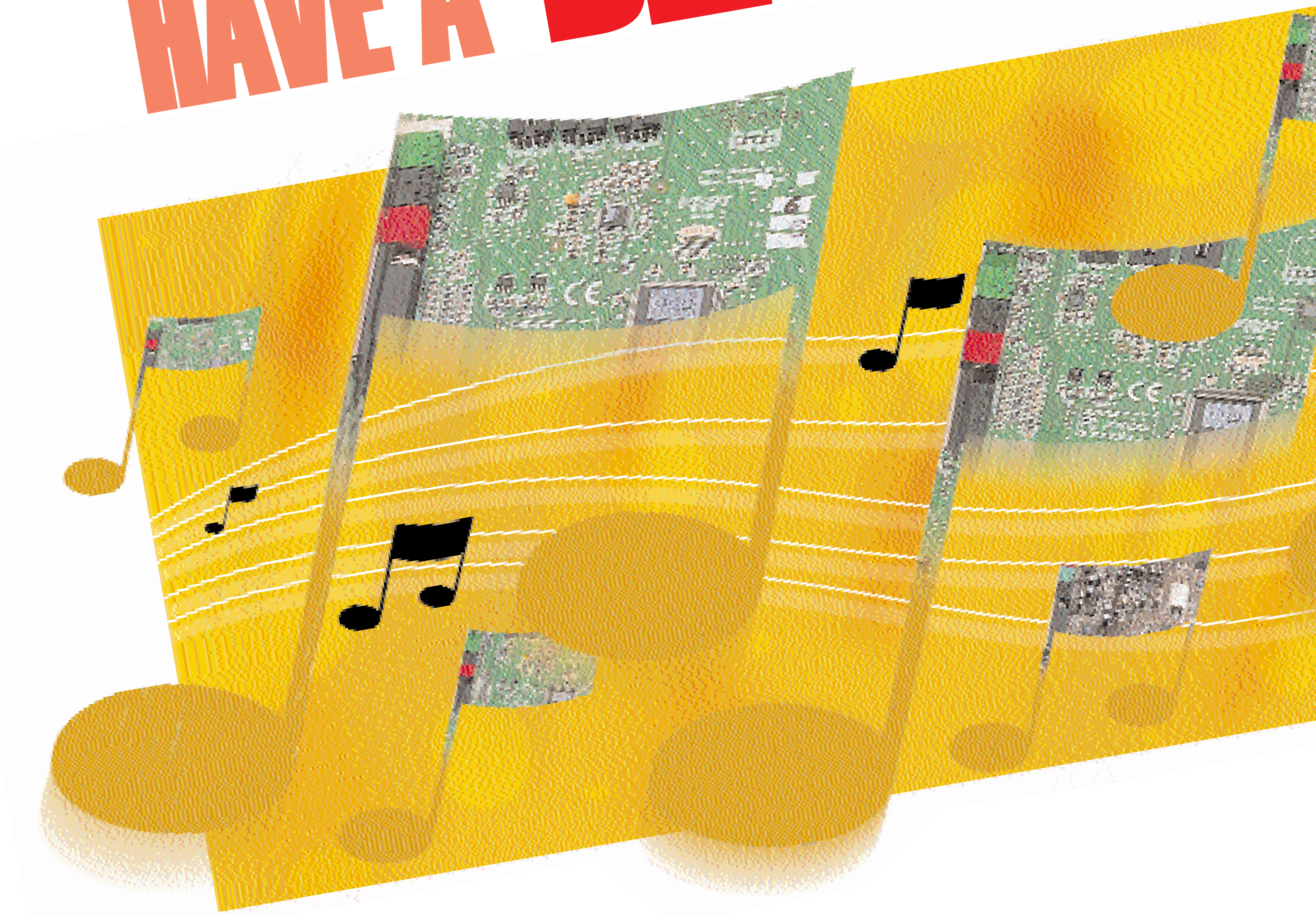
HAVE A BLAST!

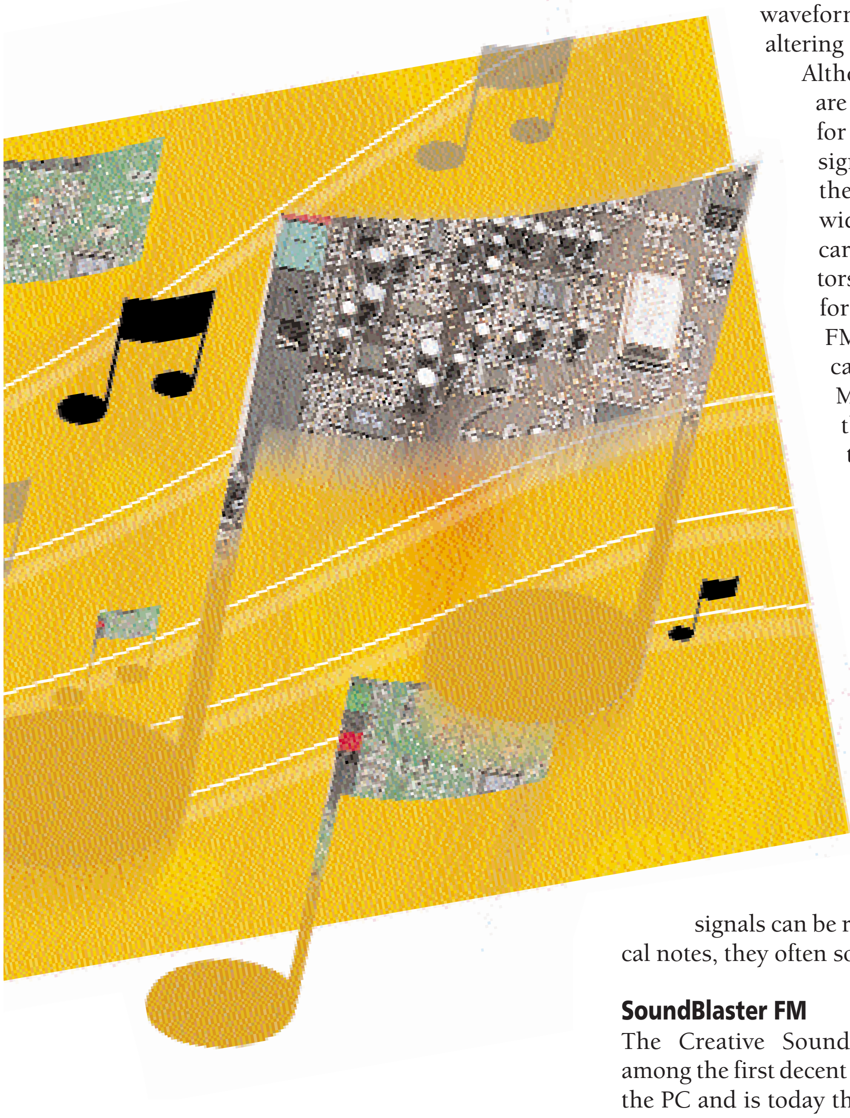


From playing a few short beeps to 16-bit stereo, soundcards in PCs have come a long way

In the beginning, all was quiet. Well, almost. IBM in 1980 decided that the PC did not need extensive audio capabilities, so they simply outfitted it with a primitive speaker that emitted short beeps when necessary. This had owners of multimedia-capable Apple and Amiga computers smirking at them so bad that IBM introduced the PCjr. But DOS hackers had by then discovered a method to get the PC speaker itself to sing, by speeding up the system clock to twice its normal speed.

This method was used in early Amiga-Tracker music module players such as ScreamTracker, and even in the most popular of all 3D games, *Wolfenstein 3D*. It however placed such a huge load on the processor that it was left incapable of doing anything else. Hence was not very popular.

Not much improved until Ad Lib introduced its FM soundcard, giving the PC a chance at proper audio. The Ad Lib soundcard is dead now, but the technology it introduced to the PC remains to this day.



Frequency Modulation

FM (Frequency Modulation) technology is used in FM radio and was discovered by John M Chowning at Stanford Artificial Intelligence Laboratories in 1973. The process of FM synthesis starts with one frequency or tone called the carrier frequency which is altered using a second one called a modulator.

When the modulator has a lesser frequency (of a few Hertz), the carrier rises and falls much like a siren, but when the two signals are close in frequency, the result is a complex wave. Trying to vary the strength of the carrier signal only varies the volume (amplitude), but varying the modulator changes the mix of frequencies in the resulting waveform signal, thereby altering its timbre.

Although two signals are all that are needed for creating an FM signal, a normal synthesiser will use a wide combination of carriers and modulators. The output waveform signal from an FM synthesiser is called an operator. Most popular synthesisers have four to six operators.

The popularity of FM synthesis comes from the fact that it is rather inexpensive to implement. It takes only a chip. The disadvantage is that FM cannot duplicate real-world sounds. Although FM signals can be recognised as musical notes, they often sound synthetic.

SoundBlaster FM

The Creative SoundBlaster card was among the first decent audio solutions for the PC and is today the standard that all other soundcards emulate. The SoundBlaster was built around the Ad Lib FM base, but introduced more features such as MIDI and Waveform (digital) audio. The original SB used the Yamaha YM3812 chip to produce FM. This chip has only one output channel and is therefore capable of only monophonic audio. Early stereo soundcards (including the SoundBlaster) used two of these chips, one per stereo channel. The YM3812 could mix up to eleven instruments, five of them fixed (non-programmable) for rhythmic instruments, and the other six for other instrument types.

The SoundBlaster 16 replaced the YM3812 chip with the Yamaha TMF262 or OPL3 chip. This chip not only provided up to 20 voices, but also used more sophisticated algorithms—mathematical equations—for audio synthesis. It provides full stereo output and is also backward compatible with the older chip.

Applications that use FM directly through the OPL3 chip are rare today. Instead, FM is now used as the audio back-end for MIDI.

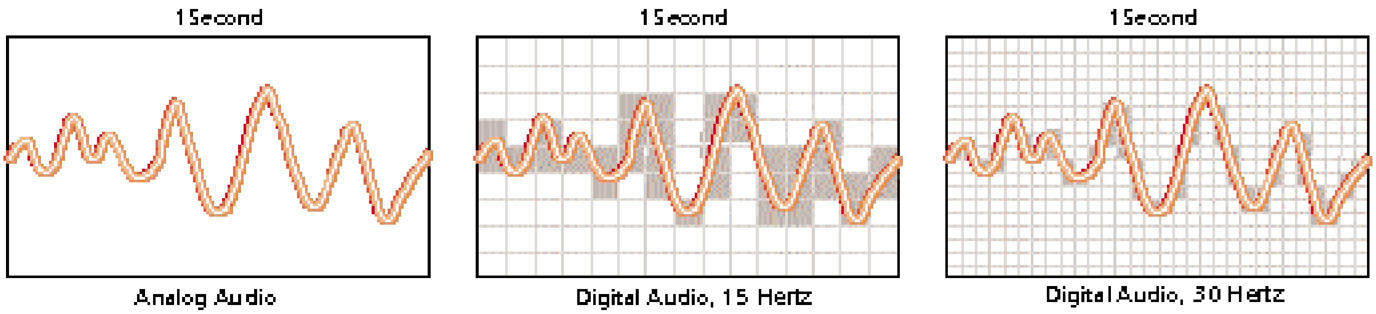
MIDI

Musical Instrument Digital Interface (MIDI) is a protocol for letting digital musical instruments interface with computers. MIDI originated as a standard protocol for transferring music data between synthesisers and computers but is now primarily used as a lightweight music file format for use in presentations and as background music on Web sites.

MIDI treats music as a sequence of musical notes from an instrument. The MIDI format allows mixing up to sixteen channels at once. Further, due to a peculiarity in the way MIDI stores data, it is possible to squeeze up to four instruments into one channel, allowing for a total of sixty-four instruments. This is functionally the equivalent of playing sixty-four instruments at once (imagine a live orchestra in concert). The MIDI instrument list recognises 128 different instruments, starting with grand piano (number 1), and a gunshot effect (number 128).

There has for long been a conflict on the assignment of instrument numbers. All the major manufacturers have their own tables that are incompatible with other manufacturers. While there has never been a clear consensus, Windows comes with a software solution to this problem, the MIDI Mapper. This hardware interface driver maps MIDI instrument numbers from your source MIDI data (including MIDI files), to the instrument numbers for your soundcard. This ensures that a MIDI file sounds the same, no matter what manufacturer's soundcard you use.

Almost all SoundBlaster-compatible soundcards (which means almost all soundcards around), use FM synthesis to play MIDI. However, FM cannot sound like the real instrument. New generation soundcards, and software-based solutions for older cards, therefore use what is



The effect of increasing sampling rate and number of bits on the purity of the output (digital) signal. The higher the sampling rate and the number of bits used, the closer is the approximation of the output digital signal with the input analog signal.

called Wavetable MIDI synthesis.

Wavetable MIDI replaces FM synthesis with a table of pre-recorded instrument sounds. All the instruments are recorded for only one note (say, note C on octave 1).

To play a different note for the same instrument, the recorded sound is played back at a different frequency. The two methods of implementing wavetable MIDI are either as a software driver, where all MIDI audio is played through the wavetable driver rather than the soundcard, and the hardware method where the soundcard itself supports wavetable.

One interesting combination of these two methods is seen in the SoundBlaster AWE64 card, which inherently supports wavetable but requires the wavetable data to be loaded by the software into system RAM.

The card itself does not store any wavetable data. This makes for a very easily upgradable system, but is also more demanding of system resources.

Digital Signal Processor

MIDI is a great way to play music with little processor-power but is very limited in what can be done since it does not allow pure waveform. What MIDI instrument would you use if you wanted to make your soundcard say “hello”?

Waveform audio is not played using MIDI, it is played using the Digital Signal Processor (DSP). The DSP is a programmable chip that has a wide variety of applications. Within your soundcard, it handles everything except MIDI and FM. Everything from setting the volume to changing the configuration to playing waveform audio, in fact.

Computing old timers will remember having to specify something like SET BLASTER=A220 I5 D1 T4 in their autoexec.bat file. This is the configuration line for the DSP. The above example indicates that the DSP is at I/O port 220 hexadecimal (hex is base 16, decimal is base 10), IRQ 5, DMA channel 1, and is a soundcard Type 4: a SoundBlaster Pro or compatible. Most soundcards offer compatibility with the SB Pro. Windows 95 automatically detects and configures your soundcard, making this setting unnecessary, but it is still used for compatibility with older DOS applications.

Waveform audio: How it works

Soundcards are digital devices. They cannot handle analog waveforms that nature generates. Soundcards therefore use two chips known as the Digital-Analog-Convertor (DAC) and the Analog-Digital-Convertor (ADC) to help convert between the human-friendly analog format, and the computer-friendly digital format.

Audio in nature is in the form of sound waves. A microphone performs the task of converting these analog sound waves into analog electrical signals, in the form of voltage that varies according to the signal.

For example, the analog signal may vary between 0 volt and 7 volt. A digital device can handle the range from 0 to 7, but cannot handle fractional values like 2.5. It has to be either 2 or 3.

The Analog-to-Digital Convertor chip takes care of this part by converting all incoming analog signals to the rounded-off digital format. The resultant digital audio with a range from 0 to 7 is called 8-bit audio (there are 8 different numbers in the 0-7 range). Modern soundcards (including all the SB 16 descendent cards) are 16-bit. CD quality audio plays at 16-bit quality, while telephone-quality is equivalent to 8-bit.

There is another aspect to be considered: being able to use a value between 0 and 7, how many such values can you capture in a second? The number of values in one second is called the sampling rate. Most soundcards can do up to 44 kilohertz (44,000 values per second). Some can even do 48 KHz. CD quality audio is sampled at 44 KHz.

The human ear can distinguish sound frequencies up to 20 KHz. Anything beyond that goes into the ultrasonic range, thereby making a constant frequency beyond 20 KHz completely inaudible. To smoothen the audio quality a CD is sampled at 44 KHz. This makes it free of any fuzzy sounds that appear at lower sampling rates.

Once in digital form, audio can be stored on your computer using applications such as Sound Recorder.

During playback, digital audio goes through a similar process.The Digital to Analog Convertor chip converts digital audio into analog electrical signals that your speakers in turn convert into pure audio.

Programming your soundcard

Want to do it yourself? Try these resources for detailed information on programming your soundcard:

ftp://x2ftp.oulu.fi

This FTP site contains a great amount of information on game programming under DOS, including stuff on soundcards.

www.s2.org/midas

MIDAS is a high-quality audio mixing library that works on DOS, Windows and Linux. The library is free for non-commercial use and comes with the complete source code.

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