SNR Advantage, FM Advantage and FM Fitting

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Introduction

FM systems for assistive listening applications are becoming increasingly sophisticated in terms of miniaturization, performance and versatility. However, users and professionals alike continue to be confused about many common issues related to daily usage, operation, and clinical fitting procedure. In particular, the gain and signal processing vary among FM systems and are not well-understood by fitters in clinical practice.

Recently, ultra-miniaturized receivers have also become programmable, thereby allowing customized adjustment of FM gain and output to achieve a desired 'FM advantage'. Given this fact, verification tools are needed for measuring the relative strengths of the FM and HI (hearing instrument) Mic signals when used simultaneously in daily life.

This paper presents a practical overview of the functioning of an FM system for fitters in clinical practice, defining commonly used terms such as SNR advantage and FM advantage, and explaining the interaction effects of programming adjustments. A straightforward procedure is proposed for reliably measuring the FM advantage, which overcomes the difficulties of measuring with sequential inputs.

FM System

An FM transmission system (figure 1) consists of two elements, the FM transmitter, also called wireless or remote microphone, and the FM receiver, which provides an audio signal to the audio input of the hearing instrument.

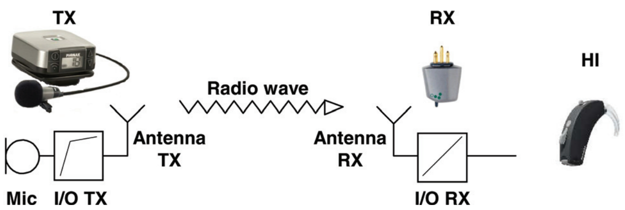

Figure 1. Illustration of an FM system, comprising a transmitter (TX) or remote microphone and a receiver (RX) connected to a hearing instrument (HI).

An FM system replaces the wire between the remote microphone and the hearing instrument's audio input by a radio transmission system.

Audio transfer within the complete FM system would ideally be fully *transparent*, i.e. would not alter the frequency or amplitude characteristics of the audio signal. This transparent signal, in turn, would serve as input to the hearing instrument. Sophisticated signal processing or 'tuning' for an individual hearing loss would be left to the hearing instrument, and omitted from the FM system.

The various commercially available FM systems differ in their ability to fulfill the demand for audio transparency. Phonak's MLx S, with its audio bandwidth of 7.5 kHz and completely flat behavior within this range, can be considered perfectly transparent with respect to audio frequencies.

In terms of the amplitude and signal strength, however, all FM systems are subject to some limitations in transparency, as will be shown in the following section.

Transmitter

Limitations in transparency are inherent to the use of radio systems in general.

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The *radio bandwidth* used by an FM system is strictly limited by telecommunications regulations and particularly may not exceed 25 kHz in the US for assistive listening applications.

As a rule of thumb, the radio bandwidth taken up by FM signal transmission is two times the sum of the *audio bandwidth, or 'frequency response'*, and the *FM deviation* (Δf_{dev},), the parameter encoding signal loudness. If we consider an audio bandwidth of 7.5 kHz, one can easily calculate that the maximum FM deviation must be limited to around 5 kHz (due to radio bandwidth restrictions).

Bandwidth limitation is the reason why each and any FM transmitter available on the market (and carrying an FCC or CE label for having been tested to comply with radio regulations) must use output compression.

Figure 2 shows the I/O characteristics of a Phonak transmitter. Other transmitters may have slightly different characteristics but a generally similar behavior to limit the maximum FM deviation. This is done by compressing signals above a so-called kneepoint. For the case of Phonak transmitters, this kneepoint lies at 72 dB-SPL, values for other manufacturers may vary.

It is important to note that, besides being unavoidable to fulfill radio regulations, the transmitter compression also has a number of beneficial effects for an FM system. It e.g. helps to keep background noise low which may be picked up by the remote microphone and come through together with the useful signal. Furthermore, it helps to keep the FM advantage (see below) relatively constant under normal use conditions.

Another limitation to FM systems is the maximum permitted radiated power of the transmitter, which directly influences the range of an FM system. European radio regulations limit the maximum permitted radiated power for transmitters used with assistive listening systems to 2 mW.

Receiver

Ear-level receivers have become increasingly preferred over conventional body-worn systems due to their improved aesthetics and ease of handling.

With recent technological advances, the output level, or gain of ear-level receivers has become programmable over a wide range, thereby overcoming one of the inconveniences of conventional systems.

The output, or gain level of the MLx S can be software-programmed over a range of 30 dB, thereby enabling individual adjustment of the FM signal strength relative to the HI microphone (FM advantage).

The overall I/O characteristics of an FM system, from the transmitter microphone to the hearing instrument's audio input, is shown in figure 3. The output of the FM system is represented as an SPL-equivalent, which gives the strength of an acoustic signal into an EK microphone (Knowles Electronics)

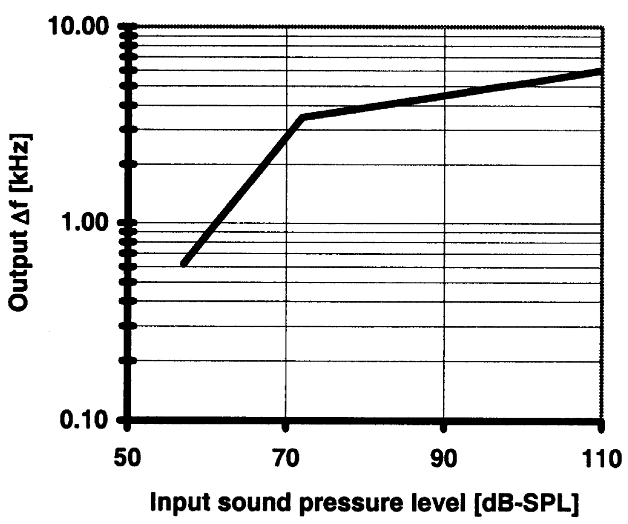

Figure 2. Typical I/O characteristics of a Phonak FM transmitter. Other transmitters have overall similar characteristics, details such as knee points and/or compression ratio may be different.

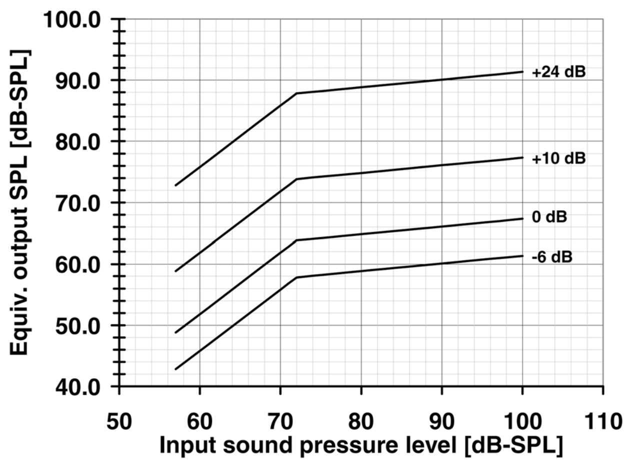

Figure 3. Overall I/O of a Phonak transmitter combined with a Phonak MLx S type receiver.

leading to the same electrical input to the hearing instrument as the FM receiver.

The uppermost curve in figure 3 represents the maximum (+24 dB) and the lowest curve the minimum (-6 dB) programmable output. The audio transfer is linear for input signals below 72 dB-SPL and exhibits a 1:5 compression above this kneepoint. This behavior is determined by the remote microphone / transmitter, as the receiver is completely linear in its I/O characteristics.

When programming the FM advantage into the MLx S receiver, the I/O curve of the entire FM system is shifted vertically within the limits shown in figure 3. This adjustment affects only the gain of the receiver, leaving the transmitter characteristics unchanged.

Hearing Instrument Architecture

The electrical audio input signal from the FM receiver may be processed differently within the hearing instrument, depending on the specific type of HI architecture. Simply stated, there are two common approaches for connecting audio input signals to the hearing instrument's signal processing core. Depending upon which architecture is present within the hearing instrument, different phenomena may influence the FM signal, thereby determining its strength relative to the HI microphone signal.

Hearing instruments with designated programmable audio input (DPAI-HI) differ significantly in their FM signal processing characteristics from hearing instruments without DPAI (non-DPAI-HI). As a rule of thumb, a DPAI is present when the hearing instrument fitting software allows individual adjustment of the FM gain with respect to the gain of the microphone. Technically speaking, a DPAI presents a high-impedance audio input to the HI, in contrast to the low impedance input (typically of the order of 4.4 kΩ.) of non-DPAI instruments.

Non-DPAI Hearing Instruments

In such hearing instruments without DPAI, which namely comprise, but are not limited to, all single-microphone and many older analog or programmable hearing instruments, the FM signal is connected in parallel to the HI microphone. Switching between the FM and the FM+M positions is achieved at the FM receiver. In the FM+M position, both the FM and the MIC signal *add* at the input of the HI (figure 4). In the FM (only) position, the FM receiver squelches the microphone by providing a low output impedance.

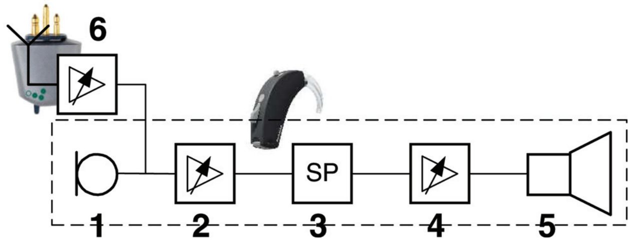

Figure 4. Simplified block schematics of a hearing instrument without direct programmable audio input (non-DPAI-HI). 1: HI Mic, 2: pre-amplifier, 3: signal processing core, 4: amplifier, 5: speaker, 6: FM receiver.

DPAI Hearing Instruments

Hearing instruments with a designated programmable audio input are typically those with advanced two-microphone audio-zoom technology. When connecting an external audio input signal such as FM, typically one of the two microphone inputs is used as the input for the external (FM) audio source.

Where DPAI hearing instruments have a separately adjustable pre-amplifier for each channel (or microphone), the corresponding fitting software of the hearing instrument often permits the independent adjustment or relative balancing of each signal with respect to one another.

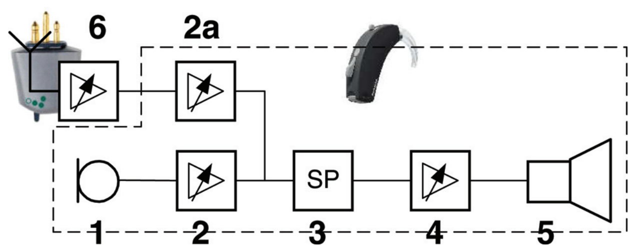

Figure 5. Simplified block schematics of a hearing instrument with direct programmable audio input (DPAI-HI). 1: HI Mic, 2: pre-amplifier (1st Mic), 3: signal processing core, 4: amplifier, 5: speaker, 6: FM receiver connected to pre-amplifier 2a (2nd Mic or audio input).

SNR Advantage

One of the primary purposes of FM systems is to provide signals with dramatically improved signal-to-noise ratio (SNR) for the benefit of the hearing-impaired user. This SNR improvement offered by the FM system counteracts the strong decrease in amplitude of e.g. a speech level with increasing distance from the speaker. There is an energy loss of roughly 6 dB every doubling of distance (a very coarse 0-th order value). More sophisticated studies describe the effects in greater detail (Boothroyd). Other effects, such as reverberation contribute to signal degradation, to result in reduced signal reception, affecting speech recognition abilities.

The *SNR advantage* refers to the benefit in signal-to-noise ratio due to the use of an FM system as compared to the situation without the FM system. The value is derived by the SNR value obtained using the FM signal transmission minus the SNR value which would be obtained without the FM system.

Figure 6 illustrates that the FM advantage can be easily as high as 20 dB when comparing a negative SNR of –5 dB (non-FM condition) for one particular location (3 m distance) and a +15 dB SNR, achieved with FM, both obtained with a speech level of 80 dB-SPL and a background noise level of 65 dB-SPL

The basic objective is to pick up the signal close to its source where it is still much stronger than the background noise and transfer it directly, i.e. maintaining its high SNR level, to the input of the hearing instrument.

The situation becomes even more striking when

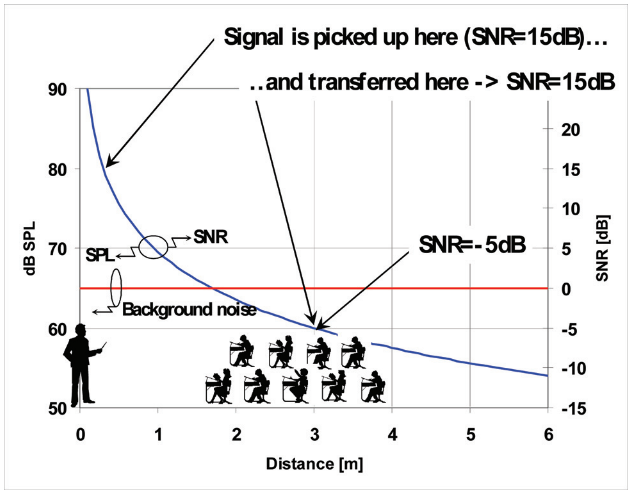

Figure 6. Speech level as a function of distance from the speech source (assuming a simplified 6 dB loss every doubling of distance) and resulting SNR values for a given background noise level of 65 dB-SPL.

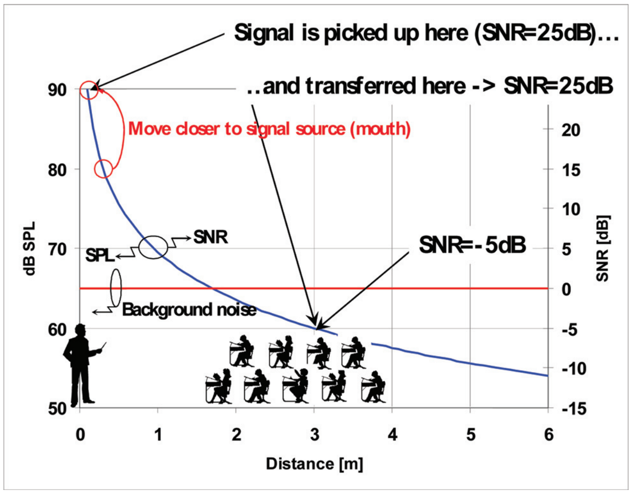

Figure 7. Speech level as a function of distance from the speech source (assuming a simplified 6 dB loss every doubling of distance) and resulting SNR values for a given background noise level of 65 dB-SPL.

one replaces the lapel microphone of figure 6 by a boom microphone, as shown in figure 7.

By picking up the signal even closer to its source, namely the speaker's mouth, one gains another ~10 dB in signal strength, and therefore SNR, against a constant background noise level. In the example of figure 7 where the unaided SNR remains at –5 dB with a 3 m signal distance, the boom microphone achieves a 30 dB SNR advantage.

FM Advantage

The SNR advantage compares the SNR of the FM signal to the SNR of the hearing instrument microphone signal and thus compares the SNR benefit with and without FM. On the other hand, the FM advantage measures the relative loudness of both signals when both, the FM signal and the HI microphone are active *at the same time*. This condition corresponds to the program or setting FM+M.

The FM advantage, as defined by ASHA (American Speech-Language-Hearing Association 2002), compares the strengths of the FM signal and the HI microphone signal for a situation where the speaker (e.g. the teacher) and the user of an FM system (e.g. the student) are two meters away from each other. In this example, the voice of the teacher will travel 30 cm to the input of the FM microphone at a strength of ~80 dB-SPL, whereas only ~65 dB-SPL will remain

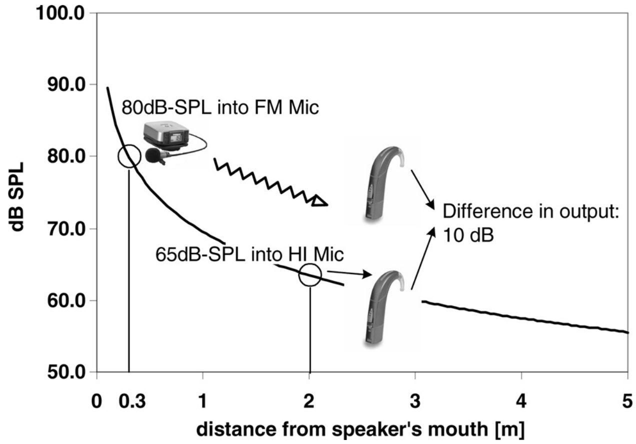

Figure 8. Definition of FM advantage according to ASHA guidelines.

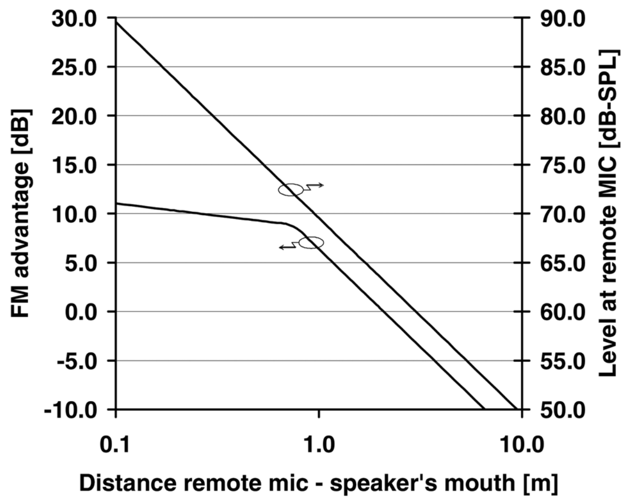

Figure 9. FM advantage (left axis) as a function of the distance remote microphone – speaker's mouth, and speech level into the remote microphone (right axis).

of this original signal after traveling the 2 m distance to the HI microphone (cf. figure 8).

The ASHA guidelines recommend that the FM signal should sound 10 dB louder than the hearing instrument's microphone signal at the output of the user's hearing instrument. In fact, the goal of this recommendation is to obtain the optimum intelligibility of the speech via the FM signal, while remaining connected to the immediate environment via the hearing instrument's microphone.

Influence of Key Parameters

It is important to note that the FM advantage is defined and can be set for one particular condition, namely well-defined signal levels into the remote (FM) and local (HI) microphones. If during utilization of an FM system, the user departs from the particular conditions for which a value of the FM advantage has been pre-determined, the FM advantage experienced by the user will be different than the desired or pre-determined value.

For example, if the remote microphone is moved closer to or further away from the mouth, the speech level at the microphone input will change, as is shown in figure 9. If a 10 dB FM advantage has been adjusted for a distance of 30 cm (per ASHA guidelines, cf. figure 8), the FM advantage will increase with increasing speech level when the FM microphone is moved closer to the mouth and will decrease with decreasing speech level when the FM microphone is moved away from the mouth. Figure 9 shows, however, that the variation in FM advantage is relatively small as long as one stays below a certain distance. As long as the transmitter is within the compressive range (above 72 dB-SPL, cf. figure 3) a change in the input signal has only a relatively small influence on the output. Outside the compressive range (below 72 dB-SPL) the FM advantage linearly scales with the speech level.

Factors and Parameters Influencing the FM Signal Strength

There are a number of factors outside the influence of the FM system, which may result in an FM signal that is stronger or weaker than expected.

General tolerances of FM transmitters and receivers may account for an estimated variation of ±1 dB. Effects relating to the hearing instrument may lead to much larger variations, as is shown in the following sections.

For the case of non-DPAI hearing instruments, there are two major effects influencing the FM signal (figure 10). First, the pre-amplifier at the input of the hearing instrument is commonly used to fine-tune the sensitivity of the HI microphone in production. This leads to a certain offset, which may vary as much as the microphone sensitivity varies. In practice, the specified sensitivity tolerance of a HI microphone is typically of the order of ±2 dB. Such a variation in microphone sensitivity, and its respective

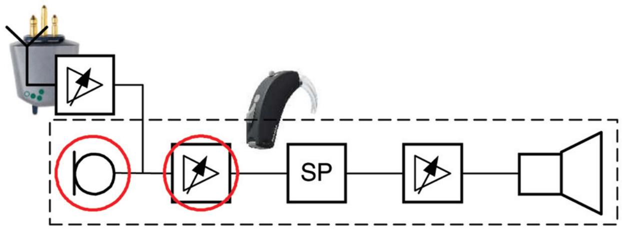

Figure 10. Schematic representation of a non-DPAI-HI, cf. figure 4. The circles indicate blocks which may influence the FM signal strength.

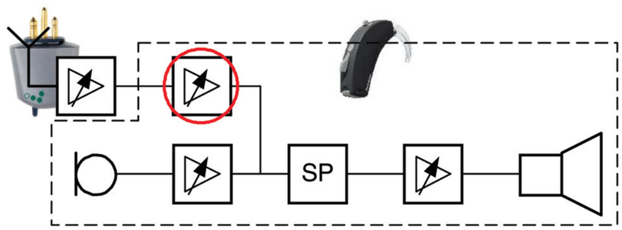

Figure 11. Schematic representation of a DPAI-HI, cf. figure 5. The circles indicate blocks which may influence the FM signal strength.

preamplifier compensation adjustment, would have the same effect in both, the FM and the FM+M positions.

Another effect is related to the impedance of the hearing instrument's microphone in the position FM+M. In fact, the FM signal is connected *in parallel* to the pre-amplifier *and* the microphone. While the amplifier typically has a high input impedance, microphones have a relatively low impedance, typically of the order of 4.4 kΩ. Therefore, the FM signal is loaded with this relatively low impedance. This loading has been taken into account in the design of the FM receiver. A microphone impedance different from the 4.4 kΩ will lead to some variation in the FM signal strength which may be as high as – 6.5 dB / +2.3 dB for the minimum and maximum specified impedance values of e.g. SonionMicrotronic microphones (SonionMicrotronic), as is shown in the following table, x being the difference to the target value due to an impedance mismatch:

Table 1. Difference x between the FM signal level and the target value as a function of the HI microphone impedance Z_{MIC}.

$Z_{MIC} = 2k\Omega$	$Z_{MIC} = 4.4k\Omega$	$Z_{MIC} = 6k\Omega$
x~ – 6.5dB	x~ 0dB	x~ +2.3dB

With DPAI-HI (figure 11) microphone impedance matching is not an issue, because the FM signal is fed *directly* into a high-impedance pre-amplifier. The remaining issue is the use of the pre-amplifier to tune the sensitivity of the microphone, which is connected to this input in normal use. Advanced hearing systems, such as e.g. Phonak Perseo, remove the offset of the pre-amplifier when an external audio signal is connected, but this certainly is not the case for all such hearing instruments.

The possible sources of offset for the FM signal outlined above make it even more important to be able to actually verify the FM advantage. The following section describes some of the difficulties related to this task and proposes the outline of a solution.

The Problem of Measuring FM Advantage with Sequential Inputs

All protocols for the verification of FM systems suggest sequential measurements. This means one first measures the response to a stimulus into the HI microphone and subsequently the response to a stimulus into the remote microphone. Such sequential measurements, however, may be misleading.

Ideally, in order to verify the FM advantage under real conditions, one should measure with *simultaneous inputs* into both the HI microphone and the remote (FM) microphone. Unfortunately, this measurement can be difficult to perform in practice as it is a rather complex (although feasible (Schall and Platz)) task to separate out the MIC and FM signal at the output of the hearing instrument.

It is important to note that the FM advantage should be verified only in program or position FM+M. In fact, although it may be easier to measure the FM response in the FM-only mode in order to prevent background noise from entering the HI microphone, it is absolutely necessary to measure the FM response in position FM+M. As has been shown above, conditions may be different in both cases. Furthermore, programming of the receiver or the hearing instrument may be different in FM+M and FM-only positions.

To illustrate why one may obtain wrong results by measuring with sequential inputs we compare the following two situations.

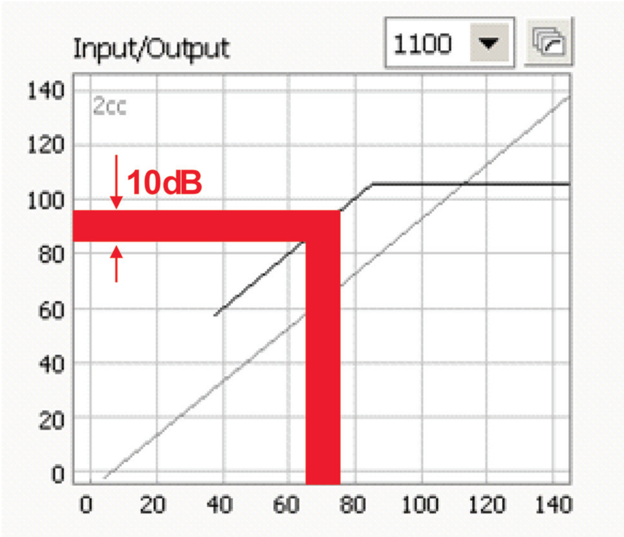

Figure 12. Linear signal processing, e.g. dLimiting.

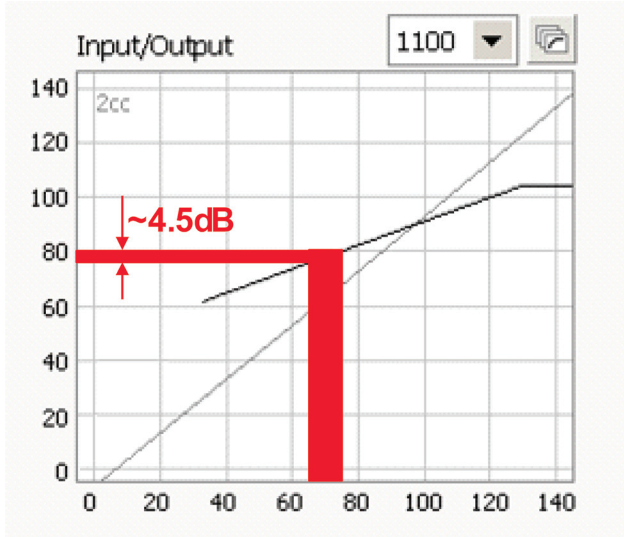

Figure 13. Compressive signal processing, e.g. dWDRC.

The first example of figure 12 illustrates the measurement of the FM system using sequential inputs. In this example, the response to a 65 dB-SPL signal into the HI microphone is an ~85 dB-SPL signal at the output, as a result of the HI gain. An 80 dB-SPL signal into the remote microphone results in a 75 dB-SPL equivalent output of the FM system if an FM advantage of 10 dB has been programmed (cf. figure 3). The response of the hearing instrument to a 75 dB-SPL stimulus is a ~95 dB-SPL signal at the output of the HI, i.e. 10 dB more than for the HI signal and therefore a 10 dB FM advantage.

This observation is accurate when the hearing instrument uses linear signal processing around 65–75 dB input levels. In such cases sequential measurements will lead to correct results.

The situation is different when non-linear signal processing is active within the hearing instrument, as shown in figure 13. The same input stimuli will lead to an apparent FM advantage of only ~4.5 dB, i.e. seemingly much too low.

In reality however, with simultaneous inputs, the FM signal and the microphone signal are added and compressed together. Therefore, the FM advantage actually experienced by the user corresponds to the *relative strengths of both signals* at the input of the hearing instrument.

In conclusion, one can say that it is rather an exception if a correct value for the FM advantage is obtained with sequential measurements.

A Proposal for Measuring FM Advantage with Sequential Inputs

Sequential inputs may be used for measuring the FM advantage if FM and MIC signal experience no or, else, the same amount of compression. Quite generally this is only true if both signals are of the same magnitude.

Based on this fact we suggest the following procedure to measure FM advantage using sequential inputs.

- Measure the response to a 65 dB-SPL signal into the HI microphone.
- Program 0 dB FM advantage into the FM receiver.
- Measure the response to a 80 dB-SPL signal into the FM microphone. The 80 dB come down to 75 dB due to the transmitter compression, the 75 dB signal result in a 65 dB equivalent output at the MLx S (cf. figure 3).
- Subtract both responses. The ideal case would be a 0 dB difference, any value different from 0 dB is an offset and has to be taken into account in order to obtain a desired FM advantage.
- Finally, program the desired FM advantage, corrected for the offset determined using the proposed procedure, into the receiver.

As an example, if an offset of +2 dB is determined using the procedure outlined above, one should program +8 dB FM advantage in order to obtain an effective FM advantage of +10 dB.

Summary

FM transmission systems for assistive listening applications are subject to limitations concerning the maximum radiated power on the one hand and transparency with respect to the signal loudness on the other hand.

The SNR advantage compares the SNR of the FM signal to the SNR of the HI signal, whereas the FM advantage measures the relative loudness of both signals when both, the FM signal and the HI microphone are active at the same time.

We have pointed out a few factors outside the influence of the FM system, which may result in an FM signal that is stronger or weaker than expected. DPAI-HI and non-DPAI-HI connect the FM signal differently to the HI signal processing core, leading to different possible effects influencing the FM signal strength. This underlines the necessity for actually verifying the FM advantage.

It has been shown that measuring the FM advantage with sequential inputs may be misleading. Therefore, we have proposed a procedure for reliably measuring the FM advantage using sequential inputs and thus avoiding artifacts due to different compression of signals of different strengths.

Acknowledgements

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References

Knowles Electronics, Data sheet EK series microphones.

Boothroyd, A. this volume.

American Speech-Language-Hearing Association. (2002). Guidelines for fitting and monitoring FM systems. ASHA desk reference.

SonionMicrotronic, Technical Data Sheet 1M024.

Schall, M., and Platz, R. to be published.

Participant Survey

An FM system has the following transfer characteristics

A. Perfectly linear
B. Compressive over the whole input range
C. Linear region followed by a compressive region above a certain threshold (kneepoint)