Multiband Speaker Protection and Dynamic Excursion Control of Audio Devices

Anonymous
Multiband Speaker Protection and Dynamic Excursion Control of Audio Devices

ABSTRACT

If the excursion limit of the diaphragm of a speaker is exceeded, the speaker exhibits nonlinear behavior which manifests as distorted sound and degraded acoustic echo cancelation (AEC) performance. Moreover, if a speaker is driven too hard, the resulting excessive heat can damage the voice-coil and speaker. While the aforementioned problems are present in speakers of all sizes, these become more acute for smaller, portable speakers, e.g., as used in smartphones, smart speakers, micro-speaker, wearable devices, etc.

This disclosure describes techniques to simultaneously protect the speaker and achieve an optimum, multi-band audio experience by modeling the speaker, dynamically adjusting the excursion, and driving the speaker to its true limit while avoiding non-linear distortion. Per the techniques, real-time variations in the speaker are directly sensed, and feedforward and feedback control procedures are applied to deliver an optimum sound performance while maintaining safe speaker operating conditions.

KEYWORDS

- Speaker modeling
- Speaker protection
- Micro-speaker
- Equalization filter
- Acoustic Echo Cancelation (AEC)
- Excursion modeling
- Excursion control
- IV-sense
BACKGROUND

A speaker is an electro-acoustic transducer that generates sound from an electric signal produced by a power amplifier. As shown in Fig. 1, the voice-coil of a speaker is attached to a diaphragm (or membrane) that is mounted on a fixed frame via a suspension. A magnetic field is generated by a permanent magnet that is conducted to the region of the coil gap.

Due to the presence of the magnetic field, an electrical current passing through the voice-coil generates a force $f_c$ which causes the membrane to move up and down. The displacement $x_d$ of the membrane is the excursion, which has a limit (or bound). If the excursion limit is exceeded, the speaker exhibits nonlinear behavior, which in turn manifests as distorted sound and degraded acoustic echo cancelation (AEC) performance. Moreover, as current in a speaker is pushed through the voice-coil, some of the electrical energy is converted into heat instead of...
sound. If the speaker is driven too hard, the resulting excessive heat can damage the voice-coil and speaker.

While the aforementioned problems are present in speakers of all sizes, they become more acute as speakers become smaller and more portable, e.g., as used in smartphones, smart speakers, micro-speaker, wearable devices, earbuds, PDAs, etc. Additionally, in such devices, the small size prevents bass frequencies (low frequencies) from presenting at the output, further reducing the overall audio performance and loudness, and resulting in nonlinear distortion that significantly degrades the AEC performance. As mentioned before, a root cause is excessive membrane excursion and voice-coil heating during speaker operation.

Traditional equalization (EQ) filters used to control membrane excursion are designed conservatively due to the wide range of operating factors, e.g., speaker variations, operating conditions, various types of audio signals with large dynamic ranges, etc. These conservative approaches do not push the speaker to its true limit. On the other hand, simultaneously protecting the speaker (by controlling the excursion) and achieving an optimum audio listening experience (by maximizing loudness and minimizing nonlinearity) is a difficult problem that is of wide importance to the industry. EQ filters also have the undesirable property that at low audio-signal energies and within EQ attenuation bands (which is when the excursion is within limits), the filters still attenuate the output audio, thereby actually contributing to the degradation of audio performance and loudness.

**DESCRIPTION**

This disclosure describes techniques to simultaneously protect a speaker and achieve an optimum, multi-band audio experience by modeling the speaker, dynamically adjusting the excursion, and driving the speaker to its true limit while avoiding distortion. Per the techniques, a
feedforward control procedure is operative if the current (I) and voltage (V) sense (collectively, IV-sense) is unavailable in the digital-to-analog converter (DAC) of the speaker. A feedback control procedure is operative for the case where IV-sense is available in the DAC, with the option of also applying the feedforward control procedure for this case. By directly sensing real-time variations in the speaker, the techniques enable the application of real-time procedures that deliver an optimum sound performance while maintaining safe speaker operating conditions. The techniques work regardless of the size of the speaker, e.g., from micro-speaker (thin, electrodynamic speakers) to large traditional speakers.

The key parameters in designing multiband excursion control and speaker protection are processing power (e.g., computational and memory complexity) and accuracy of dynamic excursion estimation and control.

![Multi-band excursion control and protection](image)

**Fig. 2: Multi-band excursion control and protection in the absence of IV-sense (feedforward approach)**
Fig. 3: Multi-band excursion control and protection if the speaker has IV-sense (feedback approach)

Fig. 2 illustrates multiband speaker protection and excursion control if IV-sense is unavailable, and Fig. 3 illustrates the same if IV-sense is available, e.g., via the DAC of the speaker. In both cases, the audio-in signal is adjusted dynamically such that the audio-out signal provided to the speaker is within the excursion limit of the speaker.

Fig. 4: A three-band filter bank scheme ($0 < f_{c2} < f_{c1} < f_s/2$)
In both cases (IV-sense present and absent), a filter bank splits the audio signal from a full-band into $M$ sub-bands, where $M$ can be, e.g., 2, 3, 4, etc. The filter bank is designed to reconstruct the signal without latency. In the three-band filter bank ($M = 3$) illustrated in Fig. 4, the low-pass (LPF), the high-pass (HPF), and the all-pass (APF) filters are of Butterworth type, with crossover frequencies denoted by $f_{c1}$ and $f_{c2}$, and sampling frequency denoted $f_s$. The sampling frequency can be, e.g., 32 kHz or 48 kHz. The APF is designed to match phase responses between all bands such that the outputs of each band are in-phase. The values of these crossover frequencies are related to speakers as follows. If the driver resonance frequency is $f_0$
Hz (e.g., 260 Hz), then the crossover frequencies satisfy the requirement of $0 < f_{c2} < f_0 < f_{c1} < f_s/2$.

Fig. 5 illustrates an example of the magnitude responses of a three-band filter bank for the case $f_{c1} = 1000$ Hz and $f_{c2} = 120$ Hz. It is seen from Fig. 5 that the total magnitude response of the filter bank is a flat, zero dB, e.g., the filter bank reconstructs the signal perfectly.

As mentioned earlier, the excursion control modules in Figs. 2 and 3 are respectively feedforward control (without IV-sense) and feedback control (with IV-sense). If the speaker has no IV-sense, excursion modeling is based on fundamental speaker parameters, and the excursion control is of type feedforward. If the speaker has IV-sense, the excursion modeling is dynamically adjusted on the basis of the current and voltage provided by the IV-sense, and the excursion control is of type feedback.

**Fig. 6: Excursion control**

Fig. 6 illustrates the excursion control procedure in detail. The excursion model in Fig. 6 is implemented by an infinite impulse response (IIR) filter. A speaker typically has five fundamental parameters, which can be estimated by using measurements or by using the speaker’s output current and voltage as provided by IV-sense. These fundamental parameters are as follows:

1. $R_{ms}$: The mechanical resistance of a driver's suspension, also known as mechanical damping factor, in units of N-s/m.
2. \( M_{ms} \): The moving mass of the diaphragm and coil, including the acoustic load, in units of grams.

3. \( C_{ms} \): Compliance of the suspension of the driver, also known as mechanical compliance, in units of mm/N.

4. \( BL \): Electromotive transformer, also known as force factor, in units of T-m, which is a measure of the speed at which the diaphragm and coil (measured by \( M_{ms} \)) move.

5. \( R_e \): The DC resistance of the voice coil, in units of Ohms.

**Excursion model if IV-sense is unavailable**

The excursion model and its impedance are derived from the above five parameters and can be implemented using the below Equations 1 and 2 respectively. Their parameters can be estimated in advance through speaker measurements.

\[
Z_{excu}(s) = \frac{BL}{M_{ms}R_e} s^2 + \frac{s}{\omega_0 Q_{ms}} s + \omega_0^2
\]

(1)

\[
Z_{imep}(s) = R_e + Z_{BEMF}(s) = R_e + \frac{(BL)^2}{M_{ms}} s^2 + \frac{s}{\omega_0 W_{ms}} s + \omega_0^2
\]

(2)

In the above equations, \( Z_{BEMF}(s) \) is the impedance from the back electromotive force (BEMF).

The driver resonance frequency \( f_0 \) and the mechanical quality factor \( Q_{ms} \) are given by Equations 3 and 4 respectively:

\[
f_0 = \frac{\omega_0}{2\pi} = \frac{1}{2\pi \sqrt{M_{ms}C_{ms}}}
\]

(3)

\[
Q_{ms} = \frac{\sqrt{M_{ms}}}{R_{ms}\sqrt{C_{ms}}}
\]

(4)

A reliable excursion model is built by measuring several, e.g., five speakers for each speaker model. Speaker impedance curves can be obtained, for example, by sweeping the five speakers with pink noise and averaging the results. The impedance model resulting from the
measurements is of the form shown below (Equation 5), where coefficients $a$, $g$, $b$, and $c$ are obtained from the above-mentioned sweep measurements.

$$H(s) = a + g\frac{s}{s^2 + bs + c}$$  \hspace{1cm} (5)

Comparing Equations 2 and 5, the estimated fundamental speaker parameters are obtained as follows.

$$R_e = a$$  \hspace{1cm} (6)

$$M_{ms} = \frac{(BL)^2}{g}$$  \hspace{1cm} (7)

$$C_{ms} = \frac{g}{c(BL)^2}$$  \hspace{1cm} (8)

$$R_{ms} = b\frac{(BL)^2}{g}$$  \hspace{1cm} (9)

Here, the force factor $BL$ can be obtained either from the speaker vendor or from laser measurements. In this manner, in the absence of IV-sense (Fig. 2), the excursion model can be established in advance via Equation 1 and can be applied to the excursion control scheme (Fig. 6).

**Excursion model if the speaker has IV-sense**

Denoting the frequency-domain output current and voltage provided by the IV-sense as $I(s)$ and $V(s)$ respectively, the actual voltage from the BEMF is given by

$$V_{BEMF}(s) = V(s) - I(s)R_e.$$  \hspace{1cm} (10)

The estimated voltage from the BEMF is

$$V'_{BEMF}(s) = I(s)Z_{BEMF}(s).$$  \hspace{1cm} (11)

Therefore the estimation error $E(s)$ is
The time-domain estimation error is given by
\[ E(s) = V_{BEMF}'(s) - V_{BEMF}(s) = I(s)[Z_{BEMF}(s) + R_e] - V(s). \] (12)

The time-domain estimation error is given by
\[ e(t) = i(t) * z_{BEMF}(t) + i(t)R_e - v(t). \] (13)

where \(i(t)\) and \(v(t)\) are respectively the current and voltage outputs of the speaker in the time domain.

The voice coil DC resistance \(R_e\) can be measured in advance by playing a pilot tone of low frequency, e.g., 55 Hz, to the speaker. The actual voltage from the BEMF is almost zero in Equation 10 for the pilot tone. Therefore, the voice coil DC resistance \(R_e\) is estimated as follows.
\[ R_e = \frac{v(t)}{i(t)}. \] (14)

A reliable excursion model is built by using several, e.g., five speakers for each speaker model for this measurement. Minimizing the estimation error \(e(t)\) in Equation 13, by taking the partial derivatives of \(e(t)\) with respect to the rest of the fundamental speaker parameters and setting them to zero, results in the determination of the rest of the fundamental speaker parameters. The excursion model can be dynamically established via Equation 1 and can be applied to the excursion control scheme (Fig. 6) for the case when the speaker provides IV-sense (Fig. 3).
The excursion model having been established, the peak estimator (Fig. 6) is implemented by taking the absolute value of its input audio signal. The excursion compressor (Fig. 6) can be implemented using the statistical curve illustrated in Fig. 7. The compression curve is defined by compression parameters, e.g., \(\text{threshold}_1\) (e.g., \(\text{threshold}_1 = 0.80 \times \text{max excursion}\)), \(\text{threshold}_2\) (e.g., \(\text{threshold}_2 = 0.95 \times \text{max excursion}\)), and compression ratios \(CR_1\), \(CR_2\) and \(CR_3\), subject to the condition \(1 = CR_1 \leq CR_2 \leq CR_3\). The compression ratio, \(CR\), is defined as a ratio between the input level difference and the output level difference as follows.

\[
CR = \frac{In_1(dB) - In_2(dB)}{Out_1(dB) - Out_2(dB)} \tag{15}
\]

The compression parameters \(\text{threshold}_1\), \(\text{threshold}_2\), \(CR_2\), and \(CR_3\) are adjustable flexibly and independently among bands to precisely reduce band level only when audio dynamics in the band exceed the corresponding excursion thresholds, precisely reduce peak-to-average ratio (PAR) over frequency bands to maximize bass and loudness, and precisely reduce speaker total-harmonic-distortion (THD).
The output of the excursion compressor (Fig. 6) is an intermediate adaptive gain. The intermediate adaptive gain \( p(k) \) in the \( k \)th band can be calculated as illustrated in Fig. 8 by using the input level \( L(k) \) (in dB) and the compression curve (Fig. 7).

If \( L(k) < \text{threshold}_1 \), then follow linear processing: \( p(k) = 0 \, \text{dB} \).

If \( \text{threshold}_1 \leq L(k) < \text{threshold}_2 \), then set \( p(k) \) as follows.

\[
p(k) = \left(1 - \frac{1}{CR_1}\right) \times (\text{threshold}_1 - L(k))
\]  

(16)

If \( L(k) \geq \text{threshold}_2 \), then set \( p(k) \) as follows.

\[
p(k) = \left(1 - \frac{1}{CR_2}\right) \times (\text{threshold}_2 - L(k))
\]  

(17)

**Fig. 8: Setting the intermediate adaptive gain for the \( k \)th band**

The linear adaptive gain \( q(k) \) can be obtained as follows.

\[
q(k) = 10^{p(k)/20.0}
\]  

(18)

The gain smoother block (Fig. 6) reduces the variation of the adaptive gain. Multiplying the delayed block of input signals by the smoothed linear gain results in a well-controlled, smooth output. It is to be noted that all the bands use the same delay values in the number of audio samples.

In Figs. 2 and 3, after the summation operation across all the bands, a full-band output audio signal is obtained. A peak limiter is deployed to prevent the system output from clipping, thereby minimizing artifacts. The audio-in signal is dynamically adjusted such that the audio-out signal provided to the speaker is within the excursion limit of the speaker.
The techniques of multiband excursion control and protection described herein produce richer and more robust sound without damaging the speaker from over-excursion. Some advantages of the described techniques include:

1. Excursion tracked even if IV-sense is unavailable.
2. Look-ahead feature provided in excursion compression.
3. The production of a rich sound and balanced sonic experience with good audio quality.
4. Workability for any number of channels and any frame size.
5. Workability at any sampling rate, e.g., 48 kHz, 44.1 kHz, 32 kHz, 24 kHz, 22.05 kHz, 16 kHz, 11.025 kHz, 8 kHz, etc.
6. Improvement of not only bass frequencies but also loudness under the maximum volumes.
7. Applicable to speakers of various kinds, e.g., of different sizes and models.

In this manner, the techniques of this disclosure achieve a cost-effective and robust multiband (including bass) excursion control and protection of a speaker while incurring minimum non-linear distortion and dynamically driving the speaker to its limit (bound). The techniques dynamically estimate the excursion by modeling the speaker, and dynamically compress the audio at bands that exceed the excursion limit. The techniques enhance audio performance and loudness, and, by minimizing nonlinear echo, also enhance AEC performance.

CONCLUSION

This disclosure describes techniques to simultaneously protect the speaker and achieve an optimum, multi-band audio experience by modeling the speaker, dynamically adjusting the excursion, and driving the speaker to its true limit while avoiding non-linear distortion. Per the techniques, real-time variations in the speaker are directly sensed, and feedforward and feedback
control procedures are applied to deliver an optimum sound performance while maintaining safe speaker operating conditions.

REFERENCES