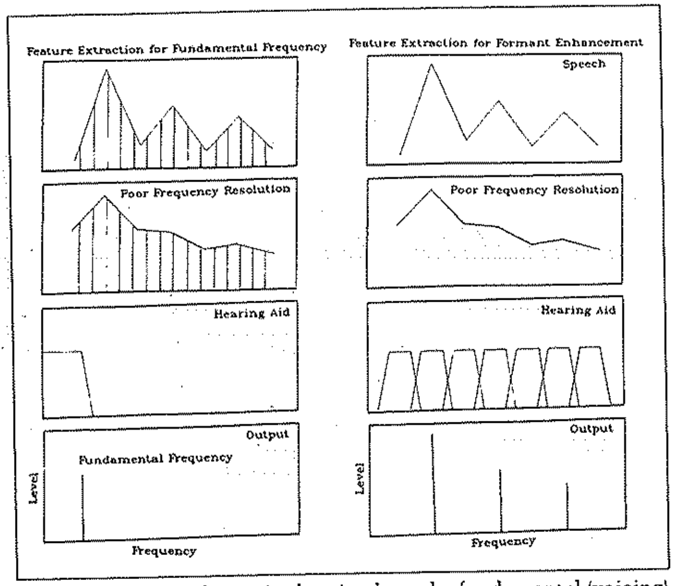


Figure 10. *Left*: Schematic showing how the fundamental (voicing) frequency of speech can be extracted, even in listening with poor frequency resolution. This signal can aid lipreading. *Right*: Schematic showing how feature extraction, in this case implemented by a multiband filter, can determine the formants and their amplitudes.

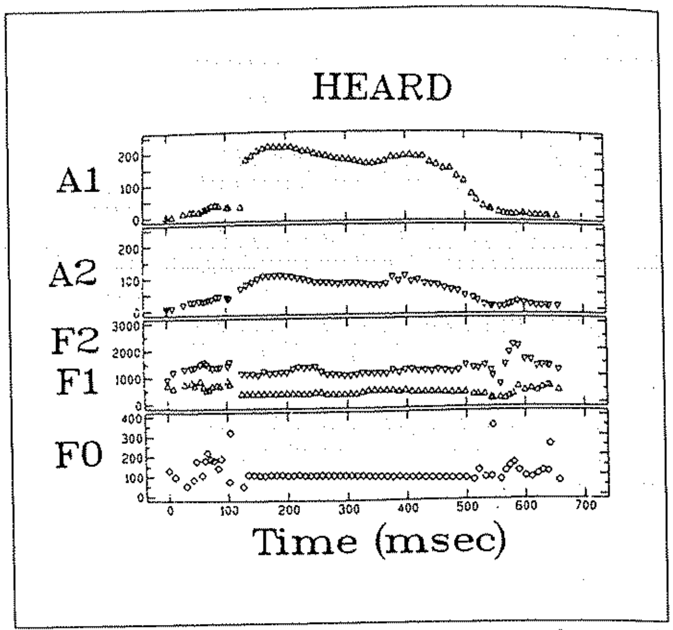


Figure 11. The output of the Nucleus Corporation implant speech processor to the word 'heard.' This is an example of feature extraction. Adapted from Blamey et al.

the ability to separate speech and noise can be impaired.

I have reviewed a number of attempts to alleviate poor frequency resolution. These efforts have not been completely successful. However, in a few individuals, particularly in some carefully controlled laboratory environments, some individuals have shown improved scores.

There might be many reasons why these attempts have not been more successful. First, there are many other auditory processing deficits, such as abnormal temporal resolution and intensity distortions, that are not compensated for when frequency resolution is 'treated' in isolation. It might be necessary to compensate for several major auditory abnormalities to have a significant overall impact on speech perception. Second, signal processing typically introduces other alterations, particularly in the time domain. Thus while the frequency representation may be improved by the processing, offsetting changes in the time domain may degrade the signal even more, so that the overall effect on performance is poor. We need to know what kind of temporal distortion in speech hearing-impaired persons can tolerate. Third, separating speech from noise by signal processing has not been solved. This is a very difficult task, and it will require a great deal of further investigation. Perhaps consideration of binaural processing strategies will be helpful, since the normal auditory system can make use of two ears to separate speech and noise.

It is also worth noting that some testing equipment may be inadequate to evaluate some of the benefits of certain signal processing schemes. For example, it is conceivable that a particular type of signal processor may reduce the "noisiness" of the signal, or reduce the listening effort, without measurably affecting intelligibility. While speech perception tests should remain the benchmark of hearing aid performance, there is a need for additional measures for evaluating signal processors.

It is important to attend to individual differences. Not all individuals have reduced frequency resolution, and therefore we cannot expect *all* patients to benefit from processing directed at alleviating the effects of impaired frequency resolution. Further work is required to address more specifically how impaired frequency resolution distorts particular aspects of speech. The perception of some speech sounds might depend heavily on frequency resolution, whereas others might not. There might be interactions between the effects of impaired frequency resolution and other distortions. For example, if the temporal distortions are large, then impaired frequency resolution might be less important. However, in aged listeners with poor central-processing skills, moderate frequency resolving abnormalities might have great consequences.

In summary, designing signal-processing hearing aids to reduce the effects of impaired frequency resolution has only been partly successful thus far. Nevertheless, several instances of well-documented benefit are evident, and a variety of different schemes are available. Digital technology will make the implementation of these schemes easier. This will facilitate trying a variety of systems on individual patients in a reasonable amount of time. As basic research about hearing loss increases, and digital-processing hearing aids are designed to make use of this information, we might see some rapid advances in the upcoming years.

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