# Classifying automatic signal processors

A classification system for level dependent frequency response automatic signal processing circuits is proposed.

**By Mead C. Killion, PhD, Wayne J. Staab, PhD, and David A. Preves, PhD**

**Automatic signal processing, often abbreviated as ASP, can take many forms. The purpose of this short article is to distinguish among some of the newer forms and to suggest a systematic classification to describe them for the sake of ease and clarity of communication.**

Circuits that automatically change not only the gain but also the frequency response of the hearing instrument as a function of the input signal have been introduced in the last decade. In contrast, the traditional automatic signal processing circuits, i.e., the automatic gain control (AGC) or compression circuits, reduce gain at high levels (compression limiting) and/or increase gain at low levels (wide dynamic range compression) but do not change the frequency response of the hearing aid in the process. Well-defined terms for fixed frequency response (FFR) automatic signal processing circuits already exist, as indicated above (Fig. 1 - left side).

This article is primarily about level dependent frequency response (LDFR) automatic signal processing circuits (Fig. 1 - right side). Three examples describing these circuits will be given. The circuits most often called ASP reduce the low frequency gain in response to high level inputs. The Manhattan™ circuit[4] operates in this fashion. The opposite circuit behavior is to reduce the high frequency gain in response to high level inputs. The K-AMP™ circuit[2] operates in this fashion. Finally, the two-channel, programmable compression ReSound circuit[6,7] is an example of a circuit that can be adjusted to exhibit either type of behavior. All of these different LDFR behaviors are called automatic signal processing, but there are no simple, rigorous terms to describe each one.

The authors decided to distinguish among these circuits in terms of their reaction to low-level rather than high-level inputs because the former yielded easy-to-remember acronyms. However, for those who have difficulty in thinking about performance at low levels, the corresponding performance at high levels is indicated as well. The authors propose the following classification for level dependent frequency response automatic signal processing circuits.

*Killion is president of Etymotic Research, Elk Grove Village, IL. Staab is a private consultant in Phoenix, AZ. Preves is vice president of research and development for Argosy Electronics, Eden Prairie, MN.*

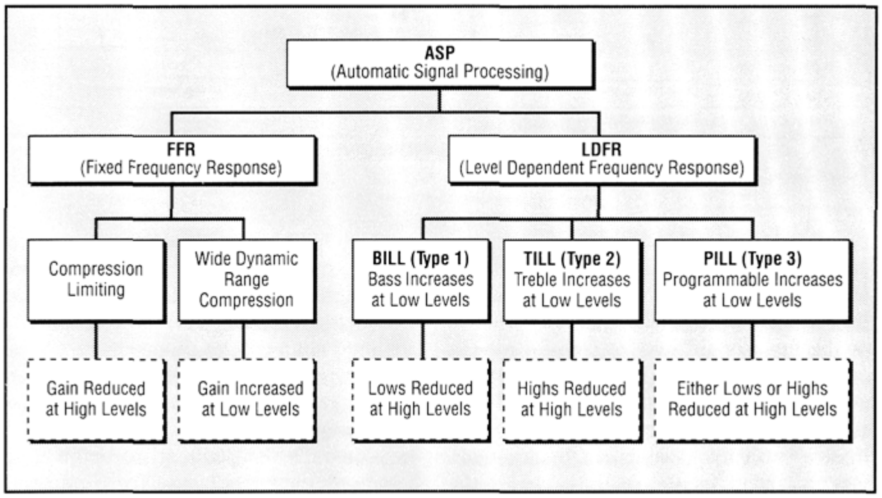


***Fig. 1. Outline of recommended classification system for ASP type instruments.***

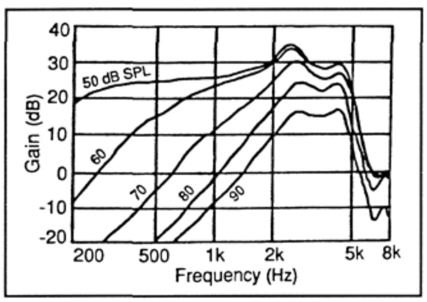


***Fig. 2. BILL (Type 1) LDFR showing reduced low frequency gain as input level increases. Graph shows 2 cc coupler measurement using speech-shaped noise input.***

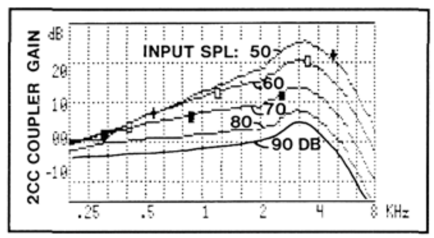


***Fig. 3. TILL (Type 2) LDFR showing reduced high frequency gain as input level increases. Graph shows 2 cc coupler measurement using speech-shaped noise input.***

## The types of LDFR

**BILL (TYPE 1)**—Bass Increases at Low Levels (bass decreases at high levels). This type of ASP describes the operation of circuits such as the Manhattan circuit which provide relatively more bass response for low level inputs than for high level inputs (Fig. 2).

**TILL (TYPE 2)**—Treble Increases at Low Levels (treble decreases at high levels). This type of ASP describes the operation of circuits such as the K-AMP™ circuit, which provide relatively more treble response for low level inputs than for high level inputs (Fig. 3).

**PILL (TYPE 3)**—Programmable Increases at Low Levels. This type of ASP describes the operation of circuits such as the ReSound circuit, which provide programmable level dependent frequency response modification, and can be adjusted to provide either bass response decreases with increasing level or treble response decreases with increasing level (i.e., either BILL or TILL) type of behavior (Fig. 4).

## Which for what?

The basic motivation for developing an ASP circuit is that it can perform a task for wearers that they would otherwise need to perform for themselves on a regular basis, so regular as to be inconvenient, annoying or perhaps even impossible. Adjusting the volume control is such a task, leading to AGC circuits. Simultaneously adjusting a tone control is a second such task, leading to LDFR circuits.

The **BILL (Type 1)** LDFR is intended for wearers who frequently find themselves in noisy environments, especially environments where low frequency noise

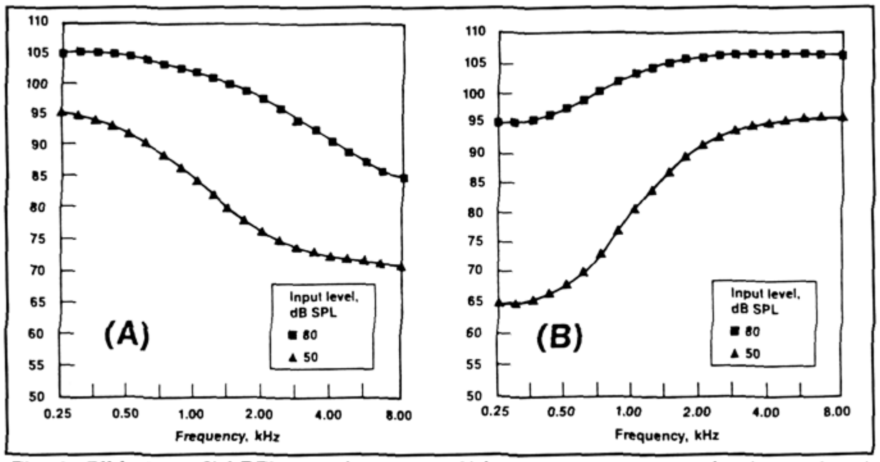


*Fig. 4. PILL (Type 3) LDFR showing either A) low frequency gain reduction as input level increases or B) high frequency gain reduction as input level increases. (Waldhauer and Villchur[7] with permission).*

predominates. Under those circumstances, there is both anecdotal evidence and some research evidence to indicate that reducing the low frequency response of the hearing aid may be helpful[1]. If the overall gain of the instrument is also automatically reduced for high level inputs so as to prevent overload distortion, a further improvement is obtained[3]. Recent indications are that one of the important reasons hearing aid wearers have difficulty hearing in high noise levels is because of the distortion introduced by the hearing instrument itself[5].

The **TILL (Type 2)** LDFR is intended for wearers with high frequency hearing loss, who typically need more high frequency gain for quiet sounds than they do for loud sounds. An amount of high frequency gain that might produce a harsh or shrill sound in a linear hearing aid may become quite acceptable if the treble boost is automatically reduced for high level inputs. If the overall gain is also automatically reduced for high level inputs, it is possible to prevent audible distortion under nearly all listening conditions.

**PILL (Type 3)** LDFR is the most versatile. Not only can it be adjusted to exhibit either BILL or TILL type processing, but each of its independent processing channels can ignore strong interference in another channel.

The acceptability of any ASP circuit depends on the success of the designer in choosing appropriate time constants for the automatic operation(s), maintaining low distortion in the gain control and/or tone control circuit elements and choosing appropriate input-output operating characteristics. Use of the present terms should permit clearer discussions of the latter choices. □

## References

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*Address further inquiries to: Mead C. Killion, PhD, Etymotic Research, 61 Martin Lane, Elk Grove Village, IL 60007.*