



## ITU DTMF Transmitter/Receiver

**SIGNALS+SOFTWARE**



### Processor

Motorola DSP56300 range.

### Background

The algorithm implemented is the ITU-T recommendation DTMF signalling system. DTMF is the most common signalling method used within the telephone network, and has now largely replaced loop disconnect (“pulse”) dialling. DTMF works by employing pairs of tones to encode the digits 0-9, #, \* and A to D. DTMF receivers need to correctly detect the presence of these tones, while eliminating background noise and allowing for distortions introduced by the network. The Mitel test tape exists to verify the performance of DTMF receivers.

DTMF applications include reception for public or private telephone exchanges; telephony and line test equipment; remote control of computer and telephone equipment

### Features and Performance

- The transmitter software produces the required tones with a frequency tolerance of 0.1%
- The receiver software has been extensively tested with all Mitel tests.

<b>Dynamic range</b>	In excess of 35dB
<b>Guard time</b>	The receiver detects digits of 34ms duration and greater
<b>Signal-to-noise Ratio</b>	The receiver identifies all digits correctly at an SNR of 14dB
<b>Mitel talk-off test (CM72910)</b>	1 false digit detected during this test; the pass level is 30
<b>Frequency offset</b>	The receiver passes all frequency offset tests
<b>Twist</b>	The receiver passes all 8 twist tests, with average attenuation at cut-off of 8dB for normal twist, and 4dB for reverse twist

**Table 1 : DTMF Performance**

DTMF	Program Memory (Kwords)	Tables (Kwords)	Stack (Kwords)	Data Memory (Kwords)	Processing Load (MHz)
Transmitter	0.26	0	0.07	n *0.01	1.7+n *0.16
Receiver	0.4	0.11	0.05	n * 0.03	2.1+n *0.51

**Table 2 : DSP Requirements for DTMF**

**Note:** Processing loads quote worst-case scenarios and n represents the number of channels.  
Program memory table values are initialisation values. 1 word equals 24 bits. Kwords equals 1024 words.

### Technical Notes

The receiver code will be highly configurable for a particular application. Output decisions are to be held in buffers of user-selectable size, allowing infrequent polling for results by the host.

The code will consist of an initialisation routine, which is normally called upon DSP reset, and by a single subroutine, which will be called once for each channel. The audio input/output format is 8kHz linear samples. Additional code may then be required to interface to the host platform, for example to extract data from a serial link.

### Interface Details

For convenience the individual software functions will be supplied as a single library module. The library will contain all the object code that is required to link in to a user's top-level application code.

### Availability

Fully optimised code is available now for a one-off payment and/or royalties depending on the commercial application. Also available for DSP56300<sup>TM</sup> are a full range of vocoders, echo cancellers, modems, IP telephony and other communication algorithms.

### SIGNALS+SOFTWARE

**SIGNALS+SOFTWARE** was founded in 1992 as a developer of high quality Digital Signal Processing application software for the communications industry. Supplying to a whole range of customers, including large blue chip corporations, **SIGNALS+SOFTWARE** has quickly established itself as the world leader in DSP software design and production.

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