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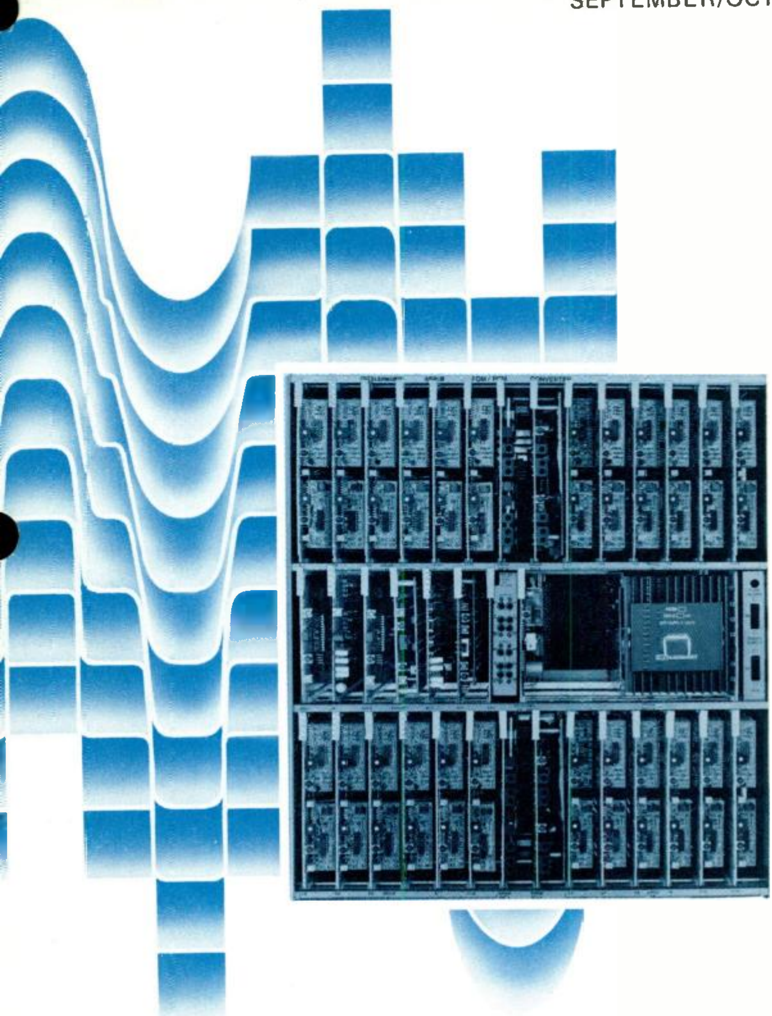

FDM/PCM Conversions

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Despite the rapidly increasing number of digital transmission and switching system installations, the majority of long-haul transmission systems use analog facilities. This fact established a requirement for an efficient, low-cost, reliable method for interfacing frequency division multiplex systems with digital transmission systems or time division switches.

Two devices, FDM/PCM Converters (Conventional Transmultiplexers) and Digital Transmultiplexers, have been developed to meet this requirement. A major feature of these devices is that they enable the user to take advantage of the cost savings realized by digital switches and PABX's without paying the penalty of obsoleting analog transmission facilities. FDM/PCM Converters and Digital Transmultiplexers are the subject of this article.

Digital toll switching offices are being placed in service throughout the United States. These offices switch signals in a time division multiplexed (TDM) form. Analog signals, including frequency division multiplexed (FDM) signals, entering and leaving these offices must be converted to and from digital form.

FDM is used world-wide for long distance telephone. By using single-sideband suppressed carrier (SSSC), amplitude modulation and electrical filtering techniques, FDM systems simultaneously transmit many information channels over a single transmission medium. FDM signals are transmitted at the same time but separated in frequency.

TDM came into general use much later than FDM. Currently, the principal TDM applications are to switching and exchange area trunks. However, the use of TDM for point to point data communications, and other applications, is rapidly increasing.

By using pulse code modulation (PCM), TDM systems transmit many information channels over a single transmission medium. TDM signals are transmitted at the same frequency but separated in time.

Analog signals lend themselves to FDM because they are in the general form of a sine wave. Digital signals lend themselves to TDM because they are in the form of a pulse stream. This is the reason why analog signals must be translated to digital form to use TDM facilities and digital signals must be translated into analog form to use FDM facilities. The translation is commonly referred to as analog to digital and digital to analog conversion. The basic circuits are called A/D or D/A Converters.

Until recently, these translations were accomplished by placing FDM

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channel bank and signaling assemblies back-to-back with PCM channel bank assemblies. Under this method the signals to be converted are demultiplexed down to voice frequencies in their original form, then remultiplexed in the other form.

Today, new devices are available which accomplish these translations much more economically. These devices are known as FDM/PCM Converters or Transmultiplexers. Two design approaches, conventional and digital are used. As its name implies the conventional approach uses the same basic idea as back-to-back channel banks. The analog/digital interface is at voice frequencies (vf).

The digital approach uses all digital circuitry with the exception of the digital to analog (D/A) and analog to digital (A/D) converters for signal processing. This processing is at high frequencies (HF). The balance of this article discusses the conventional and digital approaches and compares them with each other and the back-to-back channel bank arrangement.

Conventional Approach

Examples of the conventional approach are the Bell LT-1 and LT-1B types and GTE Lenkurt 4691A and 4691B FDM/PCM Converters. The 4691A converts two 12 channel FDM groups to one 24 channel PCM digroup. The 4691A provides an optional 3825 Hz out-of-band to PCM signaling conversion. Aside from these differences the 4691A is essentially the same as the 4691B described in the following paragraphs.

The 4691B FDM/PCM Converter is a conventional transmultiplexer that terminates four 60-108 kHz groups of twelve Frequency Division Multiplex (FDM) voice channels and bilaterally converts them to either two 24 channel Pulse Code Modulation (PCM) digroups for T1 transmission or one 48 channel double digroup for T1C transmission. In addition, information conveyed by the per channel 2600 Hz signaling tones is converted to and from the PCM signaling channel.

The signaling and maintenance features are such that the 4691B may be locally or remotely connected to a digital switch or connected via T1 or T1C type repeatered lines or digital radio to any D3/D4 compatible channel bank operating in mode 1, 2 or 3.

At this time, the conventional approach is more attractive than the digital approach for a number of reasons. This is not to say that the digital approach does not show future promise.

A comparison of a number of factors showed the conventional approach allows substantial improvements over back-to-back FDM and PCM channel banks and permits a number of important application features to be provided. These are:

1. PCM transmission in modes 1, 2 & 3 available because of the use of existing modules.
2. Individual pads for Inserted Connection Loss (ICL) selection and compensation for attenuation variations across the group band. The former is provided as a prescription adjustment and the latter adjusted to facility measurement.
3. Independent FDM and PCM facility synchronization can be maintained. The FDM modulation and demodulation is separately synchronized to a line pilot and the PCM equipment to the bit rate of the digital switch.

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4. The absolute delay of the filters and codec is low compared to the sample, process and forward time of an all digital approach.

On the FDM side, many of the circuits and components were derived from the GTE Lenkurt 46A3/46A6 FDM Multiplex. These include the Polylithic crystal channel filters and the active double balanced modulator. The 8192 kHz carrier, 104.08 kHz pilot and 2600 Hz signaling tones are derived from existing equipment.

The PCM technology was derived from the 9004A Channel Bank. This includes the active channel filters to further limit bandwidth before and after sampling, the codec per channel, the microprocessor for PCM Alarm and TMB sequences, and the Transmit, Receive and Syndes units.

4691B Equipment

The equipment arrangement of the 4691B double digroup is shown in Figure 1. All the equipment required to convert 48 analog channels (4 groups) into 48 digital channels (2 digroups) is contained in a 19.25" H x 19" W x 12" D 11 mtg. space shelf. The shelf consists of two identical card files for channel units and FDM group units and one for all of the analog and digital units common to 48 channels. This arrangement permits up to 288 channels in an 11 ft. 6 in. rack and 192 channels in an 8 foot rack, including room for frequency generation and accessory equipment. The channel units are designed for two channels per printed circuit card.

Common Equipment

Ten common equipment units (Figure 2) are used in the 4691B. In the signal path on the FDM side, the Group Receive Units and Group Transmit Units each process two groups. These provide optional group pilot detection and insertion, office cable build-out, pilot muting, channel splitting and combining, and full test jack access to each group.

The Group Carrier and Pilot Amplifier Unit buffers the external 8192 kHz reference frequency and 2600 Hz signaling frequency for application to all channel units. It also contains a 104.08 kHz group pilot amplifier. This output is usually applied to the two dual Group Transmit Units for group transmission on the FDM path. Muting of the group pilot is provided for far end Carrier Group Alarm indication in 46A. Input and output ports are provided to interface Western Electric carrier failure alarm (CFA) equipment, where the FDM facility connects to L-carrier at the far end.

The PCM side has a selection of units depending on the mode of transmission. The mode 3 Transmit and Receive Units provide two DS-1 (1.544 Mb/s) signals through the Digital Access Unit (DAU) to the associated digital switching or transmission equipment. The mode 2 Transmit and Receive Units are connected to the Syndes Unit and then to one half of the DAU. The mode 1 Transmit and Receive Units synchronously combine and divide the two digroups. They also connect directly to one half of the DAU. Modes 1 and 2 interface DS-1C facilities operating at 3.152 Mb/s.

The DAU provides bridging or splitting jack access to the DS-1 or DS-1C signals in both directions. It also provides for DS-1/DS-1C cable loss pads and equalizers. All test level adjustments are performed via the DAU using either a Western Electric

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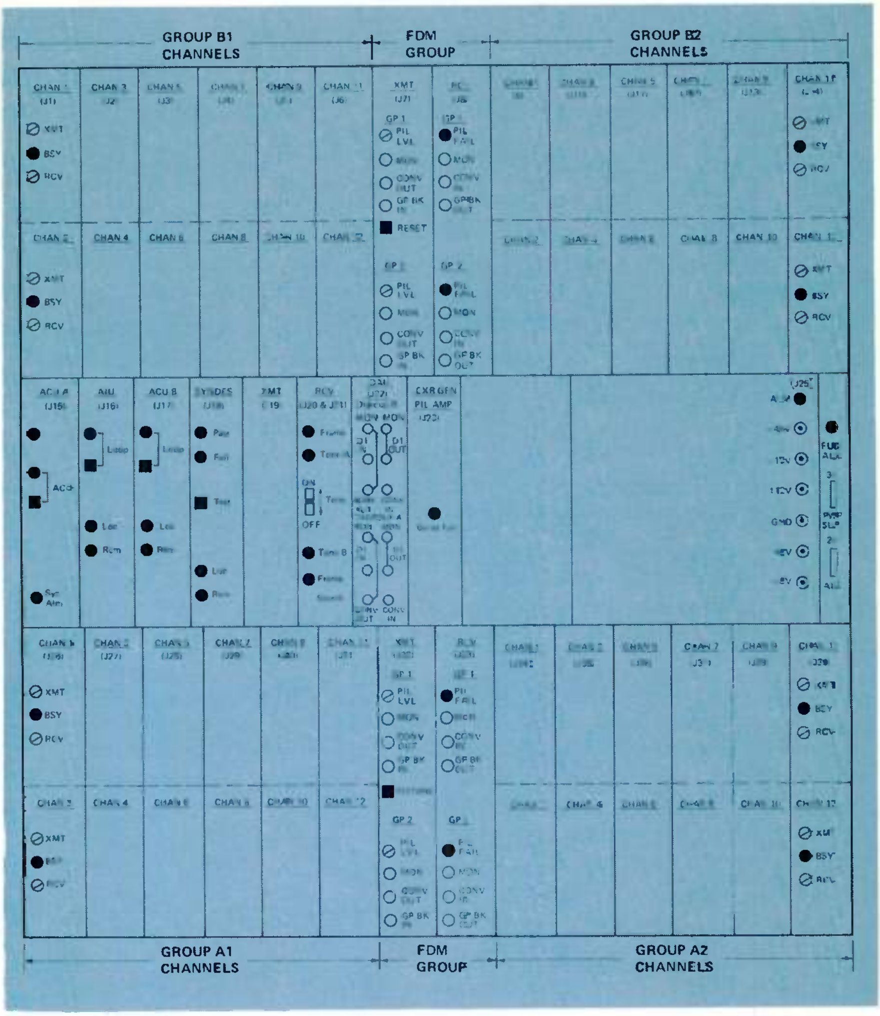

Figure 1. FDM/PCM Converter Shelf Profile.

Data Access Timeslot Selector (DATS) test set or a simplified equivalent offered by GTE Lenkurt. No other voice frequency access is necessary. Aligning-end-to-end levels in this way avoids an accumulation of level errors in the converter.

One Alarm Control Unit is provided for mode 1 and two for modes 2 and 3 (one for each digroup). These units provide PCM local and remote alarm indications. Upon receipt of a local out-of-frame indication from the Receive Unit, the Alarm Control Unit initiates all required delays to cause alarm conditions at both ends of the circuit and for simultaneous restoral. The logic and delays are implemented in a single chip, 8-bit, mask-programmed, microcomputer. This chip is fabricated in N-channel silicon gate MOS and contains a 1k x 8

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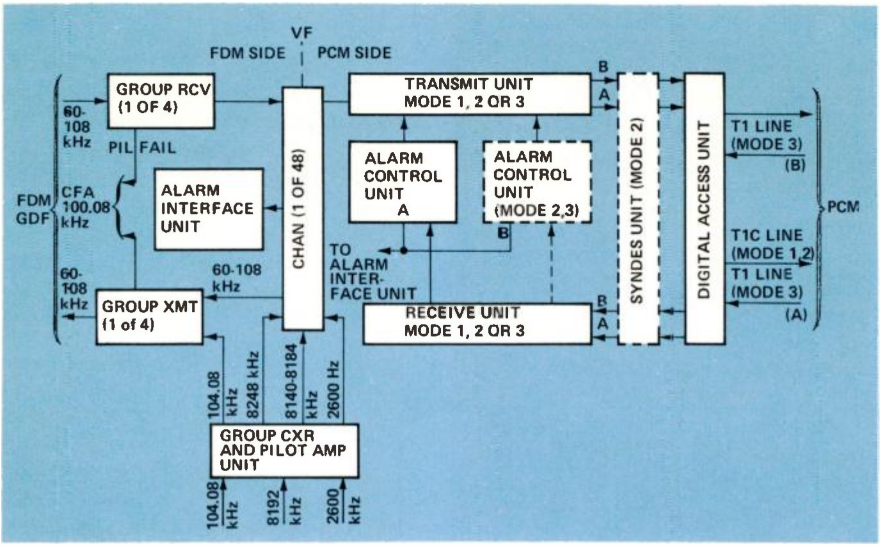

Figure 2. Simplified Block Diagram.

program memory, a 64 x 8 oscillator and clock circuit.

The Alarm Interface Unit (AIU) provides diagnostic and control features for remote, central-service facilities as well as conventional local alarm systems. The remote extensions may be via limited range metallic facilities or an Alarm and Control system such as the GTE Lenkurt System 51. A summary of the functions are:

1. One set of alarm contacts and one sense-point for loss of pilot for each of the 12 channel groups, to condition a switch or provide network management information.
2. Remote reset of group pilot when lost from failure or looped mute.
3. Individual power and carrier supply failure indicators.
4. Individual digital local (AR) and remote (AY) alarms to indicate a digital facility loss.
5. Loop back control for each digroup.
6. Loop back indicator for each digroup.
7. Individual analog and digital summary alarms for shelf or assembly.
8. Major and minor alarms for shelf or assembly including accessory equipment mounted in the same assembly.

These 4691B alarms may be remotely announced either for assembly, system or network components.

Channel Unit VF Path

The channel unit (Figure 3) is a dual unit which performs all channel functions for two complete channels on one printed circuit card.

The channel unit can be divided into four basic functions; analog modulation and channel filtering, signaling, VF filtering and the coder — decoder (codec). In the A to D (receive) direction, each channel circuit accepts the 60-108 kHz group, modulates it to the 8140 to 8188 kHz

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range with an 8248 kHz carrier, filters out the desired channel frequencies and demodulates to voice frequency with a carrier of appropriate frequency. For example, the channel 1 sideband is selected by a Polylithic filter passing 8140-8144 kHz. The output of this filter is demodulated to VF with an 8140 kHz carrier selected from a 8140-8184 kHz spectrum generated from the 8192 kHz, FDM-reference frequency supply. The VF is applied to the ICL prescription attenuator for setting trunk loss, then to the delay equalizer, active lowpass filter, codec and Tansmit Unit PCM input bus.

In the D to A (transmit) direction, the reverse process takes place. The channel 1 codec output is applied to an active lowpass filter, delay-equalizer amplifier, modulator, then to the channel-1, 8140-8144 kHz Polylithic filter where the upper sideband is selected. The 8140 kHz carrier used for this modulation step is the same as that used on the receive side.

The filter output is then modulated with an 8248 kHz carrier to place it at 104 to 108 kHz. The 12 channel outputs are summed in the Group Transmit Unit to form the 60 to 108 kHz group. The 104.08 kHz pilot is added at this point and attenuators are provided for setting the correct level at the Group Distribution Frame or Group Equipment.

The modulator used in the four functions described above is an integrated circuit, implemented in a bipolar process. The codec is a single chip N MOS device with an internal voltage reference and transmit/receive signaling I/O ports.

Signaling Pulse Correction and TMB

The signaling function performed is to interface the SF tone signaling on the FDM side with the PCM side A&B signaling channels. In the A to D direction, after the voice frequency signals have been demodulated as described above, they are passed through an active bandpass/bandstop

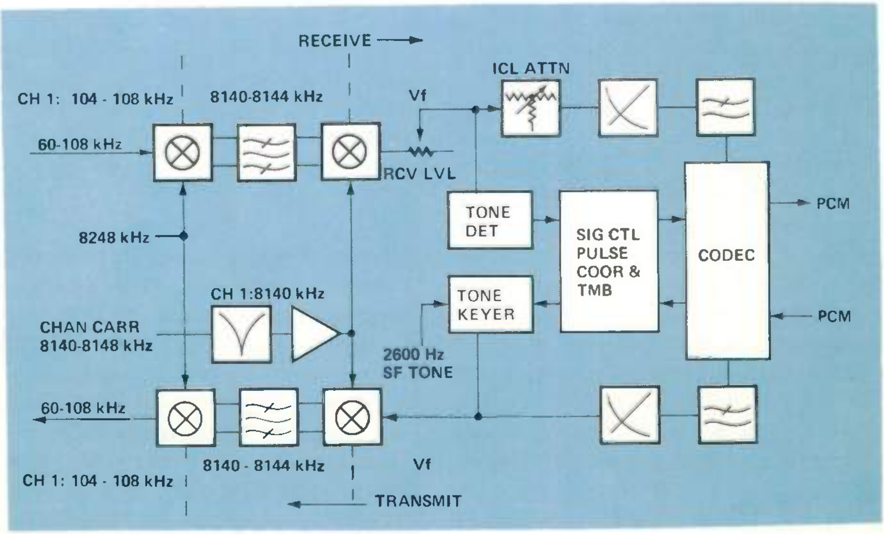

Figure 3. 1/2 E&M Channel Unit.

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filter and to a sig/guard detector which detects presence or absence of 2600 Hz tone. The guard circuit prevents talk-down by weighting the detected tone level against the level of other signals in the remainder of the VF band.

The output of the sig/guard detector is passed through the pulse corrector to the signaling input of the codec which inserts the detected signaling state into the A&B signaling bit positions for that channel. The pulse correction is performed by a single-chip, low-power N MOS microcontroller per channel. Clocking for the microcontroller is provided by the 1.544 Mb/s receive clock and no external components are required.

The microcontroller provides all logic and delays required for supervision and pulse correction in both directions including signaling filter control, guard control, tone on/off, tone enhance, line cut and busy LED driver functions. Split integration is provided with a 2:1 integration ratio. Transmit and receive pulse correction are independent of each other.

In addition to the above, the microcontroller performs a Trunk-Make-Busy (TMB) function which is required when the converter is not colocated with the switch. The alarms on the two sides are not passed through but are used to busy the outgoing channels on the un-alarmed side. For example, if a failure is detected on group A1 on the FDM side these 12 channels will be busied outgoing on the PCM side. If digroup A detects an alarm on the PCM side then the channels in both groups A1 and A2 will busied on the FDM side. Options are provided to force an on-hook or an on-hook followed by an off-hook after disconnect.

These choices are dictated by the trunk assignment, i.e. whether it is an outgoing or two-way trunk or an incoming trunk. The signaling may be completely disabled by means of a strap option to permit the E&M unit to be used in common channel, interoffice signaling (CCIS) applications.

The microcontroller is a control-oriented processor, of N channel silicon gate MOS construction, having a 1k x 8 mask-programmed ROM, a 64 x 4 RAM and an additional internal time-base counter. Its low power consumption makes it particularly applicable to a per-channel function.

The software consists of a single continuous loop with all paths through the program adjusted with no-op instructions to equal 127 instruction cycles. This requires 2.632 milliseconds. There are no subroutines and all jumps are forward. Timing is referenced to the number of times through the loop after an event or to the internal hardware timer.

Initially, dual channel units are being offered with the E&M SF tone signaling either equipped or deleted. Future dual channel units are being planned to accommodate Foreign Exchange and special service applications.

Performance

The VF performance is better than equivalent FDM and PCM channel banks back-to-back and exceeds the requirements proposed in CCITT "General Considerations on Transmultiplexing Equipments." The performance is comparable to the typical values given for the LT-1 in AT&T Technical Advisory No. 39, Issue 1, dated March 15, 1979. The in-band

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signaling performance meets all distortion, level and timing requirements for end-to-end compatibility with E&F type signaling as defined in AT&T Technical Advisory No. 39.

The –48 Vdc office battery power required, when 48 E&M channels are equipped, is approximately 1.5 watts per channel.

First Installation

Figure 4 is a block diagram of a radio office interface between Bell's long-haul, FDM carrier equipment and General Telephone of Wisconsin's Digital Number 3 EAX Switch at Wausau, Wisconsin. The interface is located at Rib Mountain and is the first 4691B installation.

Seventeen 4691B FDM/PCM Converters were placed in service in mid-June 1980. They provide an interface for 64 message, two data and two special service groups.

Operations between the number 3 EAX office at Wausau and various cities throughout the United States, including Atlanta, Ga., Chicago, Il. and New York, N.Y., have exhibited excellent performance. The advantages of the design approach have been confirmed by these operations.

Perhaps the greatest advantage of the conventional approach, used in the 4691B design, is that it makes maximum use of existing, reliable, field proven components, processes and firmware. The approach also permits the inclusion of a number of important application features. These are:

1. PCM transmission in modes 1, 2, & 3 available because of the use of existing common equipment modules.
2. Individual pads are provided for Inserted Connection Loss (ICL) selection and compensation for attenuation variations across the group band. The former is provided as a prescription adjustment and the latter adjusted to facility measurement.
3. Independent FDM and PCM facility synchronization can be maintained. The FDM modulation and demodulation is separately synchronized to a line pilot and the PCM equipment to the bit rate of the digital switch.
4. The absolute delay of the filters and codec is low compared to the sample, process and forward time of an all digital approach.

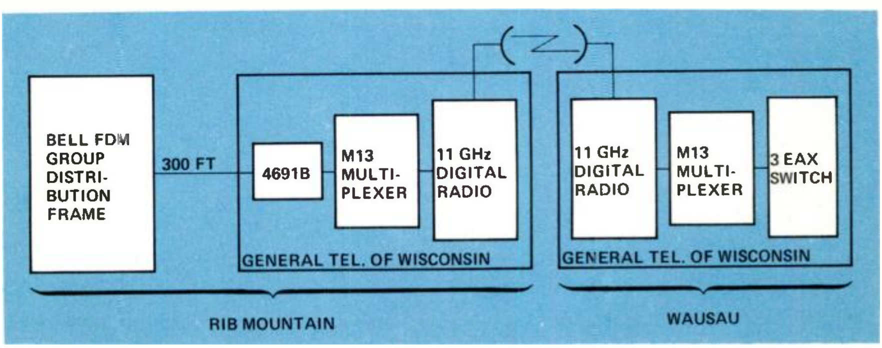

Figure 4. 4691B FDM/PCM Converter used in interface between Bell's FDM long haul carrier equipment and General Telephone of Wisconsin's No. 3 EAX digital switch in Wausau, Wisconsin.

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Compared to the back-to-back channel banks, the 4691B FDM/PCM Converter provides 38 to 45% initial cost savings. Space requirements are reduced 64 to 71% and power requirements are reduced 51 to 67%. Maintenance requirements are also reduced and the FDM/PCM Converter improves performance and increases reliability.

Figures 5 through 7 show the 4691B interfacing between various FDM and digital facilities. Figure 8 shows the 4691B interfacing between FDM and Fiber Optics facilities.

Digital Approach

The digital approach to trans-multiplexing converts SSB FDM signals to PCM TDM signals and

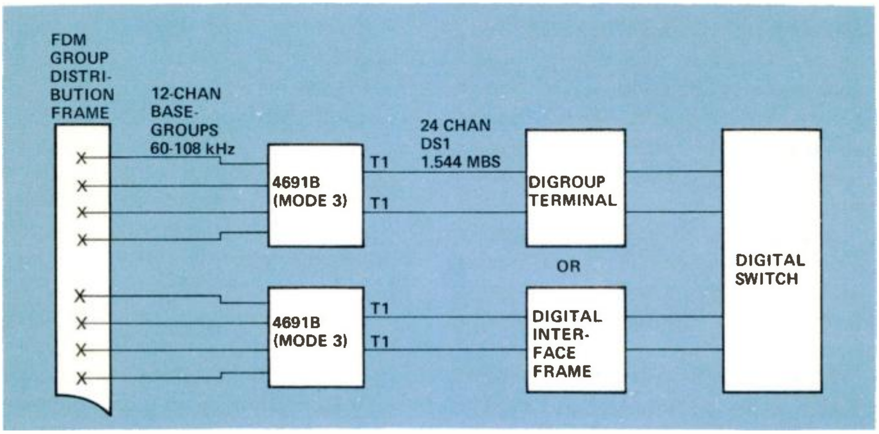

Figure 5. 4691B Interface Between a Digital Switch and an FDM Group Distribution Frame. The digroup terminal or digital interface frame may be arranged for DS1, DS1C, or DS2 input signals.

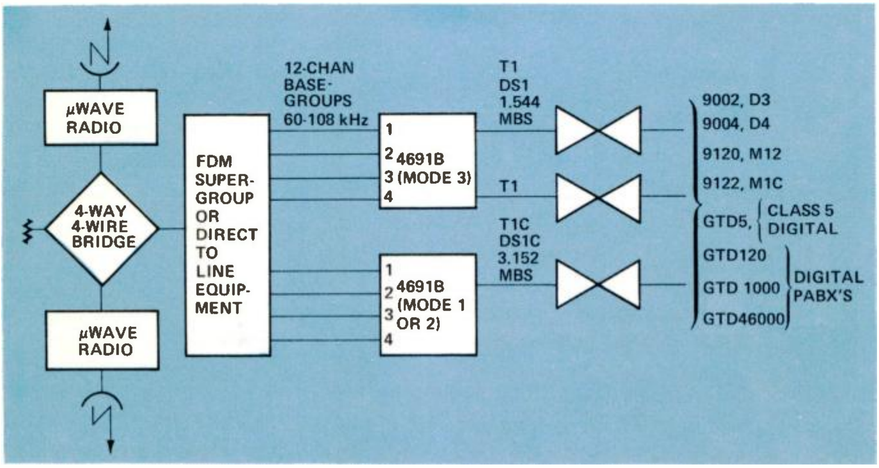

Figure 6. 4691B Interface Between FDM Microwave and 9104A Repeatered Line.

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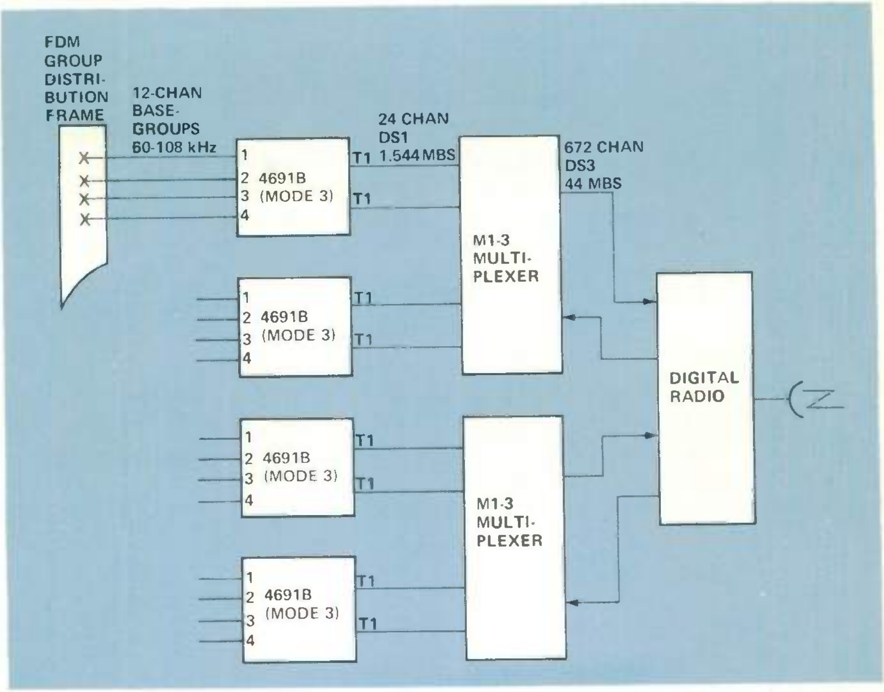

Figure 7. 4691B Interface Between Digital Microwave radio and FDM group distribution frame.

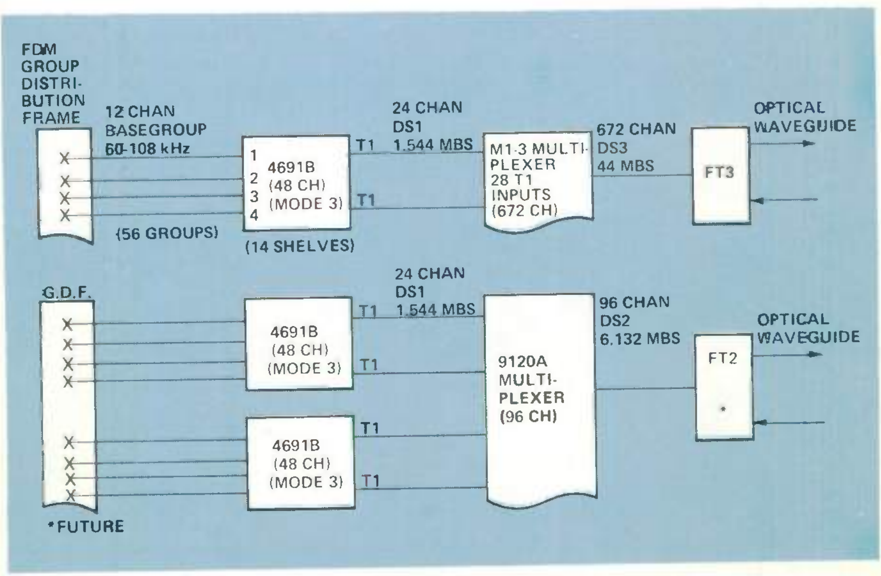

Figure 8. 4691B Interface Between Fiber Optic Systems and FDM group distribution frames.

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PCM TDM signals to SSB FDM signals at multiplex level. This is in contrast to the conventional approach which demultiplexes the signals to vf level in their original form then remultiplexes them in the other form.

A digital transmultiplexer which is currently available performs the conversions at the basic, 60-108 kHz, group level. The translation is between two 12 channel FDM groups and a North American, 24 channel TDM unit.

A transmultiplexer has also been developed which performs the conversions at supergroup level. The translation is between the 60 channel supergroup and two 30 channel TDM units. This transmultiplexer best fits the TDM hierachy used in Western Europe. Since both types use the same basic principles, our discussion is limited to the North American unit.

The transmultiplexer to FDM interface is at group level, 60 to 108 kHz. The transmultiplexer to PCM interface is at the standard DS1 level. Figure 9 is a block diagram of the system.

FDM To PCM

Referring to the figure, Group 1 and Group 2 FDM signals are connected, through separate input transformers, into the HF interface and pass through adjustable attenuators and bandpass filters. The filters pass only signals in the 60 to 108 kHz spectrum. Their outputs connect to the A/D converter inputs.

The two groups are amplified by separate preamplifiers and passed to two sample and hold circuits. The sample and hold circuits are simultaneously clocked at a 112 kHz rate by a line receiver driven by an 112 kHz trigger.

The sample and hold circuits connect to separate, 14-bit A/D converters. These converters are also simultaneously clocked by differential line receivers driven by the same trigger. The A/D converter's outputs connect to a 2 input, 1 output multiplexer which is clocked by a differential line receiver driven at a 112 kHz rate.

The multiplexer alternately passes 14-bits from each A/D converter. The multiplexer outputs are con-

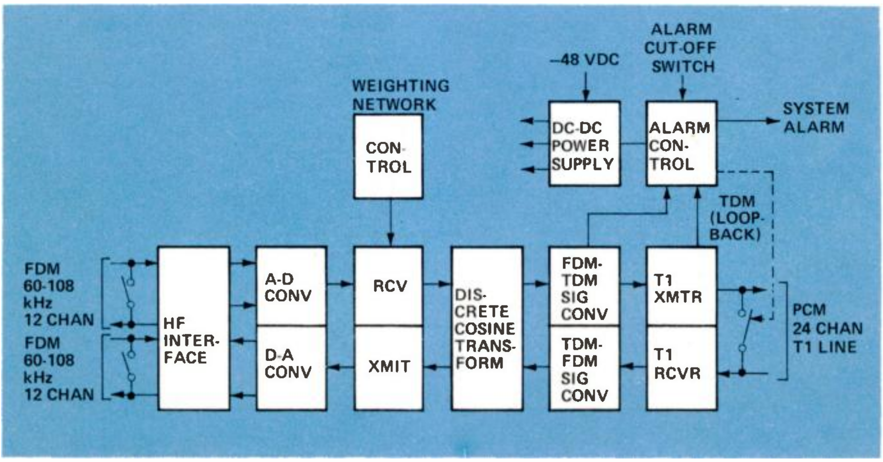

Figure 9. Block Diagram of Digital Transmultiplexer.

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nected to 14 line drivers. The line driver outputs are in a 14-bit parallel format and are transferred at a 224 kHz rate to the Receive Weighting Network. Each set of bits represents alternate basic FDM groups.

The Weighting Network takes each basic 60 to 108 kHz group and translates its 12 channels down to vf. The output signals, in a 16-bit parallel format, are connected to the Discrete Cosine Transform unit. This unit digitally demultiplexes the signals into 24 voice channels. Its 14-bit, parallel output is sent to the FDM-TDM Signal Converter.

The Signal Converter detects the 2600 Hz, inband, SF signaling tone. The detector output is sent to the FDM-TDM Signal Processor to be processed. The 2600 Hz tone is filtered out before the signal is sent to the T1 Transmit Unit.

The T1 Transmit unit converts the 14-bit code into an 8-bit, μ255 code. Multiplexers, combiners, signaling and framing generators convert the vf and signaling information into the T1 format at a 1.544 Mb/s line rate. A unipolar to bipolar converter converts the pulse train from unipolar to bipolar. The Transmit unit interfaces with the T1 line.

PCM To FDM

In the PCM to FDM direction, the 24 channel, T1 line signals enter the T1 receiver and are converted from a bipolar to a unipolar pulse train. The timing, framing and signaling information is separated and the 8-bit vf channel signal is expanded into a parallel, 14-bit linear code.

All of this information is inputted to the TDM/FDM Signal Converter. The TDM/FDM signaling converter is time shared between the 24 voice channels. Circuits in the converter digitally create a 2600 Hz signaling tone in response to the signaling bit from the PCM line. The signaling tone is transmitted at −8 dBm0 for dialing pulses and at −20 dBm0 for idle channel tone. The voice information is either passed or interrupted, depending upon the signaling state. This action improves the reliability of the signaling in the PCM to FDM path.

The timing and control of the signaling and voice interruption is controlled by a microprocessor. The microprocessor also controls the operation of the FDM to TDM Signal Converter. This unit, combined with the TDM to FDM Converter, constitutes the digital equivalent of an Analog SF Signal Converter.

The output of the TDM/FDM Converter is connected to the input of the Discrete Cosine Transform. This unit digitally multiplexes the voice data from the PCM line into two 12 channel groups. In the TDM/FDM direction the output of the Discrete Cosine Transform unit is connected to the input of the Transmit Weighting Network.

The signal at the input of the Transmit Weighting Network is in a 16-bit parallel format at a 224 kHz rate. Each successive set of bits represents an alternate basic group. If the first set of 16 bits represents group 1, then the second set represents group 2, the third set again represents group 1 and so on.

The data for group 1 is clocked into a 16 bit parallel register "A". The next set of data representing group 2 is clocked into register A while the contents of A (representing group 1) is clocked into register B. The contents of register B are then clocked into storage register C while the con-

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tents of A are clocked into storage register D.

The Weighting Network frequency translates the vf channels into the twelve 4 kHz channels in the 60 to 108 kHz spectrum. The 16 data bits in storage register C are sent along with 16 data bits from a coefficient memory into a 16x16 multiplier. The multiplier product is connected through an output buffer to an adder and overflow correction circuit. This circuit provides the input to a 16-bit output register.

The output register provides input signals to 16 differential line drivers. The line driver outputs are connected to the input of the D/A converter. The operation of the Transmit Weighting Network unit is controlled by address and timing functions generated in the Weighting Network Control unit, which also controls the Weighting Network Receive unit.

The signals from the Transmit Weighting Network to the D/A converter are in a 14-bit parallel format along with two control bits at a 112 kHz rate. They are inputted to 3-level differential line receivers. The outputs from 14 line receivers are inputted into 2 14-bit parallel registers. The registers are clocked by the two control bit outputs from their corresponding receivers. The output from each register is connected to the input of a separate D/A converter.

At this point, one of the sampled, 12 channel groups is in the 4 to 52 kHz band and the other is in the 60 to 108 kHz band. The signals in the 4 to 52 kHz band are used to modulate the 112 kHz carrier. The modulating process produces an amplitude modulated signal with lower sidebands in the 60 to 108 kHz spectrum (112-52 = 60 kHz; 112-4 = 108 kHz). The chopped, vf channel signals at the D/A converter output are connected to the input of the HF Interface Transmit/Receive Unit.

A bandpass filter at the input of the HF Transmit Interface passes only those signals in the 60 to 108 kHz spectrum. The filter also integrates the chopped vf signal pulses to form an analog signal representing the original speech inputs. The group 1 signals are coupled by an output transformer to an HF line. The group 2 signals are coupled by a separate output transformer to an HF line.

Comparisons

The all digital transmultiplexer we have described is cost competitive with back-to-back terminal arrangements but not with FDM/PCM converter arrangements such as the 4691B. It also requires substantially more input power than the 4691B (about 150 watts for 24 channels as opposed to about 100 watts for 48 channels).

The 24 channel all digital system, without a power converter, occupies about 6 vertical mounting spaces on a standard rack. The 48 channel 4691B requires 11. The dissipation of 150 watts in a 6 mounting space area could cause heating problems, if ventilating fans or similar measures are not provided.

The all digital system does not provide a method for adjusting individual channel levels, to correct for transmission loss variations within the 60 to 108 kHz output group. These variations may require adjustments that exceed 1 dB. Network requirements specify that the levels be adjusted within 0.1 dB of the preset values. This specification is difficult to meet without channel level adjustments.

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As stated earlier in this article, the 4691B has adjustments for transmission loss variations across the group band. It also has pads for inserted connection loss selection.

The all digital transmultiplexer used the 1.544 megabit, T-line frequency to demodulate the FDM signal. If there is a frequency error in the FDM signal or PCM bit rate, this will show up in the vf channel as a frequency shift.

The demodulating process needs a frequency synchronized to the incoming FDM signal to remove frequency errors. The 4691B carriers are derived from the FDM master frequency generator or line pilot and are fully synchronized.

As previously stated, the absolute delay time of the all digital approach is long compared to that of the 4691B. A long, absolute-delay time may cause echo problems. Also, the digital transmultiplexer cannot be used in non-colocated applications because it lacks trunk-make-busy (TMB) functions.

The all digital configuration we have described provides transmultiplexing of 24 voice channels whereas the 4691B provides 48. Two digital transmultiplexers could provide 48 channels in essentially the same rack space but at a substantially greater cost and power consumption.

Conclusion

From the foregoing comparisons, it is apparent that the all digital transmultiplexer is not presently competitive with the FDM/PCM Converter. However, the all digital approach is based on valid design principles.

The hardware and technology used in the FDM/PCM Converter has been refined through several generations of FDM and PCM systems. At some future date, when the all digital transmultiplexer has gone through similar hardware and technological refinements, it may equal or surpass the FDM/PCM Converter in both cost and performance.

Meanwhile, the telephone network evolution, from analog to digital, is continuing. Devices like the conventional and digital transmultiplexers are providing an orderly, economical transition without interrupting service. The Demodulator will continue to monitor the evolution and report its progress from time to time.

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