Experiment No. 1.
Telephone Systems and Dialing Tones

By: Prof. Gabriel M. Rebeiz
The University of Michigan
EECS Dept.
Ann Arbor, Michigan

The circuit diagram of a telephone circuit is shown in Fig. 1. The telephone is generally connected to the central office using a pair of twisted wires. However, new installations are using a coaxial-fiber system for wide bandwidth communications. Unlike audio systems with bandwidths of 20 Hz – 20 kHz for high fidelity sound, a telephone operates over the 300 Hz – 3.3 kHz bandwidth. The reason is that most of the energy of human voice is within this frequency bandwidth and a 3 kHz bandwidth is enough for reliable conversation (Reliable: yes, but not excellent!). The bandwidth limitation is the main reason why we have trouble distinguishing "b" from "p" from "d" over a telephone. The audio voltage swing is 5-500 mV peak, leading to a dynamic range of 40 dB, which is much lower than hi-fi system (dynamic range of 70-90 dB).

The telephone operates on a 48 V DC system supplied over a pair of lines from the central telephone office. This is historical since the telephone was invented before the AC 60 Hz power distribution system and could not be changed anymore. To grab your attention, the control office sends bursts of a 20 Hz sinusoidal signal with a 75 V rms voltage to activate the ringer. The bursts are on for 2 sec and off for 4 seconds. When a party answers the phone, the telephone switch closes, the central office detects a DC current in the circuit and stops the ringing signal. You could ask why 75 V? It is huge! The answer is that this signal was needed to activate inefficient ringers on old telephones. In newer phones with electronic ringers, a TTL (5 V) digital signal is enough to activate the ringer. However, this telephone system will not be compatible with old phones!
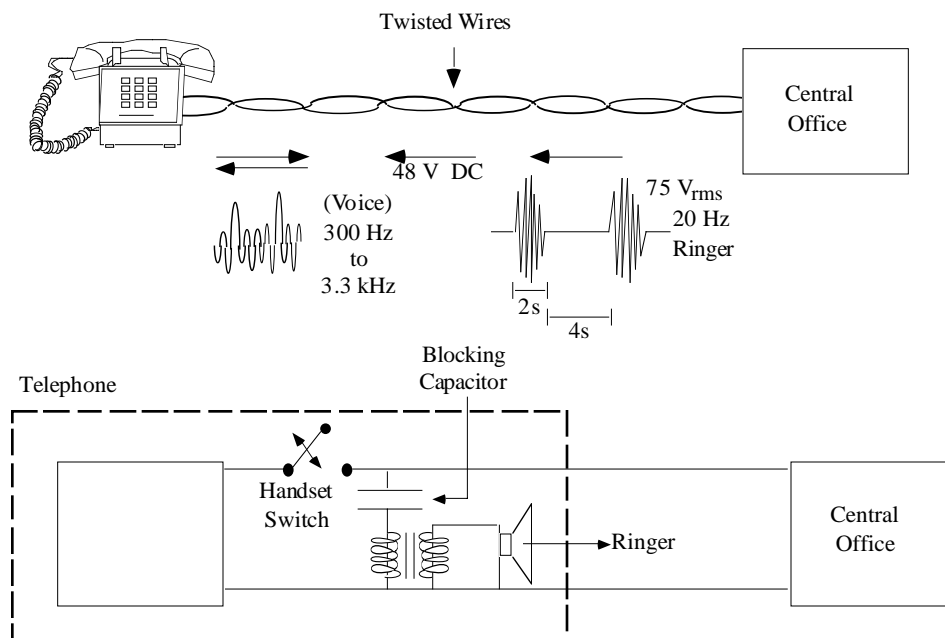
Fig. 1. The home telephone system.

The lines between the central telephone office and your home therefore carry:

DC (O Hz) at 48 V:	Powering the phone
20 Hz Bursts at 75 V_{rms}:	For activating the ringer
300-3,300 Hz at 5 mV – 0.5 V:	Voice signal

In order to dial a phone number, you need to transmit specific frequencies which are within the 300 Hz – 3,300 Hz range. However, if you assign a simple frequency to each number, then somebody whistling when you are dialing (or a large clean sound) can actually cause you to misdial! A very nice way to solve this interference problem is to send two frequencies for each number. The probability that two specific frequencies with a ratio equal to a rational number are present in the background noise when you are dialing is really very low!

The dial pad of a telephone is shown in Fig. 2. When a button is pushed, the two tones corresponding to the intersection of the vertical and horizontal axes are sent. Notice that no frequency is the harmonic of any other frequency thereby avoiding problems due to distortion and harmonic generation (see Lab. 2.2). Also, no frequency can be synthesized from the sum or difference of any two frequencies, thereby avoiding misdialing problems due to intermodulation products (see Lab. 3.2).
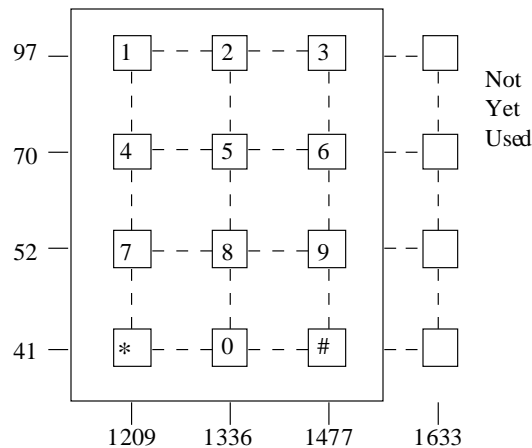

Fig. 2: A telephone dial-pad. (All frequencies are in Hz.)

Experiment No. 1.
Time & Frequency Domain Measurements,
Telephone Systems & Dialing Tones

Goal: The goal of Experiment #1 is to learn how to use the equipment in the lab, and to measure the output frequencies of a telephone dialer.

❏ Read this Experiment and answer the pre-lab questions before you come to the lab.

1.0 Time Domain Measurements (or Learning the Agilent Oscilloscope):

Equipment: Agilent 33120A Waveform Generator

Agilent 54645A Oscilloscope

1. Connect the output of the Agilent 33120A waveform generator to Channel 1 of the Agilent scope using a coaxial cable.

2. Set the waveform generator to deliver a sinewave at 1 KHz and 2 Vppk. Also, set the offset voltage to be zero (see p. 4).

3. Run the [Setup], **Default Setup** sequence. (Do NOT use the [Auto-scale] key. This key resets a large portion of the scope settings and displays the waveform. By using this keey you can develop a very bad habit and never learn how to use a scope well!)

4. Turn the VERTICAL Volts/Div knob of Channel 1 and see how you can expand or compress the waveform depending on your selection. You will find that you can easily "saturate" a scope and you should never do this. The waveform must always be within the display area.

   Turn the Position knob to see how you can move the center of the waveform up and down. Now, center the waveform and choose a 500 mV/div setting.

5. Turn the HORIZONTAL Time/Div knob and see how you can expand or compress the waveform depending on your timebase selection. If you choose a 200 µs/div setting, the 1 KHz (/msec period) sinewave "looks" expanded. If you choose a 2 msec/div setting, the 1 KHz sinewave "looks" compressed. Choose a 500 µs/div setting.

   Turn the Delay knob to the right to see how you can delay the triggering time. Look at the dark small arrows (on top and bottom of the screen). These define the actual trigger point. Return the delay back to 0.00 s (align the arrows).

6. Trigger Selection: The scope needs a signal to trigger its sampling circuitry. The triggering signal could be derived from the signal itself or from external or internal references.

   a. Press the [Source] key under the TRIGGER section. You will find on the bottom of the screen:

   Ch.1 Ch. 2 External Line

      Press Ch. 2: Since you have no input to Channel 2, you will loose your lock and the signal will not be stable on the screen.

Press <u>Line</u>: The scope is now triggering on the 60 Hz AC line voltage. Since it is not locked to the 1 KHz waveform, the signal will not be stable on the screen.

Press <u>External</u>: The scope triggers on an external signal provided by the external trigger input. Since you have no input, the signal will not be stable on the screen. You can connect the SYNC output of the Agilent 33120A waveform generator to the External Trigger input. Your signal will then be stable on the screen. Do it if you wish.

Press <u>Ch. 1</u>: The scope is triggering back on the input signal and the signal is stable on the screen. (Remove the external trigger if you have connected it.)

7. <u>Slope/Glitch Triggering</u>: A scope can trigger on the rising edge or falling edge of a waveform. Look at the left side of the screen. You will find two small dark arrows, on top and bottom of the screen. These arrows define the triggering plane.

   a. Press the [Slope Glitch] key and choose the rising edge key (see the screen). Look at the waveform at the "arrow" reference plane. Note that you are triggering at the rising slope of the sinewave.

   b. Choose the falling edge and note that you are triggering at the falling edge of the sinewave.

   c. Return to rising edge trigger mode (we do not use TV or Glitch triggering).

8. <u>Mode/Coupling Triggering</u>: A scope needs a certain voltage level to trigger. Normally, this is set automatically, but in certain cases, you want to control this level so that you do not trigger on low level signals or noise.

   a. Press the [Mode Coupling] key and look at the bottom of the screen. The <u>Auto Level</u> is highlighted.

   b. Press the <u>Auto</u> option on the screen, and turn the <u>Level</u> knob under the TRIGGER section. Look at the waveform. As the triggering level is raised (or lowered), the scope triggers a bit late (or earlier) so as to align the set level with the triggering reference plane. When the level is above the waveform peak, the scope does not trigger anymore and you loose lock.

   c. Press the <u>Normal</u> option on the screen and repeat. When the level is above the waveform peak, you loose the trigger and the waveform freezes. The scope is not running anymore.

   d. Press back the <u>Auto Level</u> option.

   e. The Coupling key, AC or DC, means that the signal is either AC or DC coupled. If it is AC coupled, then the scope will not show the DC level of the signal.

   f. The <u>Reject</u> key introduces low-pass filter with a corner freq. of 50 KHz to reject all noise above 50 KHz, or a high-pass filter with a corner freq of 50 KHz to reject all noise below 50 KHz. We rarely use this key.

9. <u>Voltage Measurements</u>: Look at the top of the scope under the <u>Measure</u> section and press the [Voltage] key. Now look at the bottom of the screen.

   a. Make sure that you are on <u>Source</u> 1 (for Channel 1).

   b. ❏ Press <u>Vpp</u>, <u>Vavg</u>, and <u>Vrms</u> and write these values in your notebook.

   c. Press <u>Clear Meas</u> and then <u>Next Menu</u>.

    d. Press <u>Vmax</u>, <u>Vmin</u>, <u>Vtop</u>, <u>Vbase</u>. Do not write them in your notebook.

10. <u>Time Measurements</u>: Press now the ⬚Time key and look at the bottom of the screen.

    a. Make sure that you are on <u>Source</u> 1 (for Channel 1 signals).

    b. ❏ Press <u>Freq</u>, <u>Period</u>, <u>Duty</u> and write these values on your notebook.

    c. Press <u>Clear Meas</u> and then <u>Next Menu</u>.

    d. Press <u>+Width</u> (positive part of waveform), <u>−Width</u>, <u>Risetime</u> (defined at 10% to 90% of the waveform).

11. <u>Cursor Measurements</u>: Press the ⬚Cursors key and look at the bottom of the screen.

    a. You have two cursors, <u>V1</u> and <u>V2</u> and you can read any voltage on these cursors. Also, you can read the difference between V1 and V2 (ΔV). Read V_{peak} and V_{ppk} using the cursors.

    b. Same for time measurements with <u>t1</u>, <u>t2</u> and Δt.

    Note that if you want to read the <u>Degrees</u> (or phase of a signal), you need to calibrate first. Put a cursor at a zero crossing (t1) and put the other cursor one period away Δt = 1 ms). Press the <u>Deg</u> key and the <u>Set 360˚</u> key. You have now calibrated the phase.

2.0 Square-Wave Risetime Measurement:

1. Connect now the Agilent 33120A waveform generator <u>directly</u> to the scope. Set the Agilent 33120A to give a 10 KHz square-wave with Vppk =2 V.

    On the Scope, choose a 500 mV/div setting for Vertical, and a 20.0 µs/div for Horizontal.

2. ❏ Go to the Measure section, press the ⬚Time key, then go to the <u>Next Menu</u> and press the <u>Risetime</u> key. Write your measurements in on your lab notebook.

3. Choose now a 50 ns/div timebase. Notice how the waveform is still triggered underneath the arrows and you only see the rising edge of the square wave. Draw the waveform in your lab notebook. (You will see some oscillations called "ringing" and you will study this in EECS 211.)

4. ❏ Measure the risetime and write it in your lab notebook.

3.0 Frequency Domain Measurements:

1. Set the Agilent 33120A to give a 10 KHz sinewave with V_{ppk} = 2V. Connect it to Channel 1 of the scope.

2. Choose the vertical settings at 0.5 V/div and a horizontal setting (timebase) at 500 µs/div. You will see a lot of sinewaves on the screen.

Entering Math Mode:

3. Press the Math Mode ⎡±⎤ button.

4. At the bottom of the screen, you will see Function 1 and Function 2.

   Function 1 does addition (+), subtraction (–) or multiplication (*) on signals of Channels 1 & 2.

   Function 2 does integration (∫dt), differentiation (dv/dt) or a Fast Fourier Transform (FFT) on signals of Channels 1 or 2, or of the resulting waveform of Function 1 (F1).

5. Press Function 2: ON and Menu

   Operand: 1, 2, F1 (Choose the operand to be Channel 1)

   Operation: FFT, ∫dt, dv/dt (Choose FFT).

 | | | |
   |---|---|---|
   | Units/div | 10 dB | (Sets the units in dB for the vertical lines) |
   | Ref Level | 0.00 dBV | (Sets the top horizontal line in dBV) |
   | FFT Menu | | (Goes into the FFT menu) |

6. Press FFT Menu, you will see at the bottom of the screen:

 | | | |
   |---|---|---|
   | Center Freq | 48.8 KHz | (Shows the center freq. Controlled by timebase (horizontal) settings) |
   | Freq Scan | 97.66 KHz | (Shows the freq. scan. Controlled by timebase (horizontal) settings). |
   | Window | Hanning | (Can be selected between Hanning, flat-top, Exponential and Rectangular. Always choose Hanning.) |
   | Autoscale FFT | | (Moves 0 Hz to left and autoscales the vertical settings. I do not like it.) |

7. Press Channel 1 button ⎡1⎤ twice to get rid of the time domain representation.

   > (To view Channel 1 in the time domain, press ⎡1⎤ at any time.)

8. Press Math Mode button ⬚ to get back into the FFT menu at the bottom of the screen.

You have in front of you a clear representation of a sinewave signal in frequency domain. Notice the single peak at 10 KHz (the rest is sampling noise).

9. ❏ Press ⬚. Using V_1, V_2 and f_1, f_2, measure the amplitude of the sinewave (in dBV) at 10 KHz and the average amplitude of the noise. Write them in your lab notebook.

10a. ❏ Change the amplitude of the sinewave to 100 mVppk and using the Cursors mode, measure the voltage in dBV. Write it down on your notebook.

Return the amplitude back to 2 Vppk.

10b. ❏ Using the knob on the function generator, vary the frequency in 1 KHz steps to 100 KHz and see the peak moving on the screen. Notice what happens above 100 KHz. The signal returns to the screen and moves backward!! This is not correct and is due to the sampling circuitry/firmwave of the scope. THEREFORE, ALWAYS BE SURE THAT YOUR FREQUENCY SPAN (AT THE TOP OF THE SCREEN) is LARGER THAN YOUR SIGNAL!

Return the sinewave frequency to 10 KHz.

11. ❏ Now comes the interesting part: Press the Square-Wave button (10 KHz, Vppk = 2V) and measure the frequency and amplitude of the fundamental (f_O) and of each harmonic ($3f_O$, $5f_O$, $7f_O$ and $9f_O$). Write them in your lab notebook in a table form.

12. Press back the sinewave button. Now play with the timebase settings and see how you change the center frequency and frequency span of the display. Always choose a setting which puts your fundamental and harmonic frequencies within the set frequency span!

To view Channel 1 in the time domain, press [1] at any time.

To view Channel 1 in the frequency domain, press [±] at any time.

The dB Scale: In time domain, the unit is volts or "V." You can have V_{ppk}, V_{rms} or V_{av}. In frequency domain, the only unit is <u>dB</u> since the rms voltage is needed to calculate the power carried by a signal. The conversion is quite easy if you have a calculator.

$$dB = 20 \log (Voltage_{rms} (V))$$

$$V_{rms} = \frac{V_{pk}}{\sqrt{2}} = \frac{V_{ppk}}{2\sqrt{2}}$$

Voltage$_{(rms)}$ (V)	dB
10	20
1	0
0.1	–20
0.001	–40
0.001	–60
(1 mV)	

and

Voltage$_{(rms)}$ (V)	dB
1.0	0
0.8	–2
0.5	–6
0.4	–8
0.2	–14
0.1	–20

4.0 Telephone Dialing Tones:

Equipment: Agilent 34401A Multimeter

Agilent E3631A Triple Power Supply

Agilent 54645A Digital Sampling Scope

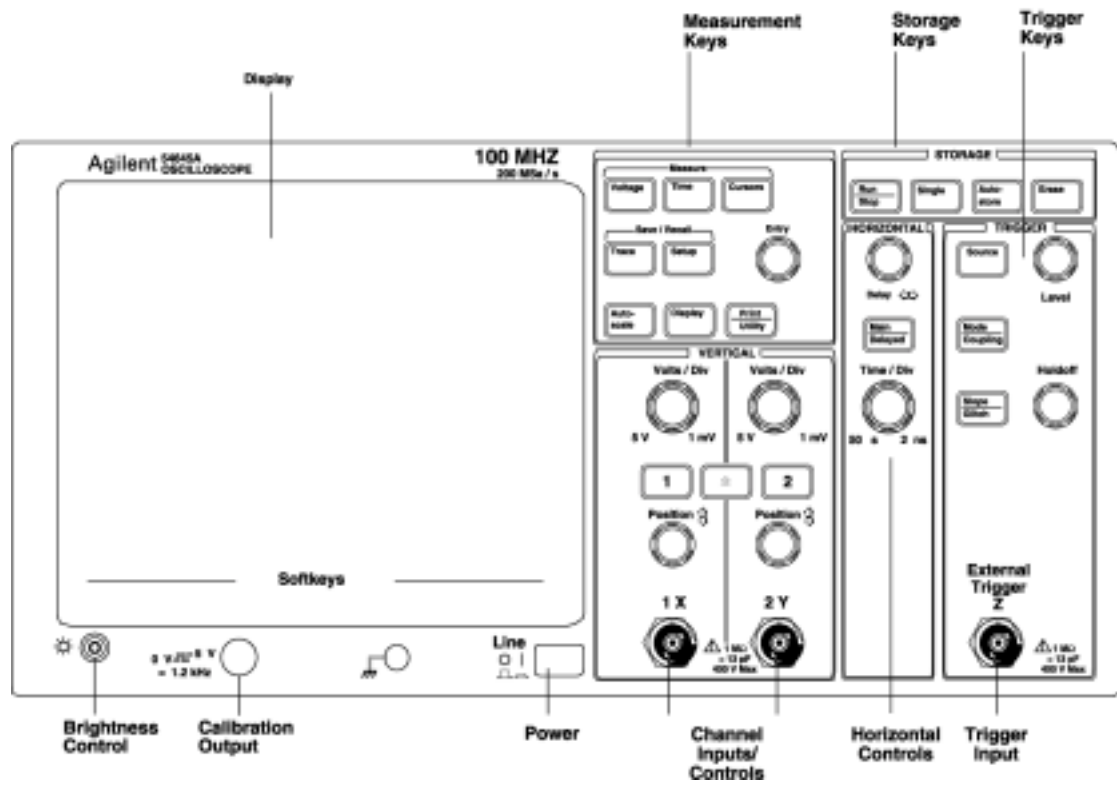Telephone Tone Dialer Mounted on a Circuit Board

Experiment Set-Up:

1. Connect the Agilent 33120A signal generator to the digital scope and measure a 1 Vppk sinewave at 1500 Hz, and a 1 Vppk sawtooth wave at 1500 Hz in time and frequency domain. (Do not take any data. This is just to get back in the groove!)

2. Set the Agilent E3631A power supply at 4 V. Set the current limit to 100 mA (see p. 7). Make sure that the (–) terminal of the Agilent E3631A power supply is connected to the ground (⏚) terminal of the circuit board, and the (+) terminal of Agilent E3631A is connected to (+) of the circuit board. Check the voltage applied to the telephone dialer unit with the Agilent 34401A multimeter .

3. Dial any number(s) you wish and listen to the generated tones.

4. ❑ Neatly graph in time domain the signal resulting from the number 8. Label few max. and min. values of the waveform. Choose a time span between 1 ms/divison and 5 ms/division which results in a signal with many peaks and valleys on the screen and "STOP" the triggering of the scope (top right section). Go into the cursor mode and try to determine the "period" of this complex waveform (this is a bit tricky) and the corresponding "frequency" (f = 1/ ΔT).

   ❑ Sketch the time waveform on your notebook and compare it to your pre-lab exercise.

   ❑ Now, "RUN" the oscilloscope again and set the FFT correctly. Go into the cursor mode and measure the signals at the two frequencies (~852 Hz, ~1336 Hz) and their amplitudes in dBV.

   ❑ Sketch the frequency spectrum on your notebook (do not include the noise) and label the axes and put the measured values on the graph.

5. Repeat above exercise with any other number of your choose (but <u>not</u> on the same column or row as #8) and the "*" key.

   ❑ Sketch the time waveform and frequency spectrum of each signal on your notebook. Measure and label the frequencies and amplitues corresponding to each signal.

6. You are now free to play for 5-10 minutes with the dialer but you are not required to take any data. See how the frequencies jump around but are understandable. See how <u>different</u> the time waveforms are from number to number, and that we cannot quickly get a lot of information from them. This is yet another indication that WE THINK IN FREQUENCY DOMAIN!

Agilent 54645A Oscilloscope Front Panel

Experiment No. 1.
Time & Frequency Domain Measurements,
Telephone Systems & Dialing Tones

Pre-Lab Assignment

1. ❏ Translate the following dBV numbers into <u>rms</u> and <u>ppk</u> voltages:

   +10 dBV   =   _____ V   +38 dBV   =   _____ V

   −10 dBV   =   _____ V   −38 dBV   =   _____ V

   −20 dBV   =   _____ V   −60 dBV   =   _____ V

   +20 dBV   =   _____ V   +60 dBV   =   _____ V

2. ❏ Translate the following rms voltages into dBV (20 log (V_{rms})).

   100 V   =   _____ dBV   10 mV   =   _____ dBV

   10 V   =   _____ dBV   60 mV   =   _____ dBV

   1 V   =   _____ dBV   6 mV   =   _____ dBV

   100 mV   =   _____ dBV   60 µV   =   _____ dBV

3. ❏ Translate the following <u>ppk</u> voltages into dBV.

   10 V   =   _____ dBV   150 mV   =   _____ dBV

   2 V   =   _____ dBV   100 mV   =   _____ dBV

   0.5 V   =   _____ dBV   22 mV   =   _____ dBV
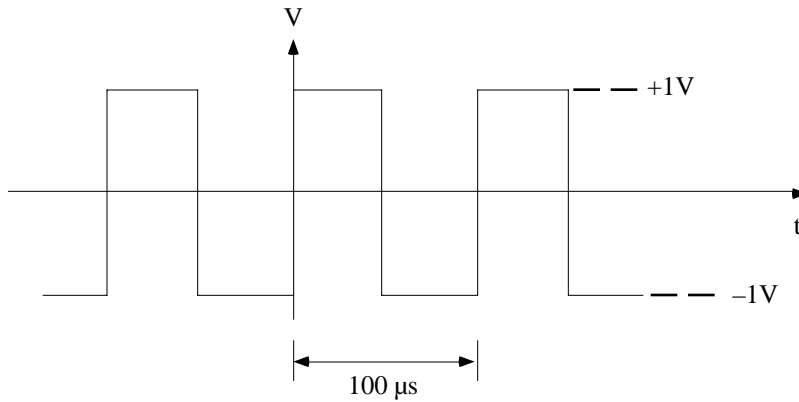
   200 mV   =   _____ dBV   3 mV   =   _____ dBV

4. ❏ What is the meaning of a signal period (T)?

   What is the period and frequency of a DC signal?

   What is the period of a 10 kHz signal in msec?

5. ❏ Derive the Fourier-Series expansion of a square-wave. Calculate the Fourier coefficients of a 10 KHz (f_O), 2 V_{ppk} square wave up to 13 f_O. The DC level of the square-wave is zero.

   Write your answers in Volts (rms) and dBV for the fundamental and each harmonic frequency in a table form. You will need this table for comparison with experimental data.

6. ❏ Take two sinusoidal signals at 852 Hz and 1336 Hz (number 8).

v_1 (t) = cos (2π (852) t)

v_2 (t) = cos (2π (1336) t)

Plot: v (t) = v_1 (t) + v_2 (t)   Use MATLAB (attach your computer code with your plot and remember to include your name and date of work in the title of your plot).

x-axis = 0 - 0.01 sec. (choose small time steps)

y-axis = −2 to + 2 V

The resulting waveform is complex. Look *carefully* at the graph: Can you find the period, T, of the complex waveform? What are the max. and min. voltages? Write down your answers.

7. ❏ If a circuit exhibits non-linear behaviour, it will generate harmonics and intermodulation products between two signals. For f_1 = 852 Hz and f_2 = 1336 Hz (number 8), calculate the following harmonic and intermodulation frequencies:

$2f_1$, $2f_2$, $f_2 - f_1$, $f_2 + f_1$, $2f_2 - f_1$, $2f_2 + f_1$, $2f_1 - f_2$, $2f_1 + f_2$.

The $(2f_1)$, $(3f_1)$, ... and $(2f_2)$, $(3f_2)$, ... are called the *harmonics* of the signals.

The $(f_1 - f_2)$, $(f_1 + f_2)$, $(2f_1 + f_2)$, ... are called the *intermodulation products* of the signals.

Do not be surprised if you find some of these frequencies on your dial-tone spectrum.

Experiment No. 1.
Time & Frequency Domain Measurements,
Telephone Systems & Dialing Tones

Lab Report Assignment

1. Compare the measured dBV values of the 10 KHz square-wave (fundamental and harmonics) with the calculated values in your pre-lab. Organize the data in a table format and show difference in dB.

2. ❑ Take the measured spectrum of the 10 kHz square-waveform at f_O, $3f_O$, $5f_O$, $7f_O$, $9f_O$ and plot the following waveforms from t=0 to t=0.3 msec. Note that V_{f_O}, V_{3f_O}, ... are rms values!

$$v_1(t) = \sqrt{2}\, V_{(f_o)} \sin\left(2\pi f_o\, t\right) \qquad\qquad \text{(fundamental)} \qquad \underline{\text{V in rms!}}$$

$$v_2(t) = v_1(t) + \sqrt{2}\left[V_{(3f_o)} \sin\left(2\pi\left(3f_o\right)t\right) + V_{(5f_o)} \sin\left(2\pi\left(5f_o\right)t\right) \right] \quad \text{(Up to the 5th harmonic)}$$

$$v_3(t) = v_2(t) + \sqrt{2}\left[V_{(7f_o)} \sin\left(2\pi\left(7f_o\right)t\right) + V_{(9f_o)} \sin\left(2\pi\left(9f_o\right)t\right) \right] \quad \text{(Up to the 9th harmonic)}$$

Do these summations using MATLAB. The 3 plots should be well detailed with an x axis in msec (0–0.3 msec) and a y-axis in volts ($\pm$2 V). Your time steps should be small enough to clearly see the waveforms.

Reminder: Do not forget to attach your MATLAB code with every plot, and include your name and date of work in the title of each plot.

The essential point behind this exercise is to clearly "see" how any periodic waveform (square wave or other) is actually composed of a fundamental component (f_O) and a series of higher frequencies called harmonics. In the case of a square wave, you need at least up to $5f_O$ before it starts <u>looking like</u> a square wave and up to $9f_O$ before it <u>becomes</u> a good square wave. This means that a 100 MHz square-wave clock computer must be designed so that the computer lines (on-chip and off-chip) pass at least up to 500 MHz sinusoidal signals for proper operation. DO NOT FORGET THIS!

3. a) Translate the meaured dBV of the spectrum of #8 into Vrms and plot the following function using MATLAB.

$$V_1(t) = \sqrt{2}\left(V_{rms_{(f_1)}} \cos\left(2\pi\left(f_1\right)t\right) + V_{rms_{(f_2)}} \cos\left(2\pi\left(f_2\right)t\right) \right) \qquad \text{where } f_1 \text{ and } f_2 \text{ are your}$$

measured frequencies for #8.

This is the ideal #8 waveform as per your pre-lab (but with a different vertical scale).

b) One frequently asked question is the use of cos (2 π ft) or sin (2 π ft) functions. You will learn later in the course that there is a difference between them, and it is called "phase shift."

To see the difference, plot V2(t) and V3(t) on the same graph using MATLAB.

$$V_2(t) = \cos(2\pi f_1 t) + \cos(2\pi f_2 t) \qquad f_1 = 852\,Hz \ \ f_2 = 1336\,Hz$$

$$V_3(t) = \cos(2\pi f_1 t) + \sin(2\pi f_2 t)$$

$$x - axis = 0 - 0.004\,sec$$

$$y - axis = -2\,to + 2\,V$$

Write a brief statement on the comparison between V2(t) and V3(t).