

ADVENT INSTRUMENTS INC.

TSA3300 TELEPHONE SIGNAL ANALYZER

***User Guide
&
Reference
Manual***

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■ Section 1

Installation & Setup

Before You Begin

Verifying the System Requirements

The TSA3300 Telephone Signal Analyzer program requires the following minimum system setup for the target computer.

- Intel 486DX33 PC computer, or equivalent
- VGA type monitor
- Microsoft Windows 3.1, 95/98 operating system
- Eight Megabytes of RAM
- One full length 16 bit ISA expansion slot

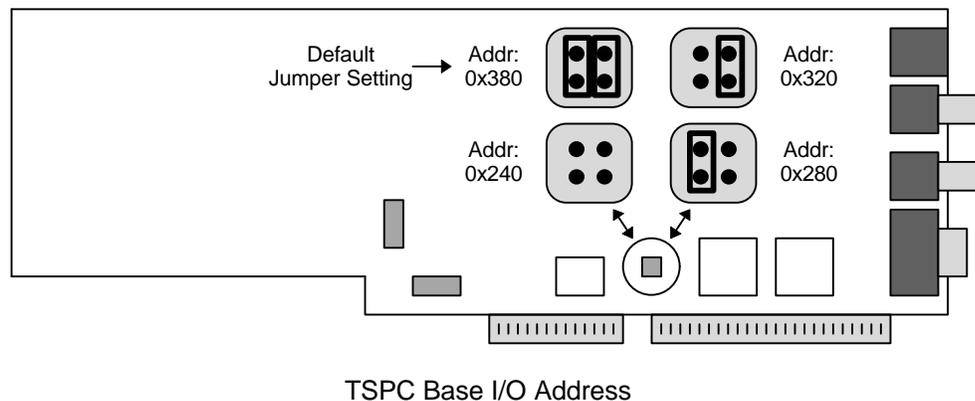
Backup the Software

Make a backup copy of the software disk before beginning the software installation. Keep the original disk in a safe location.

Installing the Telephone Signal Processing Card

Before installing the Telephone Signal Processing Card (TSPC), verify that the correct base I/O address is set. The base I/O address used is determined by the settings of two jumpers shown in the figure below. The factory default settings includes both jumpers, as shown below, for a base I/O address of 0x0380. This address should be free of conflict from most other peripheral cards; however, in case of conflict, change the jumpers to select an alternate base I/O address. The TSA3300 program will automatically scan all four possible base I/O addresses for a TSPC. Note, that the TSPC is designed to avoid interrupt and DMA channel conflicts, and as such, these parameters do not need to be configured. The only possible conflict between peripheral cards is with the base I/O address selected.

The TSA3300 program will support systems with up to four TSPC's, providing up to four independently controllable telephone lines. However it is important to insure that each TSPC installed is set to a unique base I/O address.



To install the TSPC:

1. Turn off the computer, including all external peripherals
2. Leave the power cable connected to a grounded outlet, so that the system is grounded
3. Remove the cover from the computer
4. Locate an unused 16 bit ISA expansion slot in your system
5. Insert the TSPC into the expansion slot
6. Replace the cover from the computer

Note: The TSPC can create high voltages. Ensure adequate space between the TSPC and other adjacent peripheral cards in the computer. Also, always re-attach the cover to the computer before turning it on in order to ensure that no contact can be made to the high voltage portions of the TSPC.

Installing the Windows Software

To install the software package, turn on the system and launch either the Windows 3.1, or 95/98 operating system. Insert the floppy disk into the 3.5 Disc Drive (usually drive A:). For users of Windows 3.1, from the Windows Program Manager, select menu command [FILE] [RUN]. Type "A:\setup" and press ENTER. For users of Windows 95/98, press the START button, followed by the RUN selection. Then type "A:\setup" and press ENTER.

The setup program will extract all the necessary files and install them into the directory you select. The default directory is set to "C:\AI\TSA3300"; however, alternate directories may be chosen.

The Setup program will automatically create both a program group and program icon within the Windows Program Manager

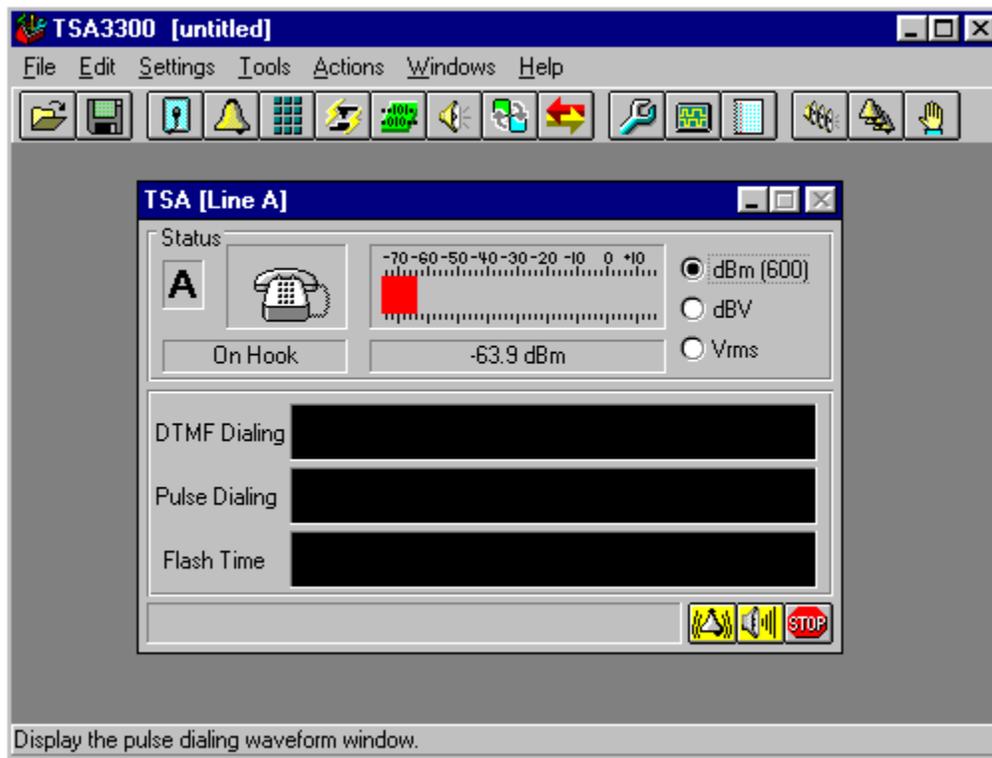
Note: The software does not require any modifications to the AUTOEXEC.bat or CONFIG.sys files.

Using Windows NT 4 and Windows 2000

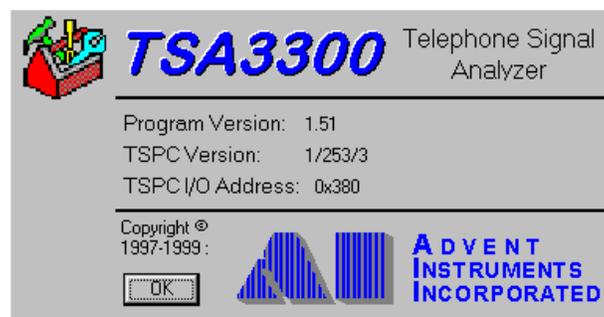
The standard release of the TSA3300 software contains drivers that support Windows 3.1, 95, and 98. The Windows NT & 2000 operating systems are not compatible with the standard TSA3300 drivers. However, a version of the driver that operates under Windows NT & 2000 is available from the installation CD or our web site at www.adventinstruments.com. The driver package includes instructions for its installation in a Windows NT & 2000 system. If problems are encountered during the installation, please contact us for assistance.

Testing the Installation

Once you have finished installing the hardware and software, you can verify the correct setup by launching the TSA3300 program from the Windows Program Manager (Windows 3.1), or from pressing the [START] button on the Task Bar (Windows 95). Once the program has finished loading itself in the system, the screen should display the following.



To verify the software version and the TSPC version codes, select the HELP menu by either clicking on HELP or pressing ALT-H. Select the "About TSA3300" option within the HELP menu. The following window will appear until you press the OK button.



The window will display the software version code, TSPC product, revision, and option codes, and the TSPC base I/O address being used. If any problems are encountered in the use of this product, include the above information in any correspondence with technical support.

■ Section 2

Getting Started

Starting Up

Once the TSA3300 Telephone Signal Analyzer program has started, it is ready to begin performing any of a wide range of functions. For example, connect a telephone to the RJ-11 jack on the rear of the Telephone Signal Processing Card (TSPC) and press the F7 key. This will start ringing the telephone with one of the pre-defined ringing patterns. Going off-hook with the telephone, or pressing F8, will cease the ringing. Sending network tones, such as dial tone, is done by simply pressing the F6 key. Again F8 can be used to halt the network tones.

System Features and Capabilities

While this program is written with a windows graphical user interface for ease of use, the program allows for sophisticated operations in order to properly test and analyze the characteristics of various telephones.

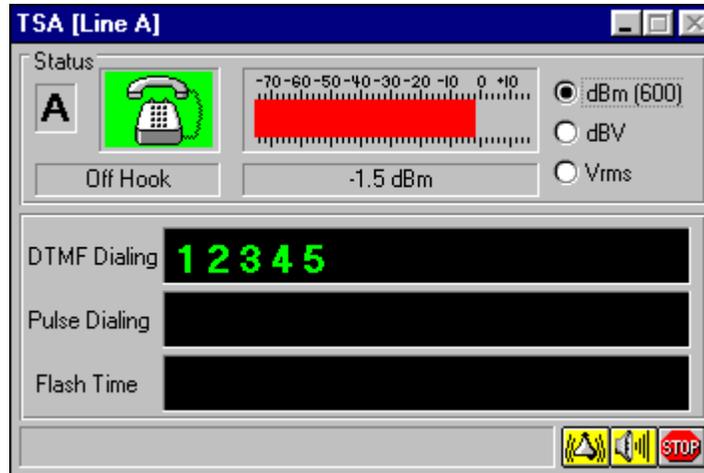
The program will provide all the basic functions required of a telephone signal analyzer. This includes performing DTMF dialing analysis, pulse dial analysis, flash timing measurement, and a programmable ringing generator. DTMF digits are analyzed for their low and high tone group levels and frequencies, along with tone duration and inter-digit delay. Likewise, pulse dialed digits are broken down into break times, make times, make-break ratios, pulses per second, and inter-digit intervals. Limits can be assigned to many of these measurements, allowing the program to automatically classify digits that fall inside or outside of the programmed limits. The ringing generator can be programmed over a wide range of frequencies, levels, and cadences, along with a selection of different wave shapes.

More advanced features include the ability to send Caller ID information with ringing (Type I), or performing off-hook Caller ID (Type II, or CIDCW). Along with the basic call setup messages, Visual Message Waiting messages can be generated for both Type I and Type II Caller ID. A range of signaling methods support most international Caller ID systems as well as the North American Bellcore system. Also, arbitrary network tones can be defined and generated at any time. This includes dial tone, busy tone, ring back, call waiting, and others.

A unique and powerful feature to the TSA3300 program allows for linking events with programmed actions. Various events, such as going off-hook, going on-hook, receiving dialing digits, line flash, key pressed, external inputs, and others, can be programmed to generate an action. The actions are user defined via a scripting language and can change various program settings, ring the telephone, or generate network tones. For example, the default setting of the TSA3300 program will link the off-hook event to an action that generates dial tone. As such, when a telephone goes off-hook, the TSA3300 will automatically start sending dial tone. This is one of the simplest of event/action relationships;

Basic Operation

Up to a maximum of four telephone lines can be supported by the TSA3300 program. At startup, the program queries the PC for the maximum number of TSPC's present in the system. Each TSPC, that supports the TSA3300 program, is assigned a letter from A to D. These letters are used throughout the program to distinguish the different telephone lines from each other. For each telephone line supported, a separate window is displayed showing the status and activity of that telephone line. An example of this main control panel for each line is shown below.



The top portion of the window displays the telephone line letter (A in this case), status of the telephone line (off hook in this case), and the current signal level present on the telephone line. Any DTMF or pulse dialed digits generated by the telephone under test are shown in the lower part of the main control panel. In the example shown above, the DTMF digits 1 to 5 have been dialed. No pulse dialed digits are displayed. If the telephone performs a line flash, the duration of the flash is displayed just below any pulse dialing digits.

The three buttons at the lower right corner of the window are used to generate ringing or network tones. Clicking the left most button, in the shape of a bell, will start the ringing generator, provided the telephone is on-hook. Likewise, clicking the middle button, will start the currently selected network tone, provided the telephone is off-hook. Finally, the red stop button will terminate any ringing or network tone generation. Since a number of different ringing patterns or network tones can be defined, clicking the right mouse button over the ringing or network tone button displays a list of defined patterns to select from.

Finally, the panel to the left of the three buttons is used to display what operations are being acted upon the telephone line. For example, when starting the ring generator, the ringing pattern name will be displayed in a yellow background color. Likewise, for a network tone, its name is displayed with a light blue background color. Any events that are triggered will be briefly shown in green, along with the corresponding action name. For an example of this, generate a line flash to clear the DTMF and pulse dialing displays. Then dial "5551111". The default settings of the TSA3300 links the DTMF dialing of 555-1111 to the generation of a busy tone. The busy tone can be stopped by flashing the line or pressing the stop button. The DTMF number of 5551234 has also been linked to generate the ring back tone. In a similar manner, other events can be associated with various types of actions.

For More Information

A more detailed description of the functions of the main control panel for each line can be found in section 3: Main Panel Operation.

For an explanation of how to change the various program settings such as the telephone interface characteristics, ringing patterns, network tone definitions, Caller ID parameters, and others, see section 4: Program Settings.

The operation of the program tools such as the tone generator and analyzer, log of events and dialing are described in section 5: Using the Program Tools.

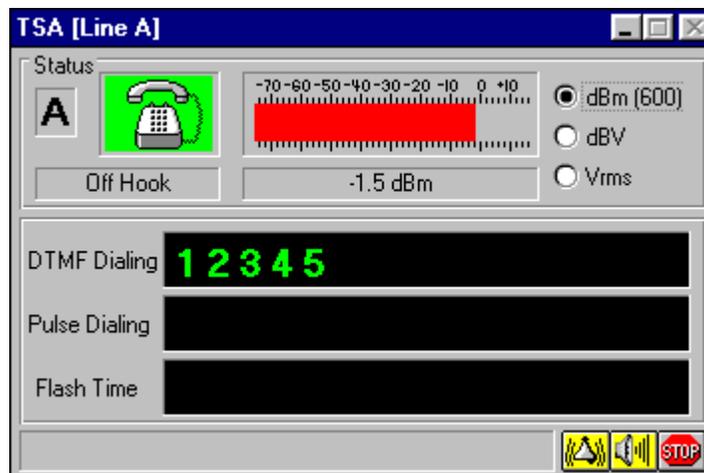
■ Section 3

Main Panel Operation

Line Designations

The main panel window is the primary means to view the actions of any telephone under test along with controlling any signals sent to the telephone. A separate window is displayed for each TSPC present in the system, up to a maximum of four. At program startup, a search is conducted for TSPC's capable of supporting the TSA3300 program. Each unit found is assigned a device letter from A to D. As such, a system with only one TSPC will have only one main panel window with the designation of A. Two TSPC's creates two main panel windows with the designations A and B respectively. And so on, until the maximum of four main panels are reached.

In order for the TSA3300 to properly locate multiple TSPC's, they must be assigned different PC I/O addresses. The I/O address is set by the insertion (or lack thereof) of two shorting jumpers present on the TSPC. The User Guide & Reference Manual, Section 1, shows the different possible jumper positions and their effect on the PC I/O address. It is important to ensure that if multiple TSPC's are present in the system, that they be set to different I/O addresses. The TSA3300 program searches for the cards starting at an address of 0x380 and works its way down to 0x240. The first card found is assigned the letter A, with the second B, the third C, and finally the fourth D (if present). Thus by knowing the PC I/O address of each TSPC, it is possible to determine which designation corresponds to which physical telephone line.



The designation for each line is shown in the upper left corner of the main panel window, as well as in its title bar. The figure above shows the window for line A.

Note: Selecting the [HELP] [ABOUT TSA3300] menu item displays the program version as well as the PC I/O address and version ID for each TSPC found in the system. This can be useful in locating TSPC's that are set to an incorrect address or that fail the search process at program startup.

Telephone Line Status & Level

The current status of each telephone line is shown in the upper left corner of the main panel window as an icon and brief text description. Up to six different icons can be displayed showing the telephone as off-hook, on-hook, ringing generator active, network tone generator active, and sending FSK Caller ID data in both the on or off hook state. All of the possible icons are shown below.



Telephone on-hook



Telephone off-hook



Ring generator active



Tone generator active



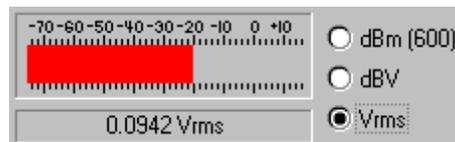
Send FSK data (on-hook)



Send FSK data (off-hook)

The brief text description matches the displayed icon and provides additional information when sending type I (on-hook) and type II (off-hook) Caller ID data.

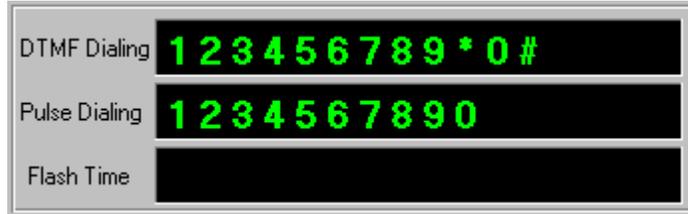
The top half of the main panel window also includes a bar graph indicating the total signal level present at the telephone line. The level meter measures the RMS level of signals generated from either the telephone under test or the TSPC over a wide frequency range of 50 Hz to 10 kHz. The level reading can be displayed in units of dBm, dBV, or Vrms by selecting the corresponding button to the right of the bar graph. The bar graph will only show the logarithmic units of dBm or dBV, while the numeric reading can also be displayed in units of Vrms. In the case that the selected units are Vrms, the bar graph uses units of dBV. Below is an example of the level meter set to units of Vrms.



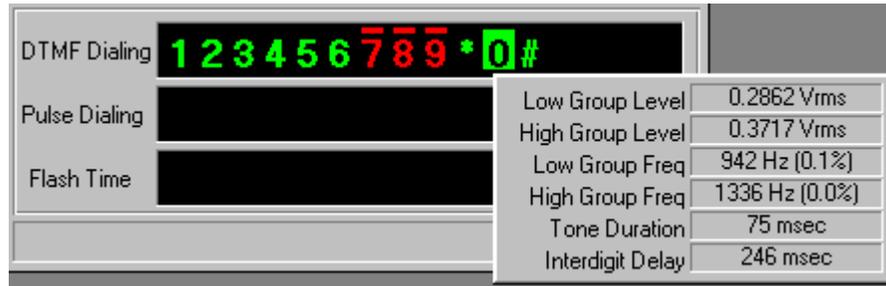
It is important to note that the units of dBm is with respect to a 600 ohm load. Therefore, a 0 dBm reading is equivalent to the voltage level needed to dissipate 1 milliwatt of power into a 600 ohm load. This voltage level is 0.7746 Vrms or -2.22 dBV. As such, the difference between the readings in dBm and dBV will always be 2.22 dB.

DTMF / Pulse Dialing

The lower portion of the main panel window is used to show the dialing and flash results from the telephone under test. DTMF dialed digits are shown in the first line, while pulse dialed digits are shown in the second line.



The above figure was created by dialing all of the DTMF keys on a sample telephone, then changing to pulse dial mode, and dialing the ten pulse dial digits. As each DTMF digit is received it is analyzed for its low tone group level and frequency as well as its high tone group level and frequency. These measurements are then compared to a set of programmable limits. If within the limits, as shown above, the digits are displayed in a green color. However, if any of the measured elements are outside the limits, the digit will be displayed in red with a bar over it. The next figure shows DTMF dialed digits in which some of them (digits 7, 8, and 9) were outside of the acceptable limits.



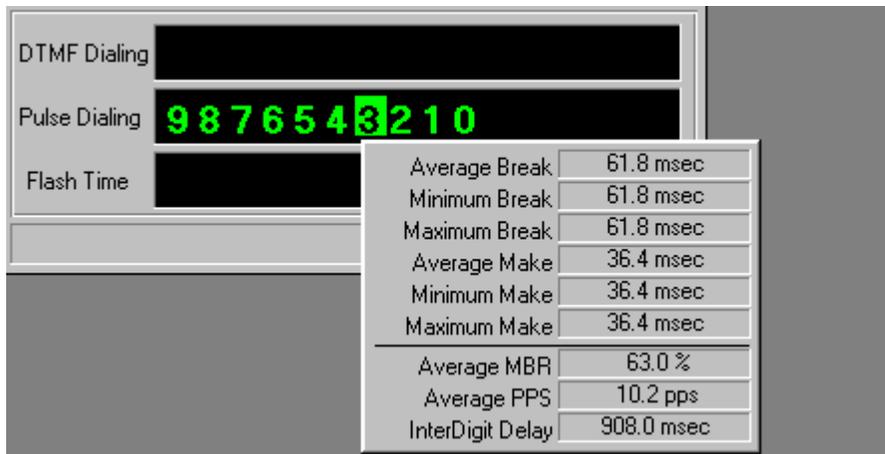
Two phones were used to generate the above figure. The first telephone has a highly accurate DTMF generator, while the second telephone exhibits a much larger frequency error. The frequency tolerance in this example was reduced from its nominal 1.5% to a much tighter 0.5%. This change was made inside the DTMF settings window, which is displayed by selecting the [SETTINGS] [DTMF DIALING SETUP] menu command. The first telephone still fell within the limits and displayed the green (passed) DTMF digits, while the second telephone was used to dial the digits 7, 8, and 9, which fell outside the tighter tolerance setting. As such, these digit are displayed in red with the bar over them.

The characteristics of any DTMF (or pulse) dialed digit can be examined by pressing and holding the right mouse button over the selected digit. A pop up window shows the DTMF levels, frequencies, duration, and inter-digit delay for any DTMF digit as long as the right mouse button is pressed. The above example shows the characteristics for the DTMF digit 0. This can be compared to the characteristics of the failed DTMF digit 8, shown below.



The highlighted item of the failed DTMF digit shows what measurement failed its limit. In this case, the low group tone frequency limit was exceeded since the limit was previously changed from its nominal 1.5% setting to 0.5%. The cause of any failed digit can be examined by simply pressing the right mouse button over the digit in question.

Note that the characteristics of the last dialed digit can be displayed in a permanent window by selecting the [TOOLS] [SHOW LAST DIGIT RESULTS] menu command. See section 5: Using the Program Tools, for more information.

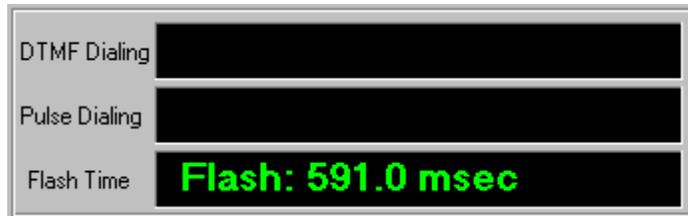


As with the DTMF digits, the characteristics of any pulse dialed digit can be examined in the same manner. Pulse dialing digits are measured in terms of their average, minimum, and maximum break time as well as their average, minimum, and maximum make time. From these measurements, the average make-break ratio (MBR) and the number of pulses-per-second (PPS) is calculated and displayed. The time interval between digits is also shown as the inter-digit delay. In the example above, the pulse digit 3 is examined. The results indicate that the digit is well within the limits for the make-break ratio and pulses-per-second. These limits, as well as the limits for the flash timing can be changed by selecting the [SETTINGS] [PULSE DIALING & FLASH] menu command. Any characteristic of a pulse dialed digit that falls outside the programmed limits is highlighted in the same manner as a failed DTMF digit.

Flash Timing

Anytime the telephone under test performs a line flash, the duration of the on-hook interval will be measured and displayed. The minimum and maximum

limits for a valid flash interval can be viewed and changed by selecting the appropriate menu command or toolbar button. In the following figure, a valid flash time of 591 msec was recorded and displayed.



As with the DTMF and pulse dialing digits, a valid flash is also shown in a green color. If the on-hook interval is less than the minimum limit or greater than the maximum limit, the flash will be shown in a red color. The default limits for the flash time is a minimum of 500 msec and a maximum of 750 msec. Since the following flash time was only 415 msec, it is displayed in red. The minimum flash time that can be measured is 200 msec. Any on-hook intervals less than this limit are assumed to be part of a pulse dialing digit and is displayed as such. For any on-hook intervals that exceed the maximum programmed flash time by less than 500 msec will be displayed as an invalid flash in red. However, on-hook intervals greater than 500 msec beyond the maximum flash time are treated as simply the phone going on-hook and then off-hook.



The DTMF and pulse dialing display is cleared once a line flash is detected, or the telephone goes on-hook then off-hook. Likewise, the results of a line flash are cleared upon dialing any digits or the telephone going on-hook then off-hook. A maximum of 16 DTMF and pulse digits can be shown at any one time. Beyond 16 digits, the display will begin to scroll to the left on any successive digit. As a result, only the last 16 digits are displayed. However the Event and Dialing Log can be set to capture the results of an unlimited number of digits.

Call Waiting Deluxe Support

Call Waiting Deluxe (CWD) services offered by telephone companies, allow a user to hold, transfer, conference, or drop callers following a Caller ID with Call Waiting (CIDCW, or Type 2) message. The term "Type 2.5" is sometimes used to refer to Call Waiting Deluxe features.

A telephone requests a CWD service by sending flash and DTMF digit after a successful CIDCW transmission. Each CWD option is represented by a unique DTMF code. The TSA3300 program will display a message representing the CWD service if a DTMF code is received within a specified time window following a hook switch flash. In the following example, "CWD: Forward" is displayed on receiving a DTMF 9 after a valid flash.



The time period between the end of the flash and DTMF digit must fall within a specific time window to be considered a valid CWD service request. If the DTMF is generated too late, the central office switch may ignore it, while if it arrives too early, the switch may not be ready to accept it. The minimum and maximum delays from the flash to the DTMF digit are specified in the Pulse Dialing & Flash Setup window. If the timing falls outside the valid range, it will be displayed in red, as shown below.



All of the flash timing, CWD interval timing, and DTMF digit characteristics can be recorded in the event log. Depending on the filter settings, the data recorded in the event log can be specified to meet various requirements. The following figure shows two CWD service requests captured in the data log window. The first was successful, while the second failed, due to an invalid delay between the hook switch flash and the DTMF digit. As seen below, the entire CWD event is broken down into 3 entries. The first records the flash time, the second records the interval time (from end of the flash to the start of the DTMF digit) and the CWD service being requested, and finally, the third entry records the DTMF digit information. For more information on the Data Log of Events, see section 5-2.

Data Log of Events		
Line A :	FLASH	Duration=503.6 msec
Line A :	CWD 9	Forward ,Time to DTMF=473.0 msec
Line A :	DTMF 9	FL/FH=854(+0.2%)/1482(+0.3%)Hz LL/LH=-5.8/-3.5dBm Dur=160msec
Line A :	FLASH	Duration=502.0 msec
Line A :	CWD 3 FAIL	Conference ,Time to DTMF=661.1 msec
Line A :	DTMF 3	FL/FH=698(+0.1%)/1480(+0.2%)Hz LL/LH=-5.6/-3.5dBm Dur=194msec

In order to detect the CWD service request, both the flash timing and DTMF digit must meet the defined limits. As such, the hook switch flash time must fall between specified minimum and maximum limits. Also the DTMF digit must fall within limits for level and frequency. If either the flash timing or DTMF digit is invalid, no CWD service request will be displayed.

Within the TSA3300 program, an event may be programmed to trigger various actions on the successful reception of a CWD service request, or an action can be triggered on an invalid CWD service request. These new events and actions are programmed within the Automatic Actions setup window (see section 4-8).

The Call Waiting Deluxe feature, described in the Bellcore GR-416-CORE document, defines the linkage between the received DTMF code and the CWD function as shown below.

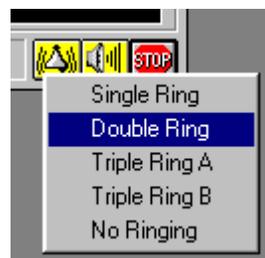
Digit	Function
3	Conference
4	Drop Last
5	Drop First
6	Hold
7	Drop
8	Announce
9	Forward

If any DTMF digits, other than those defined above, are detected following a hook switch flash, the program will display "(undefined)".

Generating Ringing or Network Tones

By pressing one of the yellow buttons located at the bottom right corner of the main panel, the TSPC will start to generate a ringing pattern, or a network tone. The first button, in the shape of a bell, starts the ringing generator, provided the telephone is on-hook. Pressing this button while the telephone is off-hook is ignored. Likewise, the middle button, in the shape of a speaker, starts a network tone, provided the telephone is off-hook. If the telephone is on-hook, no tones will be generated. Pressing the right most button, in red, will stop any ringing or network tone generation in progress.

A number of different ringing patterns and network tones can be defined within the program. The ringing patterns can be viewed and edited by selecting the [SETTINGS] [RING GENERATOR SETUP] menu command, while the network tone definitions are viewed and edited by selecting the [SETTINGS] [NETWORK TONES] menu command. When pressing the yellow ring or network tone button on the main panel, the last ringing pattern or last network tone definition is repeated. However, by pressing the right mouse button over either of the buttons, a pop-up menu appears with all of the defined patterns, as shown below. To change the ringing pattern, simply select the new pattern from the list. Likewise with the network tones.



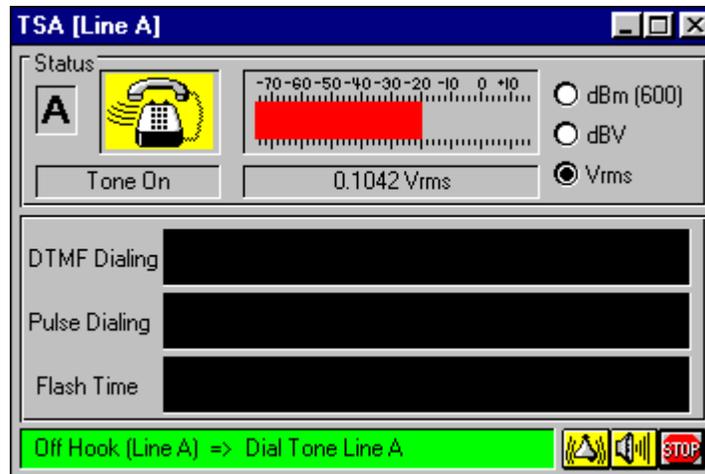
When enabled, Caller ID information is sent with any ringing pattern. Depending on the Caller ID options, this can change the actions of the TSPC before and during the ringing. Enabling or disabling the Caller ID feature is done by selecting the [ACTIONS] [SEND CALLER ID INFORMATION] menu command, or by pressing F5. Each time the command is selected, the Caller ID feature will be turned on or off. The default state is on. Depending on the signaling method selected in the Caller ID settings window, the TSPC may perform a line reversal or generate a short ringing burst before ringing the selected pattern. These extra

actions are required as part of signaling system for some Caller ID standards. If this interferes with the desired testing, disable the Caller ID feature.

Type II Caller ID information can also be sent when the telephone is off hook. Unlike the ringing, each network tone defined can enable or disable the Type II Caller ID information. However, network tones that do not have a pre-defined ending time cannot be linked to a Caller ID transmission. For example, dial tone, ring back, and busy tones do not have a pre-defined ending time. They stop when the user dials a digit, flashes the line, or hangs up. As such, they cannot be linked to a type II Caller ID transmission. Network tones that have a pre-defined ending time, like the Call Waiting tone, can be linked to type II Caller ID. Once the network tone ends, the type II Caller ID transmission will begin. Whether or not Caller ID is sent with a network tone is part of the network tone definition and can be viewed or edited by selecting the [SETTINGS] [NETWORK TONES] menu command.

Events & Actions Status Line

At the bottom of the main panel and to the left of the three buttons is the events and actions status line. Any events detected that are linked to a corresponding action are shown in the status line, as well as what ringing pattern or network tone is active. The default setting for the TSA3300 program has the off-hook event linked to an action that generates dial tone (for line A only). As such, when the telephone under test goes off-hook, the status line shows the event and the corresponding action name. The figure below shows an example of this.



The background color for the events are always in green. In this case, the "Off-Hook" event for line A was detected. This caused the execution of the action script called "Dial Tone Line A". Event messages are only shown for a brief time interval. Once this time interval expires, the status line will either contain nothing, or indicate if ringing or a network tone is active. In this case, the dial tone is activated by the action script, which is shown in the light blue background color.

Network Tone: Dial Tone

Any time a network tone is active, the name of the tone is displayed in the status line with a light blue background. Likewise, if the ringing generator is active, the name of the ringing pattern is displayed; however, the background color will be yellow instead of light blue.



Ringing: Single Ring

■ Section 4

Program Settings

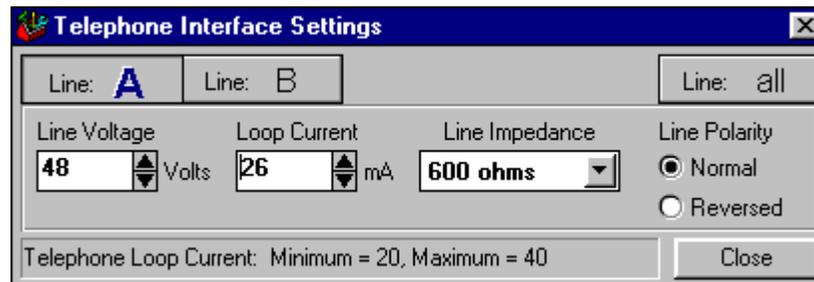
This section describes in more detail how to view and modify the various program settings. Major topics covered include the following:

- 1) Working with Multiple Telephone Line Settings
- 2) Setting up the Telephone Interface
- 3) Setting up the Ringing Generator
- 4) Setting up the DTMF Parameters
- 5) Setting up the Pulse Dialing & Flash Timing
- 6) Setting up the Caller ID Parameters
- 7) Setting up the Network Tone Definitions
- 8) Defining Events and Actions
- 9) Controlling the BNC Signal Routing

■ Section 4-1 Working with Multiple Lines

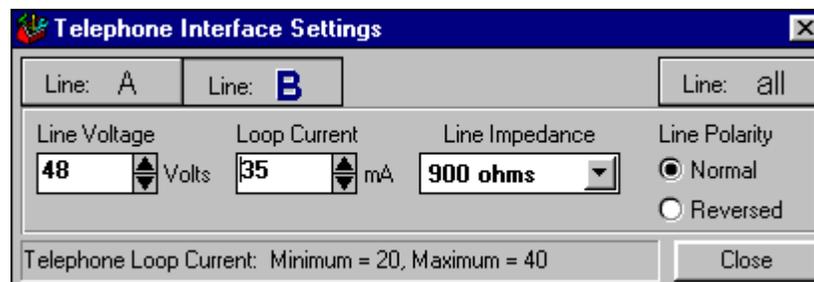
The TSA3300 program can support up to four lines simultaneously while allowing for a wide range of programmability among the various parameter settings. All of the programmable settings are broken down into eight groups, each with a separate menu item under the [SETTINGS] heading. These include controlling the telephone interface, ringing generator, DTMF parameters, pulse & flash timing, network tone definitions, and others. Toolbar buttons provide a shortcut to access the eight different parameter groups.

For example, selecting the [SETTINGS] [TELEPHONE INTERFACE] menu command displays a window similar to the following.

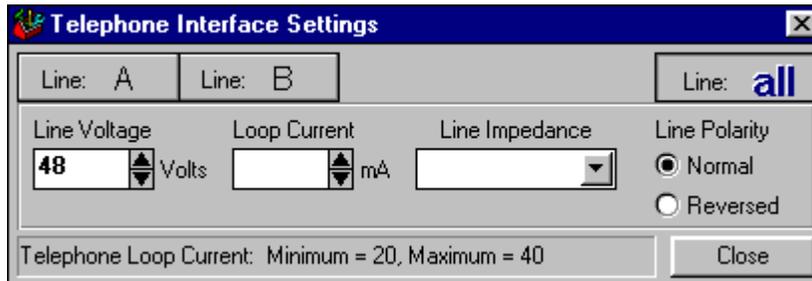


In this example the TSA3300 program is controlling two TSPC's designated line A and line B. The top section of the settings window contains a row of tabs representing each of the active telephone lines. With two lines active, only tabs A and B are visible. If four lines were present in the system, tabs A, B, C, and D would all be visible. The tabs select which telephone line settings is being viewed and can be edited. Therefore, pressing the mouse on tab A shows the parameters for line A. Tab B represents line B, and so on. If two or more lines are active, as in this example, then a fifth tab label ALL is also visible. The ALL selection is used for changing all of the lines present in the system to the same value.

In this example, the B tab is now pressed to display the settings for line B. The loop current value for line B is changed from the default setting of 26 mA to 35 mA, and the line impedance is changed from the default setting of 600 ohms to 900 ohms. The new settings for line B are shown below.



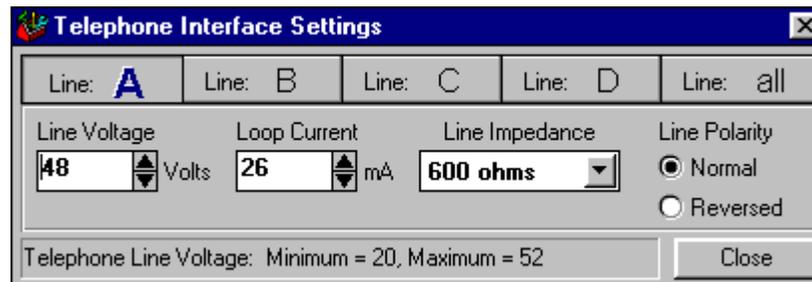
If the A tab is pressed, the settings for line A will be displayed, which are different from line B in terms of the loop current and line impedance. By selecting the tab ALL, the parameter settings for all the lines can be changed to the same value simultaneously. However, any settings that are different between the various lines will be "blanked out". As shown below, when pressing the ALL tab, no values are displayed for the loop current and the line impedance. This is because those settings are different between lines A and B. Where the settings among the lines are the same, the parameter value is shown. Thus the ALL tab provides a quick way to see if any parameter settings are different between the various telephone lines. If any value is changed while the ALL tab is selected, then all the telephone lines will be set to the new parameter value.



Note: Short cut keys ALT-A, ALT-B, ALT-C, and ALT-D can be used instead of the mouse for selecting tabs A to D respectively. The ALL tab can be selected by pressing the ALT-X key. Also, the program will remember which tab was active when the settings window is closed. As such, when opening the settings window at a later time, the last used tab will be active again.

■ Section 4-2 Telephone Interface Settings

To change the telephone interface settings for any of the lines, select the [SETTINGS] [TELEPHONE INTERFACE] menu command, or click the mouse on the equivalent tool bar button. In either case, a window similar to the following is displayed.



The tabs at the top of the window control which telephone line's parameter is being viewed or edited. In systems with fewer than four cards, the extra tabs will be removed. The tab labeled ALL can be used to change the settings of all the telephone lines simultaneously.

To change the telephone line voltage, simply type a new voltage level in the appropriate text box and then press the ENTER key. Changes to the voltage will be made immediately. As an alternative, the up/down spin buttons can be used to raise or lower the line voltage in 1 volt increments. In either case, the voltage level will be clamped to its minimum and maximum limits displayed at the bottom of the window.

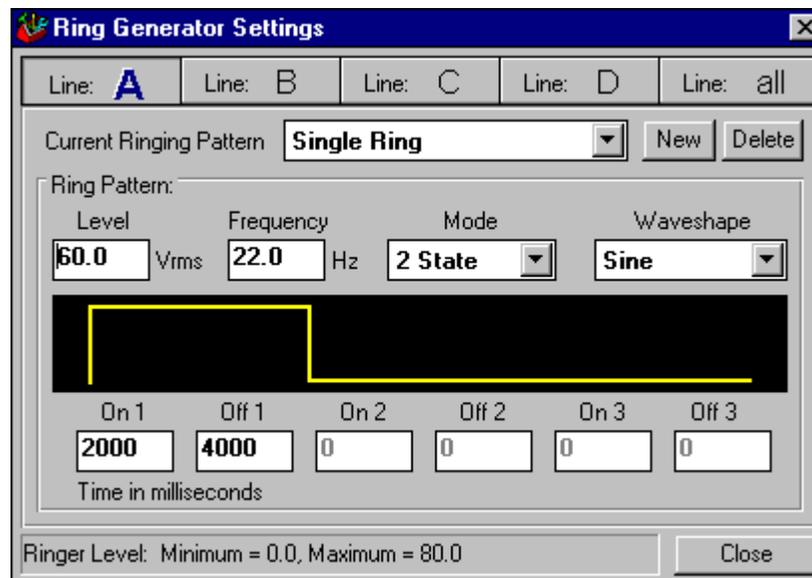
The telephone loop current setting works in the same manner as the telephone line voltage setting. Simply enter a new value and press ENTER, or use the up/down spin buttons. As with the line voltage, the minimum and maximum permissible loop currents are displayed at the bottom of the window.

The drop-down list box to the right of the loop current setting contains the selected telephone line impedance. The line impedance can be either 600 ohms, 900 ohms, or a complex impedance. The standard setting for the complex impedance is the German ZR; however, other impedances can be factory ordered, or changed by the user to provide custom impedances. To change the line impedance, simply select a new value from the drop-down list box. The changes are implemented immediately.

The last parameter controls the polarity of the telephone line. This has two settings; normal and reversed. By clicking on the desired setting, the line polarity will change to the new selection.

■ Section 4-3 Ring Generator Settings

The ringing generator can be programmed to produce various ringing patterns. The ringing pattern definitions control the frequency, level, timing, and waveshape parameters of the ring generator. To view, edit, or create new ringing patterns, select the [SETTINGS] [RING GENERATOR SETUP] menu command, or press the equivalent tool bar button. In either case, a window similar to the following is displayed.



Each telephone line in use is associated with a default ringing pattern. This is the ringing pattern generated when the yellow "ring" button is pressed on the telephone line's main panel window, or the [ACTIONS] [RING SINGLE/ALL LINES] menu items are selected. The default ringing pattern can be changed for any active telephone line by selecting the appropriate line designation tab at the top of the ring generator settings window and then choosing the desired ringing pattern from the drop-down list. If more than one telephone line is present, selecting the tab ALL can be used to set the default ringing pattern for all lines to the same ringing pattern.

The program defaults include the definitions for five different ringing patterns. These are termed Single Ring, Double Ring, Triple Ring A, Triple Ring B, and No Ringing. By changing the selection of the ringing pattern drop-down list, the definition parameters for that ringing pattern are displayed.

To change the frequency and level of the ringing, simply enter a new frequency or level value in the appropriate field and press the ENTER key. Any new values entered will be clamped to their minimum or maximum limits as shown in the panel at the bottom of the window. The ringing mode drop-down list box controls how many on/off intervals are used to define the pattern. This can be either 2, 4, or 6. Once the number of on/off intervals are set, the time intervals

for each on and off period can be changed in the fields near the bottom of the window. Changing the time intervals will re-draw the diagram outlining the ringing generator envelope. This gives a visual representation of the defined ringing pattern. The waveshape used to create the ringing can be selected from one of three possibilities. These being either Sine, Triangle, or Square. Note that limitations exist in the accuracy of very low frequency square wave ringing. In such a case, a significant amount of droop can be observed in the waveform due to a limited low frequency response of the ringing generator.

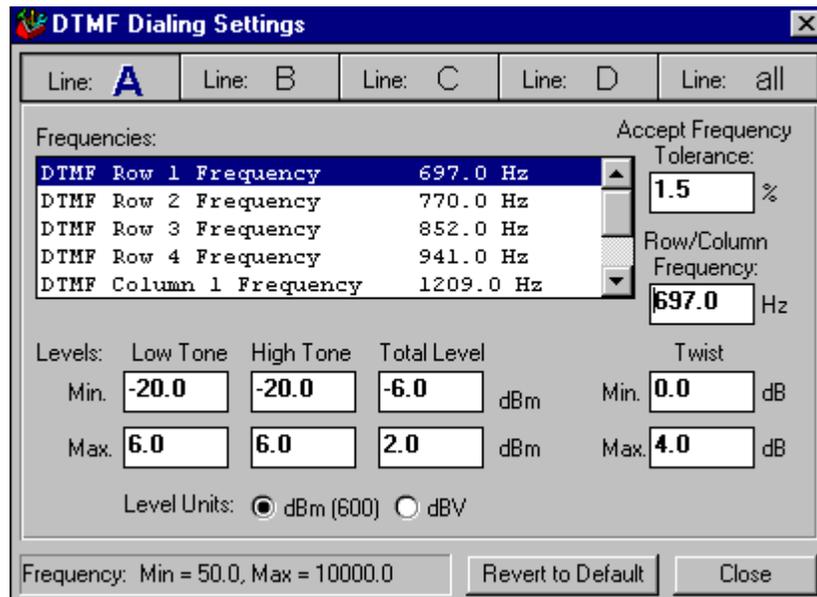
In addition to altering the default ringing patterns, it is possible to create new ringing pattern definitions. Pressing the "New" button next to the ringing pattern selection list, displays a small window requesting the name for the new pattern.



By entering a unique name and pressing the OK button, a new pattern is created. Up to 128 different ringing patterns can be defined. As with the default patterns, the new pattern can be edited in order to change its ringing frequency, level, timing, or waveshape. Patterns can also be deleted by selecting a ring pattern from the drop-down list box and clicking the DELETE button.

■ Section 4-4 DTMF Parameter Settings

The limits by which the DTMF digit analyzer classifies valid or invalid digits can be viewed or edited by selecting the [SETTINGS] [DTMF DIALING SETUP] menu command, or by clicking the mouse on the equivalent tool bar button. In either case, a window similar to the following is displayed.



As with all the other parameter groups, the limits for the DTMF digit analyzer can be altered on a line by line basis. Thus the limits for all the telephone lines in use need not be the same. The line designation tabs at the top of the window determine for which telephone line the DTMF limits are shown. In systems with more than one telephone line, the tab marked ALL can be used to change the settings for all active telephone lines together.

The limits used to qualify a DTMF digit center on the digits low and high group tone frequencies and levels. The center frequency for each of the four row and column frequencies can be specified along with an acceptable tolerance around each center frequency. Also, minimum and maximum level limits can be specified for the high and low group tone levels independently, in addition to a total signal level and twist limits. The twist level is defined as the level ratio between the low group tone and the high group tone. Positive twist occurs when the level of the high group tone is greater than the low group tone. Likewise, negative twist exists if the high group tone level is less than the low group tone.

To change any one of the DTMF row or column frequencies, select the desired frequency from the list box near the top of the window. This list box contains the current defined frequencies for the four row tones and four column tones. The selected frequency will appear in the text box just to the right of the list of frequencies. Simply type a new frequency value and press the ENTER key.

This will update the selected row or column frequency with the new value. Any value entered must fall between the minimum and maximum limits shown at the bottom of the window.

The frequency tolerance parameter, shown in the lower right corner, defines the range of valid frequencies for each of the row or column frequencies in the list above. This value can range from 0.1% to 3.0%, with a nominal setting of 1.5%. For example, a 1.5% tolerance value with a Row #1 value of 697 Hz means that any low group tone detected in the range of 697 Hz - 1.5% (686.5 Hz) to 697 Hz + 1.5% (707.4 Hz) will be accepted as a valid Row #1 frequency.

The limits for the low/high group, and total levels can be specified in units of either dBm (600 ohm) or dBV. In either case, to change the minimum or maximum acceptable value, simply enter a new level and press ENTER. Note that a warning message is displayed if the minimum tone level is set to a value higher than the maximum tone level, since this results in a situation where no tone level would be accepted.

All of the DTMF parameters can be returned to their default settings by pressing the "Revert to Default" button at anytime .

■ Section 4-5 Pulse Dialing & Flash Settings

The pulse dialing timing and flash timing minimum and maximum limits can be adjusted by selecting the [SETTINGS] [PULSE DIALING & FLASH] menu command, or by clicking the mouse on the equivalent tool bar button. In either case, a window similar to the following is displayed.

Line:	A	B	C	D	all
PPS Limits					
Min.	9.0				
Max.	11.0				
Make/Break Ratio Limits					
Min.		50.0			
Max.		70.0			
Flash Time Limits (msec)					
Min.			500.0		
Max.			750.0		
Call Waiting Deluxe (time to DTMF)					
Min.				400.0	
Max.				600.0	
Minimum Interdigit Delay	600.0				
Pulses Per Second: Min = 6.0, Max = 25.0					
Revert to Default					
Close					

Each telephone line in use, designated by the letters A to D, can operate with pulse dialing and flash timing limits independent of each other. Selecting one of the tabs A to D (if visible) will display the current limit settings for that line. Selecting the tab marked ALL will show the common limits between the various lines and allow changing of the limits for all lines to the save parameter value. Note that the ALL tab is only present in systems with more than one telephone line active.

For the pulse dialing digits, the limits are specified by the minimum and maximum pulses-per-second (PPS), minimum and maximum make-break ratio (MBR), and the minimum inter-digit delay. Changes are made by simply typing a new value over the previous one and pressing the ENTER key. The range of acceptable values, for all parameters, are displayed near the bottom portion of the window.

A valid telephone line flash is specified by its minimum and maximum duration. Typing a new value and pressing the ENTER key will change the flash timing limits.

The two settings at the far right of the window are used to define a time window for Call Waiting Deluxe (CWD) service detection. For a successful CWD event, the time between the end of the flash and the following DTMF tone must fall within the specified minimum and maximum limits.

If the PPS, MBR, Flash, or CWD timing minimum limit value setting exceeds the maximum value setting, then a warning message will be displayed when the CLOSE button is pressed. The limit value must be changed to a parameter value that will be accepted by the program.

All of the pulse dialing and flash timing parameters can be returned to their default settings by pressing the "Revert to Default" button at anytime.

■ Section 4-6

Caller ID Settings

Caller ID information may be sent with any ringing pattern generated or with a network tone. The Caller ID settings are viewed and adjusted by selecting the [SETTINGS] [CALLER ID SETUP] menu command, or by clicking the mouse on the equivalent tool bar button. In either case, a window similar to the following is displayed

As with all of the other parameter settings, the Caller ID setup may be different for each of the active telephone lines. The tabs with the line designators A to D (if visible) control for which telephone line the Caller ID parameters are being displayed. Selecting the tab marked ALL will show all the common parameter settings between the various lines and allow changing of the settings for all lines to the same parameter value. Note that the ALL tab is only displayed in systems with more than one active telephone line .

Message Types

Six different message types can be specified for the Caller ID transmission. They are:

- i) Single Data Message Format
- ii) Multiple Data Message Format
- iii) Single Message Waiting Format - Activate
- iv) Single Message Waiting Format - Deactivate
- v) Multiple Message Waiting Format - Activate
- vi) Multiple Message Waiting Format - Deactivate

The message types are selected from the drop-down list as shown in the following figure.

Message Type: 2) Multiple Message

Date & Time: 2) Multiple Message

Calling Number: 3) Single Waiting Activate

Calling Name: 4) Single Waiting Deactivate

Calling Name: 5) Multiple Waiting Activate

Calling Name: 6) Multiple Waiting Deactivate

Calling Name: jonn simon

Signaling Type: (on hook CID only) 1) Ring > Data > Ring...

Signal Level: -13.0 dBm Modem: Bell 202 V.23

The Single Data Message Format selection will only transmit the date/time and calling number (or number private/out-of-area) information. This format does not have the capability to send the calling name information. The Multiple Data Message Format selection will send the data/time, calling number (or number private/out-of-area), and calling name (or name private/out-of-area) information. The calling number and calling name fields may each contain up to 20 characters. Number absence or name absence information may be sent in lieu of the number or name by clicking the mouse on the Private or Out-of-Area buttons. If either the Private or Out-of-Area options are enabled they will be shown with a green check mark. This will disable the calling number or name fields, as that information is not sent. The date and time information always reflects the current system time of the PC.

The last four message types deal with sending visual message waiting information. In these cases, the calling number or calling name fields have no bearing on the information sent. The data sent with the message waiting messages is fixed as follows:

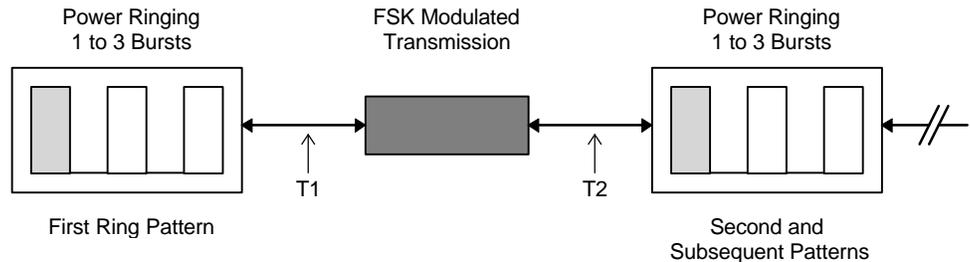
- i) Single Message Waiting Format - Activate
Bytes sent: 0x06 0x03 0x42 0x42 0x42 0x31
- ii) Single Message Waiting Format - Deactivate
Bytes sent: 0x06 0x03 0x6F 0x6F 0x6F 0xAA
- iii) Multiple Message Waiting Format - Activate
Bytes sent: 0x82 0x03 0x0B 0x01 0xFF 0x70
- iv) Multiple Message Waiting Format - Deactivate
Bytes sent: 0x82 0x03 0x0B 0x01 0x00 0x6F

The Caller ID data sent to the telephone under test can be modulated in either a Bell 202 or V.23 format. In both cases the baud rate will be 1200 bps with a single start bit, eight data bits, and a single stop bit per byte. However the mark and space tone frequencies will be different, as for Bell 202 they are 1200 Hz and 2200 Hz ; while for V.23 they are 1100 Hz and 2300 Hz respectively. The signal level for the FSK modulated data is specified in dBm (600 ohms) and can range from a maximum of +6 dBm to a minimum of -70 dBm.

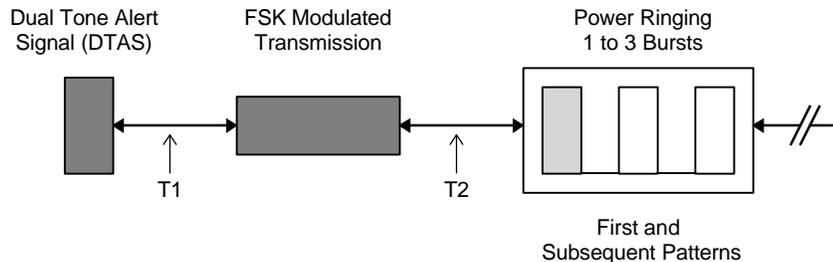
Signaling Types

Apart from the Caller ID data sent and the characteristics of the FSK modulator, five different signaling methods can be used to alert the telephone under test. The different signaling methods support many of the Caller ID standards employed in various countries. Note that this applies only to type I (on-hook) Caller ID.

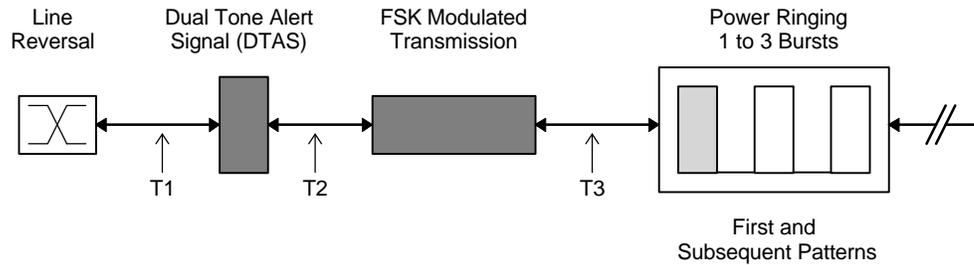
- Signaling: Ring > Data > Ring**
 This is the default signaling method. In this case, the Caller ID FSK data is sent between the first and second rings. This is the last off time defined for the ringing pattern selected. For example, if using the default Double Ring pattern, the FSK data is sent during the second off interval, which is 3000 msec in duration. The FSK signal will begin 250 msec (T1) after the end of the second ring on time. The time T2 is variable and depends on the duration of the FSK signal. It will be adjusted such that the total time of T1, T2, and the FSK transmit time equals that of the ringing off interval.



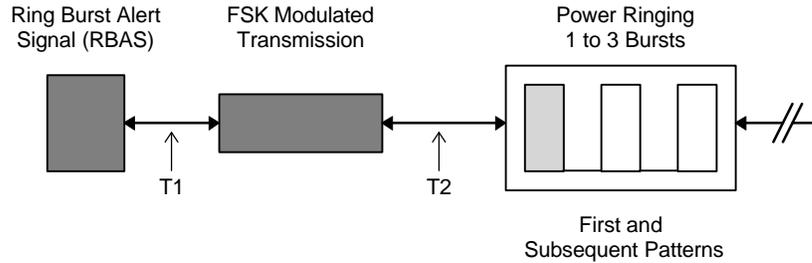
- Signaling: DTAS > Data > Ring**
 For this signaling type, a short Dual Tone Alert Signal (DTAS) is sent prior to the FSK data. The DTAS tone consists of two frequencies of 2130 Hz and 2750 Hz at an amplitude of -20 dBV for a duration of 100 msec. After the DTAS tone, a delay of 250 msec (T1) is added before the FSK data is sent. Following the FSK data, another delay of 350 msec (T2) is added before the ringing pattern starts.



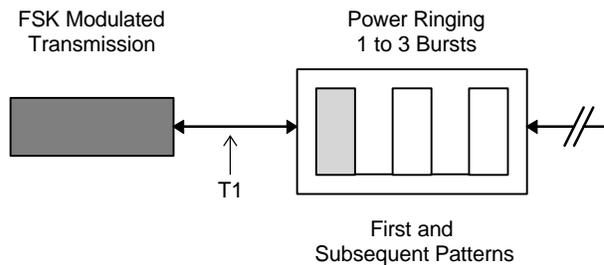
- Signaling: Line Reversal > DTAS > Data > Ring**
 The third signaling type is a slight variation from the previous case in which a line reversal now precedes the DTAS tone. The line polarity reversal occurs 200 msec (T1) before the DTAS tone. The remainder of the timing is identical to that of the previous signaling method.



4. Signaling: Ring Pulse > Data > Ring
 For this signaling type, a short ringing pulse is generated prior to sending the FSK data. The ringing pulse will have an amplitude of 60 Vrms at a frequency of 22 Hz with a duration of 250 msec. The delay between the ringing pulse and the start of the FSK data is 650 msec (T1). Finally, the selected ringing pattern follows the FSK data transmission with an inserted delay of 350 msec (T2).

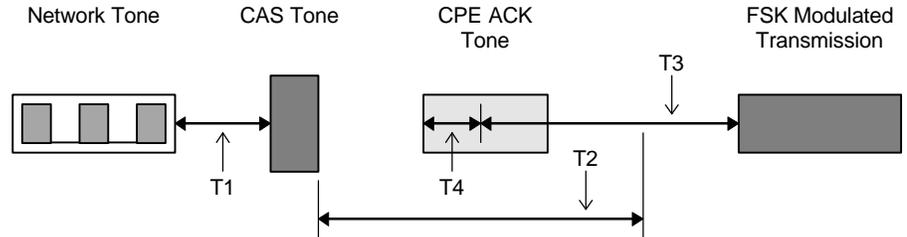


5. Signaling: Data > Ring
 The last signaling type simply sends the FSK prior to ringing the telephone under test. The delay between the end of the FSK data and the start of the ringing is 350 msec (T1).



6. Type II Caller ID Signaling:
 Type II (off-hook) Caller ID is always sent using the same signaling method. If a network tone is associated with a Type II Caller ID signal, once the network tone ends, then a CPE Alerting Signal (CAS) will be sent to the telephone under test. The CAS consists of two tones at frequencies of 2130 Hz and 2750 Hz at a level of -20.0 dBV for a duration of 80 msec. The delay between the network tone and the CAS tone is 25 msec (T1). After the CAS tone has been sent to the telephone under test, it must respond with an ACK tone (DTMF A or D) within 160 msec (T2), or the FSK data will not be sent. If the response

from the CPE is detected within the time-out interval, a delay of 200 msec (T3) is inserted before the FSK data is sent. The delay T4 represents the processing time to properly detect the ACK tone, which can range from 10 to 40 msec.



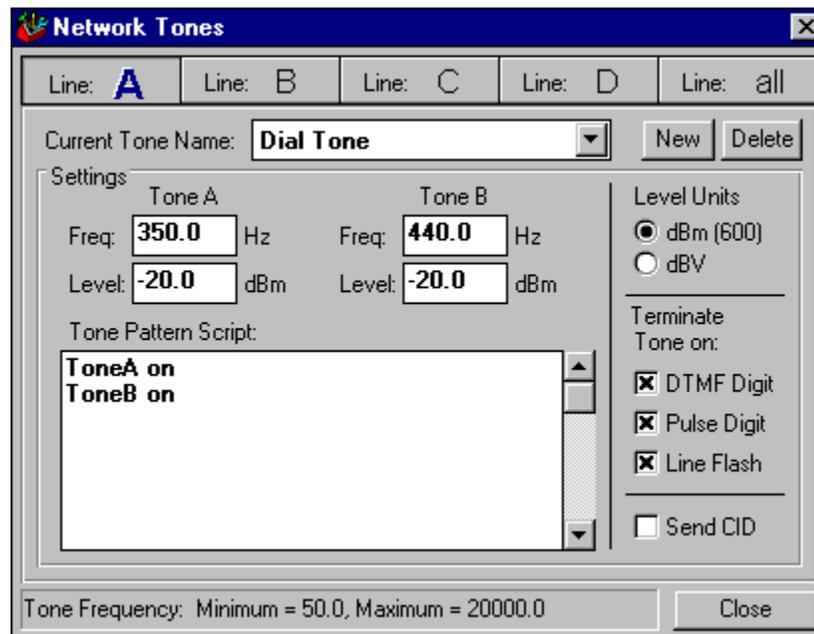
The Caller ID feature can be globally enabled or disabled by selecting the [ACTIONS] [SEND CALLER ID INFORMATION] menu command or by pressing the F5 command. If the check mark is present on the menu, the Caller ID data will be sent with any ringing signal and may be sent in conjunction with a network tone. If the check mark is not present on the menu, then no Caller ID information will be sent under any conditions.



All of the Caller ID parameters can be returned to their default settings by pressing the "Revert to Default" button at anytime.

■ Section 4-7 Network Tone Settings

Network tones are signals sent to the telephone under test in order to communicate various conditions for the call in progress. This can include dial tone, busy tone, ringback tone, call waiting tone, and others. Many different characteristics of network tones can be collected together into a complete definition of that tone. This includes the signal frequencies, level, timing or cadence, and reaction to various events. To view, edit, or create network tone definitions, select the [SETTINGS] [NETWORK TONES] menu command, or click the mouse on the equivalent tool bar button. In either case, a window similar to the following is displayed.



Similar in nature to the ringing patterns, each telephone line in use is associated with a default network tone definition. This is the network tone generated when the yellow "tone" button is pressed on the telephone line's main panel window, or the [ACTIONS] [GENERATE NETWORK TONES] menu items are selected. The default network tone can be changed for any active telephone line by selecting the appropriate line designation tab at the top of the network tone settings window and then choosing the desired tone definition from the drop-down list. If more than one telephone line is present, selecting the tab ALL can be used to set the default network tone for all lines to the same definition name.

The program default settings include the definitions for seven different network tones. These are termed Dial Tone, Stutter Dial Tone, Recall Dial Tone, Ringback, Busy, Reorder, and Call Waiting. By changing the selection of the ringing pattern drop-down list, the definition parameters for that network tone are displayed.

A network tone can consist of up to two simultaneous tones, labeled A and B. The frequency for tones A and B can range from a minimum of 50 Hz to a maximum of 20,000 kHz. The signal level of the tones can be specified in units of dBm (600 ohms) or dBV by selecting the desired units to the right of the window. To change the frequency of either tone, simply type a new value over the existing value. Likewise for any changes in the tone levels.

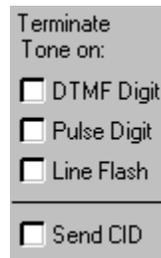
Tone Scripts

The timing or cadence of a network tone is defined by a simple script. By using a script, a wide range of cadences can be accommodated. The scripts are composed by using five commands. These are ToneA, ToneB, Loop, Loopend, Wait. The function of these commands are as follows:

- i) **TONEA (on/off)**
Following the ToneA command word must be either the word ON or OFF. If ON is specified, the tone generator for tone A will be enabled. Otherwise, specifying OFF will disable the tone generator for tone A.
- ii) **TONEB (on/off)**
This command has the same format as TONEA, except that it enables or disables the tone for generator B instead to A.
- iii) **LOOP [0 to 32000]**
The loop command is used to repeat various script commands a pre-defined number of times. The number of times a loop is to be repeated can range from 0 to 32000, or indefinitely. If no value is specified, then the loop will execute indefinitely, or until stopped by either an event or user intervention. Only one occurrence of the LOOP command is allowed per network tone script.
- iv) **LOOPEND**
This command simply marks the end of the loop. LOOPEND must appear after the LOOP command, and can only occur once in the network tone script.
- v) **WAIT [0 to 32000]**
The WAIT command is used to insert a delay in units of milliseconds before the next command is executed. The range of delays can be from 0 to 32000 msec.

Response to Events

Any network tone that is active, can be programmed to automatically terminate upon a specific event occurring. These events are the dialing of a DTMF digit, dialing of a pulse digit, or flashing the line. The following check boxes, when enabled, will stop the network tone if that event occurs.



Terminate
Tone on:

DTMF Digit

Pulse Digit

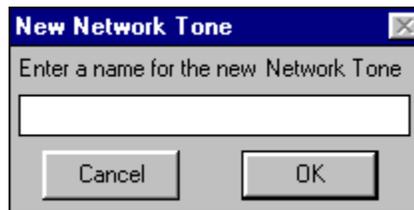
Line Flash

Send CID

This helps to mimic the actions of a central office switch. For example, the default dial tone definition is set to automatically stop upon detecting any digits being dialed.

Also, any network tone can be set to cause the generation of a type II Caller ID transmission, under certain conditions. By enabling the check box marked "Send CID", when the network tone stops, a type II CID transmission will be started. However this is only applicable if the network tone is "self-terminating". This means that the network tone must terminate by itself instead of relying on external events such as dialing, or a line flash, or pressing the STOP button on the main panel window. Tones such as the ring back and busy use the indefinite loop command which never terminates by itself. As such these tones can not send the Caller ID information, since they never self terminate.

In addition to altering the default network tone definitions, it is possible to create new tone definitions. Pressing the "New" button next to the network tone selection list, displays a small window requesting the name for the new definition.



New Network Tone

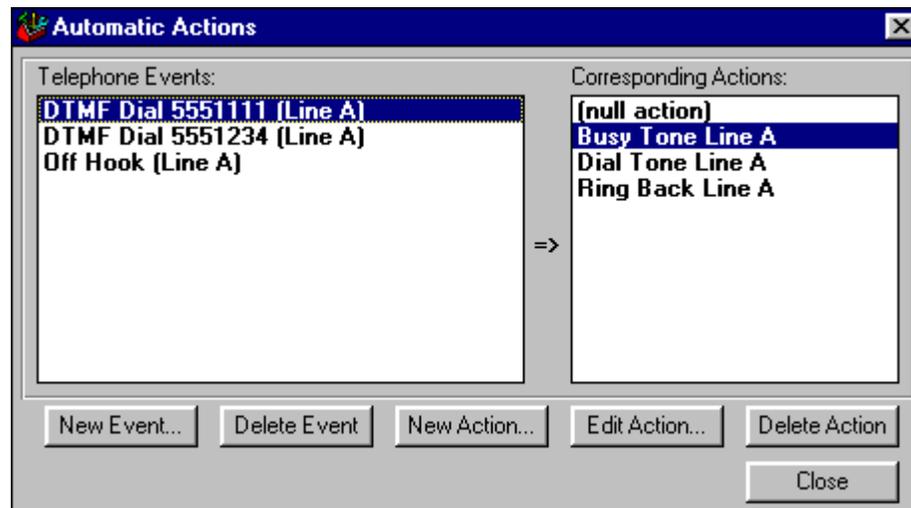
Enter a name for the new Network Tone

Cancel OK

By entering a unique name and pressing the OK button, a new pattern is created. Up to 64 different network tones can be defined. As with the default tones, the new network tone can be edited in order to change its characteristics, timing script, and reaction to various events. Network tone definitions can also be deleted by selecting a tone from the drop-down list box and clicking the DELETE button.

■ Section 4-8 Defining Events and Actions

The TSA3300 program constantly monitors for the occurrence of specific events. These events can be generated by the telephone under test, such as going on-hook, off-hook, dialing, flashing the line; or by user initiated events such as pressing a key on the keyboard or by a low to high transition at one of the auxiliary digital inputs. Any or all of these events can be linked to various actions. The actions are composed of a scripting language that can perform tasks such as changing the program settings, ringing the telephone under test, and generating network tones. To view or edit the default event/action relationships, or to create new ones, select the [SETTINGS] [AUTOMATIC ACTIONS] menu command, or click the mouse on the equivalent tool bar button. In either case, a window similar to the following is displayed.



The two lists contain all of the defined events and actions currently in use. The left hand list shows the various event names while the right hand list shows the action script names. The program default settings include three events. They are DTMF dialing the number 5551111 on line A, DTMF dialing the number 5551234 on line A, and going off-hook on line A. Clicking on any of the events with the mouse highlights the associated action name in the list to the right. As such, the defined events and the associated actions are:

- | | | | |
|------|----------------------------|-----|------------------|
| i) | DTMF Dial 5551111 (Line A) | ==> | Busy Tone Line A |
| ii) | DTMF Dial 5551234 (Line A) | ==> | Ring Back Line A |
| iii) | Off-Hook (Line A) | ==> | Dial Tone Line A |

The names of the action scripts are arbitrary, but it is helpful that they represent their underlying purpose. With this default setup, if the telephone under test were to DTMF dial the number 5551111, then the action script called "Busy Tone Line A" would be executed. This script will start the network tone called "Busy". Likewise, dialing the DTMF number 5551234 executes the action script called "Ring Back Line A", which starts the ring back network tone. Finally, going off-

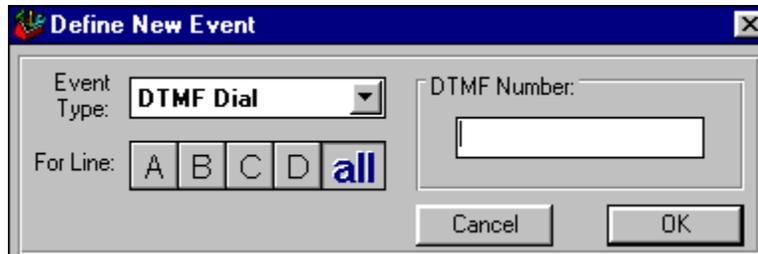
hook with line A executes the "Dial Tone Line A" action script. This script will start the generation of dial tone.

By using these events and actions, basic operation of a central office switch can be emulated. Along with generating tones and ringing, the action scripts can alter different program settings such as changing the Caller ID number or name information. Also messages can be displayed to the user along with controlling the auxiliary digital outputs and sending DTMF tones.

To change which action script is linked to a particular event, simply select the event from the left hand list. This highlights the associated action in the right hand list. To change the action, simple select a new action from the right hand list. If no action to the particular event is desired, then select the "(null action)" entry from the right hand list. The "(null action)" script can not be deleted and does nothing. Alternatively, the event can be deleted by pressing the Delete Event button.

Creating New Events

Adding a new event is accomplished by clicking the mouse on the New Event button. A new window appears in which the various events can be defined. There are twelve possible event types available to choose from. In conjunction with the different types, the events can also be associated with a specific telephone line. For example, an off-hook event can occur when only line A goes off-hook, but not any other lines such as B, C, or D (if present in the system). By selecting the ALL button, an event will be independent of which line generated it. In the previous example, the off-hook event could occur when any telephone line that was in the off-hook state, instead of just line A. The Key Press event is a special case, since regardless of how many telephone lines are present in the system, there is only one keyboard present.



The new event type is selected from the drop-down list. Below the event type selection, the associated telephone line can be chosen by clicking the mouse on the desired line designation. Press the ALL button to generate the event for all of the telephone lines present. If less than four lines are available, then some of the buttons will not be visible. At minimum though, the line A and ALL buttons will always be visible. Certain events, such as DTMF and Pulse dialing will require additional information. Like the specific dialing number used to trigger the event. This is entered in the text field on the right side of the window.

The 14 possible event types are as follows:

- 1) **Off Hook**
The off-hook event occurs when the telephone line transitions from the on-hook to the off-hook state (drawing loop current). However, brief off-

hook to on-hook to off-hook transitions that are part of pulse dialing or line flashes will be ignored. As such the telephone line state must first be on-hook for a duration exceeding the maximum flash time plus 500 msec. Only when this has occurred, will the off-hook event be issued when loop current starts to flow in the telephone line.

- 2) **On Hook**

The on-hook event occurs when the telephone line transitions from the off-hook to the on-hook state (drawing no loop current). However, in order prevent the on-hook event being issued due to pulse dialing or line flashes, the telephone line must be in the on-hook state for a duration greater than the maximum flash time plus 500 msec. Once the telephone line has been on hook for this duration, the event will be issued.
- 3) **Line Flash**

The line flash event is triggered upon detecting a valid telephone line flash. A valid flash is defined as when the on-hook interval meets the flash timing specification defined in the Pulse Dialing & Flash timing setup window. If the on-hook interval is less than the minimum limit, or greater than the maximum limit, then this event will not be triggered.
- 4) **Invalid Flash**

An invalid flash event is similar to the line flash event, except that the duration of the on-hook interval was outside the limits of a valid flash. Either the on-hook interval was less than the minimum allowable flash time, or greater than the maximum allowable flash time. Note that if the on-hook interval is less than 200 msec, it will be treated as an attempt at pulse dialing and not a line flash. Likewise, an on-hook interval that exceeds the maximum flash time by 500 msec will be treated as a separate on-hook event followed by an off-hook event.
- 5) **DTMF Dial**

An event can be issued upon DTMF dialing any number up to 16 digits in length. The number is defined as part of the event and can include the DTMF digits * and #. It is important to note that the number must be dialed following an off-hook or line flash event, and repeated dialing of the number will not issue any subsequent events. For example, a DTMF dialing event for the number "123" will only be triggered once if the telephone under test dials "123123". The first occurrence of "123" will cause the event, but any further occurrences will be ignored.
- 6) **Pulse Dial**

The pulse dial event is similar in nature to the DTMF dialing event, except that the event occurs for pulse dialed digits. Up to 16 digits can be specified and as with the DTMF dialing, the event is only issued on the first occurrence of the digit. Subsequent occurrences of the pulse dialing number will not trigger the event, unless an off-hook or line flash has occurred, which clears the dialing buffers.
- 7) **DTMF Digit Fail**

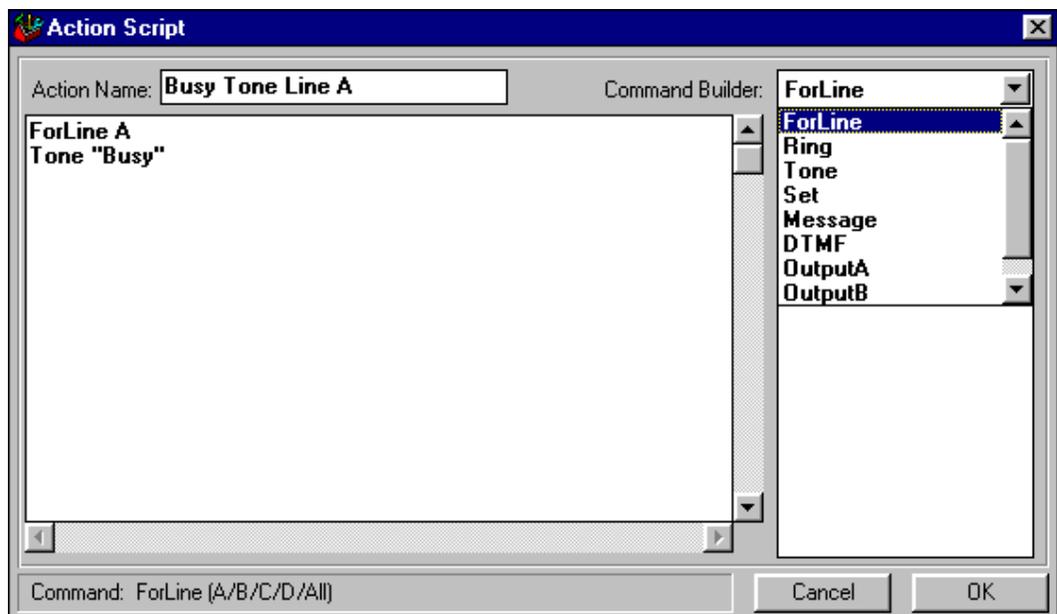
The DTMF digit fail event is issued whenever a DTMF digit has been detected, but is outside the DTMF dialing specifications for frequency or level. These specifications are defined in the DTMF Parameters settings window.

- 8) **Pulse Digit Fail**
Like the DTMF digit fail event, the pulse digit fail event occurs whenever a pulse dialing digit has been detected, but falls outside the specified limits of MBR, PPS, and interdigit delay. These limits are defined in the Pulse Dialing and Flash timing settings window.
- 9) **External Input A**
The rear DB9 pin connector contains two digital input ports. These are termed Input A and Input B. If the signal level at input A transitions from a low logic level to a high logic level, the External Input A event is issued. After the low to high transition, the logic level must remain high for a least 200 msec in order to properly detect the event and ignore any spurious signals.
- 10) **External Input B**
The External Input B event functions the same as that for External Input A, except that a low to high transition on the input B triggers the event. Like with input A, the logic level at input B must remain high for at least 200 msec after the transition, in order to properly detect the event and ignore any spurious signals.
- 11) **No CID ACK Tone**
As part of the normal handshaking process for type II (off-hook) Caller ID, the telephone under test must respond with a valid ACK tone before the FSK data is sent. The ACK signal must be either a DTMF A or D. If neither of these DTMF digits are received, this event will be triggered.
- 12) **Key Press**
The Key Press event can used to start action scripts when a specific key on the keyboard has been pressed. As part of the event definition, the target key is selected from a list of valid choices. This includes the F9 to F12 keys, 0 to 9 keys, A to Z keys, and the space bar.
- 13) **CWD (Call Waiting Deluxe)**
This event occurs whenever a valid Call Waiting Deluxe service request is detected. This consists of the telephone flashing the hook switch and then generating a DTMF digit within a specified time window. If the flash time, DTMF digit, and CWD time interval meet the defined limits, a CWD event is issued. CWD events are associated with a specific service type, such as:
- Conference, Drop Last, Drop First, Hold,
Drop, Announce, Forward, and (undefined)
- As such, a separate event can be created for each type of CWD service.
- 14) **Invalid CWD (Call Waiting Deluxe)**
The Invalid CWD event is not associated with a specific CWD service, and is triggered anytime a CWD service request has been detected, but the timing interval between the hook switch flash and DTMF digit is outside the specified limits. Note that in order for this event to be detected, both the flashing timing and DTMF digit characteristics must meet their respective limits. It is only the CWD timing interval that fell outside the defined limits.

Since only one event can be issued at a given time, conflicting events will be trapped and an error message will be displayed. Conditions of conflicts include specifying two or more of the same events, but with one of the events acting on ALL the telephone lines. For example, an off-hook event for line A would conflict with another off-hook event for all lines. If line A went off-hook, it would be undetermined which off-hook event should be issued. Also, the dialing number specified with the DTMF dialing or pulse dialing events must be unique. The same number can not be used twice.

Editing and Creating Action Scripts

Once an event has been created, it must be linked to a script before any actions can occur. The action scripts can be edited and created by pressing either the New Action or Edit Action buttons on the Automatic Actions setup window. As an example, select the DTMF Dial 5551111 (Line A) event with the mouse. This will highlight the Busy Tone Line A action. With the action script highlighted, press the Edit Action button. The contents of the script will be displayed as shown below.



This is the script editor window that is used to create new scripts or edit existing ones. At the top of the window is the name of the script. This can be changed anytime by simply typing a new name over the previous one. Below the name is the content of the script. It is made up of a series of commands that perform various functions. This particular script is only composed of two lines. The first, 'ForLine A' signifies that any following commands will effect only telephone line A. The second line, 'Tone "Busy"' will command the system to generate the network tone called "Busy". Thus when the associated event is triggered, these two lines will start the busy tone for line A.

To assist in creating the action scripts, the right side of the window contains a drop-down list box with all of the available commands. Below this, the list box will display any possible keywords associated with the selected command. Finally, the panel at the bottom of the window will provide information on the structure of the command selected in the drop-down list box. For example,

select the TONE command. The panel at the bottom of the window shows the structure of the command, which is simply the command word TONE followed by the network tone name in quotation marks. Also, the list box below the command has filled up with all the network tones defined. Double click the mouse on the "Ringback" tone. This will automatically transfer the TONE command and the selected tone to the scripting text area. Doubling clicking on any of the selections in the list box will copy the command and selection to the script text area, at the current cursor position. All of the commands function in a similar manner. By double clicking the mouse on the desire keyword, the command is transferred to the text area automatically.

Action Script Commands

1. **FORLINE** (A, B, C, D, ALL)
The FORLINE command is used to specify which telephone line the other commands refer to. This can be either "A", "B", "C", "D", or "ALL". The selected line is in effect until another FORLINE command occurs. If the telephone line has not been specified before using the other commands, and error message will be displayed when the action script is executed. The error will halt any further script execution. The only commands that are not line specific are "MESSAGE" and "CALLERID". It is a good practice to begin every action script with the FORLINE command.

2. **RING** "ringing pattern name" [Hold]
The RING command will start the ringing generator with the specified pattern name, provided the telephone is on-hook. The FORLINE command must have been used before this command in order to specify which line should be rung. If the telephone on the designated line is not on-hook, the ringing will be held until the on-hook state is reached. Once the telephone goes on-hook, the ringing will then start.

Optionally, the keyword "HOLD" can follow the ringing pattern name. With the HOLD keyword, the ringing pattern is prepared for execution, but not started. It is held until the RELEASE command is processed. This is normally done when precise timing is required between multiple telephone lines. The RELEASE command will start all network tones, ringing, or OSI's at the same time, assuming the DELAY parameter is set to zero. A non-zero DELAY parameter value will hold back that line until the delay time expires. See section 6, Synchronizing Multiple Telephone Lines for more information on the HOLD keyword.

3. **TONE** "network tone name" [Hold]
The TONE command will start generating the specified network tone name, provided the telephone is off-hook. If the telephone line is in the on-hook state when this command is executed, then no action will occur. The FORLINE command must have been used before the TONE command in order to specify which line will generate the tone.

As with the RING command, an option HOLD keyword can be added after the network tone name. It performs a similar function in suspending the generation of the network tone until the RELEASE

command is encountered. See section 6, Synchronizing Multiple Telephone Lines for more information on the HOLD keyword.

4. **SET** (parameter name) = (value)
The SET command is used to change the value of any of the program settings. The syntax for the command requires a parameter name, equal sign, and new value for the parameter, all following the SET command word. To view the entire list of parameter names, select the SET command from the drop-down list in the script editor. The list box below will be filled with the various parameter names. When selecting any of the parameters, its full name will be displayed on the panel near the bottom of the window. The parameters may require numeric or text values to be specified after the equal sign. Text values should be enclosed in quotation marks.
5. **MESSAGE** "text string" [parameter name , line]
The MESSAGE command will display text string(s) on the screen for a few seconds and then disappear. This command can be used to send comments to the user. Any messages that are displayed on the screen can be immediately cleared by clicking the mouse on the close button in the upper right hand corner of the message window.



In addition to text strings, the value of any parameter can be displayed. This is done by enclosing the parameter name and telephone line in [] brackets. For example, the command:

Message "The Caller ID name on line B is: " [CID_NAME,B]

Will display in a window (similar to above):

The Caller ID name on line B is: John Smith

6. **DTMF** "digits to dial" [level [on time [off time]]]
The DTMF command will generate a string of DTMF digits on the telephone line(s) specified by the FORLINE command. Up to 30 digits can be sent along with specifying the tone level, on time, and off time for the digits. A string of digits to send must be enclosed in quotation marks and follow the DTMF command. Optionally, the level can be specified in units of Vrms (0 to 4.0 Vrms) accurate for an unterminated telephone line. If the level is specified, then the tone on-time can be specified in units of milliseconds in the range of 1 to 20000. Likewise, if the on-time is specified, then the off-time can be specified also in units of milliseconds from 1 to 20000. Note that the DTMF command will have no effect if any ringing, network, or OSI action tone is currently active.
7. **OUTPUTA** (on, off)

The OUTPUTA command can be used to set the auxiliary digital output A to a high or low logic level. If the keyword ON is specified, the output will be set to a high logic level. The keyword OFF will set the output to a low logic level. See the section: Auxiliary Digital Input/Output for more information on the digital I/O port.

8. **OUTPUTB** (on, off)

Like the OUTPUTA command, the OUTPUTB command can be used to set the auxiliary digital output B to a high or low logic level. If the keyword ON is specified, the output will be set to a high logic level. The keyword OFF will set the output to a low logic level. See the section: Auxiliary Digital Input/Output for more information on the digital I/O port.

9. **CALLERID** (on, off)

The CALLERID command turns on or off the Caller ID feature for all of the telephone lines. If turned off, no Caller ID information will be sent with the ringing, nor will any type II Caller ID transmissions occur with the network tones. This command has the same effect as the menu command [ACTIONS] [SEND CALLER ID INFORMATION], which can be used to enable or disable the Caller ID feature.

10. **OSI** (duration) [Hold]

The OSI command will generate an open switching interval (OSI) with the specified duration. During an OSI, the DC feeding voltage is removed from the telephone line. The duration of the OSI must range between 1 and 10,000 msec. As with the TONE and RING commands, the OSI can be held back with the HOLD keyword. Once a RELEASE command is processed, the OSI will start. Note that the DELAY parameter affects the start of the OSI as well as the start of any network tones and ringing.

Note: TSPC hardware versions 2.2b and earlier do not support the generation of an OSI. If the TSA3300 program detects the older hardware versions, then a warning message will be displayed.

11. **RELEASE**

This command starts the execution of any held network tones, ringing, or OSI. Before starting the programmed actions, if the DELAY parameter for any line is greater than zero, then a delay is inserted.

12. **WAIT** (duration)

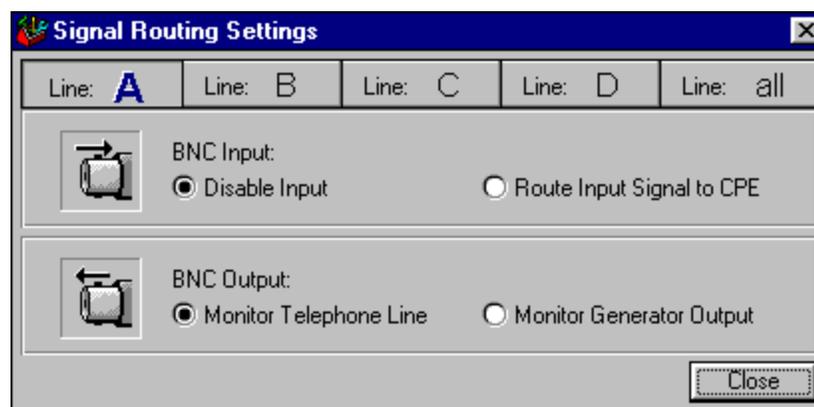
The WAIT command will suspend the script program execution by the time value specified. The duration value must be in the range of 1 to 1000 milliseconds. While the delay is in effect, no other commands, actions, or event will be processed.

Note that the resolution of the WAIT command is roughly 50 msec.

■ Section 4-9

BNC Signal Routing

The BNC connectors at the rear of the TSPC can be configured to monitor or inject signals to or from the telephone line. These settings are displayed by selecting the [SETTINGS] [SIGNAL ROUTING] menu command, or by clicking the mouse on the equivalent tool bar button. In either case, a window similar to the following is displayed.



Each telephone line in use, designated by the letters A to D, can operate with settings independent of each other. Selecting one of the tabs A to D (if visible) will display the current settings for that line. Selecting the tab marked ALL will show all the common settings between all of the telephone lines and allow changing of the settings for all lines simultaneously. Note that the ALL tab is only present in systems with more than one telephone line active.

The BNC input connector can be used to inject any arbitrary signal onto the telephone line. This is accomplished by selecting the "Route Input Signal to CPE" option. If selected, any signals present at the input connector will be sent to the telephone under test in both the on-hook or off-hook state. In this configuration, internal tone generation is disconnected. As such, any active network tones will not be sent to the telephone line. However, ringing can still occur with this setting.

The BNC output connector can be used to monitor either the signals preset on the telephone line, or the signals created by the on board tone generators. If the telephone line is monitored, then signals originating from both the telephone under test, or the internal tone generators will be present at the BNC output. If the tone generator output is selected, then the signals sent by the telephone under test will not be present at the BNC output.

For more information on the relationship between the signal levels at the telephone line and the BNC input or output connector, see the section: BNC Input/Output Levels

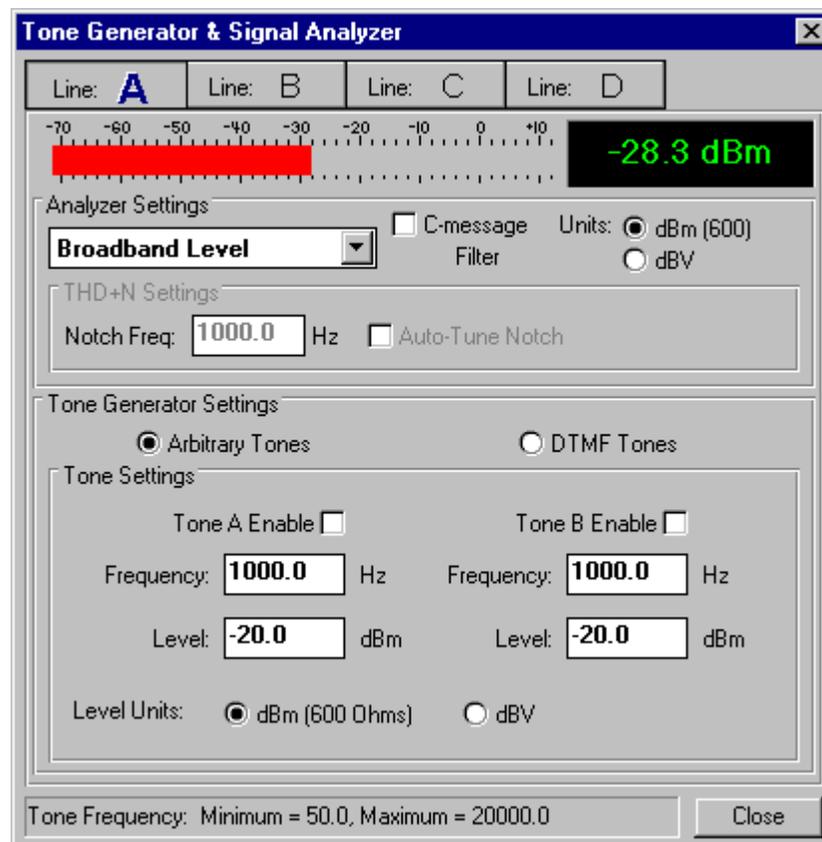
■ Section 5 Using the Program Tools

This section describes in more detail the operation of various program tools. Major topics covered include the following:

- 1) Tone Generator & Signal Analyzer
- 2) Log of Events and Dialing
- 3) Pulse Dialing Waveform
- 4) Last Dialed Digit Analysis

■ Section 5-1 Tone Generator & Analyzer

A flexible signal analyzer and tone generator can be activated by selecting the [TOOLS] [TONE GENERATOR & ANALYZER] menu command, or clicking the mouse on the equivalent tool bar button. In this mode of operation, no further DTMF digits, pulse digits, or line flashes are detected. Also, none of the action scripts are executed if an event occurs. The tone generator and signal analyzer panel is shown below.

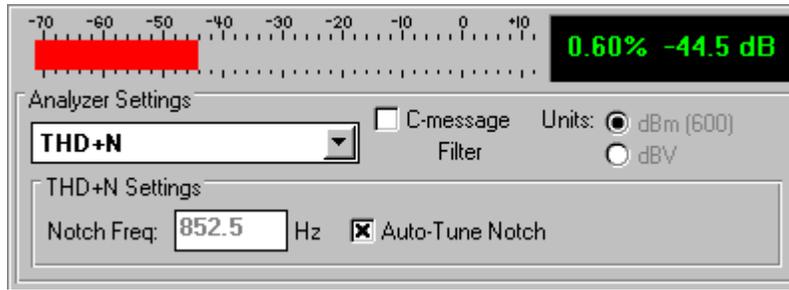


The line designation tab at the top of the window control which line is being acted upon. Any one of up to four different telephone lines can be selected. If fewer than four telephone lines are present in the system, then the extra tabs will be invisible.

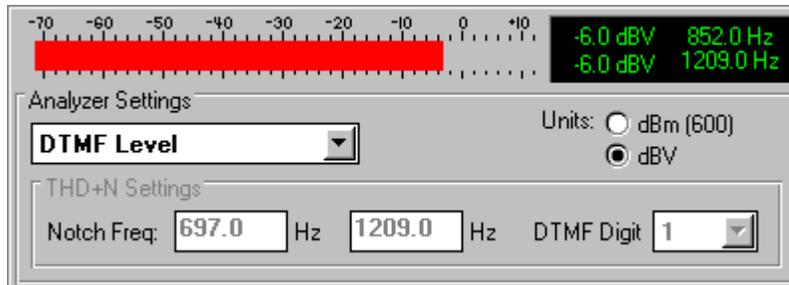
Signal Analyzer

The top half of the panel contains the signal analyzer. Four different measurement types can be chosen from the drop-down list. They are broadband level, total harmonic distortion plus noise (THD+N), DTMF low and high group level and frequency, and DTMF total harmonic distortion plus noise.

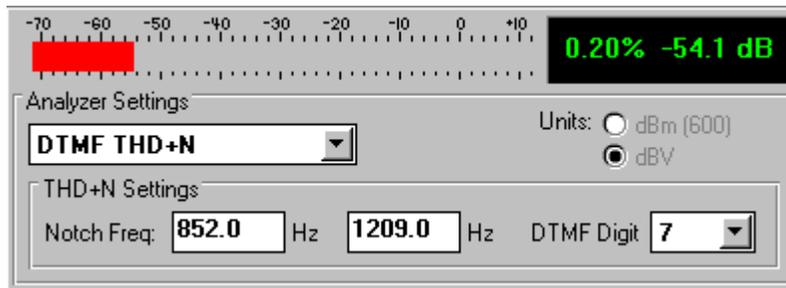
The broadband level measurement will measure the total RMS signal level present at the telephone line. The units for the level measurement can be either dBm (600 ohm) or dBV. Depending on the selection, the bar graph and numeric readout value will change slightly. Optionally, a C-message weighting filter can be inserted in the signal measurement path. This can be useful when measuring noise levels on the telephone line, which are occasionally specified with a C-message weighting filter.



A distortion plus noise measurement is made by changing the measurement mode to THD+N. The THD+N result is a ratio of the RMS signal level after a notch filter to the RMS signal level before the notch filter. The readings of THD+N will be displayed as both a percentage and decibel value. The bar graph will reflect the reading in dB's. The notch filter used in making the measurement can be set to automatically track the signal being measured. If the Auto-Tune check box is enabled, the center frequency of the notch will align itself to the signal's frequency. However, under noisy conditions the notch filter may not settle properly to an accurate frequency. In this case, it is best to disable the Auto-Tune check box and enter the notch frequency manually. As with the broadband level measurement, an optional C-message weighting filter can be inserted in the signal path, by clicking the mouse on the C-message filter check box.



An analysis of a DTMF digit low and high group tone is made by changing the measurement type to DTMF Level. The RMS level and the frequency of both the low and high group tones are displayed in the readings panel. In the above example a DTMF digit 7 is being measured. The low group tone is measured at -6.0 dBV at a frequency of 852.0 Hz while the high group tone has a level also at -6.0 dBV and a frequency of 1209.0 Hz. The bar graph shows the total combined signal level at -3 dBV. As with the broadband level measurement, the level units can be either dBm (600 ohms) or dBV. The THD+N settings have no effect on the level or frequency measurement.



The last measurement mode reads the distortion plus noise of a DTMF digit. This is selected by changing the measurement type to DTMF THD+N. Similar to the THD+N measurement, the reading of the DTMF THD+N is derived from a ratio of signal level after a notch filter to the signal level before the notch filter. However, since a DTMF digit contains two separate tones, two notch filters are employed. One notch removes the low group tone while the other notch removes the high group tone. To set the notch frequencies, select the DTMF digit being measured from the drop-down list box. In this case, the digit 7 is being measured. Changing the digit automatically updates the center frequency of the two notch filters. To manually change the notch frequency, simply type a new frequency value and press the ENTER key. The readings of the DTMF THD+N will be displayed as both a percentage and decibel value. The bar graph will reflect the reading in dB's.

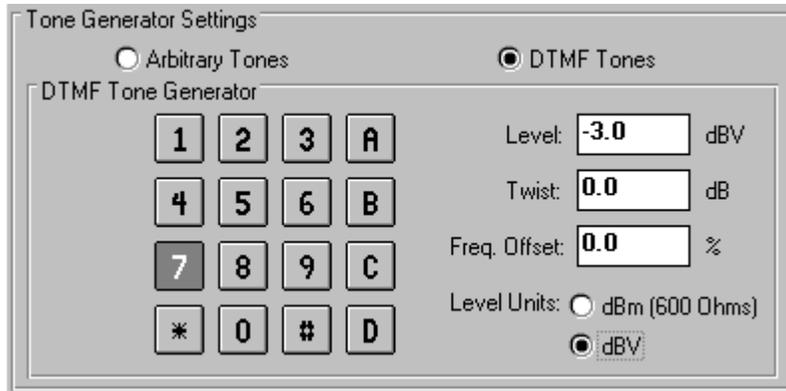
Tone Generator

The tone generator can operate in one of two different modes. The first is simply a dual tone arbitrary generator which can produce either one or two tones simultaneously over a wide range of levels and frequencies. Each tone generator, A and B, can be enabled and disabled independently by clicking the mouse on the enable check boxes respectively. To change the frequency or level, type a new value in the appropriate field and press the ENTER key. The panel at the bottom of the window will show the minimum and maximum limits for any changes made. If the new value is outside the valid range, it will be clamped to either the lower or upper limit.

The level units can be specified in either dBm (600 ohms) or dBV. There are significant differences between how the two units influence the output level. The selection of dBm (600 ohms) means that the tone generator will deliver the requested power level into the telephone under test assuming its terminating impedance is 600 ohms. As such, a setting of 0 dBm will deliver 1 milliwatt of power into a terminating 600 ohm load. If the telephone line is unterminated, then the voltage present at the telephone line will increase depending on the source impedance setting. If the source impedance is set to 600 ohms, then the voltage level will be 6 dB higher than specified. If the source impedance is set to 900 ohms, then the voltage level will be 7.9 dB higher than specified. However, it is important to note that if the source impedance is changed to the complex setting, the program can not compensate for the source impedance since the magnitude and phase of its impedance changes with frequency. In this case, the specified level in dBm will reflect that of an unterminated telephone line.

If units of dBV are selected for the tone generator, the output level will reflect the setting value for an unterminated telephone line under all source impedance

settings. For example, generating a level of 0 dBV will produce 1 Vrms of voltage across the tip and ring leads when the telephone line is unterminated. If terminated, the level will be reduced by the relationship between the source and load impedances.

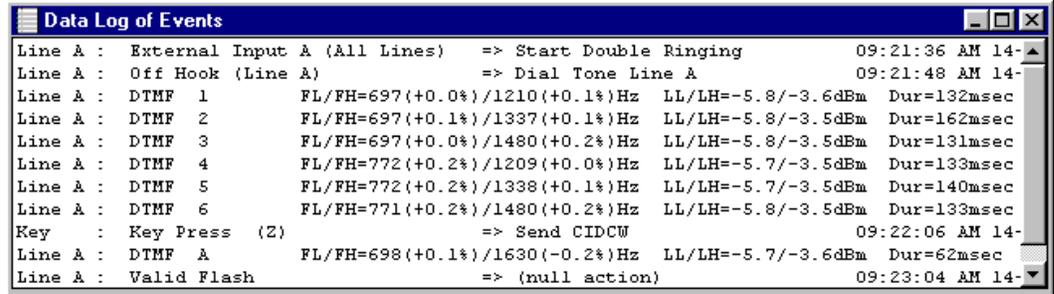


The second mode of operation links the two tone generators together in order to produce DTMF tones. Clicking the mouse on any of the 16 DTMF digits will toggle the tone generators on and off. The level, twist, and frequency error can be specified independently on the right side of the window. The twist is defined as the ratio of high group tone level to low group tone level and is entered in units of dB. Also, a frequency error of up to +/- 5% for both the low and high group tones can be specified. This can be useful in testing the detection range of DTMF decoders. As with the arbitrary mode of operation, the DTMF tone levels can be defined with the units of dBm (600 ohms) or dBV. The effect of the telephone line source impedance and load impedance is also the same as that of the arbitrary tone generator.

■ Section 5-2

Event and Dialing Log

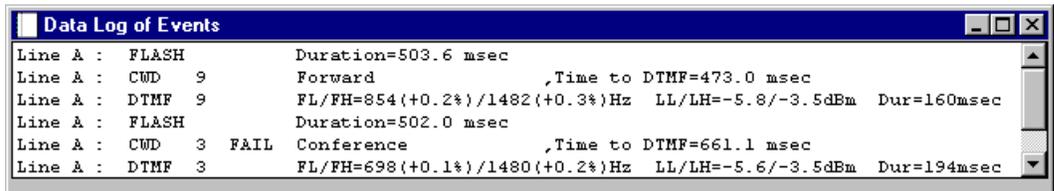
The TSA3300 program maintains a record of all the events and dialing for all of the active telephone lines. This can include the results from DTMF dialing, pulse dialing, flash timing, CWD timing, any telephone line events that may or may not be linked to an action script, key press events, and messages generated by an action script. By enabling filters, the type of data collected in the log file can be restricted in order to reduce the volume of data collected. The event log window can be viewed by selecting the [TOOLS] [OPEN EVENT LOG WINDOW] menu command. The following figure is a sample of some of the data that can be collected in the event log.



Line A	Event	Action Script	Time
Line A	External Input A (All Lines)	=> Start Double Ringing	09:21:36 AM 14-
Line A	Off Hook (Line A)	=> Dial Tone Line A	09:21:48 AM 14-
Line A	DTMF 1	FL/FH=697(+0.0%)/1210(+0.1%)Hz LL/LH=-5.8/-3.6dBm Dur=132msec	
Line A	DTMF 2	FL/FH=697(+0.1%)/1337(+0.1%)Hz LL/LH=-5.8/-3.5dBm Dur=162msec	
Line A	DTMF 3	FL/FH=697(+0.0%)/1480(+0.2%)Hz LL/LH=-5.8/-3.5dBm Dur=131msec	
Line A	DTMF 4	FL/FH=772(+0.2%)/1209(+0.0%)Hz LL/LH=-5.7/-3.5dBm Dur=133msec	
Line A	DTMF 5	FL/FH=772(+0.2%)/1338(+0.1%)Hz LL/LH=-5.7/-3.5dBm Dur=140msec	
Line A	DTMF 6	FL/FH=771(+0.2%)/1480(+0.2%)Hz LL/LH=-5.8/-3.5dBm Dur=133msec	
Key	Key Press (Z)	=> Send CIDCW	09:22:06 AM 14-
Line A	DTMF A	FL/FH=698(+0.1%)/1630(-0.2%)Hz LL/LH=-5.7/-3.6dBm Dur=62msec	
Line A	Valid Flash	=> (null action)	09:23:04 AM 14-

Various events and DTMF dialing digits have been captured in the figure above. Each entry in the log window is represented by a single line. The first line indicates that the External Input A event was triggered and caused the execution of the script named Start Double Ringing. This occurred for Line A, as shown at the beginning of the line. The current system time and date are appended to the end of the line. The second entry also shows an event captured. This was Line A going off-hook, which started the dial tone. The format for displaying any recorded events is always the same. First is the telephone line designation, followed by the event name, then action script name, then current system time.

The next six entries in the log window show that the DTMF digits 1 through 6 were dialed for line A. With each digit, the low and high group frequency and level are displayed, along with its duration. If the received DTMF digit low and high group frequencies and levels does not meet the limits specified in the DTMF Dialing Setup window, then its entry in the log file would show "FAILED" following the digit number. Pulse dialing digits are treated in a similar manner. The type of data displayed with the DTMF or pulse digits can be changed or re-ordered in the event log options window.

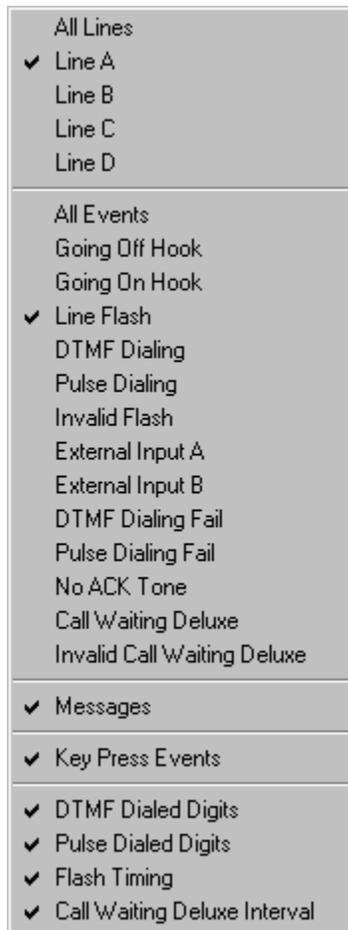


Line A	Event	Action Script	Time
Line A	FLASH	Duration=503.6 msec	
Line A	CWD 9	Forward ,Time to DTMF=473.0 msec	
Line A	DTMF 9	FL/FH=854(+0.2%)/1482(+0.3%)Hz LL/LH=-5.8/-3.5dBm Dur=160msec	
Line A	FLASH	Duration=502.0 msec	
Line A	CWD 3 FAIL	Conference ,Time to DTMF=661.1 msec	
Line A	DTMF 3	FL/FH=698(+0.1%)/1480(+0.2%)Hz LL/LH=-5.6/-3.5dBm Dur=194msec	

The above figure, shows two CWD service requests captured. The first was successful, while the second failed, due to an invalid delay between the hook switch flash and the DTMF digit. The entire CWD event is broken down into 3 entries. The first records the flash time, the second records the interval time and the CWD service being requested, and finally, the third entry records the DTMF digit information.

Filters

With multiple telephone lines, the amount of data in the event log window can become very large. Filters can be used to capture only the data or events of interest, and ignore everything else. To view the current filter settings, and to make changes, select the [TOOLS] [SET EVENT LOG FILTERS] menu command. This will display a menu of all of the filter settings along with which settings are currently active.



Clicking the right mouse button, when over the event log window, is a short cut to displaying the filter menu. Any item in the filter menu that has a check mark beside it will allow that type of record to be viewed and stored. The top section of the filter menu deals with the multiple telephone lines. If only a specific telephone line is of interest, then all the others may be disabled. This would prevent the events and dialing of any other telephone line from being recorded. The second section in the filter menu controls which specific events will be logged. This can be used to block out events that are not of any interest. Finally

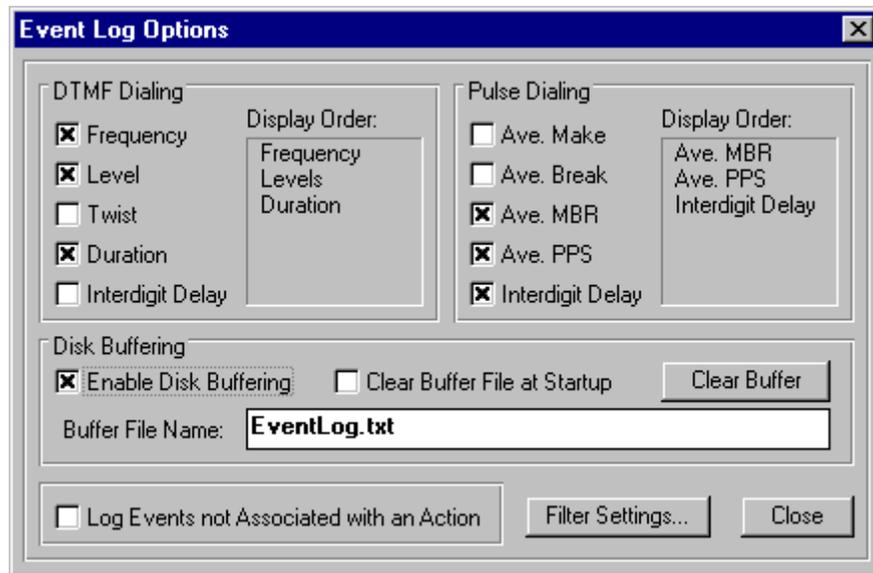
the lower portion of the filter menu controls if messages, key press events, or dialing data is stored to the log window.

Changing the filter settings has two effects. The first is that any new entry is compared to the new filter settings. If they do not match the settings, the event or dialing entry is not recorded. The second effect is that the current entries in the log window are sorted through with the new filter settings, and only the matching entries are displayed in the log window. However, any entries that did not meet the new filter requirements are not discarded. They are still stored in the event log, but just not visible. As such, it is possible to initially allow the recording of all events and dialing. Once all of the desired data has been collected, the filter settings can be changed to view specific events or dialing occurrences without losing any of the other entries.

Note that the event log window will only contain up to a maximum of 1000 entries. Beyond this, the earliest events are removed so that only the last 1000 entries are stored. However, it is possible to buffer all the entries to a file on disk, allowing for an almost unlimited amount of data collected.

Event Log Options

The event log can be customized by selecting the [TOOLS] [SET EVENT LOG OPTIONS] menu command. This displays the following window from which the dialing entries to the event log can be changed, disk buffering can be enabled, and non-associated events can be recorded.

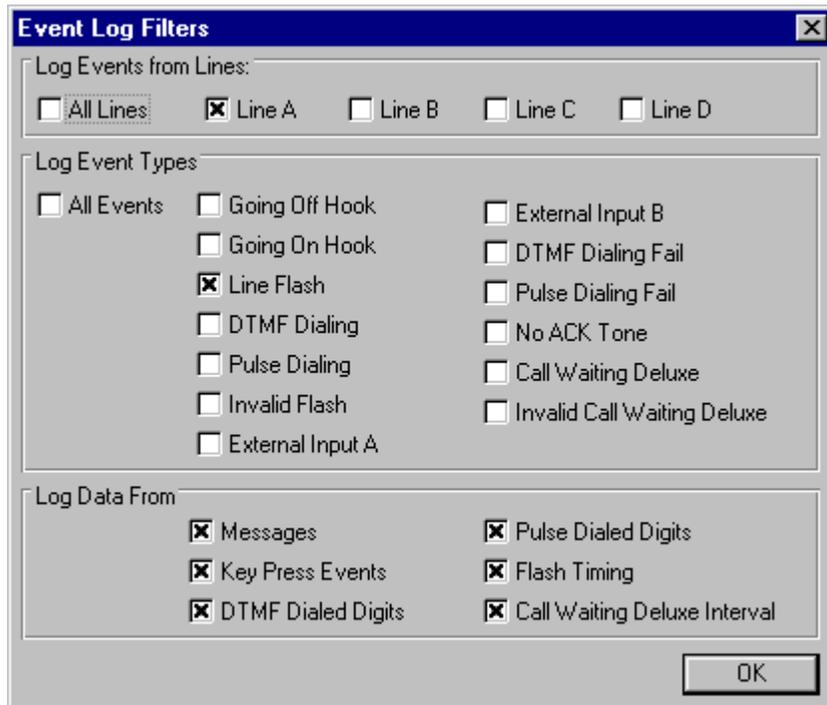


The top section of the options window controls how the results from DTMF or pulse dialing will be stored in the event log. For DTMF dialed digits, up to five different measurements can be written to the log. By clicking the mouse on the desired measurement, it can be added to or removed from the event log. The order in which the measurements are displayed depends on which option is selected first. New measurements are always added to the end of the list. So to change the order, disable all of the options and then re-enable them in the desired order. The same applies to the pulse dialing digits. Up to five different measurements can be displayed in any order.

In order to handle a very large number of events and dialing digits, the event log entries can be buffered to a disk file. To enable this feature, simply click the mouse on the "Enable Disk Buffering" check box. Once enabled, any event or dialing digit that meets the filter requirements is appended to the specified disk file. This file can grow in size until running into the limitation of the storage medium. Unlike the event log window, changing the filter settings will not re-sort or shuffle the file on disk. If the event log file name does not have a path specified, it will be stored in the same directory as the TSA3300 program. If a full path has been specified, the file will be stored in the given directory. An option exists that will automatically clear the event log file every time the program starts. This can be useful in preventing the file from reaching excessive sizes. Also, the event log file can be erased by pressing the "Clear Buffer" button.

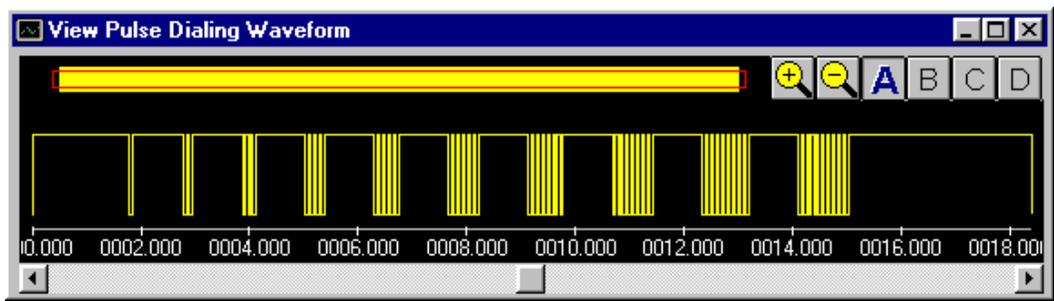
The check box at the bottom of the window controls if events not associated with an action will be recorded to the event log. Normally, the only events that will be recorded are assigned to an action script. However, at times it can be useful in knowing when an event occurred regardless of whether or not it triggered an action script. By enabling this feature, all events will be recorded regardless of if they have been linked to an action script, or not. Note that the filter settings are still applicable in this case. Unless the event passes through the filter, it will not be stored.

In addition to changing the filter settings by right clicking the mouse in the Data Log of Events window, selecting the "Filter Settings..." command button in the Event Log Options window opens a new window where all the filter settings can be adjusted. Using this method to change the filter settings is more effective when changing a large number of the options at the same time.



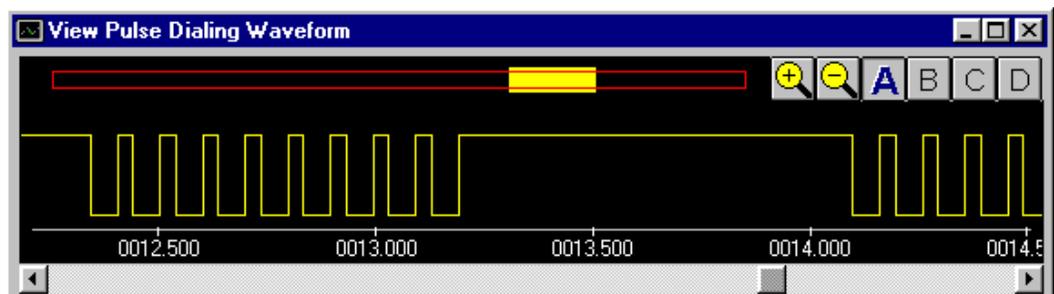
■ Section 5-3 Pulse Dialing Waveform

The pulse dialing hook switch transitions can be viewed graphically by selecting the menu command [TOOLS] [OPEN PULSE DIALING WINDOW]. The waveform represents the state of the hook switch for the telephone under test from when the telephone went off-hook to the time it returned to the on-hook state. The pulse dialing waveform for any of the four telephone lines can be viewed by clicking the mouse on the appropriate button (A to D) in the upper right hand corner of the window. In the following figure, the pulse dialing digits 1 through 9 and zero have been captured.



The starting point, or time zero, for the graph is when the telephone went off-hook. At slightly before the two second mark, the digit 1 was dialed. This was followed by all the remaining digits. The last transition captured marks the telephone returning to the on-hook state at roughly 18 seconds after going off-hook. The waveform is cleared when ever the telephone goes off-hook or at the end of a line flash.

Zooming in to, and out of the waveform is done by pressing the two zoom in/out buttons in the upper right corner of the window. Each time the zoom in button is pressed, the time scale is reduced by a factor of 2. Likewise, when zooming out, the time scale is increased by a factor of 2. The relative scale of the displayed waveform to the total waveform record is shown at the top of the window. This shows the location and zoom factor of the current waveform segment on the screen.



The horizontal scroll bar at the bottom of the window can be used to pan either left or right. The figure above shows in more detail the hook switch transitions when the digit 9 was dialed, and the beginning portion of the following digit 0.

■ Section 5-4 Last Dialed Digit Analysis

The measurement results of the last dialed digit can be viewed continually by enabling the Last Dialed Digit window. This is done by selecting the [TOOLS] [SHOW LAST DIGIT RESULTS] menu command. The window will show either the DTMF dialing or pulse dialing results for the last digit dialed on any active line. An example of the measurement results for a DTMF dialed digit is shown below.

[A] Digit = 5	
Low Group Level	-4.2 dBm
High Group Level	-1.9 dBm
Total DTMF Level	0.1 dBm
Twist Level	2.3 dB
Low Group Freq	772 Hz (0.2%)
High Group Freq	1336 Hz (0.0%)
Tone Duration	83 msec
Interdigit Delay	127 msec

The title bar of the window indicates which line had dialed the digit by showing the telephone line designation letter in square brackets. The measurement values for each digit are the same as what is displayed by pressing and holding the right mouse button over the displayed digit in the main panel window. If any of the measured results are outside the specified limits, the background color of the reading will turn to a yellow color. The signal level units for the DTMF digits are the same as the level meter for that particular telephone line. As such, changing the level meter units to dBV, or Vrms will also change the low and high group level and total signal level units to the same.

[A] Digit = 7	
Average Break	61.8 msec
Minimum Break	61.8 msec
Maximum Break	61.8 msec
Average Make	36.4 msec
Minimum Make	36.4 msec
Maximum Make	36.4 msec
Average MBR	63.0 %
Average PPS	10.2 pps
InterDigit Delay	907.9 msec

If a pulse digit is dialed, the last dialed digit window will alter slightly to show the results of the different measurements taken. As with the DTMF digits, if any measurement is outside the limits specified, then the measurement reading will have a background color of yellow instead of the normal gray.

To remove the last digit dialed window, select the [TOOLS] [SHOW LAST DIGIT RESULTS] menu command again, or click the mouse on the window's close button in the upper right hand corner.

■ Section 6

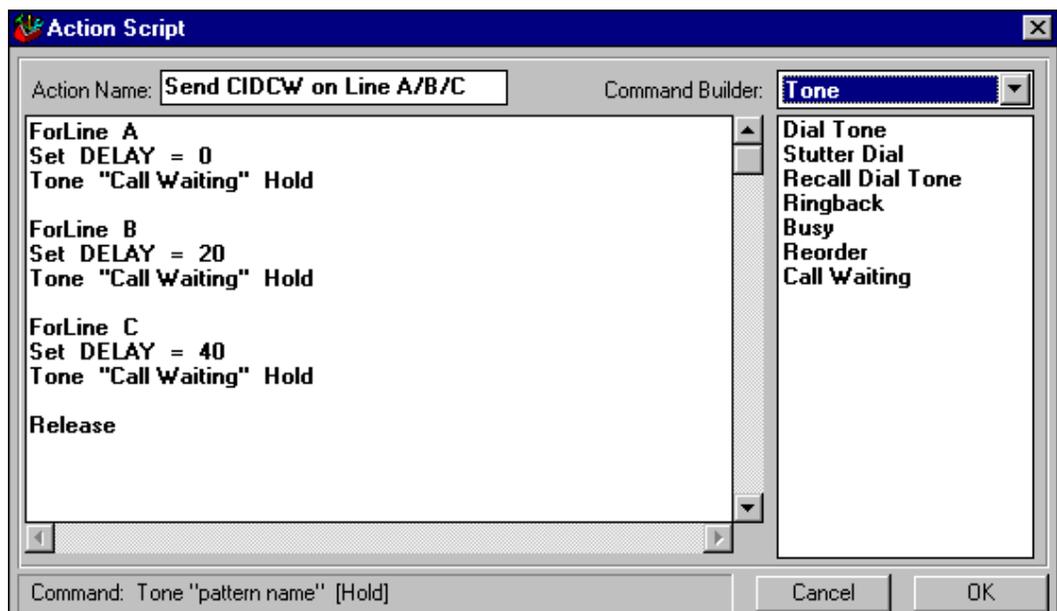
Additional Information

Synchronizing Multiple Telephone Lines

In testing multiple line telephones, it becomes important to simulate worst case conditions for any real time processing required by the telephone. An example of this is multiple Type II Caller ID transmissions, where a telephone must detect simultaneous CAS tones along with decoding the FSK on multiple lines. Other situations include detecting ringing occurring at the same time on multiple lines.

By using the action script commands RING and TONE, in conjunction with the DELAY parameter, tight control over timing variations can be achieved. The DELAY parameter controls the amount of time delay inserted before starting any ringing, network tone, or OSI action. It is unique for each line, as such, each line can have a different delay value. Appending the HOLD keyword at the end of the RING, TONE, or OSI script commands causes the action to be prepared, but holds its execution. In this manner, the actions on multiple lines can be set up in advance and at the end of the action script, they are started virtually simultaneously by using the RELEASE command.

In the figure below, lines A, B, and C are sent a Type II (CIDCW) Caller ID transmission. However, by using the DELAY parameter value, the SAS/CAS tone on line B will be delayed by 20 msec, relative to line A. Likewise, the SAS/CAS tone for line C is delayed by 40 msec, relative to line A. If the DELAY value were set to zero, then the SAS/CAS tones for all the telephone lines would commence at the same time.



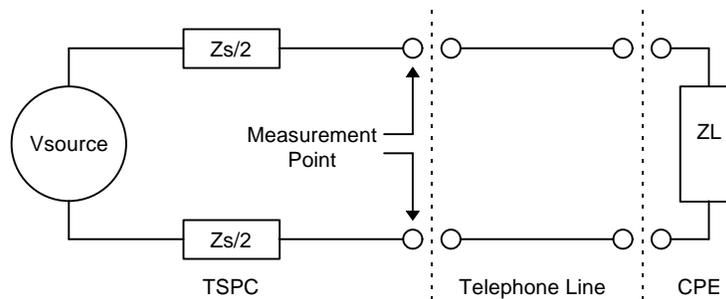
Using the HOLD keyword on each of the three TONE commands, holds back the start of the tone until the RELEASE command is processed. If the HOLD keyword is not present, that line would start the CIDCW sequence immediately, without waiting for the rest of the script to execute. This can result in significant delays between the signals on the telephone lines. The HOLD keyword can be used with the TONE, RING, and OSI commands. The range of the DELAY parameter is from 0 to 10000 msec (10 seconds). If the specified value is outside this range, it will be clamped to the closest minimum or maximum value.

It is important to note that the PC's processor speed has an influence on the timing accuracy in starting multiple actions simultaneously. This residual timing error is in the range of ± 2 msec for a Pentium class processor. Since the sequence of starting actions is from line A to line D, line A will always be the first to start and line D the last. As such, if more accurate timing is needed, it is possible to delay line A by a few milliseconds using the DELAY parameter. The accuracy of the DELAY parameter is within ± 1 msec.

Telephone Interface Signal Levels

Signal levels are a common source of confusion with most signal generation equipment. This section aims to explain how signal levels are calculated within the program.

It is important to understand how the Telephone Signal Processing Card (TSPC) delivers its AC signals to the CPE under test. The figure below shows a simplified version of the AC signal path from the TSPC to the CPE. The TSPC consists of basically a programmable AC voltage source followed by two impedance's. The two impedance's combined represent the total source impedance (Z_s) of the TSPC which can be set to 600 ohms, 900 ohms, or a complex impedance in the Telephone Interface settings window. The CPE under test is represented by the impedance Z_L .



Any signal measurements made are taken from the tip and ring leads. As such, the signal meters will measure both the signals generated by the TSPC and the signals generated by the CPE.

Generally, all of the tones generated by the TSPC can be defined in level units of dBm (600 ohms), or dBV. This setting has a significant impact on how the AC voltage source output levels are calculated by the program.

Using units of dBm (600 ohms)

The units of dBm, in the context of this program, always refers to a power level delivered into a 600 ohm load. As such, a setting of 0 dBm for a signal level will produce a voltage level equivalent to 1 milliwatt of power dissipated into a 600 ohm resistor. This voltage level works out to be 0.7746 Vrms, or -2.218 dBV. As such, the program will compensate for the different possible source impedances. With a telephone interface source impedance of 600 ohms, the program will set the AC voltage source to a level 6 dB higher than what is specified, while for an impedance of 900 ohms, its voltage level will be 7.96 dB higher. This insures that the power delivered to a 600 ohm across tip and ring will be exactly what was entered. However, if the complex source impedance is selected, then the program can not compensate for the source impedance since the magnitude and phase of its impedance changes with frequency. In this case, the specified level in dBm will reflect that of an unterminated telephone line.

If the CPE's impedance (ZL) is not 600 ohms, the power level delivered to the CPE will be different from what is specified. The voltage level can be calculated, in any case, from the following formula:

$$V_{cpe} = V_{specified} * \frac{Z_{s1} + 600}{600} * \frac{Z_L}{Z_{s2} + Z_L}$$

where: V_{cpe} is the voltage level at the CPE in Vrms
 V_{specified} is given in Vrms
 ZL is the load impedance of the CPE
 Zs1 is the source impedance of the TSPC
 Zs2 is the source impedance of the TSPC

As seen from the above formula, if the telephone line is unterminated, then the voltage level will be 6 dB (2 times) higher than specified if the source impedance is set to 600 ohms, and 7.96 dB (2.5 times) higher if the source impedance is set to 900 ohms.

In the case that the TSPC's source impedance is set to the complex setting, the program can not compensate for the source impedance since its magnitude and phase response varies with frequency. When using the complex impedance, the signal levels will always be calculated assuming an unterminated telephone line. The above formula can still be used, but substituting a value of zero for Zs1.

Using units of dBV

If units of dBV are selected for the various tones, the output level will reflect the setting value for an unterminated telephone line under all source impedance settings. Generating a level of 0 dBV will produce 1 Vrms of voltage across the tip and ring leads when the telephone line is unterminated regardless of the telephone interface source impedance. If terminated, the level will be reduced by the relationship between the source and load impedances.

For example, setting the tone generator to produce an output level of -20 dBV converts into 0.100 Vrms. This is the level at the tip and ring leads with an unterminated telephone line. However, if the telephone line is terminated, the level present at the CPE is given by:

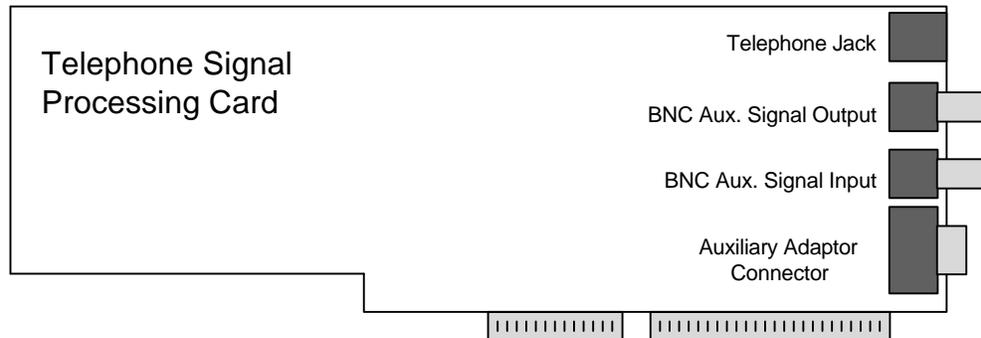
$$V_{cpe} = V_{source} * \frac{Z_L}{Z_L + Z_{source}}$$

$$Z_s + Z_L$$

where: V_{cpe} is the voltage level at the CPE in V_{rms}
 V_{source} is given in V_{rms}
 Z_L is the load impedance of the CPE
 Z_s is the source impedance of the TSPC

External BNC Signal Levels

The two BNC connectors on the rear of the Telephone Signal Processing Card (TSPC) can be used to monitor signals from the CPE, and to inject signals directly to the CPE. The figure below shows the BNC signal output connector being located below the telephone jack, and just above the BNC signal input connector.



BNC Output Connector

The auxiliary signal output has an output impedance of 600 ohms, and can either be set to monitor the internal tone generators, or the signals being received from the CPE connected via the telephone jack. When monitoring the signals generated by the TSPC, the output level present at the BNC connector (with a high impedance load) is equal to one quarter (12 dB less) of the AC voltage source driving the telephone interface.

For example, setting the Tone Generator for an output level of 0 dBm (terminated) with a telephone interface impedance (Z_s) setting of 600 ohms, the level at the BNC connector will be -8.22 dBV into a high impedance. This is because to deliver 0 dBm of power across a 600 ohm load requires a voltage level of -2.22 dBV across the load. With a 600 ohm source impedance, the generator level must be 6 dB higher, or +3.78 dBV. Since the level at the BNC connector is 12 dB less than the AC voltage source driving the telephone line, its level will be 3.78 dBV - 12 dB = -8.22 dBV. If the telephone interface impedance (Z_s) is changed to 900 ohms, the level at the BNC connector will be increased by 1.96 dB, as the AC voltage source driving the telephone line must be increased in order to maintain a level of 0 dBm across the load. In this case the BNC level is -8.22 dBV + 1.96 dB = -6.26 dBV.

See the previous section dealing with Telephone Interface Signal Levels, for a more in-depth look at the signal levels within the telephone interface. If monitoring signals generated from the CPE, the signal level present at the BNC connector (with a high impedance load) is 6 dB less than the voltage level across the CPE. So if a level of 0 dBm is present at the CPE terminals and the CPE represents a 600 ohm load the voltage level will be -2.22 dBV. Routing this signal to the BNC output connector will give it a level of -2.22 dBV - 6 dB = -8.22 dBV.

BNC Input Connector

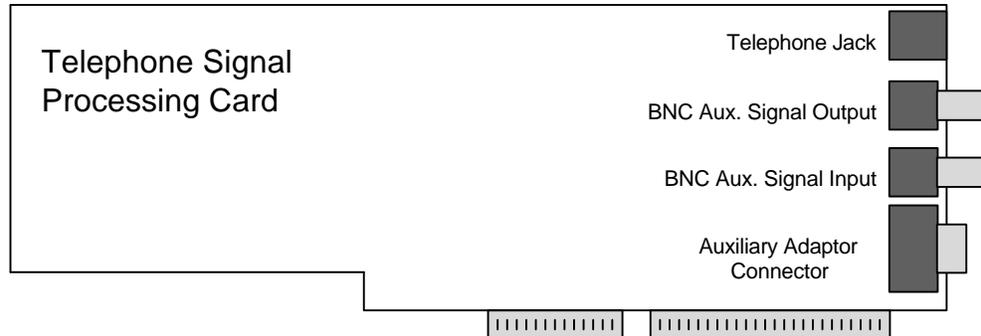
The auxiliary signal input has an input impedance of approximately 100 kohms. Signals injected in this input can be routed directly to the CPE depending on the signal routing settings. If set to route the input signal to the CPE, the voltage at the AC voltage source driving the telephone interface will be four times higher than the voltage level injected at the BNC input connector.

For example, if the CPE is disconnected (open circuit on tip and ring), the voltage at the telephone line would be 12 dB more than the voltage present at the BNC input connector. If the CPE presents a 600 ohm load and the telephone interface impedance (Z_s) is 600 ohms, the voltage level across the CPE would be 6 dB more than the voltage level at the BNC input connector. Changing the telephone interface's source impedance (Z_s) to 900, the voltage level across the CPE will be reduced by 1.96 dB.

As above, see the previous section dealing with Telephone Line Signal Levels for a more in-depth look at the signal levels within the telephone interface.

Auxiliary Digital Input & Output

The DB9 pin connector at the rear of the Telephone Signal Processing Card (TSPC) provides access to general purpose digital outputs and inputs that can be used for various functions throughout the program.



The DB9 female connector allows for the use of up to 2 digital outputs, 2 digital inputs, and a +5V supply from the PC. The digital outputs are only active when Pin 7 is connected to ground. This serves as a output port enable. The outputs are driven from 5V 'HC CMOS logic. Pin 3 is defined as Output A, while pin 4 is Output B. The two available inputs are at pins 8 (Input A) and 9 (Input B). The input voltages should be limited to ground and +5V in order to prevent damage to the internal 'HC CMOS buffers. Both Input A and Input B have 10 kohm resistors to ground in order to prevent floating inputs. The pin assignment for the DB9 connector is as follows:

- Pin 1: +5V (can draw up to 0.5A from this pin) (internally fused)
- Pin 2: Reserved Output (do not use)
- Pin 3: External Output A ('HC CMOS 5V output)
- Pin 4: External Output B ('HC CMOS 5V output)
- Pin 5: Ground
- Pin 6: Reserved Output (do not use)
- Pin 7: Digital Output Enabled (connect to ground to enable outputs)
- Pin 8: External Input A ('HC CMOS 5V input)
- Pin 9: External Input B ('HC CMOS 5V input)

The two digital outputs (A & B) can be programmed high or low by the scripting commands. This may serve to control external switching relays, or be used to trigger external equipment (oscilloscopes, logic analyzers, emulators) during key periods of interest. The two digital inputs (A & B) can be used to trigger internal program events. To properly use the inputs for an event, the logic level must change from a low level to a high level, and remain high for at least 200 msec. A simple use for the two inputs can be to connect push button switches to them that trigger the program events. These events can then cause other actions such as ringing the telephone or sending network tones. This eliminates the keyboard for simple production related testing.

Telephone Line Balance and Grounding

As with any system of instruments and devices, proper grounding is essential in order to minimize circuit hum and susceptibility to interference. It is important to understand the circuit grounds of all the devices in the test setup to ensure their proper connection.

Like most PC based instruments the TSPC's ground point is referenced to the PC's ground, which should be connected to earth ground. The ground conductor on the BNC input and output and the DB9 connector are at earth ground potential, assuming the computer is properly grounded. The tip and ring leads of the telephone interface have a negative potential with respect to ground depending on the programmed line voltage. At the default setting of 48 volts, the voltages present on tip and ring are approximately -52 volts and -4 volts with respect to the computer's ground.

Care must be taken with CPE's that do not maintain isolation from the telephone interface. This can cause grounding problems when connecting the ground leads of oscilloscopes or other equipment that is normally earth grounded. The most common cause is that the CPE connects either tip or ring to its circuit ground via a bridge rectifier. If the CPE's circuit ground is then connected to earth ground by use of an oscilloscope ground lead, an emulator, or logic analyzer, then this effectively shorts either the tip or ring line to earth ground. The imbalance in loop current is detected by the TSPC and will be shown as a line unbalanced condition.



If the CPE has an isolated telephone interface, then this condition should not occur, since the CPE's circuit ground is not connected to its telephone interface circuitry. If this is not the case, and external equipment must be connected to the CPE under test that forces its ground to be connected with earth ground, there are a few possible solutions available. In some cases, the telephone interface is not a requirement for the testing situation, and the BNC inputs or outputs can be used, since the BNC's are ground referenced unbalanced signals. If the telephone interface is required, then the grounding "loop" must be broken. This can mean isolating the PC that contains the TSPC, or the test equipment connected to the CPE, or the telephone line itself. Depending on what testing is required, the telephone line can be isolated via an AC coupling transformer and a DC feeding bridge. Of course, the DC supply would have to be isolated from earth ground, but this is generally easier to do than isolating the PC.

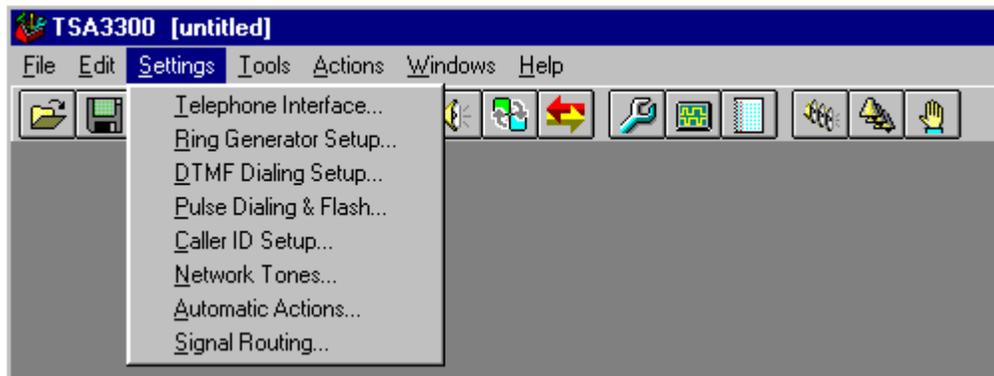
Note: Disconnecting the earth ground wire from the computer is both an improper and potentially dangerous method of isolating the computer from the remainder of the test setup. The AC isolation achieved is generally poor due to the large parasitic capacitance's to earth ground. An alternative approach may be the use of a UPS. UPS's are an excellent method of achieving near perfect isolation between circuit grounds.

■ Section 7

Program Reference

Menu Options

This section describes each command available on the program's menu bar.



The program menu bar contains the following headings.

- 1) FILE menu heading
- 2) EDIT menu heading
- 3) SETTINGS menu heading
- 4) TOOLS menu heading
- 5) ACTIONS menu heading
- 6) WINDOWS menu heading
- 7) HELP menu heading

The FILE Menu Options

The Options under the FILE menu are the following:

[OPEN CONFIGURATION] This option will display a dialog box where the user can select a configuration file to be loaded into the program. The configuration files contain an exact snap shot of all of the settings and data stored in the program at the time the configuration file is saved.

The shortcut key for this command is: **CTRL-O**

[SAVE CONFIGURATION] This menu command will save the program's configuration to the last file name specified with either the Open Configuration or Save As Configuration commands. The saved configuration file is a snap shot of all the settings and data stored in the program.

The shortcut key for this command is: **CTRL-S**

[SAVE AS CONFIGURATION] This option will display a dialog box where the user can select a file name for saving the program's configuration. The resulting configuration file will contain a snap shot of all settings and data stored in the program. This configuration file can then be loaded into the program at any time to quickly return to the same state as when it was saved.

[PRINT CONFIGURATION] Selecting this option sends a snapshot of all the program settings to the currently selected printer. The settings are defined as any parameter that can be accessed and changed under the SETTINGS menu header.

[PRINT EVENT LOG REPORT] This option prints the contents of the event log window to the currently selected printer. Unless the event log window is active (has focus), this menu item will be disabled.

When printing the Data Log, an optional comment field can be filled out. The comments can include information such as test conditions and unit under test details. These comments will appear on the printout and are saved with the program configuration.

[PRINTER SETUP] If various printing devices are connected to the current system, this option allows the user to select the preferred printer, along with various printing options.

[RESTORE DEFAULT SETTINGS] This option resets the current program settings to their default value. Any data collected in the event log is lost. Effectively, this command results in the same actions as quitting and restarting the program.

[EXIT] This option will shut down the telephone signal processing card, and end the program's operation. All data that has not been save at this time will be lost.

The EDIT Menu Options

The options under the EDIT menu are the following:

[CUT] Any text that is currently selected or highlighted will be removed and placed into the Windows clipboard. Once inside the clipboard, the text may be transferred to other Windows programs by PASTE'ing it out of the clipboard.

The shortcut key for this command is: **CTRL-X**

[COPY] Any text that is currently selected or highlighted will be copied into the Windows clipboard. Once inside the clipboard, the text may be transferred to other Windows programs by PASTE'ing it out of the clipboard.

The shortcut key for this command is: **CTRL-C**

[PASTE] If any text is contained within the Windows clipboard, it can be placed into the current text region that has the focus at the insertion point. If text within the text region has been selected, the text from the clipboard will replace the selected text.

The shortcut key for this command is: **CTRL-V**

[SELECT ALL] This option will select or highlight all of the text in the current text region.

The shortcut key for this command is: **CTRL-A**

[CLEAR] Any text that is currently selected will be deleted. This option does not effect the text within the Windows clipboard.

The SETTINGS Menu Options

The Options under the SETTINGS menu are the following:

[TELEPHONE INTERFACE] This menu item will display the Telephone Interface settings window. The line voltage, loop current, impedance, and polarity settings for any of the telephone lines active in the system can be view and altered.

[RING GENERATOR SETUP] This menu item will display the Ring Generator settings window. From this window, the current ringing pattern definitions can be edited, or new ringing patterns can be created.

[DTMF DIALING SETUP] This menu item will display the DTMF Dialing Setup window. These parameters control the measurement limits for the DTMF detector, which are used to distinguish between valid DTMF digits (displayed in green) or invalid digits (displayed in red).

[PULSE DIALING & FLASH] This menu item will display the Pulse Dialing & Flash Timing settings window, where changes to the pulse dialing limits and flash timing limits can be made. Pulse dialing digits are specified in terms of Make-Break Ratio (MBR) and Pulses-per-Second (PPS), while line flashes are specified in terms of on-hook durations.

[CALLER ID SETUP] This menu item will display the Caller ID settings window. From this window, the message type, number to send, name to send, and signaling method used can be specified for each active telephone line.

[NETWORK TONES] This menu item will display the Network Tone settings window. The definitions for the default network tones can be viewed or edited, along with adding new network tone definitions.

[AUTOMATIC ACTIONS] This menu item will display the Automatic Actions window. All of the defined program events, along with their associated actions are displayed. Events and actions can be added, deleted, or edited.

[SIGNAL ROUTING] This menu item will display the Signal Routing window. The signal routing to and from the BNC connectors at the rear of the TSPC is controlled from this window.

The TOOLS Menu Options

The Options under the TOOLS menu are the following:

[TONE GENERATOR & ANALYZER] This menu option opens the Tone Generator & Signal Analyzer panel. From this panel, various different signal measurements can be made, along with being able to generate a wide range of tones. When this window is active, normal DTMF and pulse dialing detection is disabled, as well as flash timing measurement, and the processing of any defined events.

The short cut key for this command is: **F2**

[OPEN PULSE DIALING WINDOW] This option opens the Pulse Dialing Waveform window. If the window is already opened, then this menu command has no effect. The window will display any pulse dialing occurring on the selected telephone line in a graphical format. Also any hook switch transients or glitches can be viewed from this window.

The short cut key for this command is: **F3**

[OPEN EVENT LOG WINDOW] The menu option opens the Event Log window. If the window is already opened, then this menu command has no effect. The event log window displays any events or dialing digits that have occurred for any of the active telephone lines, providing they match the current filter settings.

The short cut key for this command is: **F4**

[SET EVENT LOG FILTERS] This menu option displays a further level of menu commands that determine the filter settings for the event log. These filters are used to selectively record only the events or dialing of interest. If a check mark is present beside a filter option, then that event will be recorded. The check marks can be toggled on and off by repeatedly selecting the menu command.

[SET EVENT LOG OPTIONS] This option displays Event Log Options window. From this panel, various options concerning the recording of data to the event log is controlled.

[CLEAR EVENT LOG] This menu command will clear all entries to the event log.

[SHOW TOOL BAR] This menu option is used to enable and disable the tool bar showing all of the shortcut icons. If the check mark appears beside the command, the tool bar is displayed. This setting is remembered by the program when starting up the next time.

[SHOW HINT LINE] This menu option is used to enable and disable the hint line visible at the bottom of the main window. If the check mark appears beside the command, the hint line is displayed. This setting is remembered by the program when starting up the next time.

[SHOW LAST DIGIT RESULTS] This menu option can be used to enable or disable the Last Digit Results window. This window displays the measurements for the last digit dialed on any active telephone line. If the check mark appears beside the menu command, the last digit results are displayed.

The ACTIONS Menu Options

The Options under the ACTIONS menu are the following:

[GENERATE NETWORK TONES - SINGLE LINE] This option displays a further menu from which a network tone can be generated for a single telephone line. If the telephone connected to the line selected is on-hook, then no network tone is generated. The network definition used will be the same as the last tone sent to that particular telephone line.

The short cut key for this command is: (Line A) **Ctrl-F1**

The short cut key for this command is: (Line B) **Ctrl-F2**

The short cut key for this command is: (Line C) **Ctrl-F3**

The short cut key for this command is: (Line D) **Ctrl-F4**

[GENERATE NETWORK TONES - ALL LINES] This menu option will start generating a network tone for all telephone lines that are currently off-hook. The lines that are on-hook will not be effected.

The short cut key for this command is: **F6**

[RING SINGLE LINE] the option displays a further menu from which ringing can generated for a single telephone line. If the selected telephone line is off-hook, then this command will have no effect. The ringing pattern sent to the selected telephone will be the same as the last pattern sent to that particular telephone line.

The short cut key for this command is: (Line A) **Shift-F1**

The short cut key for this command is: (Line B) **Shift-F2**

The short cut key for this command is: (Line C) **Shift-F3**

The short cut key for this command is: (Line D) **Shift-F4**

[RING ALL LINES] This menu option will start ringing all telephone lines that are on-hook. Any lines that are off-hook will be unaffected.

The short cut key for this command is: **F7**

[STOP RINGING OR NETWORK TONE] This menu command will stop any network tones or ringing on all active telephone lines.

The short cut key for this command is: **F8**

[SEND CALLER ID INFORMATION] This menu command can be used to enable or disable the Caller ID feature. If the check mark appears beside the menu command, then the Caller ID data will be sent with every ringing pattern, and can be sent with the appropriate network tone. If no check mark is present, then no Caller ID data is sent in either on-hook or off-hook forms.

The short cut key for this command is: **F5**

The WINDOW Menu Options

The options under the WINDOW menu are the following:

[TILE WINDOWS] This menu option will arrange all open windows in a tiling pattern across the main program window.

[CASCADE WINDOWS] This menu option will arrange all open window in a cascading pattern from top left to bottom right.

[ARRANGE ICONS] This menu option will arrange all minimized windows along the bottom of the main program window.

[1 to 6] Selecting any of these menu options will bring the associated window to the front, and place all other windows behind it.

The HELP Menu Options

The Options under the HELP menu are the following:

[CONTENTS] This menu option displays the help window table of contents. From the table of contents, specific areas of program help can be reached by clicking on the underlined topic heading.

The shortcut key for this command is: **F1**

[SEARCH FOR HELP ON] This menu option displays the Windows Help Search dialog box. By typing key words, the help program will display help topics that relate to the key word entered.

[TECHNICAL SUPPORT] This menu option shows how you can contact us for questions concerning this product.

[ABOUT TSA3300] This menu option displays the program version code, along with the revision and product codes for the telephone signal processing card(s). The base I/O addresses used to communicate with the telephone signal processing card(s) are also displayed.

Tool Bar

The toolbar gives a shortcut method for accessing various program functions contained in the menus. The function of each toolbar button is described below. For more information on each menu item, see the section: Program Reference: Menu Options

ICON	Description of Action	Equivalent Menu Option
	Open Configuration File	[FILE] [OPEN CONFIGURATION]
	Save Configuration File	[FILE] [SAVE CONFIGURATION]
	Telephone Interface Settings	[SETTINGS] [TELEPHONE INTERFACE]
	Ring Generator Settings	[SETTINGS] [RING GENERATOR SETUP]
	DTMF Dialing Settings	[SETTINGS] [DTMF DIALING SETUP]
	Pulse Dialing & Flash Settings	[SETTINGS] [PULSE DIALING & FLASH]
	Caller ID Setting	[SETTINGS] [CALLER ID SETUP]
	Network Tones Settings	[SETTINGS] [NETWORK TONES]
	Automatic Actions Settings	[SETTINGS] [AUTOMATIC ACTIONS]
	Signal Routing Settings	[SETTINGS] [SIGNAL ROUTING]
	Tone Generator & Analyzer	[TOOLS] [TONE GENERATOR & ANALYZER]
	Open Pulse Dialing Window	[TOOLS] [OPEN PULSE DIALING WINDOW]
	Open Event Log Window	[TOOLS] [OPEN EVENT LOG WINDOW]
	Start Network Tone - All Lines	[ACTIONS] [GENERATE TONES - ALL LINES]
	Start Ringing - All Lines	[ACTIONS] [RING ALL LINES]
	Stop Network Tones or Ringing	[ACTIONS] [STOP RINGING OR NETWORK TONE]

Shortcut Keys

Pressing the shortcut keys are equivalent to selecting its associated menu option.

View Tone Generator & Analyzer	F2
Open Pulse Dialing Window	F3
Open Event Log Window	F3
Toggle Caller ID Enable	F5
Start Network Tones - All Lines	F6
Start Network Tones - Line A	Ctrl-F1
Start Network Tones - Line B	Ctrl-F2
Start Network Tones - Line C	Ctrl-F3
Start Network Tones - Line D	Ctrl-F4
Start Ring Generator - All Lines	F7
Start Ring Generator - Line A	Shift-F1
Start Ring Generator - Line B	Shift-F2
Start Ring Generator - Line C	Shift-F3
Start Ring Generator - Line D	Shift-F4
Stop Network Tone or Ringing	F8
Display On-Line Help	F1
Open Configuration File	CTRL-O
Save Configuration File	CTRL-S
Cut Selection to Clipboard	CTRL-X
Copy Selection to Clipboard	CTRL-C
Paste Selection from Clipboard	CTRL-V
Select All	CTRL-A

■ Appendix A Modifying the Complex Impedance

The telephone line impedance that is presented to the CPE can be set to ,under program control, to one of three possible values. These are:

- i) 600 ohms (real)
- ii) 900 ohms (real)
- iii) Complex impedance ZR

Where the complex impedance ZR is modeled by the following schematic diagram:

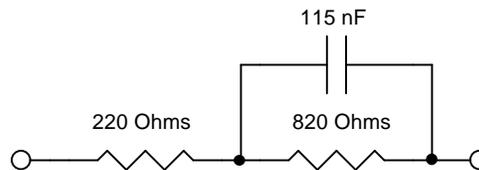


Figure 2. ZR Impedance

However, it is possible to change various components on the TSPC in order to present other custom or user defined impedance's. Figure 3 represents a network of components present on the TSPC that determine the value of the complex line impedance. The line impedance presented to the CPE will be 0.1 times the impedance of the network shown in figure 3. As such, the default complex impedance of ZR can be changed to other values simply by changing values of components R1, R2, C1, C2, and C3.

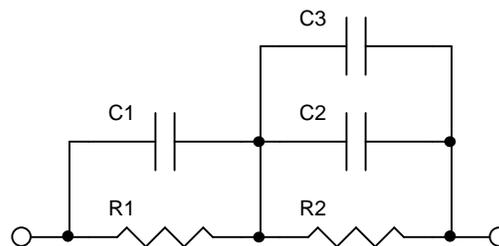


Figure 3. Complex Impedance Network

The default values for the components, in order to present a ZR impedance, are:

- R1 = 2200 ohms
- R2 = 8200 ohms
- C1 = open (not used)
- C2 = 10.0 nF
- C3 = 1.50 nF

The position of these components on the TSPC is shown in figure 4.

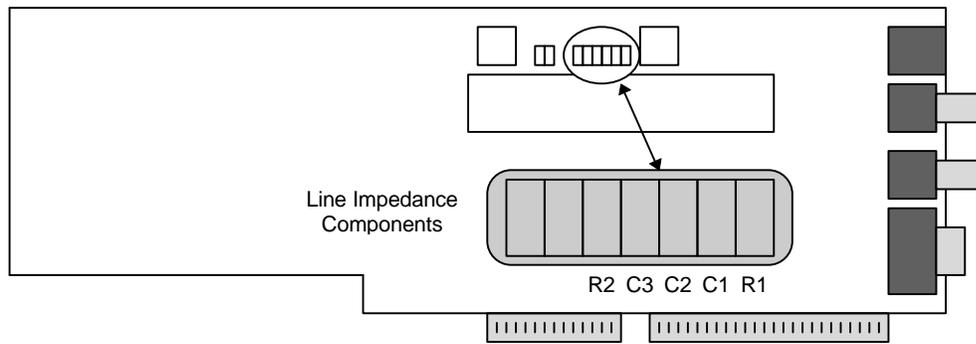


Figure 4. Complex Line Impedance Components

If the component values are to be changed, make sure that standard ESD precautions are taken during the removal of the existing components and soldering of the new components. Of course, the PC should be turned off and the TSPC removed from the PC before any changes are made.

■ Appendix B

General Specifications

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	+/- 0.1dB

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

Telephone Line

Output Impedance	600 or 900 ohms +/- 2%
Output Impedance	complex ZR +/- 3%
Loop Voltage	20 - 52 Volts +/- 1V
Loop Current	20 - 40 mA +/- 10%

Level Meter

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

■ Appendix C

Technical Support

If you encounter problems while using the TSA3300 Telephone Signal Analyzer, please contact us so that we can provide assistance. You may reach us in any one of the following manners:

Email: techsupport@adventinst.com

In North America:

Tel: (604) 944-4298

Fax: (604) 944-7488

Mail: Advent Instruments Inc.
111 - 1515 Broadway Street
Port Coquitlam, BC, V3C 6M2
Canada

In Asia:

Tel: (852) 8108-1338

Fax: (852) 2900-9338

Mail: Advent Instruments (Asia) Limited
Room 13, 23rd Floor, Peninsula Tower
538 Castle Peak Road
Kowloon
Hong Kong

Note: Please include the program version, TSPC product code, revision code, option code, and TSPC base I/O address as given in the [HELP] [ABOUT TSA3300] window in your correspondence.
