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DIGITAL COMMUNICATIONS/INTEROPERABILITY TECHNIQUES

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ABSTRACT

Contemporary long haul and tactical digital communications systems employ a variety of digital transmission formats. New code conversion equipment has been developed which digitally converts between different pulse code modulation formats and between pulse code modulation formats and delta modulation formats. Conversion is accomplished on wideband trunks without restoring the signal-to-analog baseband. The digital code conversion techniques may be applied to conventional multiplex equipment, Army tactical communications

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equipment, strategic communications systems, and TRI-TAC equipment. Conversions are accomplished with respect to data format, data rate, supervision and signaling formats and electrical interface. The code conversion techniques promise significant advantages in utility, performance and cost. System applications of the digital code converter are compared with other alternatives. Signal-to-quantization noise ratio measurements on channels which have undergone digital code conversion are presented.

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DIGITAL COMMUNICATIONS INTEROPERABILITY TECHNIQUES

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ABSTRACT

Contemporary long haul and tactical digital communications systems employ a variety of digital transmission formats. New code conversion equipment has been developed which digitally converts between different pulse code modulation formats and between pulse code modulation formats and delta modulation formats. Conversion is accomplished on wideband trunks without restoring the signal to analog baseband. The digital code conversion techniques may be applied to conventional multiplex equipment, Army tactical communications equipment, strategic communications systems, and TRI-TAC equipment. Conversions are accomplished with respect to data format, data rate, supervision and signaling formats and electrical interface. The code conversion techniques promise significant advantages in utility, performance and cost. System applications of the digital code converter are compared with other alternatives. Signal-to-quantization noise ratio measurements on channels which have undergone digital code conversion are presented.

1.0 INTRODUCTION

A wide variety of digital communications systems are in use today. Each employs a unique transmission format. The choice of a particular digital communications technique is dictated by: technology at the time of development, required trunking capacity, user performance requirements, anticipated number of multiple hops, the electrical characteristics of associated equipment, and bandwidth of the transmission medium. Each of these is described briefly in the following subparagraphs to emphasize those characteristics which must be considered for interoperability.

<u>Technology</u> - Technology at the time of development determines the maximum capacity of the system and the type of format that can be used. One example is compression. To achieve the necessary dynamic range, while maintaining a good signal to noise ratio, it is necessary to compress the signal amplitude range. Diode compressors were used in some of the earlier equipment. Matching of these compressors, which are highly dependent upon the individual diode devices, is very difficult. More recent

equipments utilize digital compression techniques. The digital technique is easily matched from unit to unit and provides much better input-output linearity.

Trunking Capacity - Trunking capacity, or the number of channels handled by any one multiplexer unit, is dictated by system constraints. Commercially, a 24 channel grouping is most often encountered. In tactical military systems, channel grouping of 3, 6, 12, and 24 may be found. In newer Continuously Variable Slope Delta Modulation (CVSD) systems, groupings of 4, 7, 8, 15, and 16 are encountered.

Performance - Performance is measured in terms of quality and intelligibility and is dictated by user requirements. In a Pulse Code Modulation (PCM) system, the quality and intelligibility is determined by the number of bits per sample used to encode the signal. Frequently, a call must be switched through several substations, being converted back to analog, switched, then reconverted to PCM. In these tandem connections, the high quality provided by 8-bit PCM is required. Quantization noise is increased 3 dB each time the number of conversions doubles (3 dB for two hops, 6 dB for four hops, 9 dB for eight hops, etc.)

Electrical Characteristics - The electrical characteristics of associated equipment dictate the terminal characteristics of most digital multiplex equipment. Multiplex equipment typically provides an analog or a digital interface which is directly compatible with the switchboards and telephones in use. Characteristics, such as signal level, input/output impedance, and frequency response are dictated by the system in which the multiplex equipment will be used. A four-wire interface is normally required to interface with multiplex equipment. In analog systems where four-wire telephones are not used, a hybrid transformer must be provided as in the Telephone Signal Converter, CV-1548, used in tactical military communications. This converter provides for the two-wire to four-wire conversion and incorporates circuits for the detection of different types of supervision and signaling tones. This brings out one of the primary advantages of digital code conversion. By operating on a multiplexed wideband trunk, the number of electrical interfaces is minimized and complex functions such as signaling conversion can be accomplished more efficiently.

Bandwidth - Available bandwidth of the transmission medium directly affects the bit rate which can be passed through it. The transmission link may consist of cable and/or radio paths.

2.0 DIGITAL COMMUNICATIONS FORMATS

A digital communications system, as the term is used herein, refers to those systems which formats the users' analog signals into a binary (two state) form. The two basic formats to be discussed are Pulse Code Modulation (PCM) and Continuously Variable Slope Delta Modulation (CVSD). PCM is already in wide use and CVSD will be, with the coming of the TRI-TAC equipment.

PCM and CVSD systems digitize analog user waveforms for efficient transmission and restoration. The purpose for digitization includes security, economy and quality.

Pulse Code Modulation (PCM)

In both PCM and CVSD systems, the user waveform must be sampled at discrete intervals in time. According to Nyquist, the signal must be sampled at least at twice its bandwidth. In PCM telecommunications systems, a sampling rate of 8 kHz is used almost universally since the nominal bandwidth is 4 kHz. As shown in Figure 1, once the waveform is sampled, it appears as multiple samples of varying amplitude. The waveform at this point is pulse amplitude modulation (PAM). When this waveform is filtered it will accurately reproduce the original waveform. Each PAM sample is then digitized to a 6, 7, or 8 bit code word for transmission. A 6-bit code word is shown in Figure 1.

An instantaneous compressor is used in PCM systems for improvement of the dynamic range. Figure 2 shows a compression curve. The abscissa indicates the analog voltage of the encoded signal at any instant in time. The ordinate indicates the integer that would be encoded for transmission. The curve shown in Figure 2 represents the 3 segment compressor used in 6-bit tactical PCM systems. The integer is encoded into a binary number made up of three fields. The first, or lefthand field contains the sign bit, the second field contains one bit indicating the particular compressor segment, and the last field contains a 4-bit representation of the PAM sample. Note that for 6-bit PCM there are only two segments in each of the two quadrants. The curve in Figure 2 approximates a logarithmic compressor with μ=100 (μ=compression factor). There are numerous PCM formats including the following:

Multiplexer Type	PCM Word Size	Compression
TD-352 TD-660	6-Bit, approx μ=100	3 segment compressor
D1 T1	7-Bit, μ=100	Diode Compression
D2 T2 TD-968 TD-1192	8-Bit, μ=255	15 segment compressor

Multiplexer Type	PCM Word Size	Compression
CCITT	7-Bit	13 segment
European A-Law	8-Bit	13 segment

Continuously Variable Slope Delta Modulation (CVSD)

Delta Modulation (DM) uses a single bit form of quantization. The sampling rate must be correspondingly higher to produce a good reproduction of the original waveform. 50 kbps delta modulation systems operate with reasonable quality. Continuously variable slope delta modulators provide acceptable quality at rates as low as 16 kbps. Figure 3 shows a functional block diagram of a CVSD modulator.

In a simple delta modulator, the analog voltage which is to be encoded is applied to one input of a comparator. This voltage is compared to a voltage developed on a simple RC integrator applied to the other input of the comparator. The output of the comparator is sampled, typically at the channel rate. If the input voltage is greater than that developed on the integrator, then a positive pulse is generated. This pulse is fed back to increase the voltage on the integrator. If the input voltage is less than that on the integrator then a negative pulse is generated, decreasing the voltage on the integrator. Only the sense, positive or negative, of the pulse is transmitted.

In this simple delta modulation system, the voltage developed on the integrator will track that of the applied input voltage. The decoder is the equivalent of the integrator which is used in the feedback loop of the encoder. The advantages of delta modulation are simplicity and single bit quantization. The single bit format eliminates the synchronization which is required to extract multiple bit code words from a serially transmitted bit stream.

As shown in the bottom of Figure 3, a simple delta modulation may be unable to track a high amplitude, rapidly varying, analog waveform, thus causing slope overload. To track such a waveform, the step size of the delta modulator must be increased; however, this has the undesirable effect of increasing quantization noise. A CVSD system, as its name implies, has a continuously variable step size. The step size is varied in proportion to the signal rate of change. This results in a relatively constant ratio of signal-to-quantization noise. Slope overload detection logic and a phonemic filter implement the variable slope algorithm.

The slope overload detector identifies the occurrence of three or more bits of the same sense (3 ones or 3 zeros) from the comparator. This indicates that the comparator is attempting to drive continuously in one direction, and that the step size should be increased. If this occurs, the output voltage from the slope overload detector is increased. The voltage change will be integrated by the phonemic filter which has a 6.4 millisecond RC time constant.

The voltage out of the phonemic filter is applied directly to the input of the loop integrator. The loop integrator, a single pole RC filter, will charge exponentially with a time constant of 1 millisecond. The CVSD algorithm described here has been selected as the TRI-TAC standard.

Multiplexing

The digitized representation of the user signal is multiplexed to achieve economy of transmission. Signals are multiplexed by sequentially picking up samples from multiple channels. In PCM systems this is typically accomplished when the waveform is in its PAM format. To achieve an 8 kbps sampling rate on each channel, a twelve channel system must be sampled at 96 kHz. The output of the sampler is then fed to a single analog to digital converter; thus, one analog to digital converter can handle multiple channels. At the receiving end, a demultiplexer redistributes the multiplex samples to their respective channels. A synchronizer is required to control the distribution process such that each channel's data is ultimately distributed back to the correct channel. Synchronization is typically accomplished by transmitting a known pattern in a dedicated slot within the multiplexed data format.

3.0 SYSTEM DESCRIPTION

Two typical digital communication systems are shown in Figure 4. The figure is intended to show the dissimilarity between two PCM systems. The first system depicts a two-wire switchboard. Telephone signals may be switched among themselves or trunked to a remote switch. The trunk is depicted as a PCM time division multiplexed link. A signal converter is required to convert the two-wire telephone interface to a four-wire interface, and to generate or detect signaling and supervision tones. The multiplexer is designed to operate with a full load signal of -4 dBm. The video interface of the multiplexer in this case is connected to a cable driving modulator and demodulator.

In the second system, four-wire telephones are shown interconnected by a four-wire switch. In this case, trunks are wired directly to the multiplexer from the switchboard. The multiplexer interface is 600 ohms and receives a -16 dBm full load signal and provides a +7 dBm output signal. This multiplexer is shown as an 8-bit PCM multiplexer. The video interface in this case is connected to a modulator and a demodulator designed for radio transmission.

The dissimilarity between the two systems depicted in Figure 4 is apparent.

4.0 INTEROPERABILITY

Consider a typical tactical field deployment. The tactical system is self sufficient; however, its proximity to commercial equipment is assured in most parts of the world. Problems of interoperability may be alleviated by simply providing two telephones to those users requiring them (possibly one military phone and one commercial phone).

In many applications, it may be desired to have multiple trunks interconnecting two communications systems. A digital code converter can be used to allow dissimilar equipment within each of the systems to be operated together. Alternatives to digital code conversion are shown in the upper portion of Figure 5. As shown in the first example, the interface may be accomplished by providing compatible multiplexers, modems, and transceivers, in each of the two communications systems which are to be interconnected. Most likely, the type of multiplexer, modem, and transceiver connected will be identical to that normally used in one of the two systems.

In the basic system, a tactical multiplexer and its associated transmission equipment have been relocated at the strategic switch. (Although a radio set is shown, a cable transmission system could be used.) The alien tactical multiplexer requires interface amplifiers and signaling converters to achieve a complete interface.

If a strategic radio set (or cable driver) is compatible with the tactical radio set (or cable driver), then only the multiplexer must be installed as shown in Figure 5, Option 1. In this case additional interface amplifiers may be required to interconnect the tactical multiplexer video signals with the strategic radio set.

Invariably, one of the systems must include alien equipment, that is, equipment which is unfamiliar to local operators which will be difficult to maintain. Assuming this can be accomplished, the problem remains of matching impedances and signal levels out of the multiplexer and also of matching signaling and supervision techniques between the two systems.

The preferred technique, using a Digital Code Converter to achieve interoperability, is shown in Figure 5 as Option 2. In this case there is no need for alien multiplex or transmission equipment.

A strategic multiplexer may be used at the strategic site, and a tactical multiplexer may be used at the tactical site. The electrical interface consists of 4 or fewer coaxial cables and is readily matched by internal circuits. This is to be compared with 12 to 24 matching circuits for an audio interface. Digital code conversion can transform data rate, channel modularity, PCM format and even signaling and supervision tones between two dissimilar systems. If a digital code converter is used, it is only required within one of the two systems which are to be interconnected. The other system remains virtually unchanged.

Installation of new systems, such as the addition of a satellite radio, is greatly simplified through code conversion. All systems (strategic, tactical, special, etc.) may utilize the same radio equipment. A code converter insures a compatible radio interface. The satellite link can utilize an existing format with code converters required at the few installations which do not use that format. Alternatively, a narrow band format may be dictated which would reduce transmission bandwidth. Eight bit PCM (64 kbps per channel) can be converted to 32 kbps CVSD for 2 to 1 compression or converted to 16 kbps CVSD for a 4 to 1 compression.

5.0 DIGITAL CODE CONVERTER TECHNIQUE

Digital Code Converters have been built which digitally convert:

12 channel, 6-bit, $\mu \hat{=} 100$, PCM $\longleftrightarrow$ 12 channel, 8-bit, $\mu \tilde{=} 255$ PCM
12 channel, 6-bit, $\mu \doteq 100$, PCM $\longleftrightarrow$ 18 channel, 32/16 kbps CVSD
12 channel, 8-bit, $\mu \doteq 255$, PCM $\longleftrightarrow$ 18 channel, 32/16 kbps CVSD

Where channel groupings between systems are different, some of the channels of the larger system are not used. The conversion is accomplished on a wideband trunk without restoring the signal to analog baseband. Trunk conversion is emphasized because of the economy realized through a single multichannel conversion.

The 8-bit PCM to 6-bit PCM code conversion method is relatively straight-forward. It is accomplished by a Read Only Memory as shown in Figure 6. PCM data of one format is serially shifted into a serial-to-parallel conversion shift register. This register addresses a Read Only Memory (ROM) whose output is the transformed code word. The output of the Read Only Memory goes to a parallel-to-serial conversion shift register. This conversion is accomplished by directly mapping one PCM code word into another.[3,4] Digital filtering is not required because identical sampling rates are used in both PCM systems.

A CVSD to PCM code conversion is more complex.[5] The sampling rate is either 32 kbps or 16 kbps for CVSD and 8 kHz for PCM. Digital filtering is required to transform the sampling rates.

Furthermore, the CVSD system uses a syllabic compressor. Decoding of the CVSD signal requires a knowledge of the past history of the channel. The 6.4 msec syllabic time constant used in the CVSD system indicates that at least a 6.4 msec history of the channel must be retained to decode the signal.

A CVSD to PCM code conversion is accomplished through the use of digital filters which emulate the performance of an analog CVSD encoder or decoder. The interconnected digital filters produce integer samples representing the signal encoded within the CVSD channel. These samples are ultimately produced at an 8 kHz sampling rate and may be directly converted from Read Only Memory to a compatible PCM format. The complexity of the CVSD to PCM code converter over that of a PCM to PCM code converter is increased because of the requirement for digital filters. However, the increase is not significant as individual digital filters may be time-shared between a number of multiplexed CVSD channels.

A block diagram in Figure 7 illustrates the PCM to CVSD and CVSD to PCM code conversion process. Read Only Memories are used to convert 8-bit companded PCM to 12-bit linear PCM which can be processed by the digital filters. For 6-bit PCM, a second ROM first converts 6-bit PCM to 8-bit PCM before conversion to 12-bit linear PCM.

An interpolator converts between the 8 kHz PCM sampling rate and the 32 kHz CVSD sampling rate by inserting 3 intermediate sampling points between each PCM sample. This yields the required 4 to 1 rate conversion. The filtered output is applied directly to a numerical magnitude comparator which is the input to the CVSD encoder. The reverse conversion uses the same filter for converting from the decoded CVSD sampling rate of 32 kHz to the 8 kHz PCM sampling rate. Look-up tables form the basis of the conversion of the 12-bit integers, first to 8-bit PCM and then, if required, to 6-bit PCM.

The digital CVSD decoder is shown in block diagram form in Figure 8. At the input is a 3-bit shift register with logic to detect 3 identical bits in a row. A detection will cause the input to the phonemic integrator to be switched to a level of 291, thereby causing the phonemic filter output to increase in value. The levels 3 and 291 represent the minimum and maximum step sizes the CVSD encoder inputs to the phonemic filter. These are at the same step size ratio, approximately 1 to 100, of an analog encoder (minimum, 30 millivolts to maximum 291 volts) and are analogous to the CVSD step sizes of the encoder. The filter output will continue to increase to a maximum of 59740 or until an input bit of different sense occurs to break the string of identical bits.

The phonemic filter consists of an adder, a register, and a multiplier. The register holds the output value of the filter which was computed during the last sample. The multiplier multiplies the last output by a factor of 0.995, emulating the discharge of a phonemic filter having a time constant of 6.4 msec. The input sample (291 or 3) is then added to the attenuated last sample to produce the current phonemic filter output. This value will be loaded into the register on the next clock pulse. The phonemic filter implements a difference equation:

$$Y_{(n)} \cong 0.995\ Y_{(n-1)} + X_{(n)}$$

where $Y_{(n)}$ is the new output, $Y_{(n-1)}$ is the previous output, and $X_{(n)}$ is the input (either 3 or 291). In this equation, 0.995 is approximated by 1019/1024 and all results are translated to integers. The two input values are determined from the CVSD algorithm itself. This phonemic filter has a time constant of 6.4 msec at a 32 kHz sampling frequency and accurately models the equivalent RC time constant. The divide by 8 is made for scaling down the value.

Each of the parallel bits from the phonemic integrator goes through "exclusive-or" gate which will selectively negate the entire output in response to the input CVSD bit sense. Thus, the output of the phonemic integrator will be either added to, or subtracted from (negative, 2's complement value), the loop integrator values. This operation is analogous to the operation of the analog CVSD.

The loop integrator is similar to the phonemic integrator except that the attenuation constant is 0.969 which is equivalent to an RC time constant of 1 msec at the 32 kHz sampling rate. The difference equation is:

$$Z_{(n)} = 0.969\ Z_{(n-1)} \pm Y_{(n)} \frac{1}{8}$$

where the addition or subtraction is controlled by the sense of the CVSD bit. $Y_{(n)}$ is the phonemic integrator output and $Z_{(n)}$ is the loop integrator output. The 0.969 is approximated by 31/32, and all results are truncated to integers.

A 2 kHz full load tone will be encoded by a 32 kbps CVSD encoder as a periodic sequence of 8 ones followed by 8 zeros. Table 1 presents a decoding example showing for each clock period: the CVSD bit; the input, $X_{(n)}$, to the phonemic integrator; the output, $Y_{(n)}$, of the phonemic integrator; the output, $Z_{(n)}$, of the loop integrator; and the decoded PCM. The decoded PCM is the output of the loop integrator, $Z_{(n)}$, divided by 16.

The division is necessary to compensate for the gain of the digital filters. The highest amplitude PCM signal is normalized to 2040 which can be represented by 12 bits. The value 2040 was chosen because it maps closely to and is a factor of the exact segment and points of the 8-bit, μ=255, PCM code.

Refer to the center of Table 1 where the CVSD bit first changes from 1 to 0. This breaks the string of identical bits and causes the input, $X_{(n)}$, of the phonemic integrator to be changed to 3. The new output from the phonemic integrator is calculated from its difference equation by adding the new input, $X_{(n)} = 3$, to the past output, $Y_{(n-1)} = 45067$, after multiplication by the factor $\frac{1019}{1024}$. The new output is $Y_{(n)} = 44850$.

The loop integrator output, $Z_{(n)}$, is calculated by dividing the phonemic integrator output, $Y_{(n)} = 44850$, by 8 and subtracting this value (in accordance with the "0" sense of the CVSD bit) from the previous loop integrator output, $Z_{(n-1)} = 22666$, after multiplication by the factor $\frac{31}{32}$. The new loop integrator output is $Z_{(n)} = 16351$. The loop integrator output, $Z_{(n)}$, is then divided by 16 to generate the final converted PCM value of 1022. This value will then be filtered by the interpolator and converted to the 8-bit PCM format.

TABLE 1 - DIGITAL DECODING EXAMPLE FOR 2 kHz TONE

CVSD	$X_{(n)}$	$Y_{(n)}$	$Z_{(n)}$	PCM
		45067	-22666	
1	3	44850	-16351	-1022
1	3	44634	-10261	-641
1	291	44707	-4352	-272
1	291	44780	1381	86
1	291	44852	6945	434
1	291	44924	12343	771
1	291	44996	17582	1099
1	291	45067	22666	1417
0	3	44850	16351	1022
0	3	44634	10261	641
0	291	44707	4352	272
0	291	44780	-1381	-86
0	291	44852	-6945	-434
0	291	44924	-12343	-771
0	291	44996	-17582	-1099
0	291	45067	-22666	-1417
1	3	44850	-16351	-1022

Note that a single 32 kHz clock line activates both filter registers and the slope overload detection shift register. The clock simplifies implementation. The same filters are economically shared in the code converter for all 12 channels and for both encoding and decoding. The filters and look-up tables are the critical components with respect to performance. However, most of the circuitry is associated with supporting functions such as synchronization, timing recovery and the electrical interface with multiplexers and radio sets.

Table 2 shows a comparison of the relative circuit complexity in the two code converters. The percentage of circuitry dedicated to each of the primary code converter functions is indicated. Frame synchronization represents one of the most complex code conversion functions. Twenty-eight percent for PCM-PCM and 18% for PCM-CVSD. The code conversion function itself requires only 6% of the PCM-PCM circuitry and 27% of the PCM-CVSD circuitry.

TABLE 2 - COMPARISON OF RELATIVE CIRCUIT COMPLEXITY

	PCM-PCM	PCM-CVSD
Input Interface	8%	7%
Timing Recovery	8%	7%
Frame Synchronization	28%	18%
Demultiplex	8%	6%
Sync/Signalling Deletion	6%	3%
Code Conversion	6%	27%
Timing Regeneration	12%	12%
Sync/Signalling Insertion	4%	3%
Multiplex	8%	6%
Output Interface	12%	11%

6.0 SIGNAL-TO-QUANTIZATION NOISE RATIO PERFORMANCE

Signal-to-quantization noise ratio, S/Nq, was the principal performance parameter used in evaluation of the digital code converter. S/Nq was measured over a range of input signal levels from -4 dBm to -44 dBm. The range in signal level is necessary to evaluate the effects of the PCM and CVSD compression techniques. For the 6-bit PCM to 8-bit PCM conversion, there was a significant improvement in S/Nq, up to 8 dB, for signal levels between -14 dBm and -28 dBm, when using the Digital Code Converter. In present PCM systems, the analog patch method is used to interface the 6-bit to the 8-bit system. Measurements for the analog patch method were performed by sending the analog signal into the 6-bit multiplexer (one A/D and D/A hop) and then through an 8-bit multiplexer (A/D and D/A hop) and measure the S/Nq on the output. The improvement was relative to a 6-bit multiplex operating back-to-back with an identical 6-bit multiplexer. This was due to the improved digital mapping at the 2nd segment end points by the Digital Code Converter.

The CVSD to 6-bit PCM conversion performance was measured and compared against predicted values. Figure 9 shows how the performance prediction was made. The performance of a 6-bit PCM multiplexer and a CVSD coder/decoder were each measured back-to-back. Figure 9 shows the S/Nq versus input signal level. The predicted value was calculated assuming that the independent noise sources from CVSD and PCM would add. Where both curves give the same S/Nq (at -14 dBm) then the predicted S/Nq was 3 dB lower.

Figure 10 shows the actual measured performance. It can be seen that the 6-bit PCM to CVSD conversion performance is significantly better than predicted. This is due to the improved digital decoding of 6-bit PCM. The analog PCM decoder is bypassed in the PCM to CVSD converter. In converting from CVSD to 6-bit PCM, the analog PCM decoder is used (the analog PCM encoder is bypassed) and the resultant dip in S/Nq for signal levels around -25 dBm is apparent. In both cases, the digital code conversion performance is better than predicted.

Figures 11 and 12 show the results of comparative tests run on the Digital Code Converter and on equipment connected back-to-back in an analog interface. An improvement of approximately 3 dB is realized through digital code conversion, depending on the particular value of the input level. In Figure 11, a test signal was supplied to a 32 kbps CVSD encoder. The digital CVSD encoder output was multiplexed with other CVSD signals, then digitally code converted to a multiplexed PCM signal and finally demultiplexed and decoded to analog. The S/Nq distortion in the analog output was measured with the results shown. In the analog test, the digital CVSD signal was decoded to analog by a CVSD decoder and then fed into a PCM multiplexer. The signal was the PCM encoded, multiplexed, demultiplexed, and decoded. The S/Nq in the output was measured and plotted for comparison. The significant improvement through digital code conversion is apparent. Figure 12 shows similar results for 16 kbps CVSD.

7.0 CONCLUSIONS

The digital code conversion interoperability technique which has been presented, offers significant advantages over other alternatives. By providing code and signal conversion at the video output of a compatible multiplexer, complete compatibility with the switchboard and local user instruments can be maintained. The alternative, the installation of an alien interface multiplexer circuit and signal conversion module which is compatible with a remote system, would necessitate an electrical conversion of signal amplitudes, impedance and possible signaling and supervision techniques. The digital code converter does not require forced air cooling and uses little power. As no controls are required for the digital code converter, a sealed enclosure may be used which simplifies EMI constraints.

The digital code converter offers significant performance improvements. In particular, the unit operates synchronously with the input data and alleviates the need for channel filters. In a PCM to PCM code conversion, it is possible to provide multiple conversions with no additional build-up of quantization noise.

Where additional degradation is permissable, an 8-bit PCM system (a 64 kbps channel) may be converted to 16 kbps CVSD providing a 4 to 1 bandwidth reduction. A digital code converter applique can be used to provide bandwidth compression[2] for existing multiplex systems. The installation of a new multiplexer would require training and extensive maintenance facilities. The digital alternative is easily maintained and requires no adjustments.

The use of these devices will extend life-cycle utilization of current inventory equipment in step with evolutionary introduction of new TRI-TAC equipment.

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5. Digital Code Converter - Final Report - Betts, W. L., Green, R. T., Electronic Communications, Inc., June 1975, Contract No. DAAB07-75-C-0304.

The Digital Code Converter equipment was developed by Electronic Systems Engineering, E-Systems, Inc., ECI Division, St. Petersburg, FL, under the technical guidance of the US Army Electronics Command under Contract No. DAAB07-73-C-0304.

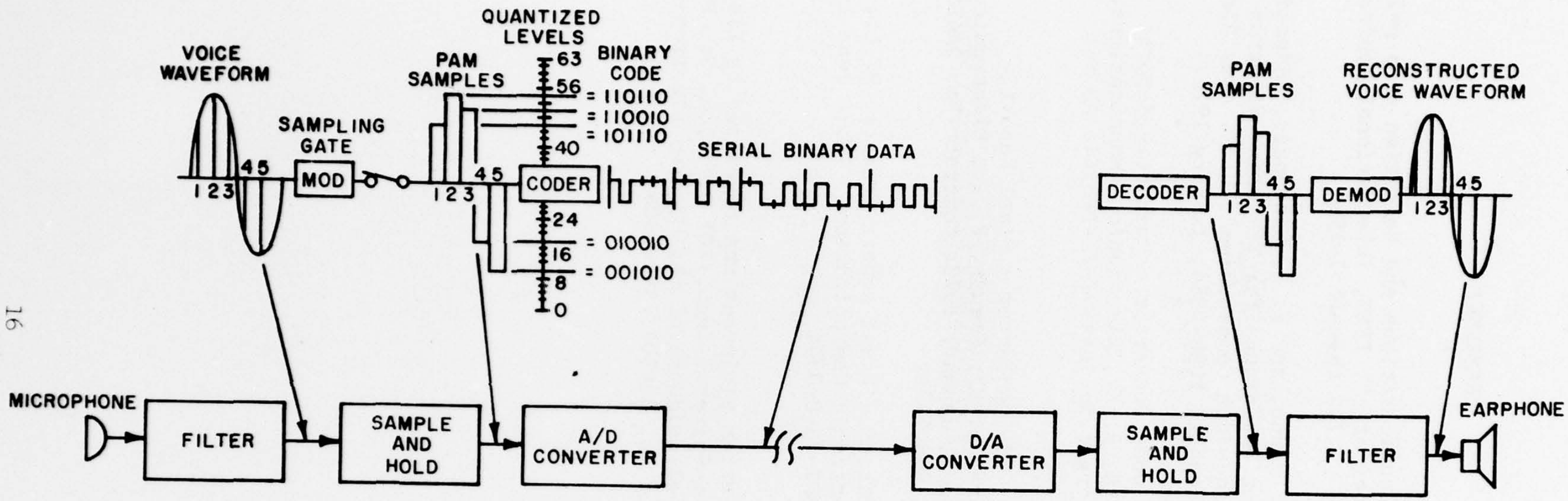


FIGURE I. WAVEFORM SAMPLING FOR PULSE CODE MODULATION

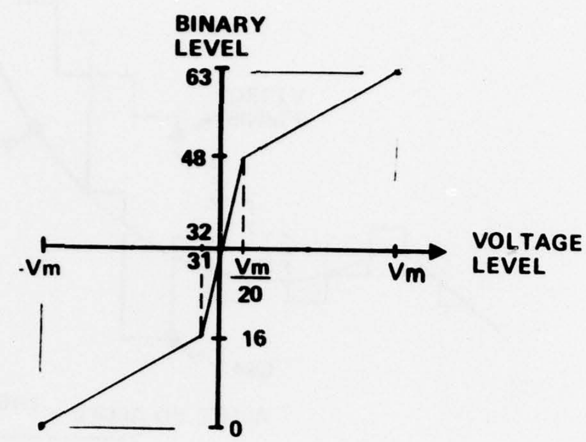


FIGURE 2. 6-BIT PCM COMPRESSION (FOR TD-352)

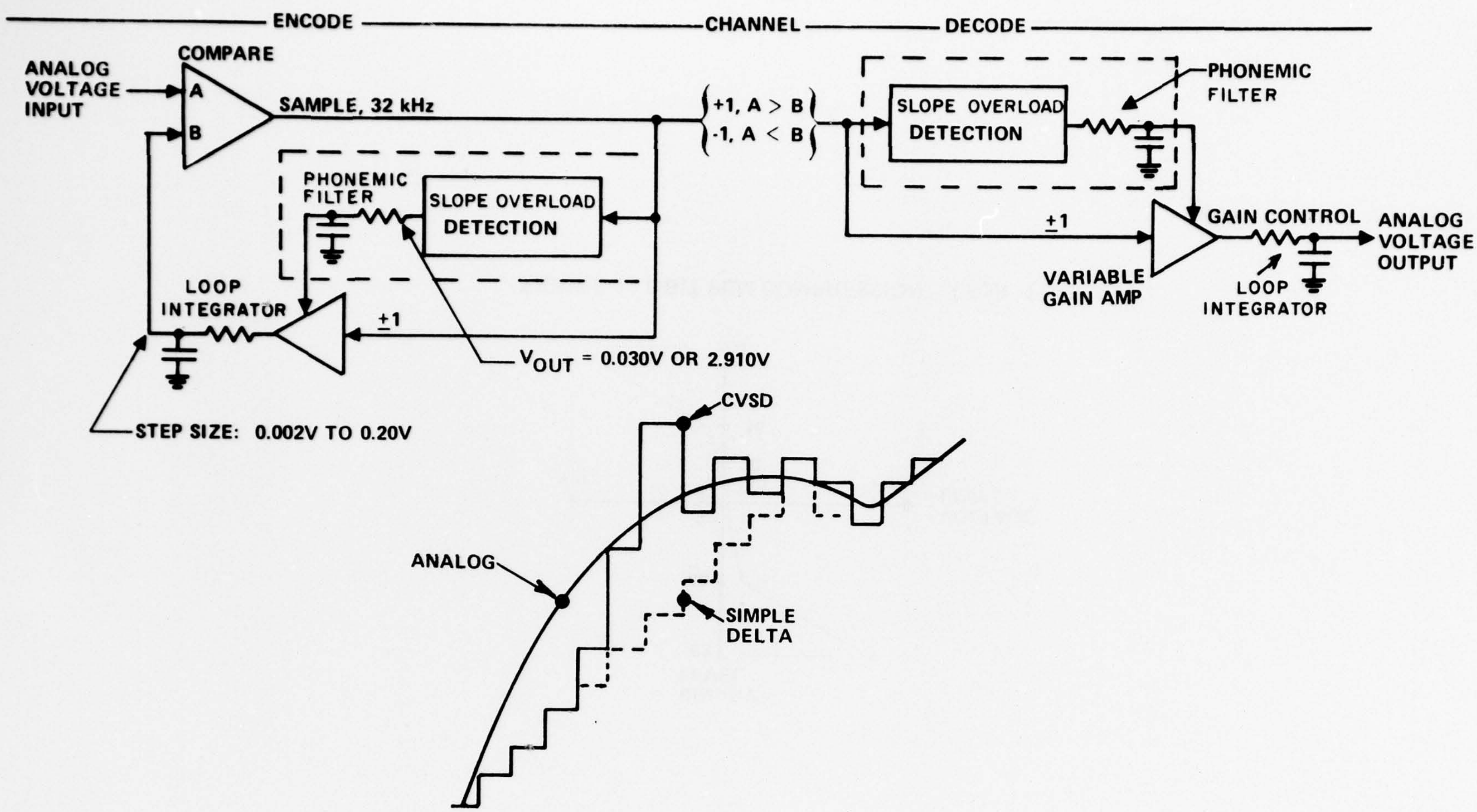


FIGURE 3. CONTINUOUSLY VARIABLE SLOPE DELTA MODULATION (CVSD)

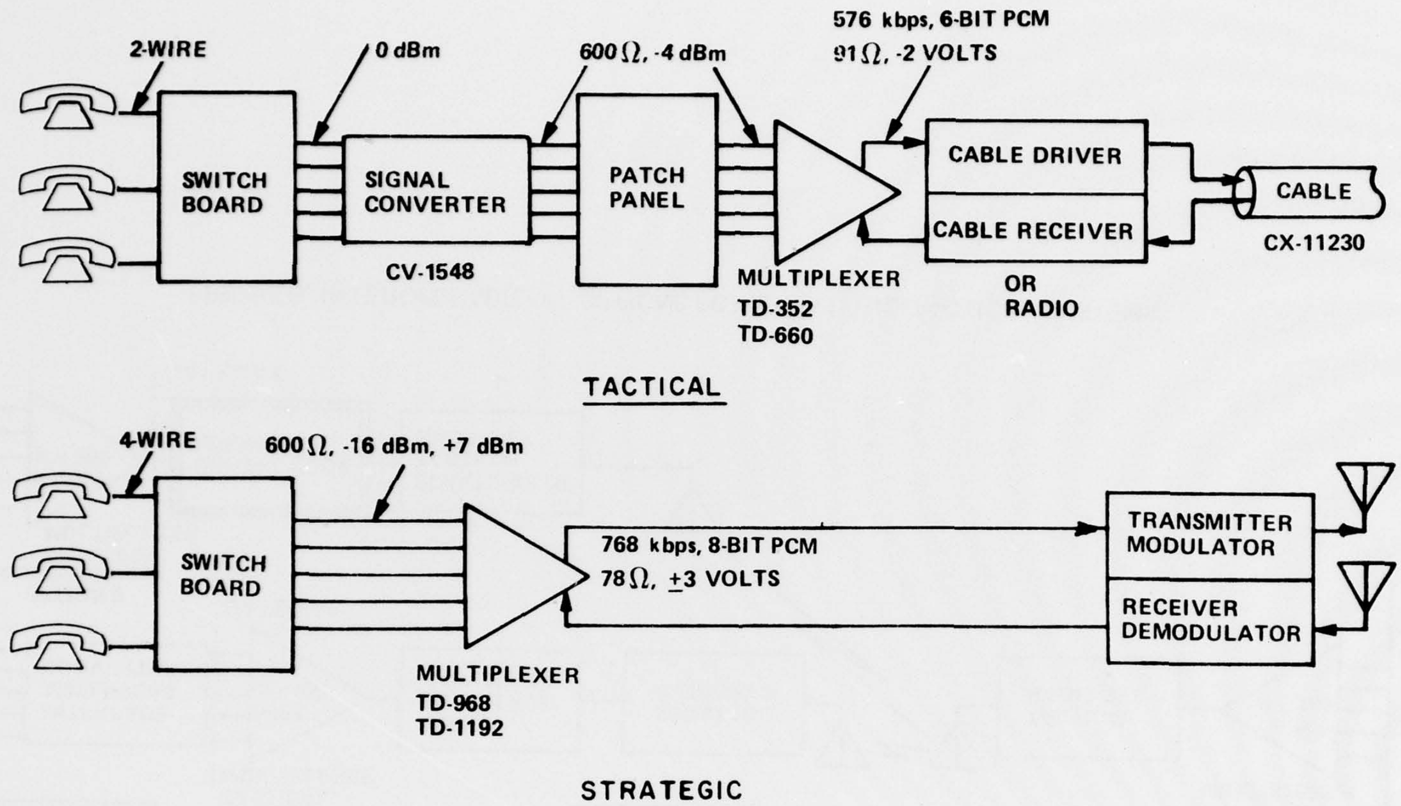


FIGURE 4. TYPICAL DIGITAL COMMUNICATION SYSTEMS

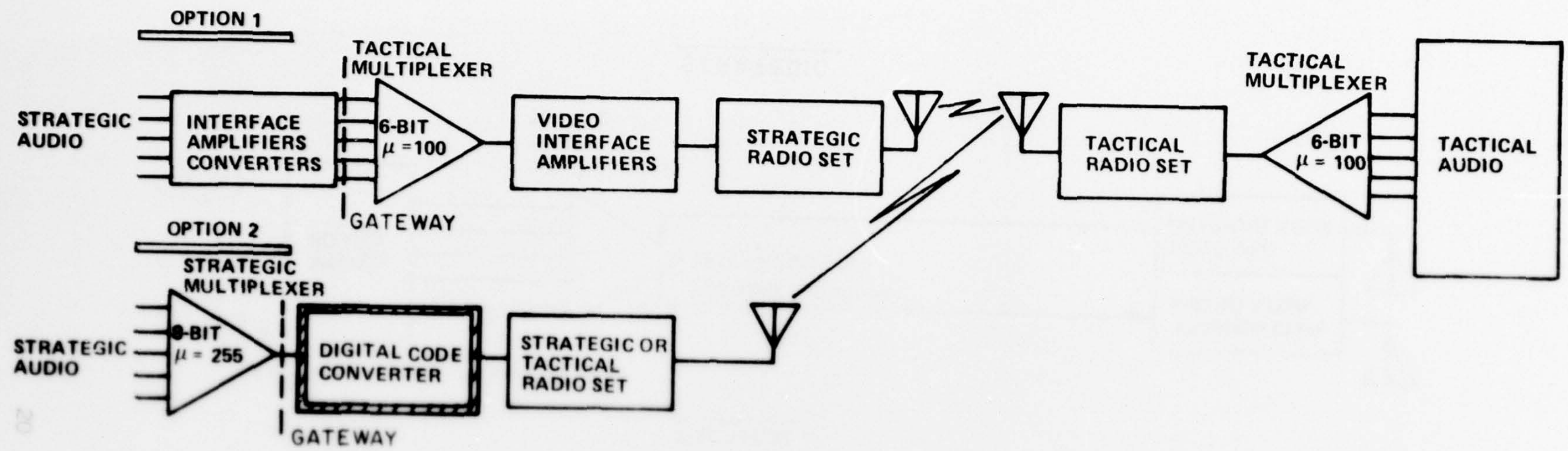


FIGURE 5. INTEROPERABILITY OPTIONS FOR STRATEGIC/TACTICAL SYSTEMS

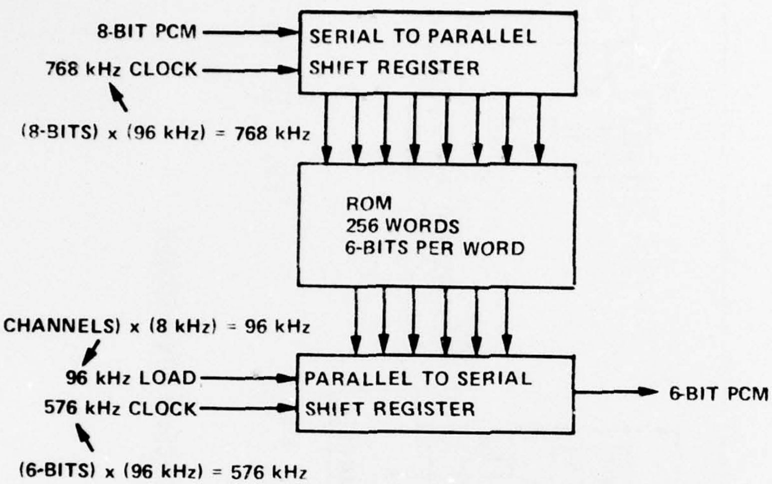


FIGURE 6. 8-BIT PCM TO 6-BIT PCM CODE CONVERSION

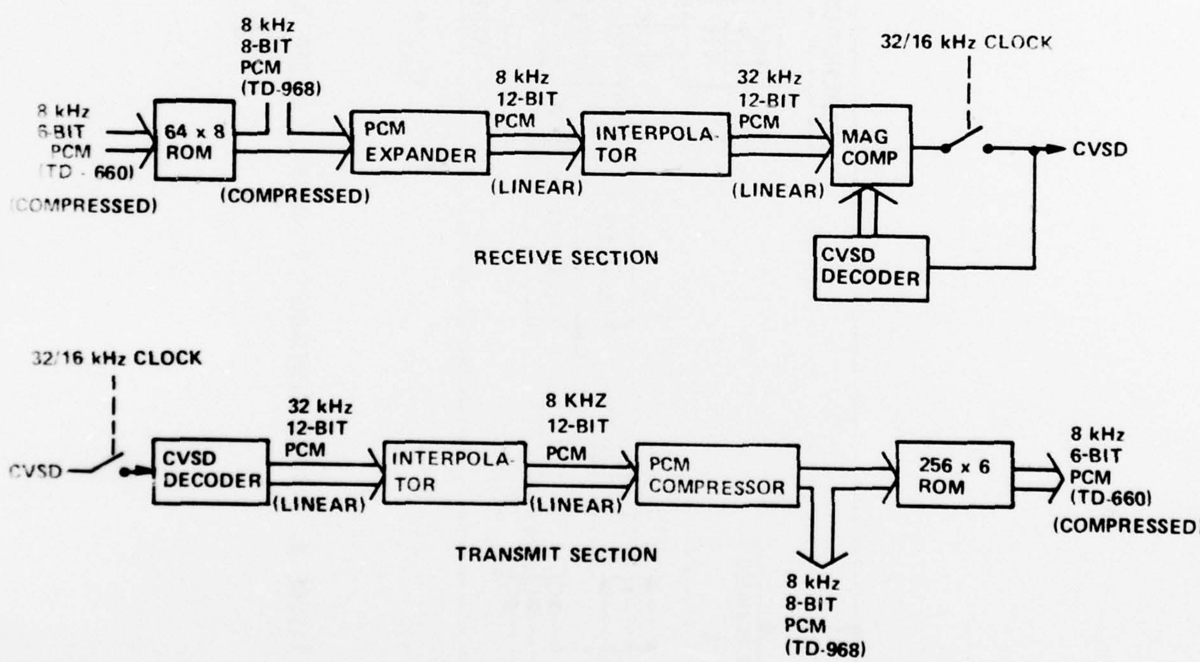
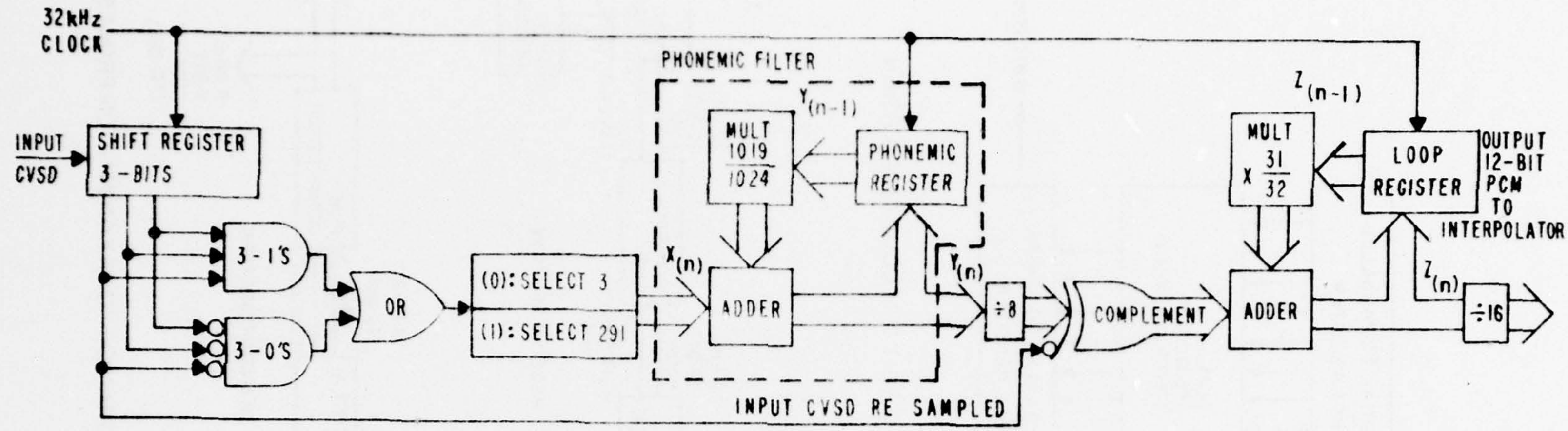


FIGURE 7. BLOCK DIAGRAM OF CODE CONVERSION PROCESS



$$Y_{(n)} = \frac{1019}{1024} Y_{(n-1)} + X_{(n)} \qquad Z_{(n)} = \frac{31}{32} Z_{(n-1)} \pm \frac{Y_{(n)}}{8}$$

FIGURE 8. DIGITAL CONTINUOUSLY VARIABLE SLOPE DELTA (CVSD) MODULATION DECODER

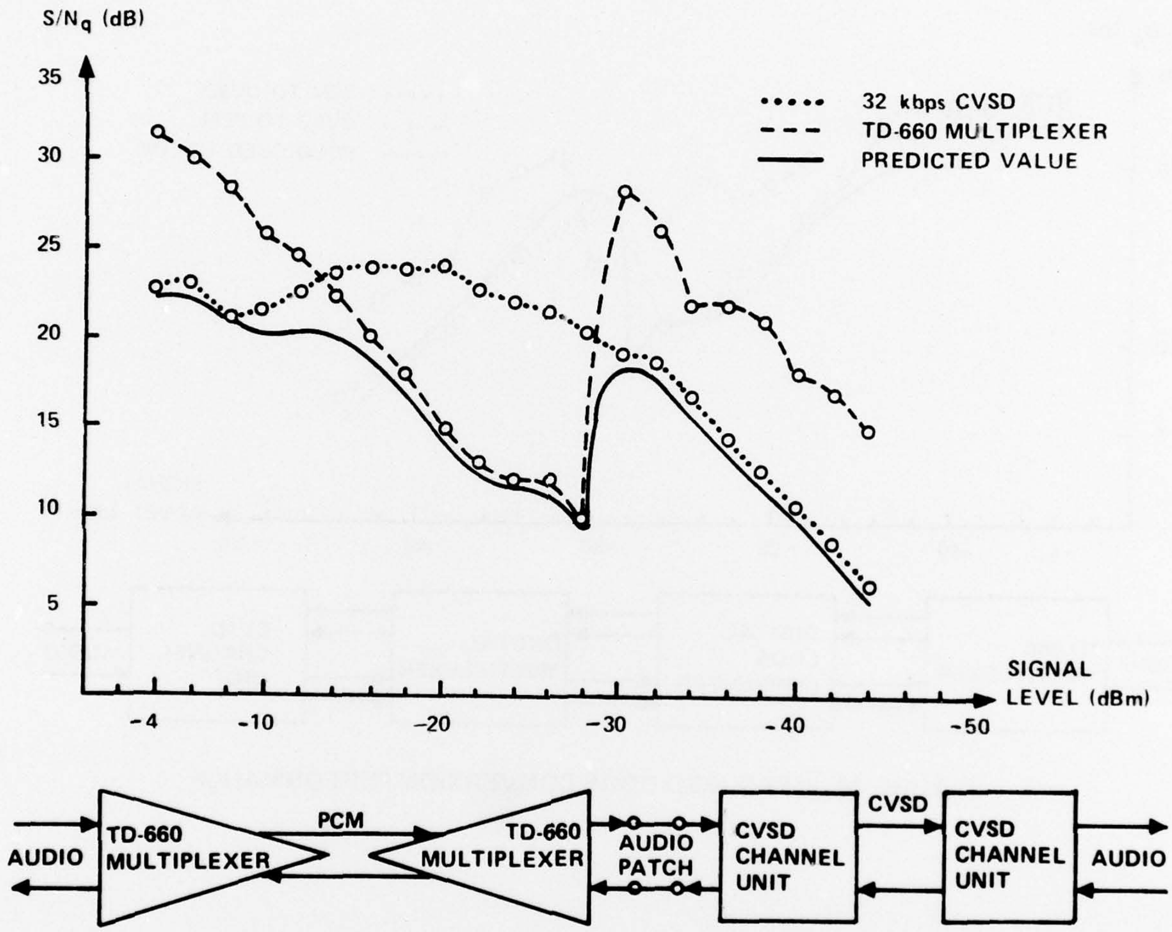


FIGURE 9. AUDIO PATCH PERFORMANCE PREDICTION

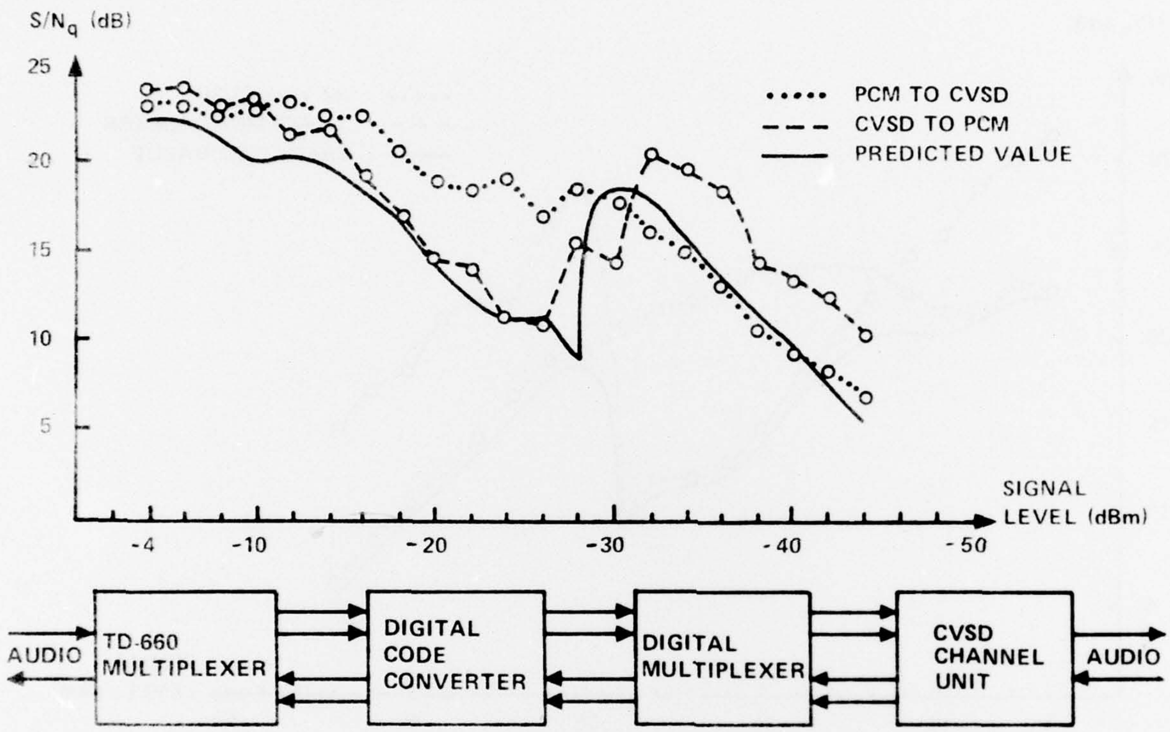


FIGURE 10. MEASURED CODE CONVERSION PERFORMANCE

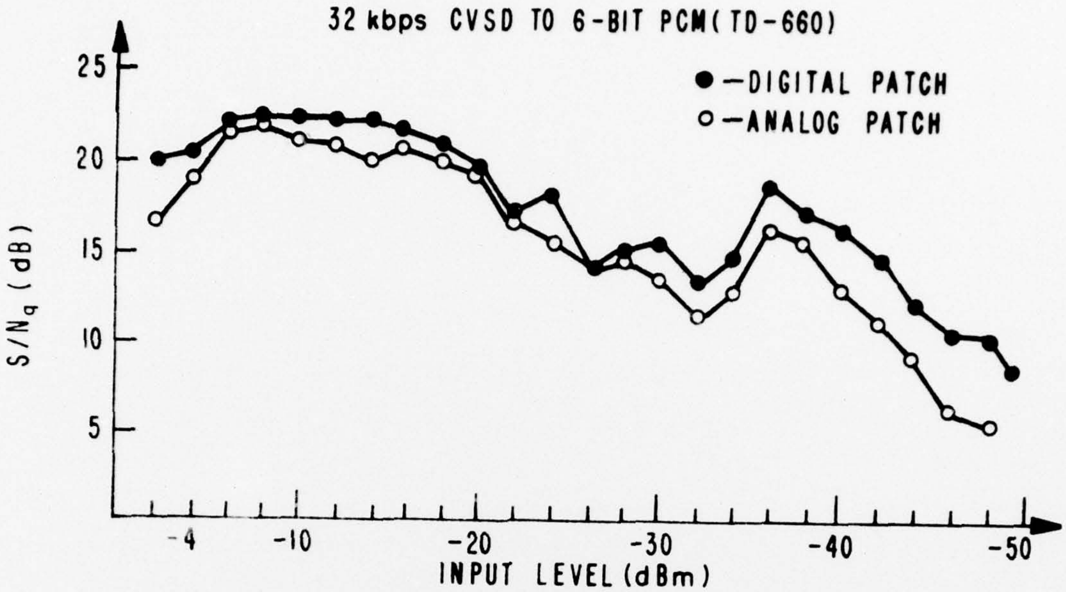


FIGURE 11. DIGITAL CODE CONVERSION VS. ANALOG PATCH

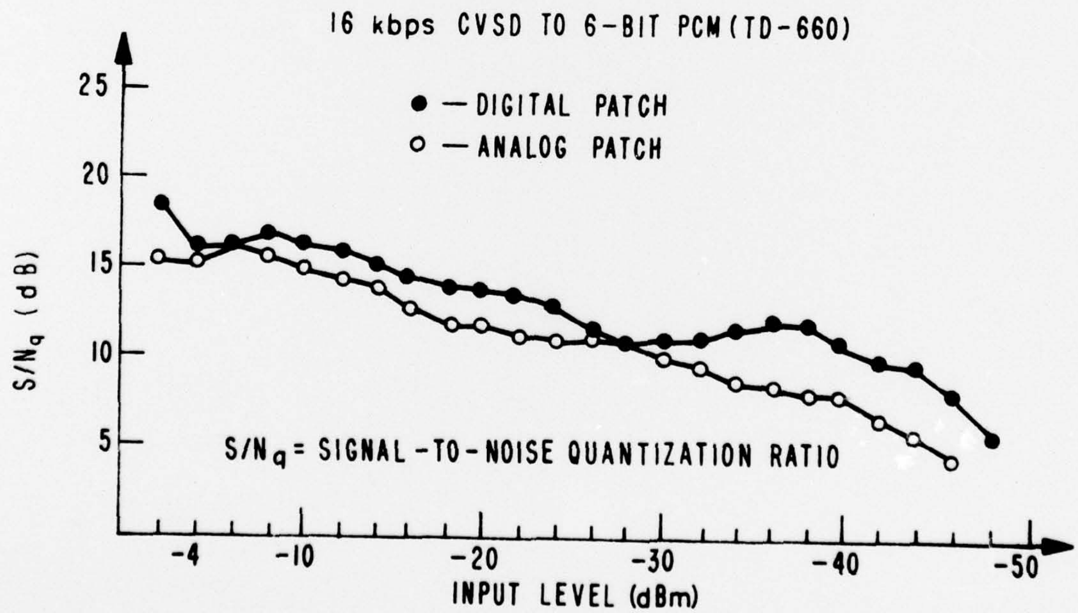


FIGURE 12. DIGITAL CODE CONVERSION VS. ANALOG PATCH