Midi Time Code to SMPTE conversion (C++ / openframeworks)

I've recently needed to work with Midi Time Code (MTC) and could not find any code to parse the midi messages and construct an SMPTE timecode. Closest I got was finding **this documentation** (which is pretty good) on how the data is encoded in the bits of 8 bytes sent over 2 SMPTE frames, each byte sent at quarter frame intervals. From that I wrote the code below (I've only really tested the 25 fps). The code is from an openframeworks application but should work with any C/C++ code.

P.S. Some info on bits, bytes and nibbles **here.**

```
class ofxMidiEventArgs: public ofEventArgs{
public:
    int     port;
    int     channel;
    int     status;
    int     byteOne;
    int     byteTwo;
    double  timestamp;
};

#define kMTCFrames      0
#define kMTCSeconds     1
#define kMTCMinutes     2
#define kMTCHours       3

// callback for when a midi message is received
void newMidiMessage(ofxMidiEventArgs& eventArgs){

    if(eventArgs.status == 240) {                       // if this is a MTC message...
        // these static variables could be globals, or class properties etc.
        static int times[4]     = {0, 0, 0, 0};                 // this static buffer will hold o
        static char *szType     = "";                           // SMPTE type as string (24fps, 2
        static int numFrames    = 100;                          // number of frames per second (s

        int messageIndex        = eventArgs.byteOne >> 4;       // the high nibble: which quarter
        int value               = eventArgs.byteOne & 0x0F;     // the low nibble: value
        int timeIndex           = messageIndex>>1;              // which time component (frames,
        bool bNewFrame          = messageIndex % 4 == 0;


        // the time encoded in the MTC is 1 frame behind by the time we have received a new frame
        if(bNewFrame) {
            times[kMTCFrames]++;
            if(times[kMTCFrames] >= numFrames) {
                times[kMTCFrames] %= numFrames;
                times[kMTCSeconds]++;
                if(times[kMTCSeconds] >= 60) {
                    times[kMTCSeconds] %= 60;
                    times[kMTCMinutes]++;
                    if(times[kMTCMinutes] >= 60) {
                        times[kMTCMinutes] %= 60;
                        times[kMTCHours]++;
                    }
                }
            }
            printf("%i:%i:%i:%i | %s\n", times[3], times[2], times[1], times[0], szType);
        }


        if(messageIndex % 2 == 0) {                             // if this is lower nibble of tim
            times[timeIndex]    = value;
        } else {                                                // ... or higher nibble
            times[timeIndex]    |=  value<<4;
        }


        if(messageIndex == 7) {
            times[kMTCHours] &= 0x1F;                               // only use lower 5 bits for
            int smpteType = value >> 1;
            switch(smpteType) {
                case 0: numFrames = 24; szType = "24 fps"; break;
                case 1: numFrames = 25; szType = "25 fps"; break;
                case 2: numFrames = 30; szType = "30 fps (drop-frame)"; break;
                case 3: numFrames = 30; szType = "30 fps"; break;
                default: numFrames = 100; szType = " **** unknown SMPTE type ****";
```

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```
            }
        }
    }
}
```

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Synchronisation and SMPTE timecode (time code)

This document is a description (mainly technical, but not requiring extensive prior knowledge) of the SMPTE time-code synchronization system, especially the Bi-Phase Mark "LTC" timecode sync tone which can be recorded onto the audio track of a video tape or onto an audio tape. Other time code variants, such as VITC time-code are also mentioned. Some supporting information on the related subjects of colour television standards and field sequences is included. There is also a concise description of the binary coded decimal (BCD) representation as used in the SMPTE (or EBU) timecode (time code) data word.

My original reason for exploring this subject technically was to develop a low cost MIDI tape sync unit, which was a frequent customer request, and a logical addition to our range of electronic music gadgets. Our Time Code tape sync unit, which supported SMPTE (including Drop Frame format) and MIDI Time Code, was called the ***TS1*** and was for over a decade a successful and highly-respected product.

Synchronisation systems

When you require pieces of audio, video, or music technology equipment (eg. tape recorder and sequencer) to work together, you may need some means to make sure that they play in time with each other. This is called 'synchronisation' or 'synchronization', which gets shortened to 'sync' or even 'synch'.

A magnetic tape-recorder track is the region of the tape that is scanned by one recording head element. It can, for example, carry an audio signal, that is an electronic analogue of a stream of sound. In order to use an audio track to record sync information, the information must be encoded into an audio-compatible signal, called a sync tone. An audio tape track on which a sync tone has been recorded is called a 'stripe'. It is desirable that the synchronised devices can join in wherever you start up the sync tone, even if it is not at the beginning of the stripe; this is called 'chasing'. The SMPTE/EBU timecode standard defines the predominant internationally accepted standard for a sync tone and it allows devices to 'chase' or locate to a precise position.

In music technology applications, with equipment which is not timecode compatible, you may be able to use a tape sync unit by way of a tempo-relative stripe system. Examples of such systems are traditional *FSK* (Frequency Shift Key) and the *Philip Rees* proprietary *FSKplus* format.

The SMPTE/EBU timecode standard

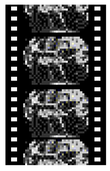
Until the advent of video recording, there was mechanical sound to film synchronisation. This mainly relied on sprockets (a row of neat holes) in the film and in special sprocketed recording tape. Relative timing adjustments could be made by slipping sprocket holes. The same sprocket holes were used to maintain synchronisation. Video tape hasn't got any sprocket holes, so when video arrived an electronic equivalent was needed to take the place of mechanical methods of synchronisation.

In 1967, The US Society of Motion Picture and Television Engineers introduced that which we call SMPTE ("simpty") time code. The audio sync tone version of SMPTE is called linear or longitudinal time code or LTC. By the way, there are also versions of timecode which can be inserted into a video signal or sent via a MIDI connection.

On the website **tvhandbook.com**, the interesting Broadcast History Timeline has this entry the year of 1969: "SMPTE timecode established to end the chaos of incompatible time codes for various editing machines".

The original uses of SMPTE timecode include accurate video editing and synchronising film sound-tracks. The timing data in SMPTE takes the form of an eight digit twenty-four hour clock. The count consists of 0 to 59 seconds, 0 to 59 minutes and 0 to 23 hours. The second is subdivided into a number of frames, which may be varied to match the various frame-rates used around the world. The frame-rate is the number of times a second that the picture is updated so as to give the illusion of continuous movement.

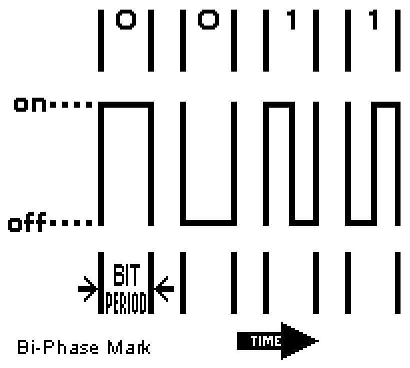
Frame-rate formats



There are four standard frame-rate formats: The SMPTE frame-rate of thirty frames per second (fps) is often used for audio in America. It is used with the Sony 1630 format for CD mastering. It has its origins in the obsolete American mono television standard. The American colour television standard has a slightly different frame-rate of about 29.97 fps. This is accommodated by the SMPTE format known as thirty Drop Frame and is required for video work in America, Japan and generally the 60 Hz (mains frequency), NTSC (television standard) world. The number of frames in each second is not an integer, so an approximation is used; it is based on 30 fps, but two frames counts are dropped (skipped) at the start of every minute, except for every tenth minute. The EBU (European Broadcasting Union) standard of 25 fps is used throughout Europe, Australia and wherever the mains frequency is 50 Hz and the colour TV system is PAL or SECAM. The remaining rate of 24 fps is required for film work, it is rarely used for audio.

In audio post-production, SMPTE has been adopted for machine synchronisation and as a reference of tape position. Essentially, the choice of frame rate for audio work is usually arbitrary.

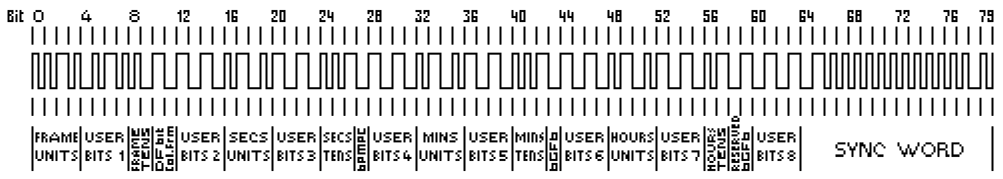
The modulation scheme of SMPTE LTC



A bit is a binary digit (which can take only the value 0 or 1). The value of a bit is often represented by the 'off' and 'on' states of an electronic switch. When the sequence of binary digits must pass through a channel designed for analogue audio signals the 'on' and 'off' states ('phase') cannot be reliably distinguished. However, audio frequencies are reproduced faithfully; to exploit this property, SMPTE Longitudinal TimeCode encodes its data into the rate (frequency) of electronic state transitions. The binary value 0 is represented by a single transition at the start (or 'boundary') of the bit-period (or bit 'cell'). The binary value 1 is represented by a two transitions - one at the start and the second in the middle of the bit period. This scheme is called 'Bi-Phase Mark' and closely resembles frequency modulation ('FM'); that is, a stream of binary ones is represented by a burst of audio at double the frequency of a burst of audio which is used to represent a stream of a binary zeros. This straightforward scheme of modulation is amazingly robust and compatible with a range of real world audio channels, including tape recorder tracks. To ensure good performance in audio channels, the rise-time of the waveform is also specified.

The data organisation of the SMPTE LTC frame

The datastream for each video frame of Longitudinal TimeCode consists of eighty bit-periods (bit cells). At the original American frame-rate of 30 fps, the bit rate would work out to 30 x 80, that is 2400 bits per second. The frequency for a stream of zeros would be 1.2 kHz and for a stream of ones it would be 2.4 kHz. The corresponding bit rate for the European 25 fps frame-rate is 2000 bits per second. in this case, the frequency for a stream of zeros would be 1.0 kHz and for a stream of ones it would be 2.0 kHz. All these frequencies are safely within the audio range, so the SMPTE LTC sync tone waveform can be recorded easily on any half decent audio track.



In each frame, twenty six of the eighty bits carry the SMPTE time or 'address', in binary coded decimal. In the diagram above, these Bits are shown as `FRAME UNITS, FRAME TENS, SECS`

`UNITS`, `SECS TENS`, `MINS UNITS`, `MINS TENS`, `HOURS UNITS` and `HOURS TENS`. The BCD digits are loaded 'least significant bit first'.

Thirty two bits are assigned as eight groups of four `USER BITS`, also sometimes called the "Binary Groups". This capacity is generally used to carry extra info such as reel number and date. The User Bits may be allocated howsoever one wishes as long as both Binary Group Flag Bits are cleared. The User Bits do not usually vary in the course of a timecode stream.

The last sixteen Bits make up the `SYNC WORD`. A timecode reader uses these Bits to find the frame boundary, the tape direction, and the bit-rate of the sync tone. The values of these Bits are fixed as `0011 1111 1111 1101`

The Bi-Phase Mark Phase Correction Bit `bPMPC` is Bit 27. This bit may be set or cleared in order that every 80-bit word contains an even number of zeroes. This means that the phase (or logical sense) of the pulse train in every Sync Word will be the same. As the LTC standard evolved, during the 1980's, timecode was being heavily used in 1" video editing. Many of the early processor systems used to synchronize the playback and record machines to an editor required that the timecode Sync Word to be properly timed to the vertical interval (per the spec). However, because of tracking (or other interchange) problems with some tapes, it was necessary to manually track those tapes for optimum playback. At the same time, not all VTRs were equipped with timecode boards, and not all of those that did performed a regeneration of the off-tape timecode (which provides a properly phased and shaped signal to the outside world). As a consequence, as the tracking was varied, the timing relationship of the Sync Word varied as well. Much to the dismay of many editors and tape operators (spoolers), it was learned too late that the timecode readers in the editor could be off by a frame if the tracking was moved too far away from spec. The *Ampex VPR-2/2B*, the work-horse of the industry at the time, had a timecode waveform display. Using it to verify the position of sync word was complicated by the constant toggling of the polarity of the waveform due to the varying number of zero's per frame. The Bi-Phase Mark Bit was added to make finding the Sync Word visibly easier. By the time the standard was fully adopted, most newer machines had already incorporated the regeneration feature which made the Bi-Phase Mark Bit obsolete. Prior to the standard being adopted, *Sony* on the *BVH-2000* had implemented a phase correction bit in one of the User Bits - this machine also had a timecode waveform display. ***credit***

Bit 58 is not required for the BCD count for HOURS TENS (which has a maximum value of two) and has not been given any other special purpose so remains unassigned. This Bit has been `RESERVED` for future assignment.

Bits 43 and 59 are assigned as the Binary Group Flag Bits `bGFb`. These Bits are used to indicate when a standard character set is used to format the User Bits data. The Binary Group Flag Bits should be used only as shown in the truth table below. The Unassigned entries in the table should not be used, as they may be allocated specific meanings in the future.

	Bit 43	Bit 59
No User Bits format specified	0	0
Eight-bit character set	1	0
Unassigned (Reserved)	0	1
Unassigned (Reserved)	1	1

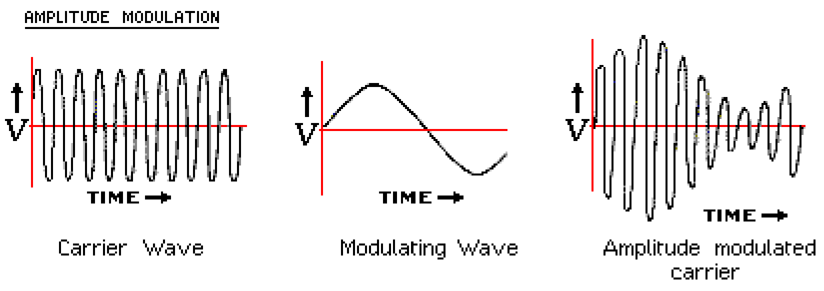
If you were designing a timecode reader to be able to read stripes from other writers, you would be best advised to make the reader ignore (mask out) Bits 27, 43, 58 and 59. If you were designing a writer to produce portable SMPTE timecode (for good compatibility) you should allocate zero values to these Bits; this is because some readers may interpret these Bits as part of the SMPTE frame 'address'; the result would appear to be invalid or illegal BCD values.

The Drop Frame flag `DF bit` is Bit 10; if the Drop Frame format is in use, this bit is set. The Colour Frame Flag `col.frm` is Bit 11; if the timecode intentionally synchronized to a colour TV field sequence, this bit is set.

NTSC and *Drop Frame* format

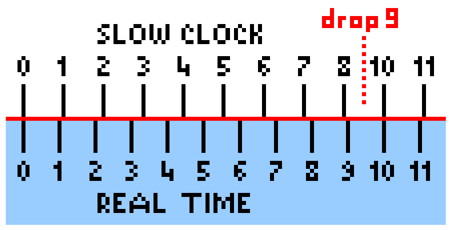
Electronic television was invented in the US by Philo T. Farnsworth, and his system had 300 lines per frame. Because of patent litigation in the US, this technology was first exploited in Britain. The original (now obsolete) British electronic television system used 25 fps and 405 lines. in 1941, the American monochrome television standard was specified to be somewhat superior, it had a frame rate of 30 fps with 525 lines. The audio signal was combined with the video signal by modulating a subcarrier wave at a frequency of 4.5 MHz.

In 1953, the American NTSC (National Television System Committee) specified a colour television standard which would be compatible with the existing black-and-white receivers. The two extra colour (chrominance) video components are combined with the monochrome (luminance) video signal by modulating additional subcarrier waves. They found that, with only minor problems, they could get away with selecting a frequency for these subcarriers which lay within the bandwidth of the main luminance component. This meant that a new colour TV channel would occupy the same bandwidth as an existing monochrome channel. The frequency for the the two chrominance components was set to be the same (227.5 times line frequency) but 90 degrees (quarter of a cycle) out of phase.



It was found however that there was a troublesome interaction between the line frequency and both subcarrier frequencies. They could probably have put this right by slightly altering the audio subcarrier frequency (4.5 MHz). Instead, they slightly altered all the other frequencies, increasing the frame period by 0.1%. This results in: a frame rate of (30 × 1000 / 1001) fps, that is about 29.97 fps; a field rate of twice the frame rate, that is about 59.94 Hz; a line rate of (525 × 30 × 1000 / 1001) Hz, that is about 15.73 kHz; the colour subcarrier ends up at (227.5 × 525 × 30 × 1000 / 1001) MHz, that is about 3.579545 MHz. [Please note: × is the symbol for the multiply operator.]

The dropped frames compensate for the fact that 29.97 is not an integral (whole) number of frames per second. This is achieved by regularly bumping forward the value of the frame count to make the slow-running clock catch up with an actual clock. This resembles, in an opposite way, how February 29th compensates for the fact that an astronomical year is not an integral number of days long. It is not so much **frames** which are dropped as frame **numbers** which are skipped. *Drop Frame* counting misses out frame numbers 0 and 1 in the first second of every minute. For example, timecode address 11:41:59:29 is immediately followed by 11:42:00:02. However, skipping two frame numbers per minute slightly over-compensates, so no frame numbers are dropped when the minute number is exactly divisible by ten. For example, timecode address 11:49:59:29 is immediately followed by 11:50:00:00. By this technique, a stream of timecode running at about 29.97 fps will repeatedly jump back into sync with an actual clock.

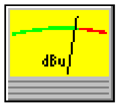


Here's a general idea of why frames are dropped

If you are working with European video, or are working on your own with audio, you can avoid the pitfalls of 30 fps SMPTE formats by sticking with the EBU 25 fps standard. If you are working with American video, you will actually be working with approximately 29.97 fps. If you use this American TV frame-rate with so-called 30 non-drop timecode the clock will actually run slow by about 1.8 seconds per hour. If you use the American TV frame-rate with the special 30 *Drop Frame* format the clock will stay correct to within a theoretical 75ms in 24 hours. If you are working with just audio, you can use 30 non-drop at a true frame-rate of 30 fps; many people do, especially in America, but this format is not truly compatible with any non-obsolete video format. Greater timing precision is sometimes cited as a reason for choosing 30 non-drop for audio work; in practice, however, the benefit is too small to be significant.

When an American TV frame-rate video production must be edited to actual clock time, Drop Frame timecode must be used. If staying in time with an actual clock not important, especially on short video productions like commercials, the bastard 29.97 fps non-Drop Frame timecode is often used. Some editors prefer this as accurate calculations can readily be made without the aid of a computer program or special hardware calculator.

SMPTE stripe signal levels

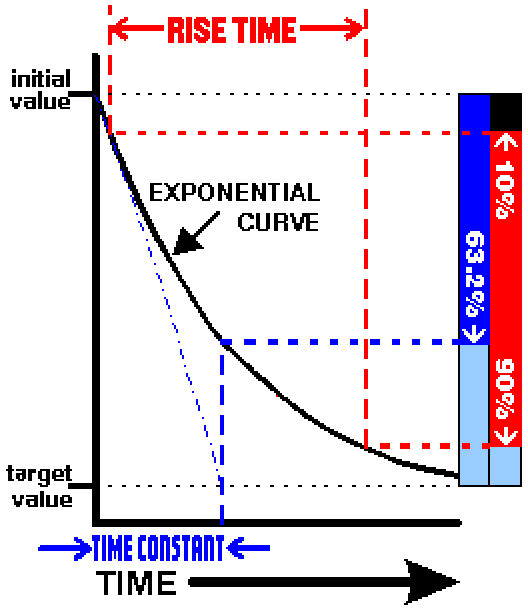


You should make sure that a SMPTE Longitudinal TimeCode stripe is recorded and played back at suitable levels. It is definitely not the case that higher levels are better. SMPTE timecode is a pulse or rectangular wave signal, where signal levels do matter. The most common error is recording at too high a level which causes waveform distortions leading to errors. Recording at too high a level will also exacerbate audio crosstalk, breakthrough of the sync tone onto audio signals. Recording at too low a level may make the stripe replay susceptible to crosstalk, leading to corruption of the signal seen by the timecode reader. Excessive volume in the stripe replay path may swamp the input circuits of a timecode reader, impairing its ability to interpret the sync signal.

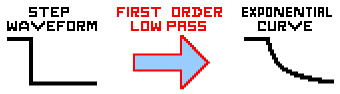
A SMPTE Longitudinal TimeCode stripe is best recorded at just a few dB above the minimum level which allows correct reading. the best way to arrive at appropriate working level settings is usually trial-and-error. Most audio tape tracks and one-inch Video Tape Recorders will work well with the stripe recorded as low as 10 dB below reference level. Audio tracks with noise reduction and Video Cassette Recorders may work better with stripes recorded as high as just 3 dB below reference level.

SMPTE LTC sync tone waveform

Here is a one second snatch of 25 fps LTC sync tone, in case you would like to hear how it sounds. You may decide to download the 22050Hz 8-bit mono Windows *.wav* sound file, Sun/Next *.au* sound file or Apple *.aiff* sound file: the files are about 42K bytes each.



The audio-compatible SMPTE Longitudinal TimeCode signal uses the Bi-Phase Mark modulation scheme. The data for each frame period is contained in an eighty bit word format. The modulation scheme, frame-rate and word length determine the range of pulse frequencies required by the LTC signal at normal speed; 960 Hz (zeroes at 24 fps) to 2400 Hz (ones at 30 fps).



The **rise time** is the time it takes the step response to traverse between ten percent and ninety percent of the full step amplitude. The **time constant** is the time it would take for the variable to traverse the full step amplitude, if its rate of change was the same as at the very start of the step response. The time constant approximately equals the rise time divided by 2.2. Where t_c is the time constant in ms, the formula for -3 dB cut-off frequency f_c in kHz is:

$$f_c = \frac{1}{2\pi t_c}$$

The standard specifies that the LTC signal should have a rise time of 25 µs ± 5 µs. This means that the pulse waveform must be low pass filtered. If the filter is of the first order (6 dB/octave), the 25 µs rise time corresponds to a time constant of about 11.5 µs. This also corresponds to a -3 dB cut-off frequency (channel bandwidth) of about 14 kHz. The resultant waveform does not have excessive high frequency energy, and is easily accommodated by a standard audio channel. [Please note: µ is the symbol for 'micro', ± is the symbol for '+/-'.]

In practice, the most common form of degradation to a SMPTE signal is loss of high frequencies: you may be able to recover such a waveform by passing through a band boost of a few dB at about 2 KHz with a Q of about 1, or a low order shelf filter which boosts 2 kHz with respect to 1 kHz. Other forms of degradation are crosstalk and noise: you may be able to reduce the effects of these by passing the signal through a high pass filter with a low frequency roll-off below about 800 Hz. The other main form of degradation is jitter; the only way that a user might help with this is by the careful adjustment of levels. The sync tone input of my ***TS1*** design had good jitter tolerance and several automatic features for waveform restitution.

Generally, the SMPTE sync tone can be handled in the same ways as an ordinary audio signal. The SMPTE LTC signal can use balanced or single-ended interconnects as required. In professional installations a balanced signal using screened pair audio cable is usually for timecode distribution. The SMPTE sync tone is an obtrusive noise and often a fairly high level signal; so it is necessary to route cables so as to avoid crosstalk. The SMPTE signal should not be subjected to audio processing (such as expansion, compression, equalization, noise reduction and automatic gain controls), which can render the waveform unreadable. The *Philip Rees* ***TS1*** has such a clever sync tone input that it can even decipher a SMPTE signal that has been recorded and played back through **dbx** processing.

SMPTE VITC and special timecode variants

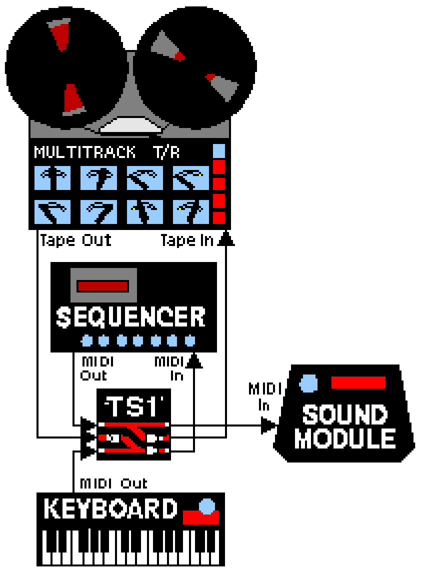
LTC is 'longitudinal' because it is usually recorded along the length of a tape. It is an audio-like signal, which can only be read while the tape is moving relative to the tape head. In video tape systems, LTC is used at normal or higher speeds.

In video tape systems, an alternative type of timecode may be inserted into the vertical sync period of the video frame signal, so it can be read even while the video tape is paused - this is called Vertical Interval Time Code or VITC ("vitsy"). Both SMPTE timecode types encode the time into a frame with a similar data format. SMPTE LTC can be overdubbed onto an already recorded video tape, whereas SMPTE VITC usually cannot (it needs to be incorporated into into the video signal during the initial recording or while copying between tapes). VITC is part of the video signal and is used at normal or lower speeds.

The VITC word is ninety bits long. The VITC word inserts nine pairs of synchronizing bits, while an eight bit error-checking "CRC" code replaces LTC's sixteen bit synchronizing word.

Many modern video recording and audio recording formats make special provision to record the timecode data. This may allow time code to be overdubbed onto an already recorded video tape without copying; yet it also avoids occupying an audio track. Even when the recorded signal is different, as with SMPTE VITC, the output datastream is usually presented in SMPTE LTC format.

MIDI Time Code



A supplement to the MIDI specification defines a standard whereby the timing information (in hours, minutes, seconds, and, frames) found in SMPTE timecode can carried by a MIDI connection. In this way, a suitably equipped MIDI sequencer may be synchronised to an absolute timing reference. This standard is called MIDI Time Code or MTC.

MTC acts as a bridge between SMPTE and the accepted standard system for controlling musical equipment, MIDI. MTC makes it possible to employ a single timing reference throughout a video, audio and MIDI system. Once MTC is implemented, a user may avoid the need to convert manually between absolute SMPTE time and tempo-relative bar and beat numbers.

Binary Coded Decimal

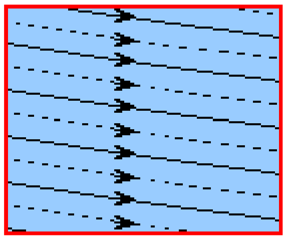
As with all digital electronic systems, the information in a SMPTE timecode signal is held in binary (base two) format. Within the SMPTE LTC word the numbers of the SMPTE timecode frame address are in binary coded decimal representation.

Binary Coded Decimal (often shortened to BCD) is a method of representing base ten (decimal) numbers using binary (base two) codes. Each decimal digit occupies four binary digits (bits). The four bits provide 2^4 (that is sixteen) possible bit patterns. Ten of these patterns are used by the BCD system to represent the decimal values zero to nine inclusive. The remaining six patterns do not correspond to legal BCD values. [Please note: 2^4 above represents two raised to the power of four.

BINARY CODE	DECIMAL	BINARY CODE	DECIMAL
0000	0	1000	8
0001	1	1001	9
0010	2	1010	*not legal BCD*
0011	3	1011	*not legal BCD*
0100	4	1100	*not legal BCD*
0101	5	1101	*not legal BCD*
0110	6	1110	*not legal BCD*
0111	7	1111	*not legal BCD*

Because some possible bit patterns are not used, a value represented as a BCD number is less compact (that is it occupies more bit cells) than the same value represented in plain binary. BCD numbers are also generally more difficult for computers to manipulate than plain binary numbers. Consequently, BCD is less commonly chosen as a number format nowadays. However, it was often preferred back when SMPTE timecode was specified, because in those days it was comparatively complex to convert data between base ten representation and base two representation.

Colour TV field sequences



To reduce the perceived screen flicker on a television, a technique called 'interlacing' is employed. Interlacing divides each video frame into two fields; the first field consists of the odd scan lines of the image, and the second field of each frame consists even scan lines.

'Component video' is employed in certain professional video formats; luminance (brightness), chrominance (colour) and colour burst, blanking and sync pulse components are kept separate to optimise picture quality. Each successive frame of black-and-white or component colour video can be edited at any frame boundary, in a manner similar to cutting and splicing film.

A 'composite video' signal is one in which the luminance, chrominance and other components have been combined into a single signal. In the various world colour TV standards (NTSC, PAL or SECAM) the schemes of component combination are somewhat different.

In the established systems, the colour subcarrier frequency (used to superimpose the chrominance component on the luminance component) is not an integer multiple of the frame-rate. Significant phase discontinuities in the colour subcarrier cannot be tolerated. Consequently, the overall pattern of the composite signal repeats over a period longer than a single frame. The accepted repetition period is four frames in PAL, and the pattern is called the eight field sequence. The repetition period is two frames in NTSC or SECAM, and the pattern is called the four field sequence. In order to avoid picture degradation, an edit should not disrupt these sequences. To maintain the sequences across edits, the timecode (used to synchronise the video sources) must be able to keep track of the sequence position of each frame.

With a four field (two frame) sequence at 30 fps Drop Frame, the field sequence synchronisation just consists of odd and even frames. In timecode, the pattern is not broken by the *Dropped Frames* (as these come in pairs) nor by the resetting of the frame count when the seconds count is incremented (there are always an even number of frames in each second). The sequence position of each frame is simply indicated by the state of the least significant bit of the frame units count, that is Bit 0 of each SMPTE word.

With a four field (two frame) sequence at 25 fps, field sequence position can be indicated via timecode from the 'exclusive or' (XOR) of the least significant bits of the FRAMES UNITS and SECONDS UNITS.

With an eight field (four frame) sequence at 25 fps (as in regular PAL), field sequence position can be indicated in timecode by the first taking the SUM of the two least significant bits of the FRAMES UNITS and two least significant bits of the SECS UNITS, ignoring the carry-out. The more significant bit of the two-bit result of the SUM should be XOR'ed in turn with the next-to-least significant bits of both the FRAMES TENS and SECS TENS; this corrects for the effect of resetting of the UNITS digits when the TENS digits are incremented. Fortunately, at 25 fps, there are no *Dropped Frames* to complicate matters, and the pattern is not broken by the resetting of the seconds count when the minute count is incremented (sixty is divisible by four). The two least significant bits of the frame units count are Bits 0 and 1 of each SMPTE word. The two least significant bits of the seconds units count are Bits 16 and 17 of each SMPTE word. The next-to-least significant bit of FRAMES TENS is Bit 9, and the next-to-least significant bit of SECS TENS is Bit 25.

The 'exclusive or' (XOR) is a Boolean operator, its basic form has two binary inputs and one binary output. When the inputs are equal the output is zero. When the inputs are non-equal the output is one. In the computer language 'C' this operator is shown by the character '^'. The XOR truth table is shown below:

```
0 ^ 0 = 0
0 ^ 1 = 1
1 ^ 0 = 1
1 ^ 1 = 0
```

Binary addition is very similar to ordinary decimal addition, except that all the digits can only take the value 0 or 1. The '+' sign here indicates addition.

```
0 + 0 = 00
0 + 1 = 01
1 + 0 = 01
1 + 1 = 10
```

Along with a couple of XOR operations, a SUM function is needed to derive the eight field sequence position as described above. This two-bit (binary digit) addition has the carry-out discarded, so the results are as follows:

```
00 + 00 = 00        00 + 01 = 01        00 + 10 = 10        00 + 11 = 11
01 + 00 = 01        01 + 01 = 10        01 + 10 = 11        01 + 11 = 00
10 + 00 = 10        10 + 01 = 11        10 + 10 = 00        10 + 11 = 01
11 + 00 = 11        11 + 01 = 00        11 + 10 = 01        11 + 11 = 10
```

The acronym PAL stands for 'Phase Alternation by Line', or words to that effect. This is the colour television standard widely used in Europe (including the UK), invariably with 625 lines and 25 fps. PAL is a derivative of the American NTSC system, but the carrier phase of the one of the colour components is inverted on alternate lines; this compensates for hue shifts which can otherwise be caused by phase distortion. The acronym SECAM stands for 'Système Electronique Couleur Avec Memoire'. This is the colour television standard used in France, many former communist bloc countries, and a sprinkling of other territories that De Gaulle was able to lean on. This standard is nowadays also associated with 625 lines and 25 fps. In SECAM the two colour components are each carried alternately on alternate lines. Video production for SECAM is often performed in PAL, the signal being converted to SECAM for transmission.

Credit: for his contribution of the expert information about the Bi-Phase Mark Bit, I am indebted to *Jim DeLuca, Senior Video Engineer, Crest National Digital Media Complex, Hollywood CA.*

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