

# Design and Implementation of a Time Division Multiplexing (TDM) Communications System

Revision 3.0

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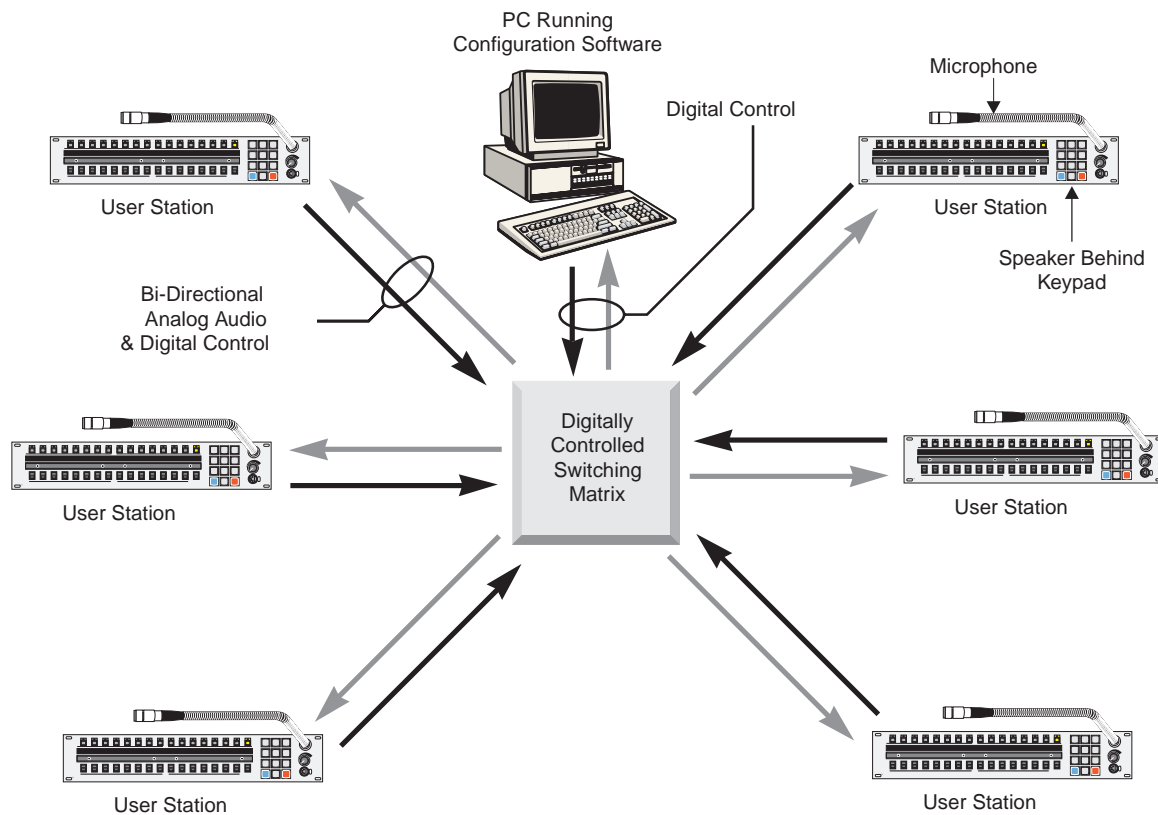
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## Intent and Scope of this Document:

This paper will examine Time Division Multiplex (TDM) techniques as applied specifically to the design of a digital audio intercom system.

## Introduction:

Telex is involved in the design and production of large, digitally controlled matrix intercom systems. Such intercoms find use in television studios, theaters, military and industrial installations, or other situations where the activities of many personnel need to be coordinated.



**Figure 1.**

Figure 1 shows a small matrix intercom system. Each user station has a microphone and a speaker, and so is both a source and destination of audio. As large intercom systems may accommodate hundreds of users, there can potentially be hundreds of audio sources and destinations within the system that need to be interconnected.

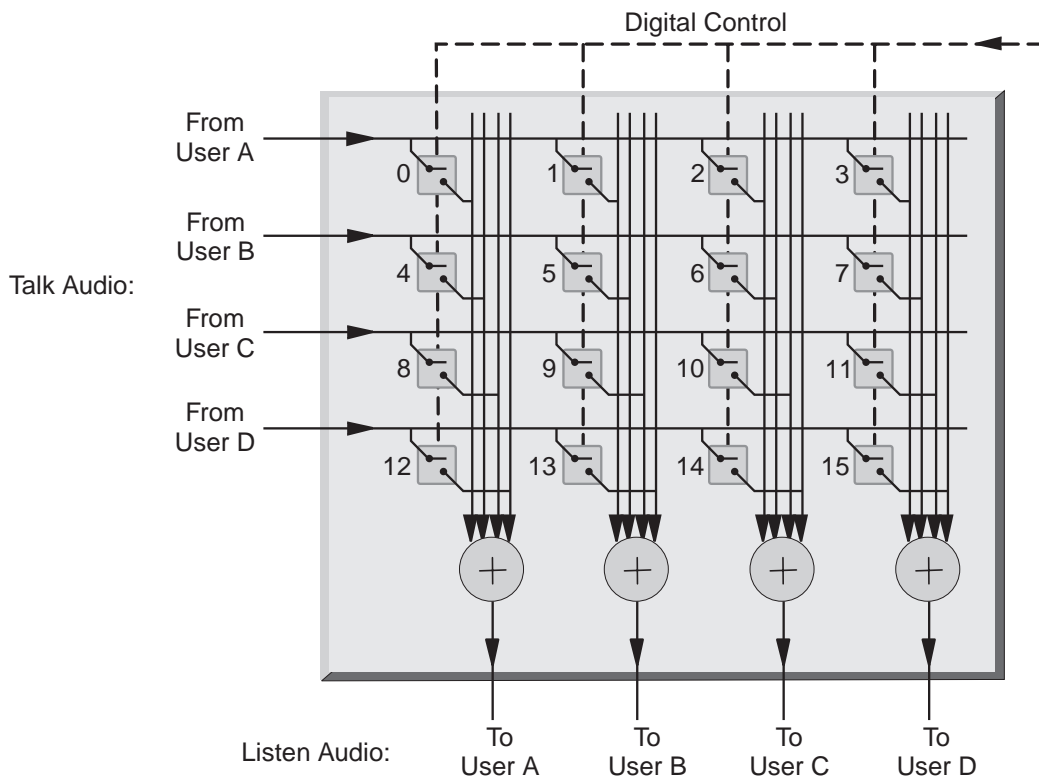
When a user at a station presses a talk key, his or her audio is sent to another station, where it is mixed with any other talker's signals that might be simultaneously arriving there. When a listen key is pressed, the selected signal is retrieved and mixed with other listen audio arriving at the user's station.

Rather than being connected directly to each other in "hard-wired" fashion, each user station is bi-directionally connected physically to a central switching matrix. The matrix then makes the actual electrical connections between source and destination.

In order to make the proper connections, the matrix must be able to sense talk and listen key presses at the user stations, so bi-directional control signals are also connected between each user station and the matrix.

Although Telex's previous generation of intercom systems provided digital control over audio connectivity, the audio paths themselves were analog from end to end. To provide computer controlled connectivity, these systems made use of a matrix of digitally controlled analog switches, called crosspoints.

In order to make a connection between a source and a destination, a single crosspoint would be closed. In order to accommodate any possible interconnection, a crosspoint had to be provided for every conceivable combination of source and destination. The resulting configuration was referred to as a crosspoint matrix.



**Figure 2**

Figure 2 displays a simple crosspoint matrix of four users, A through D. If talker C wishes to talk to listener D, for example, then the matrix will close crosspoint #11. The fixed-gain mixers across the bottom of

the matrix allow each listener to hear multiple talkers at once. Note that with four users, 4 x 4, or 4 squared crosspoints are required.

In reality, matrices usually begin at a minimum size that is much larger than four users. In the previous generation of Telex's intercom product line, for example, the smallest matrix product was a 3 RU-high chassis containing 50 x 50, or 2,500 crosspoints to interconnect 50 user stations.

As you can see, one of the major drawbacks of systems based on a crosspoint matrix architecture is that their size and resulting cost grow in proportion to the square of the number of users. This square-law growth is shown by the fact that four users require 16 crosspoint switches, 50 users require 2,500 crosspoints, and a large system of 250 users would require in excess of 60,000 crosspoints! Our 3 RU matrix for 50 users grows to nearly two entire 7-foot racks to accommodate 250 users.

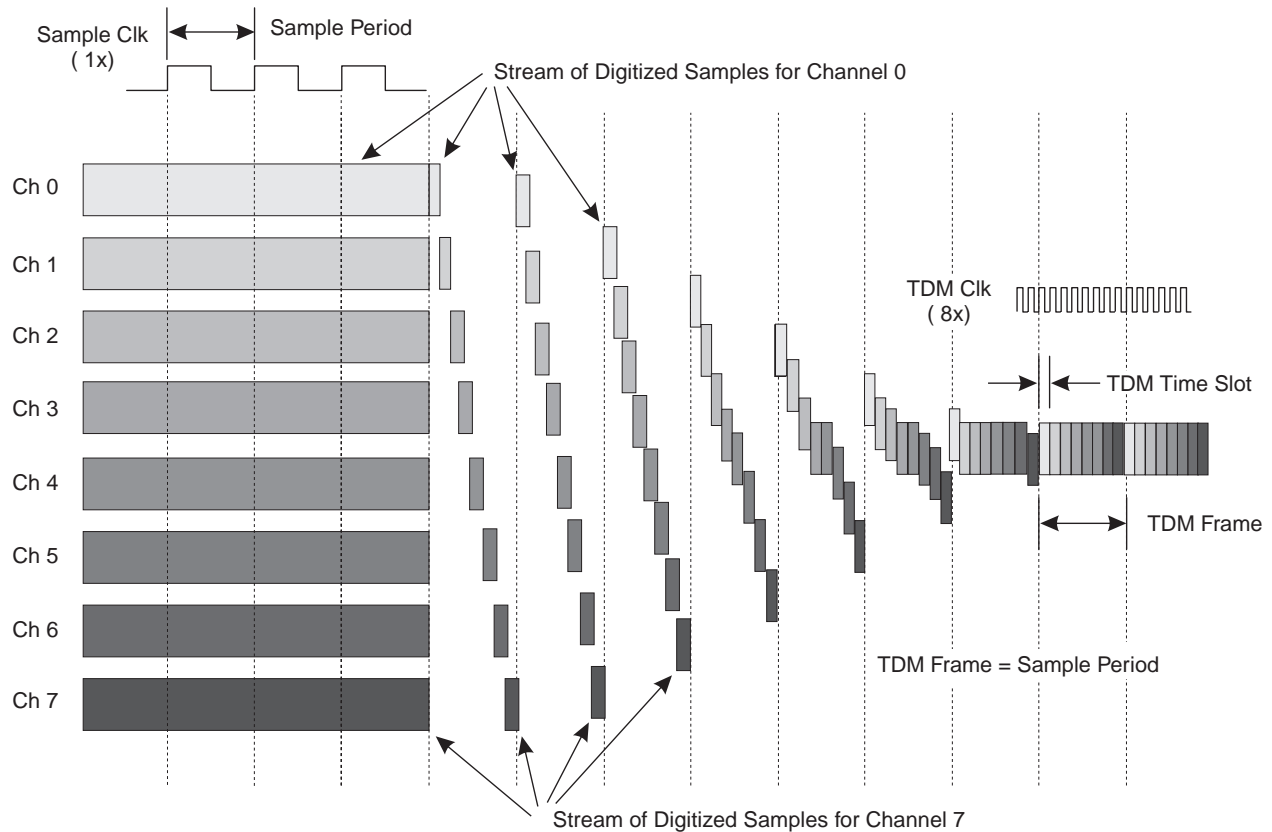
Another disadvantage of the previous generation system is that the fixed-gain mixers did not allow a user individual control over the relative levels of the signals that he or she was listening to.

As a result of these limitations, a new system was envisioned - one in which the square law of growth would no longer drive up system size and cost. The design team chose to base the system design around a Time Division Multiplexing (TDM) architecture.

## **TDM Technology Overview:**

TDM is a standard technique which allows a single physical conductive path or bus to carry many channels of information. The many channels can all "fit" because the individual channels take turns using the bus in round-robin fashion. When the last channel has taken its turn, the first channel takes another, and so on in a regularly repeating way.

Digital audio signals are inherently "TDM-Ready", because they each consist of a sequence of measurements, or samples, evenly spaced in time. A TDM scheme simply interleaves the samples of several digitized signals.



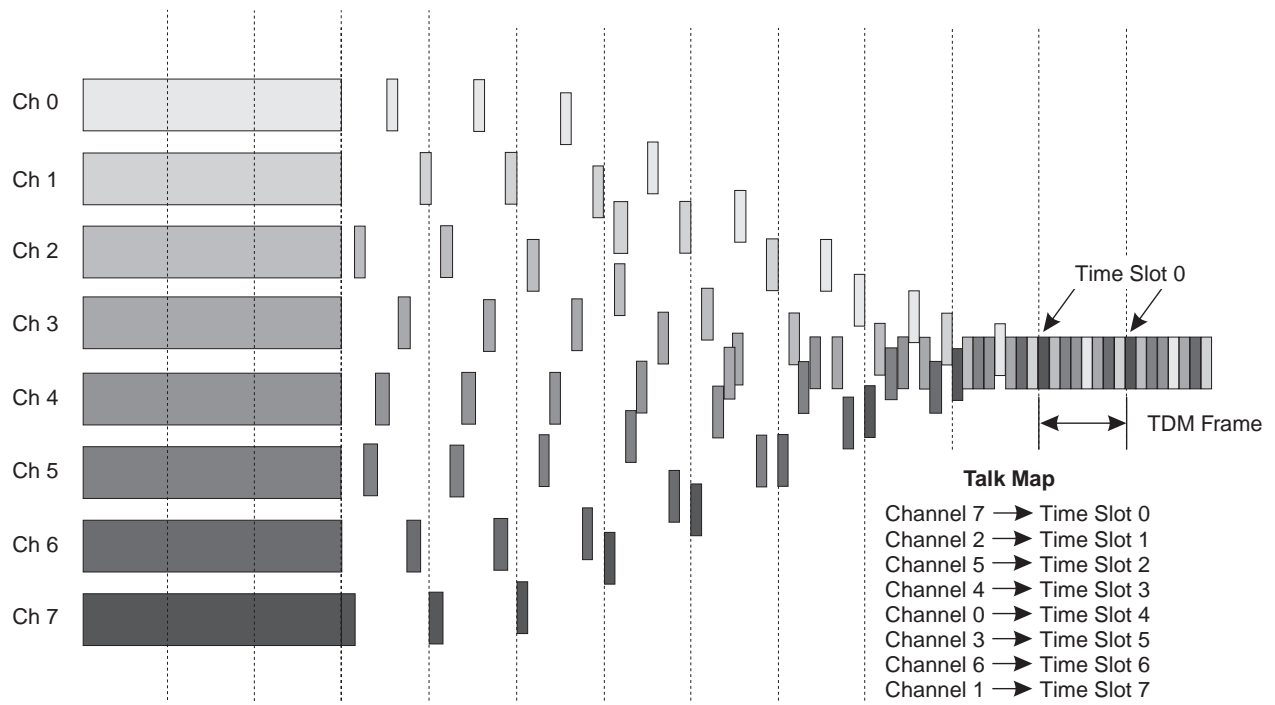
**Figure 3**

Figure 3 illustrates the “TDM-ing” of eight digitized audio source channels, or talk channels, into a single composite TDM signal. On the left are the eight individual channels of digital audio, all uniformly sampled on the rising edge of a relatively slow sample clock. At this stage, each individual channel is carried on its own bus, so its samples appear back-to-back.

Moving from left to right, the diagram then shows the eight individual sample streams being time-compressed and interleaved onto a single TDM bus. To prevent loss of information, the TDM bus clock must run at a frequency equal to the basic system sample rate times the maximum number of channels to be combined. This ensures that any given channel still gets to transmit one sample per sample period.

The space that a compressed sample occupies in time on the TDM Bus is referred to as its “time slot”. The time it takes for all the signals’ time slots to be presented once is called the “TDM frame”, and is set equal to one period of the basic system sample clock.

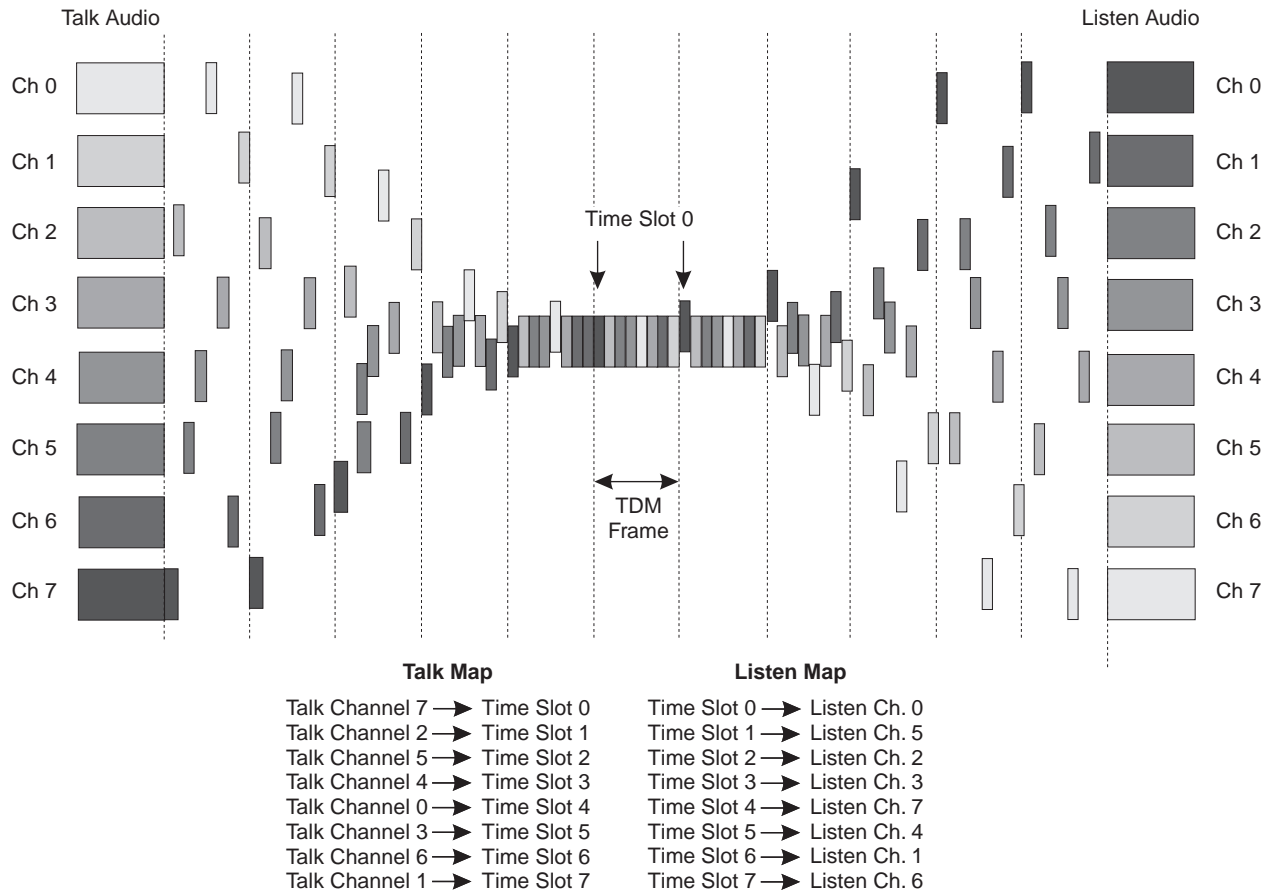
The first time slot in every TDM frame is called “time slot 0”, with subsequent time slots numbered in increasing order. For simplicity, Figure 3 shows the talk channels being assigned to time slots in order, but in the general case there need be no correlation between talk channel number and time slot number, as Figure 4 should help to make clear:



**Figure 4**

By incorporating a RAM-based mapping function called the “talk map”, any talk channel can be routed to any time slot under software control. Notice that talk channel 0 has been routed to time slot 4, while talk channel 1 has been assigned to time slot 7, etc.

Signals that have been “TDM’ed” together need to be “un-TDM’ed” before they can be used at their destinations.



**Figure 5**

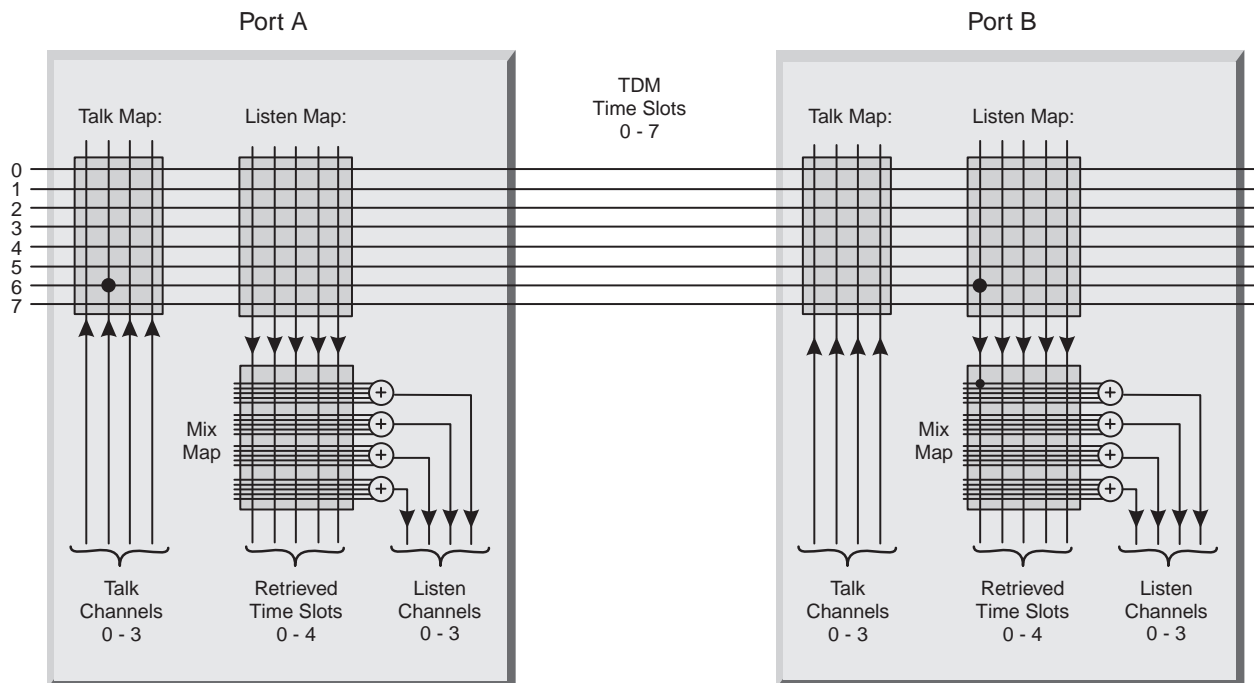
Figure 5 shows the TDM composite being separated back out into individual listen channels. Notice once again that there is no correlation between time slot numbers and listen channel numbers. By incorporating a second RAM-based mapping function called the “listen map”, any time slot can be retrieved from the bus and routed to any listen channel under software control.

Because of the talk and listen mapping functions, any talk channel may be routed to any listen channel via the TDM bus. For example, notice that the signal that starts out in life as talk channel 0 ultimately winds up as listen channel 7 after being routed, via TDM time slot 4, by the talk and listen mapping functions. To make a connection, a listener’s user station is mapped to the same time slot as the talker’s user station. The choice of time slot is irrelevant. As long as there are enough time slots to accommodate the maximum number of talk channels anticipated, then any source may be mapped to any destination - in fact to any number of destinations.

For purposes of this discussion, the combination of a circuit that interleaves samples onto the bus with a circuit that retrieves them will be called a “TDM interface”, or simply a TDM port. Rather than having a single TDM port circuit acting as combiner and separator for the whole system, a TDM matrix is usually implemented by allowing multiple driving and receiving ports to be connected to the TDM bus at various points along its length. Think of each TDM port circuit as a “bus stop” where some time slots board the TDM bus and others get off.

In addition to signal routing, a matrix intercom system requires a mixer for each user. To incorporate programmable mixing into the TDM architecture, a mixer is inserted between the retrieved time slots and the listen channels. A third RAM-based mapping function called the “mix map” is then used to dynamically control the mixing of multiple talkers to each listener site.

Figure 6 illustrates mapping in a very small example TDM system with just eight time slots and only



**Figure 6**

two ports. For simplicity, the time slots are being represented as wires, and the maps are being shown as familiar crosspoint matrices.

The TDM port design shown is a four-user port, as it has four talk and listen channels. Of the eight time slots available, each port is capable of driving any subset of four, and of retrieving any subset of five. Each port is then capable of creating four independent submixes of the five time slots retrieved. Port design specifications such as the number of time slots driven and retrieved, the number of submixes created, and the number of time slots input to each submix are all parameters that may thought of as being adjustable by the system designer to suit the needs of a particular application.



To make a connection from a talk channel to a listen channel, three map entries must be made: one entry apiece in the source port's talk map, the destination port's listen map, and the destination's mix map. Figure 6 shows the map entries as connection dots, and as an example shows talk channel 1 on port A being mixed to listen channel 3 on port B via time slot 6.

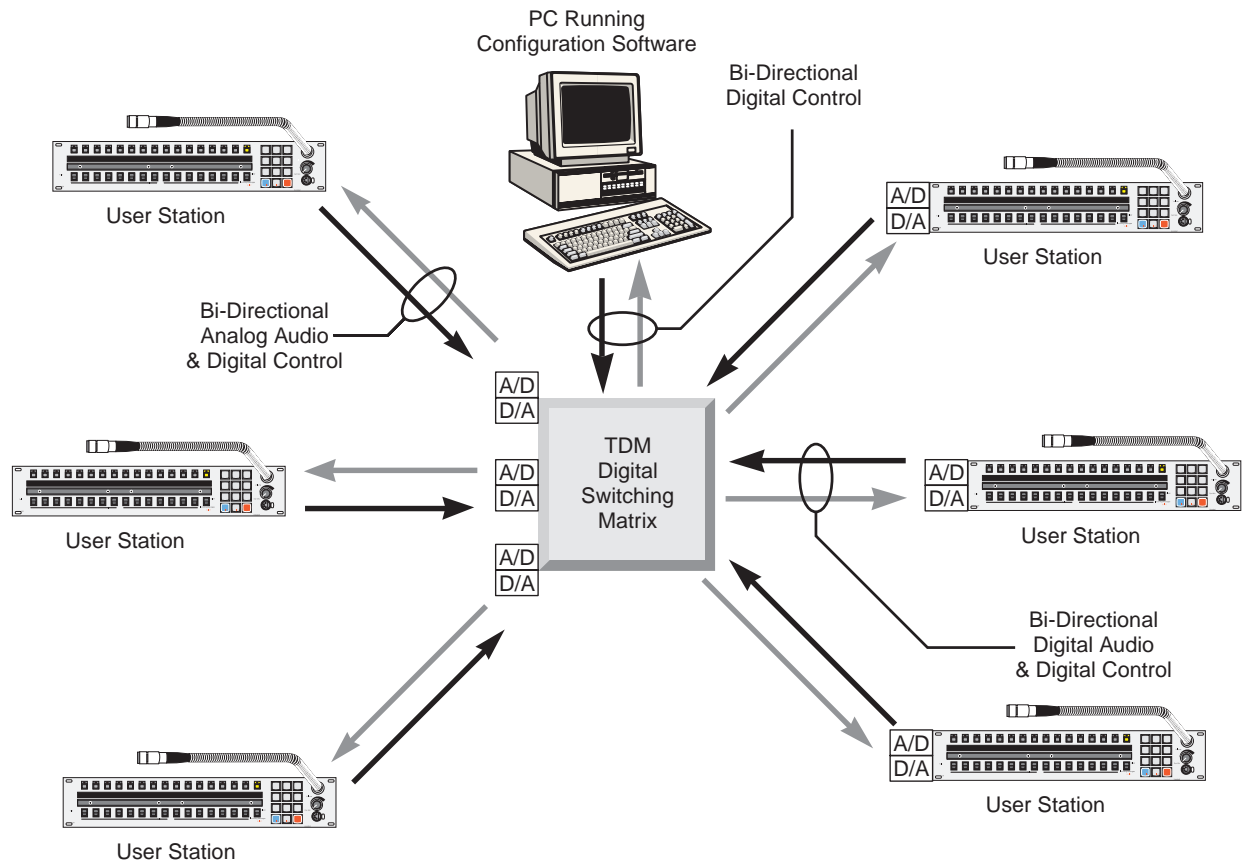
The concept of the port is what grants the TDM architecture immunity from the exponential growth rule that applies to crosspoint matrices. The TDM matrix is decentralized; a little bit of the overall matrix resides in each port. The port circuit is designed to accommodate a certain number of users. To add that number of users to the system, you just add another port. The system is also designed to accommodate a certain number of ports. The system's maximum capacity is then equal to the maximum number of ports times the number of users each port accommodates.

## **Telex TDM System Design Considerations:**

Now that the basics of a TDM architecture have been reviewed, the specifics of Telex's particular brand of TDM can be discussed, along with justifications for our choices of system design-parameter values.

- It was decided that the digital audio paths should allow transmission of CD-quality signals throughout the system. Therefore, 16-bit A/D and D/A converters were selected and the TDM frame rate was set at 44.1 kHz.
- It was decided that the TDM frame would be divided into 1,000 time slots. The system thus accommodates as many as 1,000 user stations. Conceptually, systems could be designed with even higher capacities, but the TDM clock frequency eventually gets so high that relatively exotic technology is required to keep up. The Telex TDM bus runs at 44.8 MHz, within reach of relatively inexpensive CMOS technology.
- The TDM port design has been implemented in the form of a custom ASIC, or Application-Specific Integrated Circuit, i.e., a custom chip. This chip can drive any subset of 64 time slots and retrieve any subset of 256 time slots. From the group of retrieved time slots, the mixer can create from 1 to 63 volume-weighted submixes.
- The TDM bus word-width was selected at 24 bits - sufficient to allow mixes of 256 channels of 16-bit audio to be calculated without need for rounding or truncation.
- It was decided that the TDM bus should electrically and mechanically accommodate a large number of ports (17), so that users can be added in small increments. A larger number of smaller ports results in a more modular system.

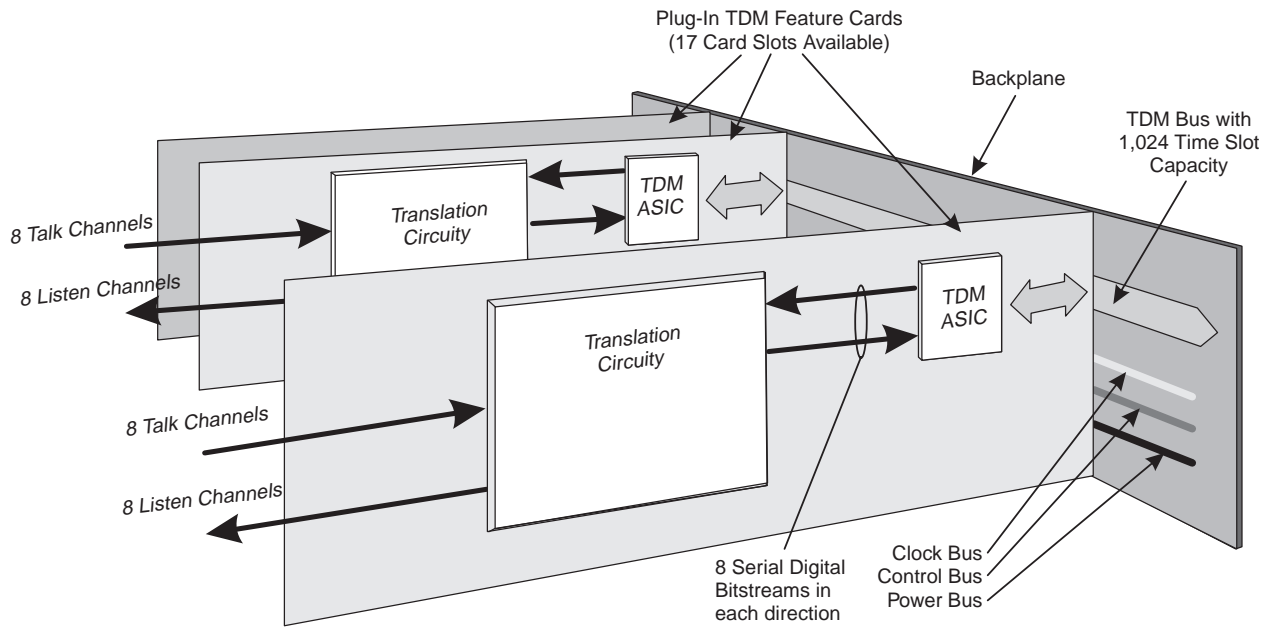
The top-level diagram of the new system has actually not changed that much:



**Figure 7**

The only noticeable difference at this level is the presence of A/D and D/A converters. On the left are three of the original analog audio user stations; on the right are three conceptualized digital audio user stations. Architecturally speaking, the only difference between analog and digital user stations is which end of the cable the converters are on. Telex has elected to keep the converters on the matrix end of the cable, preserving the tremendous number of interconnect options available to analog audio signals.

It is on the inside of the matrix where the change is more striking:



**Figure 8**

The overall matrix design physically consists of a 7 RU card cage with integral backplane and redundant power supplies. The card cage has 17 physical card slots in which to install TDM-compatible circuit cards called TDM feature cards. Every feature card includes its own TDM ASIC which allows the card to interface directly with the bus. These are the system's 17 TDM Ports. Figure 8 shows a system with just three feature cards installed.

The intent is ultimately to design an entire family of such feature cards, each of which will serve a distinct purpose in the system. Users will then be able to mix and match the various card types to construct a system with capabilities custom to their specific needs.

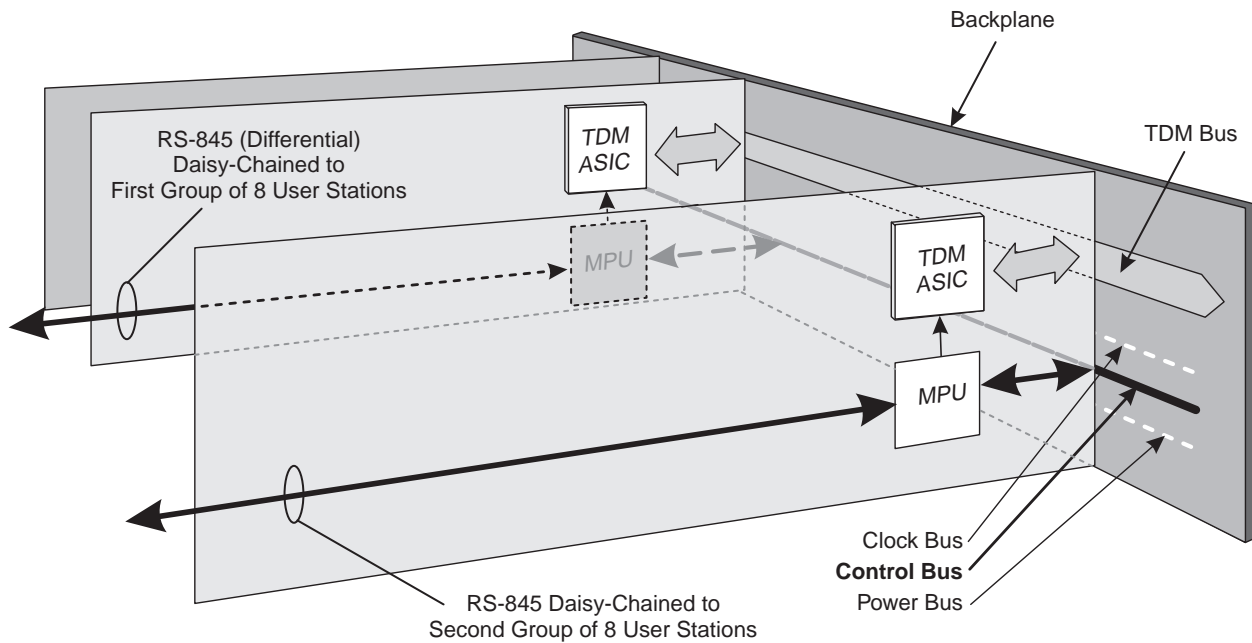
For example, one type of card is used to interface eight analog user stations into the system; another will cross-connect multiple card cages, etc.

No matter what the purpose of a TDM feature card, its design always starts with the same basic TDM “boilerplate” circuit, representing the minimum circuitry required for an interface to the TDM bus. This boilerplate circuitry includes at least one TDM ASIC as well as data and clock drivers, etc. On every card there must also be a microprocessor which is responsible for maintaining the mapping functions inside the ASIC.

What is unique about each feature card type is its so-called “translation circuitry”. The job of the translation circuitry is to convert whatever audio format is being generated and consumed by the external equipment into one of the many digital formats understood by the TDM ASIC.

Figure 8 shows the first TDM feature card actually in production. It is called the “analog I/O card”, and its purpose is to provide system interface for eight of our existing analog audio user stations. The translation circuitry on this card predictably consists of eight channels of A/D and D/A conversion.

Now that the audio pathways through the system have been described, the control pathways can be discussed.



**Figure 9**

As mentioned, every card includes a microprocessor responsible for maintaining the mapping functions inside the ASIC. These processors have access to a backplane control bus, which they can use any time they need to send a message to a processor on another card.

The Analog I/O Card's processor changes the TDM ASIC's mapping functions in response to user key presses at one or more of the card's eight user stations. Therefore, an RS-485 serial differential line is provided which "stars" out to all eight of the card's user stations in parallel.

Now suppose a user presses a talk key. The processor on the talker's card becomes aware of the key press during the very next poll of its user stations. The talker's processor first makes the audio connection from talk channel to time slot by programming its own TDM ASIC. Then it sends a message via the backplane control bus to the listener's processor, asking it to make the subsequent connection from time slot to listen channel via the mixer. The listener's processor makes this connection by programming its TDM ASIC. As soon as the final connection is made, digital audio flows from source to destination through the system without further need for microprocessor intervention or assistance. The processors are then free to await further user key presses.

In summary, we have seen how moving from an analog crosspoint matrix architecture to a digital TDM backplane bus architecture has helped create a new system which is smaller, sounds better, gives listeners control of the individual channel volumes in their mix, and provides a large new reserve of excess capacity.