RECORDED ON CASSETTE NO. 6

Inside the Yamaha GS1 and GS2

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Yamaha's latest polyphonic synthesiser, *the GS1, is to digital synthesisers what* their GX1 was to analogue instruments; very expensive and very good. This state of the art machine is a constructive synthesiser utilising the phase modulation principle. It is capable of producing sounds of much greater subtlety and definition than was possible using analogue techniques. Only the output has been changed (to analogue) *to protect the sound quality.*

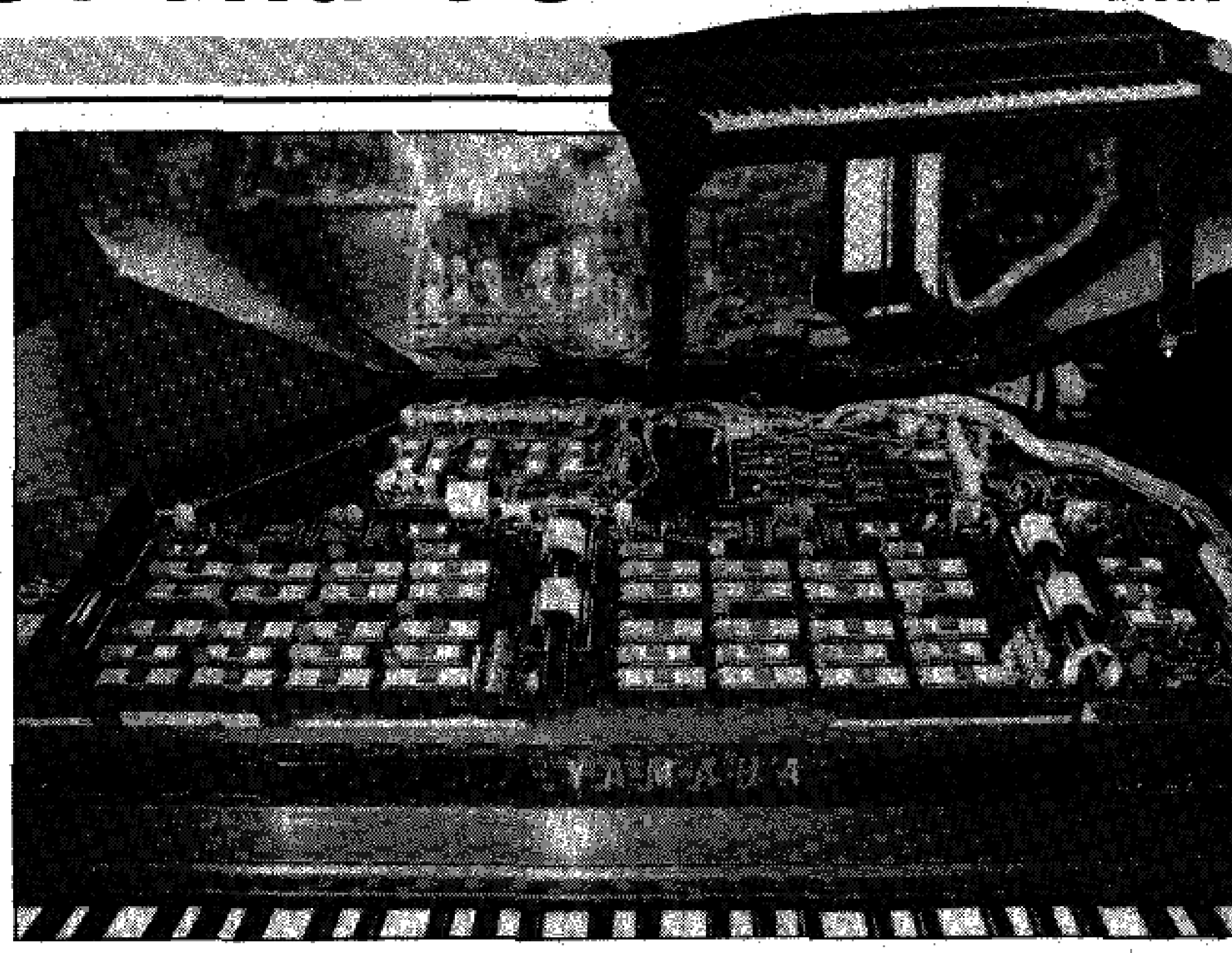
The electronics of the instrument is a rather unorthodox hybrid of radio and computer techniques, and Yamaha have chosen an equally unorthodox casework to house it. The cabinet is in the style of a mini 'grand' piano with a beautiful wood veneer. The control panel at first glance could pass for a conventional piano front. The buttons and control knobs are deliberately unobtrusive, as is the Yamaha name plate in relatively small dull brass letters. This is definitely intended to be a real musician's instrument, and the cabinet styling is an essential component in this very special relationship between player and instrument.

The FM Principle

Nearly all synthesisers create sound timbres by regulating the proportions of the various harmonics of a note. In an ordinary analogue synthesiser, this is done by filtering a harmonically rich waveform, however, only rudimentary control is possible by this means. In addition, non-harmonic timbres are not possible without extra complexity. Better control of the sound may be obtained by building up the individual harmonics *separately, a procedure known as additive,* or Fourier Synthesis. This method is well suited to digital techniques, where each harmonic is computed as a sine wave of the appropriate frequency; the individual harmonics are then added together and fed to a digital to analogue converter. The amplitude of each harmonic may be controlled separately, and the system has great flexibility; *however, a lot of computation is* required.

The GS1 uses a quite different system known as frequency modulation — FM for short — and the circuitry for achieving this is shown in Figure 1. Taking just the top line for the moment, the carrier phase generator and sine conversion block produce a digitally encoded sine wave, whose amplitude may be controlled by the 'output level' multiplier. The sine conversion can be thought of as a ROM loaded with sine values, which is stepped through by a counter (the phase generator); the speed at which this happens is controlled by the key code data, i.e. the pitch information.

The modulator phase generator and its sine converter and multiplier are a duplicate of the above; note that the keycode data need not be the same as that of the carrier, however, or even related to it at all. The sine wave values produced by the modulator section are added to the carrier phase data, with the result that the carrier sine wave is modulated by the modulator sine wave. Two very simple examples of this are shown in Figures 2a and 2b. The dotted sine waves



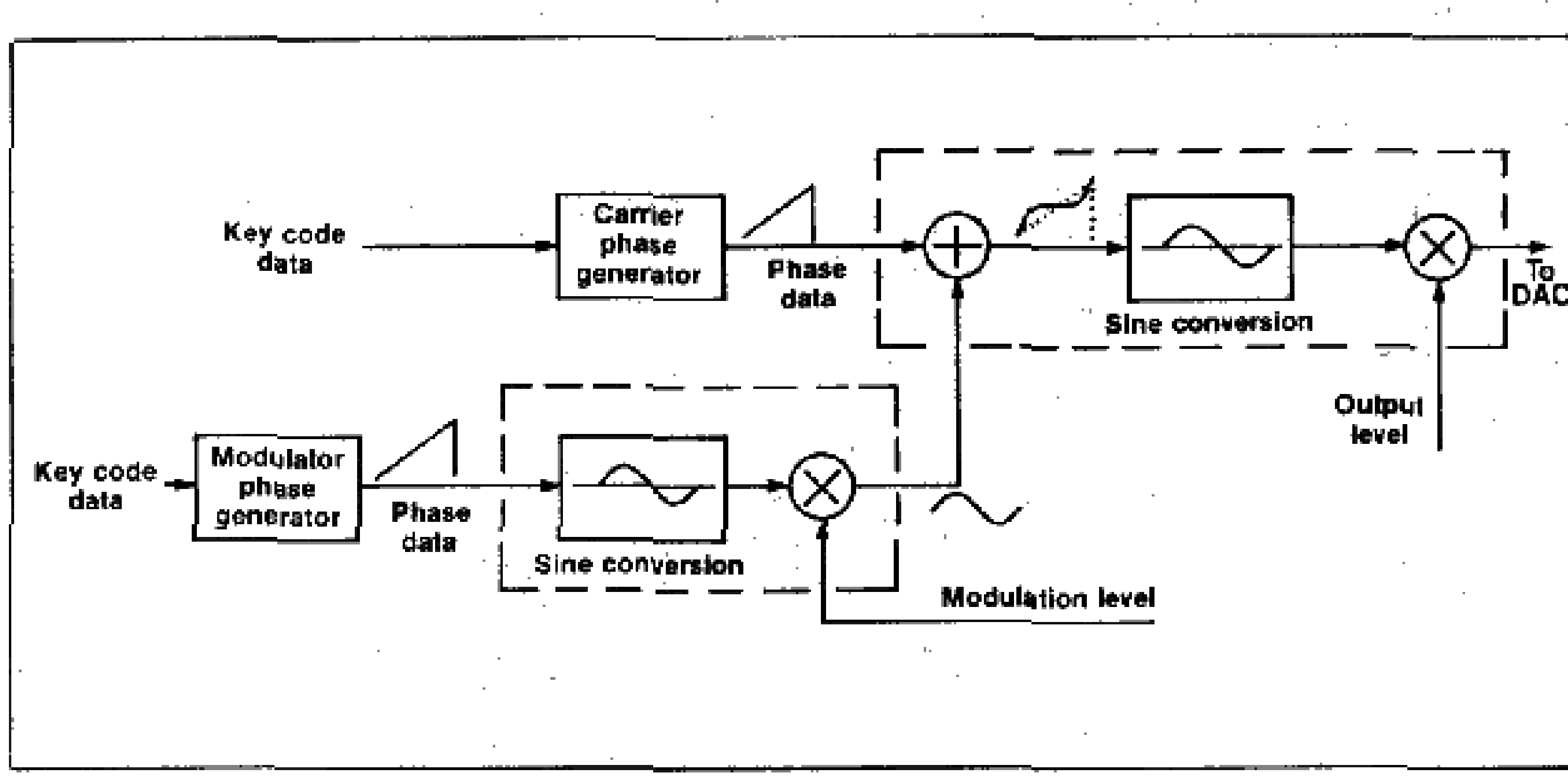


Figure 1. Simplified block diagram of one FM circuit.

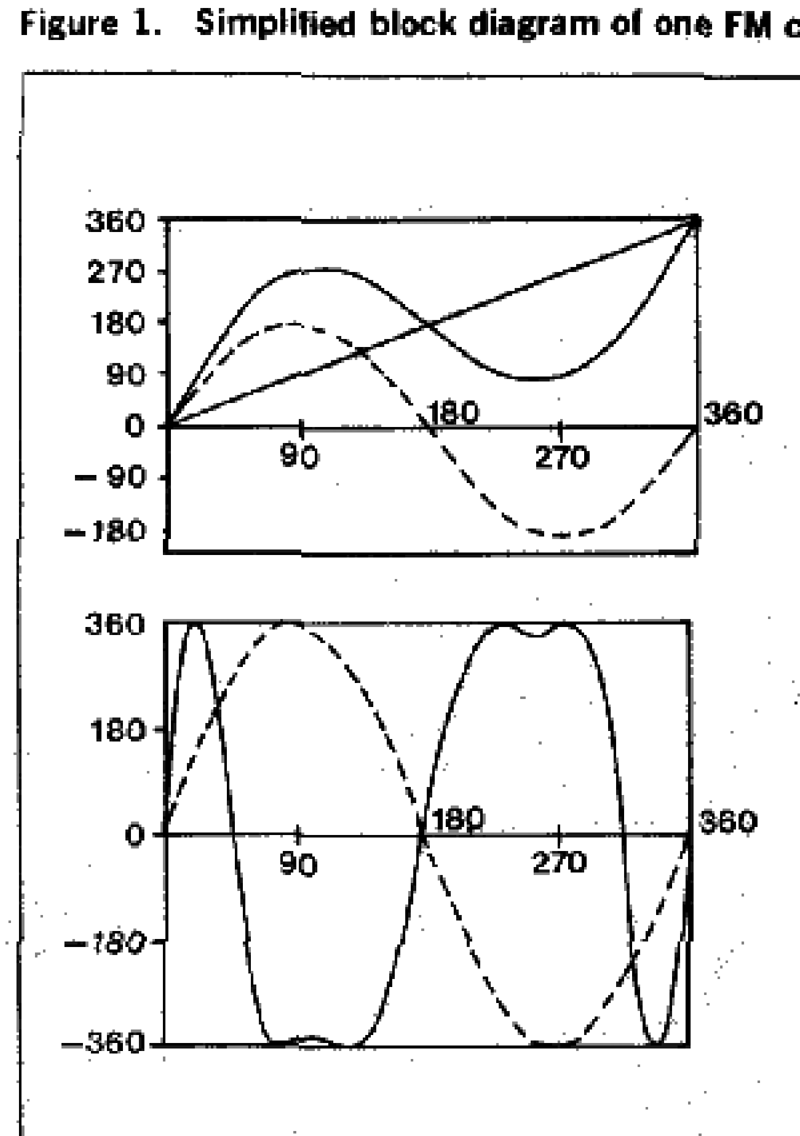


Figure 2a. FM waveform example with modulation frequency = carrier frequency.

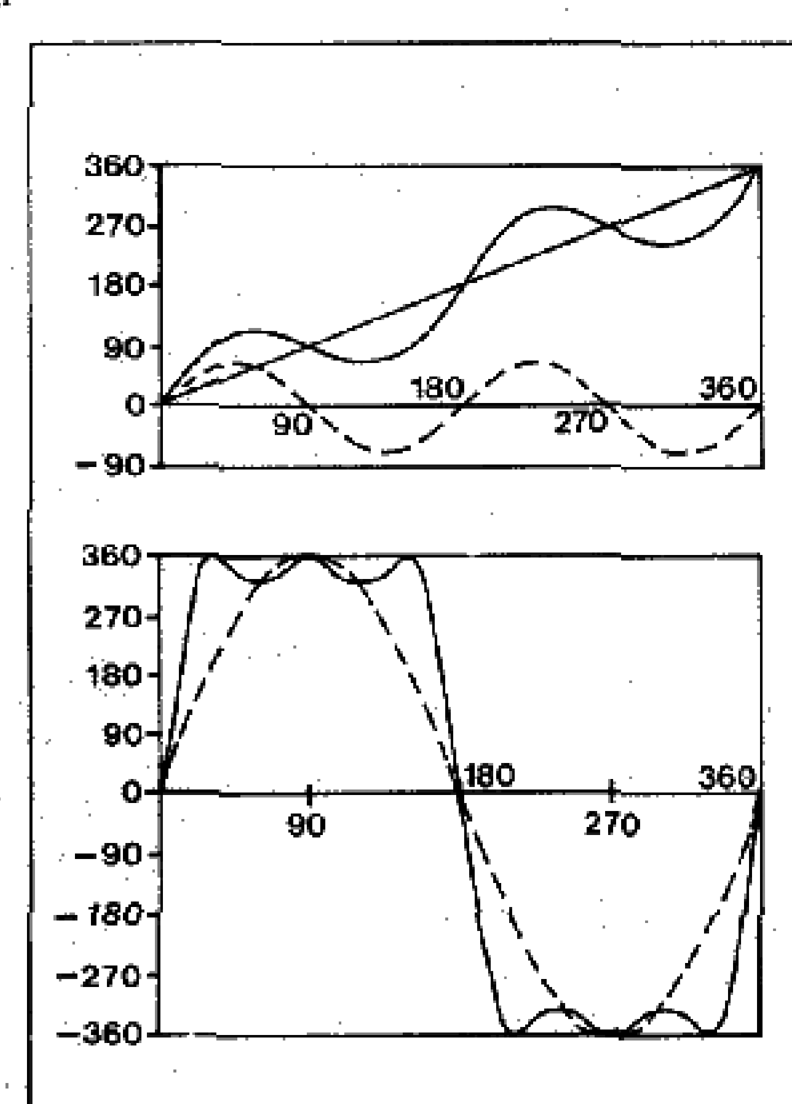
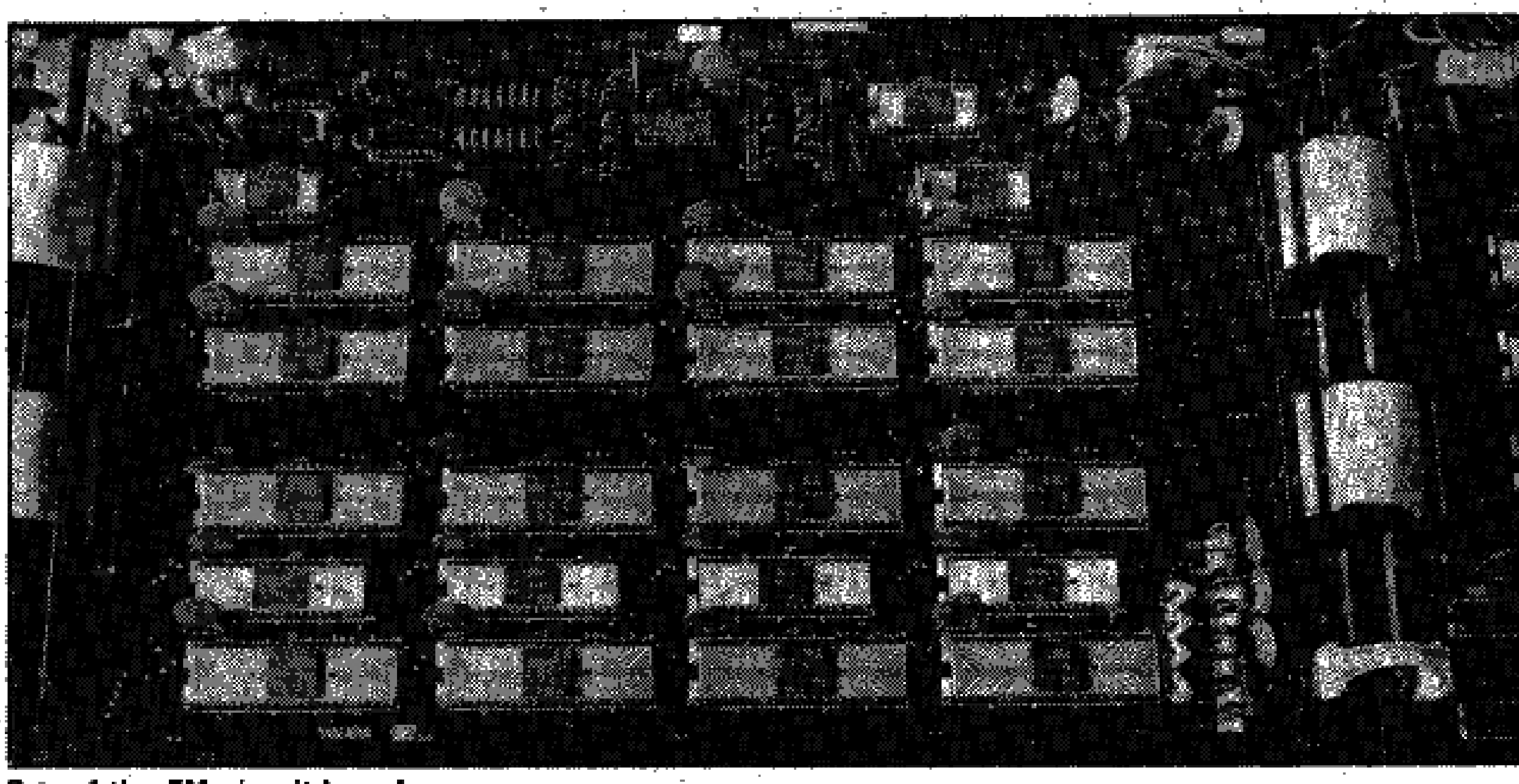
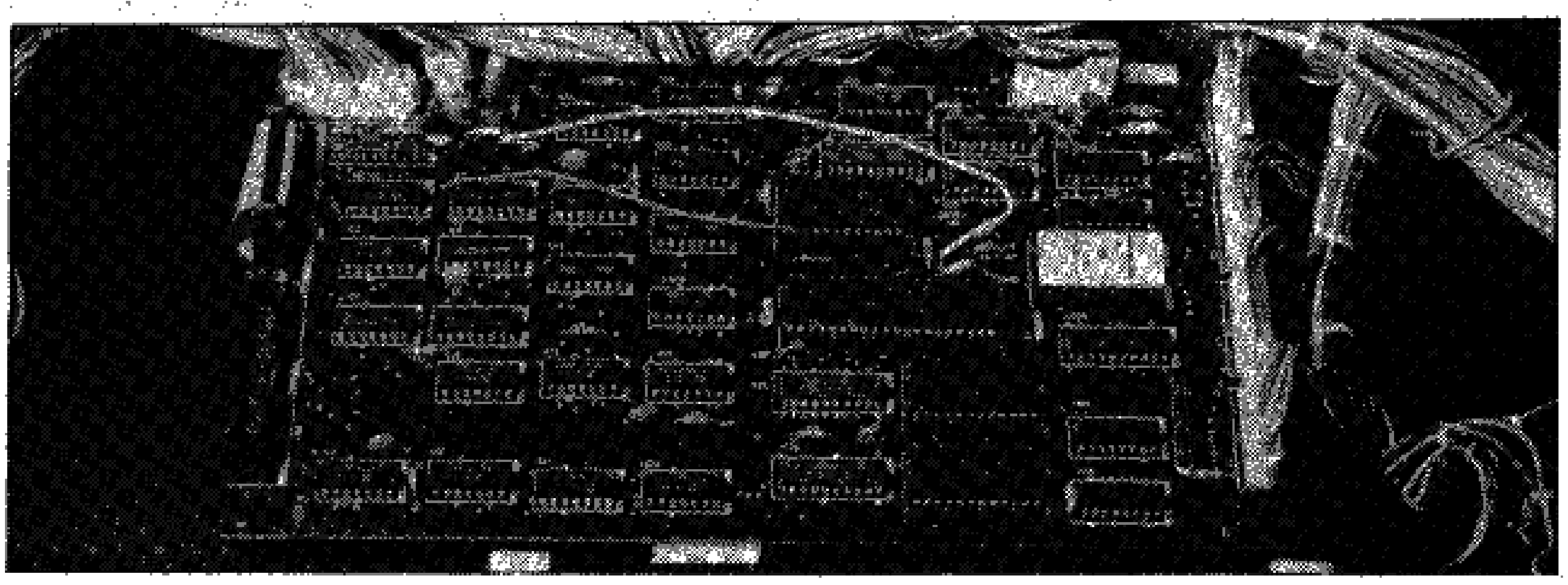


Figure 2b. Waveform example with modulation frequency = 2 x carrier frequency.



One of the FM circuit boards.



Microcomputer board.

represent the modulation frequency in the top graphs, and the unmodulated carrier frequency in the bottom graphs. The solid waveform in the top graphs shows the modulation superimposed on the phase data, and the resulting waveforms are shown solid in the bottom graphs.

Note that in Figure 2a the modulation frequency is the same as the carrier frequency, a situation known as phase modulation, whereas in Figure 2b the modulation frequency is twice that of the carrier. The waveforms may be changed during the sounding of a note by altering the amplitude of the modulation wave under the control of an envelope generator; a second EG determines the volume of the final sound. There are, in fact, four of these FM circuits in the GS1 — two per channel — and two in the GS2, so very complex sounds may be produced.

Although the block diagram of the FM circuit may appear simple, the four FM circuits, eight envelope generators and two DACs in the GS1 require a total of forty Yamaha LSI ICs, and there is more circuitry to come, as we shall see. Without Yamaha's in-house LSI capability it is extremely unlikely that such an instrument could have been built.

The advantage of the FM system is its speed; in fact, the circuitry is so fast that it may be multiplexed 16 ways. In other words, although up to 16 notes may be played at once on the GS1 or GS2, they all share the same sound generator, and this obviously reduces the electronics to a manageable amount. The disadvantage is that control of the sounds by the player is difficult, and so the GS1 is not an orthodox user programmable synthesiser. Its 16 voices are initially input from magnetic voice strips, and the control of this data is handled by a microcomputer based on the 8035. Two 16K CMOS RAMs retain voice parameter data, and program data is held in a 16K EPROM. Each of the eight envelope generators requires 256 bits of voicing data per sound type, hence 16 sounds x 8 x 256 = 32K.

The Keyboard

The 88-note touch sensitive keyboard connects via a diode matrix to a keycoder IC, which is now standard practice for Yamaha keyboards. This chip detects key information from key contacts between octave and note terminals at the IC input. The GS1 has two inputs per note; the key contacts are staggered in such a way that the velocity is sensed by the time interval between those contact points and the particular common octave terminal.

Key code data from the keycoder is output on a serial data line to a channel processor IC, which reorganises this serial data into 16 channels of parallel keycode and initial touch data. The key code data feeds the four FM circuits in parallel, whilst the initial touch data is converted to 10 bit parallel data via the initial touch generator IC. This information is used by the eight envelope controller ICs of the FM circuits.

Aftertouch information is derived from stress sensing bridges located beneath each key. The output voltages of these are sampled by four multiplexer ICs controlled by serial key code data. The voltage samples are converted by an aftertouch generator IC and A to D converter into 8-bit parallel digital data. This data is ORed with the initial touch data lines.

The 88 keys of the GS1 are divided into groups of five keys at the bottom of the keyboard, five keys at the top of the keyboard and 26 groups of three keys in between. Each group has its own envelope generator depth setting. The reactive levels of these carrier and modulator settings is the foundation of the phase modulation system voicing. By using this number of individual level controls across the keyboard range, a very high degree of accuracy and character is achieved by the synthesiser. The other sound parameters are scaled or graduated across the keyboard range, again to have a more natural response.

The Analogue Circuitry

Output data from each pair of FM circuits is added and converted to analogue, and a digital compander circuit is used to improve this performance. The rest of the circuitry is analogue, and consists of a three section tone control, tremolo (performed by voltage controlled amplifiers) and ensemble, courtesy of three *analogue* delay lines.

Specifications

Specifications for Yamaha's "Grand Synthesiser" are as follows:

88 note keyboard, A to C

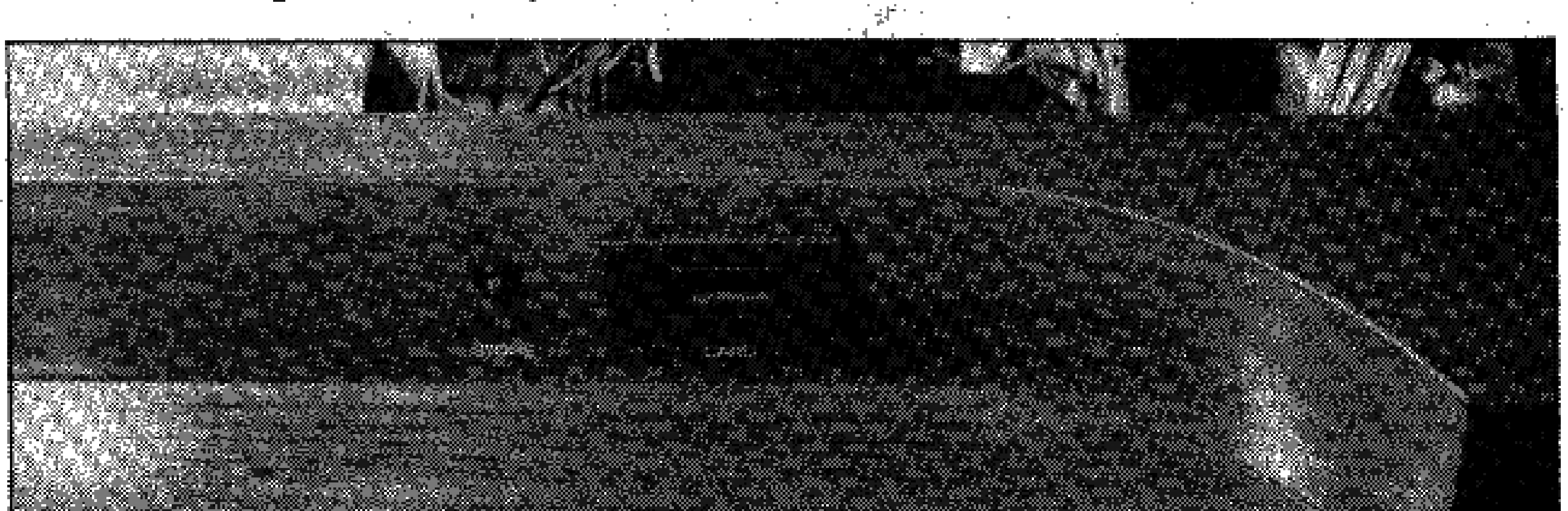
4 x FM digital tone generators (16 note polyphonic)

16 voices (stored in RAM); alternative voices may be added from a library of magnetic voicing strips.

Controls are:

1) Master pitch
2) Detune
3) Vibrato speed, depth and on/off
4) Tremolo speed, depth and on/off
5) Ensemble on/off
6) Touch response on/off
7) Equalisation: bass, middle and treble
8) Master volume
9) Store switch
10) Headphone jack output
11) Foot controller jack input
12) Line out switch
13) Pedal controls: damper, tremolo and vibrato
14) Unbalanced outputs: channel 1, channel 2 and mixed
15) Balanced outputs: channels 1 and 2

E&MM



Close up of the magnetic card reader.