Tucker Master

MODEL 2A/2B JITTER and HIT SYNTHESIZER

PRELIMINARY OPERATING INSTRUCTIONS

BRADLEY TELCOM CORP.
9 Dash Place
Bronx, New York 10463
(212) 549-8500

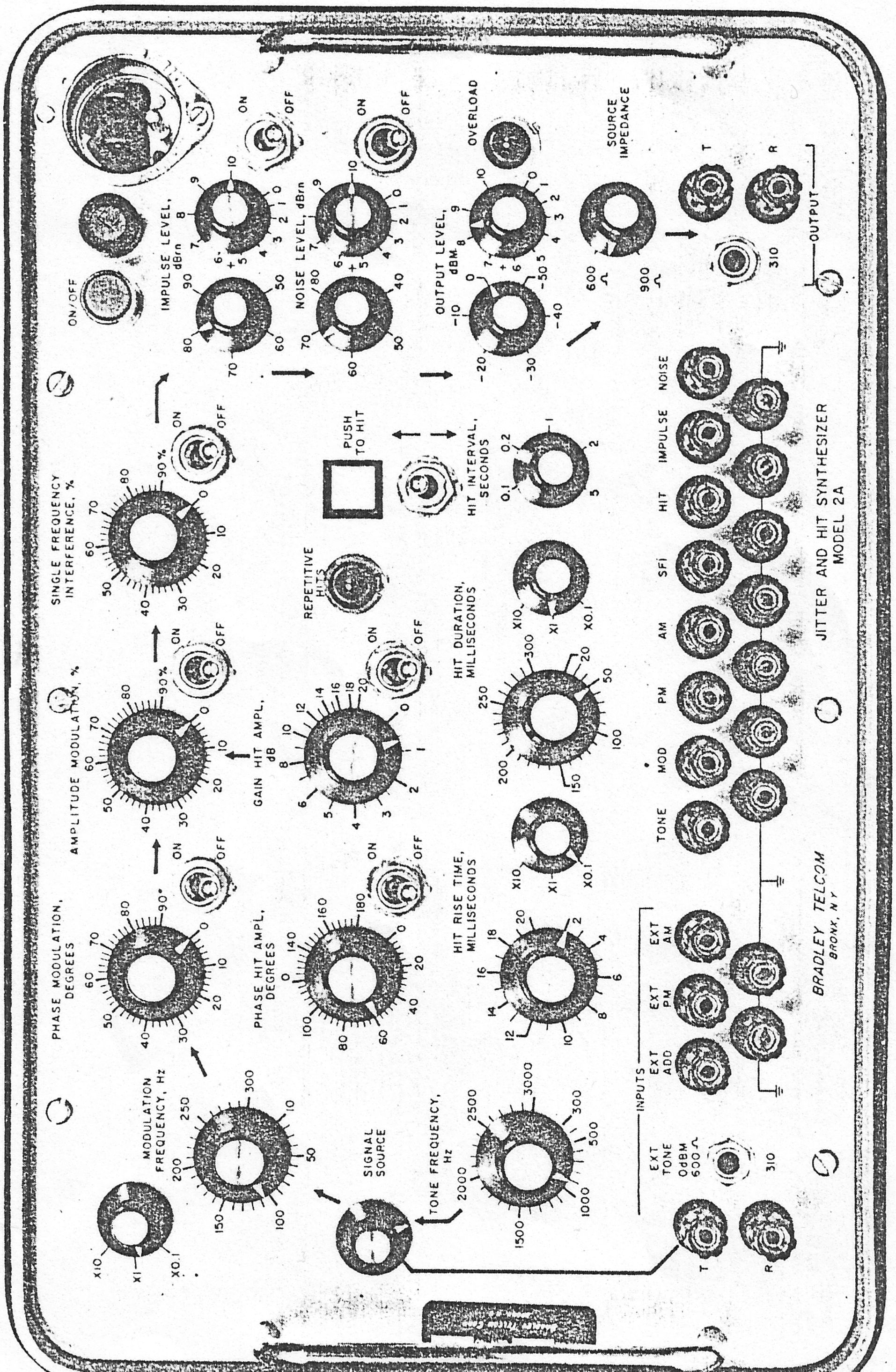
Do Not SHIP!

SYNTHESIZER MODEL 2A
and
FREQUENCY TRANSLATION AND HARMONIC DISTORTION MODULE

PRELIMINARY OPERATING INSTRUCTIONS

TABLE OF CONTENTS

TABLE OF CONTENTS (continued)



9 DASH PLACE BRONX, NEW YORK 10463 (212) 549-8500

LINE IMPAIRMENT GENERATOR with FREQUENCY TRANSLATOR and HARMONIC DISTORTION MODULE

The Model 2A Line Impairment Generator (Jitter and Hit Synthesizer) with the Model 2B Frequency Translator and Harmonic Distortion Module provide *calibrated* and *continuously variable* impairments which may be induced on an internally generated test tone (300 to 3000 Hz) or on an externally generated voice band signal.

STEADY STATE IMPAIRMENTS

Voice band white noise – 0 to −50dBm
Frequency Translation – 1 to 10 Hz, up or down
Harmonic Distortion
- 2nd – 0 to 15%
- 3rd – 0 to ±15%

These adjustments are independent of each other

Phase Jitter – 0 to 90°
Amplitude Jitter – 0 to 90%
Single Frequency Interference – 0 to 90%

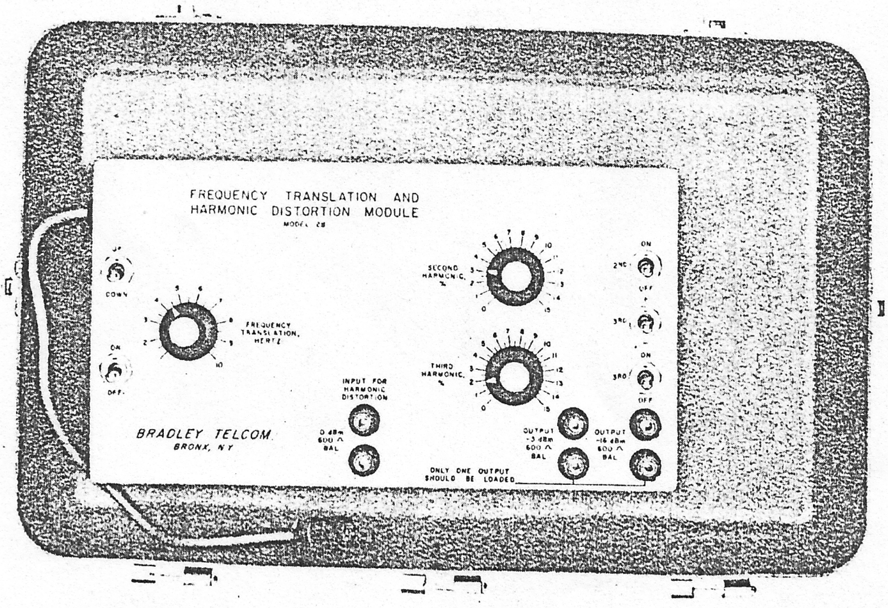
The disturbance frequency may be varied from 1 to 3000Hz (internal oscillator) or may be controlled by external input signal.

HIT IMPAIRMENTS

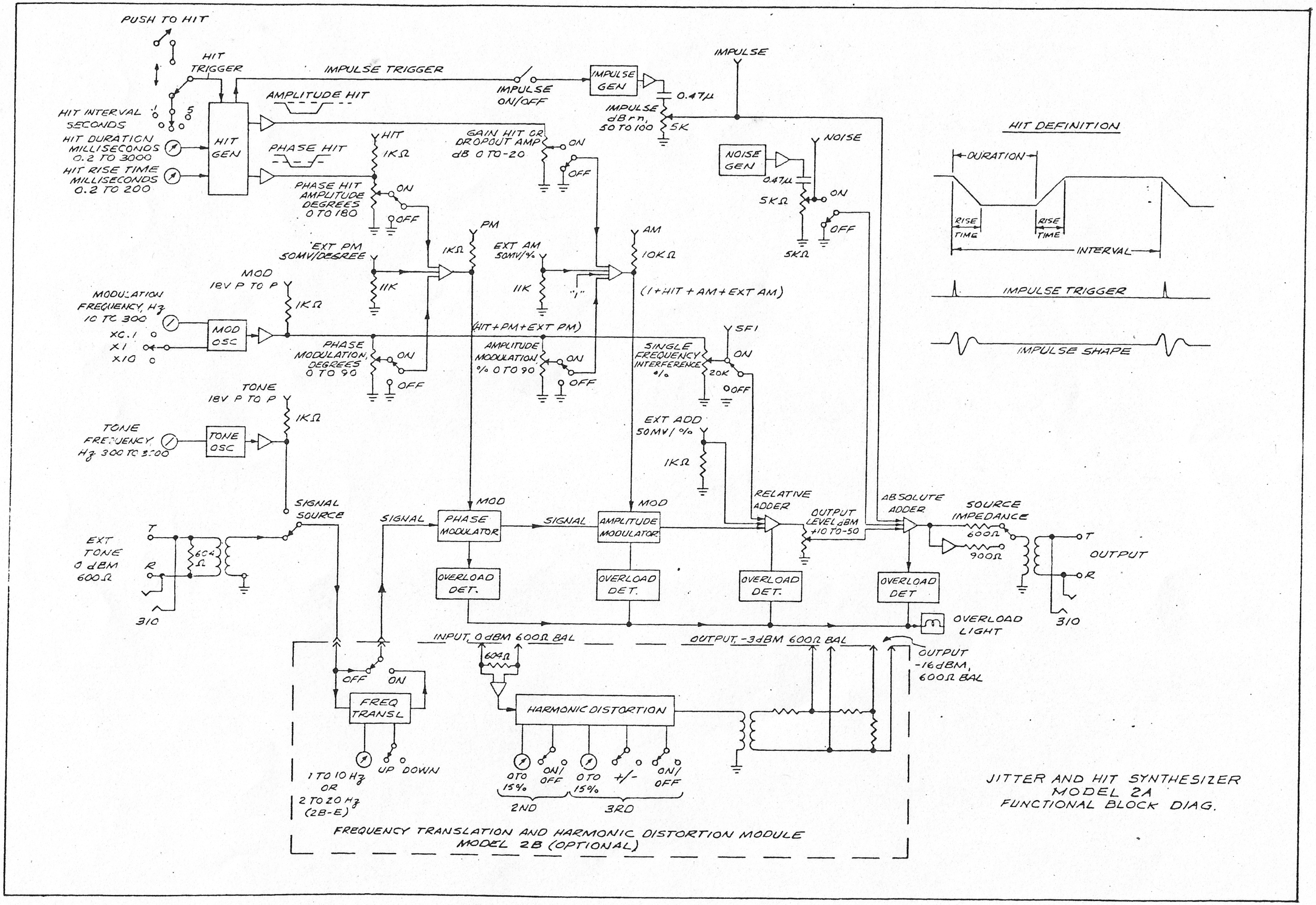
Impulse Hits	+10 to −40dBm
Phase Hits	0 to 180°
Gain Hits	0 to 20dB
Dropouts	0 to 20dB
Frequency Hits	0 to 20Hz

These hits have selectable or manually controlled occurence rate, adjustable rise time – 0.2 to 200 milliseconds and adjustable duration 2 to 3000 milliseconds

The Model 2A Line Impairment Generator includes two variable frequency audio oscillators, a voice band calibrated noise generator, a voice band calibrated phase shifter, a trapezoidal pulse generator, and an impulse generator. All of these are independently useable as laboratory instruments. (See photo on reverse side)



9 DASH PLACE BRONX, NEW YORK 10463 (212) 549-8500



JITTER AND HIT SYNTHESIZER
MODEL 2A
FUNCTIONAL BLOCK DIAG.

SECTION 1

OPERATING INSTRUCTIONS

1. INTRODUCTION

The 2A Synthesizer, without the 2B option, is furnished with a jumper connector in place. With the 2B option, the cable to the 2B is in place. One or the other of these must be plugged into the 7 pin blue receptacle in the upper left corner of the 2A panel, otherwise the signal path is interrupted.

Referring to Figure 1, the EXT TONE signal or the internal oscillator (as selected by the SIGNAL SOURCE switch) passes through a frequency translator (optional), the phase modulator, an amplitude modulator, an SFI (Single Frequency Interference) Adder, a calibrated attenuator, and a Steady State and Impulse Noise Adder. The Tone Oscillator permits the generation of test tones with calibrated impairments for evaluation and/or calibration of jitter, noise, and hit sets, as well as for the training of maintenance personnel. The Modulation Oscillator provides a second variable frequency source which may be applied in calibrated amplitude to the Phase Modulator to generate phase jitter, to the Amplitude Modulator to generate amplitude jitter, and/or to the SFI adder to generate single frequency interference. Access is provided for the application of external inputs of any desired (voice band) spectral content to the Phase and Amplitude Modulators and to the SFI Adder. The Hit Generator provides pulses at a selectable Interval, of adjustable Rise Time, and adjustable Duration. These pulses may be selectively applied to generate Phase Hits, Gain Hits, Coincident Phase and Gain Hits, or Dropouts. The modulation-impaired signal is then coupled, via a calibrated attenuator, to the Steady and Impulse Noise Adder where these latter two impairments may be added. The Impulse Generator is triggered by the leading edge of the hit pulse and generates an approximately uniform spectral energy distribution from 800 to 2200 Hz.

The harmonic distortion section of the 2B module is patchable at both input and output so that it can be placed ahead of or behind the 2A in the signal path.

2. 2A CONTROL FUNCTIONS (Refer to front panel photo of instrument)

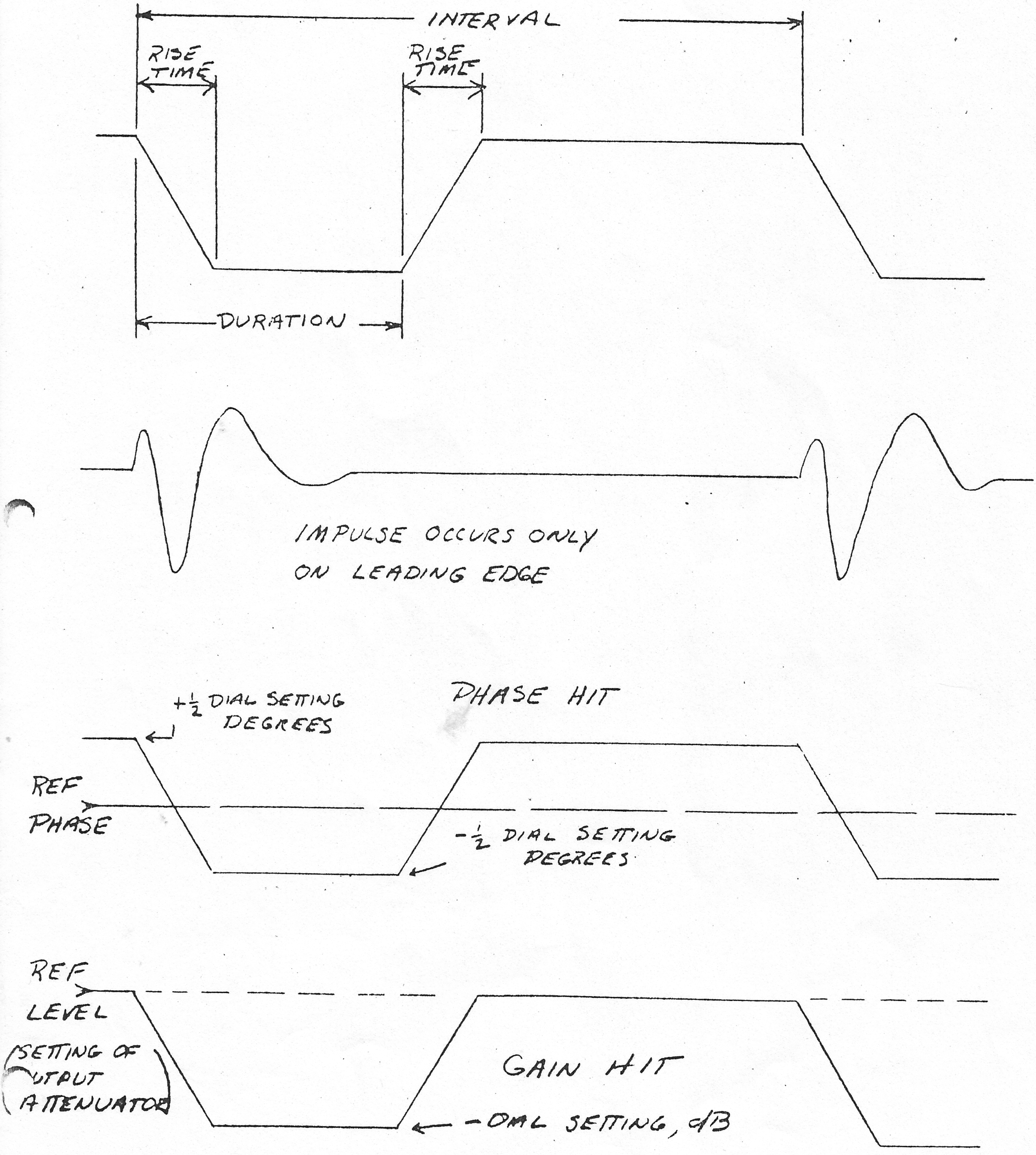
The heavy arrows starting at the lower left side of the panel and ending at the OUTPUT jack (lower right) define the signal flow path. Panel controls and indications perform the functions described below:

a. SIGNAL SOURCE - Selects EXT TONE jack or internal tone oscillator as signal source.

b. Tone Controls

TONE FREQUENCY, HZ - Adjusts internal tone oscillator frequency from 300 to 3000 Hz.

FIGURE 1



INTERVAL
RISE TIME
RISE TIME
DURATION
IMPULSE OCCURS ONLY ON LEADING EDGE
PHASE HIT
+1/2 DIAL SETTING DEGREES
REF PHASE
-1/2 DIAL SETTING DEGREES
REF LEVEL
(SETTING OF UTPUT ATTENUATOR)
GAIN HIT
-DIAL SETTING, dB

OUTPUT LEVEL selector switch and vernier - Adjusts output level between -50 and +10dBm. Add vernier reading to switch position for reading. The output level attenuator is direct reading for an EXT TONE input of 0 level and 600 ohms source impedance. The EXT TONE input must be less than 2.2 volts peak to peak. This is the peak to peak value of a 0dBm (600 ohms) sine wave.

SOURCE IMPEDANCE switch - Selects source impedance of balanced output as either 600 or 900 ohms.

c. Modulation Controls

MODULATION FREQUENCY, Hz - Range switch and vernier, adjusts modulation from 1 to 3000 Hz.

PHASE MODULATION, DEGREES - Vernier adjustment and on/off switch, adjusts amplitude of phase modulation from 0 to 90% peak to peak. Phase modulation is at MODULATION FREQUENCY.

EXT PM jack - Provides external access to the phase modulator. This input will accommodate voice band inputs of any spectral content. Scale factor is 50 millivolts per degree.

AMPLITUDE MODULATION, % - Vernier adjustment and on/off switch, adjusts amplitude of AM from 0 to 90%. AM is at MODULATION FREQUENCY.

EXT AM jack - Provides external access to the amplitude modulator. This input will accommodate voice band inputs of any spectral content. Scale factor is 50 millivolts per percent.

SINGLE FREQUENCY INTERFERENCE % - Vernier adjustment and on/off switch adjusts amplitude of additive tone from 0 to 90% of tone source amplitude. Additive tone is at MODULATION FREQUENCY.

EXT ADD jack - Provides external access to the SFI adder. This input will accommodate voice band inputs of any spectral content. Scale factor is 50 millivolts per percent.

d. Additive Disturbances

IMPULSE LEVEL - Selector switch, vernier, and on/off switch. This set of controls adds a calibrated impulse to the OUTPUT. Note all hits other than impulse occur in pairs, see below, the impulse coincides only with the first hit of each pair. Range is from 50 to 100 dBrn. This is instantaneous power at the peak of the impulse. Add switch and vernier setting for reading.

NOISE LEVEL - Selector switch, vernier, and on/off switch. This set of controls adds a calibrated amount of white noise (3KC band width) to the OUTPUT. Range is from 40 to 90 dBrn. Above 80 dBrn scale accuracy is reduced. Add switch and vernier setting for reading.

e. Hit Disturbances, see Figure 1

Method of hit initiation is selected by an "arrow-labeled" toggle switch between the HIT INTERVAL switch and the PUSH TO HIT Button. In the down " " position of the toggle, the HIT INTERVAL Switch controls, and in the up " " position the PUSH TO HIT Button controls.

HIT INTERVAL, SECONDS - Selects one of the five designated repetitive hit periods.

PUSH TO HIT - momentary push button - this switch causes a gain or phase hit on depression and another hit in the opposite direction upon release If the button is released prior to the expiration of the HIT DURATION, the second hit will not occur until the end of the HIT DURATION interval. Impulse hits occur only upon depression of the button and not upon release.

HIT DURATION, MILLISECONDS - Range switch and vernier, adjusts hit duration from 20 to 3000 milliseconds. Multiply scale switch setting and vernier reading for reading. Note HIT DURATION can be made longer than HIT INTERVAL This will cause seemingly erratic hit intervals. It is not a normal mode of operation.

HIT RISE TIME, MILLISECONDS - Scale switch and vernier, adjusts hit rise time from 0.2 to 200 milliseconds. Multiply scale switch setting and vernier for reading. Note HIT RISE TIME can be set longer than HIT DURATION and HIT INTERVAL. This will result in seemingly erratic and/or reduced hit amplitude. It is not a normal mode of operation. To avoid confusion the following conditions must be observed:

(1) HIT DURATION + HIT RISE TIME shorter than HIT INTERVAL

(2) HIT DURATION longer than HIT RISE TIME

PHASE HIT AMPL, DEGREES - Vernier and on/off switch adjusts the amplitude of the phase hit. PHASE HIT AMPL vernier offset the output phase by half of its set value in the lagging direction. Then when the hit occurs, the phase moves to an equal amount (half its set value) in the lead direction at the start of the HIT INTERVAL. This occurs with a linear ramp whose duration is set by the HIT RISE TIME. At the expiration of the HIT DURATION the phase returns to the original lagging direction, again with the same linear phase ramp.

GAIN HIT AMPL, DEGREES - Vernier and on/off switch adjusts the amplitude of the gain hit. When the hits occur, the level drops by a dB amount as indicated by the GAIN HIT dial setting. The drip occurs with a linear ramp whose duration is set by the HIT RISE TIME. At the expiration of the HIT DURATION, the level returns to its starting value with the same linear ramp. Note that the gain hit does not affect the noise, SFI, or EXT ADD components. These are added to the signal after the gain hit is introduced, see block diagram. The gain hit will affect noise which is introduced at the EXT TONE jack when that input is being processed by the synthesizer, i.e., TONE SOURCE in appropriate position.

f. Indicator Lights

ON/OFF (green) - This is an alternate action on/off switch which controls the application of primary power. Line power is fused at 1 ampere.

REPETITIVE HITS (green) - This indicates that the HIT INTERVAL switch is in control of hit operation.

OVERLOAD (red) - This light indicates that the combination of impairments called for by the control settings and/or the external inputs is causing the instrument to saturate, and that the output is not a valid representation of the input plus specified impairments.

g. Control Outputs

TONE (jack) - Nominally 20 volts peak to peak sinusoid at TONE FREQUENCY, 1000 ohm source impedance. This output is useful for oscilloscope syncing and as a general purpose oscillator.

MOD (jack) - Nominally 20 volts peak to peak sinusoid at MODULATION FREQUENCY, 1000 ohm source impedance. Useful as a general purpose oscillator, and for scope syncing to display modulation.

PM (jack) - Summation of internal phase modulation, phase hit, and EXT PM. Scale factor nominally 50 mv per degree, source impedance 1000 ohms. This signal is the phase modulation forcing function which exists at any given time.

AM (jack) - Summation of internal amplitude modulation, gain hit, and EXT AM. Scale factor nominally 50 mv %, source impedance 10K. This signal is the amplitude modulation forcing function which exists at any given time.

SFI (jack) - Summation of SFI and EXT ADD. Scale factor nominally 50 per mv per %, source impedance 0 to 5000 ohms. This signal exists at the jack regardless of the state of the SINGLE FREQUENCY INTERFERENCE ON/OFF switch.

HIT (jack) - This is the hit profile for oscilloscope display, the voltage swing is from +4.5 to -4.5V, source impedance is 1000 ohms. This voltage may be used to synchronize an oscilloscope for stationary display of hit profile.

IMPULSE (jack) - Output of impulse generator, 1000 to 5000 ohms source impedance. This voltage may be displayed on an oscilloscope. It is only present when the IMPULSE on/off switch is ON.

NOISE (jack) - Output of the noise generator independent of the state of the NOISE LEVEL ON/OFF switch. May be used as a general purpose noise source. 1000 to 5000 ohms source impedance.

3. 2B CONTROL FUNCTIONS. The 2B Frequency Translation and Harmonic Distortion Module mounts inside the lid of the Model 2A Jitter and Hit Synthesizer. It receives power (+12, -12, and ground) and tone from the 2A via the connector on the upper left panel of the 2A. It also returns the frequency translated tone via this connector. The input and output for harmonic distortion are separately patchable on the panel of the 2B to permit insertion of harmonic distortion ahead of or behind the impairments inserted by the basic 2A unit.

Panel elements of the 2B are as follows:

FREQUENCY TRANSLATION, HERTZ - Calibrated continuous adjustment from 1 to 10 Hertz of translating frequency.

UP/DOWN - Two position toggle, selects direction of frequency translator.

ON/OFF - Two position toggle, inserts frequency translation in signal path or bypasses frequency translator.

The frequency translation mechanization in the 2B introduces 2 to 4% of amplitude modulation and 3 or 4 degrees of phase jitter at the translation frequency. The UP/DOWN switch may be used to insert frequency hits, e.g., set to 2Hz, switch from UP to DOWN to put in a -4Hz frequency hit.

INPUT FOR HARMONIC DISTORTION - OdBm, 600 ohms, Balanced-Sinusoid should be at OdBm level here for accurate calibration of distortion dials.

SECOND HARMONIC, % vernier and ON/OFF toggle - control operation and level of second harmonic distortion, 0 to 15%.

THIRD HARMONIC, % vernier and ON/OFF toggle - controls operation and level of third harmonic distortion, 0 to 15%.

THIRD HARMONIC +/- controls sense of third harmonic.

OUTPUT, -3dBm, 600 ohms balanced -

and OUTPUT, - 16dBm, 600 ohms balanced -

Only one of these outputs should be loaded at any one time

Harmonic Distortion

The harmonic distortion function is patched from the panel of the 2B so that it can be inserted either ahead of or behind the other 2A impairments. The performance equation for this unit is:

$$\text{Output} = \Big\{ \Sigma A_i \cos w_i t$$

$$+ \text{Dial Set 2nd} \times \frac{2}{A_o^2} \left(\Sigma A_i^2 \cos 2 w_i t - \Sigma A_i^2 \right)$$

$$+ \text{Dial Set 3rd} \times \frac{2}{A_o^3} \left[(\Sigma A_i \cos w_i t) \left(\Sigma A_i^2 \cos 2 w_i t - \Sigma A_i^2 \right) \right.$$

$$\left. - A_i^3 \cos w_i t \right] \Big\}$$

-3dB or -16dB

Where Ao = coefficient of OdBm sine wave, i.e., Ao cos wt = "0" level
Ai cos wt are components of complex wave.

The calibration of the dials is dependent on the level and frequency distribution of the input signal, and is mathematically predictable from the above.

For a sinusoidal input at 0 level, i.e.,

Output voltage = (Ao cos wt + Dial Set 2nd x cos 2 wt + Dial Set 3rd x cos 3 wt)

-3dB or -16dB depending upon the output jack selected.

In order for the dials to be direct reading for a sine wave, the sine wave input level must be OdBm.

SECTION 2 - ALIGNMENT PROCEDURE
SYNTHESIZER MODEL 2A

1. INTRODUCTION

We have experienced perhaps a hundred machine years of synthesizer operation without, as far as we know, any requirement for field calibration. The point is that measurements indicating the need for recalibration should be double and triple checked for control setting and/or measurement errors before undertaking readjustment of the Synthesizer. Nevertheless, a systematic procedure is given below.

CAUTION There are individual potentiometers which independently affect the scale factor of each dial function. These may be adjusted and will not interact with any other adjustment. There are also adjustments which affect the quality of operation and which interact. A list is given below of the adjustments which are independent. To adjust any single impairment scaling, the appropriate potentiometer should be located and adjusted. Unless one desires to go through the entire systematic procedure also given further below, adjustment of any other potentiometers should be avoided.

To align the synthesizer it is necessary to have an extender card, a dual trace oscilloscope, a voice band level meter, a digital volt meter and a frequency counter capable of resolving 1/10 Hz up to 3000 Hz. The card locations are identified in Figure 1 of Section 2 in the adjustment potentiometer locations on individual cards are indicated in Figure 2. In many cases it is necessary to use the extender card to reach adjustments.

Note the signal path is interrupted at the 7 pin connector at the upper left corner of the front panel. Either the 2B frequency translation and harmonic distortion module connector should be plugged in here or a jumper connector which connects pin D to E and pin A to B. Caution, pins F,C, and H are +12, -12 and ground. Do not short these to each other or to the other pins.

2. INDIVIDUAL SCALING ADJUSTMENT POTENTIOMETERS

Gain from EXT TONE to Output R82 of Amplitude Modulator

Place SIGNAL SOURCE switch in EXT TONE position, OUTPUT LEVEL switch to -10, and OUTPUT LEVEL vernier to +10. Patch 1000 Hz sine wave at "0" level or lower to EXT TONE, and observe OUTPUT level to be the same as the input level.

Output Level of Internal Tone Oscillator   R4 of Phase Splitter

Place SIGNAL SOURCE switch in TONE FREQUENCY position, set TONE FREQUENCY to 1000 Hz, and OUTPUT LEVEL attenuator to +10, i.e., switch to "0" and vernier to +10. Observe that output level is +10.

SFI Scaling   R60 of Output Amplifier

After changing either or both of above check SFI scaling by observing some convenient setting such as 50%, i.e., sync scope on TONE jack, set SFI to 50%, switch SFI on and off, and observe 50% SFI. R60 adjusts this.

Phase Modulation Scaling   R69 of Phase Modulator

Set in 90° PM, sync scope on TONE. Set scope on a high gain setting (so that zero crossings are distinct) check zero of scope carefully. Adjust modulation frequency for relatively high beat between tone and modulation frequency so that pattern is easily observed. Adjust R69 for equal open and sweep traced interval.

Phase Hit Scaling   R68 Phase Modulator

Set in 90° phase hits, 10 per second, 50 milliseconds duration, 0.2 millisecond rise time. Sync DC coupled scope on TONE and observe 90° hits, using the same techniques as for PM above. R68 adjusts this.

Amplitude Modulation Scaling   R124 Amplitude Modulator

Set in some convenient AM level such as 50%, sync DC coupled scope on TONE and observe 50% AM while switching AM on and off. R124 adjusts this.

Gain Hit Zero   R100 Amplitude Modulator

Switch GAIN HIT or DROPOUT with no repetitive hits, toggle in up position. Move GAIN HIT pot from 0 to -20 and observe no change in output level. R100 adjusts this.

Gain Hit Scaling   R81 Amplitude Modulator

Set OUTPUT LEVEL to some convenient value say "0" dBm and GAIN HIT dial to -20. Depress PUSH TO HIT button and observe 20dB change in level. R81 adjusts this.

Impulse Scaling   R5 Output Amplifier

With no input to EXT TONE place SIGNAL SOURCE switch in EXT TONE position. Turn IMPULSE on and set for repetitive hits at 0.2 second interval, sync scope on impulse and observe amplitude of highest (second) peak. With IMPULSE LEVEL set for 100dBRn this should be 2.45 volts. R5 adjusts this.

Noise Scaling R34 Output Amplifier

Turn off impulse above and turn on NOISE. Set NOISE LEVEL to 90 dBRn and, using whatever weighting is desired, measure noise level. R34 adjusts this.
Note internal noise in bandwidth limited (3.5KHz flat) and the unit is shipped with this adjusted for correct reading, using a true rms level meter.

3. COMPLETE ADJUSTMENT SEQUENCE

Many of the following adjustments interact. The following breaks the adjustments down to sequences of independent groups, and indicates the order in which they must be done.

PROCEDURE 1
POWER SUPPLY

1. Adjust the potentiometer for 12 volt (+ and -) output. -12 exists on pins 1 through 7 of the power supply card and connector, plus 12 exists on pins 18 through 24, and ground exists on pins 8 through 13.
2. Observe that the power supply is balanced to within 10 millivolts. This balance is controlled by the equality of two 10K metal film resistors which are located, together with a ceramic capacitor, between the two LM301A metal can op amps in the center of the card toward the connector end. If the power supply balance is out, establish equality by shunting one or the other of the resistors to determine the change that must be made. Replace the appropriate resistors.

PROCEDURE 2
OSCILLATOR ALIGNMENT

This procedure is independent of anything except the power supply adjustment, Procedure 1.

TONE
Note: The Tone Frequency is available at the TONE output jack on the panel, and this is a convenient place to plug in a frequency counter. The level at this point is 18 volts peak to peak.

1. Set R31 for exactly 50% duty cycle of the tone gate, which is available on pin 16 of the oscillator card. This can also be done with a spectrum analyzer by adjusting for zero second harmonic at the TONE jack.
2. Phase the TONE FREQ dial for 300 Hz oscillator frequency at 300 Hz dial reading.
3. Rotate the dial to 3000 Hz and adjust R32 for 3000 Hz Oscillator frequency.
4. Check the frequency at 1500 Hz dial setting. It should be between 1450 and 1550 Hz.

MODULATION

The modulation frequency is available at the MOD jack in the front panel. This is a convenient place to plug in a frequency counter. The level at this point is 18 volts peak to peak.

1. Set R39 for exactly 50% duty cycle of the modulation gate which is available on pin 15 of the card.
2. With the modulation range switch in the x10 position, phase the MODULATION FREQ dial to 100 Hz modulation frequency at 10 Hz dial reading.
3. Rotate the dial to 300 Hz and adjust R35 for 3000 Hz frequency.
4. Set the range switch to x1 position and adjust R34 for 300 Hz modulation frequency.
5. Set the range switch to x0.1 position and adjust R33 for 30 Hz modulation frequency.
6. Check the modulation frequencies on all three ranges at approximate dial midpoint. The frequency should be within half a division accuracy.

PROCEDURE 3
HIT CONTROLS

This procedure is independent of all other procedures except for power supply adjustment, Procedure 1. It involves setting the potentiometers on the Control Unit card as well as one potentiometer on the Amplitude Modulator.

1. Set the hit selector toggle switch "down", the HIT INTERVAL switch to 0.1 second, observe pin 15 of the Control Unit connector with an oscilloscope self-sync, and adjust R29 of the Control Unit for 0.1 second pulse spacing.
2. Check 0.2, 1, 2, and 5 positions of the HIT INTERVAL switch, and trim R29 for best compromise on all five pulse spacings.
3. Phase the HIT DURATION dial against CCW stop at a dial reading of 20. Place the HIT DURATION range switch in the x0.1 position, and then adjust R35 of the Amplitude Modulator card for 2 milliseconds' duration of the pulse observed at pin 22 of the Control Unit. This pulse occurs at an interval which is controlled by the HIT INTERVAL switch and its duration is controlled by the setting of R35 of the Amplitude Modulator card.
4. Rotate the HIT DURATION dial to 300 and adjust R23 of the Control Unit for 30 millisecond duration. Recheck the dial at a reading of 20 milliseconds, and readjust R35 if necessary.
5. Set the HIT DURATION range switch to x1, and the dial to 300. Adjust R24 of the Control Unit to 300 milliseconds duration. This will require that the HIT INTERVAL be in the one-second, two-second, or five-second position, otherwise hit triggers will occur before the duration pulse is finished (and will be ignored).

6. Set the HIT DURATION range switch to x10 and adjust R25 for three seconds' duration. This will require that the HIT INTERVAL switch be in the five-second position.
7. Check mid-range for all three durations, and adjust and retouch the adjustments of all four potentiometers for best fit.
8. Observe pin 10 of the Control Unit connector while syncing the oscilloscope from pin 15. Adjust R1 for equal rise time and fall time of the hit.
9. Set the RISE TIME dial at its lower stop, 2, the RISE TIME range switch at x0.1. Adjust R11 of the Control unit for 0.2 milliseconds rise time.
10. Set the RISE TIME dial to 20 and adjust R8 of the Control Unit for 2 milliseconds rise time.
11. Check with the HIT RISE TIME range switch in the x1 and x10 positions for 20 and 200 milliseconds at the high end, and 2 and 20 milliseconds at the low end of the dial. Reset R11 and R8 for best fit all ranges. (middle of the range is most important.)

PROCEDURE 4
ATTENUATOR CARD

This procedure is independent of all other procedures. The Attenuator Card contains span and zero adjustments for the impulse, noise, and output attenuators.

1. With the impulse switch ON, and the IMPULSE level switch in the 80 DBrn position, observe with an oscilloscope triggered by the hit output jack, the impulse output jack. It is desirable that the HIT RISE TIME be fast to simplify scope triggering. Adjust R47 so that the second peak of the impulse wave shape adjusts over a range of 3.16 to 1, as the impulse vernier is varied from 10 to 0.
2. Adjust R53 so that the peak impulse steps through a 3.16 to 1 range as the impulse attenuator switch is stepped between 80 and 70.
3. With the signal source switch in the EXT TONE position, but no EXT TONE applied, i.e., tone off, observe the output jack with a level meter. Place the synthesizer source impedance switch in the 600 ohm position. Set the noise level attenuator switch to the 70 dBrn position and adjust R27 so that the level meter varies over a 10 dB range as the noise level vernier is varied from a setting of 10 to a setting of 0.
4. Switch the noise level attenuator from between 70 and 80 and set R33 so that this produces a 10 dB change in output level.
5. Turn the noise off, turn the tone back on, and with the same level meter, set the output level attenuator to -10 and adjust R7 so that rotation of the output level vernier from 10 to 0 produces a 10 dB change.
6. Adjust R13 so that switching an output level switch between -10 and -20 produces a 10 dB change in output level.

PROCEDURE 5
OUTPUT AMPLIFIER

This procedure is dependent upon first doing Procedures 1 and 4.

1. With a level meter on the output, the tone OFF, and the noise ON, adjust R34 so that the level meter reads the noise dial setting when the noise dial is set for 70 dBrn; that is with the noise level attenuator switch in the 70 position and the vernier set with a reading of 10.
   NOTE: That if an average reading calibrated rms level meter is used, this will read 1dB high for a white noise signal. Therefore the setting should be 1dB low for this kind of level meter.
2. Impulse Shape. Turn the impulse on and observe the impulse with the tone OFF. Adjust R3 to the midpoint of that region over which the wave shape is smooth and the second peak is larger than the first.
3. Impulse Level. With the impulse attenuator switch in the 80 position and the vernier in the 10 position, adjust R34 so that the second peak has a zero to peak amplitude of 0.78 volts.
4. SFI scaling see previous procedure.

PROCEDURE 6
SIGNAL PATH

This sequence involves the phase splitter, the phase modulator card, and the amplitude modulator card. It has been our experience to date that the phase splitter virtually never requires adjustment, but the entire procedure as given herein.

PHASE SPLITTER

1. Patch a zero level sine wave near 1000 Hz into EXT TONE, switch TONE SOURCE back and forth while observing pin 6 of the Phase Splitter Connector, and adjust R4 for equal amplitude. This matches the EXT TONE calibration to the internally generated tone calibration.
2. Observe, with a dual trace oscilloscope, the outputs at pins 8 and 9. Using the internal oscillator, vary the frequency from 300 to 3000 Hz. Adjust R1 for the most constant amplitude over this range.
3. R2 and R3 are independent level adjustments on the cosine and sine outputs (pin 8 and 9 respectively). Adjust these for equality of the outputs.

PHASE MODULATOR

1. Observe pin 9 with an oscilloscope, set in 90° phase modulation, and adjust R8 so that alternate cusps are equal.
2. Adjust R7 so that the bottom of the cusps reach 50% of the peak level.

3. With the tone off, set PHASE MODULATOR to maximum, OUTPUT LEVEL to +10 dBM, set the PHASE HIT dial to 0°, and adjust R5 for minimum output signal level. (at pin 16 of the Phase Modulator card, also available at pin 5 of the AMP. MOD.)
4. Adjust R4 for zero average level.
5. Set the PHASE HIT dial to 90°, adjust R2 for minimum output at the output jack, and adjust R1 for zero average level at pin 17. (pin 6 of AMP MOD.)
6. With the tone ON, set the PHASE HIT dial to 180° and PHASE MODULATION dial to zero.
7. Adjust R6 for minimum output at pin 16.
8. Set the PHASE HIT dial to 180° and adjust R3 for minimum output on pin 17.
9. Phase Modulation Scaling - see previous procedure.
10. Phase Hit Scaling - see previous procedure.

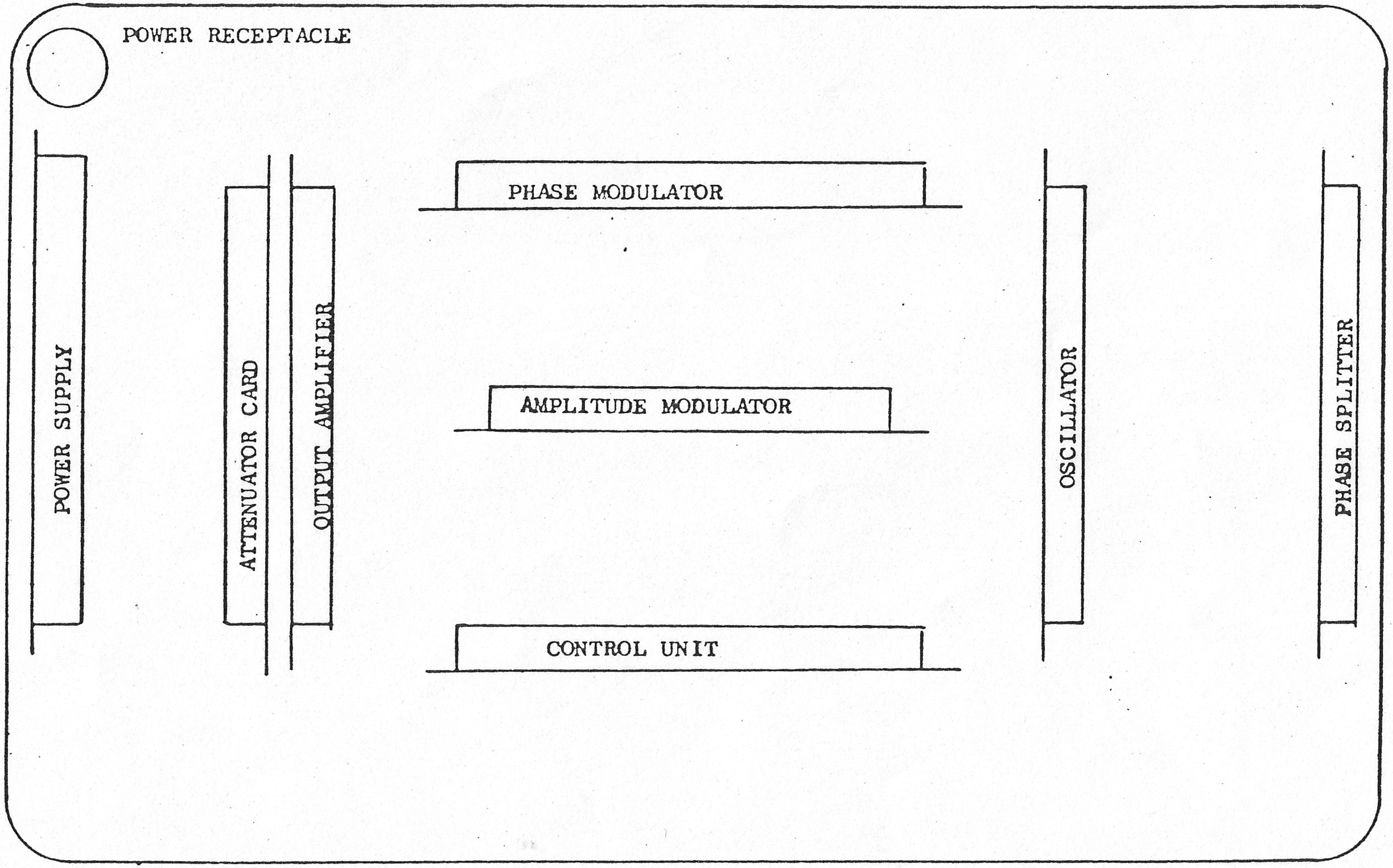
AMPLITUDE MODULATOR

1. With the tone ON and 90° of phase modulation, observe the output (available at pin 10) and adjust R83 so that the jitter modulation pattern is straight across.
2. With the tone ON, jitter OFF, gain hit ON, adjust R100 for no variation in output level as the GAIN HIT dial is varied from zero to 20 dB.
3. With the tone OFF, and the AMPLITUDE MODULATION set for maximum, adjust R123 and R121 for minimum output and zero average dc level at pin 8 of the Amplitude Modulator card. These adjustments will interact.
4. With the tone ON, and AMPLITUDE MODULATION on maximum value, adjust P122 for symmetry of the amplitude modulation.
5. With the tone ON and all impairments off, adjust P81 and P82 for zero dB output level with zero dB output level dial setting.

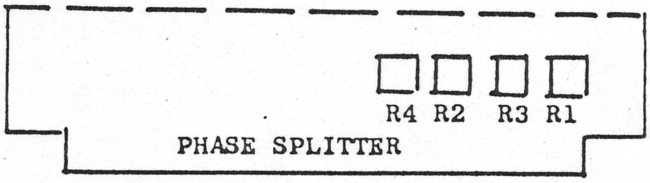
4. 2B FREQUENCY TRANSLATION AND HARMONIC DISTORTION MODULE, (see figure 3 of this section)

1. With a dual trace oscilloscope connected in the x-y mode, observe test positions 1 and 2. Display should be a circle which is traversed at the frequency shift rate. Neither axis of the circle should change dimensions as the FREQUENCY TRANSLATION, HERTZ knob is rotated over the extremes of its range. Pots R3, R7, R17, and R24, all of which interact, are adjusted for this condition.
2. Sync oscilloscope on TONE and observe test point 3. SIGNAL SOURCE switch in EXT TONE position with no input to the EXT TONE jack. Adjust R69 for minimum amplitude at the translation frequency and R70 for zero dc average level. SIGNAL SOURCE to TONE FREQUENCY position and adjust R68 for symmetric positive and negative excursion.
3. Connect to test point 4 and repeat above for pots R56, R57 and R55.

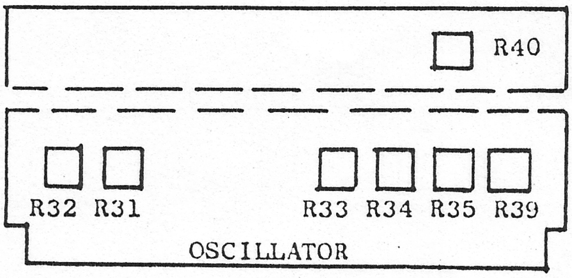
4. Using dual trace scope in x-y mode observe test points 3 and 4. Touch up R55, R56, R68 and R69 for minimum variation at the four corners of the "diamond" pattern on the display.
5. Observe OUTPUT jack with frequency translator on and adjust R33 for minimum amplitude variation.
6. Observe 90° phase jitter on frequency shifted signal (at OUTPUT jack) and adjust R50 for flat top, R43, R57 and R70 for minimum tilt.
7. Adjust R124 so that OUTPUT jack level is the same with or without frequency translation.
8. Repeat 4, 5, 6 and 7 as required to minimize the AM and PM in the output and for flat topped phase jitter. When doing this for the first time this is a lengthy procedure.
9. Using the TONE FREQUENCY Source and 0dm output level, observe test point 5 with scope synced on TONE. Adjust R86/R87 for pure 2nd harmonic and R88 for zero dc average level.
10. Adjust R98 to scale 2nd harmonic in output.
11. Observe test point 6 as above and adjust R99, R100, and R119 for pure 3rd harmonic and R101 for zero dc average level.
12. Adjust R111 to scale 3rd harmonic in output.



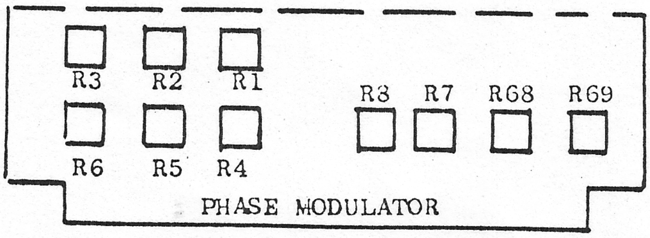
Section 2, Figure 1. Rear view diagram with cover removed.
Block is indicated on component side.



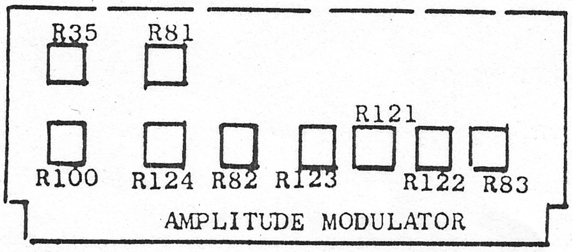
R1 Balance
R2 Cos wt Level
R3 Sin wt Level
R4 Relative level, Modem to internal oscillator



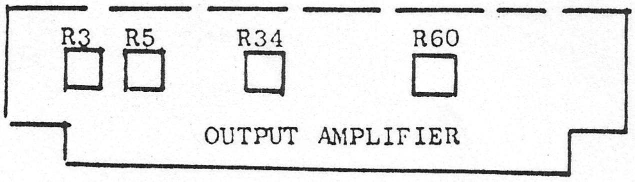
R31 Tone duty cycle
R32 Tone Frequency
R39 Modulation duty cycle
R33 Modulation frequency, x0.1 range
R34 Modulation frequency, x 1 range
R35 Modulation frequency, x 10 range
R40 Modulation oscillator, dc balance



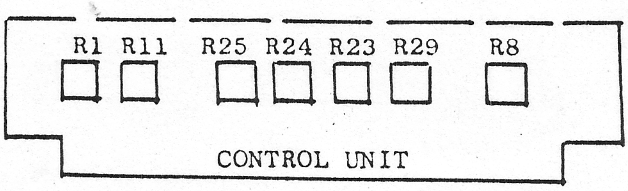
R1 Cos wt sin mt dc zero
R2 Sin mt zero
R3 Cos wt zero
R4 Sin wt cos mt dc zero
R5 Cos mt zero
R6 Sin wt zero
R7 Cos wt ±balance
R8 Cos wt peak value
R68 Phase hit scale
R69 Phase modulation scale



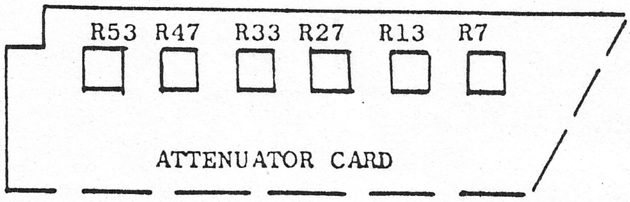
R35 Minimum duration adj
R100 Gain Hit balance
R81 Gain hit scale
R124 AM scale
R82 Output level
R121 Modulation zero
R122 Tone zero
R123 AM output balance
R83 PM sin/cos balanc



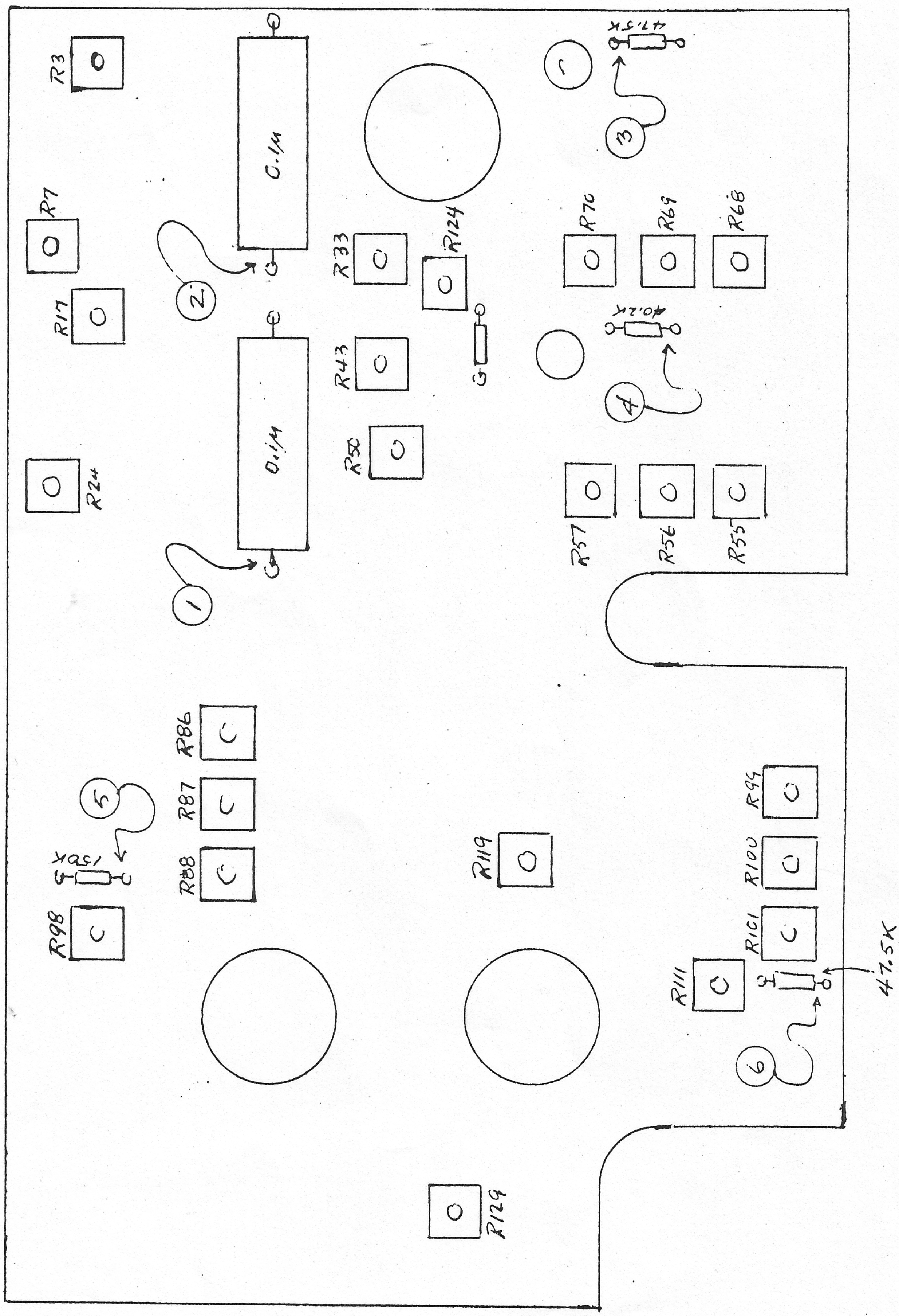
R3 Impulse shape
R5 Impulse amplitude
R34 Noise level
R60 SFI scale



R1 Rise time/Fall time symmetry
R11 Min rise time
R25 x 10 Duration adj
R24 x 1 Duration adj
R23 x 0.1 Duration adj
R29 Interval adj
R8 Rise time adj



R7 Tone vernier span
R13 Tone switch span
R27 Noise vernier span
R33 Noise switch span
R47 Impulse vernier span
R53 Impulse switch span


R3
0.1μ
47.5K
3
R7
2
R17
R33
R124
R70
R69
R68
40.2K
R43
0.1μ
4
R50
R24
R57
R56
R55
1
R86
R87
R88
5
150K
R98
R119
R99
R100
R101
R111
47.5K
6
R129

SYNTHESIZER MODEL 2A

SECTION 3 - CLASSIFICATION OF LINE DISTURBANCES

1. INTRODUCTION

Communication channel performance in data transmission is dependent upon some parameters which are defined, tariffed, and supported by the common carriers; for example, "gain vs frequency" and "envelope delay vs frequency." Performance is also dependent on other parameters which are either not defined or are inadequately defined with respect to the needs of "State of the Art" data sets. Following is a discussion of some of these undefined disturbances. To simplify the discussion, these parameters are related to single tones or simple combinations of such tones rather than to a complex signal.

$$(1)\ \text{Received test tone} = [1 + g(t)] \cos [wt + p(t)] + n(t) + h_k \cos kwt$$

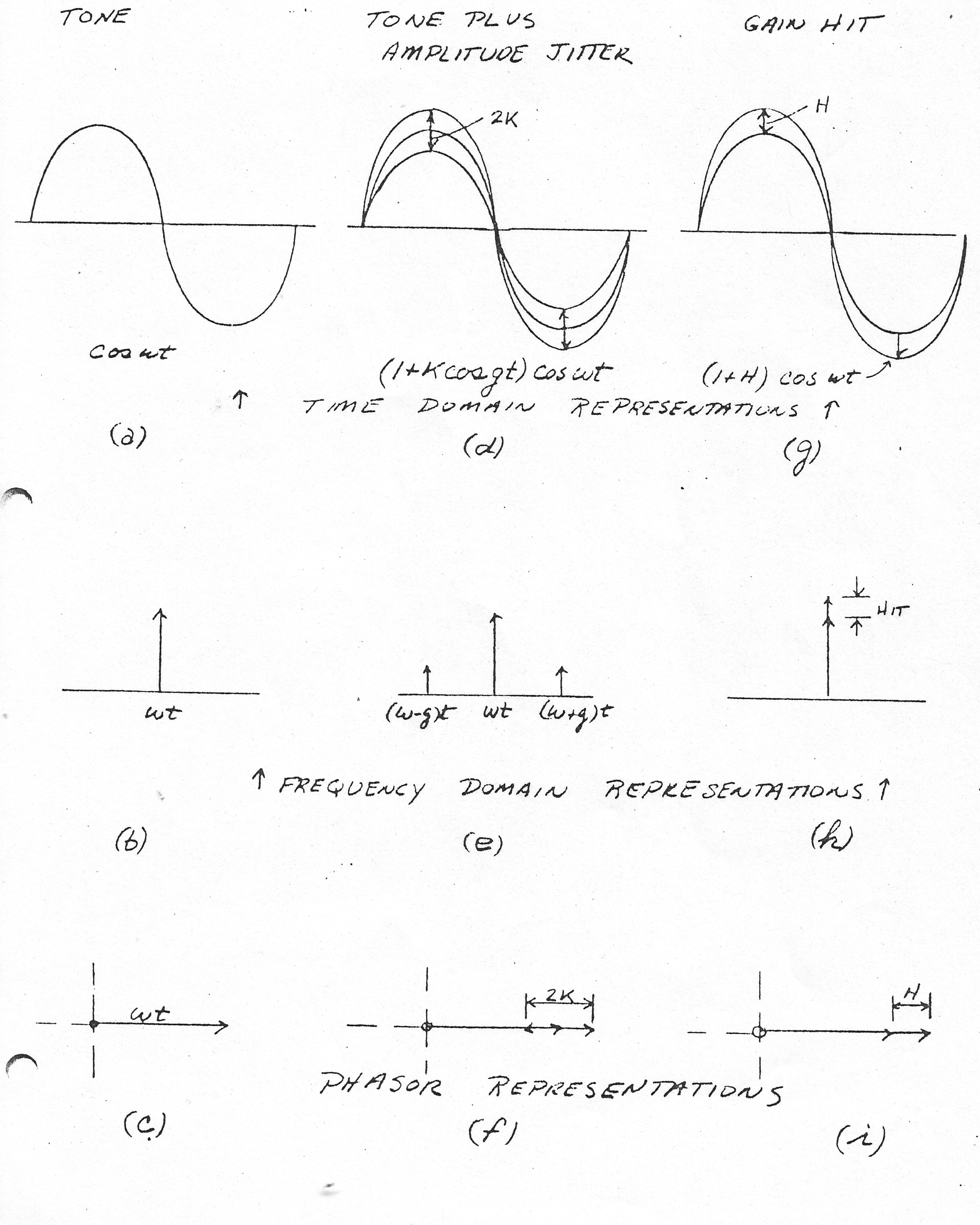
A single test tone, cos wt, as received via the communication channel, has disturbances which are synchronous with the test tone and disturbances which are not. This is reflected by equation 1 in which g(t) and p(t) are respectively disturbances on the amplitude and phase of the received tone, and $h_k$cos kwt is the harmonic distortion, where k assumes integral values of 2 and over. n(t) are disturbances which are not correlated with, or synchronous with, the received tone. Note that reference in this discussion is the received tone cos wt. This takes no account of the channel gain, phase shift, or frequency shift.

2. CORRELATED DISTURBANCES WHICH AFFECT TONE AMPLITUDE

The correlated disturbances which affect tone amplitude are amplitude jitter, amplitude hits, and dropouts. g(t) is that function of time which describes the amplitude variation of the test tone. It involves a relatively steady state component having a zero time average over the measurement bandwidth of the test instrument. This is "amplitude jitter." g(t) also has a discontinuous component having a non-zero short time average. This component is called an amplitude or gain hit. An amplitude hit is a positive or negative step in g(t). It is measured with respect to the recent average level of the test tone subject to certain rise time and amplitude criteria. A dropout is a large decrease in level or short term loss of signal measured with respect to the received tone level at the time a test is started or other appropriate reference level. Note the difference between a dropout and a gain hit. The latter is defined with respect to a recent average level of the received tone, also there is no rise time criterion for a dropout, merely a level threshold and a minimum duration. The recently promulgated definition of a dropout is "a reduction in the received tone voltage level by more than 10 db with respect to its value at the start of the test, which level stays below this threshold for 10 milliseconds or more."

Referring to Figure 1, the received tone, cos wt, is shown in a time domain representation, 1a, frequency domain representation, 1b, and as a phasor, 1c. The same tone with amplitude jitter of amplitude K at

FIGURE 1 "AMPLITUDE COHERENT" DISTURBANCES



a circular frequency mt is shown in 1c, d, and e. A representation of an amplitude hit is given in 1f, g, and h. This representation is a snapshot just before and just after the hit has occurred. Of equal interest is the time history of the hit profile during the amplitude traverse between the snapshots and, for that matter, how long does the hit persist, and does it return to reference amplitude.

3. CORRELATED DISTURBANCES WHICH AFFECT TONE PHASE

p(t) is that function of time which describes the short term phase variation. It involves a relatively steady state component having a zero time average over the instrumentation bandwidth (typically 20-300 Hz). This is called phase jitter. Figures 1d, e, and f show phase jitter in the time domain, frequency domain, and as a phasor. Note that the first sidebands of the jittered tone have an opposite sense as compared to the sidebands of an amplitude modulated tone, and that the second sidebands are AM-like sidebands. This can best be understood by considering the phasor representation. The phasor is rotated plus and minus with respect to reference. This produces a quadrature component between the reference tone and the received tone of amplitude sin θ. It produces an amplitude variation of (1-cos Φ). The amplitude variation occurs at a double frequency; that is, as the phasor rotates from zero positively and back to zero, the amplitude goes through a complete cycle. And likewise, as the phasor rotates from zero negatively back to zero, the amplitude goes through a complete cycle. The complete mathematical expansion for a single frequency of phase jitter involves a series of Bessel functions, of which the first sidebands correspond to J_o and the second sidebands correspond to J_1. For the normally encountered jitter, the first two sidebands provide a sufficiently accurate description. Figures 2g, h, and i show a phase hit. This reflects a discontinuous component in p(t) having a non-zero time average. As is the case with jitter, there is a second order in-phase effect. Figure 2 is a before and after snapshot, and it is necessary in the frequency domain representation to consider separately the sine wt and cos wt components of the frequency spectrum. Before the phase hit, the signal energy resides in the cos wt spectrum, and the sine wt spectrum is zero. The hit amounts to transfer of energy from the cosine to the sine spectrum. Note that for a true phase hit, the rms sum of the sine and cosine spectrums remains constant. Also note that there is no reference phase in a communication channel, thus there is no phase equivalent to a dropout. The test instrument tracks the phase of the received tone more or less slowly, and a phase hit is defined as an abrupt change with respect to the recent average phase of the received tone. Again, criteria are needed to define both the rate of change and the amplitude of the hit. Both step hits (no return to original phase reference) and hits which are followed by a return to the original phase reference are found on typical common carrier facilities.

4. NONSYNCHRONOUS OR UNCORRELATED DISTURBANCES

Nonsynchronous disturbances are single frequency interference, steady noise, and impulse noise. Figure 3d shows a single frequency interfering tone which, when added to the test tone, produces a frequency domain

FIGURE 2 "PHASE COHERENT" DISTURBANCES

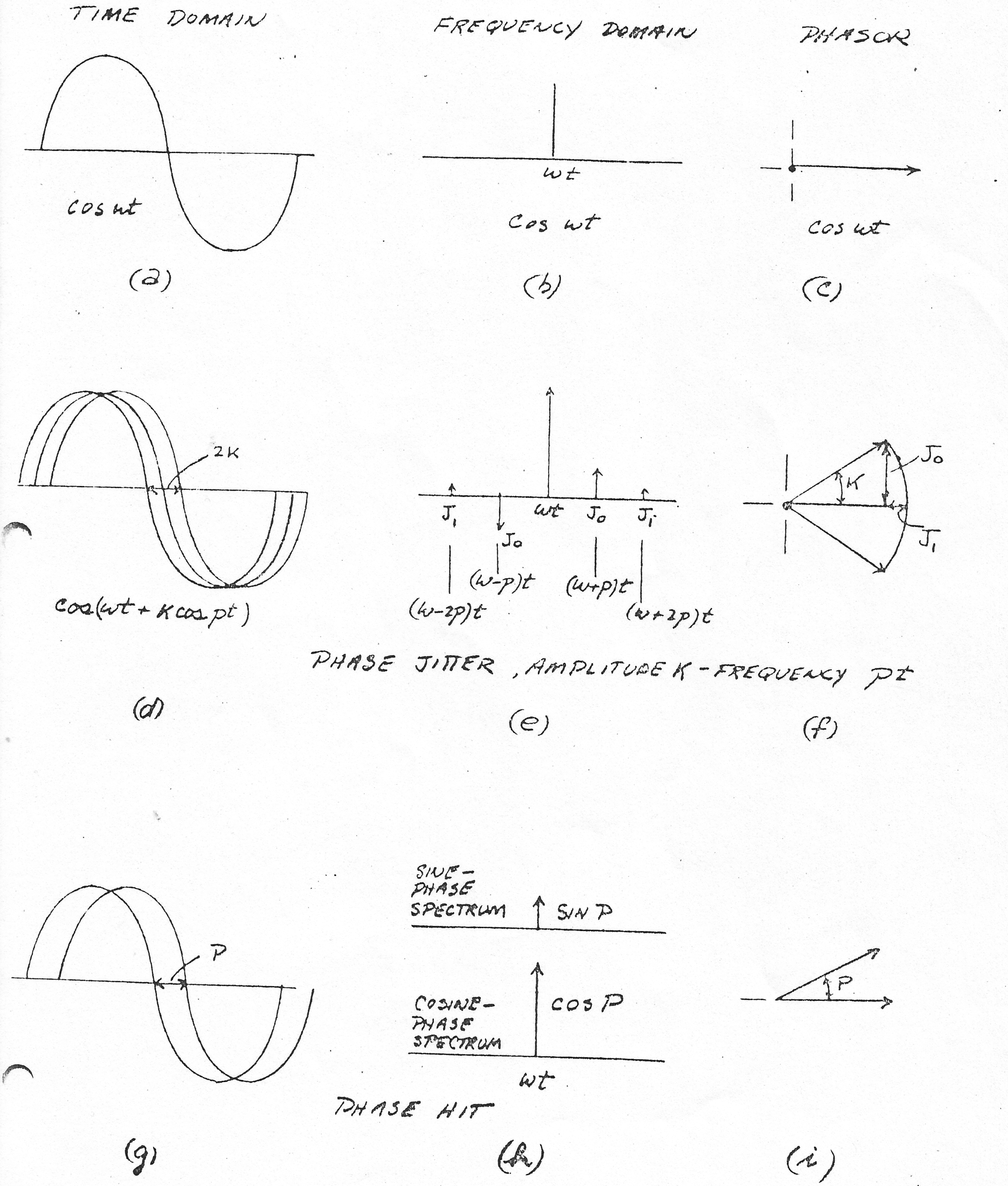
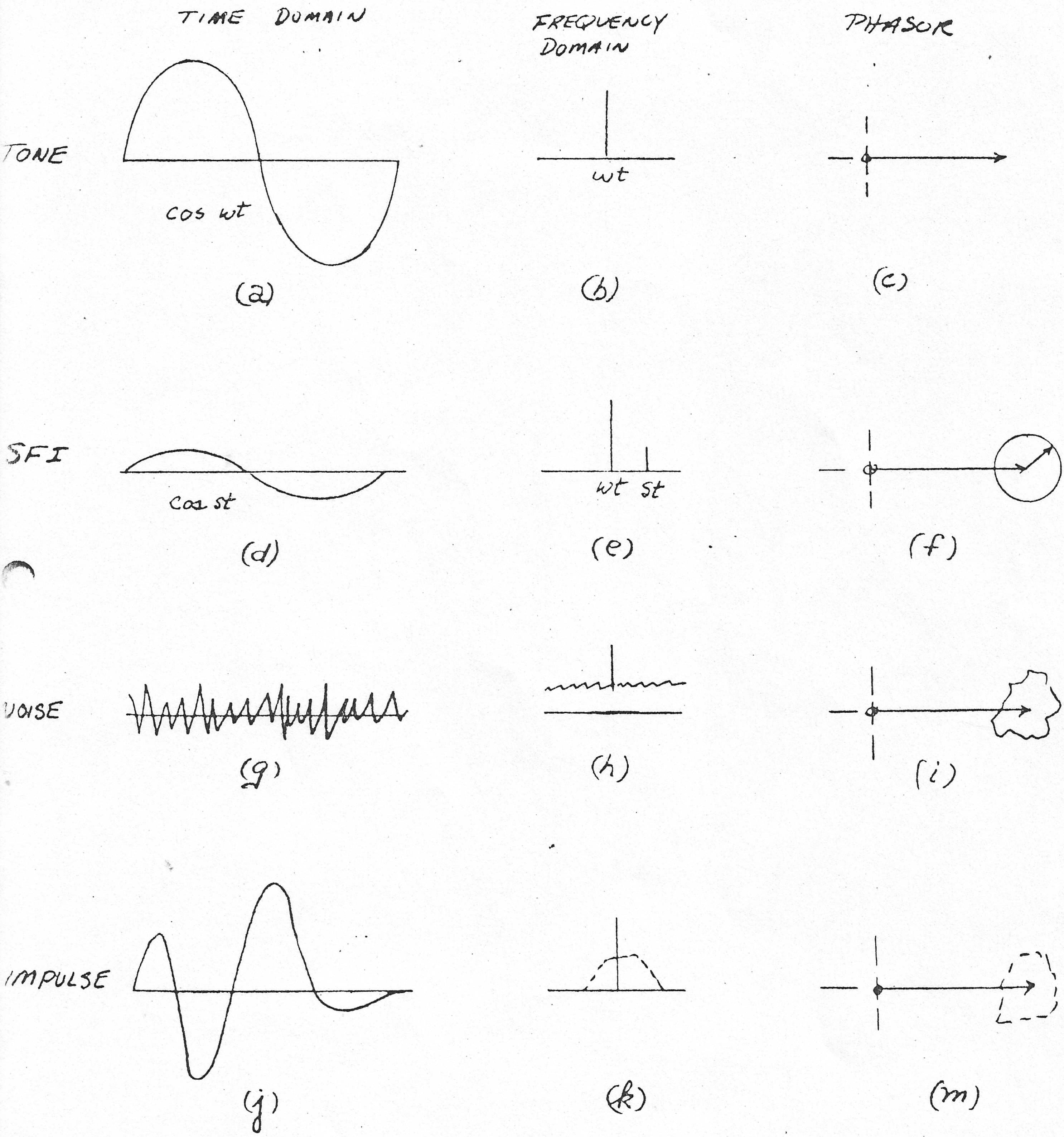


FIGURE 3 NON COHERENT DISTURBANCES



signal as shown in 3e. As represented by the phasor 3f, there is a unit length vector directed to the right representing cos wt, plus the additive SFI vector spinning around the tip of the unit vector. The amplitude of this spinning vector is the amplitude of the single frequency interfering tone relative to the tone amplitude, and the circular frequency of its rotation is the difference frequency between the test tone cos wt and the single frequency interfering tone. It is, in effect, a single sideband modulation.

Steady noise is similar to signal frequency interference except that it comprises a spectral distribution of varying amplitude interfering signals. As depicted in Figure 3h, a phasor representation, there is no single, unique spinning additive vector. The instantaneous value of the output can be any place within the relatively jagged circle. The amplitude of the "circle" is a measure of the summation of the noise factors as they reinforce and interfere with each other.

Figure 3h shows an impulse. Impulses are highly variable. The impulse shown is of the general shape selected to simulate a "typical impulse." Impulses are brief, lasting only a few milliseconds. A typical signal to steady noise common carrier objective is 24dB. A comparable signal to impulse noise objective is 5dB. The difference is acceptable partly because of the short duration of impulse noise, and partly because hit ratios are based on rms/peak, whereas steady ratios are rms/rms. Figures 3h, k, and m are representations of the instantaneous situation during the impulse which will typically have its energy in some band that is not necessarily symmetric about the tone, and which will not necessarily define a circle in the phasor diagram.

SYNTHESIZER MODEL 2A
SECTION 4 - INSTRUMENTATION

1. PHASE JITTER

Phase Jitter Test Sets typically determine the variation in zero crossing of a test tone either with respect to a reference tone phase locked to the received tone, or with respect to successive zero crossings. If a phase-locked reference is used it has a long time constant so as not to track the jitter. The test tone is normally 1000 or 1020 Hz, and the bandwidth is typically 20 to 300 Hz; that is, the test instrumentation is sensitive to phase modulation of the test tone in the 20 to 300 cycle bandwidth. Either instrumentation is incapable of distinguishing between true phase modulation and uncorrelated noise which occupies the same frequency interval as the jitter, i.e., 700 to 980 Hz and 1020 to 1300 Hz for a 1000 Hz test tone. The 1020 Hz test tone is used with "T" carrier to distribute the quantizing noise foldback products over the spectrum rather than having it concentrated at a few frequencies as happens when the tone and the sampling frequency (8KC for T carrier) are harmonically related.

One jitter set attempts to resolve noise and jitter for T carrier with a variable bandwidth filter, and by accomodating a wide range of tone frequencies. The theory is that by manipulating these variables (filter bandwidth and tone frequency), the relative magnitude of jitter and quantizing noise can be estimated. Another manufacturer's jitter set includes a spectrum analyzer, ostensibly to separate the noise from the phase jitter. Since noise in the 700 to 1300 Hz band produces the same spectral energy distribution as true phase modulation around the carrier in the 0 to 300 Hz band, such separation involves shifting the tone frequency and/or level to see which spectral components are tone dependent. A simpler approach is used on non-compandored facilities. The jitter is measured at two different tone levels. Background noise, because it tends to remain constant, makes a larger relative contribution to jitter at lower tone levels than does the true phase jitter, whose energy level varies with the tone. Thus, if the jitter reading remains relatively constant as the tone level is changed, it is indicative of true phase modulation as the predominant disturbance. If the jitter reading is inversely proportional to tone amplitude, it is indicative of (tone independent) noise as the source of apparent jitter.

2. AMPLITUDE JITTER

Amplitude Jitter exists in communications networks and, as combination phase/amplitude encoded and/or vestigial sideband modems come into greater use, one may expect a similar experience to that of phase jitter a few years ago. Indeed, the phase jitter problem in the normal (non-trouble situation) network is now under control while the effort has not yet begun with amplitude jitter. To the best of our knowledge, there is only one instrument available for the measurement of amplitude jitter, our Model 75LMA, which measures both phase jitter and amplitude jitter in the 20 to 300 Hz bandwidth. This instrument uses the same post detection filter and peak detector for either PM or AM measurement.

As a diagnostic tool in combination with phase jitter, amplitude jitter is perhaps better than the diagnostic techniques for separating noise and jitter which are described above because it can be used at fixed tone level and tone frequency, i.e., someone does not have to camp at the far end. Uncorrelated noise makes an equal contribution to amplitude jitter and to phase jitter. Thus, widely different amplitude and phase jitter readings (reduced to an equivalent sideband energy basis) are an unequivocal indication that steady noise (including quantizing noise) is not the offender. If the amplitude and phase jitter readings are equivalent, then either the amplitude jitter and phase jitter are equal (unlikely, since they come from different sources both of which must be in trouble) or the readings are the result of uncorrelated noise (steady noise or SFI). One significant advantage of this is that this diagnostic technique is independent of the type of facility (compandored or not). This makes it convenient for the common carrier craftsman; he does not have to research the channel configuration. It also makes the technique feasible for the customer, who may not have access to channel configuration information.

Of course, communication channel problems are quite diverse. It is possible to devise instrumentations and measurement techniques to identify individual problems, but the existence of other impairments, which confuse the results, is virtually invariable. The experience of the craftsman is what makes any diagnostic system effective.

3. STEADY NOISE

Steady noise is measured

1. With or without a holding tone
2. Relative to the signal level or in terms of its own absolute level
3. With a variety of filter weightings
4. With an rms or quasi rms detector

1. With a holding tone on a compandored facility, i.e. a facility with agc, a holding tone is required to establish the gains of all the agc amplifiers at some nominal level rather than having them wide open, thus emphasizing noise. In this case, steady noise is dependent upon the distribution of agc gains in the channel so that the more meaningful parameter is signal to noise ratio. The tone is notched out by filtering, its amplitude measured, and the energy in the signal minus the tone is measured. The ratio of these two quantities is displayed as signal to noise ratio. Notch noise includes steady noise, phase jitter sideband energy, amplitude jitter sideband energy, and harmonic distortion, i.e., everything but the tone fundamental.

2. When noise is measured without a holding tone on non-compandored facilities, a quiet termination is patched to the source end, and the noise level read at the receiving end. No notching or signal manipulation is involved other than detection, and the noise level is typically displayed in dBrnc. Zero level dBrnc = -90dBm.

3. Filters are used to weight the noise components. 3Kc flat, C-message, and sophometric weightings are commonly used. C-message is an experimentally derived weighting which represents the annoyance and reduced intelligibility factors of noise at various frequencies on a standard telephone instrument with reference to a 1000 cycle tone. C-message weighting was established in the United States; sophometric weighting is a similarly determined form of filter weighting which was established in Europe. Both of these have a low frequency roll-off resulting in a high attenuation at 60 Hz (and 50 Hz) relative to the 3Kc flat filter. These filters are all quite flat in the bandwidth used for data transmission, so that C-message weighting (or sophometric) is also used in data channels. Diagnostically, a high noise reading with a 3Kc flat filter relative to the reading with either of the others is an indication of the presence of power line or ringing tone.

4. Detectors. Noise level is the rms power level of the noise signal, and is properly measured by a true rms detector. It is also frequently measured by a "quasi rms detector." A quasi rms detector is a combination of the outputs of an average value and a peak value detector, with the contributions of each weighted so that the output is correct with either a pure sine wave or for a Gaussian distribution of the input signal (but not for most energy distributions other than these two). An average value detector normally consists of a conventional averager scaled so that it reads true rms value for a sinusoidal input. The average value-calibrated rms detector understates the rms value of a Gaussian distribution input by 1dB.

4. HIT COUNTERS

The definition of hits is complicated by the need to include a time history description of the event and of the measurement interval (typically 15 minutes) over which the count is accumulated.

Also, typically, hit counters have a built-in deadband of 1/7 sec.; i.e., after a hit event has been identified, it is counted in the one or several applicable registers, and the detection circuitry is disabled for 1/7 second. This procedure came about historically; early hit counters used electro-mechanical counters whose response time was limited. Not only was a great deal of data developed under these conditions, but criteria were established for relating this data to line performance on a continuous basis. See page 15 Bell System Technical Reference Pub 41008 Analog Parameters Affecting Voiceband Data Transmission-Description of Parameters-October 1971.

In the case of a dropout, after the event is counted, all hit detection circuits are disabled for the duration of the dropout. They are not again enabled until the test tone exceeds the threshold level for a specified period, i.e., 10 milliseconds.

5. IMPULSE

Impulses are measured on a peak power basis with essentially negligible duration requirements, e.g., if the peak power exceeds the threshold level for 50 microseconds, an impulse is counted. Many impulse counters have multiple threshold level detectors counting into separate registers. Logic of these instruments is such that, for a given impulse, a threshold event is counted in all registers up to the highest register which is triggered. The results displayed by the registers represent a statistical distribution of impulse amplitudes.

6. PHASE HITS

Phase hits are time coherent; i.e., synchronous with the test tone, although short duration hits of this type may not appear to be so because they start at a random time during the tone cycle. The problem of distinguishing between these correlated hits and impulse hits, which are uncorrelated, is handled by establishing a "guard band" of two to four milliseconds (depending upon particular instrumentation). This is long enough to test the signal for coherency (and quadrature). The four millisecond interval (of the older instruments) was established also because 90% of measured impulse hits are shorter in duration than four milliseconds; hence the confusion between impulse and phase hits is reduced. There are subsidiary difficulties associated with filter ringing and/or delay which reduce this apparent time margin. In order not to miss short phase hits, a shorter guard band is preferable <u>if</u> impulse hits can be discriminated against.

Another problem in measuring a phase hit is indicated in Figure 4, which plots phase vs time for a hit. The hit is not instantaneous, channel bandwidth and other considerations assure that. The phase reference, as generated by the instrument phase lock loop, tracks the test tone phase more or less slowly. Thus a higher hit threshold is reached than the hit counter threshold setting, before a count is recorded. Of course, the ideal hit counter matches the tracking rate of the modem for which the channel is to be used, but this is impractical for a general purpose instrument.

There does not yet exist a generally accepted definition of instrument response to non-instantaneous, i.e., all, phase hits and, as a result, phase hits counted by different manufacturers' hit counters are not the same (because most phase hits have a rise time from two to five milliseconds). We would propose that the phase tracking loop response be defined as follows:

The tracking time constant of the phase lock loop shall be such that a 45° hit at a rise time of 10 milliseconds shall be accurately recorded. A reasonable procedure is to use a fast rise time phase hit and adjust its amplitude so that the hit counter just counts. This should be within the hit counter tolerance. Then increase the hit rise time to 10 milliseconds and observe that the hit counter continues counting.

7. GAIN

The considerations for gain hits are essentially the same as for phase hits except that test tone amplitude rather than test tone phase is examined.

APPENDIX 2 - ABSTRACT FROM SYNTHESIZER INSTRUCTION BOOK

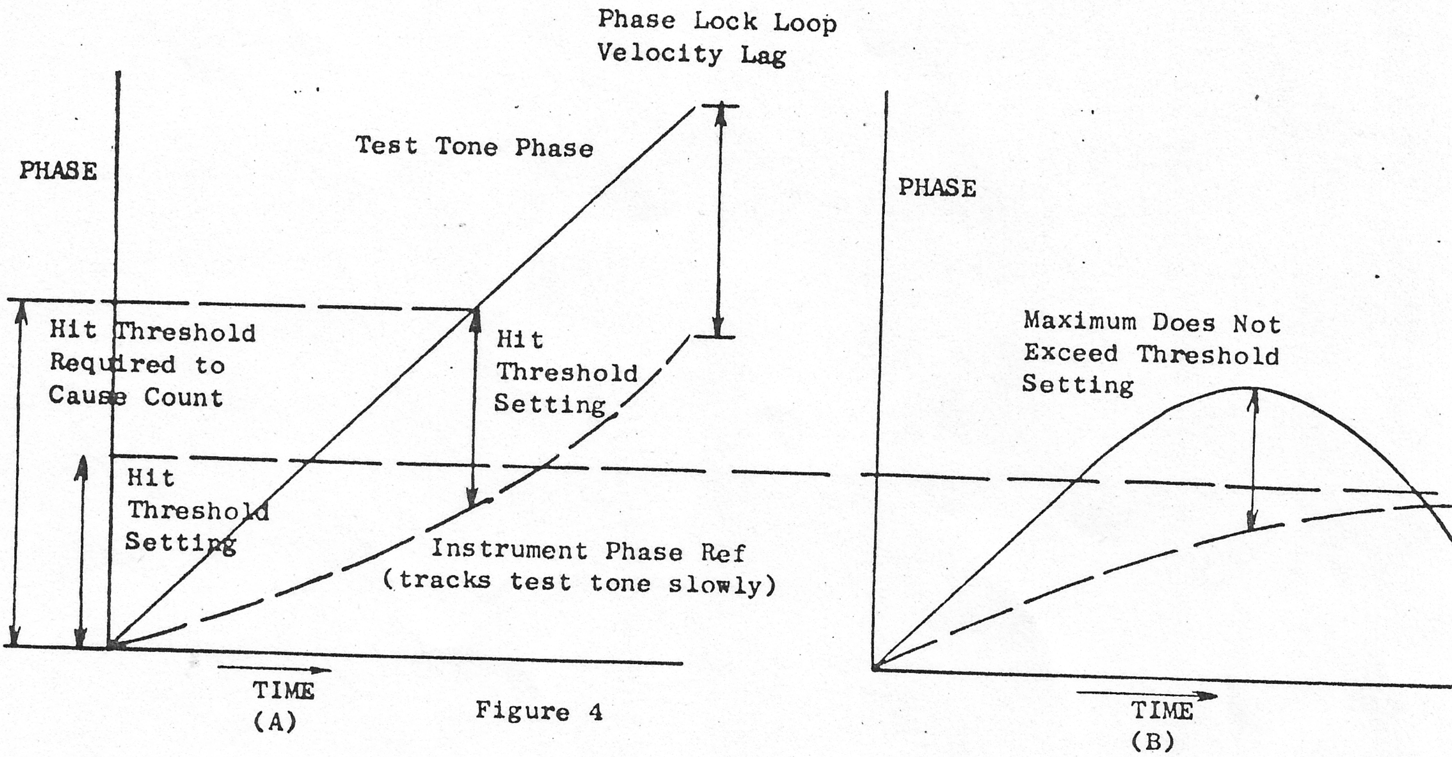


Figure 4

One of the problems involved in measuring a phase hit is depicted in Figure 4(A) above, a plot of phase vs time at the start of a phase hit, see solid line labelled Test Tone Phase. The hit is not instantaneous, channel band width and other considerations assure that. Most phase hits, as measured on the AT&T network, have a rise time between 2 and 5 milliseconds. The "Instrument Phase Ref," an internally generated test tone, tracks the test tone phase more or less slowly, and typically with some velocity lag. Thus the hit level required to cause a count is dependent not only on hit amplitude, but on hit rise time. Figure 4B indicates the situation for a hit which exceeds threshold but is not counted because its rise time is too slow. Differences of up to 4 to 1 can be found in hit counters, i.e., a hit having a slow rise time, e.g., 10 milliseconds, may have to be as large as 45° to be counted in the 15° register. This is one of the many inconsistencies in instrumentation which we hope will be gradually corrected by our Synthesizer.

8. DROPOUT

This is an absolute level threshold which is established at the start of a test. There are no rise time considerations; a dropout is counted if the level falls below this threshold for a specified time. A typical definition is 10dB below start of test for at least 10 milliseconds. There is no upper limit on the dropout time, and only one dropout is counted no matter how long it lasts, until the signal goes above the threshold for at least 10 milliseconds, whereupon the dropout is considered over. The next excursion below threshold (for at least 10 milliseconds) is counted as another dropout.

SECTION 5 - MISCELLANEOUS APPLICATION COMMENTS

1. FAMILIARIZATION PROCEDURE

To stop the internally generated tone on an oscilloscope display for viewing phase jitter, sync on the tone oscillator of the synthesizer. One can also view AM and SFI (Single Frequency Interference) in this way. In this case the tone component will be stationary and the modulation will be moving. It is also possible to sync off the modulation oscillator so that the envelope is stationary and the tone moves. It is also possible to sync on the HIT so that hit events are stationary.

1. Scope Sync External, sync off the TONE jack. Vertical input on the OUTPUT jack. The synthesizer has all impairment switches off, SIGNAL SOURCE (center left) switch in the TONE FREQUENCY position TONE FREQUENCY (lower left) dial at 1000 (Hz). OUTPUT LEVEL 0dbm (switch -10, pot +10).

   Set scope sweep rate for two or three cycles of the tone visible and the vertical gain so that the sine wave is on scale.

2. "EXT TONE (input for modem or other signal source). Switch SIGNAL SOURCE back and forth and observe the sine wave disappears. If modem signal were present, this would appear at the output in the "EXT TONE position.

3. MODULATION FREQUENCY to 150 Hz (or thereabouts) x 1 on switch and 150 on vernier.

4. PHASE MODULATION ON, move PHASE MODULATION DIAL up to cause jitter.

5. Vary modulation frequency to observe that capability.

6. PHASE MODULATION OFF, AMPLITUDE MODULATION ON and repeat.

7. AMPLITUDE MODULATION off and SFI on and repeat. SFI is additive tone at modulation frequency. SFI appears in the output as an additive tone, not as modulation. Hence it is subject to the frequency response characteristics of the output transformer and, below 60Hz is attenuated.

8. SFI off, NOISE on and vary level.

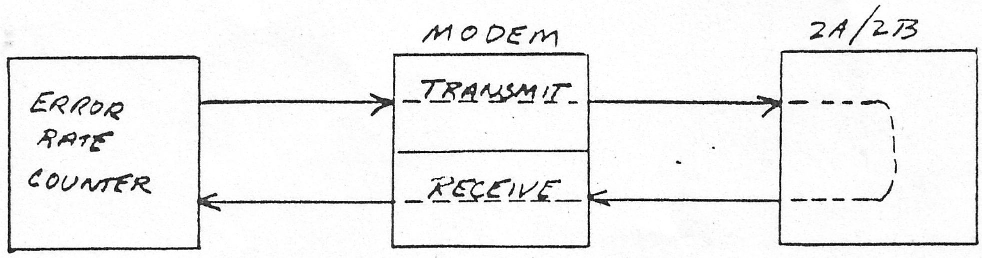
9. Switch Tone off (SIGNAL SOURCE) to observe noise by itself.

10. Also observe EXT inputs. Set the NOISE Level to a high value but leave the NOISE ON/OFF switch in the off position. Patch NOISE output jack successively to EXT PM, EXT AM and EXT ADD (with tone on). Observe pm (movement at zero crossings) and am (movement at peaks).

11. HITS either manual by PUSH TO HIT button (toggle up) or continuous (toggle down). The interval, rise time, and duration of the hits are adjustable. Note that the duration must be greater than the rise time and the interval must be greater than the duration plus the rise time. A good demonstration is 0.1 second interval, 50 millisecond duration, 0.2 millisecond rise time. Nothing happens until the Phase Hits Toggle, Gain Hits Toggle or Impulse toggle is turned on, even though the hit pulse is being generated. Turn phase hits on and adjust hit amplitudes up and down, ditto gain hits. The gain hit dial includes dropouts.

    Impulse - fixed waves shape which occurs only on lease edge of hit. Observe it with and without tone (switch to -80, pot +10).

2. MODEM EVALUATION CAPABILITY - Set up, with all impairments off, so that error rate counter, see Figure below,



is not counting any errors. This usually involves only getting the levels and impedances correct, however, it may be more complicated if the modem is of the manual equalization type.

Error rate counters generate bit sequences which are sent through a line (synthesizer). The received data stream is compared with the transmitted data stream and either, or both, bit errors and block errors are counted. A block is a group of bits of arbitrary but specified length. A block error is counted when one or more bit errors occurs in a given block.

The characteristics not simulated in the 2A/2B are the steady state impairments, i.e., gain vs. frequency and envelope delay vs frequency. The 2A Synthesizer is essentially equivalent to a Bell System C5 line. This specifies the steady state parameters above. Actually it is better in gain vs. frequency and slightly over on the C5 line spec for envelope delay vs. frequency. See data below.

TYPICAL DATA

Frequency Characteristics of 2A or off 2A and 2B with Frequency Translation Off

FREQ (HZ)	LOSS (dB)	EDD (MICROSEC)
281.	-.1	579.
437.	.0	313.
594.	.0	203.
750.	.0	145.
906.	.0	105.
1062.	.0	76.
1219.	.0	54.
1375.	.0	37.
1531.	-.1	22.
1687.	-.1	11.
1844.	-.1	.
2000.	-.1	-11.
2156.	.0	-22.
2312.	.0	-32.
2469.	.1	-41.
2625.	.1	-47.
2781.	-.1	-49.
2937.	.0	-42.
3094.	-.1	-31.
3250.	.0	-24.

Frequency Characteristics of 2A and 2B with Frequency Translation On

FREQ (HZ)	LOSS (dB)	EDD (MICROSEC)
281.	.0	1169.
437.	.0	635.
594.	.0	413.
750.	.0	293.
906.	.0	214.
1062.	.0	157.
1219.	.0	112.
1375.	-.1	77.
1531.	-.1	47.
1687.	-.1	22.
1844.	-.1	.
2000.	-.1	-21.
2156.	-.1	-40.
2312.	.0	-58.
2469.	.0	-73.
2625.	.0	-85.
2781.	.0	-92.
2937.	-.1	-90.
3094.	-.1	-83.
3250.	.0	-82.

Frequency Characteristics of Harmonic Distortion Section

FREQ (HZ)	LOSS (dB)	EDD (MICROSEC)
281.	-.2	-6.
437.	-.1	-4.
594.	-.1	-4.
750.	.0	-5.
906.	.0	-5.
1062.	.0	-5.
1219.	.0	-4.
1375.	.0	-4.
1531.	.0	-2.
1687.	.0	-1.
1844.	.0	.
2000.	.0	.
2156.	.1	.
2312.	.1	-3.
2469.	.2	-4.
2625.	.2	-5.
2781.	.2	-4.
2937.	.2	1.
3094.	.1	10.
3250.	.2	17.

Modems are equalized for line characteristics, particularly envelope delay. Envelope delay has the effect of making the high frequency components of the modem signal travel or arrive at a different time from those low frequency components which were launched into the line at the same time. Some high speed modems are equalized manually, mostly the older designs, most are equalized automatically.

When connecting a modem note that the peak to peak input level at the EXT INPUT jack is 2.2 volts maximum. This corresponds to 0dbm for sine wave but not modem signals. Modems normally have higher crest factors. Note the gain through the synthesizer is 0db (unity) when the OUTPUT LEVEL dial is set to 0dB (-10 switch, +10 pot). Frequently a loss of 16db is desired in the transmission path or rather an output level is desired of -16dB. If this is what is wanted with an input signal coming in, for example, at -3, then the OUTPUT LEVEL switch should be set to -13 to give a minus 16dBm net output level.

If a scope is kept on the output at all times it is possible to switch back and forth between the modem and the internal tone so that the impairment can be viewed as it effects the tone and hence recognized by inspection.

The modem is now playing a data stream via the synthesizer, out of, and back into the bit error rate counter. All impairments are "off" and the error rate counter is counting no errors. This will involve matching levels at the synthesizer input and output as well as matching impedances. Make sure that the peak to peak amplitude of the input signal to the synthesizer is 2.2 volts or less. This may involve transmitting at some level between -3 and 8dB, depending upon the cross factor of the modem. The output level of the synthesizer can be adjusted relative to the input level. The synthesizer has zero db gain or unity gain at zero dbm output level setting. To increase the level through the synthesizer, set it above OdB, up to +10dB (which provides 10dB gain through the 2A). To drop the level set it below OdBm by the attenuation desired.

PHASE MODULATION

Set the modulation frequency to, for example 60 Hz, switch the toggle switch on and move the phase modulation level slowly up until the error rate counter starts counting errors. Log this magnitude of jitter.

Repeat at other frequencies which may be of interest, typically, 20 to 300 Hz. The result of this is a curve which defines the sensitivity of the modem to phase jitter frequency.

AMPLITUDE MODULATION

Same as above except with amplitude modulation dial. A curve on the following page relates % AM to amount of AM sideband energy.

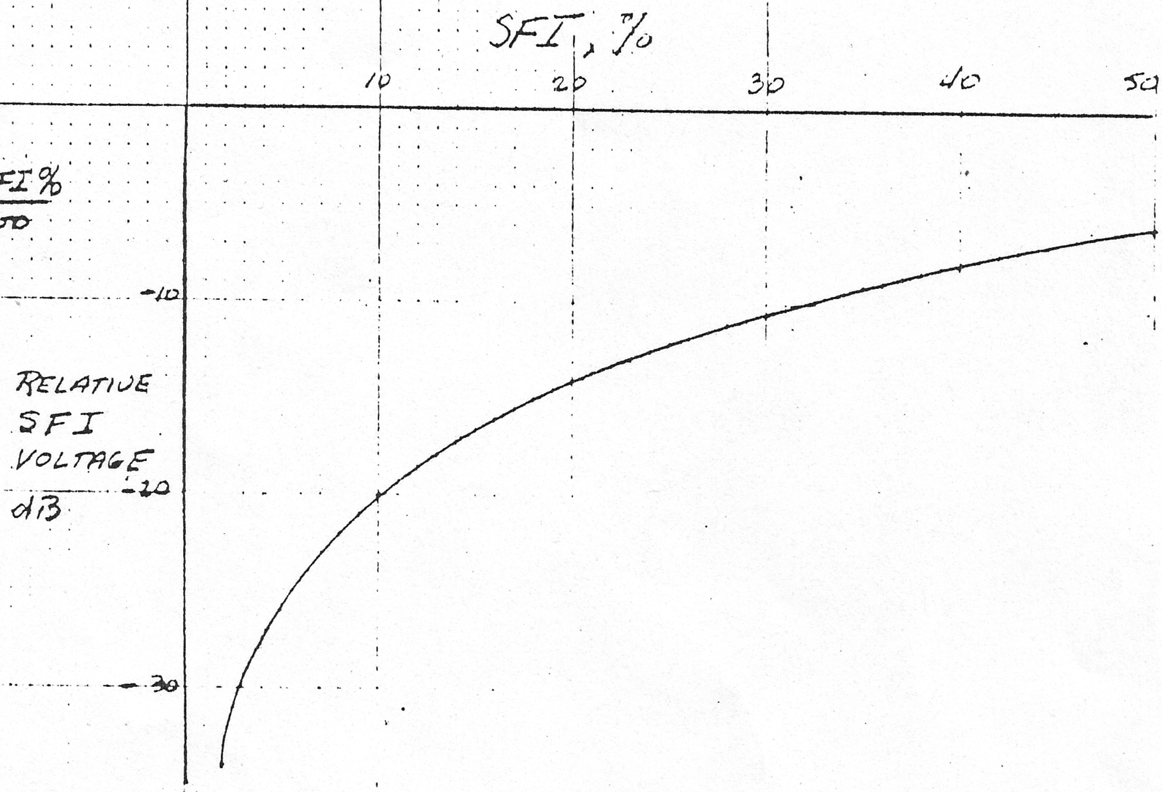
SINGLE FREQUENCY INTERFERENCE (SFI)

This is complicated because the dial is in per cent relative to zero level signal (at output) - when working from the external tone input only, see correction factor tables below. For internal tone the dial is calibrated correctly at all output levels.

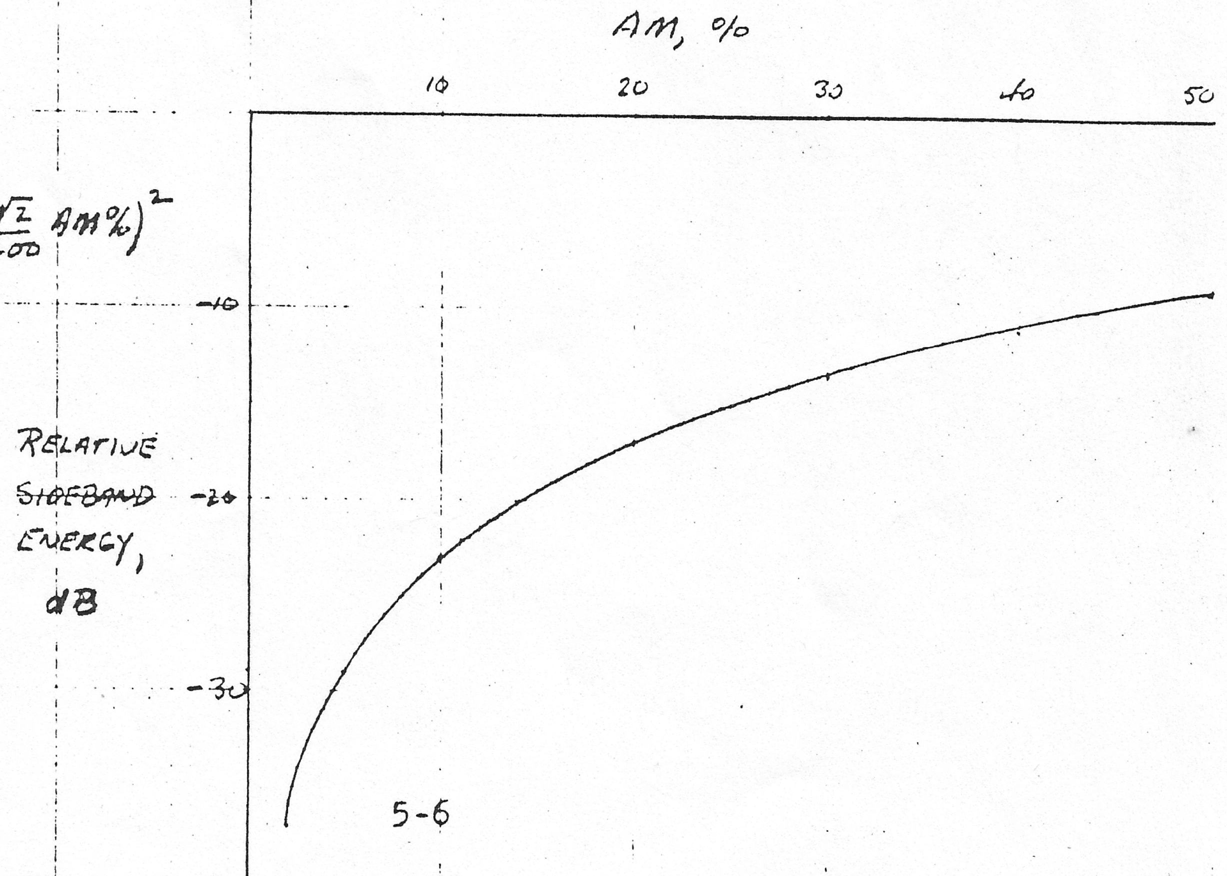
EXTERNAL TONE LEVEL AT 2A SYNTHESIZER OUTPUT JACK, dBm	SFI SCALE MULTIPLIER
0	x1
-3	x1.41
-6	x2
-9	x2.83
-12	x4
-16	x6.31
-20	x10

The SFI frequency, which is most disturbing to modems is in the vicinity of modem carrier frequency (typically from 1600 to 2200 Hz). Within above ground rules one can generate the same kind of curve for SFI. The most commonly encountered SFI tone in actual operation in the power line frequency. A curve on the following page relates % SFI to dB SFI.

$$dB = 20 \log \frac{SFI\%}{100}$$



$$dB = 10 \log \left(\frac{\sqrt{2}}{200} AM\%\right)^2$$



NOISE

Noise level is in dBrn, weighting is flat to 3.5KHz with 12db per octave roll-off thereafter. dBrn is dbm -90, i.e., OdBm = -90dBrn. The critical factor is the noise level in relation to the tone level (signal to noise ratio).

For example, a 20dB signal to noise ratio means that the noise level itself is 20dB below the tone level, i.e., if the tone level is -16dBm the noise level is -36dBm or +54dBrn (dial setting of noise attenuator).

The situation is complicated by the noise source bandwidth and/or any filtering that may be in the line for simulation purposes. Completely consistent results between simulation methods requires that the noise source have identical bandwidths or are measured in such a way that the measurement response is equivalent for the two sources.

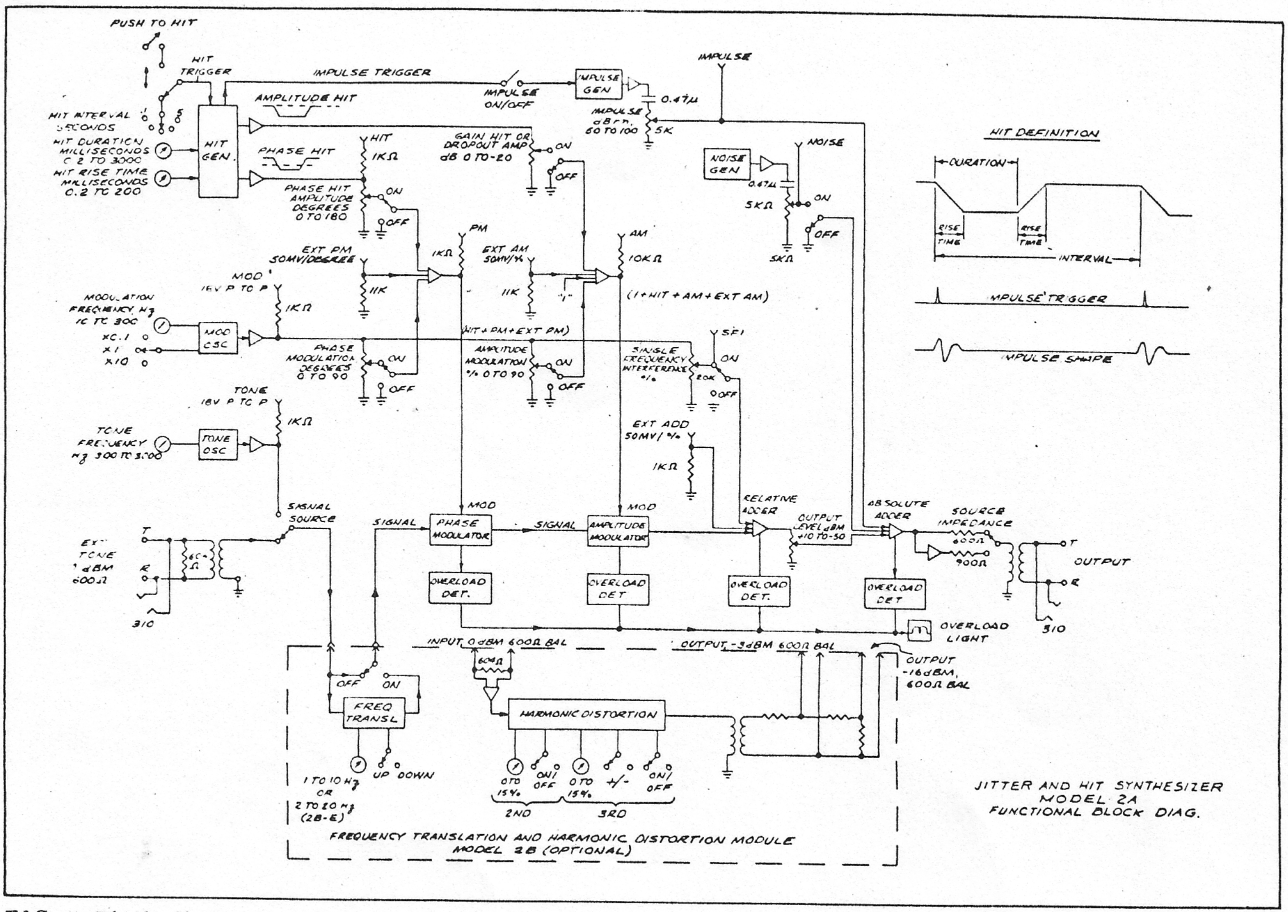


FIG. 1 Block diagram of data line synthesizer for simulating various data transmission anomalies.

Modem evaluation by simulating line impairments

Despite advances in measurement techniques, data circuits and modems don't always behave the way the instruments say they will—here's a way to get around measurement weaknesses

Frank Bradley

A VOICE BAND data communication channel can be defined in terms of performance characteristics which can be corrected or compensated for and those which cannot. In the correctable category are variations in amplitude gain or loss and envelope delay versus frequency. In the category of characteristics which cannot be compensated for are such steady-state disturbances as frequency shift, phase jitter, amplitude jitter, background noise and harmonic distortion. Impulse noise, phase hits, gain hits, coincident phase and gain hits, dropouts and frequency hits are transient in nature.

Because they do not interfere with voice communication, jitter and hit phenomena have not had the benefit of the half-century of attention which has been devoted to voice traffic impairments. More recently, however, as a result of the existing and anticipated revenue from data traffic, the industry has devoted considerable effort to identify

FRANK BRADLEY is president of Bradley Telcom Corp., New York.

and define these second order impairments which are so disturbing to data circuits. The details of the existing descriptions leave something to be desired. Perhaps this will always be. Data channels operating near their full information throughput capacity are susceptible to even brief transient disturbances. These transients are variable in nature and difficult to describe specifically.

The best defense against factors which disrupt data service is to design modems with a high level of immunity against anticipated disturbances. In the design, choices must be made which may favor a particular type of data traffic, e. g., block data versus real time, or which discriminate against one error source better than another. For example, the choice of damping in the tracking loop amounts to a guess about the duration of phase hits and the relative ratio of hits to steps.

Unfortunately, line faults do not occur predictably, and thus the design process is prolonged with lengthy field evaluations (sometimes unknowingly by the user). Corrections are devised which are based on insufficient observation of evanescent phenomena. This has led to successive generations of equipment, but not all have been forward steps.

Fault simulation

One way around this problem is to generate or synthesize simulated faults. Then individual impairments and combinations of impairments may be generated continuously, and response of the equipment to them studied. The Bradley Telcom Model 2A Synthesizer was designed for this purpose and provides a predictable, controllable and realistic source of jitter, hit and noise phenomena which are continuously variable in all significant parameters. The simulator allows the data set designer to evaluate alternatives, permits the data set tester to verify that performance criteria are met and enables the user to evaluate competitive modems. All three can use the device to calibrate test equipment and to train personnel.

LINE DISTURBANCE synthesizer is first of its kind for systematically providing means for simulating disturbances which disrupt data.

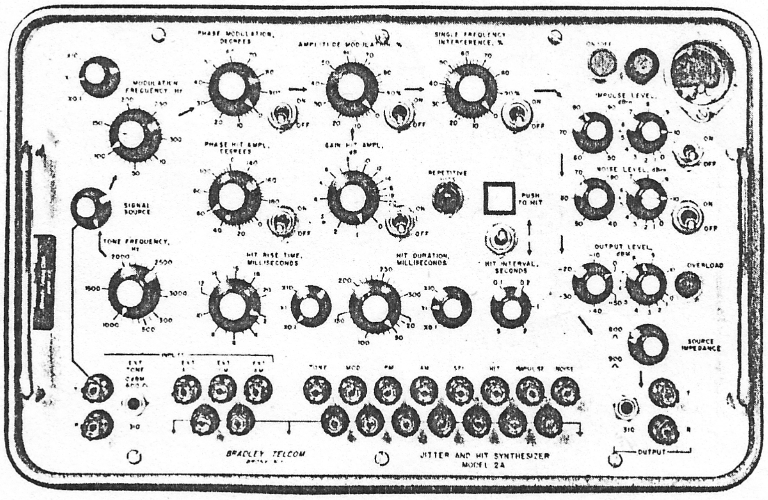


Figure 1 is a block diagram of the simulator. The data stream from the modem under test passes through a phase modulator, an amplitude modulator, an SFI (single frequency interference) adder, a calibrated attenuator and an adder for steady or impulse noise.

As an alternative input, the variable frequency tone oscillator may be used as as signal source instead of the modem signal. The tone oscillator permits the generation of test tones with calibrated impairments for the evaluation or calibration of jitter, noise and hit sets, as well as for the training of maintenance personnel.

The modulation oscillator provides a second variable frequency source which may be applied in calibrated amplitude to the phase modulator to generate phase jitter, to the amplitude modulator to generate amplitude jitter, or to the SFI adder to generate single frequency interference. Access for applying external signals of any desired spectral content to the modulators and adder is provided. The hit generator provides pulses at a selectable interval, with adjustable rise time and duration. These pulses may be selectively applied at calibrated amplitude to the phase modulator and the amplitude modulator to generate phase hits, gain hits, coincident phase and gain hits, as well as dropouts.

The modulation-impaired signal is then coupled, via a calibrated attenuator, to the steady and impulse noise-adder where noise may be added. The impulse generator is triggered by the leading edge of the hit pulse and generates a fairly uniform spectral energy distribution from 800 to 2000 Hz.

The characteristics of the simulator that make it particularly

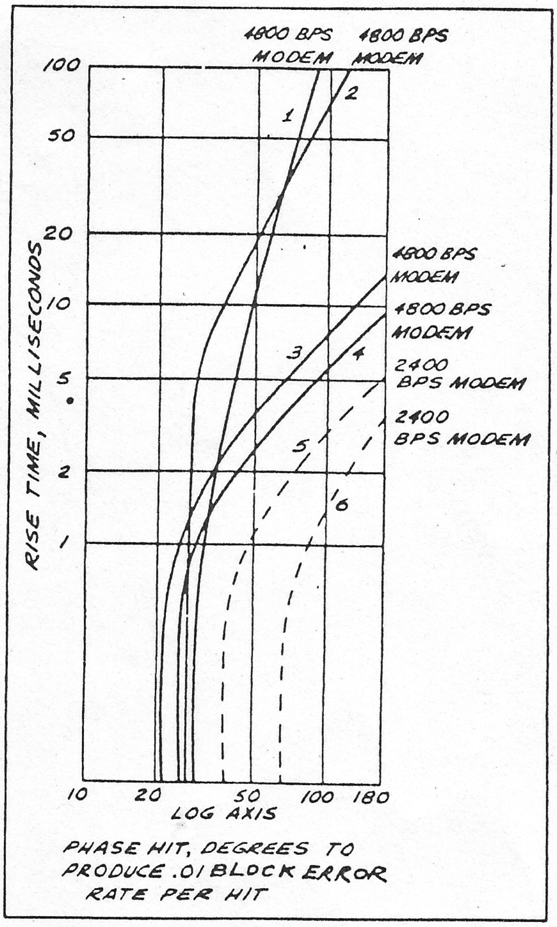


FIG. 2 Performance comparison of six well-known modems. Curves plot the dependence on modem sensitivity to rise time to produce errors under controlled circumstances.

useful are the accuracy with which it generates continuously variable impairments, the controlled-profile hit capability and the ability to accept externally generated impairments.

High grade data lines are specified in terms of their steady-state gain and envelope delay characteristics. The synthesizer, from input to output, is the equivalent of the Bell System's C-5 line. It is noteworthy that the disturbances generated by the synthesizer are, for the most part, not controlled by line specifications. Such disturbances are equally likely to appear on any line, regardless of grade. The span of all impairment variables in the synthesizer has been selected to include the normally encountered range as seen on real communications channels. All significant variables are available at panel jacks. It is possible, for example, to phase modulate with either impulse noise or steady-state noise by patching the appropriate output to the external PM input.

Modem performance evaluation

The curves which follow reflect an application of the synthesizer for evaluating modem performance with respect to phase hits. In the real world, transient phase changes occur in the form of phase steps or, more frequently, as phase hits. The phase step is a uni-directional change in phase a one-way hit. A phase hit is a brief excursion in phase—the phase departing from, and returning to, its original value. The change in phase is rarely instantaneous (in terms of channel bandwidth). The range of risetimes in voice channels is, for the most part, between 2 and 5 msec. (One commercially marketed hit counter responds to phase changes so quickly that it virtually ignores phase hits with rise times slower than 1 msec.—i. e., most phase hits as they are normally defined.) Most phase hits are longer in duration than 4 msec., with the average duration being, perhaps, around 30 msec. One speculates that hits are contributed by carrier supplies and that steps are the result of changes in path or frequency hits.

Figures 2 through 6 are concerned with single impairment phase hits. The data were taken without going through a transmission facility; the impairment, in each case, is a pure phase hit uncontaminated by any other disturbances. The independent variables are hit amplitude, rise time and duration. The dependent variable is error rate threshold. The curves represent one step in the design (or evaluation) process.

Figure 2 is a comparsion of the sensitivity to phase hits as a function of rise time of six of the

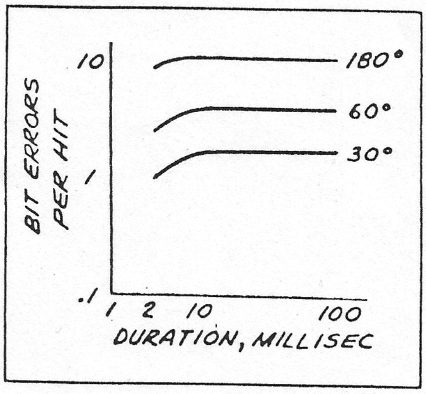


FIG. 3 Errors per hit at three different hit levels. Rise time is held constant.

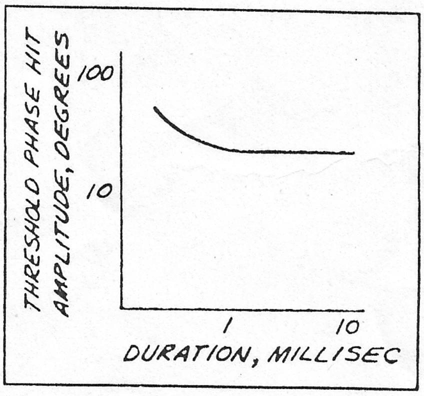


FIG. 4 Hit amplitude threshold which will produce errors, as a function of hit duration.

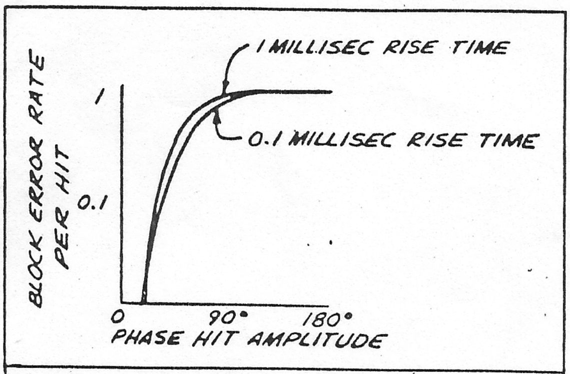


FIG. 5 Block error rate as a function of hit amplitude for two values of rise time.

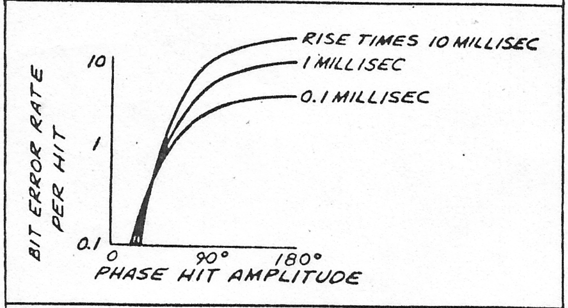


FIG. 6 Error rate per hit as a function of rise time and amplitude of hit.

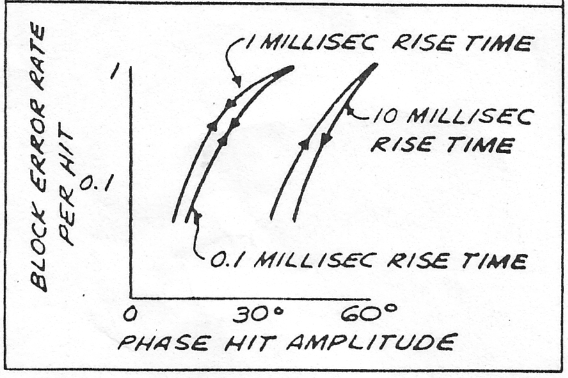


FIG. 7 Sensitivity of modem to hit polarity is revealed in this comparison. Arrows in both directions on left curve indicate curves are actually superimposed—identical error rate for both polarities. Right curves reveal greater sensitivity to positive-going phase hits.

most widely used modems. The curves show the 0.01 block-errors-per-hit threshold. These modems are either phase modulated or vestigial sideband, amplitude modulated.

The data show that the 2400 bit modems are less sensitive to hits and that faster rise time hits are more effective in causing errors. For example, where rise time is rapid, a smaller hit amplitude is required to reach the error threshold. Of the 4800 bit modems, 1 and 2 are less sensitive to fast rise time hits than are 3 and 4, but that for hits above 2 msec., (i. e. most phase hits), 3 and 4 perform better than 1 and 2.

This is, incidentally, a threshold phenomenon and a two-to-one difference in hit error sensitivity can make a much larger difference in data throughput. For example, a line with 8° to 10° steady-state phase jitter, plus some background noise, plus a steady stream of phase hits, may play well through modem 4 and poorly through modem 2.

The remaining curves illustrate the peformance of one of the 4800 bit modems compared in Figure 2. Figure 3 shows the bit errors per hit at three different hit levels for a constant rise time (1 msec.) as a function of hit duration. Note that with a 1-msec. rise time (and fall time), the shortest hit duration is 2 msec. The curves show reduced errors for short-duration hits because the phase of the signal moves over and back before this particular modem can sense the change. But beyond a certain threshold of hit duration, both the phase step at the beginning and the phase step at the end of the hit cause a full complement of errors.

Figure 4 shows the threshold phase hit amplitude required to cause errors, as a function of hit duration, for fast rise time hits (0.1 msec.).

In Figure 5 the block error rate per hit is shown, and in Figure 6 the bit error rate per hit as a function of hit rise time. Figure 7 indicates the polarity sensitivity of the modem. For fast rise times, the error rate is the same for positive- and negative-going hits. But for a 10 msec. rise time, this particular modem was more sensitive to positive-going hits than negative-going hits.

These curves are indicative of the type of testing and evaluation that can be performed with a line impairment simulator. But they only scratch the surface of a full test program for selecting the best modem for a particular application or for optimizing the design of a modem. Many families of curves can be generated in addition, if steady noise, amplitude jitter, phase jitter (level and frequency) and impulse noise are added to the investigation (as indeed they are automatically added in the real world by the transmission facility).

The device can also be used to evaluate hit counters and phase jitter test sets by using a test tone rather than a modem signal, and a major added criterion then becomes the amplitude of impulse hits—which can cause confusion in identifying the hit as either in phase or gain.

After a fifty-year delayed start, as compared to voice communications, substantial attention is now being given to data transmission. The 2A Synthesizer is but one of a new class of instrumentation designed to cope with the peculiarities of data transmission, which can be expected to appear and proliferate in growing numbers as the field responds to the surging demand for more and better data transmission. □