AUTOMATED MIXDOWN

the second generation

The

ALLISON

'MEMORY

PLUS

SYSTEM'

Reprinted from March/April 1976 issue of R-e/p.

1976 PROMISES TO BE...

THE YEAR OF THE PROGRAMMABLE AUDIO REVOLUTION. ALLISON RESEARCH PROMISES THE INDUSTRY STANDARD SYSTEM. MOST AUDIO MANUFACTURERS AGREE. HERE'S OUR STARTING LINE UP.

THE NEW 65K PROGRAMMER
- 65,000 Bits of capacity
- 3.2 millisecond access time
- No nonsense data storage on master or syncronized tape
- Self testing circuitry
- Proposed standard by most console manufacturers

THE GREAT EQUALIZER
- Fully addressable/programmable
- 24 equalizing frequencies
- 14 frequency 18dB/octave cutoffs
- Preset-able, gang-able
- Real time display of E. Q. curves
- Superior audio specs

THE MEMORY PLUS SYSTEM
- Plug in retro-fit to any console
- Programmable level, group masters assignments, solos & mutes
- Visual monitors on all parameters
- Expandable to E. Q., Panning & Echo
- Total concept human engineering

THE FABULOUS FADER
- Look ma, no knobs!
- ¼ dB precision tracking
- Fully addressable/programmable
- Noiseless, unlimited life
- Electronically settable — no nulling

PANNER PERFECT
- Fully addressable/programmable
- Visual quad pan field monitor
- Knobless control — electronically settable
- Superior audio specs

ON TAP
- Fully programmable patching, limiting expanding, compressing, etc.

AFTER ALL,
WE DID START THE WHOLE THING!

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The ALLISON

'MEMORY PLUS SYSTEM'

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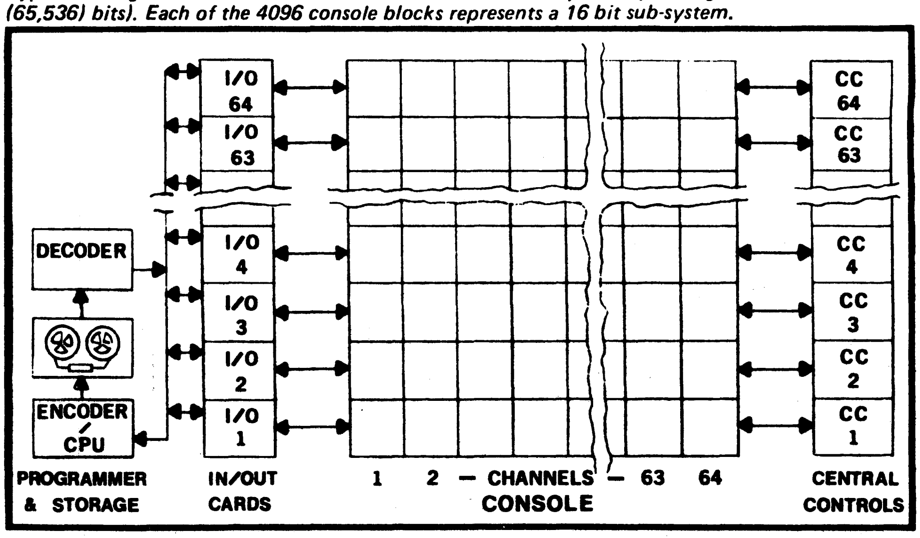
Typical configuration of Allison Research MEMORY PLUS system operating at maximum capacity (65,536) bits). Each of the 4096 console blocks represents a 16 bit sub-system.



Programmable mixing techniques have presently reached the point of industry acceptance where I think it is safe to assume that the art is here to stay.

In the design of our first generation systems, we had no previous experience to draw upon, so we went ahead in the direction we saw as most fit. At that time, our design goals were, basically, to provide the audio system with a means of remembering the positions and motions of the console controls. It was not difficult for us to foresee at that point that the industry would demand automation far beyond the control of levels, so we configured a system which could handle up to 256 analog functions, a number which represented just about the maximum possible capacity, consistent with usable scan times.

The workability of Allison Research first generation systems can be attested to by literally hundreds of hit records produced on consoles whose automation ranges from "level only" through nearly "total automation," in most continents of the world.

The purpose of this article, however, is not to discuss past performance, but rather, to pick it apart and find the shortcomings, in an effort to prescribe a more refined second generation approach.

SPEED vs. CAPACITY

The paramount key to improved system performance lies in the method employed to store the data itself. In early devices it was logical to time-share the functions, in a simple sequential fashion, and serially apply the scans of data to the tape machine for storage. The limiting factor here, of course, is the length of time required to make one scan of the console elements.

More functions = longer scan times.

It was this speed versus capacity situation which caused us to employ the unique quinary (5 level) coding in our first generation programmers. The combination of quinary encoding and word by word validation produced accumulated delays on the order of one-sixth that obtained by bi-phase, frame validation methods, and made complex degrees of automation possible.

However, the somewhat analog nature of the code placed rather stringent requirements on the alignment and general quality of the storage medium and, of course, the delays, while drastically reduced, were still a limiting factor.

ENTER PRIORITY ENCODING

Upon re-examining the data requirements for mixdown purposes, we found that simple sequential scanning is not the ultimate answer. At any given instant, most parameters of the console will be found to be stationary, or not moving. Thus, there is no point in wasting valuable time in constantly refreshing these static parameters with new data, until such time that they are changed.

This premise holds true regardless of the complexity of the system. If we were to configure a system which was able to detect changes in the console controls, and give changing parameters priority over the static ones, there would be no degradation in the effective speed of the system as more and more functions were employed.

To further understand this principal, let's analyze the operation of a really complex console which has, say, programmable levels, echo, panning, E.Q., noise-gating, limiting, assignments, patching, etc.

In spite of the fact that several thousand automation functions might be employed, operation would still consist of changing a level here, an equalizer there, and so forth. It is unlikely to expect more than two or three parameters to change at *any given instant.* Even in the unlikely event of the simultaneous change of 20 or 30 parameters, the delays would be negligible.

With this improved base to work with, it now becomes practical to utilize a more stable form of coding the data on the magnetic medium without encountering the previous problem of speed vs. capacity.

THE ALLISON 65K PROGRAMMER

In configuring the new 65K programmer, we chose to define the overall system capacity as 65.536 binary bits, a number which, although certainly sufficient to handle the most complex console, we do not consider as unrealistic in view of today's rapidly changing technology. The data stream consists of individually addressed and validated digital words, each word containing a 12 bit address, 16 bits of data, a parity bit and a parity time interval. Each such word is normally associated with some particular location in the console such as "channel No. 3 level, mute, solo and group assignment sub-section." The 12 bit address defines the systems capacity as 4096 sources of data, each source having a 16 bit data capacity.

The addressing system is broken down to matrix form such that the console, or other programmable source, is thought of as having 64 rows of data producing elements, each row having 64 discreet 16 bit locations. (64 rows X 64 columns = 4096 locations.)

For purposes of standardization, we define the first 48 columns of any row as being audio channels No. 1 through No. 48, while the remaining 16 columns are designated as master functions. (E.G. group masters, echo return channels, etc.)

Standardization is further implemented by defining the function of each row, as far as present peripherals allow. Row one, for instance, is designated to perform the task of programming faders, mutes, solos, input assignment and group master assignment. Row two handles echo systems, Row 3 performs the panning functions, etc., etc.

In short, the programmer is configured to handle consoles which, in standard form, contain up to 48 channels, and which may contain as many as 64 programmable sub-systems per channel. Obviously, by employing double rows, a 96 channel console may be implemented.

PRIORITY DETECTION

In order to determine which, if any, data sources are changing, it is necessary to examine all data in the system at periodic intervals. To achieve the minimum possible delays, we chose to examine the entire console at a rate coincident with the time required to encode one word onto tape. In other words, as one word is being encoded we search the entire system for changes, so that any such change may be encoded as the next word.

The time alloted for this tandem cycle is 3.2 milliseconds. (Since the thought of analyzing 65,000 bits in 3 milliseconds brings certain speed limitations to mind, a parallel processing approach is indicated.)

In practice, each row of data is processed by an individual in/out port which loads up the 64 potential 16 bit words once each 3.2 milliseconds. As each word is loaded it is compared, bit by bit, to the corresponding word received on the last examination cycle. If any difference exists, the address of the word in question is notated in a Random Access Memory (RAM), while the data itself, for each word, is stored in a second RAM.

Concurrent with this loading and comparing action (a parallel operation performed by the in/out ports), the master encoder continually searches the priority detection RAMs of all in/out ports, in successive approximation fashion, and locates the address of the next word to be encoded. When encoding time comes, the address is known and the latest data pertinent to that address is available in RAM memory. If multiple priorities exist, they are serviced in essentially the same order as received, with data encoded being the latest data pertinent to the changing parameter.

A secondary function of the priority system is to locate and notate word locations which have corresponding console sub-systems, so that unnecessary time is not spent in processing words

Two levels of priority then exist, "Priority one" being words which are changing and "priority two" being static words which are really there.

In order to assure that a constant stream of changing words cannot completely lock out the encoding of static parameters, the encoder is configured to allot at least one word in four to the performance of a sequential encoding of static words.

The net result is that any change in any console sub-system will be recognized and encoded within 3.2 milliseconds, assuming a singular change. In the event of multiple changes, they will suffer a one time delay of 3.2 milliseconds per 16 bit word. As for the accumulation of delays with multiple passes of tape, the individual word validation method puts decoded data back into the console immediately. The total accumulated delay, taking into account the effects of the non-synchronous encode/decode cycle, comes out 4.8 milliseconds, regardless of the number of functions employed.

Although multiple simultaneous priorities undergo an access delay of 3.2 milliseconds per word, no additional accumulation delays are incurred beyond the nominally stated 4.8 milliseconds.

In comparing these figures with those of first generation Allison systems, it can be seen that 65K series programmers, even at full function capacity, operate at 1/20 to 1/30 the delay time of a 256 function first generation system.

It should be noted that this performance is obtained with simple data storage on spare tracks of the master music recorder itself, and without any synchronizing equipment or slave data storage mechanism. In facilities where synchronizing equipment is employed, accumulated delays may be eliminated entirely by the simple expedient of advancing the data playback head 4.8 milliseconds, with respect to the record head.

DATA STORAGE MEDIA

Basically, there are three catagories of data storage media from which we might chose. Each has its advantages and its draw backs. They are: Tracks of the master audio recorder, synchronized data tape recorders, and disk or drum memories. The master music machine offers the simplest method in terms of operating complexity and cost, and assures absolute synchronization, together with the guarantee that the data will not become misplaced in storage. The disadvantages are the imposed limitation on the number of separate mixes that may be stored due to track shortages, potential leakage of the code into the audio, and the necessity to give up at least two tracks of the audio machine.

Synchronized data tape systems alleviate some of the shortcomings of audio tape storage, but tend to increase the operation complexity of the system as well as to add costs for synchronizing equipment, digital tape recorder, and tape materials themselves. An often overlooked drawback of this storage medium lies in the inconvenience encountered in storing, filing and generally keeping track of which data tape belongs to which music tape.

While disc or drum memories are highly suitable for general purpose computer usage, this author is not strong on their use for the type of system herein described, because of their finite recording time, relatively high cost and, again, the storage and operational inconveniences.

With these thoughts in mind, Allison has chosen *not to limit* their second generation equipment to one specific storage medium, but rather, to configure it to work with any of the common data storage methods.

TAPE MACHINE REQUIREMENTS

When audio recorder storage is employed, the required bandwidth is 10 kHz (3dB point) the signal to noise or signal to cross-talk ratios must be in excess of 15dB, and speed variations requirements are plus or minus 50%.

In contrast to first generation programmers, which were *somewhat prone* to the production of error signals when subjected to excess amounts of cross-talk, improperly aligned tape machines, or defective tape, second generation 65K programmers are capable of tolerating the most adverse of conditions with absolute freedom from error production. Because of multiple methods of parity validation, on a word by word basis, operation is on a strictly go/no go basis, that is to say that the decoder will either produce the exact digital word that was encoded, or it will not decode at all.

It might also be noted that while first generation programmers wrere required to skip an entire scan in the event of a dropout, the 65K decoder, being individual word addressed, needs only to skip the word affected by a tape defect. This is coupled by its ability to properly decode drop outs of approximately 10 times the depth of 256 series equipment.

As far as the potential leakage of code into music, the 15dB signal to noise ratio requirement allows the code *to be carried* at extremely low levels, thus effectively eliminating this potential problem.

CONSOLE INTERFACE

Since first generation automation programmers were primarily designed to interface to adaptions of conventional consoles, their method of data distribution in and out of the console was in terms of analog control voltages, in the range of 0 to 5 volts. While this approach is probably the simplest and most understandable method, it is far from being the most desirable, particularly for complex consoles. The most profound shortcoming of an analog interface can be found in the path from the decoder – through the console – and back to the encoder. When multiple passes of the tape are programmed, any parameters which are left in READ MODE are passed through this decoder – console – encoder loop at each programming pass of the tape. While it is expected that parameters left in READ MODE will retain the exact settings to which they were programmed, this may not be the case with an analog interface. Accuracy errors of as little as .2% and or offset voltages of as little as 5 millivolts in the console path, the A to D convertors or the D to A convertors can cause the data to deviate by one or more steps each time it is subjected to the decode – console – encode path.

In first generation Memories Little Helper systems, this potential problem was dealt with by the expedient of employing .1% resistors together with offset adjusting potentiometers on each module. Although these systems do exhibit zero error accumulation with multiple passes, it is obtained only by a careful calibration of the entire system.

If the data is passed from Decoder to console to Encoder in digital form, the path is unquestionably accurate and totally stable without any need to resort to precision components or adjustments of any sort.

The second draw-back of an analog interface lies in the inefficient use of the available data. While a gain control, for instance, requires a finely resolved and essentially variable range of control voltages, many console parameters, such as switch functions, require but a simple on or off action – exactly that which is provided by one bit of direct digital interface. Other console parameters such as equalization can be much more flexibly configured with a digital interface, since the exact number of bits required for the job may be used.

In short, a digital interface allows the peripheral designer all possible control choices from 1 bit control (on-off functions), through 3 or 4 bits (for discreet 8 or 16 position switching), to 8 or more bits (for variable parameters such as level controlling). In each case, the digital interface allows an exactly defined parameter, which is absolutely repeatable and unaffected by component tolerances, offset voltages, ground loops and other analog pests which invariably appear in any console.

THE DATA BUSS

Now that we have settled on a high capacity (65.536 bit) programmer and a digital interface, the only logical next step is to employ a data buss approach in distributing data around the system. Buss oriented systems were created and perfected by the computer industry for the same reasons that indicate their use in programmable consoles. Instead of running tens of thousands of pairs of wires around the system, we can take a common wiring buss, containing only enough wires for the 16 bits of data and the 12 bits of addressing (28 wires), and time share it, in the same fashion that we time share the tape machine for handling all of the data. Since each console sub-system has a unique address (derived from the 12 address bits), it can be strapped across this common buss and instructed by two additional bussed wires. These instructions would be: Send data and receive data. The send data command means that the programmer wants the addressed sub-system to place its data on the buss for processing, while the Receive Data command indicates that the programmer is requesting the addressed sub-system to pick-up the data which is presently on the data buss. When a sub-system is neither sending nor receiving data, its connection to the data buss is effectively opened, thus allowing free use of the buss for other sub-systems.

A further benefit of such a system lies in the fact that, since all sub-systems share a common connection and have send/receive capability, it is now possible to re-configure the system, at will and without wiring changes, to transfer data from sub-system to sub-system or in and out of peripherals which may be added at some future time. This capability makes console design a whole new ball game, wherein early obsolescence may be avoided as newer forms of control are formulated. Instead of re-wiring, the console may simply be re-instructed.

While it is quite possible to connect an entire system comprised of 4096 sub-systems on one common 28 wire buss, we have chosen to employ multiple data busses in the structure of our second generation systems. Our purpose was two-fold. By using multiple busses, we are able to achieve faster processing speeds, and we are able to de-centralize the system for increased reliability and ease of trouble shooting.

As was discussed earlier, the 65K programmer defines up to 64 rows on each of 64 columns, or channels. In our final configuration, each row is implemented by an "in/out" card in the programmer, and a data buss per row. All level controlling sub-systems (Row No. 1) then, are on a common data buss, while echo sub-systems are on a second data buss and so forth. By processing all rows simultaneously, in parallel fashion, we are able to fully instruct the entire system once every 3.2 milliseconds.

In short, the data buss approach to console data distribution offers increased system flexability and markedly reduced cost, since control wiring is reduced to a small number of wires and connectors, which are bussed, rather than discreetly wired. System reliability can be increased owing to the vastly reduced number of wires and connectors employed.

REQUIREMENTS OF THE CONSOLE CONTROLS

In order to configure a truly usable second generation system, we felt the necessity to first define what should be expected of the console, from an operational standpoint. The following is a list of what we feel the requirements are:

1. All programmable parameters shall be individually accessible, in terms of functional control and in terms of mode (READ/WRITE).
2. The status of all parameters shall be visually monitor-able, whether in READ or WRITE mode.
3. The operation of all controls shall be instinctive, with no requirement for the operator to speak "computer language."
4. All controls shall be capable of being both electrically and *physically* operated by both the automation system and by the operator. This requirement is not met by null lights and metering devices, which simply instruct the operator to physically move the controls. The automation must actually make the control settings.

4A. In setting system controls, the automation must not employ servo motors, relays or other mechanical devices.

5. All parameters shall be capable of being controlled either one at a time, or on a master basis, as an adjunct to decreasing set-up time of the console. Examples of this would be "clear all mutes," or "assign all channels to Stereo Center," or "set all Equalizers to Flat Response."

5A. All automation modes must be controllable either individually or on a master baiss. E.G. "Master Write"

6. All controls shall be large enough for simple human intervention and shall be easily reached and identifiable by the operator.
7. The console face shall be small enough to allow the operator full control of as many as 64 channels or groups, without having to stretch or move from his operational seating position.

While these requirements might appear to be impossible to achieve, it is our conviction that each one of them is urgently necessary for the successful structuring of complex programmable consoles.

Now, let's analyze these requirements and see what the solutions might be. If we are to meet requirement No. 1, we must add READ/WRITE controls to *each parameter* which is programmable. No.5A rules out simple two position mechanical switches, since they cannot be conveniently operated from an over-riding force. No. 2 requires that we add some form of visual display to each parameter so that we may know its position when it is under automatic control. No. 4 wipes out most controls found on a conventional console, since it dis-allows any control whose electrical setting bears a direct relationship to its mechanical position. No. 5 only further enforces this implication. Since requirements No.1 through No.5 indicate that we must add controls to the conventional console, while No.6 and No.7 indicate that we must remove controls, the answer is very simple and undeniable: It is impossible to successfully automate a complex console without employing radical conceptual changes in its control.

THE ANTI-REDUNDANT CONSOLE

This problem of too many eggs for the basket to hold them, is by no means a new one. It has been solved time after time in the past, and those who have been unwilling to accept the transition have died. How many of you remember those old rotary calculators with about 15 rows of 15 keys, so that you could multiply big numbers like a million, at a cost of around $1000.00. One day some person figured out that you could multiply numbers like 10 to the 99th power, using a simple 10-key keyboard and a few other buttons, for $29.95. Needless to say, they don't make rotary calculators with a key per number anymore.

The same solution is painfully evident in the interface between man and his recording console. The proposition of a control per parameter is preposterous, both in terms of suitability and cost. Since man has but two hands, how can we sanely justify thousands of controls, too small to hold in the fingers and spread out over an area twice the reach of a man's arms?

Although a complete mixdown console could, and perhaps some day will, consist of a single keyboard coupled to c.r.t. displays, it is our contention that the industry is not prepared to go that far into the concept, this soon. (Requirement No. 3)

We feel that our starting point should be a physical position for each track. A position that the operator may identify with, and upon which he may pencil in the instrument, or other music component, which it concerns. Now, if this position includes, among other controls, a momentary push button, this single "button per track" may be used to assign a nearly limitless number of parameter changes to that track, via central control devices, which may be shared by all channels. This premise is particularly effective when applied to parameters which normally require large physical areas on the channel module and large numbers of controls, yet are parameters which are not adjusted very often in the course of a mixdown.

The first such parameter which comes to mind is the output assignment section. If a centrally located matrix of buttons, numbered 1 through 32 is employed, the operator may make his output assignment by the simple expedient of pressing the singular button on the desired track, and manipulating the central matrix for the desired configuration. The operation is totally instinctive, since the operator is allowed to think "assign overhead drums to outputs 7 and 11." If full time visual monitoring of the assignment is desired, a matrix of LED's or other indicators may be located on the channel strip, to indicate the assignment status.

By adding a few more buttons to the central matrix, we can easily satisfy all of the previously stated requirements. Read and Write buttons allow us to selectively place any channel in either mode. A "Clear output Assign" button allows us to clear all 32 outputs with one operation, while an "all channels" button allows all channels of the console to be ganged, or dealt with simultaneously. The singular operation of pressing both the "CLEAR OUTPUT ASSIGN" button and the "ALL CHANNELS" button can clear the entire console, a task which could take as many as 1024 operations on a conventional console. Care must be taken in the design of the central control, however, to make catastrophic changes of this sort well protected from accidental execution. This is a simple engineering job with many solutions.

The savings which can be directly attributed to this configuration of output assignments alone add up to approximately 4 square feet of console space, and the elimination of around 1000 switches, with an inherent increase in system reliability due to the grossly smaller numbers of parts.

From a monetary standpoint, the savings effected by the removal of mechanical parts will more than likely pay for the automation added.

In structuring our second generation system, we have chosen this central control concept as our method of achieving the requirements which we initially outlined above. While not all parameters (Faders in particular) lend themselves to the implementation of this technique, most of them do. While we, at Allison Research, are currently refining methods of dealing with all parameters of a fully programmable system, we will limit the remainder of this article to the two areas in which we have actually reduced theory to production line equipment. These areas are: The equalizer section, and the level controlling section (which includes muting, soloing, input assignment and group assignment).

THE GREAT EQUALIZER

In configuring a fully programmable equalizer, we first had to make the decision of discreet or variable control. We chose discreet selection of frequencies and gain parameters as being the only justifiably correct method of control, since the digital art is one which specifically deals with discreet steps. Even when seemingly variable parameters are controlled by digital means, they are in reality a series of discreet steps. Generally speaking, the Great Equalizer is an addressable device which is controlled by 32 bits of digital data, and is TTL or CMOS compatible. Its programmable parameters are as follows:

1. An 8 frequency 18dB/octave High Cutoff filter (1.2kHz to 12kHz)
2. An 7 frequency Hi Eq. section with peak/shelf selection and ±15dB of Equalization in 15 steps (820Hz to 12kHz)
3. An 8 frequency mid Eq section with shelf/Peak/shelf selection and ±15dB of Equalization in 15 steps (220Hz to 3.3kHz)
4. An 8 frequency Lo EQ Section with Peak/shelf selection and ±15dB of Equalization in 15 steps (39Hz to 560Hz)
5. An 8 frequency 18dB/octave Lo Cutoff filter (39Hz to 390Hz)
6. A phase reverse switch, and an in/out switch.

In attempting to interface this rather complex equalizer to the operator's hands, we find that our own requirements call for visual displays of all parameters as well as positionless controls. The "instinctive operation" requirement pretty much disallows the use of a calculator keyboard.

What we're left with is a matrix of illuminated momentary buttons, one for each position of each equalizer section. The Hi Cutoff section, for instance, can be implemented with 8 buttons, one per frequency.

The equalizer fortunately falls on our list of devices which may be centrally controlled. Here is how we handle it.

The central Equalizer control panel contains the above defined matrix of momentary switches, together with LED indicators. It additionally contains a READ button, a WRITE button, a CLEAR E.Q. button and a HOLD DATA button.

Pressing the singular momentary button on an individual channel module causes the equalization presently in that channel, as well as its READ/WRITE status, to appear on the LED indicators of the central control. Since the central control has large buttons and is graphically representative, the act of "looking at an equalizer" is, in our opinion, more accurate and probably faster than looking at a conventional overly dense and graphically inferior console equalizer. If it is now desirable to change the selected equalizer, the operator need only to continue pressing the singular channel button while he makes the desired changes on the central control. From an operational standpoint, the central control, indeed, actually becomes the channel equalizer.

When it becomes desirable to gang-control the equalizers, for instance, when clearing the console, the Hold Data button comes into play. Whenever the HOLD DATA button is held down, the central control is prevented from receiving E.Q. data from the individual channels, but in turn it transfers its data to any channel whose singular button is pressed. Therefore, the act of clearing the central equalizer control to flat response, then simultaneously pressing the Hold Data button and the ALL CHANNELS button causes all channel equalizers to be cleared to this flat position, or any other desired position.

Duplicate equalization to a number of channels may be accomplished similarly by setting the desired E.Q. on the central control, holding down the HOLD DATA button, then pressing the desired singular channel buttons.

As with the output assignment section, we have again taken hundreds of controls off the console face, as well as 3 to 4 square feet of area. We have also increased the reliability through the removal of mechanical parts and the money saved in the process has paid for the automation.

The system is inherently goof proof, since the unitentional operation of any single button will cause no change in the system parameters. Only an instinctive and intentional two handed operation can enter changes. Since all buttons are momentary, it is impossible to forget that some switch which might affect the data to be entered, was set to some unexpected position.

HE FABULOUS FADER

As you might have guessed, our requirement list necessitates replacing the normal console fader with something quite different. In our quest for a device which meets all of the requirements, yet maintains the approximate feel, throw and instinctive operation of a conventional fader, we finally settled on an approach which utilizes a continuous optically encoded belt as the human adjustment mechanism. Positional indication is provided by a 32 element LED array which is located beneath the belt. From the operator's standpoint, the device appears as a 3/4" x 7" panel, which contains a single LED illuminated momentary button, and a slightly recessed slot, measuring some 5/8" by 5". Appearing in this slot, is a single point of light, which indicates the present position of the fader. The light is passing through the optical belt, which is held flat within the slot, yet is moveable by the operator's finger. The surface of the belt is somewhat textured to present a non-slip surface to the operator, and the resulting feel, in terms of sliding force, vertical pressure and horizontal scaling are essentially identical to that of a very high quality conventional fader. The device is not touch sensitive, it is movement sensitive, and the operator need not put his finger at any particular location on the belt to operate it.

If, for instance, a finger is placed, say, an inch below the point of light and is moved up and down, the light point will track the finger movement, maintaining the one inch offset exactly. The point of light, of course, is indicative of the audio gain of the channel, and may be addressed by either the operator (via the belt), the automation system, or other over-riding sources such as group masters, presets or master clearing devices.

The fabulous Fader covers a 112dB range of operation which is incremented in 1/4dB steps over the first 48dB, then 1dB steps over the remaining 64dB of its range. It is, of course, noise free in its operation and immune to analog forms of wear. The belt itself is a continuous splice free lamination of mylar, nylon and polyurethane, and, though its wear cycle is estimated at 10 to 15 years, is quickly and inexpensively replaced. Such replacement, or cleaning, would be indicated in the event of catastrophies such as cigarette burns and such.

The faders themselves are arranged on a conventional one-per-track basis, allowing the operator instant and instinctive access to all system levels.

The singular button/LED at the top of each fader, however, serves as access to centrally controlled modes associated with each channel.

The level control sub-system, a current ALLISON RESEARCH production item, includes the following centrally controlled parameters, all of which are fully programmable and are individually accessible, one channel at a time, or on an ALL CHANNELS basis:

1. Selection of WRITE or READ/AUTOMATIC UPDATE modes, with respect to levels.

2. A Clear Fader position (Fader off), and a TEST LEVEL position (Gain = Unity ±1/4dB)

3. Solo and Clear Solo positions, as well as READ/WRITE status with respect to solos.

4. Mute and Clear Mute positions, as well as READ/WRITE status with respect to Mutes.

5. Assignment of any channel to one of fifteen group masters, together with READ/WRITE status with respect to group master assignments.

6. Assignment of any channel to one of four input sources, together with READ/WRITE status with respect to input assignments.

7. Separate mute/solo systems for Group Masters, as well as READ/WRITE controls for same.

8. Four addressable RAM presets (expandable to 64), each of which can store all settings of all parameters within the level section, and can be loaded or activated either one channel at a time, or on an all channels basis.

All of the above parameters are controlled with a central matrix of 26 momentary buttons with LED indicators, which is used in conjunction with the singular button/LED located on each channel or group fader.

A unique bi-directional visual communication system allows the operator the required complete visual monitoring of all system parameters.

Since space does not permit this article to continue on much longer, I will briefly describe the partial operation of the system, and leave it up to the reader to piece together the remainder of its operation.

As with the Great Equalizer control, pressing a singular channel button causes the Level Sub-Section central control to indicate all parameters currently associated with the selected channel. If it is desired to change parameters, the desired buttons on the Central Control are operated while the channel button is held down. Reverse communication is also possible, on the following basis: Assume, for instance, that the operator wishes to see which, if any, channels are muted. Pressing the MUTE button on the Central Control causes all channels or groups which are in the muted state to indicate that fact, via their LEDs associated with their singular buttons. The same philosophy applies to all parameters associated with the system.

A further visual communication system exists in that the linear LED array associated with each fader does, indeed, indicate the actual gain of the channel, and includes the effect of mutes, solos and group masters. If a channel is solo-ed, for instance, all other channels will indicate an off condition with their LED arrays. Similarly, if a group master is moved up and down, all channels assigned to that group will indicate this up and down motion on their LED arrays. This, of course, is a direct and instinctively correct visualization of what the effects of the controls actually are.

It can easily be seen that an attempt to configure this degree of programming versatility, on a conventional basis, would result in the addition of some 35, or so, controls and indicators to each module — an addition which would render the system incapable of meeting the requirements which we have listed previously. This is to say nothing of the drastically increased costs and gross operating complexities which would inherently result.

CONCLUSION

Programmable audio systems can offer the user a powerful tool in the execution of his creative work. The real advantage of such a system lies in the operator's ability to shape the control of his precious audio to degrees which heretofore were impossible because of physical limitations.

In configuring complex systems, we, the manufacturers, have the responsibility to fulfill this promise on a basis which does not trade all of our gains for detriments, such as unmanageable size, inoperable controls, over-complexity in the human interface, unnecessary compromises in the system's capabilities, and above all, unbearable increases in the cost of the system.

We, at Allison Research, believe that this can come about only by approaching the problem with a logical and flexible pattern of thought, which is unencumbered by the shackles of conformity to past methods.

To Our Friends in the Music Business

As some of you are aware, Tin Pan Valley Corporation is dedicated to the establishment of the worlds first community specifically designed for people, like ourselves, who are actively engaged in the creative arts.

Our purpose in undertaking this task stems from our conviction that creative people, particularly music people, desire an alternative to the lifestyle and business climate offered from conventional sources.

Tin Pan Valley Corporation was formed some fifteen months ago, by active members of the music industry, to pursue these objectives. We have since located and purchased the site, a picturesque 234 acre valley, located at the outskirts of metropolitan Nashville and geographically isolated from the path of progress by high rolling hills and dense forests.

Our master plan calls for a maximum of 28 business sites and 25 homesites which are to be privately owned and legally deeded to their owners. There is ample acreage nearby for the future expansion of residential areas. The business sites are restricted to the general realm of creative arts, and are tightly controlled in terms of aesthetics and the enhancement of the natural beauty of the land.

A typical recording studio site will have 2 to 3 acres of land and will be connected to other sites by both natural walkways and by low density roadways. Eating facilities are planned, as are certain recreational facilities.

The focal point of the community is a 3½ acre community owned park, complete with a giant old tobacco barn which is much to beautiful to be torn down.

The general theme of the project is a low key, walk to work, somewhat old fashioned village affair which bustles with creative activity. A true working community whose inhabitants are proud of their environment and of their chosen line of work.

Since our project is one which we feel cannot be entrusted to conventional land developers and money men, we have spent the past 15 months learning the ropes of the real estate game. We have also spent substantial amount of our own money in purchasing the land and in engineering our master plan. Tin Pan Valley now has the support of the necessary political figures and we have gained the knowledge of what is required to make this most unusual endeavor a reality. This is where you come in.

If you believe in what we're doing, you can do us an incredible service by putting your feelings in writing and mailing them to us. If you are interested in joining us, we need to hear from you now! If you're planning a studio, a publishing house or a pottery shop for that matter, you owe it to yourself to find out more about what we're doing.

Whatever your interest might be, I can't emphasize strongly enough the importance of your relaying it to us without delay, as it could have a profound impact on the future of Tin Pan Valley. We will, by the way, be at the L.A. A.E.S. convention.

Thank you.

Paul Buff
Allison Buff
Bob Todrank

P.S. This ad has no reader service number.
If you want literature, please write or call.

Tin Pan Valley

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