ACOUSTIC SHOCK DETECTION AND CONTROL

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1. INTRODUCTION

Acoustic transient noise or impulse noise is referred as acoustic shock and defined as a sudden increase of sound pressure level within a very short time [1]. Such sudden signal level increases can result from interfering noises such as dishes breaking, hammering, a door slam, a gun shot, or any other kind of impulse noise. Acoustic shock is a well-known problem in public address systems, handsets and other telecom products as well as hearing aids. The problem of acoustic shock can be much worse for the user of hearing aids since it may cause more discomfort or further hearing damage if not handled properly by the hearing instrument [2].

Many different approaches have been developed to address the detrimental effects of such acoustic shocks. MPO (Maximum Power Output) in the frequency domain can be applied to prevent overshooting, but it is too slow to be effective. Peak-clipping in the time-domain is effective and fast, but it usually causes serious distortion of sound quality.

A sub-band-based acoustic shock algorithm has been presented in a patent application [3]. The shock detection module in the frequency domain detects if shock has occurred in a particular sub-band, and determines the sub-band energy measurement to be used for the gain calculation. The acoustic shock phenomenon is eliminated by applying the appropriate gain reduction to the signal in each sub-band. The additional time-delay is also required for the system to work.

In some shock detections, high pass filters are used since the transient noise has most of the energy at high frequencies. Low-pass filters are also often used to attenuate the transient noise without significantly affecting speech content of the signal [4]. The difficulty for the shock detection and control based on filters are to reliably separate speech from the transient noises without side effect.

What is desired is a method that can detect acoustic shock reliably and quickly with minimum computational cost, and can adaptively attenuate or cancel the shock while, at the same time, maintaining the input signal quality unaffected. This paper presents a new technology referred as AntiShock which consists of smart shock detection and adaptive shock control. AntiShock technology has been successfully applied in new generation of digital hearing aids.

2. METHOD

2.1. Acoustic Shock Definition

If a shock has a peak level of L(dB) at t1, duration T=t2-t0 where t0 is defined as the starting point of shock and t2 is defined as the half-way point between peak level and the signal floor, it looks as shown in Fig. 1.

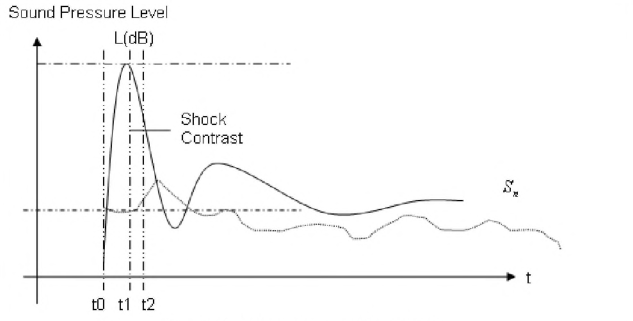

Fig. 1: A typical shock and its impact

The shock contrast level (dB) is $dL = L - S_n$, where $S_n = 10\log[\frac{1}{t1 - t0}\int_{t0}^{t1}|s(t)|\,dt]$ is the fast averaging s(t) over a short duration of about 2ms so that it can reflect the normal speech signal or the music signal change over time. The higher the shock peak level, the stronger the shock will be perceived. For the same source of shock, the perceived shock strength depends on the actual contrast level dL. With the same shock peak level, the lower the signal floor, the higher the shock contrast that will result in a stronger shock perception. Also, the longer the duration of the shock, the stronger the shock will be perceived. Therefore, it is critical to detect shock contrast in order to determine the level of shock impact and the necessary solution in controlling the shock with an appropriate anti-shock strategy.

2.2. Smart Shock Detection

The absolute shock impact level (SIL) is $SIL = \frac{L}{t_1 - t_0}$ and the shock energy strength (SES) is $SES = \frac{L}{t_1 - t_0} * (t_2 - t_0)$, which means the higher L, the shorter (t1-t0) and the longer (t2-t0) are, and the stronger the shock energy strength.

For shock detection, the following variables are included:

- Signal floor S_n (dB), which varies per acoustical environment along time;
- Peak level L(dB);
- Shock setup time (t1-t0);
- Shock duration (t2-t0)

The shock contrast level (dL) is defined as $dL = L - S_n$ and the relative shock impact level can be expressed as the Shock Index: $S\_Index = \alpha \frac{L - S_n}{t_1 - t_0}$, where α is the coefficient for Shock Index normalization. The shock index normalization constant α can be defined according to the individual's preference. In one of our preferable systems, we define α by referring to a typical dish transient noise with a shock level L=70dB in quiet ($S_n = 40\ dB$) with 0.2ms attack time and 20ms duration. Then we get $\alpha = 0.2\ ms / 30\ dB = 0.0067\ (ms / dB)$ to have $S\_Index = 1$ as normalized Shock Index. For this same dish noise, if the environment signal level increases, the shock Index will drop.

Two thresholds, minimum shock contrast level and minimum shock Index, are used for shock detection. These thresholds can be determined through a self-learning process or pre-determined measurement so that daily life non-transient signals such as speech, music, normal acoustic sound are not detected as shock, and that a transient sound such as a gun shot or a door slam will be detected as shock. A stronger and sharper shock will generate a stronger Shock Index. The duration of shock, T=t2-t0, will be used together with Shock Index as the measurement of shock strength, which is used for shock control.

The shock detection runs in real-time with the use of the thresholds of minimum shock contrast level and minimum Shock Index. According to the shock contrast level and Shock Index, it can decide whether a shock happens and how strong the shock is. Since the shock detection runs continuously, the shock can be detected anytime as long as it meets with the shock detection criteria.

2.3. Adaptive Shock Control

The objectives of shock control are:

- To reduce or minimize the shock effect;
- To keep the shock sound as natural as possible to allow awareness by the user of the type of shock event;
- To keep the relative loudness of shock so that the user can perceive the shock level;
- To keep the shock within the comfort range of the user.

The shock control will apply Gain Reduction g(t) adaptively to the input signal s(t) to get a new signal x(t). As one typical application, we may use a fast shock attack and a slow anti-shock release for the peak shock duration ($t \in (t1, t2)$) so that the shock can be efficiently controlled. In another typical application, we may need to have fast anti-shock release for the peak shock duration ($t \in (t1, t2)$), so that the useful signal is less affected. In other applications, we may want to use the same anti-shock attack and release speed in order to simplify the design.

3. RESULTS

An effective technology for acoustic shock detection and control has been invented for the new generation of digital hearing aids. In one evaluation, 30 hearing aid wearers provide preferences for AntiShock On/Off: Speech, Transients Speech + Transients. As shown in Fig. 2, the clear preference exists for the "AntiShock On" than "AntiShock Off" for acoustic shocks and there is no obvious preference for AntiShock On/Off since AntiShock has not affected any speech context.

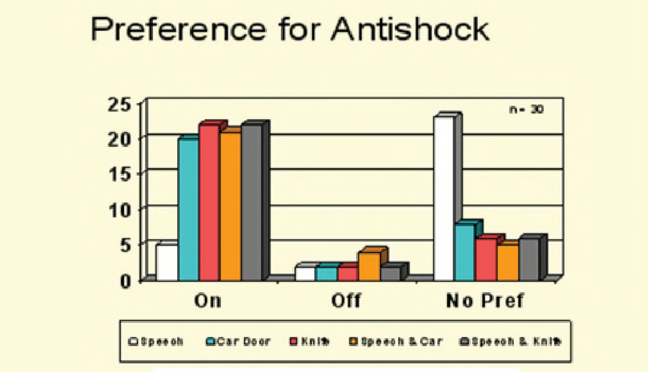

Fig.2: AntiShock Evaluation

While reducing acoustic shock adaptively, AntiShock technology keeps the natural sound quality of shock events for environmental awareness and not hampers the user's safety. AntiShock technology is capable of detecting and controlling acoustic shocks in an optimized way to cancel shock completely or keep shock in comfortable level according to applications without affecting non-shock signals such as speech and other acoustic sounds.

REFERENCES

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