

7810 / 7810-TSG IP Telephone

Network Administration Guide



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Introduction

This manual is intended for use by network administrators.

This Network Administrator's Guide explains network administration and network-based phone configuration for Tone Commander 7810 and 7810-TSG SIP VoIP telephones. Troubleshooting procedures are also included in this guide.

Document Overview

Network Setup – setting up the various servers required for a SIP VoIP system.

Configuring Telephones – configuring telephones via server-based XML files.

XML Configuration Files – details of all settings in XML configuration files.

Telephone Software Updates – updating telephone operating software from a TFTP server.

Appendix A – viewing packet statistics logs in the telephone.

Appendix B – Dial Plan syntax.

Appendix C – network and telephone troubleshooting.

Appendix D – an alphabetical index of all XML Tags.

Network Setup

Note: When setting up servers and telephones, server names can be entered in place of IP addresses.

DHCP Server

Automatic IP Configuration (DHCP)

DHCP (Dynamic Host Configuration Protocol) assigns IP addresses to telephones, and can provide other information to the phones, such as server addresses. When using DHCP, phones do not need to be configured with static IP addresses.

The DHCP server can supply:

- Phone IP Address
- Phone Subnet Mask
- Phone Domain Name
- Default Gateway IP Address
- DNS Server(s) IP Address
- SNTP Server IP Address
- TFTP Server IP Address
- SIP Proxy IP Address

Settings not supplied by DHCP must be entered locally at the phone, or in a configuration file which is supplied to the phone via a TFTP server – refer to page 27.

Configuring the DHCP Server

The DHCP server requires a scope of IP addresses that can be assigned to the phones. The scope must be configured with the router address, vendor-specific info, and the TFTP server address.

You can use an existing DHCP server for assigning IP addresses to the telephones, or add a new server.

Scope

Select the DHCP server that you will use for assigning IP addresses to telephones.

Add a new scope, and then enter the following:

- A name and description for the scope.
- The start and end of the IP address range that can be assigned to telephones. Do not include telephones and computers in the same address range.
- Any IP addresses that will be excluded from the address range.
- The lease duration for telephone IP addresses. A lease duration of seven days or longer is recommended.

When the lease expires the phone shows a diagnostic display if idle, while attempting to negotiate a new IP address lease at preset intervals. If the phone is active, the call will be unaffected and the diagnostic display will be shown when the call is cleared.

If the same IP address is offered by the DHCP server, the phone returns to operation without restarting, otherwise the phone will restart after receiving a new IP address.

- The router or default gateway IP address or server name.
- (Optional) parent domain name, DNS servers, WINS servers.

Activate the scope.

Scope Options

Set these scope options:

003 Router

If the router IP address or server name was not set when configuring the scope, enter it here.

004 Time Server

Enter the IP address or server name of the network's SNTP time server.

If your network time server is set to UTC time, enter the hex value for your location's offset from UTC time in seconds.

If your network time server is set to local time, enter **0**.

Time Zone	Offset
Pacific Standard Time	0xffff8f80
Pacific Daylight Time	0xffff9d90
Mountain Standard Time	0xffff9d90
Mountain Daylight Time	0xffffaba0
Central Standard Time	0xffffaba0
Central Daylight Time	0xffffb9b0
Eastern Standard Time	0xffffb9b0
Eastern Daylight Time	0xffffc7c0

066 Boot Server Host Name

Enter the TFTP Boot Server's IP address or server name.

151 SIP Proxy

Enter the SIP Proxy Server's IP address or server name.

SIP Registrar Server

If no IP address or server name is specified for the SIP Registrar Server in the telephone's configuration file, the SIP Proxy Server address will be used as the SIP Registrar Server address.

This address can also be manually changed at the telephone.

Outbound Proxy Server

If the system uses an optional Outbound Proxy Server for outbound requests, enter the server IP address or server name in the telephone's configuration file.

Syslog Server

The Syslog Server records error messages and Quality of Service statistics.

The IP address or server name of the Syslog Server can be specified in the telephone's configuration file, or manually entered at the telephone.


Configuring Telephones

Telephone Setup Menus

Most telephone settings can be entered locally through the Installation Options Menu. For descriptions of each setting, as well as instructions for changing or removing passwords, refer to the [7810 Installation Instructions](#) or [7810-TSG Installation Instructions](#)

Using the Setup Menus


You can enter the Installation Options menu when the phone is idle.

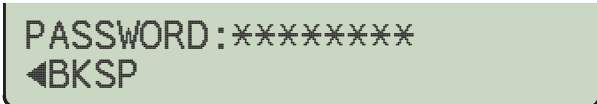
 Press the **SETUP** key.



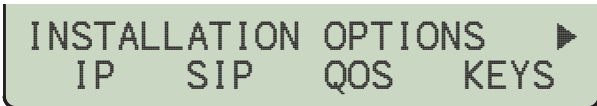
```
      SETUP MENU
INSTL ADMIN  USER
```

 Select INSTL.


 The Installation Options menu may be password protected. Enter your password with the dial pad, and then press the **OK** key.



```
PASSWORD:*****
◀BKSP
```



```
INSTALLATION OPTIONS ▶
  IP   SIP   QOS  KEYS
```

 When ▶ appears in the upper line of the display, you can press the Left or Right Arrow key to see additional menu selections.



```
INSTALLATION OPTIONS ▶
CALL PASSWD RESET UPDATE
```

XML Configuration Files

XML configuration files can be used to configure telephones from a TFTP server.

Details are in the following chapter, beginning on page 11.

XML Configuration Files

All telephone configuration options, including many not available locally from the menus, can be specified in XML configuration files. Settings in these files are loaded into the telephone from a TFTP server.

A **global configuration file**, named "tcs7000a.xml", is loaded first. Settings in this file overwrite any existing local settings. The global file is optional, and only one global file can exist on the TFTP server.

Custom configuration files are unique to each telephone, identified by the primary line ID. Settings in this file overwrite any existing local settings, and any settings loaded from the global file.

All settings are optional. Unused settings may be not included in the files or commented out.

Configuration files can contain sections for specific telephone models. In the sample file on page 13, the first section is loaded into 7810/7810-TSG with or without an 8030X Button Expansion Module. The next section applies only to phones without an 8030X, and the last section applies only to phones with an 8030X.

Editing XML Files

Files can be edited with a dedicated XML editor or any text editor. Although you can manually create the entire file, it is highly recommended that you copy one of the sample files provided by Tone Commander, and edit the copy.

File contents are made up of XML tags, elements and attributes. All tags are optional.

Do not include leading '0' characters in IP addresses.

Refer to the sample configuration file on page 13. Circled numbers identify the various sections. Blank lines between sections are shown for visual clarity, and are not required.

File Names and Location

Configuration files must be located in the TFTP server root folder.

Name each custom configuration file with the associated Ethernet MAC address (default) or the phone's line ID, and include an '.xml' extension; for example, **00048D0000F5.xml** or **4255663001.xml**. All configuration files must be located in the TFTP server root folder and must use the same naming scheme

To use a line ID naming scheme, add the following line in the global configuration file "tcs7000a.xml", enclosed within the model tag (page 12).

```
<config_file_name_base>LINE</config_file_name_base>
```

XML Declaration ①

The first line in the file defines the XML version and encoding. It is used by XML viewers and editors to control formatting. This line must be entered exactly as shown:

```
<?xml version="1.0" encoding="UTF-8"?>
```

Comments ②

Comments can appear anywhere after the first line, and must be enclosed by `<!--` and `-->`. Comments can be included on lines with XML tags or data, and can span multiple lines. *Do not embed a partial comment within another comment. Include `<!--` opening and `-->` closing tags for all comments.*

```
<!-- This is a comment. -->
```

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If you do not wish to assign a particular parameter, comment out the line or delete the entry.

```
<!-- These six lines have been commented out.
    <key num="4">SD
        <speeddial>12345678900#</speeddial>
        <label>SPDIAL 456-8900</label>
    </key>
-->
```

Root Element ③

All settings are enclosed within a single root element. The schema version is used for telephone compatibility management.

```
<tone_commander_settings schema_vers="1.1">
    :
    :
</tone_commander_settings>
```

Telephone Models ④

The model tags enclose all settings for a telephone model or group of models. A configuration can contain multiple model tags.

In the example below, the settings in the first model element apply to 7810/7810-TSG models, with or without an 8030X Button Expansion Module. The second model element contains settings that only apply to 7810/7810-TSG models without an 8030X.

```
<TCS_phone model="7810,7810 + 8030X">
    :
    :
</TCS_phone>

<TCS_phone model="7810">
    :
    :
</TCS_phone>
```

Allowed TCS_phone model values are:

7810	(7810/7810-TSG)
7810 + 8030X	(7810/7810-TSG + 8030X Button Expansion Module)

Other Settings

Other configuration settings are enclosed within the tags listed in the tables beginning on page 15.

Sample XML Configuration File

```

① <?xml version="1.0" encoding="UTF-8"?>
② <!-- Configuration file for Alexander Great (425) 566-3001 -->
③ <tone_commander_settings schema_vers="1.1">
    <TCS_phone model="7810,7810 + 8030X">
        <sip_proxy>mantaray.undersea.com</sip_proxy>
        <sip_proxy_port>5060</sip_proxy_port>
        <sip_registrar>mantaray.undersea.com</sip_registrar>
        <sip_reg_port>5060</sip_reg_port>
        <sip_outbound_proxy>10.10.10.9</sip_outbound_proxy>
        <sip_outbound_proxy_port>5060</sip_outbound_proxy_port>

        <feature_activator_list>
            <fa index="1">
                <fa_type>LOCAL CFWD</fa_type>
            </fa>
            <fa index="2">
                <fa_type>LOCAL DND</fa_type>
            </fa>
        </feature_activator_list>

④ <multi_function_key_list>
        <key num="1" to_num="3">LINE
            <line_id>8005551212</line_id>
            <label>Doctor John</label>
            <sip_name>John Johnson</sip_name>
            <sip_auth_id>IMAuthorized</sip_auth_id>
            <sip_password>PassWord</sip_password>
        </key>
        <key num="4">SD
            <speeddial>4255551234#</speeddial>
            <label>SPDIAL 555-1234</label>
        </key>
        <key num="5">SD
            <speeddial>4255554321#</speeddial>
            <label>SPDIAL 555-4321</label>
        </key>
    </multi_function_key_list>
    </TCS_phone>

④ <TCS_phone model="7810">
        <multi_function_key_list>
            <key num="9">FA
                <fa_num>1</fa_num>
            </key>
            <key num="10">FA
                <fa_num>2</fa_num>
            </key>

        </multi_function_key_list>
    </TCS_phone>

```

④

```
<TCS_phone model="7810 + 8030X">
  <multi_function_key_list>
    <key num="29">FA
      <fa_num>1</fa_num>
    </key>
    <key num="30">FA
      <fa_num>2</fa_num>
    </key>
  </multi_function_key_list>

  <directory_list>
    <directory entry="1">
      <dir_name>Ron B</dir_name>
      <dir_number>4255556789</dir_number>
    </directory>
    <directory entry="2">
      <dir_name>Ella H</dir_name>
      <dir_number>4255556263</dir_number>
    </directory>
  </directory_list>
</TCS_phone>
```

③ </tone_commander_settings>

XML Tag Tables

All valid XML tags are listed in the tables below, grouped as follows:

- General Phone Settings
 - Feature Activators
 - Multifunction Keys
 - Line Keys
 - Feature Activator Keys
 - DSS Keys
 - ACD Keys
 - Speed Dial Keys
 - Call Directory Entries
 - Call Control Settings
- Network Settings
- SIP Settings
- MLPP Settings
- Update Settings

An alphabetical index of all XML tags is on page 47.

Phone and Dialing Settings

Settings in this section include user preference settings, feature activators, multifunction key assignments, call control, call directory, and dialing plans.

General Phone Settings	
XML Tag	Data / Description
<code><silence_suppress>on/off</silence_suppress></code>	Allows audio silence suppression. The purpose is to detect silence periods in the voice signal and temporarily discontinue transmission of the signal during the silence period. This is the same as Voice Activity Detection (VAD). ON OFF (default)
<code><voice_idle_disconnect>on/off</voice_idle_disconnect></code>	When this setting is ON, and the phone does not detect voice packets or a comfort noise packet, it will disconnect the phone after 20 seconds. ON (default) OFF
<code><dial_plan>dial plan definition</dial_plan></code>	Defines the phone's Dial Plan, as described on page 17. The default dial plan is blank (no dial plan).
<code><emergency_number>number</emergency_number></code>	Defines the phone's emergency number in order to recognize when an emergency outbound call is made. The default emergency number is blank (no number).

<p><blf_console_base> line appearance number </blf_console_base></p>	<p>Determines the starting line appearance for a Sylanro server-defined Busy Lamp Field (BLF). If this setting is DISABLED, the phone will ignore Sylanro server configuration messages for this feature. 2 –50 (button number), or 0 (disabled, default)</p>
<p><conference_key>conference type </conference_key></p>	<p>Allows the definition/assignment of a function to the CONF key on the phone. TCS LOCAL – conferences are implemented locally in the phone. (default) CENTRAL 1 – conferences are implemented using a central media server.</p>
<p><display_contrast>contrast level </display_contrast></p>	<p>This sets the phone display contrast. 1–8, default value = 4</p>
<p><ring_offhk_continuous>on/off </ring_offhk_continuous></p>	<p>Determines whether the phone rings normally, or only provides a single burst of ringing when the user is on a call. ON – off hook continuous ring. OFF – off hook single ring. (default)</p>
<p><ring_pref>ringing preference </ring_pref></p>	<p>Sets the action of the phone when answering a call by going off hook with the handset, pressing the speaker key, or pressing the headset key. NONE – user must press a line appearance key to draw dial tone. IDLE – dial tone is provided on the first available idle line appearance key. RING – if a call is ringing, it is answered. (default)</p>
<p><voice_mode_headset>on/off </voice_mode_headset></p>	<p>Allows the user to select the default voice path for phone calls. When set to ON, all calls are routed to the headset jack. ON OFF (default)</p>
<p><hand_rcv>volume level </hand_rcv></p>	<p>Sets the handset receive volume. 1 – 8, default value = 4</p>
<p><hand_xmit>volume level </hand_xmit></p>	<p>Sets the handset transmit gain. 1 – 8, default value = 4</p>
<p><head_rcv>volume level </head_rcv></p>	<p>Sets the handset receive volume. 1 – 8, default value = 4</p>
<p><head_xmit>volume level </head_xmit></p>	<p>Sets the headset transmit gain. 1 – 8, default value = 4</p>
<p><spkr_xmit>volume level </spkr_xmit></p>	<p>Sets the speakerphone transmit gain. 1 – 8, default value = 4 <i>(does not apply to TSG models)</i></p>
<p><flash_update>on/off </flash_update></p>	<p>Enables/disables flash update on the phone. ON OFF (default)</p>

Dial Plan

A Dial Plan allows the administrator to configure the phone so that specific dial patterns direct the phone to:

(General Functionality)

- Automatically initiate a call upon recognition of a specific dial pattern
- Add/Replace/Delete a specific prefix pattern
- Add/Replace/Delete a specific suffix pattern
- Generate secondary dialtone upon recognition of a specific dial pattern
- Block access to specific dial patterns

(MLPP Specific Functionality)

- Add URL parameters to the outbound INVITE
- Add the r-priority to the called party ID display
- Add a Resource Priority Header to the outbound INVITE
- Add a CAL Header to the outbound INVITE

A dial plan is stored as a text string; a null `<dial-plan>` does not affect user dialing in any way. The complete BNF (Backus-Naur Form) syntax that defines the structure for the dial plan is in Appendix B on page 39.

The dial plan is set by creating an xml entry which defines how the phone should process dialed strings. The tag for this entry is `<dial-plan>`.

```
<dial-plan>dial plan definition</dial-plan>
```

The dial plan definition is broken down into one or more “components” which provide specific dialing instructions. The components are separated by a “|”. (The “|” has other uses depending on where it is used. These cases will be explained in detail later in this section.) As each digit is dialed, it is evaluated by each component from left to right.

```
<dial-plan>component1|component2|componentN</dial-plan>
```

Each component can have three sections: a prefix operation, a dial pattern, and a suffix operation. Note the placement of the brackets.

```
<dial-plan>{prefix operation}dial string{suffix operation}</dial-plan>
```

Special Characters

- { } *(left & right brace/curly brace/curly bracket, ASCII 123 & 125)*
encloses prefix and suffix operations
- ^ *(circumflex accent/caret/hat, ASCII 94)*
wildcard, matches any dialed digit
- | *(vertical bar/vertical line, ASCII 124)*
separates components
in a prefix or suffix, indicates a substitution; characters to the left of the | are replaced by characters to the right
- ` *(grave accent/acute accent/back quote, ASCII 96)*
in a dial string, instructs the phone to produce a secondary dial tone
in a suffix, indicates a meta operation

Examples

The following is a sample of a dial plan with a single “component”. The following dial plan will cause the phone to automatically initiate a call when the user dials “911”. Also, at the time of call initiation, the phone replaces “911” with “2584357”. You will notice that there are “|” characters which are not acting to separate components, but are part of the prefix and suffix operations.

```
<dial_plan>{911|2584357}911#{|}</dial_plan>
```

The components in the above dial plan can be broken down as follows:

{911|2584357} This is a “prefix operation” and is enclosed within the {}. The bar in the prefix operation tells the phone that if the digits in front of the bar are dialed at the beginning of the string, they are to be replaced with the digits after the bar.

911# This is the dial string. If the digits dialed match this string, the phone performs the prefix and suffix operations defined in this component. If the “#” occurs at the end of the dial string, it tells the phone to initiate a call after receiving digits which match those defined in the dial string.

Valid characters for the dial string are:

```
A-Z a-z 0-9 . @ _ - / \ , : ; * ` " ( ) < > ~ + = ? ! $ % & (space)
```

{|} This is a “suffix operation” and is enclosed within the {}. The bar in the suffix operation tells the phone that if the digits in front of the bar are dialed as the last digits in the dialed string, they are to be replaced with the digits after the bar. Since there are no values entered, no operation is performed.

Another single component dial plan is shown below. This dial plan automatically initiates the call after seven characters, if the dialed string begins with a 3, 4, 5, or 9. Also, at the time of call initiation, the prefix “1425” is added to the beginning of the number.

```
<dial_plan>{|1425}[3-5,9]^^^^^#{|}</dial_plan>
```

{|1425} Since there are no digits in front of the bar in the prefix operation, the digits after the bar are added to the dialed digits, which match the dial string, when the phone sends them out.

[3-5,9] ^^^^^# A couple of new things are in the dialed string. The square brackets [] indicate that the string matches the dialed digits if any of the digits within the brackets are dialed in that position in the string. The dash indicates a range of characters, so **3-5** indicates that the digits 3, 4 or 5 are a match. If any of the digits 3, 4, 5 or 9 are dialed as the first digit, the string is matched. The ^ is a “wild card” which matches any dialed digit in that position in the dial string. In this case, the next 6 digits after the first can be any valid digit. Again the # terminator indicates that once you have a matching string, perform the prefix and suffix operations and then send the resulting digits.

{|} Like the previous example, no suffix operation is to be performed on dialed digits that match the dial string.

Combining the two previous examples results in the following dial plan.

```
<dial_plan>{911|2584357}911#{|}|{|1425}[3-5,9]^^^^^#{|}</dial_plan>
```

The two components **{911|2584357}911#{|}** and **#{|1425}[3-5,9]^^^^^#{|}** are separated by a “|”. If 911 is dialed by the phone user, 2584357 will be immediately dialed by the phone. The first component matches before enough digits are dialed for the second component have a matching string. If 9234567 is dialed, the first component will stop processing after the second digit and the second component will continue process the digits since they meet the requirements of the dial string. The digit string 14259234567 will be sent out. If 811 is dialed, the dial string in neither component is matched and the phone sends out 811 after the dial timeout or after the user enters # from the keypad.

```
<dial_plan>{||}^^`{||}|{||}771900^^^^^^!{||}|{77|}771^^^^^^^^^^^#{||}</dial_plan>
```

The above dial plan has 3 components. It introduces two more special characters: ` and !. The ` can have two meanings based on where it is located. The first is as a secondary dial tone indicator when in the dial string. The second is as a meta operation marker, which we will discuss later. The ! is a block access marker. If the dialed string matches the dial string for this component, the user gets reorder and the phone does not generate a call.

{||}^^`{||} The first component will cause the phone to generate dial tone after the user dials any two digits.

{||}771900^^^^^^!{||} The second component configures the phone so that for any call that begins with 771900 and has a total of 13 digits (digits shown plus 7 wild card digits), will be restricted.

{77|}771^^^^^^^^^^^#{||} The last component will for a dialed digit string that is 13 characters long and begins with a 77, strip off the 77 and immediately send the remaining string. So if a user dials 7718005551212, the phone will immediately send 18005551212.

MLPP Meta Operation Example

For a detailed explanation of meta operations, refer to the Appendix B on page 39.

```
<dial_plan>{9[0,2,4,6,8]|}9^{|`R0,CV99}</dial_plan>
```

This dial plan has one component.

{9[0,2,4,6,8]|} The phone will strip off the leading two digits if they are in the set {90 92 94 96 98}.

9^ This component applies to any dialed string beginning with a 9 as the first digit and any digit as the second. The string is not sent until the user presses softkey send, a “#” or the dial timeout occurs.

{|`R0,CV99} The ` identifies this as a meta operation string. This string will cause the phone to include a Resource Priority Header for the “dsn” network domain with r-priority identified by the second dialed character, and include a CAL header with variable mode and access-level 99.

Feature Activators

This section defines the list of available features and associated activators/deactivators that can be assigned to Feature Activator keys. The features must be supported by the Call Server platform.

All tags in the following table must be enclosed within the `<feature_activator_list>` `</feature_activator_list>` tag.

Feature Activators	
XML Tag	Data / Description
<code><fa_index="x"></code> <code></fa></code> x corresponds to the dial pad key used to select the Feature Activator from the phone menu: 1-9, 0, *, #	(All additional tags for a feature activator are enclosed within this tag.)
<code><fa_type>type</code> <code></fa_type></code>	Feature Activator type: GENERIC (feature is defined in the XML file) LOCAL CFWD (Call Forward, default phone feature) LOCAL DND (Do Not Disturb, default phone feature) Local features are defined in the phone, and do not require any additional parameters.
<code><description>description text</code> <code></description></code>	Text shown in the phone display when assigning a feature to a multifunction key, 24 characters max. This text will be used as a key's on-screen label if there is no <code><label></code> tag for the key.
<code><label>label text</code> <code></label></code>	Key Label, 18 characters max.
<code><prompt>prompt text</code> <code></prompt></code>	Prompt shown to the user when the feature is activated, if the feature requires a prompt, 18 characters max.
<code><activation>activation code</code> <code></activation></code>	Code used to activate the feature, if required; for example, *72 . Additional characters can be included after the code as required by the feature. Refer to the Feature Activation and Deactivation Codes section on page 21.
<code><deactivation>deactivation code</code> <code></deactivation></code>	Code used to deactivate the feature, if required. Refer to the Feature Activation and Deactivation Codes section on page 21.

Feature Activation and Deactivation Codes

Feature activation and deactivation codes are SIP server specific; refer to the server documentation.

Some features require different types of functionality such as additional dialed digits, establishment of another call, and control of the feature key LED. Additional characters can be added to the code string to enable this functionality. Following are some typical feature “star codes” with additional characters:

CALL FORWARD activate	*72# D I L	(dials additional digits, an auto-answered INVITE is sent, and illuminates the LED)
CALL FORWARD deactivate	*73# - I L	(dials additional digits, an auto-answered INVITE is sent, and turns off the LED)
CALL PARK	*98# D R -	(dials additional digits, transfers the call)
CALL UNPARK	*99# D C -	(dials additional digits, establishes a call)
DIRECTED CALL PICKUP	*53# D C -	(dials additional digits, establishes a call)
GROUP CALL PICKUP	*54# - C -	(does not require additional digits, establishes a call)

The star code must be followed by a # character.

The three character following the # are defined as follows:

Character 1 – capital letter **D** to require the user to dial additional digits
 - (hyphen) if no additional digits are required

Character 2 – capital letter **C** to establish a two-way audio call
 capital letter **I** to send an auto-answered INVITE
 capital letter **R** to transfer an existing call
 - (hyphen) if none of the above is required

Character 3 – capital letter **L** to illuminate the feature key LED for an 'activate' feature, or turn off the feature key LED for a 'deactivate' feature
 - (hyphen) indicates no action

Multifunction Keys

This section defines multifunction key assignments.

All tags in the following tables must be enclosed within the `<multi_function_key_list>` `</multi_function_key_list>` tag.

All tags for a single key must be enclosed within the `<key num="x">` `</key>` tag.

Key Number	
XML Tag	Data / Description
<code><key num="x">type</code> <code></key></code> x is the key number: 7810: 1-10 8030X: 11-40	Key Type: LINE FA (Feature Activator) DSS (Direct Station Selection) ACD (Automatic Call Delivery, only enter for the first key of an ACD pair) SD (Speed Dial) UNUSED
NOTE: A range of keys of the same type can be defined within a key tag.	
<code><key num="y" to_num="z">type</code> <code></key></code>	y is the first key number, z is the last key number

Line keys are part of the multifunction keys section.

Line Keys	
XML Tag	Data / Description
<code><line_id>ID</code> <code></line_id></code>	SIP Line ID (username or directory number), 128 characters max.
<code><sip_name>name</code> <code></sip_name></code>	SIP Display Name, 128 characters max.
<code><sip_auth_id>auth ID</code> <code></sip_auth_id></code>	SIP Authentication ID, 128 characters max.
<code><sip_password>password</code> <code></sip_password></code>	SIP Authentication Password, 128 characters max.
<code><bla_shared>bridged status</code> <code></bla_shared></code>	Bridged Line Appearance: ON (shared BLA) OFF (private)
<code><alert_auto_answer>on/off</code> <code></alert_auto_answer></code>	Enables auto answering (Sylantro servers only). When ON, users can turn Voice Announce auto answering on or off via a softkey. ON OFF (default)

<code><jitter_buffer_adaptive>buffer type</code> <code></jitter_buffer_adaptive></code>	Jitter Buffer Type: ON (adaptive) OFF (fixed)
<code><jitter_buffer_control>rate</code> <code></jitter_buffer_control></code>	Adaptive Jitter Buffer Rate of Change (only used if jitter_buffer_adaptive is ON): 1 – 12 , default value = 10
<code><jitter_delay_min>ms</code> <code></jitter_delay_min></code>	Minimum Jitter Buffer Delay (only used if jitter_buffer_adaptive is ON): 0 – 280 milliseconds, default value = 10
<code><jitter_delay_max>ms</code> <code></jitter_delay_max></code>	Maximum Jitter Buffer Delay (only used if jitter_buffer_adaptive is ON): 0 – 300 milliseconds, default value = 100
<code><codec1>codec type</code> <code></codec1></code>	1 st (Preferred) Codec Type: G711_10 G711_40 G729A_30 G711_20 G729A_10 G729A_40 G711_30 G729A_20 G723
<code><codec2>codec type</code> <code></codec2></code>	1 st (optional) Codec Type: G711_10 G711_40 G729A_30 G711_20 G729A_10 G729A_40 G711_30 G729A_20 G723 NONE
<code><codec3>codec type</code> <code></codec3></code>	2nd (optional) Codec Type: G711_10 G711_40 G729A_30 G711_20 G729A_10 G729A_40 G711_30 G729A_20 G723 NONE
<code><label>label text</code> <code></label></code>	Key Label, 18 characters max.
<code><tone_type>tone type</code> <code></tone_type></code>	Determines the ringing tone type for a line. 1 – 8 , default value = 5
<code><control_type>ringing type</code> <code></control_type></code>	Determines how a line will ring. ALWAYS (default) NEVER WAIT1 WAIT2 WAIT3 WAIT4 WAIT5 WAIT6 WAIT7

Feature Activator keys are part of the multifunction keys section.

Feature Activator Keys	
XML Tag	Data / Description
<code><fa_index>feature index</code> <code></fa_index></code>	Feature preset number; corresponds to the dial pad key used to select the feature when programming at the phone. 1 – 9, 0, *, #

DSS keys are part of the multifunction keys section.

DSS Keys	
XML Tag	Data / Description
<code><line_id>ID</code> <code></line_id></code>	SIP Line (username or directory number), 128 characters max.
<code><sip_name>name</code> <code></sip_name></code>	SIP Display Name, 128 characters max.
<code><sip_auth_id>auth ID</code> <code></sip_auth_id></code>	SIP Authentication ID, 128 characters max.
<code><sip_password>password</code> <code></sip_password></code>	SIP Authentication Password, 128 characters max.
<code><speeddial>dial string</code> <code></speeddial></code>	Speed Dial String, up to 128 characters
<code><jitter_buffer_adaptive>buffer type</code> <code></jitter_buffer_adaptive></code>	Jitter Buffer Type: ON (adaptive) OFF (fixed)
<code><jitter_buffer_control>rate</code> <code></jitter_buffer_control></code>	Adaptive Jitter Buffer Rate of Change (only used if jitter_buffer_adaptive is ON): 1 – 12 , default value = 10
<code><jitter_delay_min>ms</code> <code></jitter_delay_min></code>	Minimum Jitter Buffer Delay (only used if jitter_buffer_adaptive is ON): 0 – 280 milliseconds, default value = 10
<code><jitter_delay_max>ms</code> <code></jitter_delay_max></code>	Maximum Jitter Buffer Delay (only used if jitter_buffer_adaptive is ON): 0 – 300 milliseconds, default value = 100
<code><codec1>codec type</code> <code></codec1></code>	1 st (Preferred) Codec Type: G711_10 G711_40 G729A_30 G711_20 G729A_10 G729A_40 G711_30 G729A_20 G723

<code><codec2>codec type</code> <code></codec2></code>	1 st (optional) Codec Type: G711_10 G711_40 G729A_30 G711_20 G729A_10 G729A_40 G711_30 G729A_20 G723 NONE
<code><codec3>codec type</code> <code></codec3></code>	2nd (optional) Codec Type: G711_10 G711_40 G729A_30 G711_20 G729A_10 G729A_40 G711_30 G729A_20 G723 NONE
<code><label>label text</code> <code></label></code>	Key Label, 18 characters max.

ACD keys are part of the multifunction keys section.

ACD Keys	
XML Tag	Data / Description
<p><i>NOTE: The ACD feature requires two consecutive keys. Enter the following tags and attributes for the first key only.</i></p>	
<code><group_id>ACD Group ID</code> <code></group_id></code>	ACD Group ID, typically a 4-digit extension, 128 characters max.
<code><label_1>label text</code> <code></label_1></code>	Key Label for Login/Logout Key, 18 characters max.
<code><label_2>label text</code> <code></label_2></code>	Key Label for Activate/Deactivate Key, 18 characters max.

Speed Dial keys are part of the multifunction keys section.

Speed Dial Keys	
XML Tag	Data / Description
<code><speeddial>dial string</code> <code></speeddial></code>	Speed Dial digit/character string, 128 characters max.
<code><label>label text</code> <code></label></code>	Key Label, 18 characters max.

Call Directory

Up to 100 Call Directory entries are allowed. Ten entries are displayed on each telephone screen page. Tags for all Call Directory entries must be enclosed within the `<directory_list > </directory_list>` tag.

Call Directory Entries	
XML Tag	Data / Description
<pre><directory entry="x"> <dir_name>label text</dir_name> <dir_number>string</dir_number> </directory></pre> <p>x is the entry number, 1-100</p>	<p><u>dir_name</u>: displayed entry name 19 characters max.</p> <p><u>dir_number</u>: dial digit/character string, 128 characters max.</p>

Call Control Settings

These settings control dialing, ringing, and reorder tone timeout preferences.

Call Control Settings	
XML Tag	Data / Description
<pre><phone_dial_timeout>seconds </phone_dial_timeout></pre>	<p>Determines the amount of time the phone waits after a digit is dialed before sending the dialed digits to the SIP Call Server for call processing, if the user did not initiate sending by pressing the # key or 'SEND' soft key.</p> <p>A value of 0 (zero) allows an unlimited time for dialing with no timeout.</p> <p>0 – 30 seconds, default value = 10</p>
<pre><phone_max_ringing>seconds </phone_max_ringing></pre>	<p>Specifies the maximum number of seconds that the phone will persist in the ringing state for an incoming call.</p> <p>A value of 0 (zero) allows unlimited ringing with no timeout.</p> <p>0 – 300 seconds, default value = 0</p>
<pre><phone_max_reorder> seconds </phone_max_reorder></pre>	<p>Specifies the maximum number of seconds that the phone will persist in the reorder (disconnected or error) state.</p> <p>A value of 0 (zero) allows the phone to stay in the reorder state with no timeout.</p> <p>0 – 300 seconds, default value = 180</p>

Network Settings

Network Settings	
XML Tag	Data / Description
<code><phone_domain>domain name</code> <code></phone_domain></code>	<p>The domain name for the enterprise. For example, the “tonecommander.com” portion of the SIP URI user@tonecommander.com.</p> <p>Maximum length is 128 ASCII characters.</p> <p>The default is a NULL string.</p> <p><i>May be supplied by DHCP.</i></p>
<code><phone_subnet>subnet mask</code> <code></phone_subnet></code>	<p>Subnet mask of the phone.</p> <p>Valid IP address in xxx.xxx.xxx.xxx format, omit leading zeroes.</p> <p><i>May be supplied by DHCP.</i></p>
<code><phone_port>port number</code> <code></phone_port></code>	<p>UDP port number used by the phone to receive SIP signaling messages. The form is xxxxx with leading zeros suppressed.</p> <p>1025 – 65634, default value = 5060</p>
<code><phone_rtp_port>port number</code> <code></phone_rtp_port></code>	<p>RTP starting port used to send voice packets for the first line appearance. The form is xxxxx with leading zeros suppressed.</p> <p>The associated RTCP port is always the next port (add one) from the RTP port. Subsequent line appearances will automatically be assigned to the next even numbered port.</p> <p>1025 – 65634, default value = 1760</p>
<code><sip_proxy>address or name</code> <code></sip_proxy></code>	<p>IP address or server name for the SIP Proxy Server (Call Server).</p> <p>Valid IP address in xxx.xxx.xxx.xxx format, omit leading zeroes.</p> <p><i>May be supplied by DHCP.</i></p>
<code><sip_proxy_port>port number</code> <code></sip_proxy_port></code>	<p>UDP port number used by the phone to send SIP signaling messages to the Proxy Server. The form is xxxxx with leading zeros suppressed.</p> <p>1025 – 65634, default value = 5060</p>
<code><sip_outbound_proxy></code> <code>address or name</code> <code></sip_outbound_proxy></code>	<p>IP address or server name for the SIP Outbound Proxy.</p> <p>Valid IP address in xxx.xxx.xxx.xxx format, omit leading zeroes.</p>
<code><sip_outbound_proxy_port></code> <code>port number</code> <code></sip_outbound_proxy_port></code>	<p>UDP port number used by the phone to send SIP signaling messages to the Proxy Server. The form is xxxxx with leading zeros suppressed.</p> <p>1025 – 65634, default value = 5060</p>
<code><sip_registration>on/off</code> <code></sip_registration></code>	<p>Determines if registration with the SIP Registrar is required.</p> <p>If ON is selected, the phone will determine if an IP address is available for the Registrar Server, if so, it will attempt to register with it.</p> <p>If OFF is selected, registration is not attempted at all.</p> <p>ON (default) OFF</p>

<p><sip_registrar> <i>address or name</i> </sip_registrar></p>	<p>IP address or server name for the SIP Registrar Server. If left blank (or omitted) and registration enabled, registration with the Proxy Server will be attempted. Valid IP address in xxx.xxx.xxx.xxx format, omit leading zeroes.</p>
<p><sip_reg_port><i>port number</i> </sip_reg_port></p>	<p>UDP port number used by the phone to send SIP registration messages, in the form xxxxx with leading zeros suppressed. 1025 – 65634, default value = 5060</p>
<p><sip_backup_proxy> <i>address or name</i> </sip_backup_proxy></p>	<p>IP address or server name for the SIP Backup Server or Redundant SIP Proxy. Valid IP address in xxx.xxx.xxx.xxx format, omit leading zeroes.</p>
<p><sip_backup_proxy_port> <i>port number</i> </sip_backup_proxy_port></p>	<p>UDP port number used by the phone to send SIP signaling messages, in the form xxxxx with leading zeros suppressed. 1025 – 65634, default value = 5060</p>
<p><sip_backup_reg><i>on/off</i> </sip_backup_reg></p>	<p>Determines if registration with the Backup SIP Registrar or Proxy Server is required. If OFF is selected, registration is not attempted. ON OFF (default)</p>
<p><sip_backup_reg_id> <i>address or name</i> </sip_backup_reg_id></p>	<p>IP address or server name for the SIP Registrar Server. If left blank (or omitted) and registration enabled, registration with the Backup Proxy Server will be attempted. Valid IP address in xxx.xxx.xxx.xxx format, omit leading zeroes.</p>
<p><sip_backup_reg_port> <i>port number</i> </sip_backup_reg_port></p>	<p>UDP port number used by the phone to send SIP registration messages, in the form xxxxx with leading zeros suppressed. 1025 – 65634, default value = 5060</p>
<p><ip_dhcp_enable><i>on/off</i> </ip_dhcp_enable></p>	<p>Allows the DHCP Server to supply network settings. ON (default) OFF</p>
<p><ip_gateway_svr> <i>address or name</i> </ip_gateway_svr></p>	<p>IP address for gateway (router). Valid IP address in xxx.xxx.xxx.xxx format, omit leading zeroes. <i>May be supplied by DHCP.</i></p>
<p><ip_pri_dns_svr> <i>address or name</i> </ip_pri_dns_svr></p>	<p>IP Address of the Primary DNS Server. Valid IP address in xxx.xxx.xxx.xxx format, omit leading zeroes. <i>May be supplied by DHCP.</i></p>
<p><ip_sec_dns_svr> <i>address or name</i> </ip_sec_dns_svr></p>	<p>IP Address of the Secondary DNS Server. Valid IP address in xxx.xxx.xxx.xxx format, omit leading zeroes. <i>May be supplied by DHCP.</i></p>
<p><ip_syslog_svr> <i>address or name</i> </ip_syslog_svr></p>	<p>IP Address of the Syslog Server. Valid IP address in xxx.xxx.xxx.xxx format, omit leading zeroes. <i>May be supplied by DHCP.</i></p>
<p><ip_tftp> <i>address or name</i> </ip_tftp></p>	<p>IP address or server name for the TFTP Server. Valid IP address in xxx.xxx.xxx.xxx format, omit leading zeroes. <i>May be supplied by DHCP.</i></p>

<pre><time_server> address or name </time_server></pre>	<p>IP address for SNTP time server.</p> <p>Valid IP address in xxx.xxx.xxx.xxx format, omit leading zeroes.</p> <p>May be supplied by DHCP.</p>
<pre><time_offset>hours </time_offset></pre>	<p>Offset (in hours) from UTC or time server time.</p> <p>Pacific Standard Time = -8 Pacific Daylight Time = -7 Mountain Standard Time = -7 Mountain Daylight Time = -6 Central Standard Time = -6 Central Daylight Time = -5 Eastern Standard Time = -5 Eastern Daylight Time = -4</p> <p>+12 to -12, default value is 0</p> <p>May be supplied by DHCP.</p>
<pre><vlan> <enable>on/off</enable> <voice_pri>priority</voice_pri> <signal_pri>priority</signal_pri> <id>VLAN ID</id> <pc_id>VLAN ID</pc_id> </vlan></pre>	<p>VLAN / Ethernet Layer 2 802.1Q support.</p> <p><u>enable</u>: Ethernet Layer 2 802.1Q Support (ON or OFF).</p> <p>ON OFF (default)</p> <p><u>voice_pri</u>: 802.1Q Voice Packet Priority 0 – 7, default value = 6</p> <p><u>signal_pri</u>: 802.1Q Signaling Packet 0 – 7, default value = 6</p> <p><u>id</u>: 802.1Q VLAN ID 1 – 4094, default value = 0</p> <p><u>pc_id</u>: VLAN ID on the PC port 0 – 4095, default value = 0</p>
<pre><dscp_voice>DSCP value </dscp_voice></pre>	<p>Layer 3 DiffServ Voice Packet DSCP Value.</p> <p>0 – 63, default value = 46</p>
<pre><dscp_signal>DSCP value </dscp_signal></pre>	<p>Layer 3 DiffServ Signal Packet DSCP Value</p> <p>0 – 63, default value = 46</p>
<pre><telnet_access>on/off </telnet_access></pre>	<p>Enables/disables telnet access to the phone.</p> <p>ON OFF (default)</p>
<pre><ip_phone>address </ip_phone></pre>	<p>The IP Address of the phone.</p> <p>Valid IP address in xxx.xxx.xxx.xxx format, omit leading zeroes.</p> <p>May be supplied by DHCP.</p>

SIP Settings

SIP Settings	
XML Tag	Data / Description
<sip_reg_exp>seconds </sip_reg_exp>	The number of seconds that the registration is valid before the phone must re-register with the SIP Proxy. Valid range is 120 to 7200 seconds (2 minutes to 2 hours) in the form xxxx with leading zeros suppressed. default value = 3600 (1 hour)
<sip_sub_exp>seconds </sip_sub_exp>	The number of seconds that the subscription is valid before the phone must renew its subscription with the SIP Proxy. Valid range is 120 to 4,294,467,295 seconds (2 minutes to 136 years) with leading zeros suppressed. default value = 3600 (1 hour)
<sip_backup_reg_exp>seconds </sip_backup_reg_exp>	The number of seconds that the registration is valid before the phone must re-register with the backup SIP Proxy. Valid range is 120 to 7200 seconds (2 minutes to 2 hours) with leading zeros suppressed. default value = 3600 (1 hour)
<sip_attmpt_b4_backup> unsuccessful transmissions </sip_attmpt_b4_backup>	The number of unsuccessful transmissions before the phone switches to the Backup Proxy Server in the form xx with leading zeros suppressed. 2 to 10 , default value = 8
<sip_early_media>on/off </sip_early_media>	Allows the phone to play early media (audio) when receiving a SIP 180 response with SDP to an INVITE request. If OFF, any SDP that accompanies a SIP 180 response will be ignored by the phone; early media will not be played and local ringback will be generated. ON (default) OFF
<syslog_option>logging option </syslog_option>	Logging option for the Syslog Server operation. NONE – Disabled / No Logging (default) BASIC – SIP Phone Error Logs are sent to Syslog server QOS – Error Logs and Quality of Service call packet statistics sent
<session_timer>keep alive </session_timer>	Session timer keep alive mechanism is described in RFC 4028. REQUIRED (default) SUPPORTED UNSUPPORTED
<session_timer_interval>seconds </session_timer_interval>	The maximum amount of time that can occur between session refresh requests in a dialog before the session will be considered timed out. Valid range is 1800 to 4,294,467,295 seconds (30 minutes to 136 years) with leading zeros suppressed. default value = 3600 (1 hour)

<p><session_timer_min><i>seconds</i> </session_timer_min></p>	<p>The minimum value that the phone will accept for the session. This value cannot be larger than the <session_timer_interval>.</p> <p>Valid range is 90 to 4,294,467,295 seconds (90 seconds to 136 years) with leading zeros suppressed.</p> <p>default value = 1800 (30 minutes)</p>
<p><msg_sum_sub><i>subscribed</i> </msg_sum_sub></p>	<p>Determines whether the phone subscribes to message summary events.</p> <p>ON (default) OFF</p>
<p><line_mode><i>mode</i> </line_mode></p>	<p>Defines the phone's line mode.</p> <p>NORMAL (default) WARM HOT</p>
<p><hot_warm_username><i>string</i> </hot_warm_username></p>	<p>Defines the Hotline/Warmline username portion of Request-URI.</p> <p>This value is only used when <line_mode> is not set to NORMAL.</p> <p><i>string</i> is determined by the vendor platform.</p> <p>Example: AC-XS2001-HOT_WARM (For Acme XS 2001)</p>
<p><warmline_timeout><i>seconds</i> </warmline_timeout></p>	<p>Defines the Warmline dial timeout duration in seconds.</p> <p>2 – 30 seconds, default value = 3</p>
<p><sip_invite_timeout><i>seconds</i> </sip_invite_timeout></p>	<p>Defines the Invite timeout duration in seconds. This setting abandons a call if the called party is not detected during the timeout period.</p> <p>2 – 120 seconds, default value = 40</p>
<p><endpoint_mode><i>mode</i> </endpoint_mode></p>	<p>Defines the phone's endpoint mode. If this item is set to SIMPLE, then the endpoint method cannot be set to N/A.</p> <p>NORMAL (default) SIMPLE-INFO SIMPLE-TISPAN ADVANCED</p>
<p><phone_context><i>context string</i> </phone_context></p>	<p>Defines the default phone-context to be used when calling local numbers when the SIP URL type is "tel".</p> <p>The default setting is a zero length string.</p>
<p><prack><i>option</i> </prack></p>	<p>Provisional response acknowledgement (PRACK) is defined in RFC 3262.</p> <p>REQUIRED SUPPORTED (default) UNSUPPORTED</p>
<p><preconditions_tag><i>option</i> </preconditions_tag></p>	<p>The preconditions tag is defined in RFC 3312.</p> <p>REQUIRED SUPPORTED UNSUPPORTED (default)</p>

<p><rsrc_prty_tag><i>option</i> </rsrc_prty_tag></p>	<p>The resource-priority tag is defined in RFC 4412. REQUIRED SUPPORTED UNSUPPORTED (default)</p>
<p><sip_privacy_values><i>string</i> </sip_privacy_values></p>	<p>Defines a string up to 32 characters to be used for the SIP Privacy Header IAW RFC 3323.</p>
<p><sip_transport><i>SIP transport type</i> </sip_transport></p>	<p>Defines the type of SIP transport. UDP (default) TCP TLS</p>
<p><sip_url_type><i>URL type</i> </sip_url_type></p>	<p>Defines the type of URL that is transmitted in outbound INVITE messages. SIP (default) TEL</p>

MLPP Settings

The MLPP (Multilevel Precedence and Preemption) protocol allows high-priority users to place priority calls, and, if necessary, to preempt lower-priority phone calls.

MLPP Settings	
XML Tag	Data / Description
<pre><rph_namespace> <net_domain name="network1"> <priority>priority</priority> <priority>priority</priority> <priority>priority</priority> </net_domain> <net_domain name="network2"> <priority>priority</priority> <priority>priority</priority> <priority>priority</priority> </net_domain> </rph_namespace></pre> <p><i>network...</i> is an ASCII string that identifies the network domain.</p>	<p>Default Resource Priority Header (RPH) assignment; defines a list of RPH namespaces as defined in RFC 4412.</p> <p>Include an <net_domain> element for each network domain, up to 50. All network domain definitions are enclosed within the <rph_namespace> </rph_namespace> tags.</p> <p>The phones have two predefined namespaces:</p> <ol style="list-style-type: none"> name: "dsn" priorities: "0","2","4","6","8" name: "q735" priorities: "0","1","2","3","4" <p>Each network domain can have up to eight priorities. A priority can be up to 15 characters in length.</p> <p>To clear a network domain, do not include any priority elements.</p> <pre><net_domain name="name_to_remove"> </net_domain></pre> <p>To clear all network domains except the built-in defaults, enter "default" as the domain name.</p> <pre><net_domain name="default"> </net_domain></pre>
<pre><max_rph_prty_idx>index </max_rph_prty_idx></pre>	<p>Maximum RPH Priority index.</p> <p>0 – 7, default value = 2</p>
<pre><preempt_held_call_timeout> seconds </preempt_held_call_timeout></pre>	<p>When a held call is being preempted, this defines the amount of time before the held call is cleared.</p> <p>1 – 24 seconds, default value is 2.</p>
<pre><cal_access_level>level </cal_access_level></pre>	<p>The phone's fixed mode Confidential-Access-Level.</p> <p>0 – 99</p>
<pre><cal_matrix>integer list </cal_matrix></pre>	<p>An array of 100 integers comprising the phone's Confidential-Access-Level Matrix.</p> <p>This matrix is used for variable mode access-level resolution, where a requested access-level is used as an index into this array to obtain the reflected access-level.</p> <p>The default is an array whose first element is the value 0 and subsequent elements sequentially increase by one until the last value is 99. Each element of the array must be less than 100. Separate array elements with commas.</p> <p>An element of less than zero is valid in order to indicate a requested access level that cannot be resolved.</p> <p>0, 1, 2, ... 99</p>

<pre><cal_mode>mode </cal_mode></pre>	<p>The phone's Confidential-Access-Level Mode.</p> <p>NONE (default) FIXED VARIABLE</p>
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Update Settings

These settings control automatic updating of telephone software and configuration settings from a TFTP server.

Update Settings	
XML Tag	Data / Description
<pre><config_file_name_base> name base </config_file_name_base></pre>	<p>The naming scheme for custom XML configuration files, either Ethernet MAC address or phone line ID.</p> <p>Enter this tag in the "tcs7000a.xml" global configuration file.</p> <p>MAC (default) LINE</p>
<pre><ip_tftp_enable>on/off </ip_tftp_enable></pre>	<p>Allows automatic TFTP update of the phone.</p> <p>ON OFF (default)</p>
<pre><ip_tftp_enable>on/off </ip_tftp_enable></pre>	<p>Allows automatic TFTP update of the phone.</p> <p>ON OFF (default)</p>
<pre><tftp_update>update start hour </tftp_update></pre>	<p>TFTP update start hour. The automatic update process begins at a pseudo-random interval after this time each day.</p> <p>0 – 23, default value = 1</p>
<pre><tftp_window> update window hours </tftp_window></pre>	<p>TFTP update window duration in hours. The TFTP update time for each phone is calculated from tftp_window, tftp_update, and the phone's MAC address:</p> <p>This provides system-wide pseudo-randomly distributed TFTP start times at one-minute intervals between tftp_update (start time) and tftp_update + tftp_window (window hours later).</p> <p>The recommended TFTP window duration is one hour per 60 phones on a single TFTP server.</p> <p>1 – 24, default value = 3</p>

Telephone Software Updates

Installing Files on Server

Tone Commander may supply updates to the telephone operating software.

Copy the update files to the TFTP server root folder.

Update Configuration File

The file **tcs7000b.xml** names the compressed application and boot program files to be downloaded to the phones. The file is supplied with the updates.

Copy the update configuration file to the TFTP server root folder.

Appendix A

Viewing Packet Statistics

Packet statistics are tabulated on a per call, per call appearance basis, and may be viewed while a call is in progress or after a call is completed. Counts are updated once a second while a call is in progress. Statistics are saved for the most recent call on each call appearance. As soon as a new call (inbound or outbound) is initiated, statistics for the previous call on that call appearance are lost. Expected arrival times for packet delay calculations are based on the arrival time of the first packet in the call; this reference time is re-established on underflow and overflow events.

The last 50 packet statistics are saved in the packet history.

Packet statistics are viewed through the Packet Diagnostics menu (**Setup** → ADMIN → DIAG → PACKET). Select ACTIVE to view packet statistics for the currently active call, or HISTORY to view the last 50 recorded packet statistics

For the active call, the codec and jitter buffer settings are displayed first; jitter delay is displayed first when viewing the history. Press the Right or Left Arrow keys to view additional statistics.

A packet statistics summary can also be viewed for calls in the call log by selecting DETAIL.

The following statistics are recorded and displayed:

Codec/Jitter Buffer Settings – codec and associated jitter buffer selections for the current call (not shown in the history, but available for the current call and for calls in the call log). These values are negotiated on a per-call basis.

Jitter Delay – current “start of call” average jitter buffer setting (playout delay) for either fixed delay or calculated adaptive delay. The value displayed is based on the jitter buffer settings of the primary line.

Concealed Packets – total number of packets that were concealed during audio playback; also expressed as a percentage of total packets (concealed packets)/(total number of expected packets). This measurement is done at the audio playback point and correlates to audible dropouts in the voice path due to lost packets, packets received but delayed beyond the jitter buffer playback time, or jitter buffer underflow (no packets in the buffer). During packet concealment, the last received packet is replayed at a reduced level to minimize the audio interruption. Silence is played if multiple packets must be concealed.

Lost Packets – total number of expected packets that were not received; also expressed as a percentage of total packets (lost packets)/(total number of expected packets). Lost packets are computed by comparing the expected packet count (based on RTP packet sequence numbers) to the count of actual packets received. Lost packet counts are a result of network performance and cannot be improved by local jitter buffer settings.

Lost packets = (last RTP sequence number - first RTP sequence number) - number of packets received.

Not Delayed – total number of packets received earlier than 20ms after the expected arrival time; also expressed as a percentage of total packets (not delayed packets)/(total number of expected packets). These are normal packets that have average transmission delay, but with minimal jitter delay or packets that arrive early.

Delayed >80ms – total number of packets received later than 80ms after the expected arrival time; also expressed as a percentage of total packets (>80ms packets)/(total number of expected packets). Delayed packets may or may not be played, depending on jitter buffer settings.

Delayed >70ms – total number of packets received between 70ms and 80ms after the expected arrival time; also expressed as a percentage of total packets (70-80ms packets)/(total number of expected packets).

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Delayed >60ms – total number of packets received between 60ms and 70ms after the expected arrival time; also expressed as a percentage of total packets (60-70ms packets)/(total number of expected packets).

Delayed >50ms – total number of packets received between 50ms and 60ms after the expected arrival time; also expressed as a percentage of total packets (50-60ms packets)/(total number of expected packets).

Delayed >40ms – total number of packets received between 40ms and 50ms after the expected arrival time; also expressed as a percentage of total packets (40-50ms packets)/(total number of expected packets).

Delayed >30ms – total number of packets received between 30ms and 40ms after the expected arrival time; also expressed as a percentage of total packets (30-40ms packets)/(total number of expected packets).

Delayed >20ms – total number of packets received between 20ms and 30ms after the expected arrival time; also expressed as a percentage of total packets (20-30ms packets)/(total number of expected packets).

Underflow Events – total number of jitter buffer underflow events. An underflow occurs when the jitter buffer “runs dry”, usually due to an interruption in the packet stream. This causes an audible dropout in the audio playback until enough additional packets are received to fill the jitter buffer to the average value setting.

Overflow Events – total number of jitter buffer overflow events. An overflow sometimes occurs when a burst of packets arrives that exceeds the capacity of the jitter buffer. In this instance, the most recent packets are retained and the earliest packets in the jitter buffer are dropped to make room. This causes an audible “skip” in the audio playback to restore the jitter buffer contents to the average value setting. In some cases, an overflow event may follow an underflow event if a group of packets experience unusual burst delay. An overflow event can also occur on a long-duration call, due to slight differences in packet rates between sender and receiver.

Total Packets – total number of expected packets in the call, based on RTP sequence numbers (last received RTP packet sequence number) – (first received RTP packet sequence number). This number may be higher than the actual number of packets played during a call, since it also includes lost packets and underflow packets.

Appendix B

Dial Plan Syntax

The complete BNF syntax that defines the structure for the dial plan is as follows:

```
<dial-plan> ::= <component>*
<component> ::= <prefix-operation> <dial-pattern> <suffix-operation>
<prefix-operation> ::= "{" <subst-pattern-element>* "|" <dial-character>*}"
<subst-pattern-element> ::= <dial-character> | <wildcard> | <bracket-expression>
<dial-pattern> ::= <dial-pattern-element>+ [<dial-pattern-terminator>]
<dial-pattern-element> ::= <dial-character> | <wildcard> | <bracket-expression>
    | <secondary-dialtone-marker>
<dial-pattern-terminator> ::= <initiate-call-marker> | <block-access-marker>
<suffix-operation> ::= "{" <subst-pattern-element>* "|" <dial-character>* [<meta-operation>]"
<meta-operation> ::= <meta-operation-marker> <meta-operation-identifiers>
<meta-operation-marker> ::= "`"
<meta-operation-identifiers> ::= <meta-operation-identifier> ["," <meta-operation-identifier>]
<meta-operation-identifier> ::= <priority-id> | <rph-id> | <cal-id>
<priority-id> ::= "P" <priority-index>
<rph-id> ::= "R" <priority-index>
<priority-index> ::= (0 to 49)
<cal-id> ::= "CF" <access-level> | "CV" <access-level>
<access-level> ::= (0 to 99)
<dial-character> ::= <numeric> | <upper-case-alpha> | <lower-case-alpha> | <dial-symbol>
<wildcard> ::= "*"
<bracket-expression> ::= "[" <bracket-element> { "," <bracket-element> } "]"
<bracket-element> ::= <dial-character> | <dial-range>
<dial-range> ::= <dial-character> "-" <dial-character>
<secondary-dialtone-marker> ::= "`"
<initiate-call-marker> ::= "#"
<block-access-marker> ::= "!"
```

A null <dial-plan> does not affect user dialing in any way.

Dial plan components are evaluated from left-to-right until a dial pattern within a component is recognized or the last component is processed. Each <component> specifies a <prefix-operation>, a <dial-pattern>, and a <suffix-operation>. Note that <dial-character> is limited to the following ASCII characters:

```
A-Z a-z 0-9 . @ _ - / \ , : ; * ` " ( ) < > ~ + = ? ! $ % & (space)
```

The <dial-pattern> may contain any sequence of <dial-character>, <wildcard>, <bracket-expression>, or <secondary-dialtone-marker>, and it may end with a single <initiate-call marker> or <block-access-marker>. The <secondary-dialtone-marker> within the <dial-pattern> directs the phone to generate secondary dial tone, if the current dialed string entry matches the <dial-pattern> sub-string preceding the marker. An <initiate-call-marker> at the end of the <dial-pattern> directs the phone to automatically initiate a call, if the current dialed string entry matches the <dial-pattern> preceding the marker. A <block-access-marker> at the end of the <dial-pattern> directs the phone to immediately cancel the outbound call with reorder tone, if the current dialed string entry matches the <dial-pattern> preceding the marker.

Prefix and suffix operations are applied at the time of call initiation, and only if the <dial-pattern> matches the dialed string entry.

If the leading characters of the dialed string entry matches the prefix pattern preceding the "|", then the leading characters of the dialed string entry are replaced by the substring following the "|". If the prefix pattern preceding the "|" is null, then the substring following the "|" are inserted at the beginning of the

dialed string entry. If the substring following the “|” is null, then the leading characters of the dialed string entry are deleted.

If the trailing characters of the dialed string entry match the suffix pattern preceding the “|”, then the trailing characters of the dialed string entry are replaced by the substring following the “|” and preceding any `<meta-operation>`. If the suffix pattern preceding the “|” is null, then the substring following the “|” and preceding any `<meta-operation>` are inserted at the end of the dialed string entry. If the substring following the “|” and preceding any `<meta-operation>` is null, then the trailing characters of the dialed string entry are deleted.

Meta Operations

Meta operations construct message headers that meet MLPP requirements.

The `<meta-operation>` includes the following combinations:

- `<priority-id>`
- `<rph-id>`
- `<cal-id>`
- `<prio-id>`, `<cal-id>`
- `<rph-id>`, `<cal-id>`

If the `<suffix-operation>` includes a `<priority-id>`, then the phone displays the r-priority identified by the second digit of the dialed string entry. If the `<suffix-operation>` includes an `<rph-id>`, then the phone displays the r-priority and includes a Resource Priority Header (RPH) in the outgoing INVITE. The RPH includes the network domain identified by the `<priority-index>` and the r-priority identified by the second digit of the dialed string entry. If the `<suffix-operation>` includes a `<cal-mode-id>`, then the phone includes a Confidential Access Level (CAL) header in the outgoing INVITE. The CAL header includes the mode identified by the `<cal-mode-id>` and the access level identified by the `<access-level>`.

Appendix C Troubleshooting

Power-up & Connection Troubleshooting

Whenever power is applied or a connection is made to the LAN or WAN, the phone initiates a startup routine, with progress shown in the display. When the phone and network are fully initialized, the idle display, indicating date and time, will be shown. In cases where full initialization is not attained, the following displays or conditions will be shown continuously until corrected.

Problem Observed	Remedial Action
<i>No display information is shown</i>	Check power connections and source.
NO ETHERNET CONNECTION	Check connections to the LAN or WAN.
LINE ID= DELETE CLEAR 123	The Line ID is a null value. Enter the appropriate Line ID number.
DHCP ERROR RETRYING PRESS SETUP TO CONFIGURE	Verify that the DHCP server is operating and accessible. If the LAN or WAN does not include a DHCP server, disable IP configuration via DHCP and enter the appropriate IP values (phone, default gateway, subnet mask, TFTP) using the INSTL/IP Menu.
PHONE= CLEAR	The phone IP address is a null value. Enter the appropriate phone IP address or name.
GATEWAY= CLEAR	The default gateway IP address is a null value. Enter the appropriate gateway IP address or server name.
SUBNET=000.000.000.000 CLEAR	The subnet mask is a null value. Enter the appropriate subnet mask.
TFTP= CLEAR	The TFTP IP address is a null value. Enter the appropriate TFTP IP address or server name.
AUTH ID= DELETE CLEAR 123	The AUTH ID is a null value. Enter the appropriate AUTH ID.
AUTH PSWD= DELETE CLEAR 123	The AUTH PSWD (password) is a null value. Enter the appropriate AUTH password.
PROXY= DELETE CLEAR 123	The Proxy server address is a null value. Enter the appropriate Proxy server address or server name.
REGISTERING LINES	The primary line has not registered with the SIP server. Verify all entries (LINE ID, AUTH ID, AUTH Password, all IP addresses and subnet mask) and re-enter as required.
REGISTERING LINES CONTINUE	The primary line has registered with the SIP server; however, additional lines have not. Verify the LINE ID, AUTH ID and AUTH passwords for all additional lines and re-enter as required.

Problem Observed	Remedial Action
RESTART WITH NEW VALUES? YES	Certain critical values have changed and a restart is required. Select YES.
PHY:100MBPS IP:DHCP FAIL	Upon lease expiry, the phone was unable to negotiate a new lease with the DHCP server. Verify that the DHCP server is operating and accessible.
PING UNSUCCESSFUL PING	If all attempts to ping IP addresses fail, check Layer 2 802.1Q (VLAN) programming at the phone, using the QoS menu. If "LAYER 2 802.1Q=ON", verify that the network supports this packet prioritization standard. If it does not, set LAYER 2 802.1Q to "OFF". Ping valid addresses using the ADMIN→DIAG→PING Menu.

Call Control Troubleshooting

After the phone is fully initialized (idle display showing), the following call control anomalies may be encountered.

Problem Observed	Remedial Action
Going off-hook does not automatically select a line key and draw dial tone. Selecting a line key while idle does, however.	This is the behavior of a phone that is programmed with "Call Preference=NONE". If this behavior is not desired, change the value of the attribute to either "RING" or "IDLE", using the USER→PREF Menu.
All lines indicate the arrival of inbound calls via the key indicators; however, the phone does not ring for some of them.	Verify that the affected lines are not set for NEVER or an extended WAIT interval, using the USER→RING→CONTROL Menu. If this behavior is not desired, change the value of the attribute to "ALWAYS" where applicable.
All lines indicate the arrival of inbound calls via the key indicators; however, the phone never rings.	Verify that "RINGER OFF" is not showing in the display. If it is, use the Volume Up key to set the ringer level to a value higher than the OFF.

Diagnostic Troubleshooting

After the phone is fully initialized (idle display showing), the following diagnostic information may be reviewed, using the ADMIN/DIAG Menu.

Diagnostic Mode	Information Provided
LINK option selected:	<p>The LINK option provides information about the network and SIP server status as follows:</p> <p>1) Status about the network will be displayed: PHY:100MBPS IP:DHCP OK <i>(DHCP used for IP addressing at phone)</i> or PHY:100MBPS IP:NO DHCP <i>(fixed IP addresses are in use at phone)</i></p> <p>2) Registration status for each line will be indicated via line status LED for 2 seconds.</p> <p>Solid Green – Line registration successful with server. Alternating red/green – Line registration in process and undeterminate. Solid Red – Line registration failed. Flashing red – The primary line is active, and secondary line(s) are not registered.</p>

Contacting Technical Support

If you need assistance configuring your VoIP phones, contact Tone Commander Customer Technical Support.

Tone Commander
Technical Support Department
11609 49th Place West
Mukilteo, WA 98275-4255 USA

Phone: (425) 349-1000
(800) 524-0024

Fax: (425) 349-1010

E-mail: tech@tonecommander.com

Web: www.tonecommander.com

Tone Commander is committed to meeting the product needs of our customers. Please write or call us with any suggestions for improvement.

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