CHAPTER 12
KINEPLEX DATA TRANSMISSION

1. INTRODUCTION

The tremendous growth of industry has resulted in an increased demand for better communication. To satisfy this demand, there has been a large scale expansion of radio and wire facilities to provide additional communication channels. The paramount problem in long range radio communication is congestion of the frequency spectrum and the heavy investment attendant with expansion. The steady demand for new circuits has rapidly diminished the availability of frequency spectrum for additional channels. A major effort has been directed toward developing a new signaling and detection technique for the transmission of binary information which has much greater efficiency in regard to power and spectrum utilization when compared to standard signaling practices. The signaling and detection technique accomplishing this objective incorporates kinematic filtering and signal multiplexing and accordingly has been named "Kineplex."

Historically, it always has been more economical to spend time encoding and decoding a message in order to make it suitable for transmission by way of a simpler transmission medium. An example is the smoke signal which can be transmitted with the most primitive of apparatus. When communication between two individuals is required, speech communication has always been greatly desired because of the speed with which ideas can be transmitted and responses given back. However, telegraph type communication, in which the sender causes a receiving device to print a message, is often even more desirable because it results in a written record of the complete message.

In many cases, messages must be integrated and evaluated to learn their full significance. Where the number of such messages is relatively small, the integration and evaluation can be done manually, as on a plotting board, and voice communications can be used. However, when the volume of such messages becomes large, machine methods are required and telegraph data links, with connections to a computer, are usually required. Since such record-type communication systems can frequently require much more capacity than previously has been in existence, Kineplex was designed with far greater message carrying capacity in a given bandwidth than systems previously available.

2. CODES USED FOR DATA TRANSMISSION

Coding is essential to the successful transfer of information by binary elements. The variety of codes in existence today is as varied as their methods of transmission. Each has a purposeful existence because of universal acceptance or certain desirable qualities that fulfill the demands of their users. Two important factors exist in any code, its efficiency in conveying information, and the freedom from errors introduced by the transmission medium. Actually they conflict with each other so that a given code is not necessarily the most suitable code for use in all circumstances. Similarly the methods of transmission cannot be separated from the code entirely.

Two familiar codes are the dot-dash Morse code, and the mark-space five-element teletypewriter code. The Morse code, which is important historically, suffers in efficiency and general adaptability to modern automatic transmission methods. The five-element start-stop code has proved successful in land line applications and has been generally adapted for radio-telegraph links. This highly efficient teletypewriter code is physically composed of five information elements, plus a start pulse and a stop pulse. The two conditions may be represented by a signal, or current flow, for mark and a no signal, or no current, for space. The stop pulse is made 1.42 as long as the other equal pulses so that the teletypewriter machine which is sending, and the teletypewriter machine which is receiving, will both have time to stop. This permits the two machines to start each new letter or character simultaneously. This action avoids the necessity of having both machines running at exactly the same speed and gives rise to the name "start-stop" telegraph. The addition of the start and stop elements reduces the efficiency by about 30%.

This code, which is often referred to as a 7.42 code, has a possibility of 32 permutations. A minimum of 5 bits is required to encode the entire alphabet so this allows additional functions such as, letters or figures, line feed, carriage return, etc. When reduced to equal bit lengths to utilize the benefits of synchronous predicted-wave techniques, the code is referred to as a 7.0 code. The processing of digital binary codes is essentially the same as the telegraph code since both feature two-condition on-off characteristics.

3. TELEGRAPH SIGNALING

The methods of sending codes are varied considerably. Some methods, while superseded by systems yielding more capacity and reliability, retain their position because of the simplicity of transmitting and receiving equipment. In its earliest form, telegraph consisted of a single wire circuit as shown in figure 12-1. At one end of the circuit, there was a battery and a key and at the other end a buzzer with a ground return. Closing of the key would cause the buzzer to operate so that with some kind of code, intelligence could be transmitted. This (on-off) circuit is still basic to all telegraphy although it does not appear in such simple form. Analogous in radio transmission was the on-off keying of a transmitter. With either method of transmission, the signals suffered deterioration. Filtering at the transmitter or receiver was limited since accurate message reproduction entailed the recovery of the coded form of the transmitted signal.

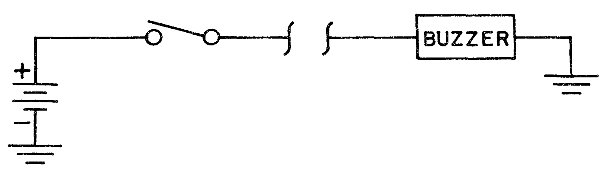

Figure 12-1. Basic Telegraph Circuit

4. FREQUENCY SHIFT KEYING

It soon became apparent that the function of the receive equipment was to reproduce the information content of the transmitted message and not the Fourier components of the transmitted signals. The encoding of marks and spaces on higher and lower carrier frequencies (FSK) enabled a uniform power output to be transmitted. The detector had only to decide which of two frequencies was the larger, which provides an advantage where signal distortion and a high noise level are present. The reduction of bandwidth by additional filtering substantially reduces extraneous noise at the detector.

Although frequency shift keying is widely accepted and used, there is still much to be desired in the way of reliability and efficiency of existing circuits. A classic equation shows that:

$C = BW \log_2 \frac{S + N}{N}$ where BW is the bandwidth in cycles per second; S is the signal power and N the noise power.

With a signal-to-noise ratio of unity and the minimum spacing consistent with practical experience, 120 cycles, very little over a 60 wpm teletypewriter rate or 45 bits per second, could be expected while the theoretical capacity was 120 bits per second. It is to be noted that the information capacity is a much stronger function of bandwidth than the rms signal-to-noise ratio. But since increasing the bandwidth decreases the S/N ratio without an increase in radiated power, there is not a great deal to be gained in terms of information capacity by increasing the bandwidth beyond the value where the signal-to-noise ratio falls below unit. Moreover, any improvements gained by widening the transmission band are limited to conditions where favorable signal conditions prevail.

5. PREDICTED WAVE SIGNALING

Aware of the implications surrounding the increased bandwidth requirements, Collins Radio Company directed their attention to improved detection whereby the bandwidth can be narrowed and an increase in signal-to-noise ratio attained. To determine the optimum means of measuring the polar amplitude of a wave in the presence of wide band random noise, assume the transmission of a rectangular pulse as shown in figure 12-2A. The transmitted pulse is attenuated by the transmission path and contaminated by additive noise to give a received signal as shown in figure 12-2B. From this it is desirable to obtain a best estimate of the polar amplitude of the transmitted pulse (figure 12-2A) which carries the transmitted information. The equipment used for this measurement consists of a gate controlling the period during which the received signal-plus-noise is accumulated in an integrating device. The problem is when to open and close the gate to obtain an optimum measurement of amplitude. A wide gate, as shown in figure 12-2C, will result in the accumulation of the entire integrated value of the signal plus the accumulation of noise when there is no signal. Similarly a narrow gate, as shown in figure 12-2D, will permit the accumulation of only

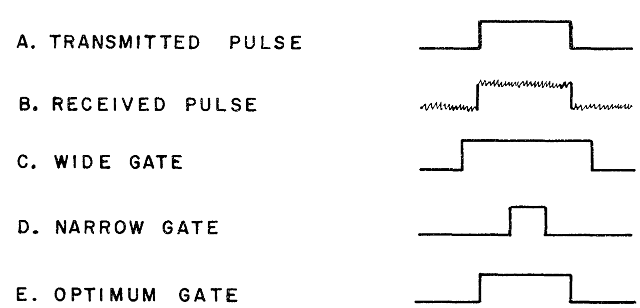

Figure 12-2. Predicted Wave Detection

a fraction of the integrated signal value and would integrate the noise over a corresponding fraction of the pulse duration. The narrow gate is a poorer choice than the wider gate because the signal contribution increases linearly with the duration of the pulse, while the noise contribution only increases by a factor of the square root (assuming the noise is to be wide band noise). This latter fact is due to the lack of coherence between the noise pulses which are being added to the integration device. The ideal condition is to open the gate at the start of a pulse and close it immediately when the pulse ends. Optimum results are obtained then by multiplying the received signal (plus noise) by a pulse of the shape of the expected signal pulse and integrating and placing the multiplying function (or weighting function) coincident in time with the expected signal pulse. It is desirable to have available locally at the receiver all the information concerning the incoming signal except the information it is designated to convey. This is called predicted wave signaling.

6. KINEPLEX

Using the basic principles of predicted wave signaling and incorporating FSK or phase shift keying, the Kineplex Data System through evaluation tests has proven conclusively the theoretical predictions of the benefits to be gained. The performance of the system is extended by the inclusion of near absolute frequency stability so that the bandwidth may be reduced to a minimum. Predicted wave signaling utilizes: synchronization so that the detector is given information on the time of arrival of the start and finish of each data pulse; gated very high Q integrating circuits, employing mechanical resonators, the response of which matches perfectly the energy distribution of the transmitted pulse; sampling of the detector outputs at the end of each pulse so that full integration of the received pulse may be utilized; encoding of binary information so that the theoretical minimum bandwidth for a given binary signaling rate may be approached.

An investigation of some of the unique features, such as phase shift keying of two bits of information on a single tone, the use of electromechanical resonators as high Q optimum weighting functions, the recovery of phase information from received signals, and the sequence of timing to provide synchronous gating action will advance the understanding of a basic system.

In order to avoid the complexity of timing circuits and to provide equal synchronous bit lengths, a code converter retimes nonsynchronous information, such as teletypewriter information. This standard code has a stop pulse which is longer than the other pulses by a factor of 1.42.

Since there are fractional accumulations of stop pulses (0.42 time elements) caused by the shortening

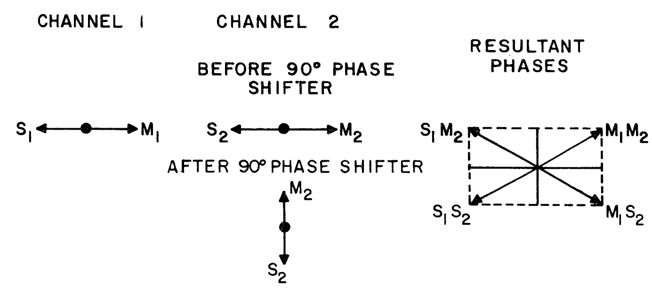

Figure 12-3. Phase Modulation Vector Diagrams

of the stop pulse, provision has been made to halt the readout of synchronous output to allow the input to catch up. This is accomplished by inserting an extra stop pulse whenever the output tends to overrun the input by a full element. While the extra stop pulse is being inserted, incoming information is read into a storage circuit. Readout information alternates between direct and stored material depending upon the time differential of the input and output. The outputs of all code converters are time synchronized. A major aspect of the Kineplex Data System is the phase modulation of two information channels upon a single tone. The resultant vector also functions as a phase reference for the succeeding element. Refer to figure 12-3. Let a mark for channel 1 be represented as an in-phase condition M1, and a space as an out-of-phase condition as S1. A second channel of information is phase shifted so that a 90° vector in reference to the first mark is assigned to the second mark, M2, and an out-of-phase condition to S2. When these two information sources are combined together, four possible phase positions are assumed by the resultant vector:

M1 M2 —— 45°
S1 M2 —— 135°
S1 S2 —— 225°
M1 S2 —— 315°

The joint action of the tone generator and keyed filter pair produces tones which are phase modulated in accordance with mark-space transitions as previously described. Each resultant becomes the phase reference for the next element.

A phase detector could easily separate the resultant vectors into components with the resultant of the preceding element as a phase reference. Assume the first element transmitted at T1, shown in table 12-1, consisted of mark information in each of the two channels. The resultant is represented at 45°. At time T2, two spaces will be transmitted from each channel. The

TABLE 12-1. KINEPLEX PHASE RELATIONSHIPS

TIME	ASSUMED SIGNALS	PHASE	STANDARD VECTOR	PREVIOUS BIT VECTOR	TRANSMITTED VECTOR
T1	M_1 M_2	45°	0	0°	45°
T2	S_1 S_2	225°	0	45°	270°
T3	S_1 M_2	135°	0	270°	45°
T4	M_1 S_2	315°	0	45°	0°

resultant vector will be at 225°. Since the previous 45° vector is a source of reference, this 225° resultant vector is transmitted at 270° as shown. At time T3, a space is transmitted on the first channel and a mark on the second channel. The resultant when combined together is a vector at 135°. However, when the previous vector at 270° is used as a reference, the phase transmitted becomes once again 45°. In like manner each new resultant is referenced to the previous resultant. At the receive end is present a reference vector without actually transmitting a separate phase reference. This phase comparison, element by element, so that the reference phase is maintained at the receiver integrator is a criterian of optimum predicted wave detection.

An electromechanical resonator is a device that will accept electrical energy, convert it to mechanical energy, and produce electrical energy at its output. Full utilization of these devices is made in their use as phase storage, filtering, and signal integration. Very high Q, when supplemented by a positive feedback, assumes an infinite Q. When driven by a signal (incoming), they provide a linear integration in time of the received signal thereby providing, when gated in synchronization with the incoming signal, a perfect weighting function. Their narrow bandwidth characteristics are equally acceptable as selective filters.

Two resonators with associated amplifiers are used in each keyed filter pair. A quench pulse gates a negative feedback around the resonator removing any residual information. An incoming signal is gated

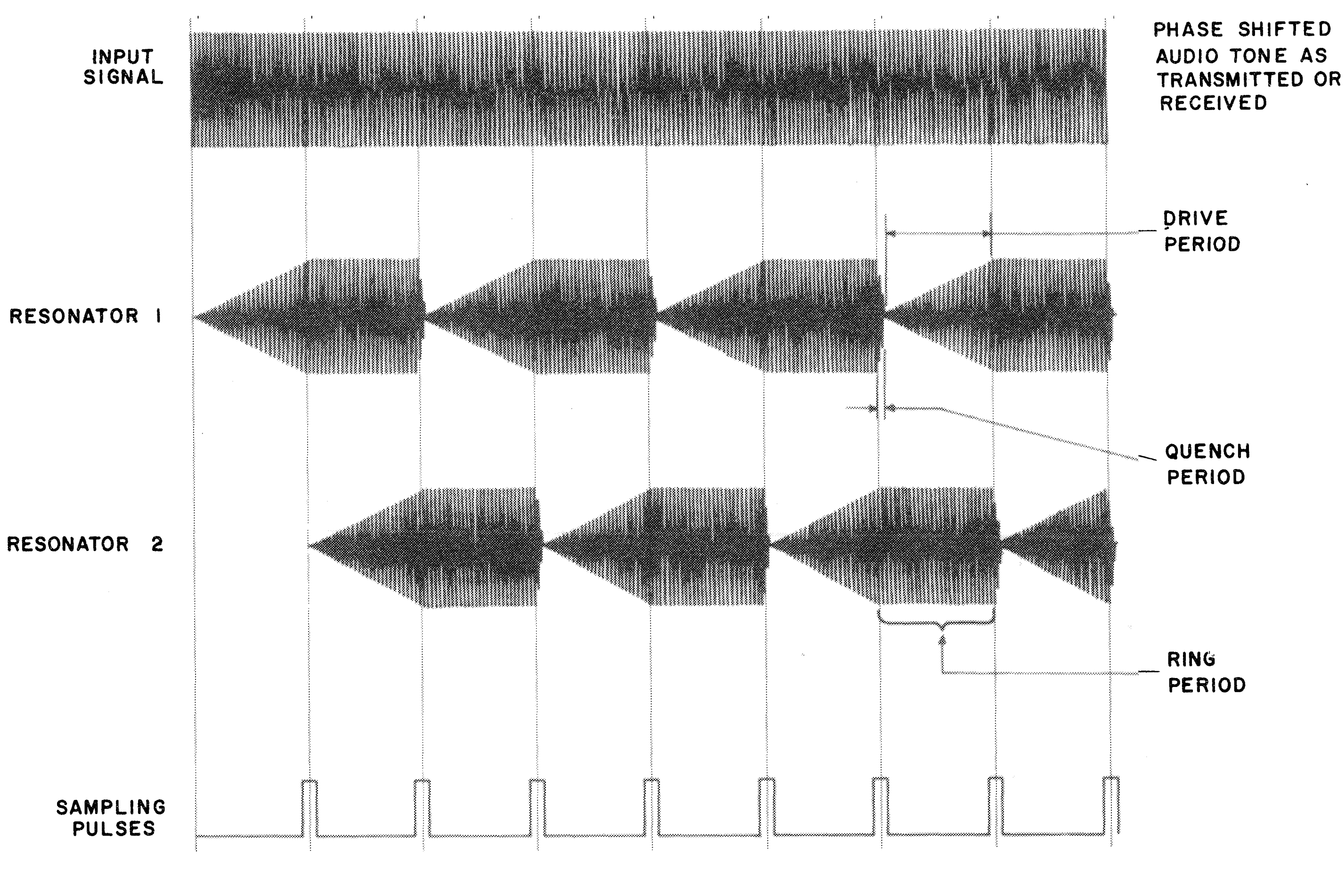

Figure 12-4. Keyed Filter Pair, Quench-Drive-Ring Sequence

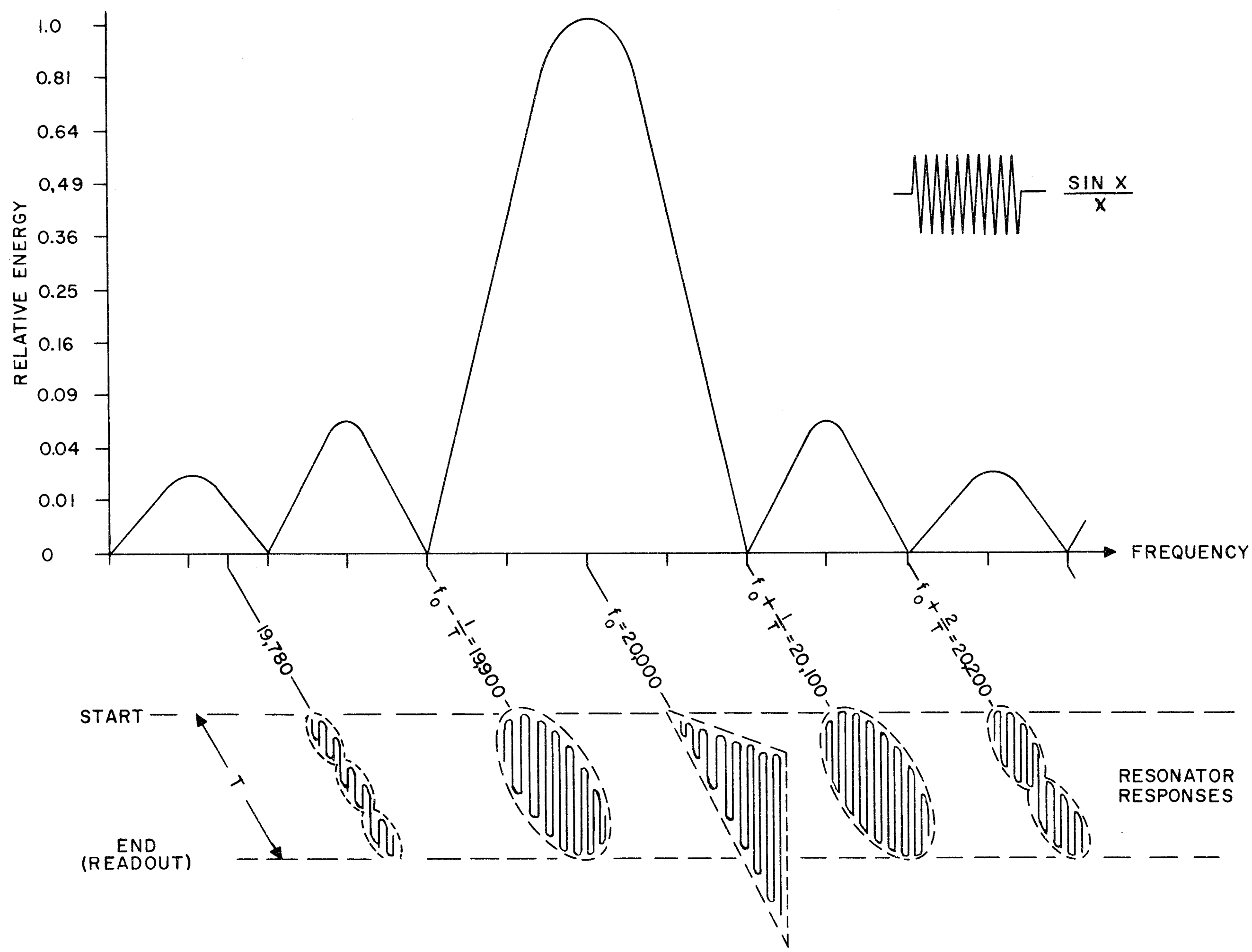

Figure 12-5. Energy Versus Frequency Distribution of Transmitted Pulses and Infinite Q Resonator Responses

into the input by drive pulses. At the end of the drive pulse, when the signal has built to a maximum, the energy is allowed to ring at the phase and frequency of the driving signal for a time equal to a quench and drive pulse. The two sections of a keyed filter pair work in alternation with one another. While one is ringing, the other is being quenched and driven, then while the first is quenched and driven, the second is ringing. Refer to figure 12-4. The resonator section that is ringing contains the frequency and phase of a driving signal that was present a time interval before. The resonator section that is being driven contains the frequency and phase information of the driving pulse that is now present. At the output of the two sections, we have information about a signal that is now being produced (or received) and equal information concerning the preceding signal. In this manner a reference is provided which is identical to that required.

A pulsed audio sine wave may be expressed by the function $\frac{\sin X}{X}$. Perfect integration of such a pulsed wave by a weighing function, such as that provided by a mechanical resonator, allows the recovery of the energy content of the sin X/X function. A relationship exists between the length of the pulse T and 1/T that orthogonal or null points occur at the resonant frequency plus or minus 1/T. This is illustrated in figure 12-5 which is an illustration of energy versus frequency distribution of transmitted pulses and infinite Q resonator responses at 100 wpm operation. The upper portion shows the energy-frequency relationship of the transmitted pulse for a resonator whose center frequency is 20,000 cycles. The lower section shows sketches of the oscillograph waveforms of the resonator response obtained for various input frequencies. In this instance the pulse length is 10 milliseconds, and the bandwidth 1/T is equal to 100 cycles. It will be observed that the resonator amplitude builds up linearly to the sampling time at the end of the pulse. A signal 100 cycles removed from the center frequency of the resonator at 19,900 or 20,100 cycles will result in a resonator response which has a linear buildup close to start but which reaches a maximum amplitude halfway through the duration of the pulse and returns to zero at the end of the pulse period. If two resonators were available with one cut to 20,000 cycles and the other cut to 20,100 cycles, both resonators could be fed at once with both frequencies, but neither resonator would respond to the resonant frequency of the other. Similar null responses occur at every hundred-cycle increment of frequency.

As shown at 20,000 cps plus 2/T, the number of loops in the response curve is equal to the value n in the expression $n\triangle f$, where $\triangle f$ is the difference between the resonator frequency and the frequency of the first null point. Tone spacings or the value of $\triangle f$ can be varied at will. In a practical system, it is possible to choose frequency spacings over a wide range merely by altering the timing signals.

In the presence of severe noise and multipath distortion, the number of information channels can be reduced. The increase in the power-per-tone attained can be discerned through the four fold increase in power-per-tone realized by reducing the number of channels by one half.

It is permissible to restrict the bandwidth emission of the transmitter by filtering to eliminate the side energy beyond approximately plus or minus 3/T. The amount of energy contained in the signal beyond this third orthogonality is so small that it can be filtered out with small effect on operation. This feature is an important point in limiting intersystem interference.

The timing functions of the Kineplex Data System integrates the operations of the individual units. Refer to figure 12-6 for the relation of pulses used throughout the system. These values apply for 60 wpm operation. The time relations of the Q and D pulses are altered at other repetition rates.

All pulses are derived by time bases which are supplied driving pulses by a standard operating at a $16F_1$ rate, where F_1 is the element repetition rate. Pulses marked with a prime are identical to their counterparts except for being 180° out of phase. The F_1 pulses provide precise timing for the information data. The F_2 pulses gate resonators alternately for signal storage during tone generation and detection. The synchronizing pulse is also gated at an F_2 rate. The sampling pulse determines the sampling time of the phase-detected signal, while the F_2D pulse forms the basis of comparison signals in the synchronizing unit. A Q pulse and a D pulse occupy the same time as a code element. The Q pulses open a gate to quench or wipe out information which has been stored in a resonator before the D pulses open gates to provide resonator build up.

Note how the Q_1 and Q_2 and the D_1 and D_2 pulses alternate in time to produce the alternate quench-drive-ring sequence of the keyed filter pairs. Table 12-2 contains the repetition rates of three standard transmission rates and the associated element rate and quench and drive pulse lengths.

Figure 12-7 represents a typical frequency tone allocation of a basic system. Each of the twenty tones with frequency spacing from 605 cps to 2695 cps are allotted two channels of information. The 110 cps spacing between tones provides for readout on the second orthogonality for 60 wpm operation while at 75 wpm and 100 wpm rates, the readout occurs at the first null point. If 500 cycles protection against unfavorable delay at band extremities is allowed, a total bandwidth of 3300 cycles is employed, which permits the positioning of a synchronizing tone at 2915 cps. Actual band use suggests the use of 55 cycles for two channels which compares quite favorably with the minimum of 120 cycles per channel with frequency-shift keying.

TABLE 12-2. TIMING FREQUENCIES AND PULSE WIDTHS

TELETYPE RATE	ELEMENT RATE	$16F_1$ cps	F_1 cps	F_2 cps	QUENCH (ms)	DRIVE (ms)
60 wpm	45 bits/sec	727.27	45.45	22.72	3.8	18.2
75 wpm	55 bits/sec	888.8	55.55	27.77	8.5	9.1
100 wpm	75 bits/sec	1185.2	74.2	37.1	4.4	9.1

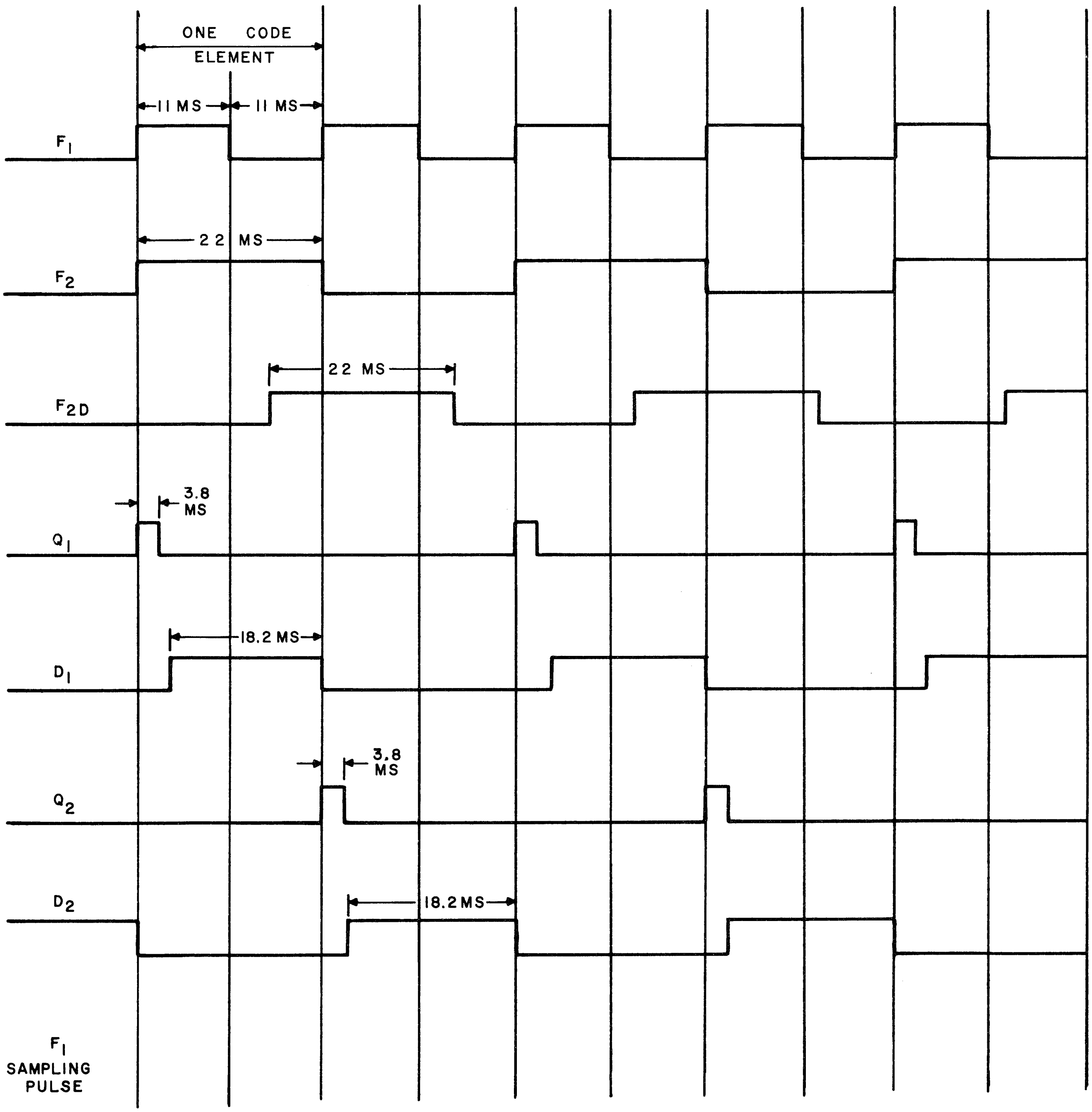

Figure 12-6. Timing Pulse Relation

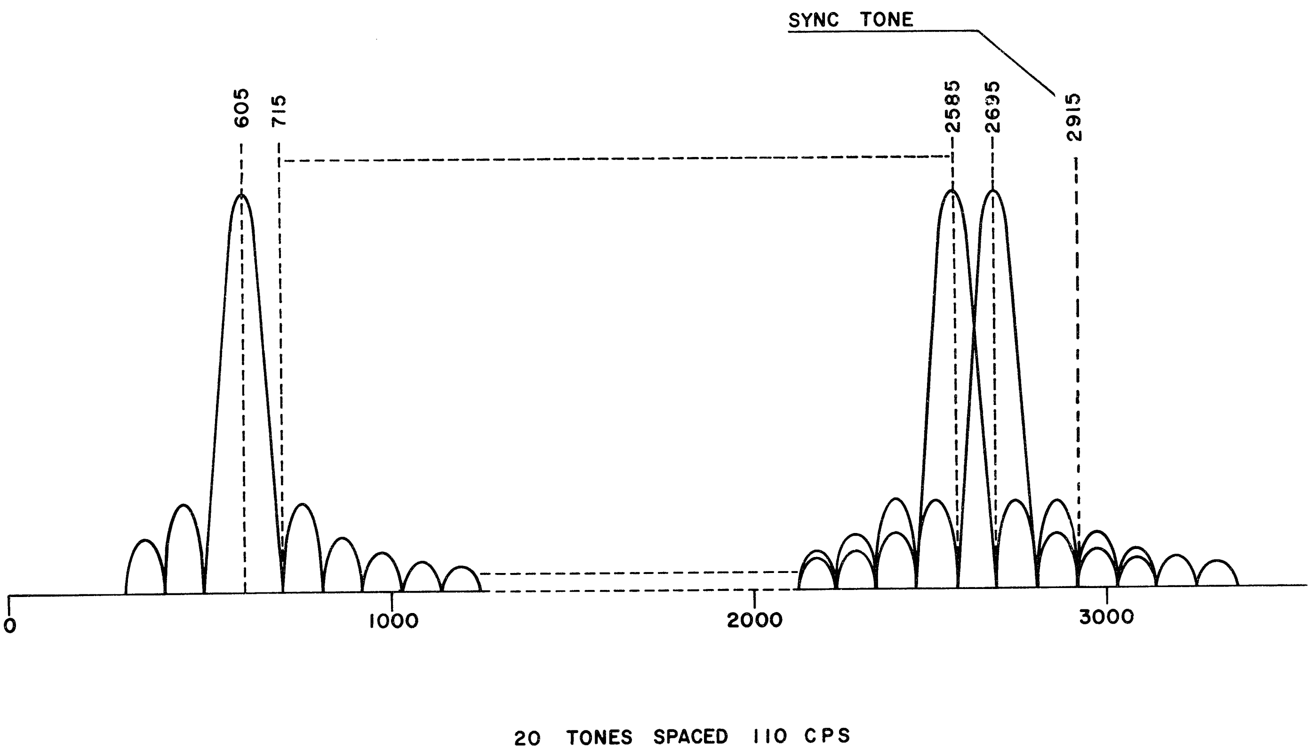

Figure 12-7. Channel Allocation at 100 WPM

With the aid of block diagram 12-8 the previous discussion of Kineplex theory will be applied to a basic system. Only two of the possible forty inputs are shown for convenience. Assume that a 60 ma d-c teletype signal is received from a transmitter-distributor or other teletypewriter device by each code converter. Since the transmitter distributors are not synchronized in any way, mark-space transitions may occur at any time on any of the channel inputs. The outputs of the code converters are gated so that all outputs occur simultaneously. Also the long stop pulse is shortened to the same length as the other pulses.

The outputs of two code converters modulate a single tone through the action of the phase shift generators and the keyed filter pairs. The output from the keyed filter pair is gated alternately to a dual-balanced modulator by an F_2 timing pulse which phase modulates the tone in accordance with the signals received from the code converters. One of the balanced modulators includes a 90° phase-shifting network in its output so that the phase-modulated tone from both balanced modulators can be combined as shown in table 12-1 and amplified for transmission. The same output is also fed into the input of the keyed filter pair to one of the two resonators for storage as a zero phase reference. The output of the tone generator is shifted in phase for each new element. Twenty such outputs are combined together with the synchronizing tone for transmission. Since all elements are occurring synchronously, the output is constant in amplitude thus assuring full utilization of the transmission medium.

Binary information from two sources modulating a single audio tone by phase-shifting techniques is not necessarily confined to the method described in the phase shift generator. The desired phase is recognized to be integral multiples of 45° of the reference phase. A 45° vector represents 1/8 of an F_1 cycle so that eight increments represent any of the possible phase conditions. A selection of any of these conditions would constitute a phase-shifted vector. The retention of this vector product provides the reference point for the next vector selection. The combination is a reconstructed sine wave gated at an F_1 rate containing the two channels of information.

The distributing unit in the receive section is an amplifier with a tone-operated level control which amplifies the incoming tone for application to the keyed filter pairs and synchronization unit. All other tones except the synchronizing tones are filtered at the synchronization unit input. The filtered output of a detector supplies an automatic level adjustment voltage

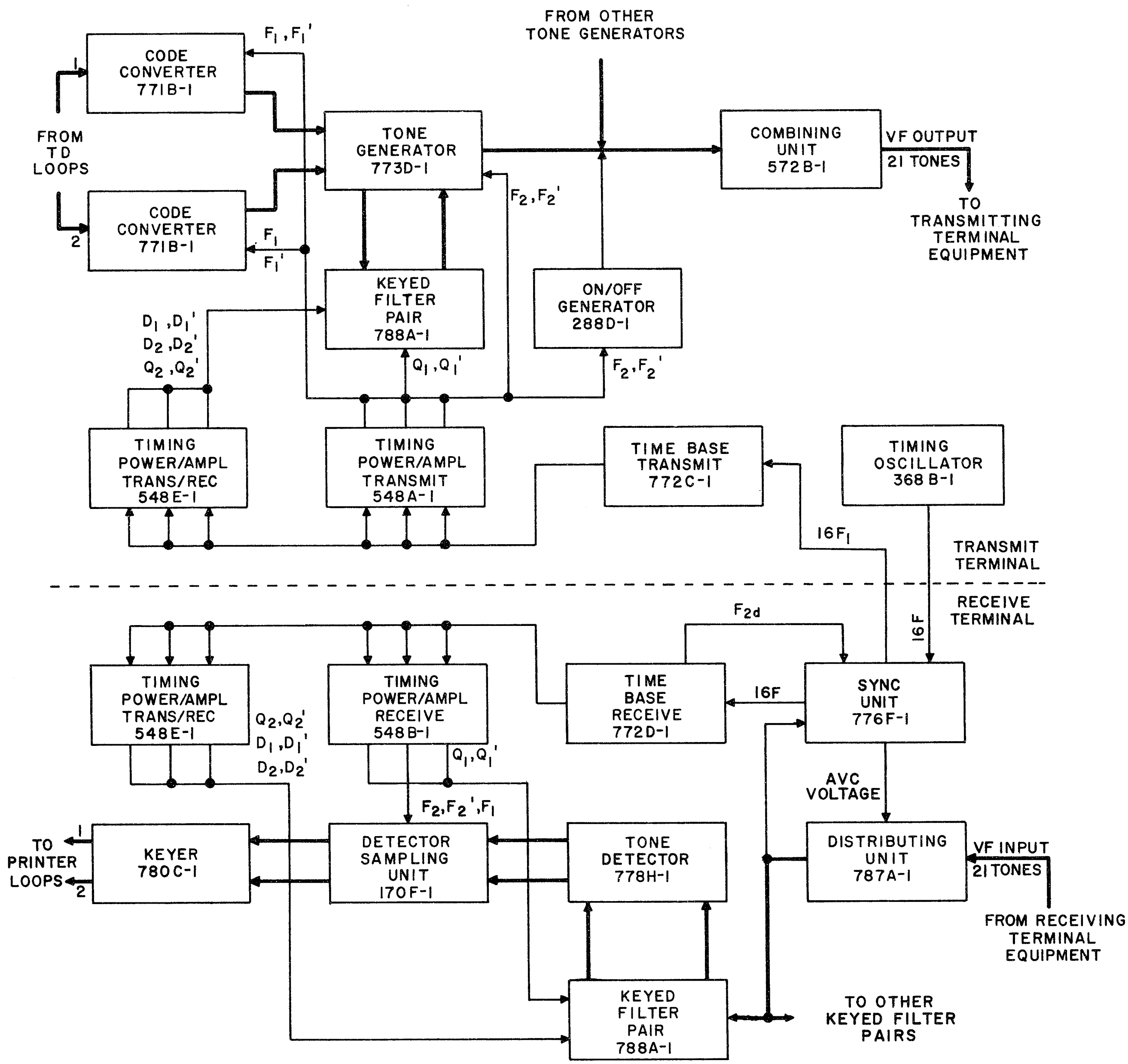

Figure 12-8. Kineplex (Frequency Division) Block Diagram

back to the distributing unit. The synchronizing tone, which is at an F_2 rate, is supplied to a phase detector. An F_2D pulse, derived from the receiver time base but delayed in respect to the other pulses to compensate for the delay of the input F_2 pulse in passing through the filter, is supplied to the same phase detector. The difference, if any, drives a magnetic amplifier whose windings are interconnected with a servomotor. This servo is geared to a resolver. The resolver shifts the timing frequency phase as necessary to effect synchronization with the incoming signal.

The third function of the synchronizing tone, automatic frequency control, may be accomplished with auxiliary equipment. A controlled modulator operates in conjunction with an error detector and a controlled oscillator in the following manner. An incoming signal is fed into a modulator where it is mixed with a variable-frequency signal containing translation error information from the controlled oscillator and converted into sidebands in the 250 kc frequency range. The resultant upper sideband is selected by a mechanical filter and reconverted to the audio range by heterodyning with a fixed 250 kc signal. This resultant a-f signal is applied to the error detector, which is a frequency discriminator operating with a 2915 cps electromechanical resonator as a reference, to produce an error voltage proportional to the deviation from the incoming synchronization tone. There the error voltage thus produced is applied to the controlled oscillator to produce the variable-frequency signal containing the error information referred to above, and produces the compensatory correction for any frequency translation to insure that the received channel frequencies correspond exactly to the transmitted frequencies.

The keyed filter pairs receive all the tones from the distributing unit, each keyed filter pair accepting a tone to which its resonators are tuned. One element is used to drive one of the resonators in the keyed filter pair and then stored in that resonator during the second element, while the second element is being used to drive the second resonator. At the end of the second bit element, a phase comparison is made, and the resonator which was ringing is quenched and driven with the third element while the other resonator is allowed to ring during the third element period. This preserves the signal received during the second element for comparison at the end of the third element.

By passing signals through a direct and a 90° phase-shifted path in the tone detectors, two outputs are obtained for the two channels which were modulated on the single tone. The output is now represented by differences in voltage levels for the two conditions of mark and space.

In addition to amplifying and wave shaping, the detector sampler determines whether the signal is mark or space during a short interval at the end of each drive period.

The keyer serves to convert the voltage excursions from the detector sampler unit to definite mark and space transmissions similar in all respects to the input information at the distant end. At the same time, the power level is increased to drive teletypewriter printers or whatever kind of output device is used.

Adequate protection against phase and frequency-shift errors introduced through motion of stations in respect to one another is afforded by auxiliary equipment.

Thus far the discussion has been concerned only with frequency multiplexing. Time division multiplexing is accomplished by reducing the length of the transmitted pulses. Time division multiplexing offers the advantage of long pulses where severe delay distortion is present. However, for high speed transmission pulses, as short as 1 millisecond may also be accommodated if their use is dictated. The increase in bits per second results in the increase of bandwidth or a reduction in the signal-to-noise ratio. Multiple tones may be utilized to increase the capacity for a specific application.

Either frequency diversity or space diversity may be applied to the basic Kineplex system. With frequency diversity, the added feature is accomplished with the reduction of the number of individual channels that may be transmitted. Space diversity requires only the duplication of keyed filter pairs and tone detectors in the receive section. The detector sampling unit has the additional function of combining inputs when diversity reception is used. Although the previous discussion has centered around the transmission of teletypewriter material, the inputs and outputs may be equipped with converters to process any kind of series or parallel binary data, synchronous or nonsynchronous. Such applications would include magnetic tape, punched cards, or other types of storage devices. The inputs would not be confined to one type for they may be conveniently intermixed at will.

Kineplex may be readily integrated into systems utilizing frequency division or time division multiplex. Optimum signal transfer characteristics permit a high capacity, flexible system for the transmission of binary data over open wire lines, cable facilities, carrier telephone channel circuits, multiplex channel circuits, and microwave system basebands. When equipped with appropriate conversion units, the Kineplex Data System will accept and transmit binary information for various services, such as teletypewriter, business machine, telemetering, supervisory control, and facsimile. Analog information transfer is best accomplished by conversion to digital information for transmission as binary data. This establishes a reliable communication system in the presence of noise that needs only to establish the quality of the signal rather than the quantity. With proper devices, almost any type of information could be transmitted including voice.

The basic system is readily adapted to accommodate the various input and output levels and impedances encountered in different types of transmission. Similarly power requirements are such that simple conversion units makes any available power source acceptable. Channels may be easily added or deleted from a basic system to adapt to the number of information channels required. Mobile and portable stations are as feasible as fixed stations since the over-all design incorporates minimum power and weight requisites consistent with reliability.