1 # CHAPTER 7

DISTORTION

2 ## Preamble

- **Deliver sound wave (or electrical signal) to a system**
- **If the system reproduces the waveform <u>faithfully</u>, the signal is <u>undistorted</u>**
- **If the shape of the waveform is <u>altered</u>, the signal is <u>distorted</u>**

3 ## Three Types of Distortion

- **Frequency**
- **Transient**
- **Amplitude**

4 ## FREQUENCY DISTORTION

- **Panel A: Measurement of amplitude response of a system**
  - **The input signal must have equal amplitude for all frequencies delivered to the system**
- **Panel B: No distortion at output of system**
  - **All frequencies at output have equal amplitude, just as at input**

5 ## FREQUENCY DISTORTION

- **Note panels C & D**
  - **All frequencies <u>not</u> reproduced with same amplitude**
  - **System was frequency selective; filtering occurred**
  - **Signal underwent <u>frequency distortion</u>**

6 ## System Transfer Function

- **The three functions in panels B, C, & D describe the <u>system transfer function</u>, or <u>amplitude response</u>, of the system**
- **The amplitude response reveals evidence of frequency distortion**

7 ## Linear Systems

- **Linear systems alter only the amplitudes and phases of a signal**
- **They <u>do</u> produce frequency distortion**
- **“Frequency response” of an audio device might be described as**
  - **100 Hz to 10,000 Hz**
  - **Flat ± X dB**

8 ## Linear Systems

- **Panel A: Input-Output (I/O) function of <u>linear</u> system**
  - **As input amplitude increases (e.g., by 5 dB), output amplitude increases <u>proportionally</u> (5 dB)**
  - **Output amplitude need not <u>equal</u> input amplitude; the change is a proportional one**

9 ## Linear Systems

- Panel B: The characteristics of the input sine wave are preserved faithfully in output sine wave; Why?
  - ☑ The system is linear

10 TRANSIENT DISTORTION

- The amplitude response of a sine wave is not really a line spectrum as shown in panel A; Why?
  - ♦ Its duration is finite

11 Effects of Duration on Amplitude Response

- Panel A: Infinite duration
- Panel B: 100 ms tone burst

12 Effects of Duration on Amplitude Response

- ♦ Energy spread to surrounding frequencies, and amplitude spectrum is continuous
- ♦ Nulls at integer multiples of reciprocal of duration (100 ms)

>> 1 /.1 = +/- 10 Hz

>> 2 /.1 = +/- 20 Hz

>> 3 /.1 = +/- 30 Hz, etc.

13 Effects of Duration on Amplitude Response

- Panel C: Duration shortened from 100 ms to 4 ms
  - ♦ 1 /.004 = + / - 250 Hz
  - ♦ 2 /.004 = + / - 500 Hz
  - ♦ 3 /.004 = + / - 750 Hz, etc
- Thus, width of each lobe is inversely proportional to duration

14 Effects of Rise-Decay Time on Amplitude Response

- Amplitude rises over time from zero to max, and falls over time from max to zero
- Panel A shows amplitude envelope in the time domain
- Panel B shows amplitude spectrum in the frequency domain

15 Effects of Rise -Decay Time on Amplitude Response

- Panel B
  - ♦ Energy spread to other frequencies: Continuous spectrum
  - ♦ Initiation & termination of signal produces transients

16 Effects of Rise -Decay Time on Amplitude Response

- ♦ The longer the rise-fall time, the less the transient distortion
- ♦ ANSI S3.6-1989
  - >> Time from -20 dB to -1 dB shall not be less than 20 ms

17 AMPLITUDE DISTORTION

- Panel A: I / O function for a linear system, compared with
- Panel B: I / O function for a nonlinear system
- Panel C: As input amplitude to linear system (from panel A) changes, output amplitude

changes proportionally; no distortion

18 AMPLITUDE DISTORTION

- Panel D: Input amplitudes do not exceed limits of linearity of nonlinear system; changes in output amplitude still are proportional to changes in input amplitude; no distortion

19 AMPLITUDE DISTORTION

- Panel A: Do some of the instantaneous amplitudes of the input signal exceed the limits of linearity of the system?
  - ☑ Yes
- What is the result?
  - ☑ The instantaneous amplitudes at or near maximum amplitude are "clipped off"-- output amplitude is not proportional to input amplitude -- the signal has been peak clipped; distortion
- Panel B: more severe peak clipping; more severe distortion

20 AMPLITUDE DISTORTION

- The distortion is called amplitude distortion; Why?
  - ☑ Amplitude is the waveform parameter that was altered
- Amplitude distortion is also called nonlinear distortion; Why?
  - ☑ The distortion arose from operating on the nonlinear portion of the I/O function

21 Effects of Amplitude, or Nonlinear, Distortion on Amplitude Spectrum

- Input signals in panels A & B are sinusoids
- Output signals are what kind of sound waves?
  - ☑ Complex periodic waveforms
- At what frequencies should you expect to see energy in a complex periodic waveform?
  - ☑ Harmonics
- Thus, amplitude, or nonlinear, distortion also can be called harmonic distortion (if the input waveform is sinusoidal)

22 Effects of Amplitude, or Nonlinear, Distortion on Amplitude Spectrum

- The figure displays the output spectrum from a nonlinear system
- At the input, f = 100 Hz
- At the output, note energy at harmonics of f_0

23 Percentage Harmonic Distortion

- The concept: Percentage harmonic distortion is proportion of total energy that is undesired energy
- Measure the voltage (V) of the output spectrum with a 1/3-octave filter and voltmeter
- Set f_c of the filter to equal, progressively, the frequency of each harmonic selectively
  - ♦ V_2, V_3, V_4, and V_5 reflect undesired energy
  - ♦ V_1, V_2, V_3, V_4, and V_5 reflect total energy
- Next, compute the proportion of total energy that is undesired energy

24 Calculation

- % = (Undesired Energy / Total Energy) $\times$ 100
  - ♦ $= \frac{f(V_2, V_3, V_4.....V_n)}{f(V_1, V_2, V_3.....V_n)} \times 100$, where
    >> f is some undefined function

>> Vs cannot be summed

25 ## Calculation

- = $f'(V_2^2 + V_3^2 + V_4^2 + ...V_n^2) \times 100$ $f'(V_1^2 + V_2^2 + V_3^2 + ...V_n^2)$
  >> Vs are squared ($W \propto V^2$) so summing now permissible
- = $\sqrt{(V_2^2 + V_3^2 + V_4^2 + ... + V_n^2)} \times 100$ $\sqrt{(V_1^2 + V_2^2 + V_3^2 + ... + V_n^2)}$
  >> Return to voltages by taking square root of sum ($V \propto \sqrt{W}$)

26 ## Approximate Percentage Harmonic Distortion

- % = $\sqrt{V_2^2 + V_3^2 + V_4^2 + ... + V_n^2} \times 100$ $\sqrt{V_1^2}$
- How do we justify the approximation?
- Most of total energy in denominator comes from f_0 (V_1)
- Contribution of V_2, V_3, V_4, and V_5 is usually negligible

27 ## Amplitude Response and Dynamic Range

- Panel A: I/O functions for several driving frequencies from .1 kHz to 10 kHz
  - On each I/O function, location of the filled dot identifies point of maximum permissible harmonic distortion

28 ## Amplitude Response and Dynamic Range

- Panel B: The dots from I/O functions are connected and redrawn in the frequency domain to form the upper curve
  - That defines the amplitude response of the system
  - Electrical noise floor (ENF) of system is shown by lower curve

29 ## Amplitude Response and Dynamic Range

- Dynamic range is distance in decibels from ENF to amplitude response at maximum
- Does ENF vary with frequency?
  - ☑ Yes

30 ## Intermodulation Distortion

- Driving signal is complex
- Experience nonlinear distortion
- Result called <u>intermodulation distortion</u>

31 ## Intermodulation Distortion

- Suppose <u>input signal</u> has two frequency components
  - f_1 = 100 Hz
  - f_2 = 110 Hz
- Frequency components of <u>output signal</u> include two types of distortion products:
  - Harmonics
  - Combination Tones

>> Difference tones
>> Summation tones

32 ## Intermodulation Distortion

- 1. Harmonics of each frequency component
  - $1f_1$ = 100 Hz & $1f_2$ = 110 Hz
  - $2f_1$ = 200 Hz & $2f_2$ = 220 Hz
  - $3f_1$ = 300 Hz & $3f_2$ = 330 Hz, etc.

33 ## Intermodulation Distortion

- 2. Difference tones:
  - $|1f_1 - f_2|$ = 10 Hz
  - $|2f_1 - f_2|$ = 90 Hz
  - $|3f_1 - f_2|$ = 190 Hz, etc.

34 ## Intermodulation Distortion

- 3. Summation tones:
  - $1f_1 + f_2$ = 210 Hz
  - $2f_1 + f_2$ = 310 Hz
  - $3f_1 + f_2$ = 410 Hz, etc.

35 ## Combination Tones

- Equation 7.3 defines frequencies of all harmonics, difference tones, and summation tones
  - $mf_1 \pm nf_2$, where
  - m and n are assigned all integer values