three packet periods or approximately 5ms. The overall auxiliary mix system delay is about twice that or approximately 10ms.

Additional features of the mix are cut and solo. Cut removes a specified channel from the monitor mix and solo removes all channels except a specific channel from the monitor mix. Cut and solo are applied at the mix node so the application delay is equal to the console scanning delay plus the Ethernet/DMA delay plus two packet periods, i.e., less than 10ms.

VI. Processing

The processing done in each channel node is modeled after the processing section of the Neve VR console. The only difference is the addition of an adjustable delay line in each channel to allow the synchronization of input transients. An example of its use is when differing microphone distances must be compensated for when an ensemble of instruments is recorded in a hall using multiple microphones. Following the delay line the input has a trim control to scale the amplitude of each channel input. Next, there is a phase button which inverts the phase of the input waveform from zero to pi or vice versa. Next there are two frequency adjustable cut filters. The frequency range of the low cut filter is 31.5Hz to 315Hz. The frequency range of the high cut filter is 7500Hz to 18kHz. These filters have 0dB gains in the pass band and roll-off at a slope of -12dB per octave. Next is the dynamics processing section, about which more will be said later. Finally, there is the equalization section. This is a structure of four filters, low, high, mid1 and mid2. The low and high filters can either be set to be shelving filters having gains +18dB to -18dB, or boost filters having ±18dB gains and Q values of 0.71 or 2.0. The frequency range of the low filter is 33Hz to 370Hz. The frequency range of the high filter is 1500Hz to 17kHz. The mid filters have ±18dB gains and Q values of 0.5 to 9.0. The frequency range of the mid1 filter is 190Hz to 2kHz, and that of the mid2 filter is 800Hz to 8.7kHz.

All the filter coefficients are stored as sets of five floating point numbers in the 3Mb of user memory on the RTH board. When a console filter knob is moved a new pointer into the filter coefficient table is computed. Then a block of data is sent via DMA to the local memory of the appropriate channel node. The five coefficients are written to the new coefficient buffer and a semaphore is set to indicate that a change has occurred.

The dynamics processing section is between the two cut filters and the four equalizing filters. The section consists of a compressor/limiter and a gate/expander. The advantage of working

in the digital domain is that present dynamics processing decisions can be made contingent upon the future states of the signal because an entire packet of "future" samples is always available for inspection. Compression is implemented when the input has exceeded a specified amplitude threshold for a specified length of time. When the compression ratio is set at K:1, the portion of the signal to be compressed is decreased in amplitude so that its peak dB level above threshold is K times lower than its original value. For example, if the ratio is 2:1, a signal with peak at 6dB above threshold is reduced to a signal with peak 3dB above threshold. Limiting is compression with a ratio of infinity to one, i.e., clipping. Expansion is the inverse of compression, i.e., a signal with amplitude below a specified threshold is increased so that its new lowest amplitude value is K times higher in dB relative to the threshold than previously. Gating decreases any portion of a signal below a specified threshold level to a new level a specified dB "depth" below the threshold. A second "unmute" level above the threshold can be specified which is the level a signal must attain again before gating is halted.

VII. Analysis

In order to aid the audio engineer in the task of equalization and signal processing in general, the stereo output of the monitor mix is sent as a digital stream to an analysis subsystem. The monitor mix is identical to the master mix unless a solo button is pushed in which case the monitor mix is simply the solo channel (useful for the EQ of a particular channel). The analysis system consists of a PC with VGA monitor and a card containing a DSP32C, 256Kbytes of RAM and a digital interface.

The spectral analysis consists of the following. The left channel, right channel or channel sum is windowed and an 8K FFT is performed ten times per second and displayed in either 2D or 3D graphical form along with the present SMPTE timecode. The linear resolution of the FFT is approximately 5Hz and there are three selectable frequency scales: 20Hz to 20kHz, 2Hz to 2kHz and 0.2Hz to 200Hz. There are also three selectable log amplitude scales: -60dB to 0dB, -90dB to 0dB and -180dB to 0dB. In addition to the graphical display, audio segments can be analyzed and that information captured to hard disk. When a capture is made, the spectral amplitude at each of 512 log frequencies is recorded for each 100ms frame. If the cursor is moved to any log frequency, the SMPTE time corresponding to the occurrence of the amplitude peak in that band is displayed. Then the captured graphics information can be replayed on the screen and halted at the occurrence of the selected peak in order to observe