Signal processing strategies – Advanced

Signal processing strategies essentially instruct the hearing instrument on how to handle or process an incoming sound signal.

Channels and bands

With analogue hearing instruments, the term frequency band is used interchangeably with the term channel. These bands are separated by filtering from other frequency regions. This is known as band splitting. In each band, only gain is adjusted.

With digital hearing instruments, it is possible to adjust gain, compression, noise reduction and feedback reduction. Although not an internationally accepted definition, it has become convention that if only gain is adjusted, the frequency region is a 'band'. If more than gain is adjusted, the frequency region is a 'channel'. With this convention, it is possible to have more bands than channels but not the other way around.

The steepest slope found in analogue hearing instruments is 24dB/octave. However, with digital signal processing, the slope between adjacent channels can be almost infinitely steep. This means that turning up the gain in one frequency band may not affect the gain in adjacent bands.

Some digital signal processing strategies operate based on time—the time domain—while others operate based on frequency—the frequency domain. Finite impulse response (FIR) and fast Fourier transform (FFT) are the two most popular signal processing strategies.

Example

Signal processing strategies are similar to a symphony orchestra with all its sections, such as woodwind, string and percussion. Each section is very good at what it does but it is the conductor that leads the orchestra and makes all the sections come together. In the same way, digital hearing instruments have various algorithms for adjusting gain, frequency and noise reduction, but it is the signal processing strategy that indicates how this should be done.

Jargon Buster

A **sloping hearing loss** is a hearing loss that becomes gradually worse moving from low frequencies to high frequencies.

Frequency transposition is when an entire portion of a frequency range is moved to a different portion of the frequency range. For example, all of the frequency 2000-6000Hz could be transposed to 1000-4000Hz.

Frequency compression is when all frequencies are transposed by the same factor.

Peak clipping refers to removing the excessively loud portions of an amplified signal. The loud parts are 'clipped' so that the amplified sound is not too loud for the patient.

Intermodulation distortion products are peaks that occur in a spectrogram

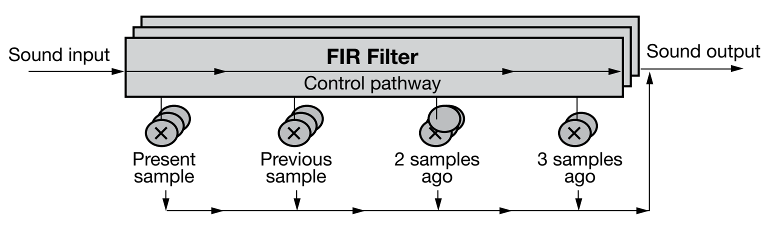
A **transposing speech vocoder** is one that transposes the speech spectrum.

FIR

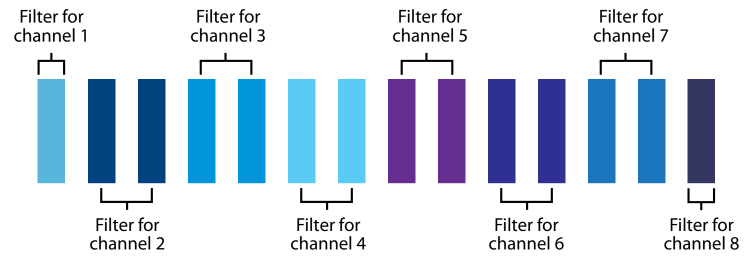
When the frequency response is divided into only a small number of channels, it is most efficient to operate in the time domain. FIR is an example of a multi-channel filter bank compressor. The FIR filter assigns one of any finite series of quantisation values to each successive sample of input over time.

If there are a finite number of samples, perhaps 15,000 samples per second, a new sample can be taken or quantised every 50 microseconds (every one millionth of a second). First, the desired output frequency response is fed into the hearing instrument. Then the FIR filter assigns a quantised value to each new input sample taken so as to achieve the desired output frequency response. Each new quantised sample is added to all the other samples previously collected in order to constantly update the entire frequency response over time.

Several FIR filters together create fixed channels. For channels situated between the lowest and highest frequency channels, two FIR filters are required to produce one channel (one for the low-frequency side of the channel and one for the high-frequency side of the channel). The lowest frequency channels only have a high-frequency FIR filter and the highest frequency channels only have a low-frequency FIR filter. This means that an eight-channel digital hearing instrument has 14 FIR filters.



Example of an 8-channel hearing instrument using FIR



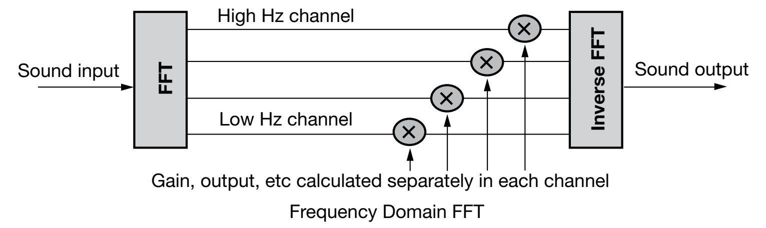
FFT

Frequency-domain hearing instruments operate in small units of time. All the samples from each parallel band within each unit of time are processed together. These processed units are moved along to the receiver like small packages along an incredibly fast conveyor belt. The length of the processing time is only as long as the length of these units of time.

Processing time delays occur in all hearing instruments in terms of milliseconds. Digital systems always take longer to accomplish the same filtering as analogue systems. The time delay can be heard when it starts to approach 20 milliseconds. When this happens, if the patient were to clap their hands while wearing the hearing instrument, there would be a short delay before hearing the clap.

Digital hearing instruments operating in the frequency domain have longer processing times than those operating in the time domain. However, the advantage is that the slopes of the channels can be made steeper without increasing processing power.

Digital hearing instruments operating in the time domain offer little processing delay. However, the disadvantage is that they tend to require more processing power than instruments operating in the frequency domain.



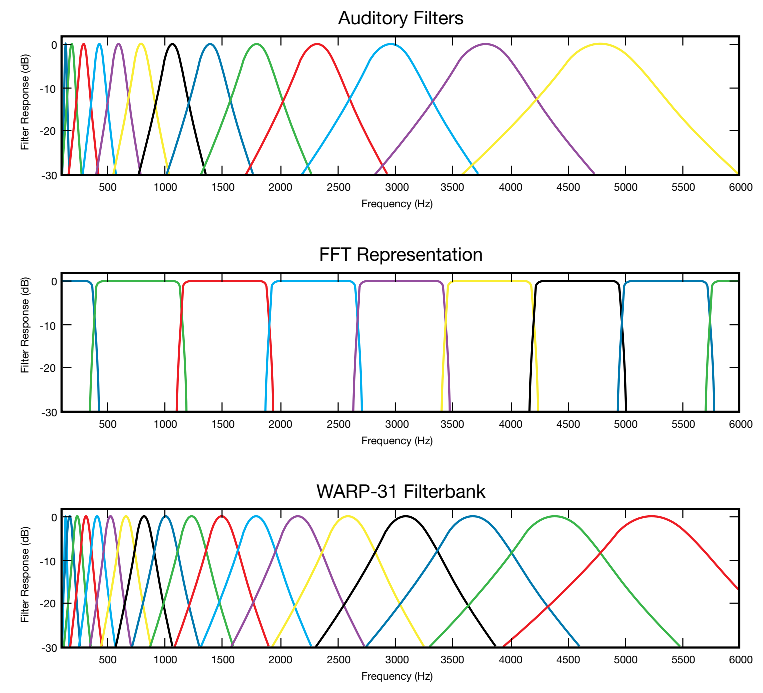
Related information

Also refer to the section in this manual on **Open fittings**, **Noise reduction**, **Feedback suppression**, **Hearing instruments and hearing loss**.

WARP

ReSound uses frequency warping which is similar to using an FFT filter, except much faster. Instead of using uniform frequency spacing, WARP processing uses non-uniform frequency spacing. The frequency spacing is based on the auditory bark scale so that it more closely resembles natural auditory perception. Sound is still sampled and analysed.

The advantages of this system are faster processing times, excellent frequency resolution and excellent sound quality.



Frequency transposition

Research has shown that when there is a sloping hearing loss with high-frequency thresholds of 70dB HL or more, the high-frequency components of speech contribute no information. For these patients, making the high-frequency components of speech audible can actually decrease their ability to hear useful information from the low- and mid-frequency components of speech. In order for these patients to be able to access high-frequency portions of the speech spectrum, information must be moved down to another frequency region where the patient is able to analyse sounds more effectively. This is known as frequency transposition.

There are several ways to shift frequency information. One approach is to use extreme peak clipping. This results in intermodulation distortion products occurring at frequencies quite different from the frequencies of the original input signal.

A more advanced approach is to take a certain input range and move it to a lower frequency range. The problem with this approach is that natural energy still occurs in this lower range so when there is a low-energy sound, it can be difficult to identify whether it comes from the original input signal or the transposed energy.

The ability to identify transposed consonants improves with training. Nonetheless, many patients with severe high-frequency hearing loss report that this strategy improves speech clarity. Speech cues are shifted using a transposing speech vocoder. Speech is filtered into adjacent narrow bands and the level of speech is detected within each band. These detected levels from high-frequency bands can then be used to modulate low-frequency bands.

Frequency compression

A limitation of frequency transposition is that transposed energy from the high-frequencies can compete with low-frequency energy. This problem can be solved by transposing the entire frequency range by some factor.

Every input frequency could be shifted to an output frequency that is half of the input frequency. This has the same effect as changing the fundamental frequency or formant frequencies. This could result in female voices sounding like male voices.

Transposition of all the frequencies by the same factor is called frequency compression.

Jargon Buster

Gain is the difference in dB SPL between the hearing instrument's input and output.

In **compression**, air molecules are moved closer together.

Noise reduction is an algorithm in a hearing instrument that is designed to reduce background noise.

Feedback reduction refers to the process of reducing or eliminating amplified sound from being re-amplified to cause howling (i.e the process of reducing or eliminating feedback).

Frequency spacing refers to the width of a frequency band or a frequency channel.