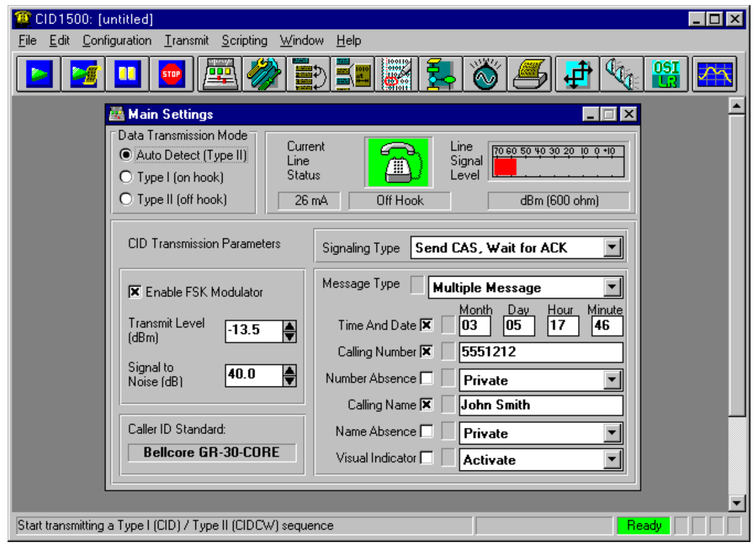
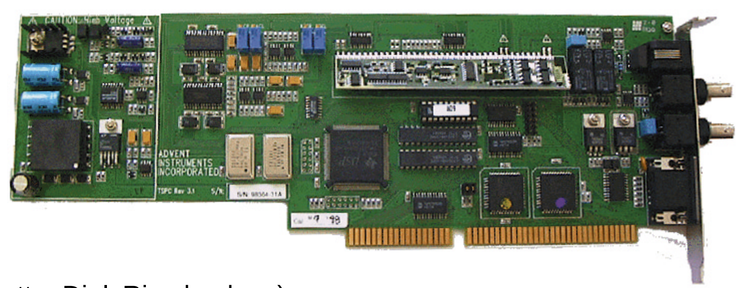
AI-100 SERIES

CALLER ID SIGNAL SIMULATORS

Key Features

- **Type I / II** Caller ID Signal Generation for **Bellcore**, **ETSI** (European Countries), Australia, China, Japan, ...
- Single and Multiple Data Message Formats
- Complete control over data Format and Packet order
- Programmable Preamble, Mark and Markout times
- Byte-wise and Bit-wise data editing
- Programmable FSK Frequency, Level, and Twist
- Programmable Signal to Noise Ratio
- Programmable Loop Voltage and Current
- Programmable 600/900/Complex Impedance for Bellcore or ETSI Standards
- Programmable Ringing level, Frequency and Cadence
- Programmable FSK carrier dropout
- Programmable OSI and Line Reversal Impairments
- Restorable Configuration Files
- Real Time Signal Capture for CAS and ACK Tone Analysis in both time and frequency domain
- Line Impairments including FSK Echo Simulation, White noise, and Interfering Tones
- FSK Loopback Testing
- BNC connectors for monitoring and loop through





- Programmable Network Tone Generation (Dial, Busy, Stutter Dial, Ring back, ...)
- Auxiliary Digital Inputs for Closed Loop Testing
- Auxiliary Digital Outputs for External Control and Triggering
- Programmable Scripting Language for automatic testing. Key features includes variable declaration, subroutine calls, user input, looping, data log for report generation or database management, ...
- Windows 3.1,95, and 98 Graphical User Interfaces

Applications

- Hardware design engineers who need to test the operational extremes of the circuits they design.
- Software design engineers who test the protocol algorithms and the error handling routines.
- Manufacturing Engineers who require simple automated testing sequences to speed production throughput.
- QA/QC engineers who conduct certification and evaluation of Caller ID capable devices.
- Marketing executives who need to demonstrate and evaluate the operation of Caller ID devices.

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Product Architecture

AI-100 Series Caller ID Signal Simulators are PC based test equipment systems. Each system consists of both hardware and software modules which include:

1. A full length PC ISA bus plug-in card called a Telephone Signal Processing Card (TSPC).
2. Windows programs controlling the TSPC for the signal generation of various Caller ID Standards. Three control programs are currently available:
   a. CID1500 supports ALL FSK based Called ID Standards except Japanese NTT .
   b. CID750D supports ALL DTMF / Multi-Frequencies (MF) Caller ID Standards.
   c. CID950N supports the Japanese NTT Caller ID Standard.

Product Models

AI-150: Standard Version - TSPC-SG, CID1500, CID750D, CID950N

The AI-150 is the most powerful simulator package we offer, and has the capability to test both the FSK signaling and Dual Tone Multiple Frequency, DTMF, based signaling systems. It is specially designed for product development, production verification, and product demonstrations.

The AI-150 also offers the ability to be controlled by other software applications. This provides the user with the ultimate in programmable convenience. The user applications can be written in either DOS, with a Turbo-C library, or Windows utilizing a 16/32 bit DLL control module. These low-level drivers can be downloaded from our website at no charge.

AI-120: Production Version - TSPC-PG, CID1500, CID750D, CID950N

The AI-120 is specifically designed for production line testing. This product is based on the same hardware platform and so has the same flexibility as the AI-150. The AI-120 uses the same Window programs as the AI-150; however, program control is limited to the scripting language only. The program windows allow for a quick verification of the simulator settings while limiting the control of the settings with the scripting language. Therefore, the AI-120 is ideally suited for the production environment which generally requires simple automated testing sequences.

Product Options

AI-E001: CAS/ACK Tone Analyzer Option for AI-150 Only

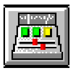
The FFT analysis option can be added to the AI-150 to capture the real time CID Type 2 transmission and to perform advanced signal analysis on the captured signal in both time and frequency domains.

AI-K001: One Year Extended Warranty Contract for AI-150 or AI-120 (Parts & Labor)

AI-K003: Three Year Extended Warranty Contract for AI-150 or AI-120 (Parts & Labor)

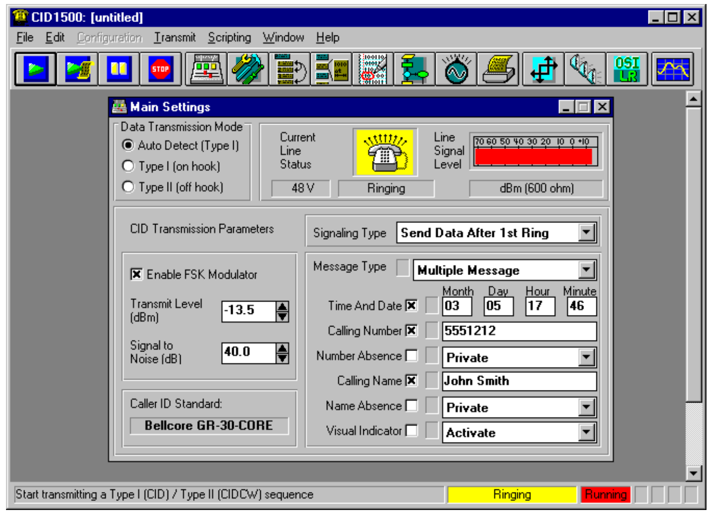
CID 1500 Software

The CID 1500 is one of the software components to the AI 100 series simulators. It implements the FSK based Caller ID standards, in an easy to use windows software architecture. This allows for precise and sophisticated control over the simulator, yet maintaining an easy to use graphical interface. Major functions are grouped into sub-windows, but can be quickly accessed via the tool bar. A hint line is included in the bottom status bar to indicate the operation of the selected control. The on-line Help with Hyper-text moves you to the help you need when you are having difficulty. Useful menu functions include saving and loading of configuration files and script program files. The ability to print a complete listing of all the parameter values, the script file, and the log file, which contains the analysis data taken during a CIDCW (Type II) Caller ID transmission.



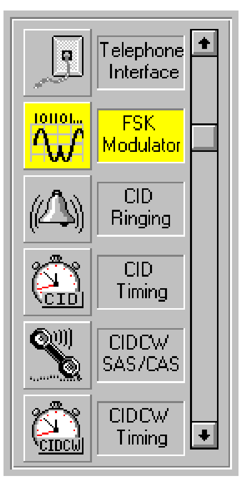
Main Settings

The main settings window includes all the primary controls for quick operation. The status section indicates the telephone state as well as what operation the simulator is currently performing. Also, a wide band level meter shows the signal level present on the telephone line. The Caller ID transmission mode can be selected as either CID (Type I) or CIDCW (Type II), or the Auto Detect feature will automatically select the appropriate transmission mode depending on the state of the hook switch. Message types can be selected along with the primary packet fields. The packet ordering is completely arbitrary and can be changed to any order, along with changing the number of mark bits inserted after the packet fields. After selecting the desired transmission signal level and signal to noise ratio, starting the transmission sequence is initiated by simply pressing the start button on the tool bar.



Changing the Parameters

Changing the operational parameters begins with a click on the Advance Setup button in the tool bar. Organized into six logical subgroups, each parameter can be found simply by clicking on the subgroup and selecting the desired parameter from a list. Changing a parameter outside the standard maximum or minimum limits illuminates a reminder flag due to the unusual setting of the parameter. The following list shows a few of the programmable parameters:



FSK Modulator Settings:
- Mark Frequency
- Space Frequency
- Transmission Level
- Transmission S/N Ratio
- Twist Level
- Baud Rate
- Mark Space Bit Skew
- Interfering Tone Frequency.
- Interfering Tone Level

Ringing Signal Settings:
- Ringer Frequency
- Ringer Level
- Ringer Cadence
- Ringer On Time
- Ringer Off Time
- Ringer Cycles

Type I Caller ID Timing:
- Time to Line Reversal
- Time to DTAS
- Time to Ring Burst
- Time to Data Transmission
- Time to Ringing

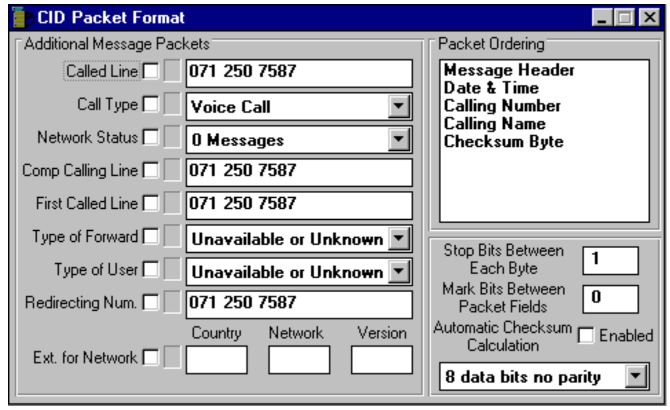
Type II Caller ID Timing:
- Time to SAS Tone
- Time to CAS Tone
- Time-out Period for ACK Tone
- ACK Minimum Trigger level
- ACK Frequency Tolerance
- Valid ACK digits
- Time to Data Transmission
- Transmit Even After
- Time-out Enable

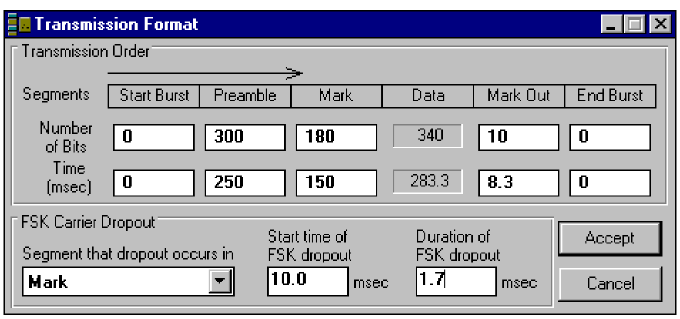
SAS/CAS Tone Settings:
- SAS Tone Frequency
- SAS Tone Level
- SAS On Time
- SAS Off Time
- SAS Duration
- CAS Low Tone Frequency.
- CAS High Tone Frequency.
- CAS Low Tone Level
- CAS High Tone Level
- CAS Tone Duration
- Noise During SAS/CAS

Telephone Line Settings:
- Line Voltage
- Loop Current
- Line Polarity
- Line Impedance

Additional CID Parameters and Packet Ordering

The CID Packet Format Window provides a visual indication of the currently selected packets and the order they will be sent. The packet ordering can be quickly changed by clicking on the packet and then simply dragging it to the new location. No complicated programming is required. Although most standards implement the data with 8 bits and no parity, for the unique systems that do not follow this the convention, the number of data bits and the parity are selectable. Additional stop bits and mark bits can be inserted between the packets to simulate the conditions caused by many central office system by just typing the number required.



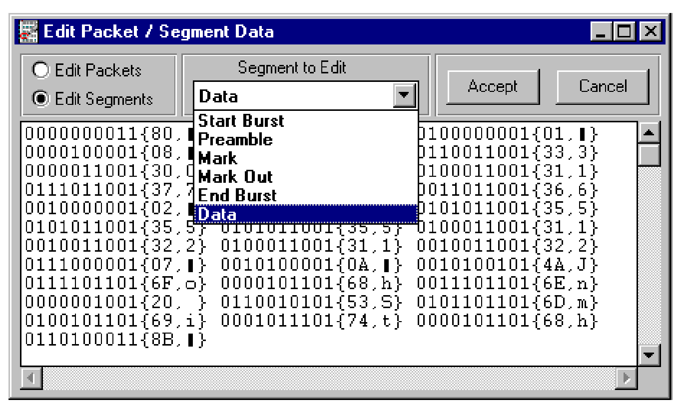


Visualizing the Transmission Data

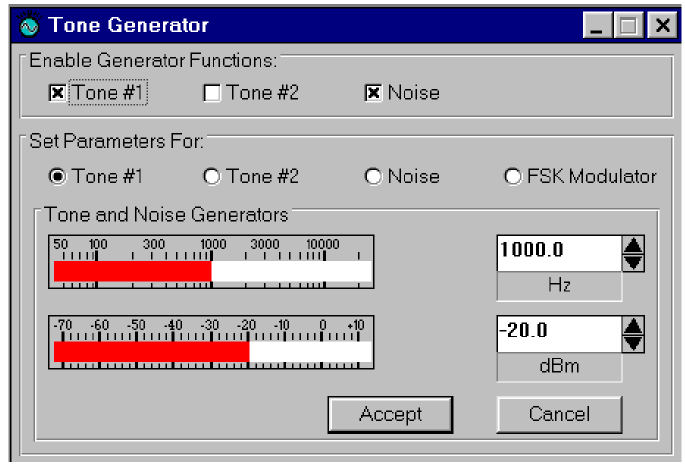
The Transmission Format Window shows the various segments of the current Caller ID data packet being sent in terms of the number of bits or transmission time. The Preamble, Mark, and Markout segments can be altered on the spot by entering either the number of bits to send, or the desired transmission time. Two additional segments called Start Burst and End Burst are made available for testing unusual exception conditions. Defining and controlling FSK signal dropouts are also performed using this window. The dropout can occur in any segment including the channel seizure and mark signal as defined in Bellcore SR-3004. If any segment has been bitwise altered from its nominal state, its title block becomes illuminated to serve as a reminder of the alteration.

Editing the Data Packet

The Editing Data/Segment Window allows byte-wise or bitwise modifications of the current Caller ID data stream. This helps to create virtually all possible exception conditions for designing and testing the embedded software of Caller ID devices. Both the information carrying packets and the segments can be edited by clicking on the appropriated radio button. A drop down list shows all the available packets/segment that can be edited. Existing bits or bytes can be modified, deleted, or new ones added. After clicking the Accept button, the new packet/segment is checked for editing errors and if none exist, the changes are acknowledged. Reminder flags will illuminate, signaling the alteration of a data segment or packet.



Flexible Tone Generation Capabilities



Apart from the Caller ID functions, a very flexible tone generation system is available in the software. Access to this feature is accomplished by simply clicking the Tone Generator button in the tool bar. One or two high purity sine wave tones can be generated with arbitrary frequency and amplitude. Changing either the level or frequency is done by typing in a new value for accurate tone generation, or by clicking on the desired value in frequency/level bar graphs for fast access. To assist in evaluating and developing FSK decoder circuits, one of the tone generators can be configured as an FSK modulator. Just as with the Caller ID FSK generator, this one is completely programmable in terms of frequency, level, baud rate, and bit skew. The FSK modulator can be set to generate mark tones, space tones, an alternating mark/space bit pattern, or a user defined bit pattern. The user defined pattern can be specified as either ASCII characters, hexadecimal values, or individual bit values. All the above options can be set in either a single shot or continuous operational mode. Engaging a special loop back mode allows for error rate testing which is helpful in verifying a FSK decoder design.

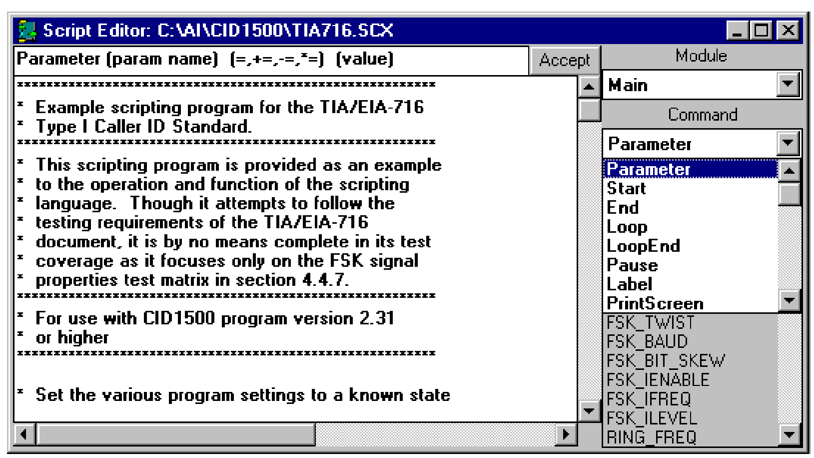
Scripting Language

With the programmable scripting language, complex or automated testing sequences are greatly simplified and can be saved or recalled at any time. A complete library of test can be used to fully qualify the Caller ID compliance of the equipment under test. The built-in script editor is designed to assist the writing of script files without having to study manuals and learn syntax. Writing a script file is as easy as clicking in the pull down box, which lists all of the commands available. Then a double click will have the command transferred to the composing line. The composing line is filled with a template of the selected command and always shows what is required next. A simple press of the enter key is all that is required to transfer the line to the script file in the editing window.

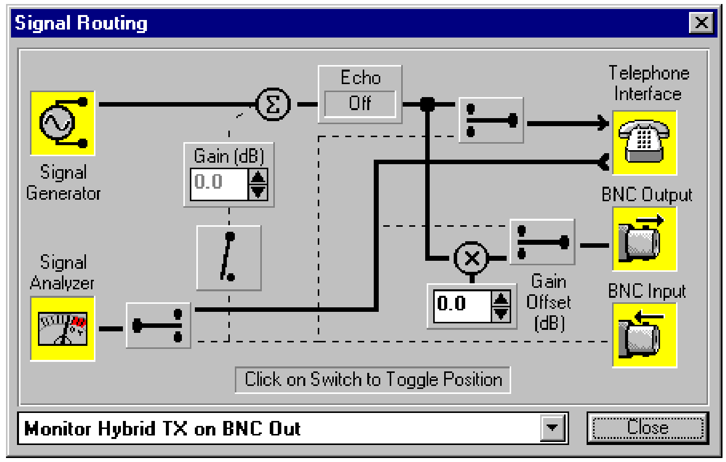
For easy programmability and maintenance, the script language also supports advanced data constructs including labels, branching, if... then... else..., loops and subroutines. Together with variable declarations, advanced user input, and flow control capabilities, very complicated test sequences can be programmed easily and quickly.

Script language commands also support advanced data logging and report capabilities. This allows users to generate test reports automatically and efficiently analyze test data. The data can be exported to maintain a database or used to create elaborate reports of a product's performance. Script language commands also allow direct contol of external hardware with digital outputs or monitoring of digital inputs for closed loop testing.

Sample scripts for both Bellcore and TIA/EIA testing are available for demonstration and evaluation purposes. Specifically, an example script file based on section 4.4.7 of TIA/EIA-716 is included with the CID1500 software. This script outlines a series of 53 separate tests effecting the FSK signal properties, including multiple impairments. This script program can also be used as a template to expand the testing to include the other physical layer and data layer tests specified in the TIA document. These sample scripts allow the user to start performing sophisticated tests right away without the need to learn the scripting language.



Controlling the Signal Flow



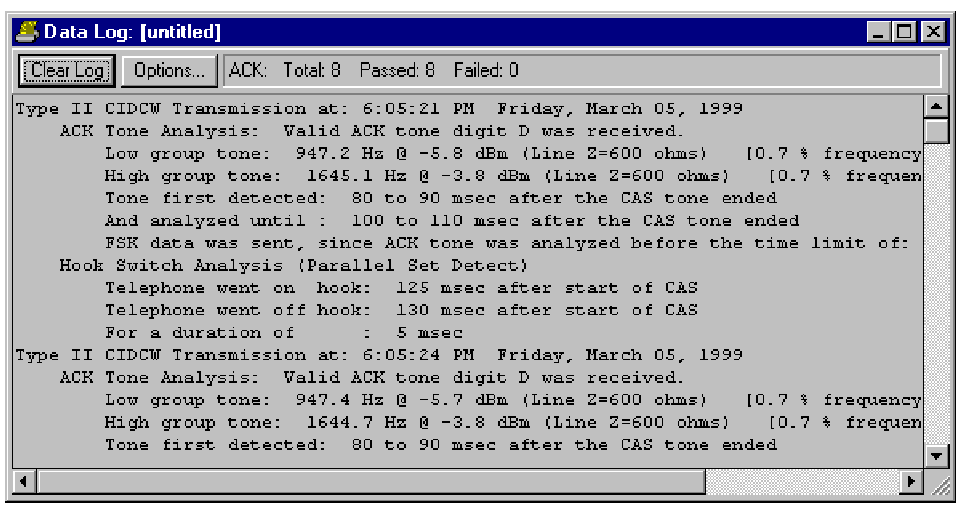
Routing of various signal paths is accomplished by simply clicking on the switch or selecting it from the drop down list for the most common configurations. These include monitoring the transmit signal, monitoring the receive signal, and using the BNC ports for direct access to the telephone interface. Loop through options for both the transmit and receive signal paths allow for the insertion of custom filters or impairments. Direct signal access to the telephone interface allows the telephone line signals to be injected and monitored without having to worry about line balance and impedance matching. A built-in signal mixer with gain offset for injecting external noise or audio during Caller ID transmission is helpful in simulating real life CO conditions and in performing TALK-OFF and TALK-DOWN tests on a CAS tone detector. Signals from the generators can be increased or decreased by up to 20 dB relative to the signal level at the telephone line interface. This feature can be handy when using the BNC output connector for development work, as adding a gain offset eliminates the need for remembering the level differences between the telephone interface and the BNC output. Plus, the telephone interface can be programmed to various line voltages and loop currents, with the option of reversing line polarities and selecting different line impedances.

Data Log File

The Data Log File holds the results of any Type II Caller ID transmissions and the progress of the script programs. An example of the contents of the data log file after a CIDCW transmission is shown below. Included in the message is an analysis of the ACK tone generated by the CPE, and any timing information concerning hook switch flashes that may have been generated.

An ACK counter displayed across the top of the data log window shows a running count of the total number of Type II Caller ID transmissions sent. Included with this is the number of ACK tones that were classified as being valid or passed, and the number which failed to meet the specified criteria. Scripting commands can be used to branch program execution if the last Type II transmission resulted in a passed or failed ACK tone. The values of the ACK counter can be displayed on the screen or written to the log file.

Also present on the data log window are two command buttons to clear the contents of the log window and to display the options for buffering the file to the hard disk.

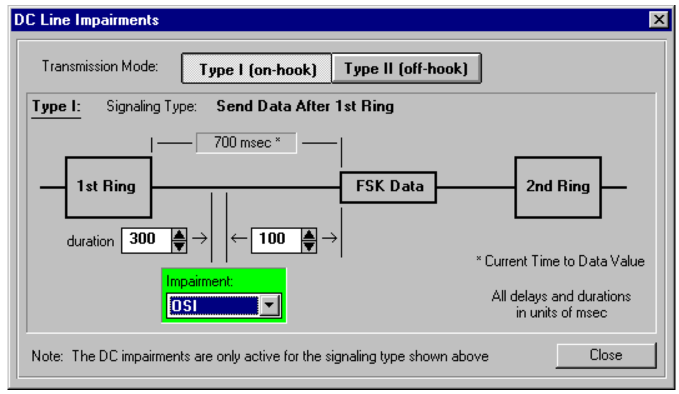


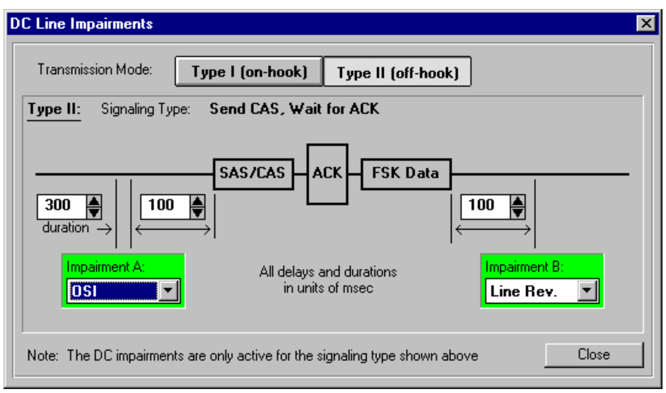
OSI & Line Reversal Impairments

DC line impairments such as Line Reversals and Open Switch Intervals (OSI's) can also be included in Caller ID transmissions. The addition of these impairments simulate possible conditions generated by various central office switches. With the addition of these impairments, the simulator is able to more accurately simulate real world conditions that can disturb the normal reception of the FSK data.

Type I Line Impairments

For Type I Caller ID transmission, a line reversal or OSI event can be inserted after the 1st ringing pattern and before the FSK data. The window to the right graphically displays the position and type of impairments that the user can generate for the Type I Caller ID transmissions. In this example a 300 msec OSI is being generated 100 msec before the FSK data is sent to the CPE.



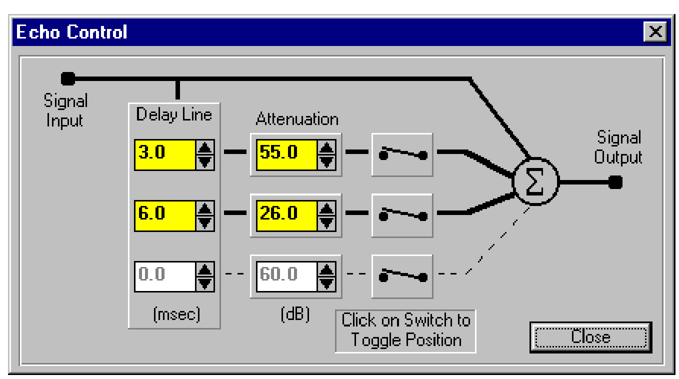


Type II Line Impairments

In a similar manner the DC impairments are also defined for the Type II Caller ID transmission. A line reversal or OSI event can be inserted before the SAS/CAS tone and after the FSK has been sent. The figure to the left shows a condition of an OSI event before SAS/CAS tone and a line reversal after the FSK data.

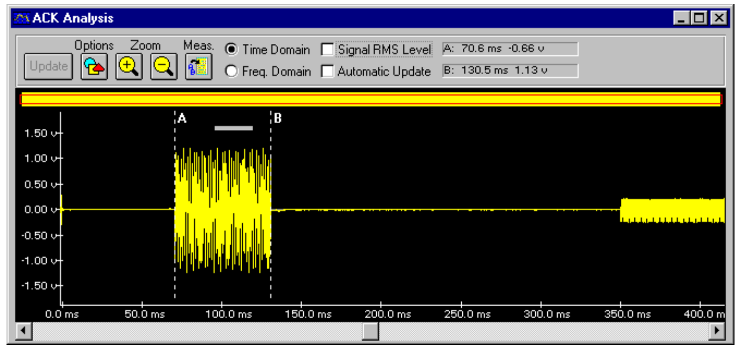
Signal Echo Generation

The AI-150 Simulator also has the ability for testing to the TIA/EIA-716 Standard for Type 1 Caller ID Testing. It supports the generation of up to three signal echoes at arbitrary delays and attenuation levels. The echoes can be enabled for any signal generated by the software as well as external signals injected at the BNC audio input. For each of the three possible signal echoes, the delay can range from 0 to 20 milliseconds, while the attenuation of each echo can be programmed to be in the range of 0 to 60 dB. The figure on the right shows the setup for creating two echoes at time delays of 3.0 and 6.0 milliseconds with attenuations of 13.0 and 26.0 dB respectively.

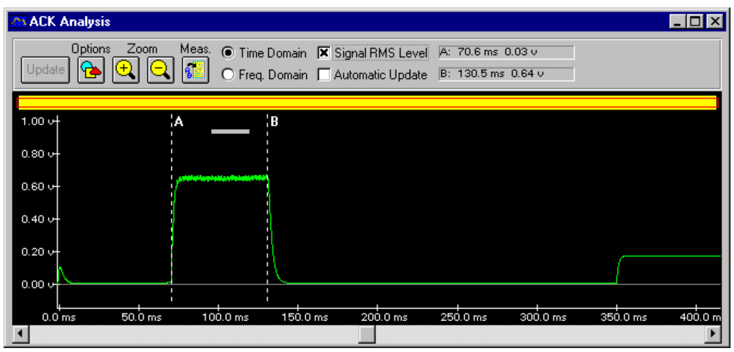


CAS/ACK Tone Time/Frequency Domain Analyzer (Optional)

A powerful CAS/ACK analysis tool is available as an optional hardware upgrade with the CID1500 program. With this option enabled, the CID1500 program will sample and process the CAS/ACK tone during a Type II caller ID transmission. The resulting data can be viewed much like a digital storage oscilloscope. This shows the user the exact timing of the ACK tone in relationship to the CAS/DTAS signal generated by the CID1500.



In the example shown to the right, an ACK tone from a CPE was captured and is displayed in its full span. In this case the ACK tone was generated 70 msec after the end of the CAS/DTAS tone. A small transient signal can be seen to be generated by the CPE at a time of 40 msec. The lower level signal starting at a time index of 350 msec is the beginning portion of the FSK data being transmitted to the CPE. Two cursors can be used for reading measurement points from the waveform. By simply clicking the mouse buttons, the cursors can be positioned anywhere on the waveform.

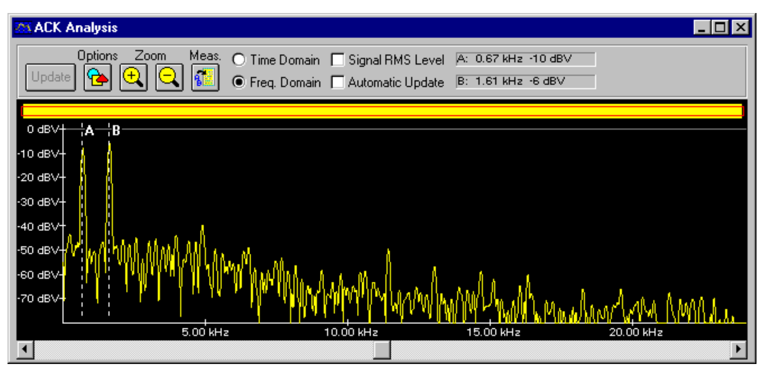


Displaying the RMS Amplitude

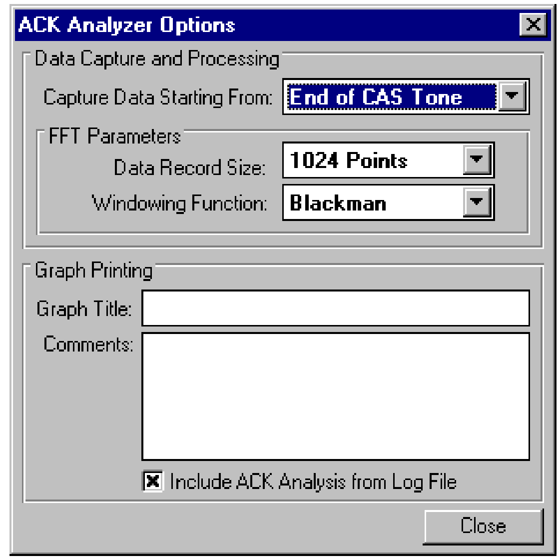
The analysis window to the left shows the RMS signal level of the ACK tone. The FSK data signal starting at a time index of 350 msec after the end of the CAS/DTAS tone. Displaying the RMS signal level can be useful in verifying that the ACK signal level is constant over its entire period, such that there are no slow ramp ups, or fluctuations in the signal level during the ACK tone that could disturb its detection.

Displaying the Frequency Domain

The last example to the right shows the frequency domain response of the ACK tone. A 1024 or 2048 point FFT can be calculated on the displayed data. The short horizontal bar displayed over the figure above represents the time span over which the FFT is calculated. Since this bar is positioned over the ACK tone, the FFT will generate a frequency domain plot of that signal. Moving the FFT bar is as simple as clicking the mouse on it, and sliding it to a new position in the time domain window. The FFT shows the full span view from 0 kHz to 24 kHz. Two distinct spectral peaks are clearly visible representing the two frequency components of the ACK tone. Since this particular CPE responds with a DTMF A tone, the spectral components are at a frequency of 697 Hz and 1633 Hz. Like the time domain view, the display can be zoomed into and cursors used to measure the signal levels at various frequencies.



FFT Analysis Options

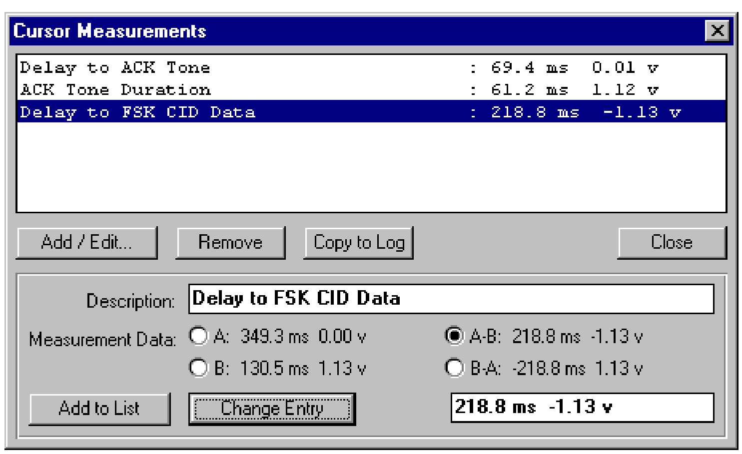


A number of options for the ACK tone analyzer shown to the right, can be accessed. The trigger point for the data capture can be selected from the Start of SAS Tone, the Start of CAS Tone, the End of CAS Tone (default setting), When ACK is Detected, and the Start of FSK Data Transmission. This allows for examining other sections of the Type II Caller ID transmission sequence.

The size of the FFT record can be changed from either 1024 points to 2048 points. Using a larger record size increases the frequency resolution; however, the record time span also increases from approximately 21 msec to 42 msec. As such, to obtain accurate results any signal measured will have to be stable for a longer time period. In addition to the variable record size, different windowing functions can be specified for the FFT. Aside from the default Blackman window, the Hanning, Hamming, Nutall, and rectangular windows can be used. The different windows provide a basic trade-off between FFT frequency resolution and side lobe attenuation. The Blackman window provides excellent side lobe attenuation, but its main lobe is wider than with the Hanning or Hamming windows. The rectangular window provides the best frequency resolution due to its narrow main lobe; however, its poor side lobe attenuation makes it difficult to resolve any low level signals.

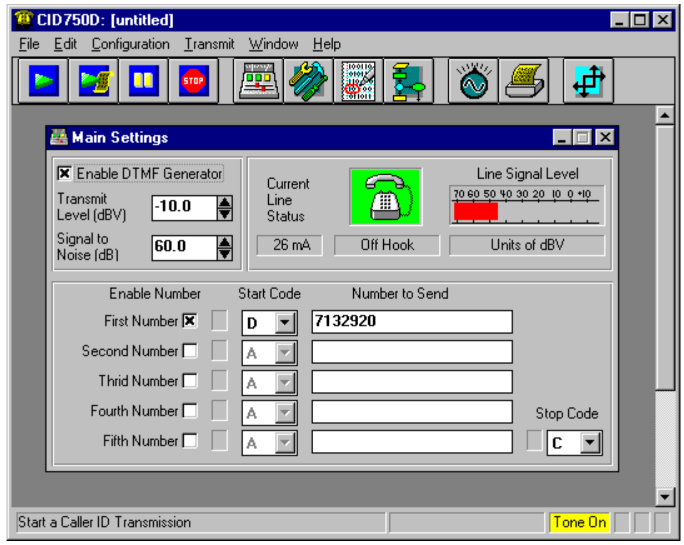
Programmable Cursor Measurement

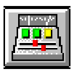
The cursor measurements add a valuable tool to quickly measure many of the needed timing relationships in a Type II Caller ID transmission. The ACK option allows the user to record the measurements made on the waveform and then add them to the log file for a permanent record. In the Cursor Measurement Window shown below, a list of up to 10 arbitrary measurements can be created, edited, or cleared. The data recorded can be either absolute measurements at the cursor location or differential measurements between two cursors.



CID 750D Software

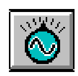
This software is intended for the DTMF based caller ID formats. The CID 750D has the same look and feel to the CID 1500 and shares many of the same features. This enables the operator to get up to speed right away. The common features include viewing and editing the transmission data, writing script files to automate the testing sequences and routing the signals to the BNC connectors via the signal routing window.





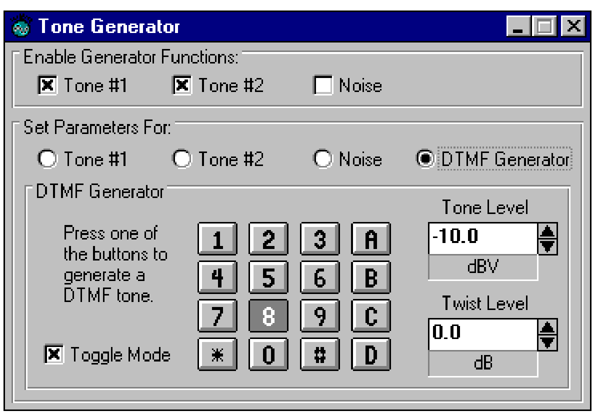
Main Settings

The main settings window contains all the primary controls for quick operation. The DTMF signal level and signal-to-noise ratio is specified in this window along with the data that will be sent. Up to five different DTMF numbers can be sent, with any of the 16 possible DTMF tones used in the numbers. An arbitrary DTMF start code can be assigned and a stop code is specified from a list of DTMF digits, or the stop code can be left out all together if required.



Tone Generator

In the 750D the tone generator can be configured to form a DTMF generator. In this mode of operation, clicking the mouse on the desired DTMF tone will produce that tone at the specified signal and twist level. Access to this feature is accomplished by simply clicking the Tone Generator button in the tool bar and then selecting the DTMF generator. The tone generator also allows for the generation of one or two high purity sine wave tones which can be configured as in the CID 1500 software.



DTMF Generator Settings:

- Signal Level
- Signal to Noise Ratio
- Twist Level
- Row #1 Frequency
- Row #2 Frequency
- Row #3 Frequency
- Row #4 Frequency
- Column #1 Frequency
- Column #2 Frequency
- Column #3 Frequency
- Column #4 Frequency
- Frequency Deviation
- DTMF Tone On Time
- Pause Time Between Tones

Caller ID Timing:

- 1st Line Reversal Enable
- Time to 1st Line Reversal
- Ring Burst Enable
- Duration of Ring Burst
- Time to Ring Burst
- 2nd Line Reversal Enable
- Time to 2nd Line Reversal
- Wait for Off Hook Enable
- Off Hook Timeout
- 3rd Line Reversal Enable
- Time to 3rd Line Reversal
- Time to DTMF Tones
- Ringing Enable
- Time to Ringing

Programmable Parameters

Like the CID 1500, for FSK Caller ID, the CID 750D provides a large number of parameters controlling every aspect of the Caller ID transmission. Organized into four logical groups each parameter can easily be changed manually or through the scripting language. If in the case a parameter is outside its normal operating range a reminder flag illuminates. The following is a list of a few of the programmable parameters:

Ringing Signal Settings:

- Ringer Frequency
- Ringer Level
- Ringer Cadence
- Ringer On Time
- Ringer Off Time
- Ringer Cycles

Telephone Line Settings:

- Line Voltage
- Loop Current
- Line Polarity
- Line Impedance

Hardware Architecture

The Telephone Signal Processing Card (TSPC) is a PC plug-in card utilizing a powerful DSP based architecture. This DSP approach allows for complete control over all signal generation and analysis. New measurement algorithms, new standards and new features can all be downloaded from the PC to the TSPC by simply running a new piece of software. This allows for an easy upgrade path in the future without the requirement for new hardware. The operator also has control of the subscriber line interface including the line voltage, line impedance, line polarity and loop current. An extremely flat frequency response allows for accurate generation of the test signals. The card also includes a ringing generator which produces a low distortion sine wave output of up to 80 Vrms. This no compromise approach to the simulator means that it is able to generate the real world signals the receiving equipment is likely to encounter.

System Requirements

Processor: 386DX min., 486DX recommended
Ram: 8 MB minimum
Operating System: Windows 3.1, 3.11, 95
Full length AT ISA slot

Warranty

A limited one-year warranty is standard, with extended service agreements and calibration services available from the factory. Contact us for further details.

Tone Generator

Output Level	-70 dBm to +6 dBm +/- 0.3 dB
Frequency Range	50 Hz to 10 kHz
Flatness	+/- 0.3 dB
THD+N	0.09% C-message
Harmonic Distortion	> 65 dBc
Frequency Accuracy	0.015%

FSK Generator

Output Level	-60 dBm to 0 dBm +/- 0.3 dB
Frequency Range	100 Hz to 5 kHz
Flatness	+/- 0.1 dB

Noise Generator

Output Level	-60 dBm to -8 dBm +/- 0.3 dB

Telephone Line

Output Impedance	600, 900 +/- 2%
	Complex +/- 3%
Loop Voltage	20 - 52 Volts +/- 1V
Loop Current	20 - 40 mA +/- 10%

Ring Generator

Output Level	0 Vrms to 80 Vrms.
Frequency Range	10 Hz to 600 Hz
Flatness	+/- 0.2 dB
THD+N	0.1%
Frequency Accuracy	0.015%
Ringer Load	5 REN

Analyzer

Level Accuracy	+/- 0.2 dB
Frequency Range	10 Hz to 10 kHz
Flatness 100 Hz to 5 kHz	+/- 0.2 dB
Maximum Input	+14 dBm
Residual Noise	<-70 dBmC

ACK Analyzer

Level Accuracy	+/- 0.2 dB
Maximum Input	+6 dBm per tone
Minimum Input	-20 dBm
Frequency Resolution	0.1 Hz

Note: Specifications are subject to change without notice.

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Ordering Information

Product Name	Model	Options	Possible combinations with other Models
Caller ID Signal Simulator Standard Version	**AI-150**	**AI-E001** **AI-K001** **AI-K003**	AI-150/240 AI-150/330 AI-150/240/330
Caller ID Signal Simulator Production Version	**AI-120**	**AI-K001** **AI-K003**	

Related Products

AI-80: Standalone Universal Caller ID Generator

AI-80 is a low cost stand-alone Caller ID Simulator which is specially designed to automate production testing of Caller ID CPEs. Test scripts or sequences are user programmable and can be downloaded to the AI-80 via a computer RS-232 interface.

AI-240: CAS Evaluation System - TSPC-CG, CAS2200

The testing to the Bellcore test plan SR-TSV-002476 is critical to the successful deployment of the CIDCW capable product. The AI- 240 brings together all of the equipment required, data logger, equalizer, CAS generator, and P.56 speech level meter, to perform the talk-down and talk-off testing. Included in the software is the probability calculator which calculates the talk-down and talk-off rates automatically.

AI-150/240: Dual System Upgrade - TSPC-CSG. CID1500, CID750D, CID950N, CAS2200

This model offers the greatest flexibility to the operator. It can be used as a Caller ID simulator with the full functionality of the AI-150 or configured to perform talk-down and talk-off testing like the AI-240 by starting the appropriate software package.

AI-330: Telephone Signal Analyzer - TSPC-TG, TSA3300

The AI-330 combines the functions of a Telephone Signal Analyzer and a basic Caller ID Signal Simulator together.

AI-410: TSPC with Low Level Driver - TSPC-DG, DDP1100

The AI-410 provides a basic building block for user who would like to integrate the simulator into a more complex automated testing environment with the low-level driver provided.

99/1.0