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## Key Words
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# ICRA Noises: Artificial Noise Signals with Speech-like Spectral and Temporal Properties for Hearing Instrument Assessment

## Ruidos ICRA: Señales de ruido artificial con espectro similar al habla y propiedades temporales para pruebas de instrumentos auditivos

## Abstract
Current standards involving technical specification of hearing aids provide limited possibilities for assessing the influence of the spectral and temporal characteristics of the input signal, and these characteristics have a significant effect on the output signal of many recent types of hearing aids. This is particularly true of digital hearing instruments, which typically include non-linear amplification in multiple channels. Furthermore, these instruments often incorporate additional non-linear functions such as "noise reduction" and "feedback cancellation". The output signal produced by a non-linear hearing instrument relates to the characteristics of the input signal in a complex manner. Therefore, the choice of input signal significantly influences the outcome of any acoustic or psychophysical assessment of a non-linear hearing instrument. For this reason, the International Collegium for Rehabilitative Audiology (ICRA) has introduced a collection of noise signals that can be used for hearing aid testing (including real-ear measurements) and psychophysical evaluation. This paper describes the design criteria, the realisation process, and the final selection of nine test signals on a CD. Also, the spectral and temporal characteristics of these signals are documented. The ICRA noises provide a well-specified set of speech-like noises with spectra shaped according to gender and vocal effort, and with different amounts of speech modulation simulating one or more speakers. These noises can be applied as well-specified background noise in psychophysical experiments. They can also serve as test signals for the evaluation of digital hearing aids with noise reduction. It is demonstrated that the ICRA noises show the effectiveness of the noise reduction schemes. Based on these initial measurements, some initial steps are proposed to develop a standard method of technical specification of noise reduction based on the modulation characteristics. For this purpose, the sensitivity of different noise reduction schemes is compared by measurements with ICRA noises with a varying ratio between unmodulated and modulated test signals: a modulated–unmodulated ratio. It can be anticipated that this information is important to understand the differences between the different implementations of noise reduction schemes in different hearing aid models and makes.

## Sumario
Los estándares actuales de las especificaciones técnicas de los auxiliares auditivos proporcionan posibilidades limitadas para conocer la influencia de las características temporales y espectrales de la señal de entrada y estas características tienen un efecto significativo en la señal de salida de muchos tipos actuales de auxiliares auditivos. Esto es particularmente cierto en el caso de los instrumentos digitales, que típicamente presentan una amplificación no lineal en canales múltiples. Incluso, estos instrumentos a menudo incorporan funciones no lineales adicionales como la "reducción de ruido" y la "cancelación de la retroalimentación". La señal de salida producida por un instrumento no lineal se relaciona en una forma muy compleja con la señal de entrada, por lo que la selección de la señal de entrada influye significativamente en la salida de cualquier prueba acústica o psicoacústica de un instrumento auditivo. Por esta razón el Colegio Internacional de Audiología Rehabilitatoria (ICRA) ha introducido una serie de señales de ruido que pueden ser utilizadas en las pruebas de auxiliares auditivos (incluyendo mediciones in-situ) y evaluaciones psicoacústicas. Este trabajo describe los criterios de diseño, el proceso de realización y la selección final de 9 señales de prueba en un CD. También se describen las características espectrales y temporales de estas señales. Los ruidos ICRA son un conjunto muy específico de sonidos vocales con forma espectral acorde al género y el esfuerzo vocal y con diversas dosis de modulación vocal que simulan uno o más hablantes. Estos ruidos pueden ser utilizados como ruidos de fondo muy específicos en experimentos psicoacústicos. También pueden servir como señales de prueba para evaluar auxiliares auditivos digitales con reducción de ruido. Se demuestra que los ruidos ICRA tienen la efectividad de los esquemas de reducción de ruido. Con base en estas mediciones iniciales, se proponen algunos pasos iniciales para desarrollar métodos estándar de las especificaciones técnicas de reducción de ruido tomando en cuenta las características de modulación. Para ello se compara la sensibilidad de diferentes esquemas de reducción de ruido mediante las mediciones de ruidos ICRA con un rango de variación entre señales de prueba modulares y no modulares. Se puede anticipar que esta información es importante para comprender las diferencias al implementar diferentes esquemas de reducción de ruido en diferentes modelos y tipos de auxiliares auditivos.



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## Introduction

The utilisation of digital technology in hearing aids has increased the occurrence of non-linear hearing aids with complex signal-processing algorithms. Assessment of such hearing aids is difficult, because sounds are processed by their circuitry in different ways, depending on the properties of the sound field in which they operate. This means that the sound signals used for the assessment must be reproducible in every detail in order to ensure reproducible results. This concern applies not only to the verification of their sound reproduction in specific user situations, but also to the clinical assessment, which typically involves speech recognition in some kind of background noise.

*Hearing aid measurements*

Traditionally, hearing instrument characteristics are measured in an acoustic coupler in response to a sweep-tone at a constant level. This type of measurement is adequate for describing the performance of linear hearing aids, but proves insufficient or even misleading in connection with non-linear hearing instruments.[1,2] Modern hearing instruments are no longer simple linear amplifiers that can be characterised completely by fixed frequency–gain curves. The introduction of different forms of non-linear amplification calls for more intelligent ways to specify the properties of the hearing instrument.

Traditional measures have already been shown to be inadequate to describe the different effects of compression, especially when applied in a multichannel hearing instrument. The spectral and temporal properties of the test signal play an important role. For broadband compressors and for multichannel compression hearing aids with inter-channel coupling, the local gain at one frequency will typically be influenced by the levels at other frequencies. This implies that a narrowband approach, as used by input–output curves for a specific frequency, or the use of locally active sweep-tones are less suitable methods for describing the non-linear transformations of a broadband signal such as speech. Especially for multi-channel compression hearing aids with inter-channel coupling, as applied in some digital hearing aids, this approach is too simplistic. This problem has been addressed in a number of amendments to IEC 118, which recommend that a speech spectrum-shaped unmodulated noise signal be used for the measuring signal. The dynamic properties of the test signal are also important. Only in compression schemes where the attack and recovery times do not exceed a few milliseconds does the dynamic input–output curve resemble the static curve. In most wide dynamic range compression hearing aids, dynamic input–output curves differ substantially from static curves.[3,4]

The recent introduction of noise reduction based on the signal dynamics complicates the technical specification of hearing instruments even further and makes the traditional approach completely inappropriate. In these algorithms, the envelope signal is analysed in different frequency channels. Speech usually shows much more modulation than noise. The modulation behaviour is used to decide whether the signal in each frequency channel is dominated by a speech signal and should be amplified or, alternatively, by the background noise and should be reduced to a certain degree. The technical specification of these forms of signal processing usually lack necessary details, but the signal processing usually requires long time constants (up to 15 seconds), and continuous test signals will reveal other properties than signals with speech-like variations.

The different forms of non-linear processing call for more realistic test signals because there is no simple relationship between the outputs produced in response to different input signals.[5] Therefore, the measurement procedures proposed by IEC (or by ANSI)[6] are inadequate to describe the sound reproduction of a significant portion of today's hearing aids in daily-life situations.

*Clinical tests*

A clinical assessment of hearing instruments or hearing instrument processing methods typically involves a determination of performance for speech recognition in background noise. The results of such performance measures will be largely determined by the spectral and temporal properties of the speech signal and the concurrent background noise. Especially in the psychophysical evaluation of relatively slow noise reduction algorithms, the experimental conditions should be chosen with great care.[7] Therefore, the psychophysical evaluation of hearing aids in a clinical or laboratory setting relies on noise signals with well-defined properties. The application of noise signals with well-described properties makes it possible to duplicate the same experimental conditions in different laboratories.

In conclusion, there is a need for the specification of new test signals that better reflect the spectral and temporal characteristics of the sounds of everyday life situations. Among the variety of such signals, we have given preference to signals with a speech-like character. The following general requirements have been set up for the test signal criteria:

1. Signals with average spectral characteristics similar to those of speech produced with various degrees of effort.
2. Signals with temporal characteristics similar to those of speech coming from one or more speakers.

Such signals can be used to describe the interdependence between the different frequency bands in a non-linear hearing aid in a number of realistic listening situations. The signals can also be used to evaluate the dynamic characteristics of non-linear processing stages with special attention to the modulation characteristics that may be used in hearing aids to activate their noise reduction algorithms. Inevitably, additional demands will be placed on hearing aids of the future to process a variety of test signals. There is a definite need now for a collection of broadband noise signals with precisely documented acoustical properties. Comparative measurements in different hearing instruments can then be carried out effectively.

Based on these constraints, a working group from the International Collegium of Rehabilitative Audiology (ICRA) began the process of designing well-defined artificial noise signals. Their work was introduced as a recommendation for hearing instrument assessment, both for technical measurements and for applications in clinical and psychophysical evaluations. This paper reports on the design criteria and actual realisation.

## Design criteria for a set of test noises

Owing to their background in advanced signal processing and clinical evaluations, the members of the ICRA group wished to develop generally accepted guidelines for hearing instrument evaluation protocols. They decided to contribute to this by specifying and making available a set of background noises that could be used both for technical measurements and for clinical testing of speech recognition in noise. A working group was formed within the ICRA in 1993, and the first results were reported in 1997.

The working group specified that the noise signals should have long-term average frequency spectra similar to those of typical speech signals produced under realistic conditions. To allow for international use, it was further decided that the noise signals should not consist of intelligible speech segments. A set of broadband noise signals with the following parameters were created:

- Speech noise shaped according to average vocal effort (normal/raised/loud).
- Speech noise shaped according to the gender of the speaker (male/female).
- Speech noise modulated according to a number of well-described communication situations without affecting the overall frequency response of the noise.

With respect to the variations in modulation, the goal was to simulate a number of realistic background noise conditions with decreasing amounts of modulation:

A condition with pronounced modulation, equivalent to a single-speaker interference.

- A condition with moderate modulation, equivalent to two competing speakers.
- A condition with slight modulation, equivalent to a six-talker babble with two speakers in the front and four speakers at some distance.
- A condition without clear modulation (continuous noise), equivalent to a situation with loud cocktail party noise.

An artificial babble noise with different temporal characteristics was implemented by adding artificial speech signals that represented a single speaker. In all cases, the spectral and temporal properties were controlled and had a close resemblance to real-life communication.

Acoustical conditions of reverberation may vary in many conditions, specifically in the spectral characteristics. Their temporal characteristic will be one of smoothing and reducing the temporal envelope. The effects of reverberation will therefore be similar to adding extra speakers, and have been left out for this reason.

## Realisation of the ICRA noises

*Equalization of the spectrum and preservation of the temporal envelope*

All signals originate from an English text read by a female speaker. The text explains the system of arithmetical notation (number scale) and stems from the EU SAM project.[8]

Figure 1 shows a block diagram of the noise generation process. The original speech signal consisting of running English text was filtered by band-split filters. The number of bands in the band-split filter has been subject to discussion. If filtering is omitted, i.e. if an all-pass filter is used, the speech envelope in different frequency bands will be perfectly correlated. On the other hand, if too many bands are used, the resulting signal will be intelligible. As a compromise, three bands are used. (The single-speaker noises generated by this three-band approach may have some intelligible fragments, at least for native speakers who concentrate on the noise alone. Our hypothesis is that this will not disturb the results of psychoacoustical tests in which these noises serve as maskers, but this has to be verified in formal listening experiments.) The crossover frequencies of the filters were selected so that the low-pass filter covers the first formant area of voiced speech sounds, and the 3-dB cut-off frequency is at 800 Hz. The high-pass band was selected to cover the frequency range of unvoiced fricatives, like /s/, and the cut-off frequency is set to 2400 Hz. The width of the mid-frequency band is thereby defined at 800–2400 Hz, which approximately covers the second formant area of voiced speech sounds. The slope of the filter flanks exceeds 100 dB/octave and the damping outside the pass-band exceeds 50 dB.

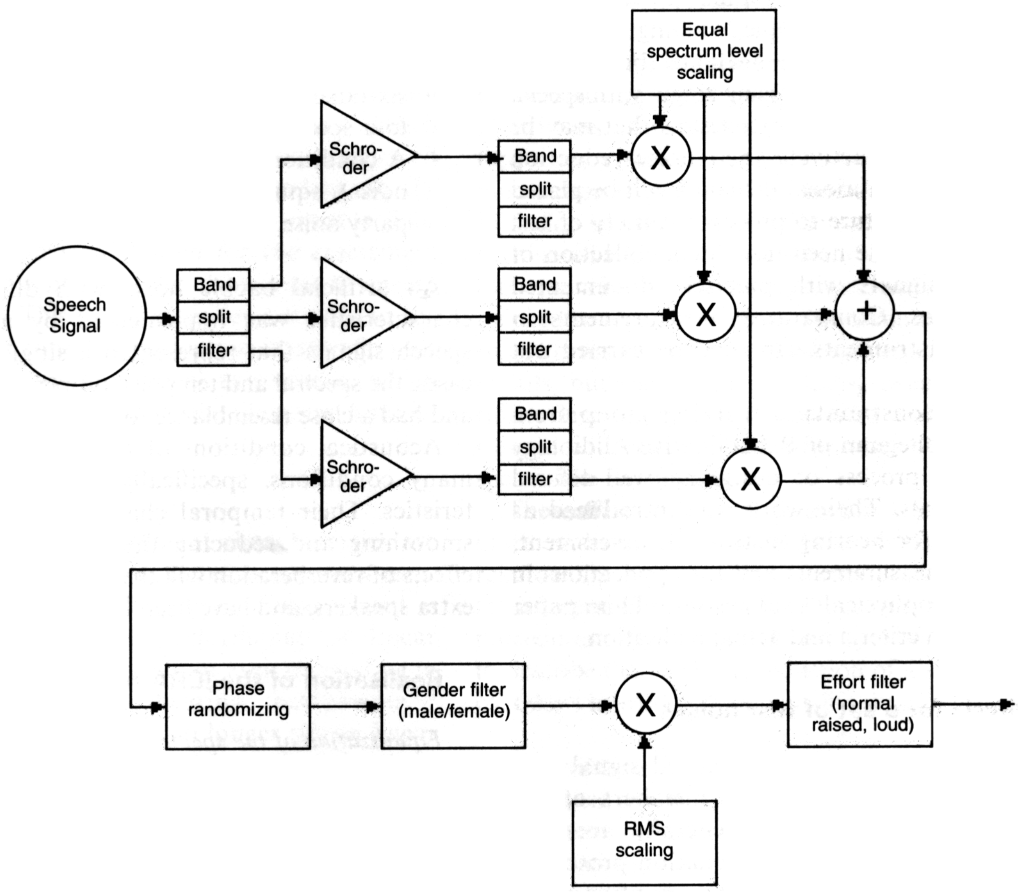


**Figure 1.** The principle of the noise generation.

Next, the filtered signal in each of the three bands was processed according to Schroeder.[9] With a probability of 50 per cent, the sign of each sample is randomly reversed or kept unaltered. Since this process preserves the numerical values of all samples, each of the modified signals has the same modulation properties as the original speech signal. However, they are unintelligible and have a flat (white) spectrum. Next, the filters that originally separated the samples are used, and the filtered signals are then scaled to the same spectrum density level as the three Schroeder-processed signals. Now the three bands are added together, forming one signal with a white spectrum. The original modulations that were separately preserved in each of the three frequency ranges remain.

*Spectral shaping according to gender and vocal effort of the speaker*

In order to obtain the desired spectrum, the signal was subsequently filtered with a gender filter to resemble the overall spectral shape of male or female speech (in accordance with the ANSI S3.5,[6] proposed standard for the calculation of the SII). Throughout most of the frequency range these spectra were consistent with the universal LTASS, measured by Byrne et al.[10] The main difference is that for frequencies above approximately 4 kHz, the frequency spectra described by Byrne et al have a more gradual slope. This high-frequency "boost" is often found in speech recorded in front of the speaker, but not in speech recorded in a reverberant sound field far from the speaker. This difference probably means that sound at high frequencies is mainly radiated in a frontal direction. We found that the shape of the speech filter, according to ANSI S3.5, represents an average of speech recorded at different azimuths, while the Byrne et al spectrum represents speech recorded at an azimuth of 45°. The filter characteristics of the male and female filters are +12 dB per octave below 100 Hz, respectively 200 Hz, then flat up to 500 Hz and −9 dB/octave above 500 Hz. Since these signals had an unpleasant scratchy sound, the phases where modified in a 512-point FFT procedure. This was accomplished by randomising the phase and overlapping/adding the segments after an inverse FFT with a 7/8 overlap.

Finally, the signals were filtered according to the speech spectrum of normal, raised and loud vocal efforts. The shapes of these filters were derived by subtracting the spectral levels of normal speech from the spectral levels of raised and loud speech (see ANSI S3.5). The resulting signals had long-term spectra resembling the speech spectrum of choice, as well as the modulation characteristics of the natural speech of a single speaker.

*The composition of nine standard signals*

The final step was the composition of nine test signals meeting the design criteria described above (for details see Tables 1 and 2). The signals in the two channels were practically uncorrelated, one track being a 30-second delayed replica of the other. All levels were equalised in terms of long-term RMS.

The noises described in Tables 1 and 2 were recorded and put on a CD. The CD contains the following tracks:

> Tracks 1, 2 and 3 are unmodulated noises for the male voice at normal, raised and loud vocal effort, respectively.
> Tracks 4 and 5 represent the single-speaker noises at a normal vocal output for the female and the male voice, respectively.
> Track 6 represents a two-speaker background noise with two equally loud speakers of different gender, speaking at normal vocal effort.
> The remaining tracks represent multi-talker babble noises for conditions with a male and a female speaker at a short distance and two female and two males at some distance (with a 6-dB level reduction to simulate the distance). The differences between the two-speaker background noise and the multi-talker babble are found only in the temporal characteristics (less modulation). The only spectral differences are due to the fact that tracks 7, 8 and 9 represent similar multi-talker babble noises, but measured at normal, raised, and loud vocal efforts, respectively.

In all cases, the spectral and temporal properties have been carefully controlled and have a close link to real speech.

## Verification of the noises

To verify the spectral contents of the recorded noises, a 2-minute track of each signal was taken for analysis. Each track was sampled at 25 kHz, resulting in 1465 50 per cent-overlapping frames of 4096 samples. The 1465 (40-μs) time signals were Hanning-windowed and the power spectrum of each frame was calculated using FFT. The average power spectrum was divided into third-octave bands with mid-frequencies from 125 Hz to 8 kHz, and the FFT components within each third-octave band were summed, after which 10 times the $\log_{10}$ was taken to transform the output into decibels.

The results of the spectral analysis are presented in Figure 2. The left-hand panels show the spectral shapes relative to the RMS level of each individual noise signal. The right-hand panels show the absolute spectra relative to the RMS of the calibration tone. The spectral differences as a function of vocal effort (tracks 1–3) can be seen in the upper panels. In the middle panels, the spectral differences as a function of gender (at a normal vocal effort) are represented (tracks 4 and 5). These panels also show that the spectral characteristics of the two-speaker noise (track 6) are close to the multi-speaker noise at a normal vocal effort (track 7). For the other multi-speaker noises (tracks 8 and 9, presented in the lower panels), the same differences as a function of vocal effort are present as for the continuous noises (see upper panels).

The input of the same tracks is used for the temporal analysis. The envelopes of the signal are calculated according to Houtgast et al.[11] The absolute values of the signal are low-pass filtered (100 Hz) and down-sampled

**Table 1** Overview of the nine standard noises that have been selected for the ICRA CD with test signals.

| *Track number* | *Character of the noise* | *Short name* | *Type of modulations* | *Gender** | *Vocal output*†‡ | *Duration (min)* |
|---|---|---|---|---|---|---|
| | Continuous normal | Normal | Unmodulated | | Normal | 2 |
| 2 | Continuous raised | Raised | Unmodulated | | Raised | 2 |
| 3 | Continuous loud | Loud | Unmodulated | | Loud | 2 |
| 4 | One-speaker female | Female | One-speaker | | Normal | 5 |
| 5 | One-speaker male | Male | One-speaker | | Normal | 5 |
| 6 | Two-speaker | 2-sp | Two-speaker | | Normal | 10 |
| 7 | Babble normal | 6-sp normal | Multi-speaker | | Normal | 20 |
| 8 | Babble raised | 6-sp raised | Multi-speaker | | Raised | 10 |
| 9 | Babble loud | 6-sp loud | Multi-speaker | | Loud | 10 |

*Male weighted spectrum: HP 100 Hz 12 dB/oct.
Female weighted spectrum: HP 200 Hz 12 dB/oct.
†Idealised speech spectrum according to ANSI S3.79 draft v3.1–07/03/1993.
‡Raised and loud effort according to ANSI S3.79 draft v3.1–07/03/1993.

**Table 2.** Overview of the track numbers for the ICRA noises, according to their spectral (rows) and modulation (columns) characteristics.

| | *Single speaker* | *Two speakers* | *Six-speaker babble* | |
|---|---|---|---|---|
| | | 6 (F+M) | 7 (F+M) | |
| | | | | 2 (M) |
| | | | | 3 (M) |

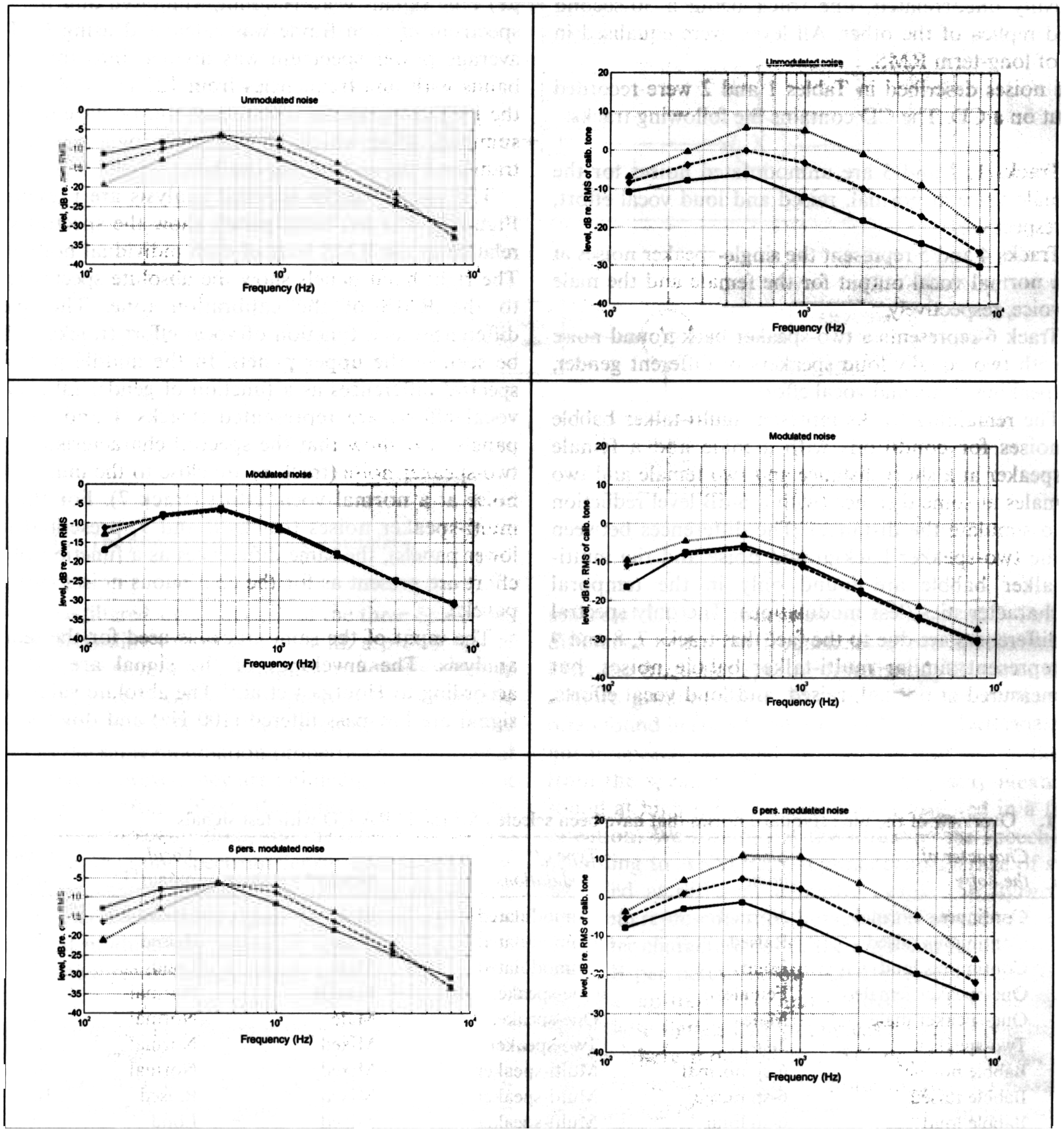


**Figure 2.** Frequency spectra of the ICRA noise signals. In the left-hand panels the spectra are presented in octave levels relative to the RMS level of each individual noise. In the right-hand panels the spectral levels are presented relative to the RMS level of the calibration tone.

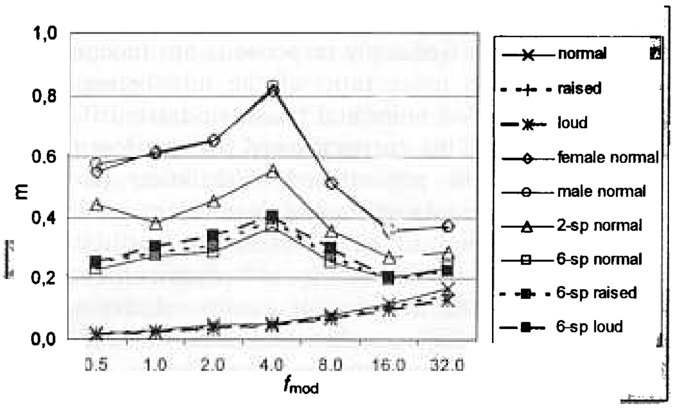


**Figure 3.** Modulation spectra of the ICRA noise signals.

single-speaker modulated noises to the unmodulated noises. Unmodulated noises and single-speaker noises are obviously clustered together, whereas multi-speaker-modulated noises show a slight difference in overall modulations.

## Application of ICRA noises to characterise noise reduction

We measured a three-channel digital hearing aid with noise reduction based on modulations (Widex Senso C8), with the noise signals ICRA-1 (unmodulated speech noise from a male speaker at normal level) and ICRA-5 (single-speaker modulated speech noise from a male speaker at normal level), presented at a level of 80 dB SPL. It can be seen from Figures 2 and 3 ("normal" versus "male") that the frequency spectra of these noises are almost identical, but that the modulation spectra are quite different.

During the start of the measurements the responses to both signals are very similar, but after some seconds the noise reduction algorithm becomes activated for the continuous noise (ICRA-1) and the response curve drops to a lower output level. In contrast, the response curve for the modulated noise (ICRA-5) remains constant, because the noise reduction is not activated by the speech-modulated signal. The difference between the two response curves increases over 10–15 seconds and, after 15 seconds, a steady response is obtained as shown in Figure 4. The difference between the curves shows the full effect of noise reduction, as measured for this particular hearing aid. The ICRA noises can thus be applied to verify the effectiveness of noise reduction based on signal modulation. It should be emphasised that, in most cases, the amount of noise reduction depends on the setting of the hearing aid. This applies not only to the parameters of the noise reduction algorithm itself, but possibly also to other settings related to the individual hearing loss (in this case corresponding to a flat hearing loss of 50 dB).

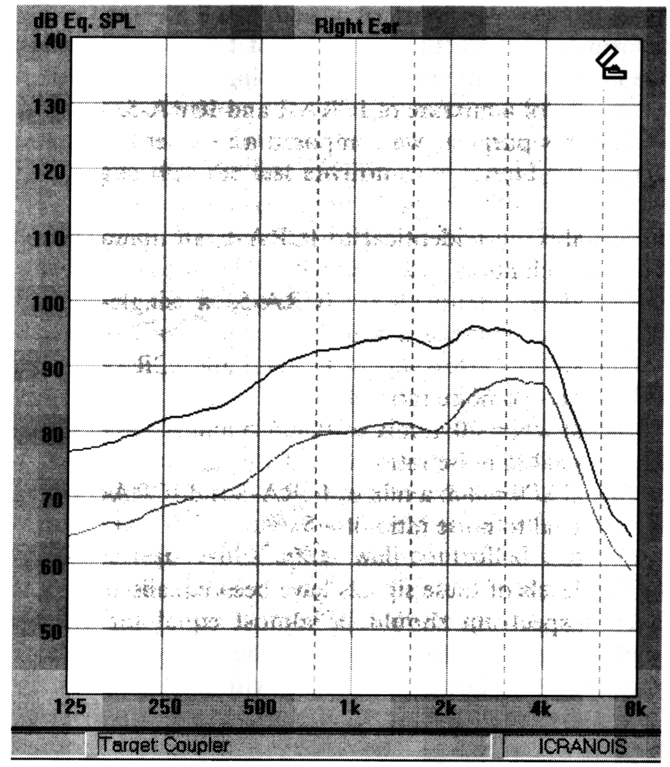


**Figure 4.** Output responses for the test hearing aid in an identical setting for ICRA noise 1 (continuous noise, lower curve) and ICRA noise 5 (single-speaker modulated noise, upper curve).

Although useful data have been obtained with regard to the amount of noise reduction as a function of frequency, it is also clear that these measurements are only a first approach to a full documentation of the noise reduction schemes that are currently available in digital hearing instruments. There are at least four characteristics which determine the properties of each specific noise reduction scheme:

1. The number of channels with independent noise reduction controls.
2. The attack and release times (which may also be frequency-dependent).
3. The degree of noise reduction as a function of frequency, being the difference between the long-term responses for modulated and unmodulated noises.
4. The sensitivity of the noise reduction algorithm, being the amount of noise reduction as a function of the ratio between modulated and unmodulated components in the signal.

The first two elements of information are usually provided by the manufacturer (although the frequency dependence of the adaptation times is not routinely given). The third characteristic can be derived from straightforward measurements with two ICRA noises. For information about the sensitivity of the noise reduction we need an additional approach using composite signals consisting of a mixture of ICRA-1 and ICRA-5.

For this purpose, we composed a new set of five test signals:

Signal N: identical to ICRA-1, an unmodulated speech noise
Signal S: identical to ICRA-5, a single-speaker modulated speech noise
Signal S/N=−5: a mix of ICRA-5 and ICRA-1 with a signal to noise ratio of −5 dB
Signal S/N=−0: a mix of ICRA-5 and ICRA-1 with a signal to noise ratio of 0 dB
Signal S/N= +5: a mix of ICRA-5 and ICRA-1 with a signal to noise ratio of +5 dB.

The levels of these signals have been equalised and the overall spectrum should be almost equal for all five signals. In order to verify this, the signals have been presented to a linear hearing aid with ample range. The outputs of this hearing aid for the five different input signals have been plotted in Figure 5, and they indeed show almost identical curves.

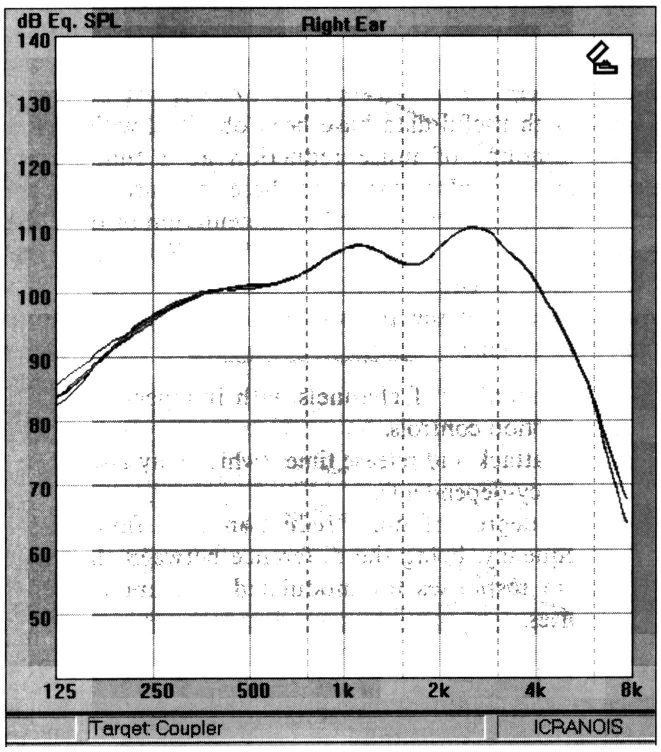


**Figure 5.** Output responses for a linear hearing aid in an identical setting for ICRA-noise 1 (continuous noise) and ICRA-noise 5 (singe-speaker modulated noise) and for three mixed signals of ICRA-1 and ICRA-5 with "signal to noise ratios" of −5, 0 and +5 (for details, see text).

In a digital hearing aid with noise reduction based on modulations, the frequency response is not independent of the signal to noise ratio of the mix between the modulated ICRA-5 noise and the unmodulated ICRA-1 noise. In general, the curves showed less gain/output for signals with less pronounced modulation (a more unfavourable modulated/unmodulated ratio, equivalent to a lower signal to noise ratio). To facilitate the interpretation of the results, Figure 6 presents the relative differences of the five output curves relative to the response of the single-speaker modulated noise (S). The results along the ordinate are expressed in dB noise reduction, i.e. the number of dB less gain and output due to noise reduction. Figure 6 shows that the maximum amount of noise reduction is less for the higher frequencies, but that the sensitivity for less pronounced modulation (e.g. at signal to noise ratio=+5) is about equal for all frequencies. For the lower frequencies, noise reduction is activated more gradually than for the higher frequencies as the modulations disappear, but the maximum amount of noise reduction is highest for the lower frequencies. These curves characterise the properties of this specific implementation of noise reduction. At the moment we are running systematic comparisons between different makes and models. The analysis shows important differences between different implementations of noise reduction based on modulations.

## Discussion

It has been argued that the evaluation of complex signal-processing schemes in digital hearing aids needs more advanced test signals that should be standardised to be comparable across studies. ICRA noises can be applied in technical evaluations with coupler measurements and

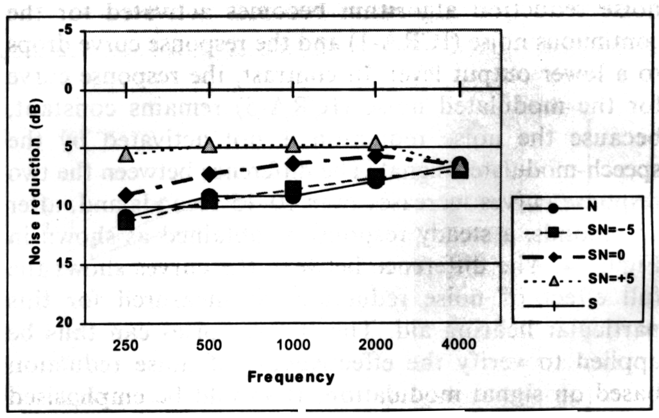


**Figure 6.** Calculated amount of noise reduction (in dB) as a function of frequency, relative to the response for the single-speaker modulated ICRA noise (S, ICRA-5). Curves are presented for the continuous ICRA noise (N, ICRA-1) and for three mixed signals with a "signal to noise ratios" of −5, 0, and +5 (for details, see text).

real-ear measurements as well as in psychophysical evaluations using speech intelligibility tests and comfort measurements.

For *technical evaluation by coupler measurements*, ICRA noises have been demonstrated to be powerful. The noises not only illustrate the effect of new noise reduction techniques, but they can also be used to quantify the effect as a function of frequency. A complication is that the amount of noise reduction may be dependent on the setting of a large number of parameters in the hearing aid under test. This complicates the comparison between aids, although this effect should become smaller if settings are compared that have been selected to fit the same hearing loss. For an objective comparison between aids, it makes sense to define a number of reference hearing losses. Different makes and types of hearing aids can be compared for their manufacturer-proposed target settings belonging to these reference audiometric losses.

Our first measurements indicate that there are at least four dimensions in which currently implemented noise reduction schemes can be compared. A systematic study across hearing aids of different types and makes is needed to make an inventory of the differences and to interpret different results in different studies. This study can be executed with composite ICRA signals, as shown above. Based on the results of such a study, recommendations could be formulated to improve the technical specification of digital hearing aids at this specific point.

The application of the ICRA noises *in real-ear measurements* is even more interesting from the audiological point of view. As with the approach used in the coupler measurements, the amount of noise reduction can be measured as a function of frequency for the individualised setting of the hearing aid. Also, the effects of the earmould can thus be taken into account. Noise reduction is often used to reduce the level of low-frequency sounds. Venting may interfere with the theoretical value of noise reduction being achieved. Insertion-gain measurements with ICRA noises can be used to analyse the net effect of the signal processing hearing aid and the earmould.

On the other hand, the application of ICRA noises in real-ear measurements calls for some extra precautions with respect to the test equipment, the procedure of testing and the test environment.

The basic problem is that the reference microphone has not been designed to deal with modulated signals. In the case of real-ear measurements with the comparison method, the compressor of the equipment will try to follow the modulations. Therefore, the time constants of the compressor should be longer than the relevant modulations in the speech signal. Also, the reference method can be applied, starting with a measurement with an unmodulated noise signal at the reference microphone, followed by the measurement with the modulated noise at the probe microphone. Because this type of measurement is non-simultaneous, the result is more sensitive to head movements and unwanted background noise. Also, one may argue that the test environment should not be too reverberant, because reverberation will smear the modulations. It is important that the hearing aid user is in the direct sound field of the test signal.

Finally, the noises can be applied in *psychophysical assessment* such as speech-in-noise testing. An earlier study by our group[7] showed that minor details in the test conditions can have large effects on the outcome. The objective evaluation of hearing aids with advanced signal processing using long adaptation times can hardly be evaluated by the common objective evaluation procedures used so far. ICRA noises will help to set standards for objective test results under well-controlled conditions, and also in terms of the modulation characteristics. Specific noises can be used to evaluate speech recognition in noise with or without compression and with or without noise reduction. For testing the latter, the noise should be presented continuously or, at least, the noise should start early enough to activate the noise reduction algorithm before the speech material is presented. It is mandatory to take into account the long reaction times of the noise reduction algorithms (usually 10–20 seconds). The characteristics of both the frequency spectrum and the modulation spectrum are carefully under control as long as the ICRA noises are used.

It is important to note that the development of ICRA noises provides a reasonably powerful tool for the current signal-processing schemes. However, not all aspects of the speech signal are imitated by the ICRA noises. ICRA noises have the same long-term spectra as the corresponding ideal speech signals. In addition, they have dynamic properties, which largely represent natural speech. However, the ICRA noises do not have a harmonic structure like vowels and voiced speech sounds. Some future signal-processing techniques may be expected to rely on features not accounted for in the ICRA noises, e.g. on the detection of voiced segments. In such cases, the ICRA noises may not be suitable for assessment.

Reducing the level of uncertainty in current applications of digital hearing instruments is an important concern. With respect to both technical specifications and individual fitting and fine-tuning procedures, the ICRA noise system appears to be a useful tool.

The HACTES (Hearing Aid Clinical Test Environment Standardization) working group under the ICRA continues to consider different environmental noises that can be used for preference measurements and/or the selection of comfort programs in a multi-program device.

## Conclusions

At the 1997 ICRA meeting, the signals described in this paper were recommended by the ICRA for use in hearing instrument assessment and for measurement of frequency responses, especially for hearing instruments that change their frequency response function according to the spectral and temporal properties of the input signal. The present paper contains a detailed description of the generation of this noise and a description of its spectral and temporal characteristics based on control measurements on the noise signal on the CD. (The ICRA CD is free for use with proper reference. In fact the ICRA seeks to stimulate a broad distribution among researchers, clinicians, audiologists, and hearing aid manufacturers. You can order a copy, with a $30 fee for reproduction, postage and handling from the first author: (w.a.dreschler @amc.uva.nl.)

Based on the work presented in this study, the following conclusions can be formulated:

ICRA noises can be used to investigate the amplification characteristics of modern digital hearing aids, including hearing aids with advanced signal processing, such as noise reduction.

2. ICRA noises can also be used to evaluate other properties of hearing aids under realistic conditions, e.g. power consumption and distortion.
3. ICRA noises can be used as background noise in speech recognition tests, in which signals with documented special properties that also vary systematically with modulation are required.
4. There is need for a systematic comparison (and documentation) of noise reduction schemes that are currently implemented in digital hearing instruments.

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