Selection of Pilot Relaying Communication Channels – A Case Study

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ABSTRACT

Communications channels and channel devices for Pilot Relaying channels traditionally reside in the Relay Engineers' domain. Today, many conventional communication channels are being replaced by digital communications networks, which are governed by other departments in the utility. The use of these channels for Relaying requires the Protective Relay Engineers to have a working knowledge of digital communications. Many Relay Engineers are responsible for specifying and/or designing Protection Schemes that include Pilot Communication channels. Some Protection Engineers have little expertise in the area of Communications making it difficult to select or specify an acceptable criterion for a reliable channel in terms that are understood by the Communications Engineers.

This paper will look at how one Utility evaluated their options for Pilot Relaying Communications channels. It will cover their selection criteria and the qualities they require in the products they use. It will address reliability concerns from substation environment to delay characteristics of communications equipment. The paper is deigned to enlighten the Protective Relay Engineer in the area of communications as applied to protective relaying and the considerations one Utility used to evaluate their choices.

DUKE POWER BACKGROUND

In 2002, Duke Power engineers worked on the design for a relay and control upgrade for Tiger Tie Station, which is a major transmission station.

The station consists of:

1. Seven (7) 230 kV tie lines.
2. Four (4) 230/100/44 kV autotransformer banks.
3. Twelve (12) 100 kV tie lines.
4. Two (2) 100 kV radial lines.
5. Two (2) 100 kV, 79.2 MVAR capacitor banks.
6. Four (4) 44 kV lines.

The upgrade solution consisted of a new drop-in control house following Duke Power's substation automation strategy. All cable was to be replaced along with yard boxes, and auxiliary yard equipment. The automation solution's communications needs included a WAN (Wide Area Network) connection to the main office, a DNP SCADA link to the Transmission Control Center and network phone lines.

For the Relay engineers, the seven (7) 230 kV tie lines were of particular interest. The seven (7) lines go to four (4) different substations and all were installed in the 1950s and 1960s. They all were protected with Directional Comparison Blocking (DCB) power line carrier schemes. The coupling equipment (coupling capacitors, tuners and wave traps) was the originally installed equipment and previous false trips had been attributed to equipment problems. If the carrier were to remain then significant dollars would have to be spent replacing the coupling equipment. A more secure Permissive Overreaching Transfer Trip (POTT) scheme using RS-232 relay-to-relay communications was thought to be a good solution if the required communications medium could be obtained. Some of the 100 kV lines also used the DCB pilot scheme and two (2) of the 100 kV lines also used an Audio Tone Direct Transfer Trip over leased phone lines to a high profile major customer with generation.

Duke Power Communications Infrastructure

Duke Power Relay engineers contacted Duke Power Telecommunications Engineering to determine if communications facilities existed that could be used for the 230 kV line relaying as well as providing the needs of the automation solution. The initial meeting was used to discuss the existing communications infrastructure and the total communications needs of the project.

It was quickly determined that there weren't company telecommunications facilities in place that would benefit the 100 kV lines and there was no clear economic justification for adding them at that time. The 230 kV lines did hold some possibilities. All of the 230 kV lines involved in this project had OPGW (Optical Ground Wire) installed forming an OC3 ring (to be described later) and there was multiplexing equipment at three of the four remote stations (North Greenville Tie, Pacolet Tie and Peach Valley Tie) along with Tiger Tie itself. The fiber had been installed for company needs as well as for commercial traffic.

While the fiber did pass through the fourth station (Shiloh Switching Station), none of the fibers were dropped there and there was no multiplexing equipment. All of that station's communications needs were being met using leased phone circuits and all pilot line relaying utilized power line carrier. The Telecommunications engineers did show an interest in dropping fibers for relaying use and station communication at Shiloh.

UNDERSTANDING COMMUNICATION NETWORK COMPONENTS

The Duke Power Communications Department utilizes several components that may or may not be familiar to a Relay Engineer. In order to understand the network that Duke Power deployed, it is important to have a basic understanding of the technology used. The following is a list of components with descriptions deployed by Duke Power with more emphasis put on the T1 Multiplexers since it was those units that became a point of contention in this paper.

SONET

SONET (Synchronous Optical NETwork) is the American National Standards standard for synchronous data transmission on optical media. The international equivalent of SONET is SDH, Synchronous Digital Hierarchy. Together they ensure that digital networks can interconnect internationally and that existing conventional transmission systems can take advantage of optical media through so called "tributary" interfaces. SONET features are:

- Application, service and protocol transparency. There is no service, protocol, nor application which cannot ride on the SONET backbone network.

- SONET is relatively inexpensive when compared to Asynchronous Transfer Mode (ATM) equipment for multimedia networks. Equipment costs can be as much as 60% less than an ATM equivalent product.

- An open systems optical architecture for Multi-Vendor Inter-working

One of the main drivers behind SONET is the Multi-Vendor interoperability. Earlier, standards existed for the electrical level only, and to connect to another manufacturer's equipment, back-to-back connected devices were required. With SONET establishing standards for the optical signal levels, a change of equipment can be made "mid-span', i.e. one vendor's multiplexer can be connected to the fiber network at one terminal and another vendor's multiplexer connects to the same fiber at the other end.

Some of the most common SONET (and SDH) applications include transport for all voice services, Internet access, frame relay access, ATM transport, cellular/PCS cell site transport, inter-office trunking, private backbone networks, metropolitan area networks and more. SONET operates today as the backbone for most, if not all, interoffice trunking as well as trans-national, and trans-continental communications.

Listed below are the levels (bandwidths) in the SONET standard and how these relate to the North American Digital Hierarchy. STS-n is the Synchronous Transport Level where 'n' signifies the level (or bandwidth). STS is the electrical signal rate used within SONET prior to its optical conversion. OC-n is Optical Carrier Level where, again, 'n' is the bandwidth level. OC is the optical signal. DS stands for Digital Signal Level. DS-0 is one voice channel and occupies 64 kbps. DS-1, also called T1, is 24 DS-0's or 1.544 Mbps.

SONET Synchronous Rates and Formats (North America)

LEVEL/INTFC	BIT RATE (Mb/s)	VOICE CHANNELS	# DS1's
VT1.5 *	1.728	24	1
STS-1**	51.84	672	28
EC-1	51.84	672	28
OC-1	51.84	672	28
OC-3	155.52	2,016	84
OC-12	622.08	8,064	336
OC-48	2,488.32	32,256	1,344
OC-192	9,953.28	129,024	5,376

* = Internal Level: "encapsulates" a DS1
** = Internal Level: "native" or can "encapsulate" a DS3

OC = Optical Carrier
EC = Electrical Carrier
STS-1 = Synchronous Transport Signal

NOTE: SONET Levels 9, 18, 24, and 36 are specified but not in common use in North America

Figure 1. SONET Rates

SONET Network Components

A typical arrangement of equipment used to interface relaying to a SONET network is shown in Figure 2.

Typically, the substation devices are connected to a T1 multiplexer, which in its turn connect to a SONET multiplexer. The T1 multiplexer provides various interfaces; synchronous 64 kbps for current differential relaying, RS-232 asynchronous data ports, contact interface inputs, and voice and data ports.

Note that the multiplexer needs to be equipped with a channel card, that is suitable for the type of application device that is being used, e.g. relay, status, voice, etc. Luckily, multiplexers are very flexible and channel cards can be added or exchanged in an existing installation as needed.

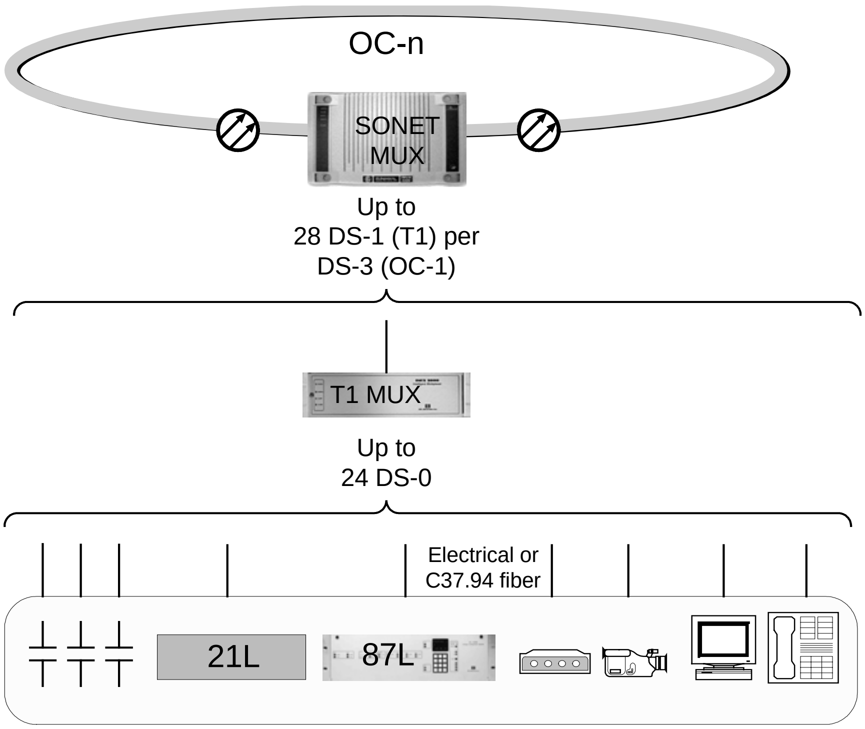

Figure 2. Network components

SONET Topology

Nodes, or terminals, in a SONET network are often arranged in rings to provide an alternate route or protected path in case of a fiber cut or failure of a node. Ring networks are less expensive than point - to - multipoint or star configured fiber optic networks since only two fibers (versus two for each location) are required to support all users or network elements on the ring. Traffic can be routed in either direction around the SONET ring. In case the primary path is cut, traffic is very quickly re-routed to the secondary path.

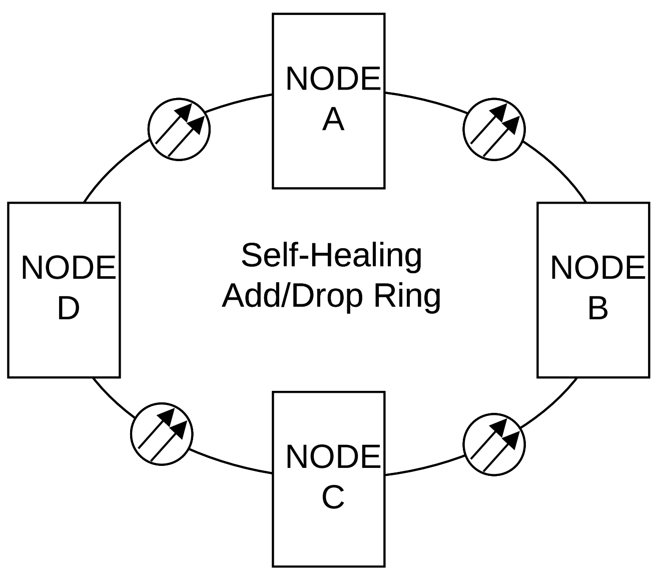

Figure 3. SONET ring topology

Often several topologies are combined in one network and interconnected. As the communications network does not take the same routing as the power system lines, two relays that are not physically distant might have a long communications link over the network. Delays imposed by each node the signal has to pass through need to be taken into account when checking possible channel delays.

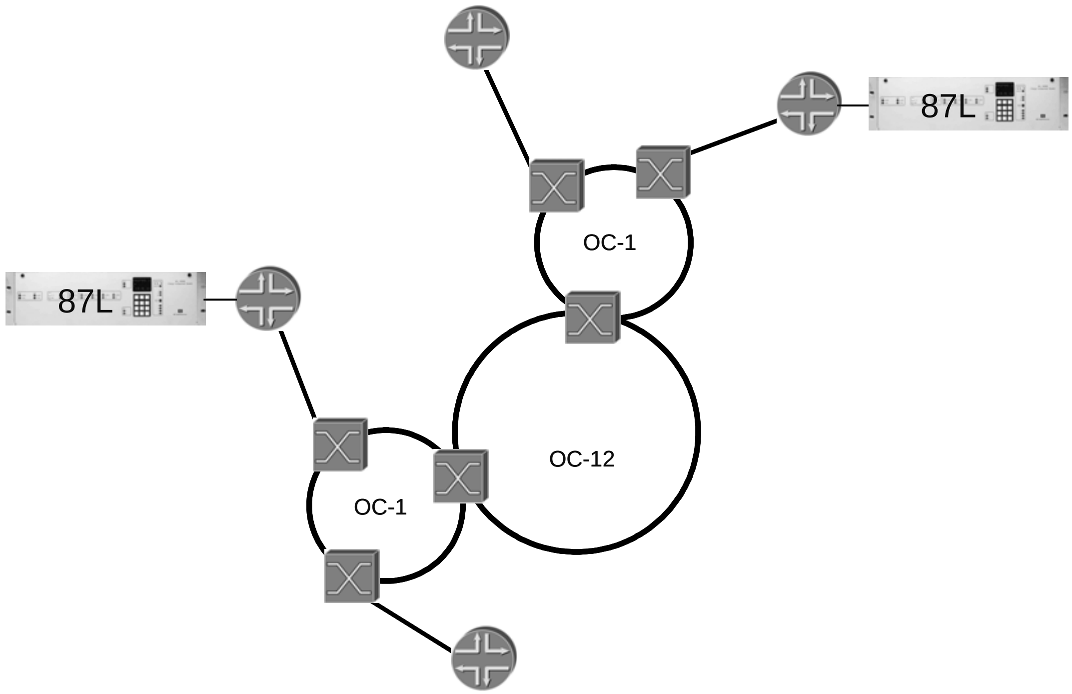

Figure 4. SONET network

CHANNEL BANKS

A channel bank is the foundation for all digital telecommunication transmissions. It is the part of a carrier -multiplex terminal that multiplexes a group of channels into a higher bit-rate digital channel and de-multiplexes these aggregates back into individual channels. A channel bank changes analog voice and data signals into a digital format. It is called a "bank" because it can contain enough processing power to convert a bank of up to 24 individual channels to a digital format, and then back to analog again. The 24 channels comprise a T1 circuit.

In other words, a channel bank to a Telecommunications Engineer is what many of us consider a T1 Multiplexer.

T1 MULTIPLEXING

T1 is a term for a digital carrier facility used to transmit a DS-1 formatted digital signal at 1.544 megabits per second. T1 was developed by AT&T in 1957 and implemented in the early 1960's to support long-haul pulse-code modulation (PCM) voice transmission. The primary innovation of T1 was to introduce "digitized" voice and to create a network fully capable of digitally representing what was, up until then, a fully analog telephone system.

Today T1 is used for a wide variety of voice and data applications. They are widely embedded in the network distribution architecture as a convenient means of reducing cable pair counts by carrying 24 voice channels in one 4-wire circuit. T1 multiplexers today are also used to provide DS0 "access" to higher order 'transport' multiplexers such as 'SONET'.

T1 Frame

A T1 frame consists of 24 eight-bit words plus a framing bit. Each timeslot of the frame contains 8-bits of binary information. Each timeslot is called a Digital Signal Zero (DS0), which is sampled 8000 times per second. This sampling rate was chosen because it can adequately represent voice characteristics of a human speaker when using Pulse Code Modulation (PCM). Therefore, each DS0 contains 64kb/s (8k samples/sec x 8 bits/sample) of user information. Time Division Multiplexing (TDM) is used to combine 24 DS0's into one T1 frame. Since there are 24 DS0's in a T1 frame, the effective data rate is 1.536 megabits per second. Also, each frame contains one framing bit, which is used primarily for frame synchronization. This bit adds an additional 8kb/s of overhead to the frame thereby primarily for frame synchronization. This bit adds an additional 8kb/s of overhead to the frame thereby increasing the information rate from 1.536 Mb/s to 1.544 Mb/s. This 1.544 Mb/s is commonly referred to as a Digital Signal One or DS1. Note that the word T1 and DS1 are used interchangeable, however this isn't really accurate. A T1 refers to the digital transmission system, which happens to operate at DS1 rates.

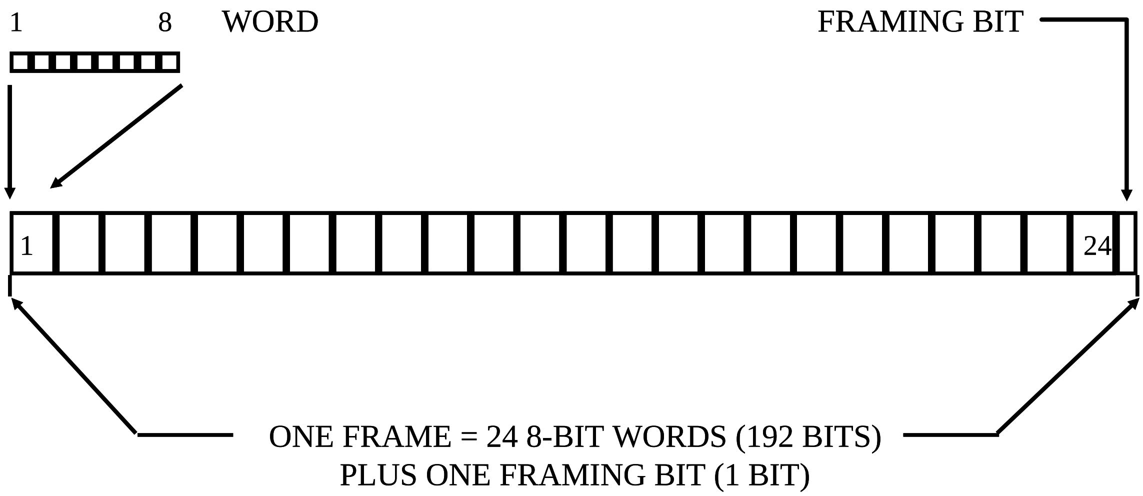

Figure 5. T1 frame

Framing

Framing is used on T1 circuits primarily to synchronize individual T1 frames without the need for external clocking devices. D1 framing was the first framing pattern to be used for transmission of T1 signals. Within a D1 frame, each timeslot contained seven-bits of digitized voice and one-bit used for signaling (call setup and routing). The framing bit identified the boundary between frames. As T1 networks developed, other framing and signaling methods needed to be developed. The SuperFrame (SF) or D4 framing was the first one introduced.

A SuperFrame consists of 12 individual T1 frames. The framing bit in every odd frame is used for terminal framing while the framing bit in every even frame is used for signaling framing. Terminal framing and signal framing are used to form a 12-digit word (100011011100). Notice that the even bits used to identify signaling frames are sequenced as X0X0X1X1X1X0. Signaling information is marked by the change in the bit pattern. Frame six transitioned to a one and frame twelve transitioned to a zero. Thereby signaling information is contained within frames six and twelve of a SuperFrame. The sixth and twelfth frames are used the same in D1 framing. Only two of the 12 frames contain signaling information within each timeslot.

Today, most T1 facilities use a framing technique called Extended SuperFrame (ESF). ESF consists of 24 individual T1 frames. The 24 framing bits are classified into three different categories; alignment or terminal framing (2kbs), CRC (2kbs), and data link (4kbs). The terminal framing bits are used to identify frame boundaries and positions of other framing-bits. The CRC (cyclic redundancy check) is used to monitor the performance of the ESF and the data link is used to send performance information as well as other messages between multiplexers.

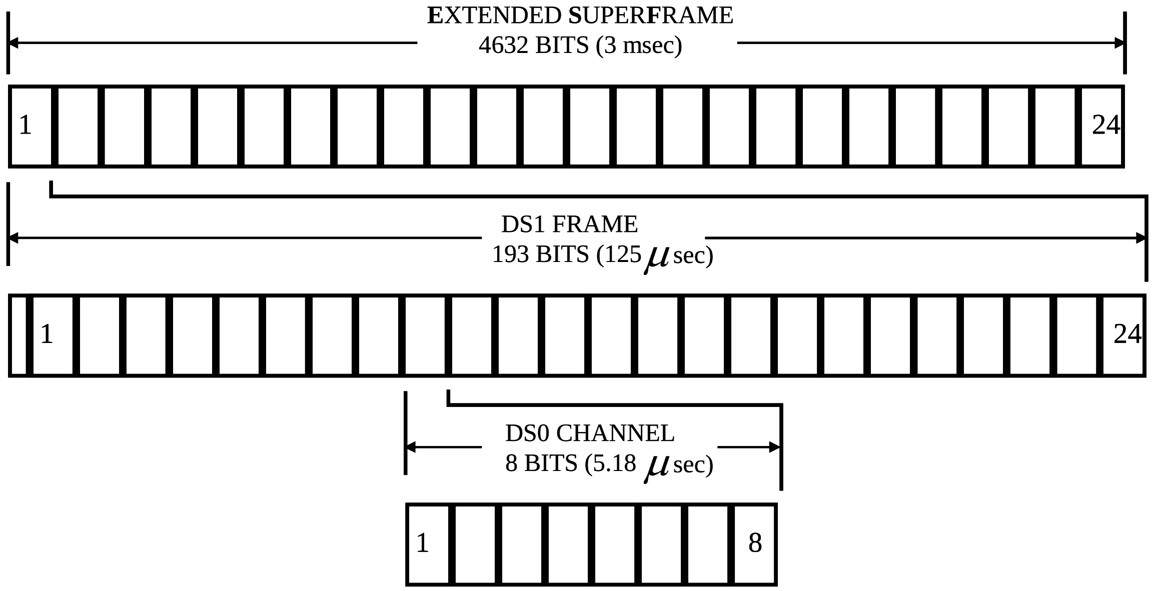

Figure 6. Extended Superframe

Line Coding

A T1 signal is transmitted on the link in a binary format (ones and zeros). This binary format is encoded onto the link using a technique known as alternate mark inversion (AMI). The format is alternating pulses (+3/-3 V) denoting a one and no-pulse denoting a zero. The benefit of this encoding is that it has a built-in method of error detection. Whenever consecutive pulses are detected of the same polarity, a bipolar-violation (BPV) is indicated. Therefore, we know that the frame experienced some type of error. A disadvantage to this encoding is the transmission of an all zero's pattern.

To correctly identify DS-1 input, the multiplexer must know when to sample the bipolar signal to determine whether a “0” or a “1” is being transmitted at any given time. To ensure proper sampling, the multiplexer relies on a timing method that uses the binary pulses (i.e., ones) to maintain synchronization with the network equipment that is transmitting the DS-1 signal.

Since pulses are critical to maintain power signal timing, all DS-1 signals are required to meet specific “one’s density standards”. These standards require that at least one pulse be transmitted within any eight-bit sequence (i.e., 12.5% ones density). Further, since long strings of consecutive zeros between digital values can also hinder signal timing, ones density standards prohibit the transmission of more than 15 zeros in succession.

Success in meeting ones density requirements can vary based on application. For example, since the size and content of the bit patterns that represent human speech are consistent, acceptable ones density in voice applications is a virtual certainty. But since digital data is highly variable in size and content, conformance to ones density standards cannot always be guaranteed. This technical problem is why a coding technique known as “*bipolar with 8-zero substitution*” (B8ZS) has gained in popularity.

B8ZS uses intentional bipolar-violations (BPVs) to break up long strings of zeros, allowing their transmission through the T1 link without violating the ones density standard. With B8ZS, network equipment replaces any string of eight consecutive zeros with two intentional BPVs before the DS1 signal is transmitted over the T1 link: the first BPV replaces the fourth zero, the second replaces the fifth and seventh zeros. Additionally, the eight-zero bit, which normally would be coded as a zero, is assigned a pulse value. Using this format, the DS1 signal can pass through the multiplexer on the T1 link with an acceptable level of pulse density. When the signal arrives at the receiving network equipment, the pattern is recognized as the B8ZS substitute for eight consecutive zeros; the equipment then replaces the international BPVs with their zero value.

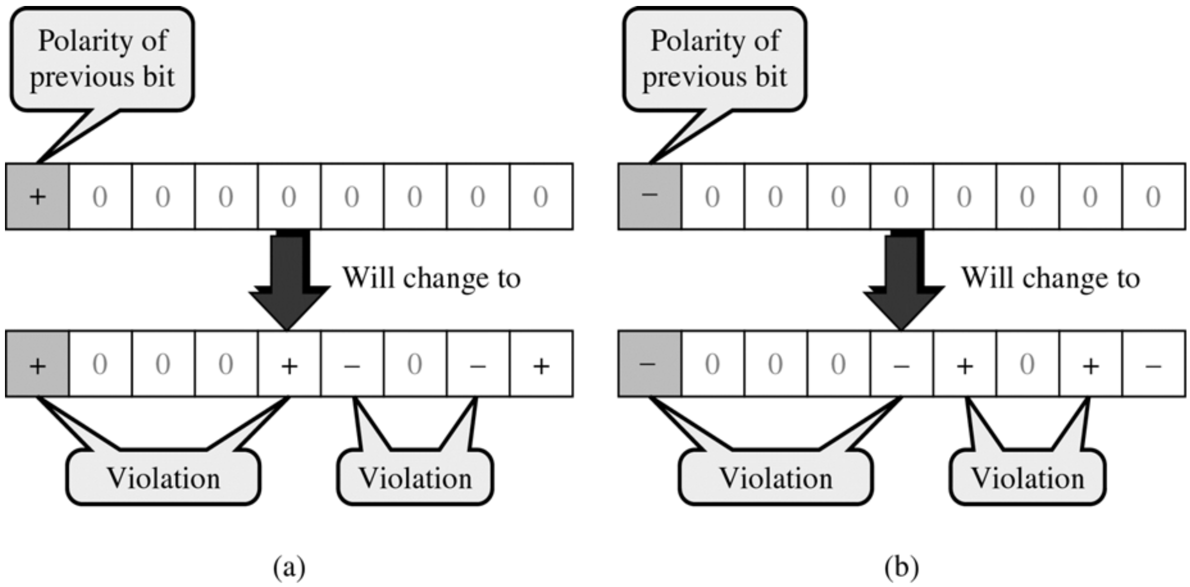

Figure 7. Bi-polar with 8-zero substitution (B8ZS)

Network Timing

T1 networks are designed to be primarily synchronous networks. That is, data clocked in at one point in the network has a fixed timing relationship to the point in the network at which the data is clocked out. Technically, this means that the speeds at both points are the same, and there is a fixed frequency relationship between the clocks which strobe the data in and out. This condition is usually referred to as "frequency locked".

The North American T1 network is derived from a series of a higher-order network multiplexer with unsynchronized clocks, using a technique called "pulse stuffing" to overcome clock inaccuracies and fluctuations. The use of unsynchronized clocks for higher-level TDM networks does not, however, preclude network-synchronized clocks for higher level T1 or DS1 signals. Synchronization of the network at the DS1 level is achieved by framing (D4 or ESF in North America) the data streams and frequency locking the node and network clocks. Loss of synchronization or unlocked clocks results in frame slips. A "frame slip" is a condition in which framing is momentarily lost, as well as network timing information, typically resulting in data loss.

Channel	Bit Rate	Composition
DS0	64 Kbps	DS0 (1 channel)
DS1	1.544 Mbps	24 DS0 (24 channels)
DS1C	3.152 Mbps	2 DS1 (48 channels)
DS2	6.312 Mbps	4 DS1 (96 channels)
DS3	44.736 Mbps	28 DS1 (672 channels)

Figure 8. North American Digital Hierarchy

DACS

A DACS is an acronym for Digital Access and Cross-connect System. A DACS is a digital switching device in telecommunications, for routing T1 lines. The DACS can support what is known as Time Slot Assignment (TSA) and Time Slot Interchange (TSI). TSA is the assignment of a time slot (channel) in a forward Time Division Multiplexing (TDM) facility in order to accommodate traffic from a tributary TDM facility or in reverse. In other words, the DACS can cross-connect any T1 line in the system with any other T1 line also in the system. TSI is the interchanging of time slot between TDM links. That is to say that a DACS can also connect any DS-0 channel or group of channels on a T1 line to any DS-0 time slots of any other T1 line. The DACS can also perform 'Hairpinning" a term used in telecommunications to describe a 'loopback' condition. Hairpinning is a term for data going into a network Element (NE) making a 'hairpin turn' and going right back out. This type of operation is most useful in commissioning or for troubleshooting purposes but one should always be cautious when performing a loopback on a DS0 when a protective relay or teleprotection unit is attached. A protective relay that is in service could possibly false trip unless some type of channel addressing scheme or other safeguard is in use.

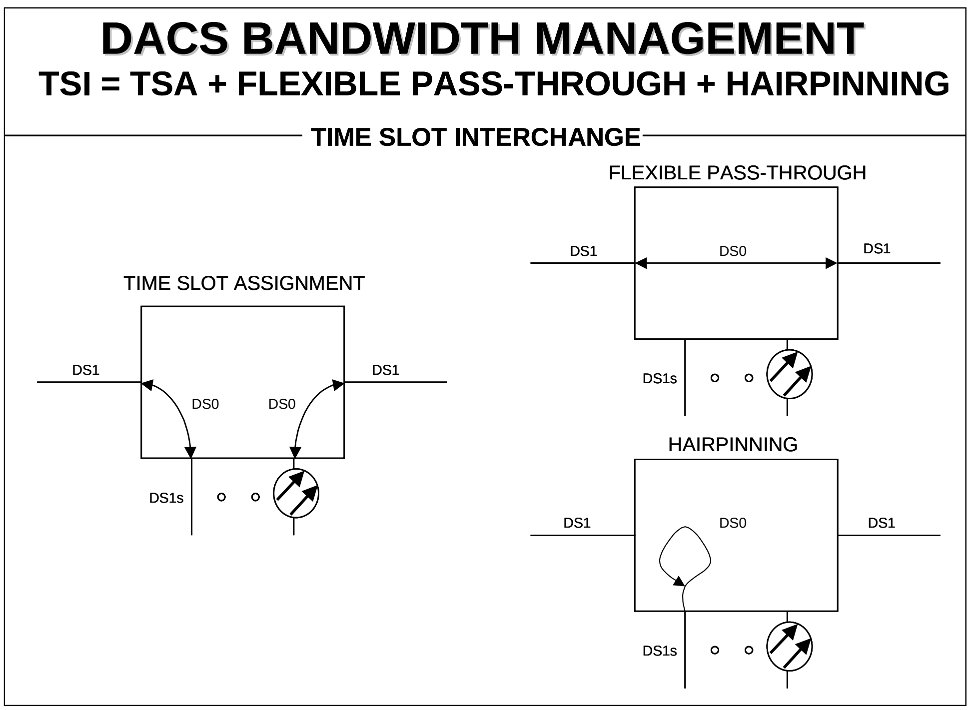

Figure 9. DACS Bandwidth Management

DUKE POWER PROTECTIVE RELAY REQUIREMENTS

The multiplexing equipment already installed at Tiger Tie, North Greenville Tie, Pacolet Tie and Peach Valley Tie stations had drawbacks for Relaying use. At that time, the communications department was using two different vendors channel banks at these locations. However, this created a problem because there was no compatibility between the two vendors digital data cards, so the same multiplexer vendor would have to be used at all terminals. Also, the type of data card required for the relaying channel wasn't even available by one of the vendors being used by the communications department. Prior to the relaying requirement, the communications department was using only voice communications from these sites. There are well-established standards for analog channel cards so there was no compatibility issue between vendors up until this point.

As discussions progressed, it was agreed that sufficient bandwidth was available that would allow a T1 channel to be given to Relaying to use as they pleased. A T1 multiplexer that would have all the needed channel cards including the digital ones needed for the RS232 relay-to-relay POTT schemes could then be installed at each station for relaying purposes.

DUKE POWER COMMUNICATIONS NETWORK

The Duke Power network is a typical telecommunications Central Office (CO) application. In telephony, a CO is a telecommunications office centralized in a specific locality to handle the telephone service for that locality. In Utilities this is typically where they drop their voice and SCADA requirements. The Duke Power Communications group used OC-3 SONET nodes connected in a ring topology for the higher order transport and T1 channel banks interconnected to the SONET nodes to provide DS0 access. All T1 traffic is routed back this central location where they are interconnected to a mainframe DACS which is used to cross-connect the DS0 payload traffic from one T1 to the corresponding DS0 of another T1. Both the SONET and mainframe DACS equipment have hardware protection to protect its operation during single point of failures. The DACS was also interconnected to a larger OC-48 SONET ring, which would transport non-relaying traffic to the main office.

The topology design agreed upon for this project consisted of a T1 from each of these stations - Tiger Tie, North Greenville Tie, Pacolet Tie, Peach Valley Tie and Shiloh Switching Station would be routed to the mainframe DACS located in the communications building at North Greenville Tie where each circuit could be groomed and routed to the appropriate substation or main office. The communications building at North Greenville Tie is located several hundred feet from the relay/control building so a fiber T1 path was used instead of the electrical T1 path used at the other stations where the T1 multiplexer was located only a few feet from the OC-3 fiber transport equipment. In all stations the relay panels were located close enough to the T1 multiplexer that the RS232 path was well within acceptable distances.

The point of concern for the Relay group was the T1 multiplexers that would provide the DS0 access for their pilot communications channels. The Communications group had standardized on two different style Telecom grade T1 channel banks for their networks and had no use or desire to add a third to their mix. This would mean more T1's to learn and service in the long run, which is something they had little time to do. Yet the Relay department was persistent and wanted more say in the type of T1 to be used for Relaying. Finally, Duke Power turned to an outside Consultant to get their independent analysis and recommendation. The results favored the Relay department and with that in mind, the Relay department performed several tests on T1 Multiplexers to help finalize their selection process. The Reliability issues that they considered are discussed below.

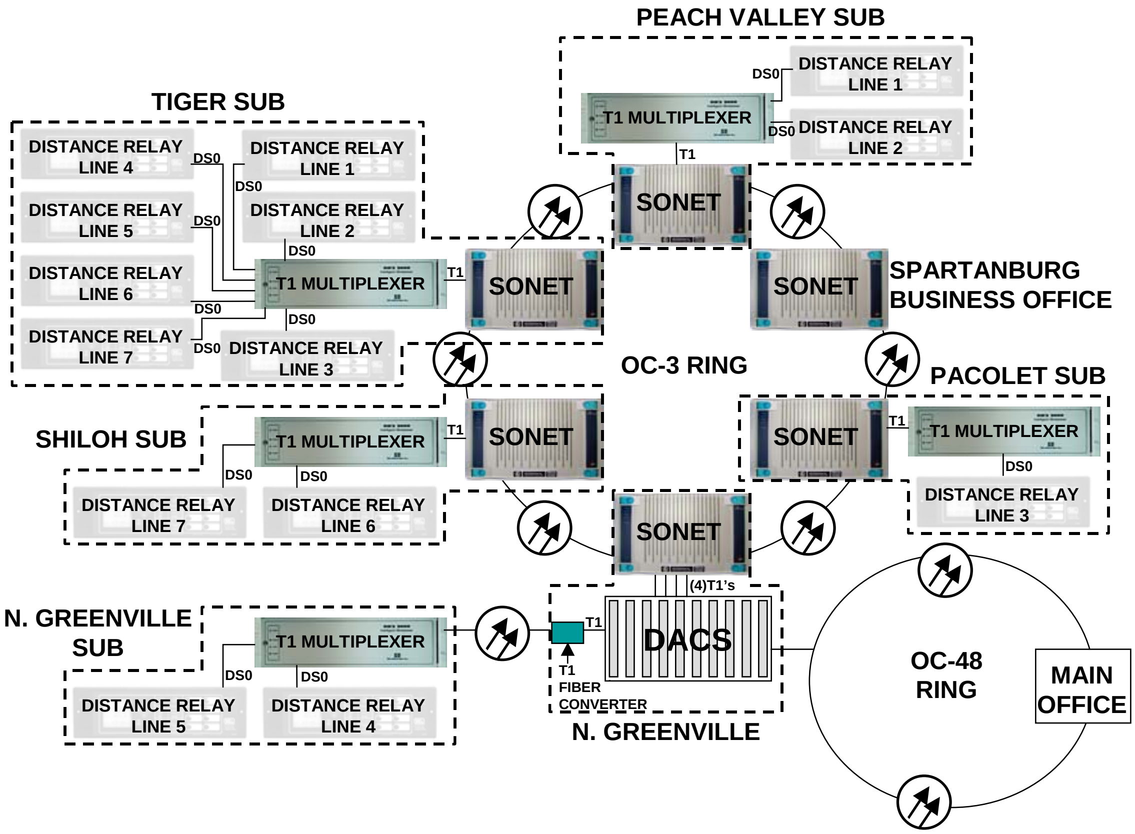

Figure 10. Duke Power Network

T1 MULTIPLEXER RELIABILTY CONCERNS

Several considerations were taken into account when selecting the multiplexer for the Duke Power protection scheme. Between the Consultant and Relay Engineers the majority of those concerns are listed with an explanation of their importance.

- Substation Hardened
  The multiplexer should meet the requirements for the ANSI/IEEE standards for Surge Withstand Capability (SWC), Fast Transient and EMI as specified in C37.90, C37.90.1 and C37.90.2. The advantage of this feature is to insure proper operation during adverse fault conditions when the multiplexer is needed most.

- Cyber Security
  The Protection Engineers favored a multiplexer that offered a selection between hardware or software configurable cards. They liked the idea of hardware settings because the Protection Engineers did not want anyone capable of changing the settings on a DS0 that was carrying a Protective Relaying signal. Because the Multiplexer could be remotely monitored and provisioned, the Relay Engineers were concerned about cyber security. They also felt that the Multiplexer would fall under the Communications Group for maintenance and did not want anyone outside their department having the capability to change a setting on a Relay circuit

- Reframe time
  "Reframe Time" is defined as: "The amount of time it takes a multiplexer to re-synchronize to the network should there be an interruption in the incoming signal." This amount of time equates to an

outage of system availability during that interruption. This is why this time should be kept to a minimum.

- Through-Channel Delay
  "Through-Channel Delay" is defined as: "The amount of delay time incurred by a channel passing through a drop-and-insert location." Since many T-1 applications are multi-point, certain channels will typically pass through several locations before reaching their final destination. This amount of delay incurred may become prohibitive to certain relaying applications (current differential or blocking) should these delays not be kept to a minimum.

- Fallback Timing
  "Fallback Timing" is a necessity that guarantees system clock is available should there be a loss of the incoming clock source. This ensures the remainder of the system will continue to operate from that new clock source.

- DS0 Synchronization and De-synchronization delay in/out of T1 payload
  This is the amount of delay that it takes to input and extract the DS0 signal into and out of the T1payload. The longer the delay the greater the effect on the relaying circuit.

- Channel Addressing
  Channel Addressing is an added layer of security on a DS0 circuit that is run over a switched network or one with a DACS in it. The user puts a separate channel address, different from a time slot assignment, which is totally unique to that circuit. This is done in case a DS0 is accidentally misconnected to the wrong DS0 in a network. The channel address would not match and this would prevent them from communicating improperly. Most modern Relays offer this capability but older Relays could benefit from this feature if it is available in the channel cards most used for Protective Relaying.

- Hardware Redundancy
  Depending on the importance or the type of protection scheme deployed, certain levels of hardware redundancy could be of concern. Some T1 multiplexers offer power supply and common logic redundancy to help eliminate single point of failures. Many consider the power supply as the 'heart' of a multiplexer. Since Multiplexers carry multiple circuits, added reliability can easily be obtained by adding a redundant power supply, which would allow the network to survive if one of the power supplies fails. The common logic card is usually considered the 'brains' of the multiplexer. It takes in the 24 channels of parallel data and forms a single serial T1 interface. For added reliability some may consider adding a redundant common logic, which would also keep the Multiplexer operational should one of the common cards fail.

- ANSI/IEEE C37.94
  C37.94 is a DS0 fiber standard which defines the interface between Teleprotection equipment and Multiplexers. This standard addresses the connectivity between relays and multiplexers on an optical level and was designed to eliminate copper wire in substations which is susceptible to noise interference and Ground Potential Rise. When choosing a multiplexer for relaying purposes this capability is of great importance, especially for new installations.

- Asynchronous Data Interface Card
  When using an RS232 asynchronous pilot channel make sure to evaluate the jitter specifications for the T1 multiplexers channel card. In typical oversampling methods sometimes used to digitize asynchronous data into a synchronous 64Kbps DS0, generally the faster the RS232 baud rate the more jitter at the receiving end. If the jitter becomes too excessive, the Relay may start to ignore some of the messages, making it less reliable. So faster is not necessarily better. The user should find a happy medium between the baud rate of the data and the jitter tolerance of the channel card as determined by the Relay performance. Different multiplexer manufacturers utilize different techniques to help keep jitter to a minimum.

- Noise in Digital Multiplexer Voice Channels
  Noise characteristics and channel hopping are a phenomenon that occurs in digital voice grade channels of Multiplexers that are not in frame. This is discussed in greater detail below.

EFFECTS OF DIGITAL FRAME LOSS ON AUDIO TELEPROTECTION

Although the network this paper discusses had the requirement for an RS232 pilot communications channel, Duke Power had other installations where they wanted to multiplex existing analog teleprotection over a voice channel of the T1. The Consultant was quick to note that audio teleprotection over a digitized voice circuit can be subject to some adverse effects should the network become unstable. Duke Power took this into consideration when selecting a T1 for protection purposes. The subject matter is discussed below.

Audio teleprotection has been applied over various communications systems over the years. Most of the schemes based on techniques perfected from experience with analog communications. Such communications systems include pilot wire, leased telephone and analog microwave. Emerging digital communications schemes are now being employed but analog expertise is not a guarantee of digital success.

The newest revision of C37.93 states:

"Note: The characteristics of failed or failing multiplexer voice grade channels may be different from those of analog transmission systems. Digital multiplexer circuitry designed for voice channels may mimic the presence of valid audio-tone signals for a number of milliseconds after an interruption of the signal path. Audio tone receiver noise and alien tone detection circuitry designed for the characteristics of analog channels may not be effective under these conditions, possibly leading to false trip outputs."

Most digital communications schemes rely on sending high-speed strings of "zeroes" and "ones" along a single bit wide communications path. Fiber, digital microwave, digital leased circuits are all one bit wide. Parallel transmission sends multiple bits at the same instant in time, giving defined hardware boundaries where one character ends and the next one starts. Serial transmission sends a block of bits and no hardware indication of character start position.

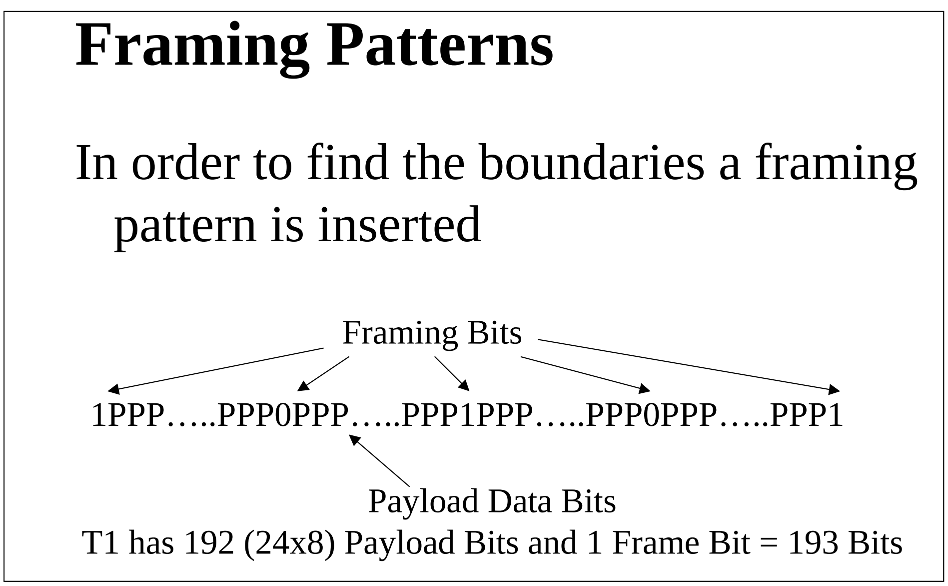

Figure 11. Framing Patterns

The serial stream is broken down into groups where the data for a given channel occurs at a point in time. This point in time is relative to a framing pattern that allows the receiver to correctly pull out the correct data for a given channel. The receiver locks to a framing pattern and is therefore able to output a given channel to a given teleprotection device. The serial stream can be continuous or packetized. In either case the receiver must be able to tell where each character starts and stops. In a continuous stream of bits this is done by looking over a long time and finding a pattern that matches a predetermined rule.

Corrupted bits interfere with this process despite most framing rules allowing for slight mismatch.

Everything works great if the serial data stream is uninterrupted and uncorrupted. When the data stream is altered the receiver may not be able to correctly find the framing pattern. At this point the data output to the protection devices is temporarily or permanently disturbed. An instantaneous disturbance, which causes a loss of frame causes data to be mis-framed for the time it takes to detect loss of frame and time to reframe. During the unframed period, the receiver continues to output data. However, since the start of frame is incorrectly positioned, the output for each channel is taken from the wrong section of the serial bit stream.

Loss of frame can be caused by many conditions.

- Degraded optical fiber connections leading to excessive loss
- Path or line switching in a fiber optic ring.
- Physical interference in a digital microwave path
- environmental interference in microwave i.e. stratified air
- Failure or degradation of communications equipment

Different media can affect the data in different ways. Fiber is usually robust until a hardware failure occurs. This failure need not even be in the data path. A failure somewhere in a fiber optic ring can cause a timing problem, which could cause temporary disturbances in the rest of the ring. Digital microwave is much more likely to drop in and out of frame regularly with varying environmental conditions. With the right weather conditions digital microwave can come and go hundreds of times an hour.

Channel hopping is a phenomenon where the erroneous frame alignment is off by exactly one channel time. This caused valid data from other channels to be output on the desired channel.

Pink noise generation can occur when the frame alignment is off by a non-channel multiple. The data output is a mix of data from adjacent channels and their harmonics. This is further complicated by the mu law compression that takes place in PCM channels.

Audio teleprotection was developed in the age of analog communications. It is very robust in the presence of a clear channel, a valid channel with random noise, and complete random noise. It is not very good at detecting channel hopping.

A worse case scenario is a utility, which has many audio protection sets running over the same multiplexer. Further assume that all of these channels use the same trip and guard frequencies and some of the channels are in the trip state when a hop occurs. A hop to a channel, which is in the trip state is guaranteed to develop a trip output.

Audio teleprotection is designed to pull a trip signal out of a very noisy channel. Protection sets will often work down below 0 db SNR. If the pink noise has larger trip frequency components then guard components, a trip output is possible.

Designing a system which never loses frame is ideal but is not practical. Designing a system where Loss Of Signal (LOS) is quickly detected and corrected is the best approach.

Sometimes rapid detection isn't easily accomplished. Loss of frame detection can be limited by a predetermined framing pattern i.e. T1 can take 2 ESF frames to detect a loss of frame (6 ms). Adding additional framing patterns, which repeat more frequently reduces this time at the expense of payload.

Channel Addressing is a common technique used in digital teleprotection where specific transmitters and receivers have matched addresses that are sent along with the data. This guards against erroneous misconnection to other similar device during switching or channel hopping. Addressing could be applied to analog systems by stealing bits from the payload. The time required to detect a misconnection would be long in analog without stealing quite a few bits.

Giving all teleprotection pairs different frequencies would guard against channel hopping but requires a very programmable teleprotection device. Having hundreds of different frequencies can be a maintenance problem as well. This will not significantly help pink noise issues. The best protection against channel hopping and pink noise is to quickly, faster than the teleprotection can operate, detect the bad framing and squelch all output data until the framing is reestablished. A utility relaying grade multiplexer should be designed to do just that.

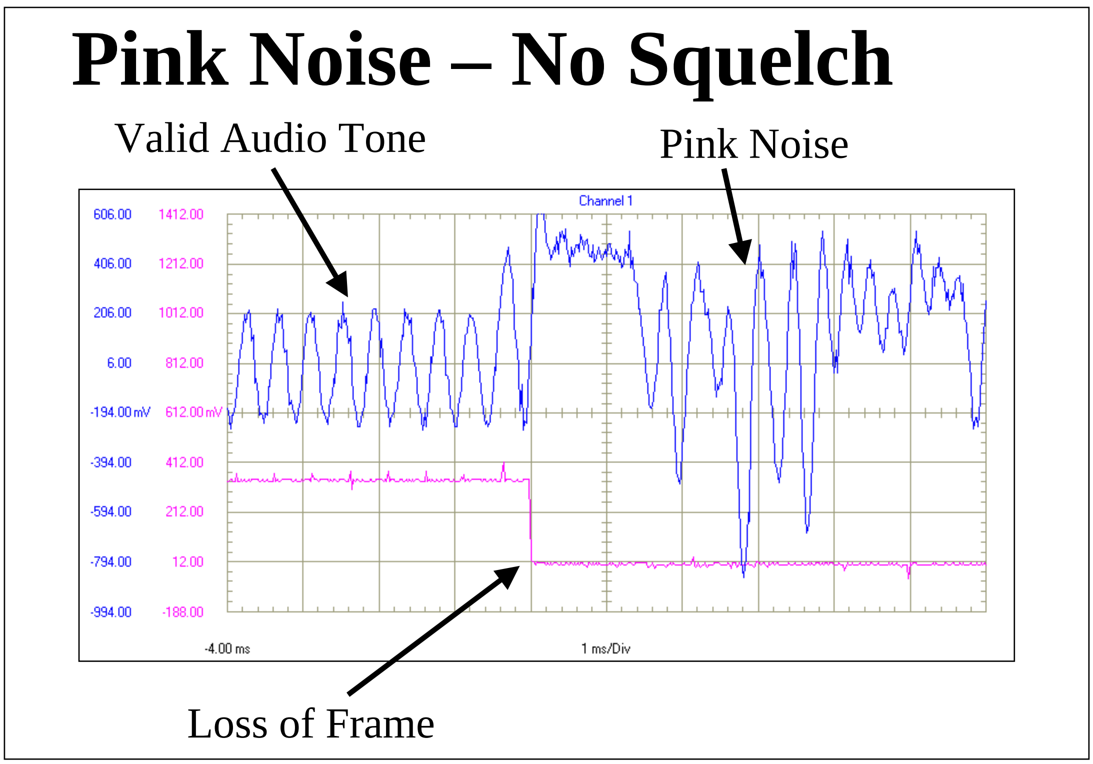

Figure 12. Pink Noise – No Squelch

Figure 12 shows the output of a PCM Audio channel on a T1 multiplexer. The audio is a single tone at about 2200 Hz. The purple line indicates where the loss of frame occurs.

The output is obviously disturbed but two points are very interesting to note.

1. The amplitude of the noise is very similar, within 5 db of the original signal.
2. The frequency of the noise in the fourth millisecond after the break is clearly about 2500 Hz.

This is a perfect example of a pink noise situation, which could cause a false trip on a system with 2200 Hz guard and 2500 Hz trip frequencies.

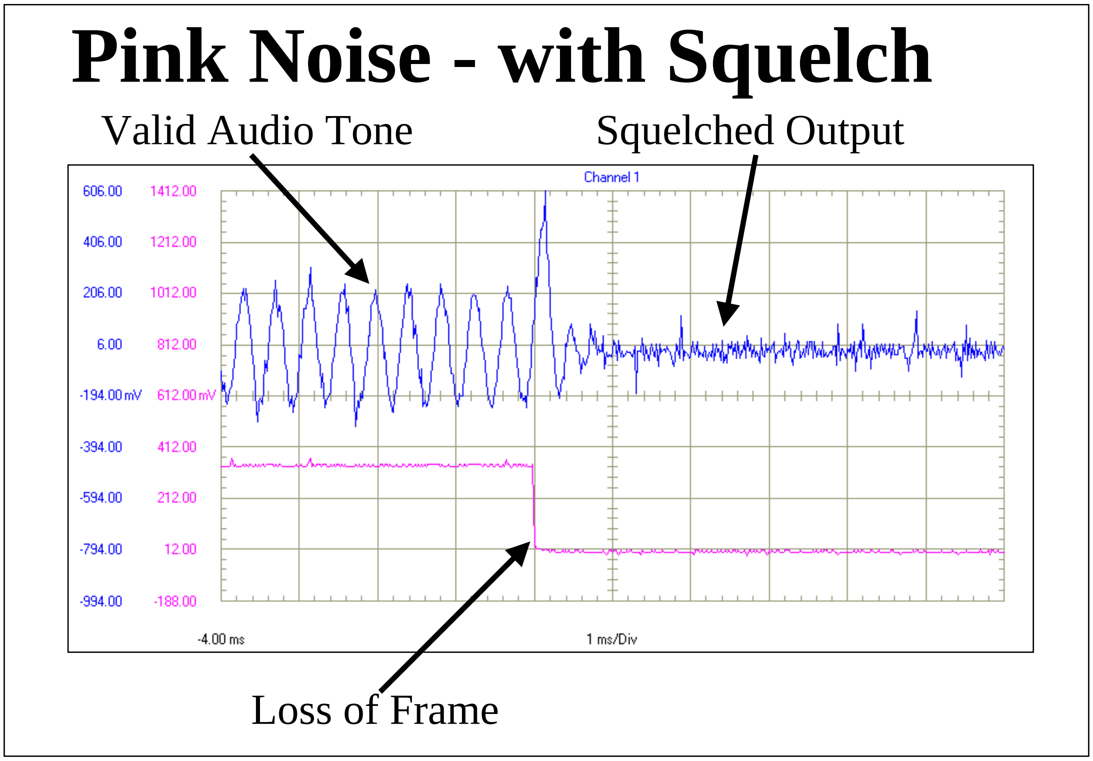

Figure 13. Pink Noise – With Squelch

Here is the same situation with DS0 squelching enabled on the T1 multiplexer. The noise output drops dramatically within ½ ms of the break. This would not cause an audio teleprotection channel to trip.

FINAL PROJECT DESIGN

The Duke Power Communications Department had been reluctant to use the T1 multiplexer chosen for relaying for station communications needs but the economics drove them to agree that it made sense at Shiloh where no equipment previously existed. Since they were going to accept it at Shiloh, then it also made sense to use the new multiplexer at Tiger Tie for the station communications needs and the existing multiplexer could be retired. Consideration has been made to also eliminate the Telecom grade multiplexer at all the stations involved in this project.

FUTURE USES

Since the completion of the Tiger Tie project, expansions of the use of this multiplexing equipment has been planned and implemented. An eighth 230 kV line is being added to Tiger Tie. It will be the second 230 kV line to Pacolet Tie. Additions to the 100 kV system in the area has made it justifiable to add communications facilities at one of the remote 100 kV stations and a POTT scheme is replacing the DCB scheme.

SUMMARY

The ability to intermix voice and data and preserve the transmission characteristics of each is a primary requirement of T1 multiplexers. Voice transmission, for example, can tolerate a few bit errors and not affect the quality of the voice at the receiving end. Speed of delivery is important as any delay is noticeable in voice conversations. In telephone company channel banks, multiplexers data transmission is treated just the opposite. Error free transmission is a higher priority than speed of delivery. It is this practice that makes most Telecom grade T-1 systems unusable for critical real-time applications such as pilot line protection relaying.

REFERENCES

[1] Pilot Protection Communication Channel Requirements, Solveig Ward & Tom Dahlin

[2] Newton's Telecom Dictionary, 20th Ed., Harry Newton

BIOGRAPHIES

Thomas M. Dahlin
Tom has spent 26 years with RFL Electronics Inc. after graduating from Metropolitan Technical Institute in 1979. Tom has held numerous positions within RFL including Test, Customer Service, R & D Engineering, Systems Engineering, Sales Engineer and now Director. Tom spent many years working with Protective Relaying applications before changing his focus to telecommunications. Today, Tom spends most of his time designing Multiplexer networks for Power Utilities. Tom is an active member of the IEEE and resides on several working groups under the Power Systems Communications Committee and Power Systems Relaying Committee.

James M. O'Brien (Jim)
Jim received his BSEE from North Carolina State University in 1976 and is a registered professional engineer in North Carolina. He has spent his entire career in Protective Relaying at Duke Power. After a short stint working as a Field Test Engineer he joined the Relay Design Group where he spent more than half his career including supervising that group from 1988 through 1996. Currently he is a Senior Engineer in the Relay Calculations and Analysis Group of the Power Delivery Division where he focuses on transmission relaying. His responsibilities include specification and technical support of relay communications equipment. Jim is a member of the IEEE and active in the PSRC where he is a member of several working groups.

Solveig M. Ward
Solveig received her M.S.E.E. from the Royal Institute of Technology, Sweden in 1977. The same year she joined ABB Relays. She has held many positions in Marketing, Application, and Product Management. Assignments include a six-month period in Montreal, Canada and two years in Mexico. When Ms. Ward returned to Sweden, she was responsible for the application aspects in the development of a numerical distance protection relay and in charge of marketing the product. After transferring to ABB in the US 1992, she was involved in numerical distance protection application design, and was Product Manager for ABB's line of current differential and phase comparison relays.
Solveig has written, co-authored and presented several technical papers at Protective Relaying Conferences. She is a member of IEEE-PSRC Main Committee and holds one patent.
In 2002, Solveig joined RFL Electronics Inc. as Director of Product Marketing. She is presently involved in the development of new products.