ABSTRACT

Wideband Data Recording Using FM Multiplexing on Analog Tape Recorders

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Introduction

It wasn't so many years ago that, the primary means of recording wideband data measurements was FM multiplexing on magnetic tape. Users would purchase several hundred modulators, select one of several multiplex format configurations that matched their data bandwidth requirements, and multiplex 8 to 10 or possibly up to 20 of these modulators onto various tracks of an analog reel-to-reel recorder. Depending on when in time you used this type system, transducer outputs as low as 100 Hz required FM multiplexing because digital recording just couldn't hack it. With improved digital recording techniques and much faster digital circuitry, digital recording has replaced FM recording as the preferred method of acquiring analog data. Today, a few test programs, primarily in the military aircraft jet engine test community, still use FM multiplexing. In these programs, data bandwidths up to 60 kHz are common, and channel needs to 200 measurements still dictate the need for FM multiplexing. A new FM system developed by Metraplex Corporation offers the test engineer a reason to reconsider FM multiplex recording for his new data acquisition system.

This paper will discuss this new digitally implemented, FM multiplexing system, that uses digital signal processing (DSP) technology to create FM modulators and demodulators. The use of application specific integrated circuits (ASICs) and programmable array logic (PAL) circuitry provides the user with a programmable system for recording data bandwidths up to 120 kHz and beyond. Gone are the screwdrivers for adjusting the equipment for drift. Gone are the piles of plug-ins to adapt to different recording speeds. Gone are the time base errors due to tape recorder flutter. Today's FM multiplexing system is fully programmable, uses the GPIB bus like other test equipment for set up and control, and doesn't require a technician to maintain it. We, as data recording engineers must revisit this recording technique and use it where it fits. Let's teach the young engineers coming out of college that a super solution to wideband recording still exists and should be considered when designing data acquisition systems.

Background

Some of you may ask, what is FM multiplex recording? Others may have forgotten it is used every day on national test ranges for transmitting wideband data in restricted bandwidth telemetry links. FM multiplexing provides a method of dividing a baseband into specific frequency slots, the width of each slot depending on the data to be transmitted and the ability to recover that information with minimal degradation.

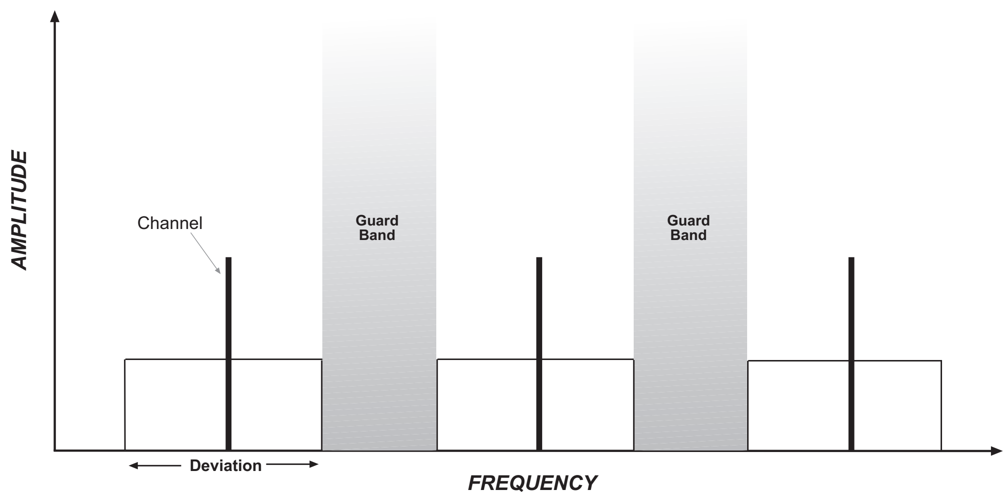

Figure 1. FM Multiplex Channels in Baseband

The best known FM multiplexing system is the FM radio transmissions we enjoy each day. Here, many channels are transmitted into air, each with a deviation of ±75 kHz and a data bandwidth of 15 kHz (high fidelity music). You tune your FM demodulator (receiver) to the channel you want to hear, and immediately you get clean, noise free music. Data recording uses the same approach except the FM channels are summed together and recorded on tape, and transmitted over radio or carried from a remote location by metallic fiber cable. In nearly all cases, the signal is recorded on analog tape. On playback, a demodulator, or group of demodulators, are tuned to the channel of interest. Clean, noise free data, the result of our recording effort, is available for processing.

Although, any carrier frequency and deviation may be used for FM modulation, most systems use standards approved by the U.S. Government. The IRIG 106-96 standards, developed by the Inter Range Instrumentation Group or the National Range Commanders Council, has been in use now for over 50 years. Using IRIG constant bandwidth formats, up to 21 channels can be recorded on a single track of an analog tape recorder. Assuming a user needs to record 16 kHz data, a 28 track recorder would accommodate 588 data channels. As the data bandwidth increases, the number of channels that can be recorded decreases. At the widest constant bandwidth system, three 128 kHz channels can be recorded on a single tape track. Again, using a 28 track recorder, 84 channels of 128 kHz data can be recorded.

	IRIG A ±2 kHz CENTER FREQUENCY (kHz)		IRIG B ±4 kHz CENTER FREQUENCY (kHz)		IRIG C ±8 kHz CENTER FREQUENCY (kHz)		IRIG D ±16 kHz CENTER FREQUENCY (kHz)		IRIG E ±32 kHz CENTER FREQUENCY (kHz)		IRIG F ±64 kHz CENTER FREQUENCY (kHz)		IRIG G ±128 kHz CENTER FREQUENCY (kHz)		IRIG H ±256 kHz CENTER FREQUENCY (kHz)
**	8	*0A*	16	*1B*	32	*3C*	64	*7D*	128	*15E*	256	*31F*	512	*63G*	1024
	16	*1A*	32	*3B*	64	*7C*	128	*15D*	256	*31E*	512	*63F*	1024	*127G*	2048
	24	*2A*	48	*5B*	96	*11C*	192	*23D*	384	*47E*	768	*95F*	1536		3072
	32	*3A*	64	*7B*	128	*15C*	256	*31D*	512	*63E*	1024	*127F*	2048		
	40	*4A*	80	*9B*	160	*19C*	320	*39D*	640	*79E*	1280		2560		**
	48	*5A*	96	*11B*	192	*23C*	384	*47D*	768	*95E*	1536		3072		
	56	*6A*	112	*13B*	224	*27C*	448	*55D*	896	*111E*	1792		3584		
	64	*7A*	128	*15B*	256	*31C*	512	*63D*	1024	*127E*	2048				
	72	*8A*	144	*17B*	288	*35C*	576	*71D*	1152		2304				
	80	*9A*	160	*19B*	320	*39C*	640	*79D*	1280		2560				
	88	*10A*	176	*21B*	352	*43C*	704	*87D*	1408		2816				
	96	*11A*	192	*23B*	384	*47C*	768	*95D*	1536		3072				
*	104	*12A*	208		416		832		1664		3328				
	112	*13A*	224		448		896		1792		3584				
	120	*14A*	240		480		960		1920		3840				
	128	*15A*	256		512		1024		2048						
	136	*16A*	272		544		1088		2176						
	144	*17A*	288		576		1152		2304						
	152	*18A*	304		608		1216		2432						
	160	*19A*	320		640		1280		2560						
	168	*20A*	336		672		1344		2688						
	176	*21A*	352		704		1408		2816						

Figure 2. IRIG Constant Bandwidth Multiplex

You may ask at this point, what is the limitation in using a serial digital recording system to handle wideband data. The problem is bit rate and the ability to record that bit rate. A general formula for determining what the equivalent bit rate for recording data compared with FM multiplexing is given below:

Bit rate = # of channels X bits per word X data response per channel X sample per cycle of highest frequency

Let's take as an example the 84 channel system above with each channel having a data bandwidth of 128 kHz. Using the formula above:

Bit rate = 84 chs X 10 bits X 128,000 Hz X 4 sample/cycle = 430,080,000 bits/second

Have you ever seen a serial tape recorder that can accommodate this bit rate at a comparable price to a 28 track, direct record, analog recorder? Have you noticed an abundance of data systems that can multiplex and digitize data at these rates?

Based on this example, you can see that there are test requirements now that can only be performed practically using FM multiplexing.

FM Multiplexing Subsystem

The basic elements of today's FM data recording system are essentially unchanged from years gone by. A typical end-to-end FM data acquisition system is shown in figure 3. The changes come in the architecture of the system design. Virtually every system component is fully programmable through the GPIB instrumentation bus. Digital design eliminates electronic calibration. Tape recording is set under computer control. Data processing and display are immediately available after the test. Complete flexibility is provided to select recording bandwidths and frequency scaling of the reproduced data by changing tape speeds automatically. Since you will be discussing tape recorders as part of this meeting, the remainder of this paper is devoted to the essential elements of the system, the FM multiplexing and demultiplexing hardware.

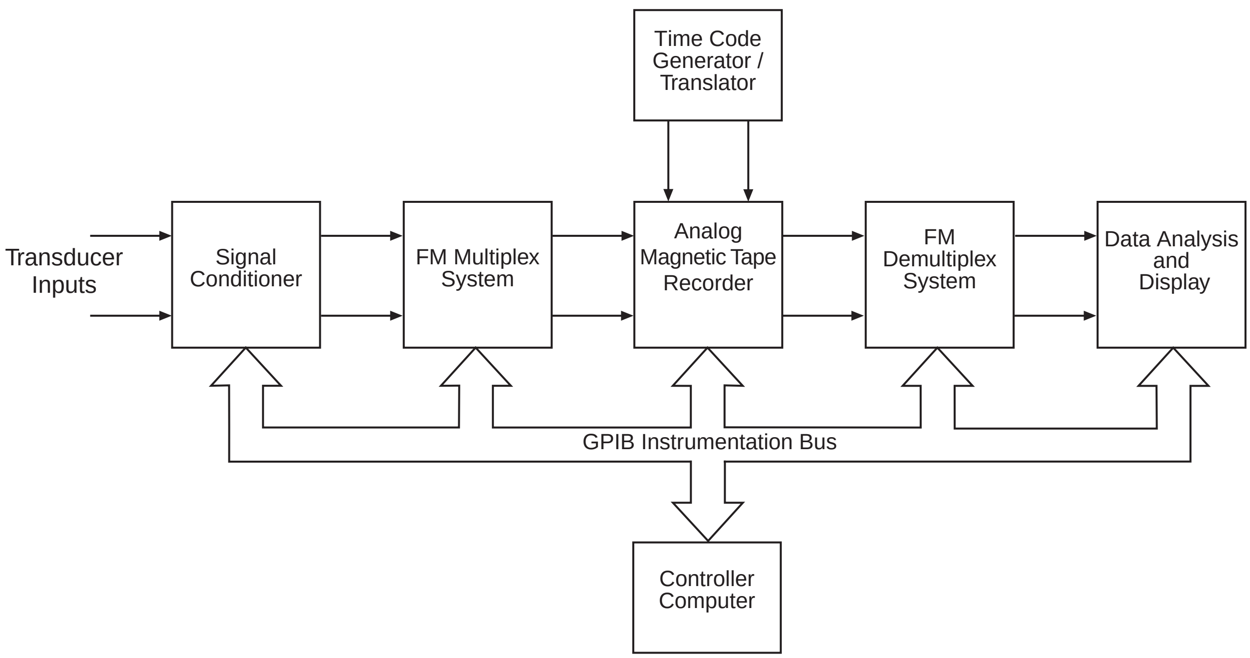

Figure 3. Typical End-to-End FM Data Acquisition System

Metraplex introduced its' Digital Frequency Multiplexer (DFM) in 1993. The previous analog multiplexing systems used modulators based on a mutli-vibrator design, the DFM system utilizes DSP design technologies, implemented with proprietary application specific integrated circuits that perform digital synthesis and modulation. The numerically controlled digital oscillator can operate at any carrier frequency from 256 Hz up to 4.0 MHz. After amplification and pre-sample filtering, the analog input to be multiplexed is digitized and applied to the oscillator, thus modulating the carrier. Deviation is limited to a range from 0.5% to 50% of the carrier frequency, not to exceed 128 kHz in the present design. The digital oscillator outputs are digitally summed, then applied to a digital to analog (D/A) converter, analog filtered and output as an FM multiplex. Since the digital oscillators have virtually no harmonics of the carrier, the output multiplex is exceptionally clean compared to the older analog systems. The digital oscillators are fully programmable eliminating all plug-in channel selectors. Even more important, they are devoid of drifts, the real problem in analog systems.

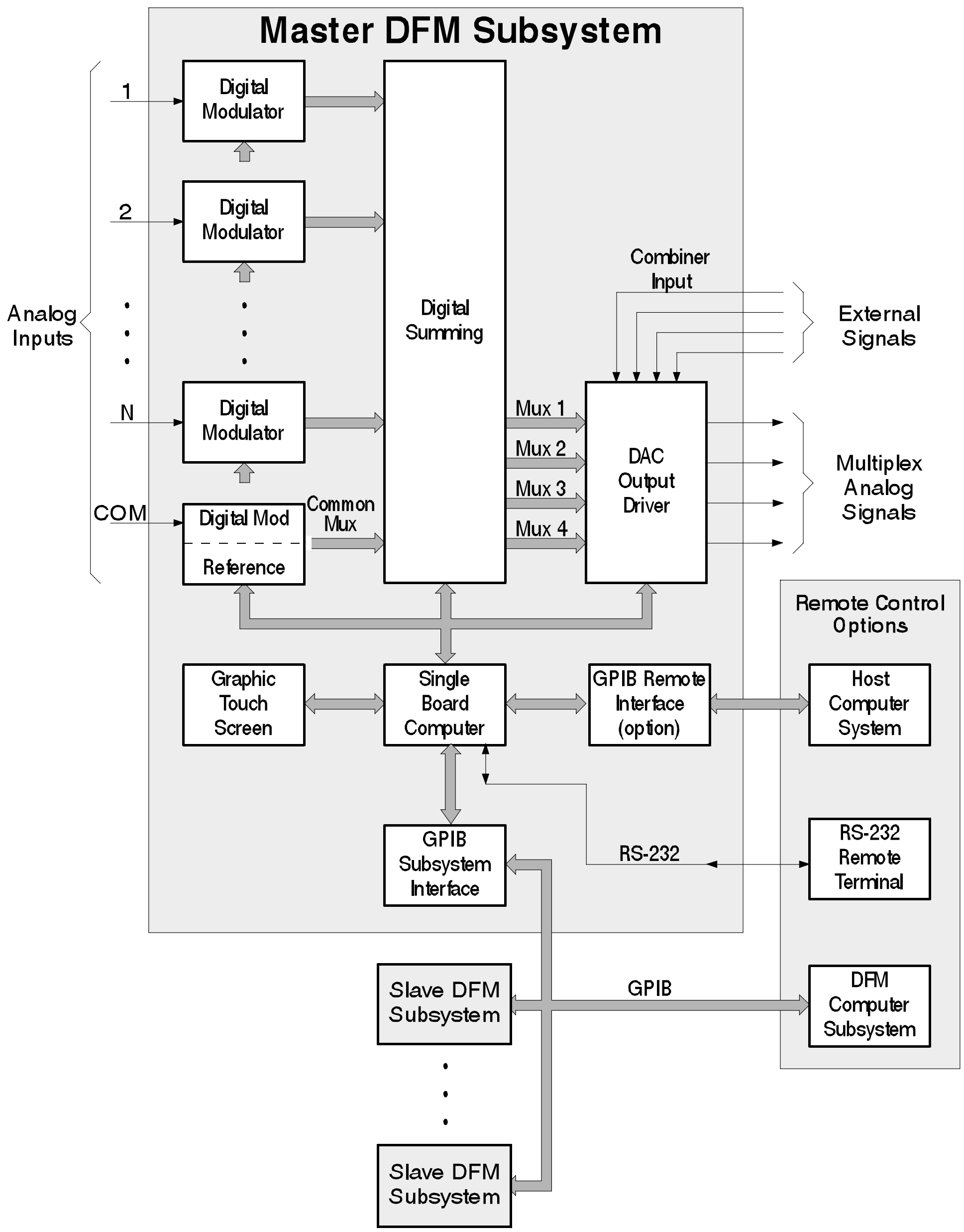

Figure 4. Model DFM Digital Frequency Modulator Block Diagram

A block diagram of the DFM is shown in Figure 4. The system uses a master/slave design to build up multiplexes for the tape recorder. There are 36 oscillators per chassis (9 cards, 4 channels per card) and these channels may be summed into one of four multiplex outputs. Selecting a card puts all four channels in the selected multiplex. You can configure a wide range of multiplex configurations using the IRIG standards. Four output multiplexes per chassis are provided. Typical multiplex configurations used by many customers are; 3 groups of 12, 4 groups of 8, 2 groups of 16, etc. The number of channels per multiplex is dependent on the direct recording bandwidth for the record speed selected, and of course, the transducer data bandwidth, which we already discussed. Slave units are identical to the master unit and are programmed by the master through a proprietary GPIB bus. Remote programming by an IBM-PC compatible computer or host computer is provided.

A unique feature of the DFM system is the ability to change the multiplex configuration in step with the tape recorder tape speed commands. For example, assume you wanted to record 8 channels of 32 kHz and 120 IPS, then reduce the record speed to 30 IPS, to record the same channels at a bandwidth of 8 kHz. The DFM can sense the tape speed selection via GPIB commands from the recorder and automatically reprograms the digital oscillators to the new carrier frequencies and deviation.

For those who are interested in the interior design of the DFM, Appendix A provides a detailed description of the product.

FM Demultiplexing Subsystem

Once the desired data measurements are multiplexed and recorded on magnetic tape, a wide range of analysis tools are available for data processing. The primary area of interest for the data user is spectrum analysis. At the present time, most analysis is done on workstations using FFT analysis software. These computers provide power spectral analysis, transmissibility measurements and correction evaluation. With the bandwidths associated with acoustic and vibration measurements, usually only one or two measurements can be processed simultaneously due to the throughput and storage capacities of these processing systems. The data analysis engineer then needs a programmable FM demultiplexing system to select specific data channels from the recorded FM multiplexes for analysis. By playing back the tape several times, all the data channels are processed through his analysis system under the control of the workstation.

The Digital Frequency Demultiplexing (DFD) system (figure 5) developed by Metraplex, consists of a master subsystem, capable of playing back two FM multiplexes simultaneously, and slave units programmed through the master subsystem to play back additional multiplexes. In most cases a single master subsystem is required for data processing due to the high frequency data content to be processed. However, in some uses such as viscosity research being conducted on new wing designs by NASA

Dryden Flight Test Center, up to 7 demultiplexer units are required to play back all 14 tape tracks simultaneously. Having access to all data channels simultaneously allows the user to compare any two channels of interest. It should be pointed out that having all data channels recorded on one recording media, and using the inherent precise time correction of FM multiplexing, the user can compare time relationships between data channels with good accuracy.

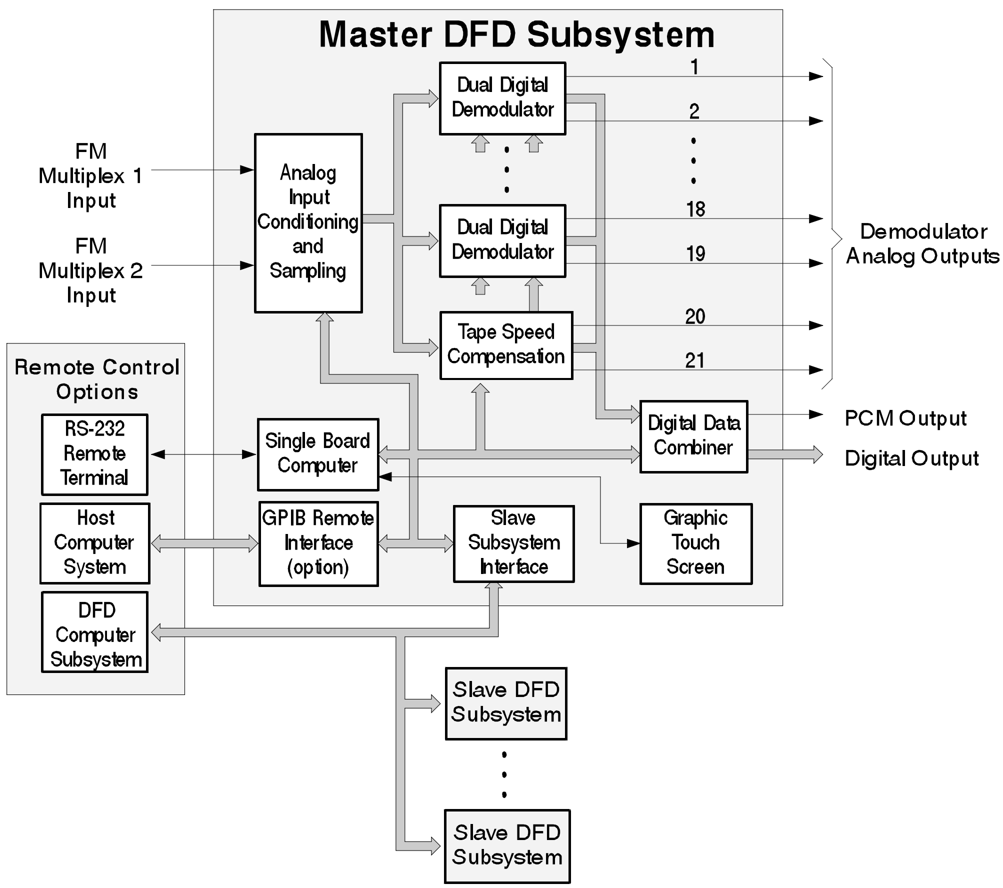

Figure 5. Model DFD Digital Frequency Demultiplexer Block Diagram

Each DFD master or slave subsystem will accommodate up to 20 data demodulators. The data demodulators can be allocated under program control, to either input multiplex. As an example, Pratt & Whitney Aircraft uses two different multiplexes, a 12 channel multiplex for low frequency measurements below 40 kHz and a 6 channel multiplex for data up to 64 kHz. These multiplexes are recorded on alternate tape recorder tracks. On play back, a single master subsystem will allow them to look at tracks 1 and 2, rewind the tape and process tracks 3 and 4 and so on. Of course, for large multiplexes, with up to 18 or 20 channels, a full demultiplex box would be needed. With the programmability of the demultiplex system and the tape recorder, it is quite easy to set up an automated process under computer control to process the recorded data.

Since recording on analog tape is not a perfect process, noise introduced due to tape speed errors (wow and filter) must be eliminated from the data. Built into each demultiplexer chassis are two tape speed compensation (TSC) demodulators, one for each input multiplex. Each TSC demodulator detects amplitude and time base errors from a reference signal recorded on each tape track and properly corrects the output data channels for these errors. The improved performance of the new flutter compensation system is quite significant relative to the older analog system.

With the introduction of DSP technology in today's equipment design, the filters used in data acquisition systems are considerably better than past systems. This is particularly apparent in the DFD. The demodulator bandpass filter, although having the same inband characteristics as the older analog demodulator, has noticeably sharper attenuation out of bandwidth without causing data distortion. The output filter is a Butterworth type filter flat to within 0.1 dB at cutoff but down nearly 60 dB at 2 times cutoff. Thus, greatly reducing crosstalk between data channels and improving the signal to noise response of the output signal. You can take advantage of this new filter by extending the data bandwidth to a deviation ratio of 1. With crosstalk and noise about the same as a deviation ratio of two in the older analog discriminators, channel bandwidth is in effect doubled. Another unique feature of the digital output filter is the linear phase relationship over the output data bandwidth. Whereas the older analog demodulators could be either constant amplitude or linear phase, the new demodulator provides both capabilities in one unit, there by, significantly reducing data distortion.

Since digital signal processing is used to perform the FM demodulation, the output of each demodulator after filtering is a digital signal. Sixteen bit processing is used in the digital circuitry, which provides in the best conditions, a dynamic range of 96 dB. Great care was taken in the design to minimize output noise. In each demodulator card, the noise measured in digital counts is less than ±5 in most units. This provides a signal to noise ratio of 6400 (32,000/5)or 80 dB. This measurement is made using the DFM as the data source. Therefore, the back to back signal to noise of the multiplexing system excluding the effort or tape approached this performance. Taking into account the noise contribution of the analog tape recorder, it is reasonable to expect an FM recording system performance of 65 dB (11 bits) and resolution approaching 14 bits. An analog output is provided for each demodulator for display on oscilloscopes, strip channel recorders and analog spectrum analyzers. Since the digital output must be converted to an analog voltage and reconstructed with a special output filter, the analog output is not as perfect as the digital output. Based on a 10 volt peak-to-peak output (3.53 volts RMS) and a typical noise level of 3.5 millivolts RMS, the throughput signal to noise using the DFM as the input source, will be

65 dB. The addition of the tape recorder noise effort would reduce the overall signal to noise to 55 dB, comparable to the analog multiplex system. In both cases, the deviation ratio is 5. Lowering the deviation ratio to 2 would drop the system signal to noise about 3 to 5 dB. Although most customers use the analog output initially, replacing the old analog system with the newer DFM/DFD system, customers should use the digital output, thereby significantly improving data accuracy.

To ease the interfacing of the demultiplexer output to computers and other digital devices, Metraplex has built into the demodulation system, an output digital multiplexer. This card takes all or selected digital discriminator outputs and multiplexes them into a parallel demand response interface for direct computer input or a serial digital output for transmission over fiber cable, satellite lines, or other communication links to a remote processor. In the case of the computer interface all 16 bits are provided along with an 8 bit channel ID number and a 24 bit time tag developed from the internal clock. The clock can be synchronized to an external time source. The serial data output is truncated to 12 bits and formatted into an IRIG NRZ-L PCM format that can be decoded using a PCM decommutator.

Conclusion

With the advent of digital signal processing, it is possible to build FM multiplexing systems that take advantage of the attributes of digital systems. Gone are the days of system configuration with different sets of plug-in filters and hours of manual adjustments to calibrate systems of many channels. Now, under computer control, all aspects of the system are programmed including, carrier, frequencies, deviations, output filters, input and output voltage levels and the like. Formats can be saved and recalled quickly for changing test conditions. Digital outputs allow the recorded data to be processed immediately by computer without the data degradation and time loss experienced when pre-processing the recorded tapes with an analog-to-digital conversion system.

The new digital FM multiplexing system provides a significant improvement in system accuracy when the user switches over to the digital output of the demodulator. Factory tests, substantiated by customer feedback, show up to a 10 dB improvement in signal noise performance.

Most important of all is this system provides a vehicle for recording large numbers of data channels with bandwidths up to 64 kHz on a single recording media. This capability is not currently available on any digital recording device. The throughput requirements for these systems would approach 400 megabits per second for a digital system. That is why test operations at Pratt & Whitney, General Electric and the U.S. Air Force Arnold Engineering Research Center rely on the DFM/DFD for their jet engine test programs.