

Engineering 2133

COMMUNICATIONS 1

Laboratory Manual

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References

1. "Electronic Communications" [Class text]
Dennis Roddy and John Coolen
Prentice Hall, 4th ed.

Lakehead University
Department of Electrical Engineering

POLICY AND RULES FOR LABORATORY EXERCISES

- No Food, No Beverages allowed in the laboratory room!!
- Keep clothes, bags, etc. OFF the benches with equipment on them.
- Be On Time ! Being late is annoying to your lab partners and if the experiment has progressed too far you may not be credited with doing the experiment!
- Safety precautions must be observed at all times to prevent electric shock, damage to instruments, etc.

COME PREPARED! BOTH THE WRITTEN LAB REPORTS AND LAB PERFORMANCE (INCL. ATTITUDE, PUNCTUALITY, PREPAREDNESS) WILL BE CONSIDERED FOR THE FINAL LAB MARK!

Lab Exercises

General

The maximum number of students in a lab work group is indicated on the sign-up sheet.

Should students leave a work group for whatever reason such that only one student remains in a group, this student may join another team provided there is still room in that team without exceeding the above maximum number.

Missed Lab Exercises

It is mandatory to perform all lab exercises according to course requirements.

Failure to perform one or more lab exercises results in a grade of "F" for the course.

When a student misses a lab exercise for whatever reason he/she must notify the instructor as soon as possible. If the reasons given for the absence are satisfactory to the instructor, a make-up opportunity may be arranged.

There may be a chance to let the student join another team to perform the missed lab provided this does not then exceed the max number of students in that group. If it is not possible to accommodate this then a final make-up date will be arranged to take place within one week after the end of classes. Should the student fail to attend this appointment he/she will be required to provide sufficient proof of inability to attend (medical certificate, air ticket, etc.) to avoid the "F" grade. The onus of proof lies entirely with the student! The student then must immediately make another appointment with the lab instructor!

The lab report in such a case is due within one week after performance, else the "F" stands.

Lab Reports

General

It is mandatory that all lab reports must be submitted on time as specified by the lab instructor!

Plagiarism

Plagiarism will not be tolerated and may result in a grade of "F" for the course.

Any material taken from sources like books, manuals, web sites, magazines, etc. must be clearly referenced as a footnote or under a bibliography!

A re-write of a report will be granted only under exceptional circumstances!

Missed Lab exercises

In some courses group lab reports may be allowed by the lab instructor. The group's composition is also determined by the lab instructor.

If a student fails to perform a lab exercise with his group he/she will have to write his/her own individual lab report!

Late Submission

The submission schedule (due date) will be made known by the attending lab instructor.

Late submission results in a deduction of 0.5 marks per day out of 10 full marks.

No or Partial Submission

A final date for submission will be clearly indicated on the sign-up sheet and/or announced by the lab instructor. If after that date not all lab reports have been submitted, the student will receive a grade of "F" for the course!

The deduction of marks for late submission still applies.

- The report **must** be typed. Graphs may be produced by computer, provided the software is suited for that use, i.e. grid, proper scaling and correct labelling can be achieved and the plot is smooth. If a graph is used to extract data or to provide some precise information, show precisely how the information is obtained (here it may be better to draw it on graph paper by hand - usually is faster, too).
- **For your report use YOUR OWN words to present your report concise, clear and clean. As pointed out above, copying etc. will be considered as plagiarism and will be severely punished by reducing marks or, in severe cases, served with an "F" as mentioned above (the provider/lender of the original work included)!**
- The notes/sheets containing the raw data taken by each student during the experiment are to be initialized by the attending technologist before leaving the lab and attached to the written report. Reports with the raw data missing are subject to a deduction of one full mark (= 10%)!

The student is encouraged to develop and use his/her own personal style for writing and presenting his/her report. However, standard procedures in industry and research laboratories require certain information to be documented. Therefore, adhere fairly loosely to a general format like the following:

- Title page

(Please make an exception here: Pages **stapled - no folders, plastic covers**, etc):

Course number, Experiment number, Experiment Title, **Name**, Lab partners, Date of performance

- Abstract

Statement of objective of the experiment (one or two sentences)

Concise and pertinent outline of the theory underlying the experiment (max one page)

- Experiment and Analysis

If the experiment consists of two or more parts, keep the experimental and analysis sections together – the reader of your report does not want to continuously flip pages back and forth to look for data etc.!

Brief outline of the method of investigation (whatever is applicable):

Procedure

Measurement techniques

Schematic diagrams

Equipment identification

Data (tables)

Observations relevant to the experiment and the results

Arrange experimental data, and do the necessary calculations (if applicable, at least a sample calculation), to prepare for analysis

Theoretical calculations (at least sample calculation)

Comparison of experimental results with theory (preferably in form of tables/graphs)

Probable causes and magnitude of errors

- Conclusion (Summary)

- Review questions

- Raw data notes, attached to report, and initialized by the attending lab technologist.

GOOD PRESENTATION IS OF THE UTMOST IMPORTANCE, AS IS CORRECT ENGLISH AND GRAMMAR. EXPECT 20% OF YOUR LAB MARK ASSIGNED TO THIS AREA!!

GENERAL INFORMATION

EQUIPMENT

If equipment needs to be signed out, contact one of the technologists. The person signing it out is responsible for it! Your marks will be held back until all equipment, books, data manuals, tools etc. are returned (i.e. your graduation might depend upon it!).

Signed-out equipment has to be returned to the same technologist from whom you signed it out!

Assure your name is then removed from his sign-out list!

No equipment may be removed from any of the laboratories without explicit permission!

As you see, this is very important! TAKE LABS VERY SERIOUSLY!

DESIGN OF A RESISTIVE ATTENUATOR

Objective

To design and test a T-type or π -type attenuator which gives an insertion loss as required by the lab instructor while maintaining 50Ω matching at input and output.

Theory

A purely resistive circuit used to lower signal levels between a source and a load is called an 'attenuator pad'. It does not introduce any phase shift. The pad usually also provides input and output matching,

If low-loss impedance matching is required, matching circuits containing only reactive components are used.

See Reference 1, class text, Chapter 1.2, for discussion of insertion loss, attenuators and matching pads.

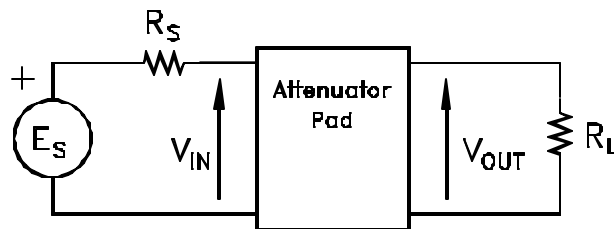


Figure 1

Assuming $R_s = R_L$, and the system being matched, the insertion loss is given by

$$InsertionLoss = -20 \log_{10} \left(\frac{V_o}{V_{IN}} \right) dB$$

(eq. 1)

Experiment

Equipment

DC Power Supply	2 - 50Ω resistors (49.9Ω)	1 - BNC _m -banana adapter
Function Generator	1 - prefab attenuator	1 - BNC _m -BNC _m cable
DMM with manual	1 - breadboard	1 - Yellow grabber cable

Design of the Attenuator

The lab technologist will specify the type (T or Pi) and the desired insertion loss. Calculate the resistor values to build a symmetrical attenuator with input and output impedances of 50Ω (see class text Fig. 1.2.2 and 1.2.3 respectively). Choose single resistors with nominal resistances closest to the calculated values and construct the pad.

DC Test

Set the DC Power Supply to 2V and apply directly to the input of the attenuator. Measure the output and compute the insertion loss. Is the result what you expected?

The attenuator was designed with an input impedance of 50Ω and an output impedance of the same value. The internal resistance of the power supply, however, is considerably less than 50Ω and the input resistance of the DMM considerably more than that. Thus, the power supply and the DMM need to be matched to the attenuator pad!

Do the matching, and again measure the voltage at the attenuator input and attenuator output. Adjust the power supply voltage if necessary to keep the voltage at the attenuator input at 2V.

Compute the insertion loss.

AC Test

Disconnect the DC power supply!

Set the Function Generator to a sine wave, $2V_{\text{RMS}}$, 1 kHz output (use the DMM to measure all voltage levels!)

Measure the insertion loss applying similar techniques as above.

With regard to the matching, consider the following:

- a) the generator has an output impedance of 50Ω .
- b) the following excerpt from the function generator's User's Manual:

The HP 33120A generator has a fixed output source resistance of 50Ω (see Fig.2). During calibration, output amplitudes are calibrated for both the open-circuit voltage (no load) and the terminated output voltage (loaded). The terminated output amplitude is calibrated for an exact 50Ω load. Since the function generator's output resistance and the load resistance form a voltage divider, the measured output voltage of the function generator will vary with load resistance value and accuracy. Thus, for example, if the function generator's output is measured with no load connected, the output will be approximately twice the displayed amplitude (V_{GEN} instead V_{LOAD}).

Repeat the measurement at 10 kHz, 100 kHz, and 1 MHz.

If your measurements don't come out as expected you might want to consider a look at the specifications in the DMM's User Manual.

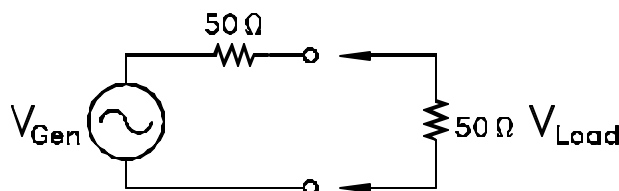


Figure 2

Unknown attenuator

Re-adjust the frequency to 1 kHz.

Note the number of the ready-made attenuator provided by the lab technologist.

Also record the type (T or π) and the resistor values.

Measure the insertion loss, using the same method as in the AC Test. If there is any matching required, do so.

Make a complete sketch of your measuring circuit!

Analysis

1. Using the chosen resistor values of your attenuator design, re-calculate the new (actual) insertion loss.
- 2a. For the DC test, compare the measured to this actual Insertion Loss.
- 2b. Show how you matched the dc power supply and the DMM to the pad's input and output respectively (sketch!), and state your reasoning.
Show at which points of the complete circuit you measured V_{IN} and V_{OUT} !
- 3a. For the AC test, compare the measured to the actual Insertion Loss (as in 2a).
- 3b. Again, show what you did about the matching problem (sketch).
4. Using the resistor values you noted from the 'unknown' attenuator pad, calculate the I.L. and compare with your measurement.

Questions

5. For the attenuator you designed, show that the resistance seen by the source (looking forward into the pad) is actually 50Ω , and the resistance seen by the load (looking back into the pad) is also 50Ω , i.e. in- and output are matched as intended.
6. If the 50Ω resistor used to match the DC supply to the attenuator input were not provided, could the output/input voltage ratio still appear to be correct?
If so, would the measurement be correct considering the conditions stated above at the beginning of the experiment (objective)? Prove your findings!
7. Comment on Questions 5 and 6.
8. Looking at the results of the AC test, are any of the measurements at 10 kHz, 100 kHz, and 1 MHz, off the expected value by more than 10%? If so, would you have an explanation? Is there anything to be learned from this?

Conclusions

Comment on your experiment.

PERIODIC WAVEFORM ANALYSIS

Objective

To measure the Frequency Spectra for various periodic voltage-time functions, and to compare measured results with theory.

Theory

See Reference No. 1, Class text, Chapters 2.1-2.9.

Study in particular eqs. (2.7.1), (2.9.1), (2.9.2), (2.9.3), (2.9.4), and the associated diagrams.

The oscilloscope is by far the most common used instrument to analyze signals in the *time domain* (i.e. representation of signal amplitude versus time). However, the analysis of signals in the *frequency domain* (i.e. the representation of signal amplitude versus frequency) requires the use of a group of instruments called analyzers, the most versatile of this group being the Spectrum Analyzer.

In this experiment we will use the spectrum analyzer to find the harmonics of the investigated waveforms. The oscilloscope serves only to observe the proper waveform selection, but is not used to make measurements.

Experiment

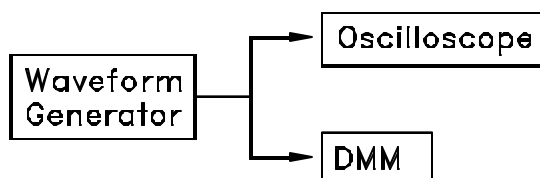
Equipment

Function Generator HP 33120A
Oscilloscope Fluke PM 3370B or HP 54645D
Spectrum Analyzer HP E4411B
DMM Fluke 45
50 Ω resistor

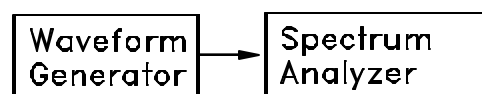
BNC-banana adapter
BNC_F to BNC_F adapter
BNC-T adapter
1 BNC cable, short
1 BNC cable, long

For students not familiar with the instruments, the following notation may help with the use of the instruments:

[Function Keys] = Labeled Keys
<Menu Keys> = Unlabeled Soft Keys



Waveform Set-up



Measurement Set-up

A. Sine Wave

We start out with the most straight forward waveform, to get acquainted with the spectrum analyzer.

LINEAR MEASUREMENT

Set the Function Generator to [Sine], [Frequ] = 100 kHz, [Amp] = 200 mV_{P.P.}.

To set up the desired waveform properly we need to pay attention to the generator's output matching requirements (remember Exp. #1, Fig. 2).

- a-1) Connect the signal to the oscilloscope. Does it correspond to what the function generator's display indicates? Record!
- a-2) Connect the signal to the DMM. Measure on the DC range and on the AC range. Do these measured data correspond to the generator's display? Record!

dB MEASUREMENT

Connect the signal to the Spectrum Analyzer.

Set the Spectrum Analyzer as follows:

[Frequency], <Start frequency> = 0 kHz (press 0 on numeric pad, then <kHz>)
<Stop frequency> = 1 MHz
[Span], Should read 1MHz, i.e. horizontal base is 100 kHz/division
[Amplitude], <Ref Level> = 100 mV (press 100 on numeric pad, then <mV>)
<Scale Type> = 'Lin'

The bottom scale now reads frequency from 0 Hz to 1 MHz, i.e. 100 kHz/div.

The left (vertical) scale now reads from the Ref.level = 100mV on the top to 0 V at the bottom, i.e. 10mV/division.

- a-3) Close to the left end of the frequency scale, at $f = 100$ kHz, the analyzer shows the amplitude of the fundamental (or 1st harmonic) frequency component. Measure this amplitude.

Note:

the spectrum analyzer input impedance is 50 Ω

the spectrum analyzer displays the RMS-value!

Check the amplitude of this frequency component: does it correspond to the value you expect to see? Is it correct? Record!

Compare with the function generator's output display. Does everything make sense? Note!

B. Square Wave

Now, on the function generator, switch to [square wave].

LINEAR MEASUREMENT

The set-up is the same as for the sine wave. Now,

- b-1) On the spectrum analyzer, read and record the amplitudes of the fundamental and the harmonics up to the 9th harmonic.

For this measurement, you may like to use the 'Peak Search' function: Press [Peak Search], then <Next Right>, etc. (Some analyzers have only 'Search' printed on the button, but the function is the same).

dB MEASUREMENT

Normally, spectrum analyzer measurements are NOT done using a linear display as shown above, but rather using a logarithmic scale to increase the dynamic range. Amplitudes are displayed in *dBm* (this is a power measurement, displayed in dB with reference to 1mW), and the *differences* between amplitudes (power levels) can be read directly in *dB*.

To get familiar with this type of measurement, we repeat the experiment using the dB display.

Press [Marker] and <Off> to clear all markers from the screen.

Press [Amplitude] and change from <Lin> to <Log>. Amplitudes are now displayed in *dBm*.

If not already there, press <Ref Level> and move the peak of the fundamental or 1st harmonic to the top graticule line by turning the knob. The fundamental is now the *reference* against which all other harmonics are measured.

Since we know by now from the linear measurements of the square wave spectrum that only odd harmonics exist, move the marker - using the [Peak Search], <Next Pk right> routine - to the (frequency) location of the 3rd harmonic: The *difference* of the power level indicated now to the level measured for the fundamental shows how much the power level of the 3rd harmonic is below the level of the fundamental spectral component. The difference is measured in *dB*.

- b-2) Using the approach just described, measure the power levels of the fundamental and the harmonics up to the 9th.

Incidentally, a more convenient way of measuring the above is to set the fundamental as 0-dBm reference and read the harmonic's power levels directly. To do this, press [Peak Search]: the marker jumps to the highest peak. Then press [Marker] and <Delta>. This sets the level of the marker (at the fundamental) at 0 dB, and using the [Peak Search], <Next Pk right> routine allows to quite easily read the levels of the other harmonics directly with reference to the fundamental.

Note: the Delta function also sets the frequency of the fundamental to Zero – ignore since this is obviously not the case!

If you would like to save the screen display of the analyzer on disk, see Appendix A.

C. Pulse Train Wave

Disconnect the spectrum analyzer and press the green [Preset] button. The spectrum analyzer switches back to its default values.

The set up is the same as for the previous measurements. Again, pay attention to the generator's output matching requirements.

Assuming that we still have the square set up from the previous experiment, adjust the [Amplitude] to 350mVpp and [Offset] to +175 mV dc. The oscilloscope should display the bottom of the square wave to be on the GND level.

Now change the duty cycle to set the pulse waveform: [Shift] [% Duty], set to 20% [Enter].

This waveform has a dc spectral component which, too, has to be measured. For this you use the DMM.

Observing the oscilloscope and the DMM, check whether the displays show exactly what you expect them to show. If you *are* satisfied, go ahead with your measurements. If you are *not* satisfied, think what could be wrong and what could be done to remedy the problem. Remember the function generator's output characteristics...

Once everything is all right,
c-1) record the dc component

Then remove all present connections and connect the function generator directly to the spectrum analyzer.

Set the spectrum analyzer up similar as in part A, with the Start frequency at 0 Hz and the Stop frequency at 1 MHz. The frequency scale now reads from 0 Hz to 1 MHz with 100 kHz/div.

LINEAR MEASUREMENT

c-2) Measure the spectrum similar as done for the square wave (linear display), up to the 10th harmonic.

The analyzer's 'Peak Search' function may not mark very small components (smaller than about half a division). To get a measurement for these use the knob to move the marker or estimate directly from the screen.

dB MEASUREMENT

c-3) Measure the spectrum similar as done for the square wave (log display), up to the 10th harmonic.

Analysis

1. For both waveforms, compute the amplitudes of the fundamental and the harmonic frequency components, in the linear mode. Do the same for the log (dB) mode, but this time find the amplitude of the higher harmonics with *reference to the fundamental*, i.e. setting the fundamental to 0 dB and for the other harmonics find the difference to the fundamental in dB
Show at least one sample calculation for each waveform!
Don't forget the dc component.
2. Present both the measured and the computed data in tables for comparison.
3. Sketch the frequency spectrum for both waveforms on graph paper to scale, linear amplitude of each frequency component vs. frequency, for the number of harmonics measured. Use single lines to represent the frequency components.

Questions

4. When you did the measurements for the Square Wave using the Log-scale, the display most likely indicated the presence of harmonics not only at the odd frequency locations, but also at the locations of the even harmonics, though at a much lower level. What could be the cause of that, and is this acceptable? Consider the power level of these even harmonics shown by the analyzer, and also compare with the linear display!
5. Consider a common telephone channel with the bandwidth limited from 300 Hz to 3400 Hz by an ideal band pass filter. You apply a square wave of 400 Hz and 1V amplitude to the input. At the output of this filter, what would
 - a) the signal waveform approximately look like (time domain)? Sketch!
 - b) the spectrum look like (frequency domain)? Sketch!

Conclusions

Comment on your results.

AMPLITUDE MODULATION (AM)

Objective

To investigate the characteristics of an amplitude-modulated wave, and to compare the results with theory.

Theory

See reference No.1, (class text), chapter 8, and in particular sections 8.1 - 8.5.

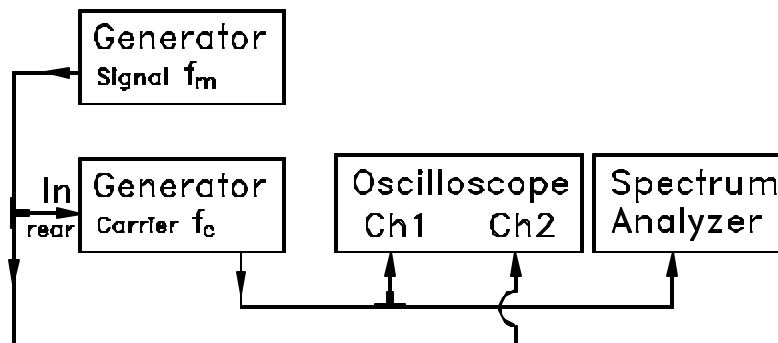
Experiment

Equipment

2 Function Generators HP 33120A (or one HP and one other)	2 BNC T-adapters
Oscilloscope HP 54645D	2 short BNC cables
Spectrum Analyzer HP E4411B	2 med. BNC cables

Basic Set-up

The HP function generator is used to set up the carrier wave. A second generator provides the modulating signal and externally modulates the carrier wave.



The Oscilloscope provides an amplitude-time display (*time domain*), while the Spectrum Analyzer provides an amplitude-frequency display (*frequency domain*).

CARRIER SET-UP

- Connect the HP function generator *providing the carrier* to channel 1 of the oscilloscope.
- Set the carrier to $f = 300 \text{ kHz}$, $A = 200 \text{ mV}_{\text{p-p}}$.
Note: The scope shows 2x the amplitude displayed by the generator since it represents an impedance almost equivalent to open circuit. However, this is of no consequence in this case, because we dealing with ratios of amplitudes.
- Press [Shift][Recall Menu], display shows "A:MOD MENU"
- Press [ω], display: "1:AM SHAPE"
- Press [$\rightarrow$], display: "2:AM SOURCE"
- Press [ω], display: "EXT/INT"
- Press [$\rightarrow$], display: "EXT"
- Press [Enter]
- Press [Shift] [AM], display: "AM,EXT"
i.e., now the generator will be AM-modulated by the external input only.

MODULATING SIGNAL SET-UP

- On the function generator *providing the modulation signal*, set $f = 10 \text{ kHz}$, $A = 2\text{V}_{\text{p-p}}$.
- Using a BNC T-adaptor, connect this signal to the external input at the rear of the carrier HP generator (labeled "AM Modulation") as well as to CH2 of the oscilloscope. (An external amplitude of $\sim 5\text{V}_{\text{p-p}}$ provides 100% modulation. The *actual amplitude* of the signal *applied to the modulator circuit of the carrier* is of course much smaller and can only be found from the oscilloscope waveform displays.)
- On the HP scope, press [Autoscale], adjust the time base for a convenient display. If the display can't be made stable, trigger the other channel or press [Run/Stop].

SPECTRUM ANALYZER SET-UP

- Set [Frequency] to the carrier frequency and [Span] to 50 kHz.
- On [Amplitude], change 'Scale Type' from <Log> to <Lin>. Press [Ref Level] and adjust the carrier amplitude until the peak of the carrier just reaches the top graticule (= reference) line. Thus the carrier is now normalized to 1, or 10 divisions.

Section One : Amplitude-Time Representation (Time Domain)

We first ignore the spectrum analyzer and concentrate on the oscilloscope to investigate the two common methods to find the modulation index m :

- a) Amplitude-Time Method
- b) Trapezoidal Method

Make adequate sketches throughout the experiment!

a) Amplitude-Time Method

Move the channel 2 position so that the trace of the modulating signal lies on top of the AM-envelope. Press [A2], then switch 'Vernier <ON>'. Now trace 2 can be adjusted in small increments and you can verify that the modulation signal trace corresponds exactly to the AM envelope trace.

Then switch channel 2 <OFF>.

Find the modulating index for the following conditions:

- a-1) Set the amplitude of the modulating generator to $1V_{p-p}$, sinusoidal waveform.
 - record the amplitude of the carrier in V_{p-p} from the scope display by temporarily disconnecting the modulation signal (if the display is unstable change trigger to channel 1 by pressing [Edge] and <A1>
 - record the amplitude of the modulating signal in V_{p-p} from the *scope display* (not from the modulating generator display!)
 - make sketches of the waveforms, indicating your measurements
- a-2) Set the amplitude of the modulating generator to $2V_{p-p}$ and change the modulating signal to a square wave.
 - repeat the measurements as in a-1.
- a-3) Set the amplitude of the modulating generator to $3V_{p-p}$ and change the modulating signal to a triangular wave.
 - repeat the measurements as in a-1.
- a-4) Set the amplitude of the modulating generator to $4V_{p-p}$ and change the modulating signal to a ramp wave.
 - repeat the measurements as in a-1.
- a-5) Set the amplitude of the modulating generator to $5V_{p-p}$ and change the modulating signal back to a sine wave. If over-modulation occurs, lower the amplitude so that 100% modulation is achieved (i.e. the upper and lower envelopes just touch)
 - repeat the measurements as in a-1.

b) Trapezoidal Method

On the oscilloscope, clear all cursors, switch Ch2 'on' and adjust the scale to observe the modulating signal.

To see the trapezoidal display,

- On the HP scope, exchange CH1 and CH2 connections, press [Autoscale], [Main/Delayed], then <XY>

- b-1) For the same conditions as stated in a-1, use eq. 8.3.2 class text to find m.
The size of the trapezoidal display can be changed for better viewing by varying the vertical deflection of Ch1 and Ch2.
Make sketches as above.
- b-2) through b-5)
Repeat this measurement under the same conditions as given in a-2 through a-5.

When finished, go back to the amplitude-time display:

- On the HP scope, clear cursors, press [Main/Delayed], then <Main>. Change time base for convenient display and if necessary adjust the trigger : [edge], <A1>

Section Two : Amplitude-Frequency Representation (Frequency Domain)

Make adequate sketches throughout the experiment!

Now concentrate on the Spectrum Analyzer. Begin with the <Lin> display.

What you see there now is the amplitude spectrum for a sinusoidally modulated carrier wave, as shown in Fig. 8.5.1, class text.

Set [Span] back to 100 kHz , and the modulating generator to the sine wave form.

*The following measurements are for **sine wave** only.*

Remove the modulation (disconnect the modulating generator) and observe the carrier on the screen: re-adjust the peak of the carrier to the top graticule line if necessary, turning the knob *slowly (and waiting a second or two until the analyzer has caught up with the new level)*.

Now the carrier can be considered normalized to 1, scaled to 10 divisions.

- c-1) Re-connect the modulating signal and set it to $2V_{p-p}$.
Record the amplitude of the side frequencies, in terms of % or divisions of the unmodulated carrier.
- c-2) Set the modulating signal to $3.5V_{p-p}$ and repeat the measurement as in c-1.
- c-3) Set the modulating signal to $5V_{p-p}$ and repeat the measurement as in c-1.
- c-4) Change display from 'Lin' to 'Log'. The carrier should be at the top graticule line. For the same conditions as in c-1 through c-3, measure and note the amplitude level of the sidebands *relative to the carrier* (in dB).
(How? Remember Experiment No. 2)

Section Three : Application

Record the inventory number on the top front edge of spectrum analyzer (EA-xx)!

Disconnect the input cable to the spectrum analyzer.
Press [Preset] to put the analyzer in its default condition.

Press [File], then <Load>, then <Trace>.

Drive C: should be highlighted.

Turn knob to highlight the filename. In our case, the desired filename for 2nd year students is "AM2", for 3rd year students "AM3". *Record the filename you used !*

Press [Enter]

From the display, record

- d-1) center frequency
- d-2) span
- d-3) positions (frequencies) of the lower and upper side frequency
- d-4) scale type (Lin or Log)
- d-5) with the carrier set to reference (= top) line, record the amplitudes of the side frequencies with reference to the carrier

Analysis

1. Collect all data in a suitable form in a clearly arranged table.
2. *Section One, part a)*, amplitude-time method:
Using eq. 8.3.1. class text, find the modulating index m for the given modulating conditions a-1 to a-5. Show how you did that (sketches, sample calculation)

3. *Section One, part b*), trapezoidal method:
Using eq. 8.3.2 class text, find m for the given modulating conditions b-1 to b-5. Show how you did that (sketches, sample calculation)
4. *Section Two*:
Using eq. 8.5.1 class text, find m for the given modulating conditions c-1 to c-3 ('Lin'-display). Show a sample calculation and sketch the spectrum.
5. *Section Two*:
Show the relation of the side frequencies to the carrier, *with the carrier set to 0 dB as the reference*, i.e. the difference in dB ('Log'-display) between side frequencies and carrier. Show a sample calculation and sketch the spectrum.
6. *Section Three*:
From the recorded data, find
carrier frequency f_c ,
modulating signal frequency f_m ,
modulating index m ,
bandwidth

Conclusions

Comment on differences, advantages and disadvantages of the amplitude-time and the trapezoidal method

Questions

1. To control an AM radio station's signals, which type of measuring method, amplitude-time or trapezoidal, would you use to monitor m , and why?
2. If an amplitude-modulated carrier of amplitude 1V is set at 1 MHz and the function generator supplying the modulating signal would be swept with a sinusoidal signal over a range of 2 kHz to 20 kHz and $m=60\%$, what would
 - a) the linear spectrum look like (sketch, with amplitude to scale)?
 - b) the bandwidth be?

FREQUENCY MODULATION (FM)

Objective

To study some aspects of frequency modulation.

Background

Compared to amplitude modulation, frequency modulation provides enhanced noise performance, i.e. a better signal-to-noise ratio, at the expense of increased bandwidth. Together with amplitude modulation, FM is one of the classic modulation techniques.

Theory

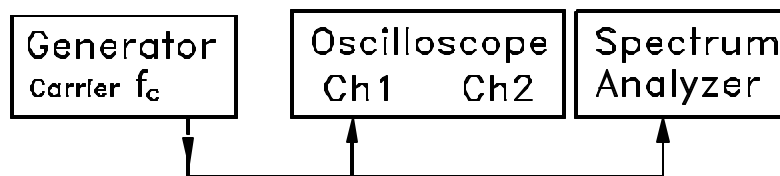
See Reference No.1, class text, Chapter 10, sections 10.1 - 10.5.

Experiment

Equipment

Function Generator HP 33120A
Oscilloscope HP 54645D
Spectrum Analyzer HP E4411B

BNC T-adaptor
2 short BNC cables



Notations and formulas used

Carrier frequency	f_c
Modulating frequency	f_m
Frequency deviation	Δf
Modulation Index	β
Bandwidth	B_{FM}

Modulation Index
$$b = \frac{\Delta f}{f_M} \quad (\text{Eq. 4-1})$$

Bandwidth
$$B_{FM} = 2(\Delta f + f_M) \quad (\text{Eq. 4-2})$$

Eq. 4-2 is known as Carson's Rule. The occupied bandwidth is usually considered using the side frequencies larger than 1% of the unmodulated carrier (as a guideline).

FM Spectrum

Set the function generator up as follows:

carrier frequency	$f_C = 100 \text{ kHz}$
carrier amplitude	$A = 200 \text{ mV}_{P-P}$
select [Shift][FM]	
modulating signal frequency	$f_M = 10 \text{ kHz}$
max. deviation	$\Delta f = 5 \text{ kHz}$

Observe the resulting waveform on the oscilloscope (try time base $20\mu\text{s}$). It is not all that informative, but at least gives an impression what a frequency modulated signal looks like.

However, we are more interested in the frequency spectrum. Thus, the spectrum analyzer is the instrument exclusively used for our measurements in this experiment.

A) Amplitude level measurements

On the Spectrum analyzer, set

- [Frequency] <center frequency> to f_C ,
- [Span] to a suitable range
- [Amplitude] to <Lin>

On the generator, press [Shift] [FM], to switch the modulation off. The scope now shows the unmodulated carrier.

On the spectrum analyzer, press <Ref Level> and adjust the carrier peak to the top graticule (= reference) line. This corresponds to the amplitude of the *unmodulated* carrier's spectrum being normalized to 1, over 10 vertical divisions.

- a-1) With Δf still set to 5 kHz, switch the modulation on again and record the amplitudes of the sinusoidal spectrum components of the modulated carrier and the side frequencies, with the normalized carrier as reference. (In practical terms, measure the divisions: a side frequency with an amplitude of for example 3 divisions equals $0.3 \times$ the carrier amplitude).

It is sufficient to record only the *significant* side frequencies. (*Consider significant side frequencies as those with an amplitude of 1% or more of the unmodulated carrier amplitude*).

Record the positions of the side frequencies on the frequency scale with regard to the carrier, that is, the distance of the side frequencies to the carrier frequency.

- a-2) Change the deviation to $\Delta f = 24 \text{ kHz}$.

Repeat the above measurements (you may want to change Span to a more convenient value to accommodate all significant side frequencies).

- a-3) Change the deviation to $\Delta f = 40$ kHz.
Repeat the above measurements, again changing Span if necessary.

B) Power level measurements

On the spectrum analyzer [Amplitude] menu, switch from <Lin> to <Log>.

Switch FM off again and verify that the unmodulated carrier aligns to the top graticule line as the reference level.

Press [Marker] and <Delta>. This sets the carrier reference to 0 dBm and allows the *difference of the amplitudes of the side frequencies to the amplitude of the unmodulated carrier* conveniently to be recorded in dB. The [Peak Search] or [Search] function may be useful, too.

Switch FM on again and for the same number of side frequencies as seen on the linear display, or seen on the Bessel's table, record the levels of the sinusoidal spectrum components with reference to the *unmodulated carrier* (i.e. the difference) in dB :

- b-1) Do these measurements under the same modulation conditions as done in a-1.
b-2) Repeat the measurements with the same modulation conditions as in a-2.
b-3) Repeat the measurements with the same modulation conditions as in a-3.

C) Deviation

As on the oscilloscope, the deviation of a carrier is not readily observed on the spectrum analyzer (switch to linear display) under realistic modulation conditions. *For the purpose of demonstration only*, set up conditions as follows:

$$\begin{aligned}f_C &= 100 \text{ kHz} \\f_M &= 0.1 \text{ Hz (100 mHz)} \\ \Delta f &= 10 \text{ kHz} \\ \text{span} &= 100 \text{ kHz}\end{aligned}$$

Switch the FM modulation off. Note the position of the carrier.
Switch the FM modulation on again. Make notes of what is happening.

Change f_M to 0.2 Hz, then to 0.3 Hz. Make notes of observations.
Change Δf to 20 kHz, then to 30 kHz. Make notes of observations.

D) Bandwidth

We now have a look at the bandwidth occupied by the spectrum. We set up a carrier and modulate it with various modulating signal frequencies while keeping the deviation constant.

Take all [Markers] <Off>.

Set $f_C = 500$ kHz, amplitude = 200 mV_{p-p}
 $\Delta f = 75$ kHz
Center frequency to f_C
Span to 200 kHz
Amplitude to <Lin>
Ref level to 10 mV

- d-1) Set $f_M = 100$ Hz
Measure the approximate bandwidth. Easiest way to do this:
Press [BW/Avg], change <Resolution Bandwidth> from <Auto> to <Man>. Then turn knob to set resolution bandwidth to 1 kHz.
Press [Marker], , then set to . Turning the knob symmetrically moves two markers and the frequency difference (bandwidth) can be read directly off the screen.
- d-2) Set $f_M = 1$ kHz
Repeat the above measurement.
- d-3) Set $f_M = 10$ kHz
Repeat the above measurement. Change [Span] if necessary.

E) Application

Record the inventory number on the top front edge of the spectrum analyzer (EA-xx)!
Disconnect the signal input from the spectrum analyzer.

Load a file:

Press [Preset] to put the analyzer into the default state.
Press [File] and <Load>, then <Trace>
Drive C: should be highlighted. If not, change to C: with <Select> function
Turn knob to select the filename. In this case the desired filename for 2nd year students is "FM2", and for 3rd year students "FM3". *Record the filename you used!*
Then press [Enter].
The file should be loaded and the screen should show some spectrum.

Record carrier and side frequency amplitude levels and the spectrum component's frequency locations, i.e. the complete spectrum (assume the unmodulated carrier level was set to the top graticule line as reference)
Record also center frequency and span.

Bessel's table:

Mod Index β	Carrier J_0	Side Frequencies						
		J_1	J_2	J_3	J_4	J_5	J_6	J_7
0.25	0.98	0.12	0.01					
0.5	0.94	0.24	0.03					
1.0	0.77	0.44	0.11	0.02				
1.5	0.51	0.56	0.23	0.06	0.01			
2.0	0.22	0.58	0.35	0.13	0.03	0.01		
2.4	0	0.52	0.43	0.20	0.06			
3.0	-0.26	0.34	0.49	0.31	0.13	0.05	0.02	
4.0	-0.40	-0.07	0.36	0.43	0.28	0.13	0.05	0.02

*Un-modulated
carrier
amplitude
= 1.0*

Negative values are of mathematical interest only. For the modulation always use the absolute value, of course.

Analysis

Part A

- For the modulating conditions given in a-1 through a-3, find the modulating index β from the Bessel's table.
- From the Bessel Functions table, which lists the amplitude coefficients of the modulated carrier and the side frequencies with regard to a normalized carrier, find the theoretical amplitude levels corresponding to your measurements.
- Neatly tabulate the experimental together with the theoretical data for convenient comparison.

Part B

- Similarly to part A, show the results in terms of dB.

Part C

- Interpret your observations.

Part D

- For the given modulating conditions, compute the bandwidth of the occupied spectrum using Carson's Rule.
- Compare your experimental data with the computed data.
- With regard to part D, how closely do the measured bandwidth and the bandwidth given by Carson's Rule correlate? What conclusion would you draw from these results with regard to the validity of Carson's Rule?
- Consider the same variation of f_M as in part D applied to an AM signal. How does it affect the bandwidth here, compared to FM?

Part E

- From your recorded data, find f_C , f_M , Δf and β .

Conclusions

Comment on the experiment and your results.

PULSE CODE MODULATION (PCM)

Objective

To investigate aspects of quantization, synchronization, companding and multiplexing processes in a PCM system.

Background

To make use of the many advantages of transmitting digital signals over transmitting analog signals (speed, precision, security etc.), a number of schemes are available to modulate a pulse train with the desired information.

The most important of these methods is Pulse Code Modulation where an analog signal is sampled and "translated" into a binary signal consisting of strings of a fixed number of On- or Off-states.

Theory

See Reference No.1, class text, Chapter 11.1-11.3 (sections about quantization, compression and receiver).

For more detailed information regarding this particular circuit see the the following outline, schematics and the attached extract from the manufacturer's data book.

Experiment

System Description

Read the description and details presented in the specification sheets and study the circuits. The board layout on (Fig.1) should help you find your way through the experiment.

Condensed Outline

The system is provided on a PC board and self-contained except for a power supply and measuring equipment. The actual system consists mainly of two ICs, a CODEC (COder/DECoder) and an IC containing filters. Both are designed by the manufacturer to work together as a unit. One set handles both the transmission (coding) as well as the reception (decoding).

The analog signal enters the board at A_N and passes through a filter via a switch into the CODEC. The switch allows a DC voltage (also provided on the board) to enter the CODEC instead of the analog signal. The CODEC samples the incoming signal at a rate of 8 kHz, assigns an 8-bit digital number to the sampled level and shifts this number out

to be transmitted via PCM_{TX} . In our case, the signal is looped right back. The signal received at PCM_{RX} , i.e. the received digital bit stream, is decoded by the CODEC, resulting in a sort of stepped waveform, the steps corresponding to the received pulse sequence. This waveform is passed through a filter to smoothen out the steps and now, representing the original signal, is finally available at A_{OUT} .

The CODEC may be used in multiplexed systems. Each CODEC can handle one out of 32 available channels of 8 bits each. The 32 channels times 8 bits each make up one **frame**. A frame synchronizing pulse derived from the system clock marks the start of a new frame. As mentioned above, one digital word is 8 bits long.

As mentioned above, one of 32 channels (called time slots in the specs) can be assigned to one CODEC (see the table in the attached codec spec sheet). Each CODEC has to be programmed to assign a time slot for its channel, and in which mode the CODEC is to be operated in this slot, i.e. to encode, decode, do both or to be powered down. A programmer circuit designed for this purpose is integrated on the board. The desired time slot and mode is set with a multi-switch. Pressing the time slot programmer pulse TS push button switch sends control clock pulses and an 8-bit series of control data pulses to the CODEC initiating the assignment. The whole process is controlled by the frame synchronizing pulse FS occurring at an 8 kHz rate.

Procedure

Equipment:

- PC-board
- Power Supply
- Oscilloscope
- DMM
- Function Generator

Use the 10x probes with the oscilloscope at all times.

Make notes and sketches of your observations throughout the experiment!

This is important since it will help to clarify the various concepts encountered in the experiment, which simulates a telephone link as it is used today.

1. Set-Up

The CODEC handles transmitting AND receiving at the same time. We use this feature by feeding our transmitted PCM right back to be received again. This allows an easy comparison of the outgoing and the received signal.

Therefore, set the mini switches to assign the time slot mode to "Encoder and Decoder" ($B_1 = 0$, $B_2 = 0$), and to time slot 1 ($B_3 \dots B_8 = 0$). Note that setting a "0" in this circuit is to set the switch to "ON" (IC 15, Fig. 4).

If not already in place, set the jumper-shunts to ' A_N ' and 'Not Signalling'.

Apply a sine wave approximately 2 kHz to the 'A_N' BNC-input, with an amplitude of 350mV_{P-P} measured at TP2 (remember: the scope probes are 10x). This provides a signal of approx. 2 V_{P-P} applied to the CODEC input.

2. Signal path through the system

To get an overlook of the whole system, follow a signal applied to the input through the whole system until received and re-constructed at the output:

Connect CH1 probe to A_N (TP1). Trigger the oscilloscope on CH1 and adjust the time base to observe the incoming signal.

Connect the CH2 probe to TP2, showing the signal after the filter. It is unchanged except for the amplitude since the filter provides some amplification. This (analog) signal enters now the CODEC.

Unfortunately, nothing can be seen of the intermediate steps applied to the signal by the CODEC (quantization, pulse amplitude modulation, etc.), until the PCM pulse train appears at the output at PCM_{TX} (TP4). The output looks somewhat strange, but we will have a closer look at that shortly.

Feeding this signal back into the receiver part of the CODEC, the output of the CODEC shows the decoded signal at TP6. The waveform on the oscilloscope may not look like much of the original signal yet. On the function generator, change the frequency to approx. 300 Hz, adjust the trigger and Hold Off to get a stable display. The waveform can now readily be observed and reminds the observer of a staircase shape. This shows clearly how the CODEC now performs the reverse procedure of quantizing, assigning a discrete voltage level to each particular 8-bit number or sequence.

Connect the other probe to A_{OUT} (TP7). The original signal is reconstructed, showing a fairly clean sine wave after passing through the filter, and also some amplification with regard to the input signal.

Change the frequency back to the previous value

3. FS-Puls

Connect CH1 probe to the Frame Synchronizing pulse (TP3). Adjust V/div to see the pulse, and expand the time base so that the pulse is approximately 1 division in length.

Connect CH2 probe to the transmitted PCM_{TX} (TP4). You will see then a particular pattern on the scope screen, consisting of an upper trace (1-level) and a lower trace (0-level). You may have to adjust V/div to observe this properly. Turning the intensity fully clockwise you can also distinguish clearly a set of 8 bits, representing a binary number related to voltage level at the sampling moment. Of course, since the sine wave level changes continuously, the eye is unable to follow this at a 8 kHz rate and perceives to see all 8 bits at the same time.

Temporarily connect CH2 probe to the system clock (TP10) and compare the clock period to

- a) the width of the FS-pulse (TP3), and
- b) the width of one bit of the PCM-Signal (TP4)

Since the FS-pulse synchronizes the whole system time-wise, this pulse lends itself as the best trigger source for most measurements on the scope.

4. PCM Signal

Under procedure No. 3, it has been observed that the PCM signal consists of an 8-bit sequence for a certain voltage level, but because of the time varying nature of the applied sine wave the value of any single bit (1 or 0) cannot be observed.

If we want to have a closer look at the PCM signal we therefore need to find some special arrangement to "slow things down".

To see the PCM sequence for a particular voltage level would require the continuous sampling of the *same* level, thus encoding the *same* 8-bit 'word' or number every time. This can be done using a little "trick" by applying a variable DC source as the input signal. Then the sampled signal level can be left constant or be changed in any desired interval or step and the 0- or 1-value of each bit of the 8-bit number (sequence) representing this level can be observed on the screen.

Set all mini switches $B_1 \dots B_8$ to "0" (On), i.e. assign time slot 1.

Change the jumper from A_{IN} to EXT DC IN. The DC voltage level can be monitored on the voltmeter and should cover the same range as the peak-peak AC signal, that is $\pm 2V_{DC}$. Like the AC signal before, the DC voltage is now applied to the CODEC input.

Trigger on CH1 (FS-pulse). Adjust the time base to display the FS-pulse, with a width of one division.

Connect Ch2 probe to PCM_{TX} (TP4). Turn the DC control (on the board) and you can now see the 8 bits taking values of 0 or 1, depending on the DC voltage level.

Since with this time base setting 1 bit is approximately 1 division wide, it is easy to read the 8-bit sequence. The falling edge of the FS-pulse marks the beginning of the first bit (the MSB; the LSB is the one at the right end).

5. Time Slot

The 8-bit sequence represents the first of 32 time slots (= channels) available. Other time slots may be assigned using the table provided in the spec sheet of the CODEC making it possible to run 32 CODECs or 32 lines at one time (Multiplexing). (See the table on the CODEC spec sheet).

Decrease the time base until you can see one complete frame (i.e. two FS-pulses). The 8-bit sequence or number is now 'squashed' together but still quite distinguishable, and representing time slot 1.

Set the mini switch B_8 to "1" (i.e. to "Off").

Pressing the TS-button initiates the assignment of the new time slot. On the screen you see the 8-bit sequence move to the right, into the second time slot.

Setting $B_7 = 1$, $B_8 = 0$ and pressing TS moves the 8-bit number further right, to the third time slot. Setting $B_7 = 1$, $B_8 = 1$ assigns time slot no. 4, and so on.

Setting $B_4 \dots B_8$ to "1" assigns the 32nd time slot.

Assign time slot 1. Expanding the time base again you can see that the MSB (the first bit counting from the left) starts with the falling edge of the FS-pulse. Assign slot 32. Now you can observe that the LSB (the 8th bit counting from the left) ends with the falling edge of the FS-pulse.

6. Companding

To enhance the S/N-ratio of a small signal on the transmission channel a process of COMpressing the signal before transmission and exPANDING it upon reception is frequently being used, the combination of both thus called COMPANDING. Instead of dividing the range from -2V to +2V into 256 equal steps for the A/D conversion, the range is divided into segments with progressively smaller voltage increments from ± 2 V towards 0 Volts.

In our experiment we will have a close look at the compression part. We could include the expansion part, too, but it will be omitted here due to time limitations in the lab.

Assign time slot no. 1.

Move the shorting bridge from ' A_{IN} ' to 'Ext DC In'.

Vary the DC voltage and note the binary sequence according to the following table. These are the minimum number of measurements, taking more readings will provide improved results.

See table 5-1 to collect the data. The resulting curve is somewhat idealized but provides a good perception of the principles involved.

Table 5-1 :

$V_{DC} \text{ (IN)}$ [M]	Binary Sequence MSB LSB	Decimal Equivalent to the 8-bit sequence
- 2.00 - 1.80 - 1.60 - 1.50 - 1.40 - 1.20 - 1.00 - 0.80 - 0.60 - 0.50 - 0.40 - 0.33 - 0.26 - 0.20 - 0.10	0 0 0 0 1 0 1 1	11
~ < 0 ~ > 0	0 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1	127 255
+ 0.10 + 0.20 + 0.26 + 0.33 + 0.40 + 0.50 + 0.60 + 0.80 + 1.00 + 1.20 + 1.40 + 1.50 + 1.60 + 1.80 + 2.00	1 0 0 0 1 0 1 1	139

Analysis

1. Describe and sketch the steps of a signal passing through the system, from input signal to transmitted signal, from received signal to output.
2. Compare and sketch the relationship between clock pulses, FS-pulse and PCM signal sequence.
3. Sketch a full frame showing the 32-channel organization, with respect to the FS-pulse.

4. Using your data from procedure no.6, plot Decimal Equivalent (ordinate) vs. $V_{DC(IN)}$ (abscissa) on graph paper.
Note the "flip" of the binary sequence and the decimal equivalent at 0 Volts. That means you need to adjust your ordinate scaling to accommodate a smooth transition!
The shape of the graph is the result of 'compressing', in our case according to a signal processing scheme called ' μ -law'. When the signal is reversed it undergoes a similar decompressing process, restoring the original form.

Conclusions

Comment on your experiment

Questions

1. What is the purpose of the input filter?
2. What is the purpose of the output filter?
3. To show the bit sequence for the companding part, we changed from an AC to a DC signal. Why was this necessary?
4. Demonstrating on your graph of the compression process, and in your own words, attempt a short interpretation of the purpose and effect of this feature.

PC BOARD LAYOUT

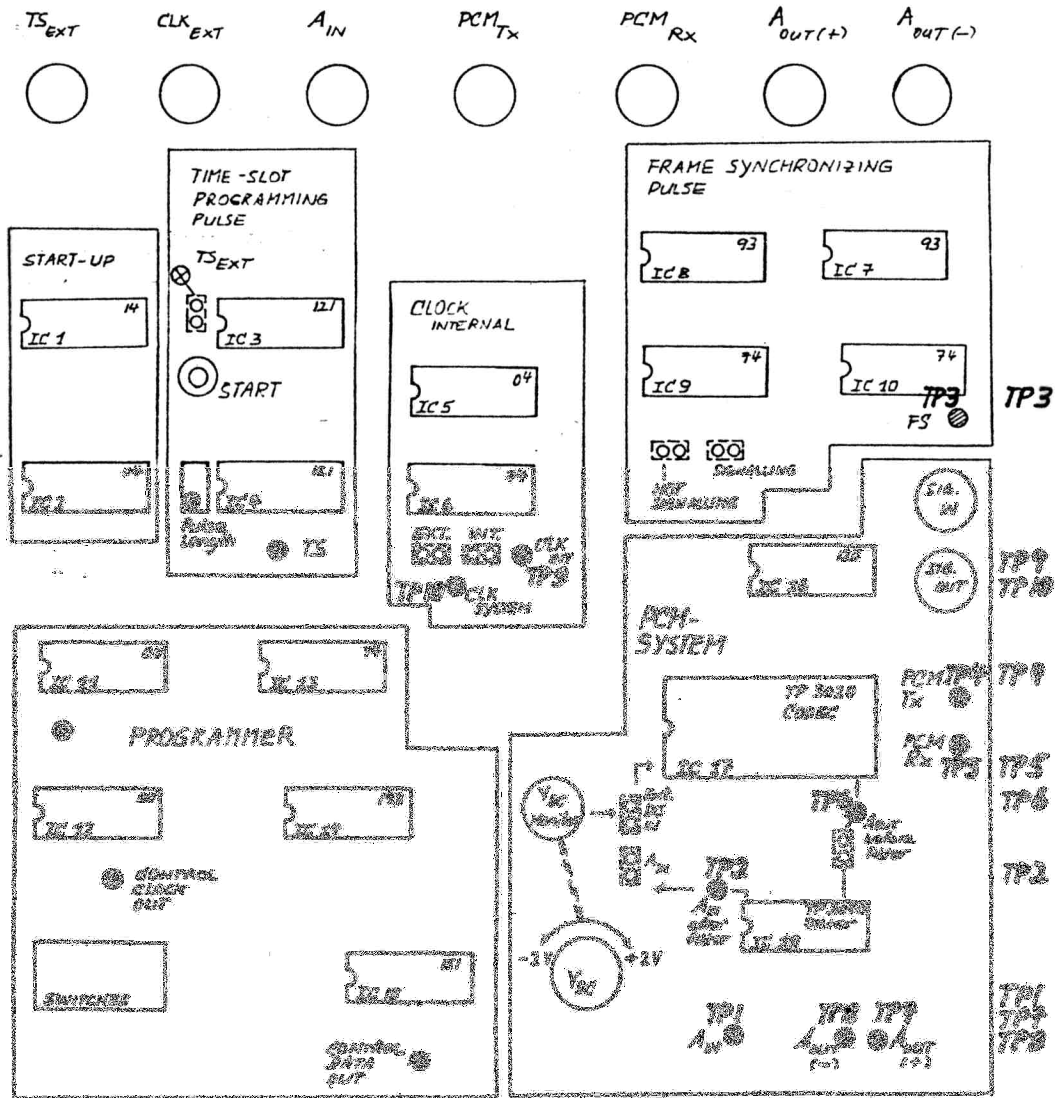


Fig. 1



Fig. 3

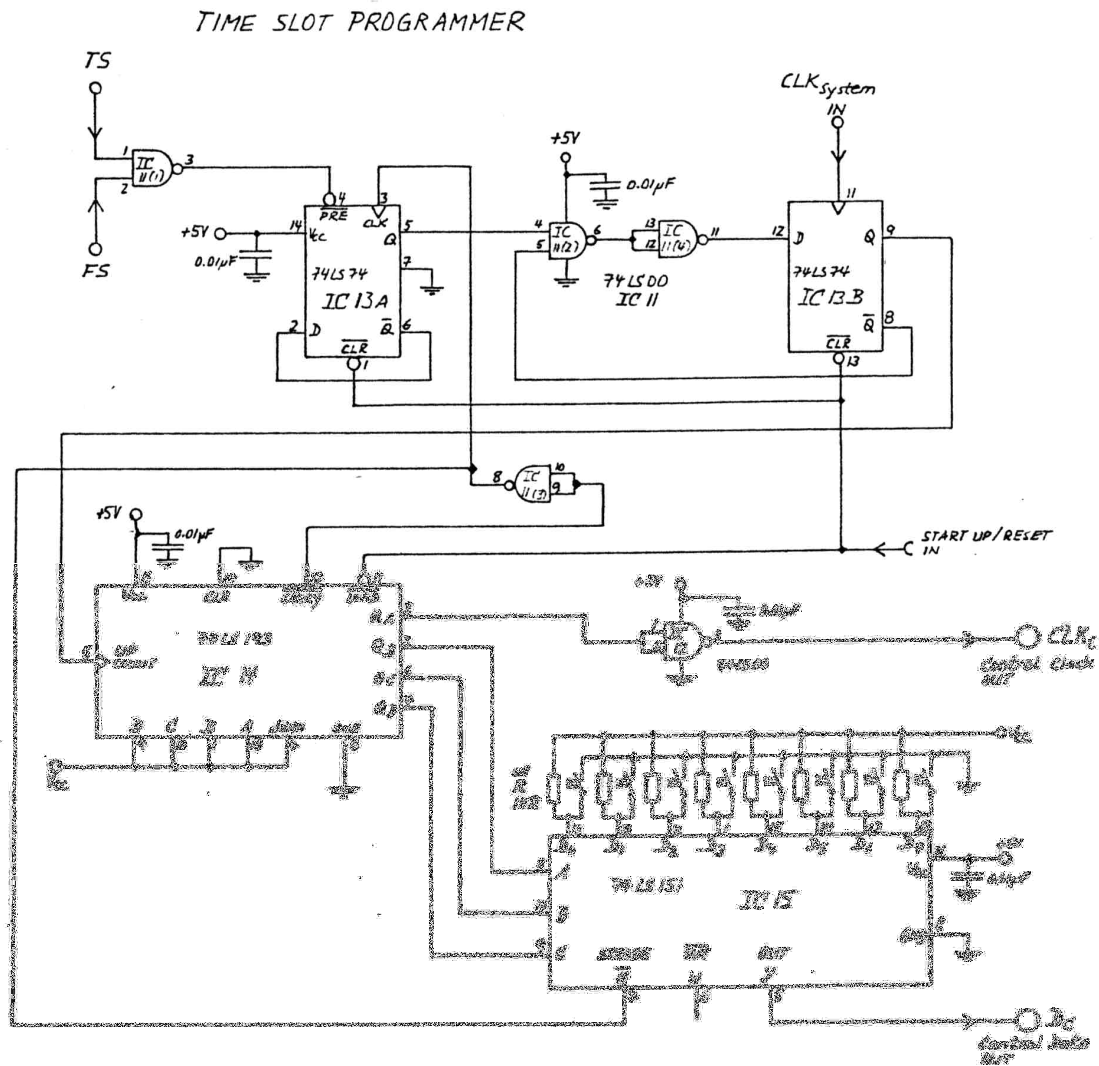


Fig. 4

TP3020, TP3020-1, TP3021, TP3021-1 Monolithic CODECs

General Description

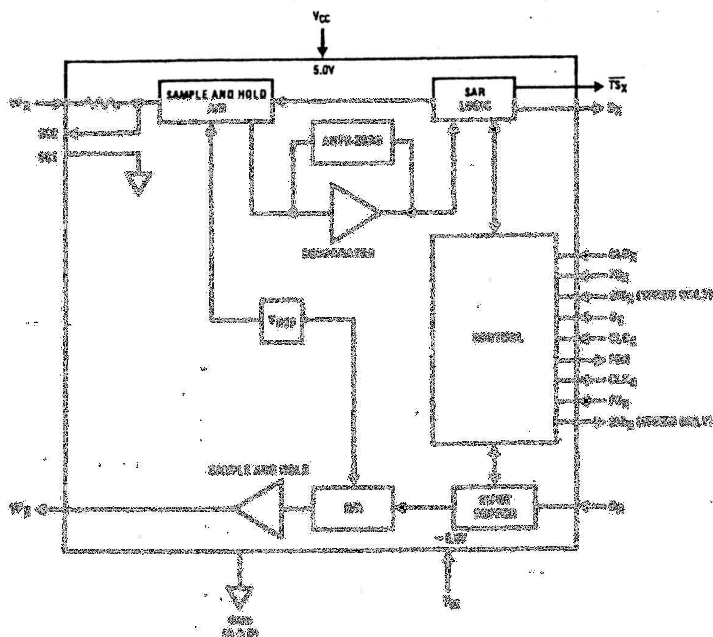
The TP3020 and TP3021 are monolithic PCM CODECs implemented with double-poly CMOS technology. The TP3020 is intended for μ -law applications and contains logic for μ -law signaling insertion and extraction. The TP3021 is intended for A-law applications.

Each device contains separate D/A and A/D circuitry, all necessary sample and hold capacitors, a precision voltage reference and internal auto-zero circuit. A serial control port allows an external controller to individually assign the PCM input and output ports to one of up to 32 time slots or to place the CODEC into a power-down mode. Alternately, the TP3020/TP3021 may be operated in a fixed time slot mode. Both devices are intended to be used with the TP3040 monolithic PCM filter which provides the input anti-aliasing function for the encoder and smoothes the output of the decoder and corrects for the $\sin x/x$ distortion introduced by the decoder sample and hold output.

Features

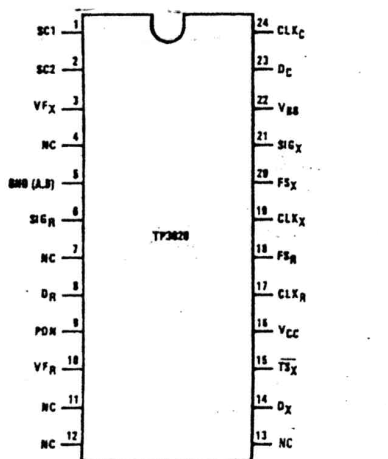
- Low operation power—45 mW typical
- Low standby power—1 mW typical
- $\pm 5V$ operation
- TTL compatible digital interface
- Time slot assignment or alternate fixed time slot modes
- Internal precision reference
- Internal sample and hold capacitors
- Internal auto-zero circuit
- TP3020— μ -law coding with signaling capabilities
- TP3021—A-law coding
- Synchronous or asynchronous operation

Simplified Block Diagram



TP3020/TP3021-1

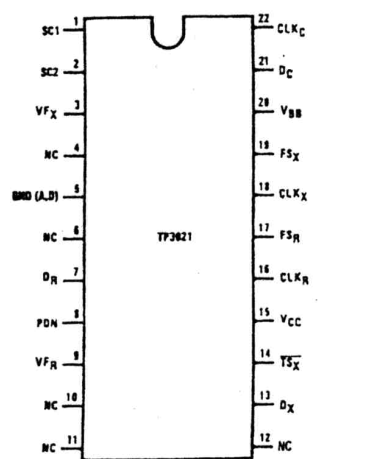
Connection Diagrams



TL/H/5538-3

Top View

Order Number TP3020J or TP3020J-1
See NS Package Number J24A



TL/H/5538-4

Top View

Order Number TP3021J or TP3021J-1
See NS Package Number J22A

Description of Pin Functions

Symbol	Function
1 SC1	Internally connected to GNDA.
2 SC2	Connects VF_X to an external sample/hold capacitor if fitted for use with pin-compatible NMOS CODECS. Ensures gain compatibility.
3 VF_X	Analog input to the encoder. This signal will be sampled at the end of the encoder time slot and the resulting PCM code will be shifted out during the subsequent encode time slot.
5 SG_R	Analog and digital ground. All analog and digital signals are referenced to this pin.
6 SG_X	Receive signaling bit output. During receive signaling frame the least significant (last) bit shifted into D_X is internally latched and appears at this output— TS_X will then remain valid until changed during a subsequent receive signaling frame or reset by a power-down command.
8 D_X	Serial PCM data input to the decoder. During the decoder time slot, PCM data is shifted into D_X , most significant bit first, on the falling edge of CLK_X .
9 PDN	TTL output level which goes high when the CODEC is in the power-down mode. May be used to power-down other circuits associated with the PCM channel.
10 VF_R	Analog output from the decoder. The decoder sample and hold amplifier is updated approximately 15 μ s after the end of the decode time slot.
21 SG_X	Transmit signaling input. During a transmit signaling frame, the signal at SG_X is shifted out of D_X in place of the least significant (last) bit of PCM data.
22 V_{BB}	-5V ($\pm 5\%$) input.
23 D_C	Serial control data input. Serial data on D_C is shifted into the CODEC on the falling edge of CLK_C . In the first time slot mode, D_C functions as a power-down input.

Symbol	Function
NC	Unused
14 D_X	Serial PCM TRI-STATE® output from the encoder. During the encoder time slot, the PCM code for the previous sample of VF_X is shifted out, most significant bit first, on the rising edge of CLK_X .
15 TS_X	Time slot output. This TTL compatible open-drain output pulses low during the encoder time slot. May be used to enable external TRI-STATE bus drivers if highly capacitive loads must be driven. Can be wire ANDed with other TS_X outputs.
16 V_{CC}	5V ($\pm 5\%$) Power Supply.
17 CLK_X	Master decoder clock input used to shift in the PCM data on D_X and to operate the decoder sequencer. May operate at 1.530 MHz, 1.544 MHz or 2.048 MHz. May be asynchronous with CLK_X or CLK_C .
18 FS_R	Decoder frame sync pulse. Normally occurring at an 8 kHz rate, this pulse is nominally one CLK_X cycle wide. Extending the width of FS_R to two or more cycles of CLK_X signifies a receive signaling frame.
19 CLK_C	Master encoder clock input used to shift out the PCM data on D_X and to operate the encoder sequencer. May operate at 1.530 MHz, 1.544 MHz or 2.048 MHz. May be asynchronous with CLK_X or CLK_C .
20 FS_X	Encoder frame sync pulse. Normally occurring at an 8 kHz rate, this pulse is nominally one CLK_X cycle wide. Extending the width of FS_X to two or more cycles of CLK_X signifies a transmit signaling frame.
24 CLK_C	Control clock input used to shift serial control data into D_C . CLK_C must pulse 4 times during a period of time less than or equal to one frame time, although the 4 pulses may excite a frame boundary. CLK_C need not be synchronous with CLK_X or CLK_R . Connecting CLK_C continuously high places the TP3020/TP3021 into the third time slot mode.

Functional Description

POWER-UP

Upon application of power, internal circuitry initializes the CODEC and places it into the power-down mode. No sequencing of 5V or -5V is required. In the power-down mode, all non-essential circuits are deactivated, the TRI-STATE PCM data output D_X is placed in the high impedance state and the receive signaling output of the TP3020, SIG_R , is reset to logical zero. Once in the power-down mode, the method of activating the TP3020/TP3021 depends on the chosen mode of operation, time slot assignment or fixed time slot.

TIME SLOT ASSIGNMENT MODE

The time slot assignment mode of operation is selected by maintaining CLK_C in a normally low state. The state of the CODEC is updated by pulsing CLK_C eight times within a period of 125 μ S or less. The falling edge of each clock pulse shifts the data on the D_C input into the CODEC. The first two control bits determine if the subsequent control bits B3-B8 are to specify the time slot for the encoder (B1=0), the decoder (B2=0) or both (B1 and B2=0) or if the CODEC is to be placed into the power-down mode (B1 and B2=1). The desired action will take place upon the occurrence of the second frame sync pulse following the first pulse of CLK_C . Assigning a time slot to either the encoder or decoder will automatically power-up the entire CODEC circuit. The D_X output and D_R input, however, will be inhibited for one additional frame to allow the analog circuitry time to stabilize. If separate time slots are to be assigned to the encoder and the decoder, the encoder time slot should be assigned first. This is necessary because up to four frames are required to assign both time slots separately, but only three frames are necessary to activate the D_X output. If the encode time slot has not been updated the PCM data will be outputted during the previously assigned time slot which may now be assigned to another CODEC.

FIXED TIME SLOT MODE

There are several ways in which the TP3020/TP3021 may operate in the fixed time slot mode. The first and easiest method is to leave CLK_C disconnected or to connect CLK_C to V_{CC} . In this situation, D_C behaves as a power-down input. When D_C goes low, both encode and decode time slots are set to one on the second subsequent frame sync pulse. Time slot one corresponds to the eight CLK_X or CLK_R cycles starting one cycle from the nominal leading edge of FS_X or FS_R respectively. As in the time slot assignment mode, the D_X output is inhibited for one additional frame after the circuit is powered up. A logical "1" on D_C powers the CODEC down on the second subsequent FS_X pulse.

A second fixed time slot method is to operate CLK_C continuously. Placing a "1" on D_C will then cause the serial control register to fill up with ones. With B1 and B2 equal to "1" the CODEC will power-down. Placing a "0" on D_C will cause the serial control register to fill up with zeroes, assigning time slot one to both the encoder and decoder and powering up the device. One important restriction with this method of operation is that the rising transition of D_C must occur at least 8 cycles of CLK_C prior to FS_X . If this restriction is not fol-

lowed, it is possible that on the frame prior to power-down, the encoder could be assigned to an incorrect time slot (e.g., 1, 3, 7, 15 or 31), resulting in a possible PCM bus conflict.

SERIAL CONTROL PORT

When the TP3020/TP3021 is operated in the time slot assignment mode or the fixed time slot mode with continuous clock, the data on D_C is shifted into the serial control register, bit 1 first. In the time slot assignment mode, depending on B1 and B2, the data in the RCV or XMT time slot registers is updated at the second FS_R or FS_X pulse after the first CLK_C pulse, or the CODEC is powered down. In the continuous clock fixed time slot mode, the CODEC is powered up or down at every second FS_R or FS_X pulse. The control register data is interpreted as follows:

B1	B2	Action				
0	0	Assign time slot to encoder and decoder				
0	1	Assign time slot to encoder				
1	0	Assign time slot to decoder				
1	1	Power-down CODEC				

B3	B4	B5	B6	B7	B8	Time Slot
0	0	0	0	0	0	1
0	0	0	0	0	1	2
0	0	0	0	1	0	3
0	0	0	0	1	1	4
.	.	.	.	.	.	.
.	.	.	.	.	.	.
.	.	.	.	.	.	.
1	1	1	1	1	0	63
1	1	1	1	1	1	64

During the power-down command, bits 3 through 8 are ignored. Note that with 64 possible time slot assignments it is frequently possible to assign a time slot which does not exist. This can be useful to disable an encoder or decoder without powering down the CODEC.

SIGNALING

The TP3020 μ -law CODEC contains circuitry to insert and extract signaling information for the PCM data. The transmit signaling frame is signified by widening the FS_X pulse from one cycle of CLK_X to two or more cycles.

When this occurs, the data present on the SIG_X input at the eighth clock pulse of the encode time slot is inserted into the last bit of the PCM data stream. A receive signaling frame is indicated in a similar fashion by widening the FS_R pulse to two or more cycles of CLK_R .

During a receive signaling frame, the last PCM bit shifted in is latched into a flip-flop and appears at the SIG_R output. This output will remain unchanged until the next signaling frame, until a power-down is executed or until power is removed from the device. Since the least significant bit of the PCM data is lost during a signaling frame, the decoder interprets the bit as a "1/2" (i.e., half way between a "0" and a "1"). This minimizes the noise and distortion due to the signaling.

ENCODING DELAY

The encoding process begins at the start of the encode time slot and is concluded no later than 17 time slots later. In normal applications, this PCM data is not shifted out until the next time slot 125 μ S later, resulting in an encoding delay of 125 μ S. In some applications it is possible to operate the CODEC at a higher frame rate to reduce this delay. With a 2.048 MHz clock, the FS rate could be increased to 15 kHz reducing the delay from 125 μ S to 67 μ S.

DECODING DELAY

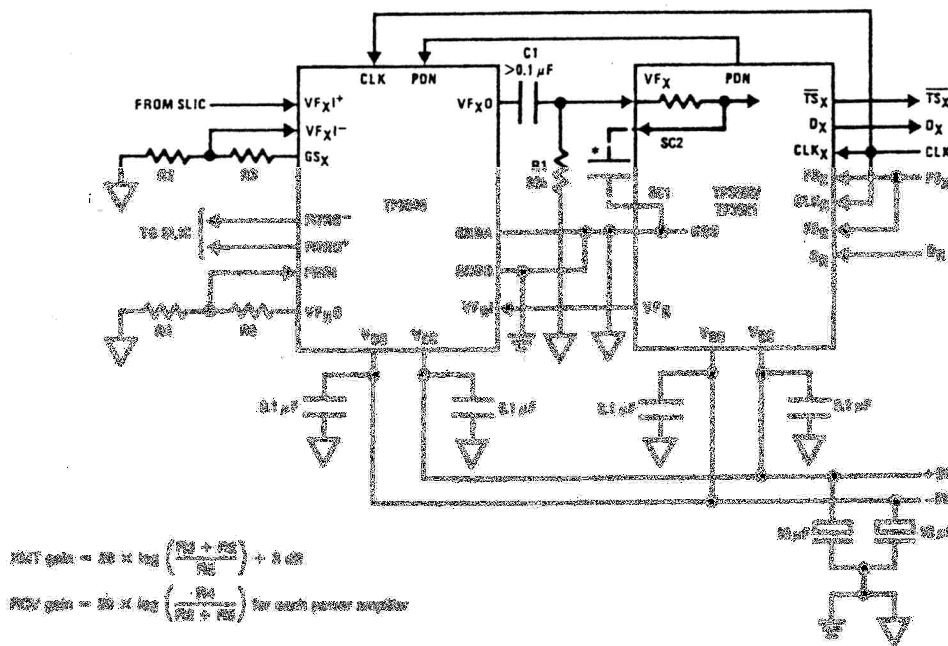
The decoding process begins immediately after the end of the decoder time slot. The output of the decoder sample and hold amplifier is updated 28 CLK_D cycles later.

The decoding delay is therefore approximately 28 clock cycles plus one half of a frame time or 81 μ S for a 1.544 MHz system with an 8 kHz frame rate or 76 μ S for a 2.048 MHz system with an 8 kHz frame rate. Again, for some applications the frame rate could be increased to reduce this delay.

TYPICAL APPLICATION

A typical application of the TP3020/TP3021 used in conjunction with the TP3040 PCM filter is shown. The values of resistor R1 and DC blocking capacitor C1, are non-critical. The capacitor value should exceed 0.1 μ F, R1 should not exceed 160 k Ω , and the product R1 $\times$ C1 should exceed 4 rms. 0.1 μ f power supply bypass capacitors should be used and placed as close to the device as possible.

Typical Application



TL/H4022-2

The power supply decoupling capacitors should be 0.1 μ F in order to take advantage of the excellent noise performance of the TP3020/TP3021/TP3040, care must be taken in board layout to prevent coupling of digital noise into the sensitive analog lines.

*The external average/hold capacitor required for use with pin-compatible NARS CODECs introduces attenuation due to the capacitive divider formed with C1. The SC pin connects VF₂ to this sample/hold capacitor (via a 500 Ω resistor) to ensure gain compatibility. The TP3020/TP3021 itself does not require an external sample/hold capacitor.

TP3040, TP3040-1, TP3040A, TP3040A-1 PCM Monolithic Filter

General Description

The TP3040/TP3040-1/TP3040A/TP3040A-1 filter is a monolithic circuit containing both transmit and receive filters specifically designed for PCM CODEC filtering applications in 8 kHz sampled systems.

The filter is manufactured using microCMOS technology and switched capacitor integrators are used to simulate classical LC ladder filters which exhibit low component sensitivity.

TRANSMIT FILTER STAGE

The transmit filter is a fifth order elliptic low pass filter in series with a fourth order Chebyshev high pass filter. It provides a flat response in the passband and rejection of signals below 200 Hz and above 3.4 kHz.

RECEIVE FILTER STAGE

The receive filter is a fifth order elliptic low pass filter designed to reconstruct the voice signal from the decoded/demultiplexed signal which, as a result of the sampling process, is a stair-step signal having the inherent $\sin x/x$ frequency response. The receive filter approximates the function required to compensate for the degraded frequency response and restore the flat passband response.

Features

- Exceeds all D3/D4 and CCITT specifications
- +5V, -5V power supplies
- Low power consumption:
 - 45 mW (0 dBm0 into 600Ω)
 - 30 mW (power amps disabled)
- Power down mode: 0.5 mW
- 20 dB gain adjust range
- No external anti-aliasing components
- $\sin x/x$ correction in receive filter
- 50/60 Hz rejection in transmit filter
- TTL and CMOS compatible logic
- All inputs protected against static discharge due to handling

Block Diagram

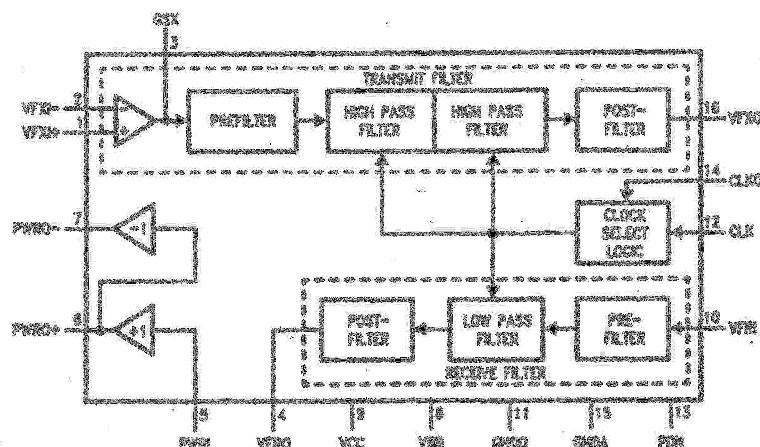
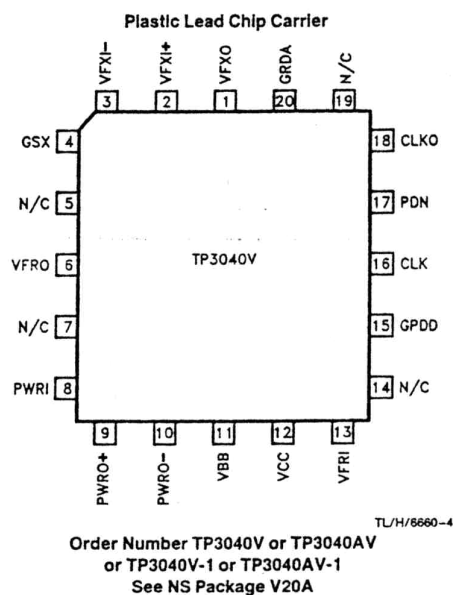
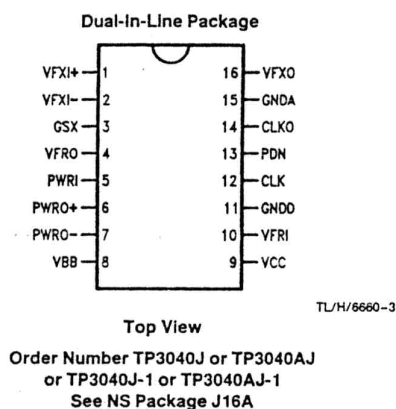


FIGURE 1

TL/H/0000-1

Connection Diagram



Description of Pin Functions

Symbol	Function	Symbol	Function
1 VFXI+	The non-inverting input to the transmit filter stage.	11 GND0	Digital ground input pin. All digital signals are referenced to this pin.
2 VFXI-	The inverting input to the transmit filter stage.	12 CLK	Master input clock. Input frequency can be selected as 2.048 kHz, 1.544 kHz or 1.538 kHz.
3 GSX	The output used for gain adjustments of the transmit filter.	13 PDN	The input pin used to power down the TP3040/TP3040A during idle periods. Logic 1 (VCC) input voltage causes a power down condition. An internal pull-up is provided.
4 VFRO	The low power receive filter output. This pin can directly drive the receive part of an electronic hybrid.	14 CLKO	This input pin selects internal counters in accordance with the CLK input clock frequency.
5 PWRI	The input to the receive filter differential power amplifier.	CLK	Connect CLKO to:
6 PWRO+	The non-inverting output of the receive filter power amplifier. This output can directly interface conventional transformer hybrids.	2048 kHz	VCC
7 PWRO-	The inverting output of the receive filter power amplifier. This output can be used with PWRO+ to differentially drive a transformer hybrid.	1544 kHz	GND0
8 VBB	The negative power supply pin. Recommended input is -5V.	1538 kHz	VBB
9 VCC	The positive power supply pin. The recommended input is 5V.		An internal pull-up is provided.
10 VFRI	The input pin for the receive filter stage.	15 GND1	Analog ground input pin. All analog signals are referenced to this pin. Not internally connected to GND0.
		16 VFXO	The output of the transmit filter stage.

Functional Description

The TP3040/TP3040A monolithic filter contains four main sections; Transmit Filter, Receive Filter, Receive Filter Power Amplifier, and Frequency Divider/Select Logic (*Figure 1*). A brief description of the circuit operation for each section is provided below.

TRANSMIT FILTER

The input stage of the transmit filter is a CMOS operational amplifier which provides an input resistance of greater than 10 M Ω , a voltage gain of greater than 5,000, low power consumption (less than 3 mW), high power supply rejection, and is capable of driving a 10 k Ω load in parallel with up to 25 pF. The inputs and output of the amplifier are accessible for added flexibility. Non-inverting mode, inverting mode, or differential amplifier mode operation can be implemented with external resistors. It can also be connected to provide a gain of up to 20 dB without degrading the overall filter performance.

The input stage is followed by a prefilter which is a two-pole RC active low pass filter designed to attenuate high frequency noise before the input signal enters the switched-capacitor high pass and low pass filters.

A high pass filter is provided to reject 200 Hz or lower noise which may exist in the signal path. The low pass portion of the switched-capacitor filter provides stopband attenuation which exceeds the D3 and D4 specifications as well as the CCITT G712 recommendations.

The output stage of the transmit filter, the postfilter, is also a two-pole RC active low pass filter which attenuates clock frequency noise by at least 40 dB. The output of the transmit filter is capable of driving a ± 3.2 V peak to peak signal into a 10 k Ω load in parallel with up to 25 pF.

RECEIVE FILTER

The input stage of the receive filter is a prefilter which is similar to the transmit prefilter. The prefilter attenuates high frequency noise that may be present on the receive input signal. A switched capacitor low pass filter follows the prefilter to provide the necessary passband flatness, stopband rejection and sin x/x gain correction. A postfilter which is similar to the transmit postfilter follows the low pass stage. It attenuates clock frequency noise and provides a low output impedance capable of directly driving an electronic subscriber-line-interface circuit.

RECEIVE FILTER POWER AMPLIFIERS

Two power amplifiers are also provided to interface to transformer coupled line circuits. These two amplifiers are driven by the output of the receive postfilter through gain setting resistors, R3, R4 (*Figure 2*). The power amplifiers can be deactivated, when not required, by connecting the power amplifier input (pin 5) to the negative power supply V_{SS} . This reduces the total filter power consumption by approximately 10 mW–20 mW depending on output signal amplitude.

POWER DOWN CONTROL

A power down mode is also provided. A logic 1 power down command applied on the PDN pin (pin 13) will reduce the total filter power consumption to less than 1 mW. Connect PDN to GNDD for normal operation.

FREQUENCY DIVIDER AND SELECT LOGIC CIRCUIT

This circuit divides the external clock frequency down to the switching frequency of the low pass and high pass switched capacitor filters. The divider also contains a TTL-CMOS interface circuit which converts the external TTL clock level to the CMOS logic level required for the divider logic. This interface circuit can also be directly driven by CMOS logic. A frequency select circuit is provided to allow the filter to operate with 2.048 MHz, 1.544 MHz or 1.536 MHz clock frequencies. By connecting the frequency select pin CLK0 (pin 14) to V_{CC} , a 2.048 MHz clock input frequency is selected. Digital ground selects 1.544 MHz and V_{SS} selects 1.536 MHz.

Applications Information

GAIN ADJUST

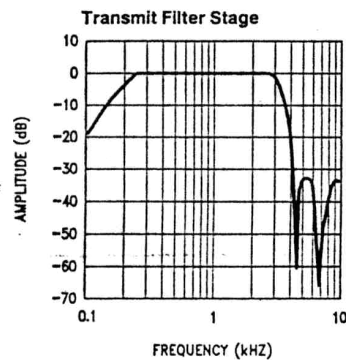
Figure 2 shows the signal path interconnections between the TP3040/TP3040A and the TP3020 signal-channel CODEC. The transmit RC coupling components have been chosen both for minimum passband droop and to present the correct impedance to the CODEC during sampling.

Optimum noise and distortion performance will be obtained from the TP3040/TP3040A filter when operated with system peak overload voltages of ± 2.5 V to ± 3.2 V at V_{F1O} and V_{F2O} . When interfacing to a PCM CODEC with a peak overload voltage outside this range, further gain or attenuation may be required.

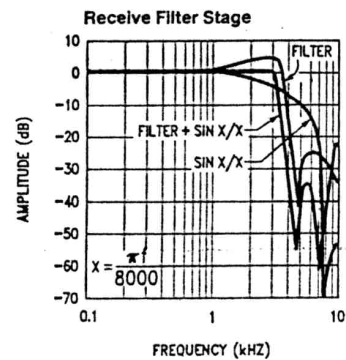
BOARD LAYOUT

Care must be taken in PCB layout to minimize power supply and ground noise. Analog ground (GNDA) of each filter should be connected to digital ground (GNDD) at a single point, which should be bypassed to both power supplies. Further power supply decoupling adjacent to each filter and CODEC is recommended. Ground loops should be avoided, both between GNDA and GNDD and between the GNDA traces of adjacent filters and CODECs.

Typical Performance Characteristics

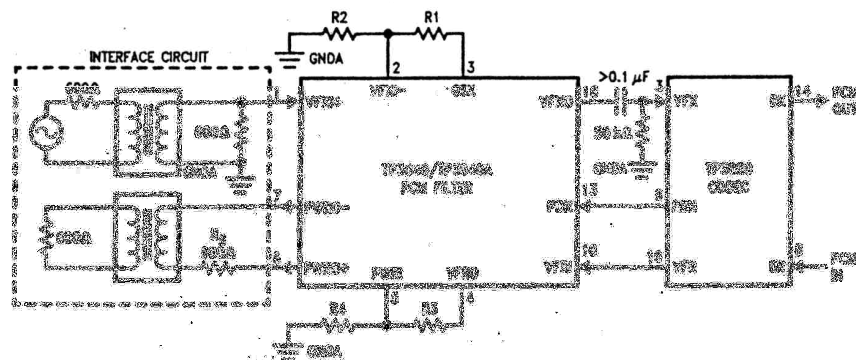


TL/H/6660-5



TL/H/6660-6

Typical Application



TLH/6660-3

Note 1: Transmit voltage gain $\frac{R_1 + R_2}{R_1} \times 6$ (The filter itself introduces a 6 dB gain, $R_1 + R_2 = 10k$)

Note 2: Receive gain $\frac{R_4}{R_3 + R_4}$

($R_3 + R_4 = 10k$)

Note 3: In the configuration shown, the receive filter power amplifier will drive a 600Ω T to R termination to a maximum signal level of 0.5 dBm. An alternative arrangement, using a transformer winding ratio equivalent to 1.414:1 and 600Ω resistors, R₃, will provide a maximum signal level of 0.5 dBm across a 600Ω termination impedance.

FIGURE 2

Appendix A

Saving screen display of the Spectrum Analyzer (bitmap)

Once the display is on the screen (either from an input waveform or loaded from the internal hard drive),

- press [File]
- press <Save>
- press <Screen>
- change destination :
 - press [Tab ⇒]
- choose filename, using soft keys :
 - press [Tab ⇒]
- choose drive :
 - press <Select>
 - A - should be highlighted
 - press <Select>
 - press [Enter]

The file is now saved on A: as a bit map (.gif) file and can be imported into a word processor.

For example, to import into WORD,

- click 'Insert'
- click 'Object'
- click 'Create from file'
- click 'Browse'
- open drive A:, choose file
- click 'OK'
- click 'OK'
- size and move to desired spot