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AUTOMATION AS APPLIED TO THE MIX DOWN PROCESS

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INTRODUCTION:

Automation has proved its merits in several industries where a repetitive predictable process must be performed. The film and television industries, for example, have recently begun to make heavy use of automated techniques to increase efficiency. The increasing complexity of contemporary studio recording techniques, particularly in areas where repetition is prevalent, suggests automation may have an application here.

Existing forms of automation have been used in the recording process mainly to simplify console set ups, monitor switching, routing assignments and tape machine track modes.

The most efficient use of automation, however, is realized during the mix down process. Here, the physical limitations of the man/machine interface are most apparent. Numerous dynamic functions must be performed by one person on a real time basis with split second timing accuracy.

There is usually a trade off — function accuracy vs function complexity vs the quantity of attempts to achieve the most acceptable mix.

AN AUTOMATED MIXDOWN CONSOLE:

An automation system has been developed to overcome the majority of problems described. The system consists of two basic pieces of hardware -

1] a programmable mixing console
2] a digital encoder/decoder

The combination of these two systems achieves what might be called a console with a memory. The system stores in memory information which will allow it to duplicate a mix as often as required. Each time the mix is repeated, changes may be incorporated with the final composite mix always available for instant recall.

The system has numerous advantages to producers in particular. Complex mixes can be built-up step by step. If one track of a multitrack master must be re-recorded, the mix information need not be destroyed. When the new track is satisfactory, the programmer provides the new mix instantly.

OPERATION:

The operation of the automated console is as follows. The programmer and console are interconnected. The programmer's encoder output is connected to the record input of a vacant tape channel on a multitrack recorder. The playback output from this channel is connected to the decoder input.

The multitrack master tape is played back through the console in the conventional manner. At the same time, the programming track is placed in the record mode. As the tape is mixed down manually, the encoder monitors all of the control voltages and generates a digital code which is recorded on the master tape. When the tape is played back a second time, the programmer decodes the digital signal into the programming information required to duplicate the mix.

If particular levels are not suitable, new manual information is mixed with the existing control information and re-encoded into a new digital signal and recorded on a second vacant tape track. This process may be repeated as often as required with the encoded information alternating between the two coded tracks.

THE PROGRAMMABLE CONSOLE:

The programmable console differs from a conventional console in that certain functions may be controlled externally. In addition, the manual controls for these functions may be read externally.

Programmable functions fall into three basic categories: switching functions, level functions and equalization functions.

Switching functions include input and output assignments, bus selections, signal routing and monitor modes. These functions are normally handled by FET switches driven by logic circuits.

Level functions include input levels, submaster and master levels, stereo pan and quad position controls, echo levels and monitor mix levels. These are handled by voltage controlled attenuators. Stereo pan controls use two attenuators with an encoding matrix to permit a single control voltage to control both attenuators. Quad position controls use four attenuation elements and an encoder to permit two control voltages to establish x, y position.

Equalization functions may be handled by one of two methods. The first method uses voltage controlled filters. This method provides a continuously variable equalizer but requires considerable memory space. An alternate method provides an equalizer with discrete incremental positions. This is complex but requires less memory space. In the latter case, each particular equalization setting can be described by a digital word of a fixed bit length.

Of the three basic categories only the level functions have been implemented to date. This has required the development of a high performance voltage controlled attenuator.

Most conventional approaches to attenuator design suffer from one or more of the following problems:

1] Poor linearity
2] Poor tracking
3] Thermal drift
4] Excessive leakthrough in the inifnity position
5] Audio performance degradation in the form of distortion, noise, poor frequency response or poor dynamic range

An attenuator has been designed to overcome all of these problems. The final design provides an attenuator with a linearity of better than 0.25 dB over the first 50 dB of the total range. This linearity assures a tracking accuracy of better than 0.5 dB. The unit will experience some thermal drift at the low end of its range if the ambient temperature rises considerably due to heat sources such as power supplies and lamps. In a proper environment, however, it does not drift any appreciable amount.

The maximum attenuation of the element itself is approximately 80 dB. Preliminary audible tests indicated this to be insufficient in console applications. An off gate subsequently was added to the circuit which provides in excess of 110 dB attenuation in the infinity mode. This off gate takes the form of an FET driven by a comparator sensing attenuation control voltage. Harmonic Distortion is below 0.1% over a 60 dB range of attenuation, frequency response is DC to 50 kHz and noise at the output is approximately -90 dBm at any attenuation except off, when the noise is a function of a buffer amplifier alone. The device uses a transconductance principal and is packaged in monolithic format.

In the console, each fader, rotary control or x-y positioner assembly generates a control voltage between 0 and +10 Volts DC. Each attenuator assembly may, at the option of a panel switch, be controlled by either the DC voltage from the decoder or the DC voltage from the manual controls. The encoder will constantly monitor the DC output from the manual controls.

A centrally located comparator display may be switched to compare the DC voltage from the programmer with that of the manual control. The display allows the mixer to adjust the manual control to the same value as the decoder output and thus be able to switch between automatic and manual modes without a level change. This feature is important when modifying mixes where only a small portion of a total mix must be changed.

THE PROGRAMMER:

The programmer contains within a single rack mounting card enclosure an encoder and decoder capable of programming up to 64 functions. The master control logic is designed in such a way that a single standard mainframe can be supplied in function capacities from 16 to 64 channels. The programmer may be field modified up or down in capacity by simply plugging or unplugging input/output cards. Each set of cards contains circuitry for 16 channels permitting capacities of 16, 32, 48 or 64 functions.

The programmer uses advanced MOS and LSI circuitry techniques to achieve the accuracy and function density imposed by the system requirements. The encoder consists of a multiplexer, sampler, ranger, analog to digital convertor and code generator. Finally a circuit converts the code into a signal which may be handled by a conventional audio tape recorder.

The decoder consists of a signal conditioner to accept information from the tape, code separator, digital to analog convertor memory demultiplexer and holding circuit. The internal MOS memory buffers the digital information and will bridge any gap presented by tape dropouts. In the event of prolonged absence of digital input the DC outputs will remain indefinitely at the level established prior to cessation. In addition, a parity comparator ensures that extraneous pulses, noise and transients will not generate an erroneous output.

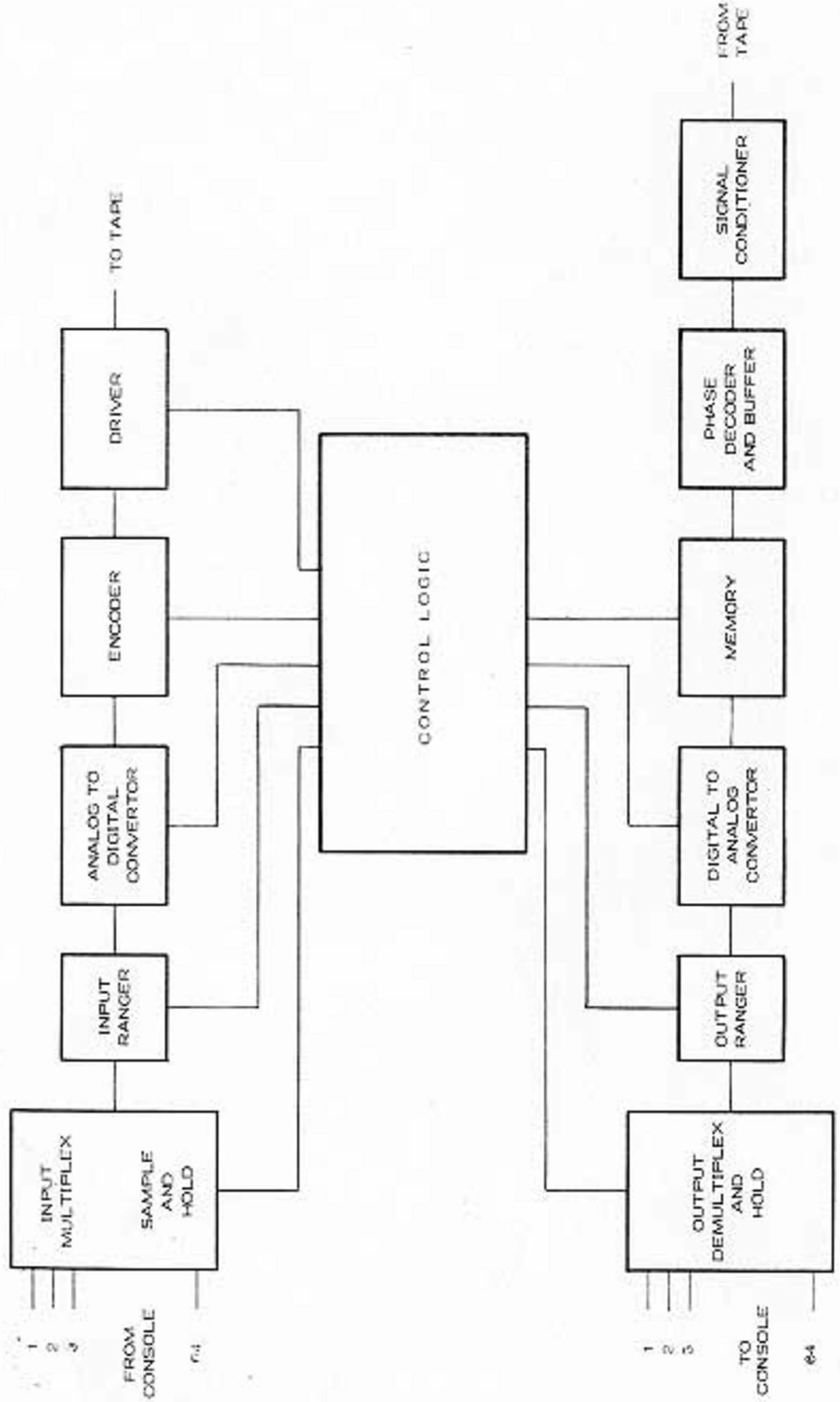
The digital signal at the output may be recorded on any audio tape recorder having a minimum 8 kHz bandwidth and 30 dB signal to noise ratio. The decoder is insensitive to tape flutter, skew and speed variation up to ±5%.

The resolution and accuracy of the complete programmer chain assures a repeat accuracy on the console of better than ±1 dB worst case and typically ±0.25 dB.

CONCLUSION:

Current advances in analog and digital technology have made an automated mix down console both technically and economically feasible. The practical realization of such a console has shown that the system meets the objectives of the project. The digital and analog philosophies employed have proven to be both consistent and reliable.

The future promises even more. Under development are modules to permit automation of more console functions such as switching and equalization. As console complexity grows operation will become simplified. The ultimate goal will see a console with an almost unlimited degree of flexibility allowing a greater artistic freedom yet with a more simplified control panel.



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AUTOMATED REMIX PROGRAMMER
SYSTEM BLOCK DIAGRAM