APPENDIX D

Configurable Parameters for the SIP IP Phone

This appendix describes configurable SIP parameters in the SIPDefault.cnf file and the SIP IP phone. Parameters are in alphabetical order, except in Table D-4, which lists options in the order that they appear on the phone. Optional parameters are so noted.

Table D-1 Configuration File Parameters

Parameter	Description
anonymous_call_block	(Optional) Configures anonymous call block. Valid values are as follows: • 0—Disabled by default, but can be turned on and off using the user interface. When disabled, anonymous calls are received. • 1—Enabled by default, but can be turned on and off using the user interface. When enabled, anonymous calls are rejected • 2—Disabled permanently and cannot be turned on and off locally using the user interface. Specify this parameter in the phone-specific configuration file. • 3—Enabled permanently and cannot be turned on and off locally using the user interface. Specify this parameter in the phone-specific configuration file. Default is 0.
auto_answer	(Optional) Configures the intercom functionality so that the user can define one or more of their lines for this feature. It is an integer field that represents a bit mask of on or off for each line key. The bit mask reads from the least significant bit of 0 equal to line 1 to the most significant bit of 5 for line 6. Valid values are as follows: • Cisco IP 7960 phone— 0 to 63 • Cisco IP 7940 phone— 0 to 3 **Note** This parameter *cannot* be set in the configuration file.
autocomplete	(Phone-specific; optional) Configures automatic completion of numbers. Valid values are 0 (disable autocompletion) and 1 (enable autocompletion). Default is 1.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
call_hold_ringback	(Phone-specific; optional) If you have a call on hold and are talking on another call, when you hang up the call, this parameter causes the phone to ring, letting you know that you still have another party on hold. Valid values are as follows: • 0—Off by default, but can be turned on and off locally using the user interface. • 1—On by default, but can be turned on and off locally using the user interface. • 2—Off permanently and cannot be turned on and off locally using the user interface. Specify this parameter in the phone-specific configuration file. • 3—On permanently and cannot be turned on and off locally using the user interface. Specify this parameter in the phone-specific configuration file. Default is 0.
call_stats	(Optional) Includes RTP statistics in BYE requests and responses. Valid values are 0 (disable) and 1 (enable). Default is 0. If this parameter is enabled, the phone inserts the headers RTP-RxStat and RTP-TxStat as follows: • RTP-RxStat: Dur=a,Pkt=b,Oct=c,LatePkt=d,LostPkt=e,AvgJit=f • RTP-TxStat: Dur=g,Pkt=h,Oct=i where the following apply: • Dur—Total number of seconds since the beginning of reception or transmission. • Pkt—Total number of RTP packets received or transmitted. • Oct—Total number of RTP payload octets received or transmitted (not including RTP header). • LatePkt—Total number of late RTP packets received. • LostPkt—Total number of lost RTP packets received (not including late RTP packets). • AvgJit—Average jitter, which is an estimate of the statistical variance of the RTP packet inter-arrival time, measured in timestamp unit and calculated according to RFC 1889. • a, b, c, d, e, f, g, h, and i—Integers.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
call_waiting	(Phone-specific; optional) Configures call waiting. Valid values are as follows: • 0—Disabled by default, but can be turned on and off using the user interface. When disabled, call waiting calls are not received. • 1—Enabled by default, but can be turned on and off using the user interface. When enabled, call waiting calls are accepted. • 2—Disabled permanently and cannot be turned on and off locally using the user interface. Specify this parameter in the phone-specific configuration file. • 3—Enabled permanently and cannot be turned on and off locally using the user interface. Specify this parameter in the phone-specific configuration file. Default is 1.
callerid_blocking	(Phone-specific; optional) Configures caller ID blocking. When enabled, the phone blocks its own number or e-mail address from phones that have caller identification enabled. Valid values are as follows: • 0—Disabled by default, but can be turned on and off using the user interface. When disabled, the caller identification is included in the Request-URI header field. • 1—Enabled by default, but can be turned on and off using the user interface. When enabled, “Anonymous” is included in place of the user identification in the Request-URI header field. • 2—Disabled permanently and cannot be turned on and off locally using the user interface. Specify this parameter in the phone-specific configuration file. • 3—Enabled permanently and cannot be turned on and off locally using the user interface. Specify this parameter in the phone-specific configuration file. Default is 0.
cfwd_url	(Optional) Configures the call forwarding feature. The maximum allowable characters for the string is 128. The character can be a telephone number or a URL. **Note** This parameter *cannot* be set in the configuration file.
cnf_join_enable	(Optional) Whether the conference bridge, when it hangs up, should attempt to join the two leaf nodes. Valid values are as follows: • 0—Do not join the two leaf nodes. • 1—Join the two leaf nodes. Default is 1.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
date_format	(Optional) Format for dates. Valid values are as follows: • M/D/Y—Month/day/year • D/M/Y—Day/month/year • Y/M/D—Year/month/day • Y/D/M—Year/day/month • Y-M-D—Year-month-day • YY-M-D—4-digit year-month-day Default is M/D/Y.
dial_template	Template with which you specify which file to download for your dial plan.
directory_url	(Optional) URL of the external directory server. This URL is accessed when you select Directory > External Directory. For example, use directory_url: "http://10.10.10.10/CiscoServices/Directory.asp".
dnd_control	(Phone-specific; optional) Sets the Do Not Disturb (DND) feature. Valid values are as follows: • 0—Off by default, but can be turned on and off locally using the user interface. • 1—On by default, but can be turned on and off locally using the user interface. The phone blocks all calls placed to the phone and logs those calls in the Missed Calls directory. • 2—Off permanently and cannot be turned on and off locally using the user interface. Specify this parameter in the phone-specific configuration file. • 3—On permanently and cannot be turned on and off locally using the user interface. This setting sets the phone to be a call-out phone only. Default is 0.
Domain Name	Name of the DNS domain in which the phone resides.
dst_auto_adjust	(Optional) Whether daylight savings time (DST) is automatically adjusted on the phones. Valid values are 0 (do not adjust) and 1 (adjust). Default is 1.
dst_offset	(Optional) Offset from the phone time when DST is in effect. When DST is over, the specified offset is no longer applied to the phone time. Valid values are hour/minute, –hour/minute, +hour/minute, hour, –hour, and +hour.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
dst_start_day	(Optional) Day of the month on which DST begins. Valid values are the following: • 1 to 31 (for days of the month) • 0 (ignore this field and use the value in the dst_start_day_of_week parameter instead) Default is 0.
dst_start_day_of_week	(Optional) Day of the week on which DST begins. Valid values are as follows: • Any of the following: Sunday or Sun, Monday or Mon, Tuesday or Tue, Wednesday or Wed, Thursday or Thu, Friday or Fri, Saturday or Sat, Sunday or Sun • 1 to 7 (1 is Sunday; 7 is Saturday) Name of the day is not case-sensitive. In the United States, the default is Sunday.
dst_start_month	(Optional) Month in which DST starts. Valid values are the following: • January, February, March, April, May, June, July, August, September, October, November, and December • 1 to 12, with 1 being January and 12 being December When the name of a month is specified, the value is not case-sensitive. In the United States, the default is April.
dst_start_time	(Optional) Time of day on which DST begins. Valid values are hour/minute (02/00) or hour:minute (02:00). In the United States, the default is 02:00.
dst_start_week_of_month	(Optional) Week of month in which DST begins. Valid values are 1 to 6 and 8 (1 is the first week; each subsequent number is a subsequent week; 8 is the last week in the month regardless of which week the last week is). In the United States, the default is 1.
dst_stop_day	(Optional) Day of the month on which DST ends. Valid values are as follows: • 1 to 31 (for the days of the month) • 0 (ignore this field and use the value in the dst_stop_day_of_week parameter instead) Default is 0.
dst_stop_day_of_week	(Optional) Day of the week on which DST ends. Valid values are Sunday or Sun, Monday or Mon, Tuesday or Tue, Wednesday or Wed, Thursday or Thu, Friday or Fri, Saturday or Sat, Sunday or Sun, and 1 to 7, with 1 being Sunday and 7 being Saturday. When the name of the day is specified, the value is not case-sensitive. In the United States, the default is Sunday.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
dst_stop_month	(Optional) Month in which DST ends. Valid values are January, February, March, April, May, June, July, August, September, October, November, and December or 1 to 12 with 1 being January and 12 being December. When the name of a month is specified, the value is not case-sensitive. In the United States, the default is October.
dst_stop_time	(Optional) Time of day on which DST ends. Valid values are hour/minute (02/00) or hour:minute (02:00). In the United States, the default is 02:00.
dst_stop_week_of_month	(Optional) Week of month on which DST ends. Valid values are 1 to 6 and 8, with 1 being the first week, each number thereafter being subsequent weeks, and 8 being the last week in the month regardless of which week the last week is. In the United States, the default is 8.
dtmf_avt_payload	(Optional) Configures the payload type for Audio/Video Transport (AVT) packets. Range is from 96 to 127. If the value specified is null or invalid, default is 101.
dtmf_db_level	(Optional) In-band DTMF digit tone level. Valid values are as follows: • 1—6 dB below nominal • 2—3 dB below nominal • 3—nominal • 4—3 dB above nominal • 5—6 dB above nominal Default is 3.
dtmf_outofband	(Optional) Configures the out-of-band signaling (for tone detection on the IP side of a gateway). **Note** The Cisco SIP IP phone supports out-of-band signaling using the AVT tone method. Valid values are as follows: • none—Do not generate DTMF digits out-of-band. • avt—If requested by the remote side, generate DTMF digits out-of-band (and disable in-band DTMF signaling); otherwise, do not generate DTMF digits out-of-band. • avt_always—Always generate DTMF digits out-of-band. This option disables in-band DTMF signaling. Default is avt.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
dyn_dns_addr_1	(Optional) IP address of a new dynamic DNS server. If a new DNS server address is specified, it is used for any further DNS requests after the phone uses the initial DNS address upon bootup. The DNS addresses are used in the following order: 1. dyn_dns_addr_1 (if present) 2. dyn_dns_addr_2 (if present) 3. DNS Server 1 4. DNS Server 2 5. DNS Server 3 6. DNS Server 4 7. DNS Server 5 The dynamic DNS address is not stored in flash memory. Only dotted IP addresses are accepted. This value can be cleared by removing it from the configuration file or by changing its value to a null value “ ” or to “UNPROVISIONED.” **Note** The dynamic DNS address is not stored in flash memory.
dyn_dns_addr_2	(Optional) IP address of a second dynamic DNS to be used for DNS requests. See dyn_dns_addr_1 for more information.
dyn_tftp_addr	(Optional) IP address of a new dynamic TFTP server. After initially querying the default TFTP server, the phone rerequests the default and phone-specific configuration files from the new TFTP server. The dynamic TFTP server address is not stored in flash memory. The number of dyn_tftp_addr values supported by the phone is limited to prevent the phone configuration being downloaded repeatedly from multiple TFTP servers. Only dotted IP addresses are accepted. This value can be cleared by removing it from the configuration file or by changing its value to a null value “ ” or to “UNPROVISIONED.”
enable_vad	(Optional) Enables voice activation detection (VAD). Valid values are 0 (disable) and 1 (enable). Default is 0.
end_media_port	(Optional) Configures the Real-Time Transport Protocol (RTP) end range for media. Valid values are 16384 to 32766. Default is 32766.
garp_enable	(Optional) Enables Gratuitous ARP for the phone. Valid values are as follows: • 0—GARP is disabled. • 1—GARP is enabled.
Host Name	Unique host name assigned to the phone. The value in this field is always SIP*mac* where *mac* is the MAC address of the phone.
http_proxy_addr	(Optional) IP address of the HTTP proxy server. You can use either a dotted IP address or a DNS name.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
http_proxy_port	(Optional) Number of the HTTP proxy port. Default is 80.
image_version	Firmware version that the phone should use. Enter the name of the image version as it is released by Cisco. Do not enter the filename extension (.bin). **Note** You cannot change the image version by changing the filename because the version is also built into the file header. Trying to change the image version by changing the filename causes the firmware to fail when it compares the version in the header against the filename.
language	(Optional) This parameter is for future use. English is the only value that is currently supported.
line*x*_authname	(Phone-specific; optional) Name used by the phone for authentication if a registration is challenged by the proxy server during initialization. It is required only if a proxy server requires authentication from phones. If a value is not configured for the line*x*_authname parameter when registration is enabled, the default name is used. Default is UNPROVISIONED. The *x* argument can be 1 or 2.
line*x*_displayname	(Phone-specific; optional) Identification as it should appear for caller-identification purposes. For example, instead of jdoe@company.com appearing on phones that have caller ID, you can specify User A in this parameter to have User A appear on the called party display. If a value is not specified for this parameter, nothing is used. The *x* argument can be 1 or 2.
line*x*_name	(Phone-specific) Number or e-mail address for use when registering. Enter a number without dashes. For example, enter 555-0100 as 5550100. Enter an e-mail ID without the host name. The *x* argument can be 1 or 2.
line*x*_password	(Phone-specific; optional) Password used by the phone for authentication if a registration is challenged by the proxy server during initialization. If a value is not configured for the line*x*_password parameter when registration is enabled, the default password is used. Default is UNPROVISIONED. Valid values for *x* are (Cisco IP 7969G phone) 1 to 6 and (Cisco IP 7940G phone) 1 to 2.
line*x*_shortname	Labels a line key with a name other than the directory number.
local_cfwd_enable	Whether the phone can do local call forwarding. This is a boolean field; it is either enabled or disabled. Valid values are 0 (disable) and 1 (enable). Default is 1.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
logo_url	(Optional) Location of the company logo file. This logo appears on the phone display. The background space allocated for the image is 90 x 56 pixels. Images that are larger than this will automatically be scaled down to 90 x 56 pixels. The recommended file size for the image is from 5 to 15 Kb. For example, use logo_url: "http://10.10.10.10/companylogo.bmp". **Note** This parameter supports Windows 256 color bitmap format only. CMXML PhoneImage objects are not supported for this parameter. Using anything other than a Windows bitmap (.bmp) file can cause unpredictable results.
messages_uri	(Optional) Configures the voice-mail number that is dialed when the messages button is pressed. Value is typically a phone number but can be a URI.
mwi_status	(Optional) Displays the message waiting status. **Note** You *cannot* set this parameter in the configuration file.
nat_address	(Optional) WAN IP address of the Network Address Translation (NAT) or firewall server. Value is either a dotted IP address or a DNS name.

nat_enable	(Optional) Enables NAT.
	Valid values are 0 (disable) and 1 (enable). Default is 0.
	• If NAT is enabled, the Contact header appears as follows:
	`Contact: sip:lineN_name@nat_address:voip_control_port`
	If the nat_address is invalid or UNPROVISIONED, the Contact header appears as follows:
	`Contact:`
	`sip:lineN_name@phone_ip_address:voip_control_port`
	and the Via header appears as follows:
	`Via: SIP/2.0/UDP phone_ip_address:voip_control_port`
	• If NAT is enabled, the Session Description Protocol (SDP) message uses the nat_address and an RTP port between the start_media_port and the end_media_port range in the C and M fields. All RTP traffic is sourced from the port advertised in the SDP message.

nat_received_processing	(Optional) Enables NAT received processing. Valid values are 0 (disable) and 1 (enable). Default is 0. If nat_received_processing is enabled, and the received= tag is in the Via header of the 200 OK response from a REGISTER, the IP address in the received= tag is used instead of the nat_address in the Contact header. If this switch occurs, the phone unregisters the old IP address and reregisters with the new IP address.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
network_media_type	(Optional) Ethernet port negotiation mode. Valid values are as follows: • Auto—Port is autonegotiated. • Full100—Port is configured to be a full-duplex, 100-MB connection. • Half100—Port is configured to be a half-duplex, 100-MB connection. • Full10—Port is configured to be a full-duplex, 10-MB connection. • Half10—Port is configured to be a half-duplex, 10-MB connection. Default is Auto.
network_port2_type	(Optional) Configures the device type that is connected to port 2 of the phone. Valid values are Hub/Switch and PC. Default is Hub/Switch. **Note** If the value is PC, port 2 can be connected only to a PC. If you are not sure about the connection, use the Hub/Switch default value. Specifying the PC option and then connecting port 2 to a switch results in spanning-tree loops and network confusion.
outbound_proxy	(Optional) IP address of the outbound proxy server. You can use either a dotted IP address or a DNS name.
outbound_proxy_port	(Optional) Port number of the outbound proxy server. Default is 5060. When an outbound proxy is enabled, all SIP requests are sent to the outbound proxy server instead of to the proxy*x*_address. All responses continue to reconcile the normal Via processing rules. The media stream is not routed through the outbound proxy. NAT and outbound proxy modes can be independently enabled or disabled. The received= tag is added to the Via header of all responses if there is no received= tag in the uppermost Via header and if the source IP address is different from the IP address in the uppermost Via header. Keep the following rules in mind: • If a received= tag is in the uppermost Via header, the response is sent back to the IP address contained in the received= tag. • If there is no received= tag and the IP address in the uppermost Via header is different from the source IP address, the response is sent back to the source IP address.
phone_label	(Phone-specific; optional) Text to display on the top right status line of the LCD. This field is for end-user display only and has no effect on caller identification or messaging. For example, a phone label can display “User A’s phone.” Limited to 11 characters.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
phone_password	(Phone-specific; optional) Password to be used for console or Telnet access. Limited to 31 characters. Default is cisco.
phone_prompt	(Phone-specific; optional) Prompt to display during Telnet or console access. Limited to 15 characters. Default is SIP Phone.
preferred_codec	(Optional) Codec to use when a call is initiated. Valid values are g711alaw, g711ulaw, g729a, and none. Default is g711ulaw.
proxy_backup	(Optional) IP address of the backup proxy server or gateway. Enter this address in IP dotted-decimal notation. **Note** You must specify at least one address or the phones cannot register.
proxy_backup_port	(Optional) Port number of the backup proxy server. Default is 5060.
proxy_emergency	(Optional) IP address of the emergency proxy server or gateway. Enter this address in IP dotted-decimal notation.
proxy_emergency_port	(Optional) Port number of the emergency proxy server. Default is 5060.
proxy_register	(Optional) The phone must register with a proxy server during initialization. Valid values are 0 (disable registration during initialization) and 1 (enable registration during initialization). Default is 0. **Note** You can also use this parameter in a phone-specific configuration file. After a phone has initialized and registered with a proxy server, you can remove the registration by changing this value to 0 in the phone-specific configuration file. To reinitiate registration, change the value back to 1. **Note** If you enable registration and authentication is required, you must specify values for the line*x*_authname and line*x*_password parameters in the phone-specific configuration file.
proxy*x*_address	IP address of the SIP proxy servers that are used by the phones. Enter the addresses in IP dotted-decimal notation or use the FQDN. The "*x*" argument is representative of server addresses. Valid values for "*x*" are 1 to 6. If the proxy*x*_address parameter is provisioned with an FQDN, the phone sends REGISTER and INVITE messages by using the FQDN in the Req-URI, To, and From fields. If the value of x is not specified in the proxy*x*_address parameter, the phone uses proxy1_address as the default value.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
proxy*x*_port	Port number of the SIP proxy server that will be used by phone lines other than line 1. The *x* variable represents a phone line. Valid values are 2 to 6. **Note** For additional phone lines, the proxy*x*_port parameter and the proxy*x*_port parameter can be used to assign different proxy addresses to different phone lines. The "x" in the parameters represents a phone line. The value of "x" can be from 1 to 6. If the value of the "x" is not specified in the proxy*x*_address parameter, the phone uses proxy1_address as the default.
remote_party_id	(Optional) The Remote-Party-ID header supports network verification and screening of a call participant's identity (for example, name and number) and provides privacy for call participants. Valid values are as follows: • 0—Remote party ID is disabled. The phone does not send or accept the remote party ID. • 1—Remote party ID is enabled. The phone sends the remote party ID, and can accept the remote party ID. Default is 0.
rfc_2543_hold	(Optional) Determine the SDP that a phone uses to place a remote party on hold. If this value is 1, the phone uses the RFC 2543 method and sets the media address to 0.0.0.0. If this value is 0, the phone uses the RFC 3264 style and instructs the other side to be in recvonly mode. Default is 0.
semi_attended_transfer	(Optional) Whether or not the caller can transfer the second leg of an attended transfer while the call is ringing. Valid values are as follows: • 0—Semi-attended transfer is disabled. • 1—Semi-attended transfer is enabled. Default is 1.
services_url	(Optional) URL of the services BTXML files. This URL is accessed when the Services button is pressed. For example, use services_url: "http://10.10.10.10/CiscoServices/Services.asp."
sip_invite_retx	(Optional) Maximum number of times that an INVITE request will be retransmitted. Valid value is any positive integer. Default is 6.
sip_max_forwards	(Optional) The phone uses the value specified in this parameter in the Max-Forwards header of the SIP requests that it generates. Default is 70.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
sip_retx	(Optional) Maximum number of times that a SIP message other than an INVITE request will be retransmitted. Valid value is any positive integer. Default is 10.
sntp_mode	(Optional) Mode in which the phone listens for the SNTP server. Valid values are unicast, multicast, anycast, or directedbroadcast. Default is anycast
sntp_server	IP address of the SNTP server from which the phone obtains time data.
speed_label*x*	(Optional) Configures the speed-dial key label. The *x* variable is from label 2 to label 6. There are five possible labels that can be configured on the Cisco IP 7960G but only one on the Cisco IP 7940G. The *x* variable is a string of up to 15 characters. **Note** This parameter *cannot* be set in the configuration file.
speed_line*x*	(Optional) Configures the speed-dial keys so that the user can set up one-touch dialing. There are five possible numbers that can be configured on the Cisco IP 7960G but only one on the Cisco IP 7940G. The *x* variable is a string of up to 128 bytes. **Note** You *cannot* set this parameter in the configuration file.
start_media_port	(Optional) Start RTP range for media. Range is from 16384 to 32766. Default is 16384.
stutter_msg_waiting	(Optional) Enables a stutter dial tone when there is a message waiting. It is disabled by default. Valid values are 0 (off) and 1 (on).
sync	(Optional) Value against which to compare the value in the syncinfo.xml file before a remote reboot is performed. Limited to 32 characters.
telnet_level	(Optional) Enables Telnet for the phone. Valid values are as follows: • 0—Disabled. • 1—Enabled, and no privileged commands can be executed. • 2—Enabled, and privileged commands can be executed. Default is 0.
tftp_cfg_dir	Path to the TFTP subdirectory in which phone-specific configuration files are stored. **Note** This parameter is only required if the phone-specific files are in a subdirectory and not in the root directory.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
time_format_24hr	(Optional) Whether a 12- or 24-hour time format is displayed by default on the user interface. Valid values are as follows: • 0—12-hour format is displayed by default but can be changed to a 24-hour format using the user interface. • 1—24-hour format is displayed by default but can be changed to a 12-hour format using the user interface. • 2—12-hour format is displayed and cannot be changed to a 24-hour format using the user interface. • 3—24-hour format is displayed and cannot be changed to a 12-hour format using the user interface. Default is 1.
time_zone	(Optional) Time zone in which the phone is located. Valid values are the time-zone abbreviations shown in Table 3-5 on page 3-18. Abbreviations are case sensitive and must be in all capital letters. Default is PST.
timer_invite_expires	(Optional) Amount of time, in seconds, after which a SIP INVITE expires. This value is used in the Expire header field. Valid values are any positive number; however, we recommend 180. Default is 180.
timer_register_delta	Configures the time interval at which reregistration will occur. This is a numeric field in which the time interval is measured in seconds. Valid values range from 32767 to 0. Default is 5 (phone will attempt to reregister 5 seconds before its registration period expires).
timer_register_expires	(Optional) Amount of time, in seconds, after which a REGISTRATION request expires. This value is inserted into the Expire header field. Valid values are any positive number; however, we recommend 3600. Default is 3600.
timer_t1	(Optional) Lowest value, in milliseconds, of the retransmission timer for SIP messages. Valid values are any positive integer. Default is 500.
timer_t2	(Optional) Highest value, in milliseconds, of the retransmission timer for SIP messages. Valid values are any positive integer greater than timer_t1. Default is 4000.

Table D-1 Configuration File Parameters (continued)

Parameter	Description
tos_media	(Optional) Type of service (ToS) level for the media stream being used. Valid values are as follows: • 0—IP_ROUTINE • 1—IP_PRIORITY • 2—IP_IMMEDIATE • 3—IP_FLASH • 4—IP_OVERRIDE • 5—IP_CRITIC Default is 5.
user_info	(Phone-specific; optional) Configures the "user=" parameter in the REGISTER message. Valid values are as follows: • none—No value is inserted. • phone—The value user=phone is inserted in the To, From, and Contact Headers for REGISTER. • ip—The value user=ip is inserted in the To, From, and Contact Headers for REGISTER. Default is none.
voip_control_port	(Optional) UDP port used for SIP messages. All SIP REQUESTS use voip_control_port as the UDP source port when nat_enable = 1. Range is from 1025 to 65535. Default is 5060.

Table D-2 Network Parameters

Parameter	Description
Admin.VLAN Id	Unique identifier of the VLAN to which the phone is attached, for use in switched networks that are not Cisco networks. **Note** If you have an administrative VLAN setting assigned on the Catalyst switch, that setting overrides any changes made on the phone.
Alternate TFTP	Whether to use an alternate remote TFTP server rather than the local one. Valid values are Yes and No. If you set this parameter to Yes, you must change the IP address in the TFTP Server parameter to the address of the alternate TFTP server. Default is No.

Table D-2 Network Parameters (continued)

Parameter	Description
Default Router 1 to 5	IP address (1) of the default gateway used by the phone and (2 to 5) of the gateways that the phone attempts to use as an alternate gateway if the default gateway is unavailable. **Note** Default Router 1 always takes precedence. If Router 1 is unavailable, the phone moves down the list of available routers in order based on the configuration in Routers 2 through 5. **Note** If you have an administrative VLAN setting assigned on the Catalyst switch, that setting overrides any changes made on the phone.
DHCP Address Released	Whether the IP address of the phone can be released for reuse in the network. Valid values are Yes and No. When set to Yes, the phone sends a DHCP release message to the DHCP server and goes into a release state. The release state provides enough time to remove the phone from the network before the phone attempts to acquire another IP address from the DHCP server. When you move the phone to a new network segment, first release the DHCP address.
DHCP Enabled	Whether the phone uses DHCP to configure network settings (IP address, subnet mask, domain name, default router list, DNS server list, and TFTP address). Valid values are Yes and No. Default is Yes. To manually configure your IP settings, you must set this parameter to No.
DHCP Server	IP address of the DHCP server from which the phone received its IP address and additional network settings.
DNS Servers 1 to 5	IP address of the DNS server used by the phone to resolve names to IP addresses. The phone attempts to use DNS servers 2 to 5 if DNS server 1 is unavailable. **Note** DHCP must be disabled.
Dynamic TFTP Server	IP address of a dynamic TFTP server. After initially querying the default TFTP server, the phone rerequests the default and phone-specific configuration files from the new TFTP server. The dynamic TFTP server address is not stored in flash memory.

Table D-2 Network Parameters (continued)

Parameter	Description
Erase Configuration	Whether to erase all of the locally defined network settings on the phone and reset the values to the defaults. Valid values are Yes and No. Yes reenables DHCP.
HTTP Proxy Address	IP address of the HTTP proxy server. You can use either a dotted IP address or a DNS name (a record only).
HTTP Proxy Port	Number of the outbound proxy port. Default is 80.
IP Address	IP address of the phone that is assigned by DHCP or is locally configured. **Note** DHCP must be disabled.
MAC Address	Factory-assigned unique 48-bit hexadecimal MAC address of the phone.
Operational VLAN Id	Unique identifier of the VLAN of which the phone is a member. This identifier is obtained through Cisco Discovery Protocol (CDP).
Subnet Mask	IP subnet mask used by the phone. A subnet mask partitions the IP address into a network and a host identifier. **Note** DHCP must be disabled.
TFTP Server	IP address of the TFTP server. **Note** If you have an administrative VLAN setting assigned on the Catalyst switch, that setting overrides any changes made on the phone. **Note** DHCP must be disabled.
GARP Enabled	Enables or disables generation of the Gratuitous ARP packets from the phone.

Table D-3 SIP Parameters

Parameter	Description
Authentication Name	(Phone-specific) Name used by the phone for authentication if a registration is challenged by the proxy server during initialization. **Note** Required when registration is enabled and the registrar challenges registration.
Authentication Password	(Phone-specific) Password used by the phone for authentication if a registration is challenged by the proxy server during initialization. If a value is not configured for the Authentication Password parameter when registration is enabled, the default logical password is used. The default logical password is SIP*mac-address*, where *mac-address* is the MAC address of the phone. **Note** Required when registration is enabled and the registrar challenges registration.
Display Name	(Phone-specific) Identification as it should appear for caller identification. For example, instead of jdoe@company.com appearing on phones that have caller ID, you can specify User A in this parameter to have User A appear on the callee end instead. If a value is not specified for this parameter, the Name value is used.
Name	(Phone-specific) Description phone number or e-mail address used when registering. When entering a number, enter the number without any dashes. For example, enter 555-0100 as 5550100. When entering an e-mail address, enter the e-mail ID without the host name.
Proxy Address	(Phone-specific) IP address of the primary SIP proxy server that will be used by the phone. Enter this address in IP dotted-decimal notation.

Table D-3 SIP Parameters (continued)

Parameter	Description
Proxy Port	(Phone-specific) Port number of the primary SIP proxy server. This is the port that the SIP client will use. The default is 5060.
Short Name	(Phone-specific) Name or number associated with the line*x*_name as you want it to display on the phone LCD if the line*x*_name value exceeds the display area. For example, if the line*x*_name value is the phone number 111-222-333-4444, you can specify 34444 for this parameter to have 34444 display on the LCD instead. Alternatively, if the value for the line*x*_name parameter is the e-mail address "username@company.com", you can specify the "username" to have just the username appear on the LCD instead. This parameter is used for display only. If a value is not specified for this parameter, the value in the Name variable is displayed.

Table D-4 Manual SIP Parameter Configuration

Parameter	Description
Line 1 settings	Displays the SIP settings that are defined in Table D-2.
Line 2 settings	
Line 3 settings	
Line 4 settings	
Line 5 settings	
Line 6 settings	
Messages URI	(Optional) Configures the voice-mail number that is dialed when the messages button is pressed. Value is typically a phone number but can be a URI.
Preferred Codec	(Optional) Codec to use when a call is initiated. Valid values are g711alaw, g711ulaw, g729a, and none. Default is g711ulaw.

Table D-4 Manual SIP Parameter Configuration (continued)

Parameter	Description
Out of Band DTMF	(Optional) Configures the out-of-band signaling (for tone detection on the IP side of a gateway). **Note** The Cisco SIP IP phone supports out-of-band signaling using the AVT tone method. Valid values are as follows: • none—Do not generate DTMF digits out-of-band. • avt—If requested by the remote side, generate DTMF digits out-of-band (and disable in-band DTMF signaling); otherwise, do not generate DTMF digits out-of-band. • avt_always—Always generate DTMF digits out-of-band. This option disables in-band DTMF signaling. Default is avt.
Register with Proxy	(Optional) The phone must register with a proxy server during initialization. Valid values are 0 (disable registration during initialization) and 1 (enable registration during initialization). Default is 0. **Note** You can also use this parameter in a phone-specific configuration file. After a phone has initialized and registered with a proxy server, you can remove the registration by changing this value to 0 in the phone-specific configuration file. To reinitiate registration, change the value back to 1. **Note** If you enable registration and authentication is required, you must specify values for the line*x*_authname and line*x*_password parameters in the phone-specific configuration file.
Register Expires	(Optional) Amount of time, in seconds, after which a REGISTRATION request expires. This value is inserted into the Expire header field. Valid values are any positive number; however, we recommend 3600. Default is 3600.

Table D-4 Manual SIP Parameter Configuration (continued)

Parameter	Description
TFTP Directory	IP address of the TFTP server. **Note** If you have an administrative VLAN setting assigned on the Catalyst switch, that setting overrides any changes made on the phone. **Note** DHCP must be disabled.
Phone Label	(Phone-specific; optional) Text to display on the top right status line of the LCD. This field is for end-user display only and has no effect on caller identification or messaging. For example, a phone label can display “User A’s phone.” Limited to 11 characters.
Enable VAD	(Optional) Enables voice activation detection (VAD). Default is No.
VoIP Control Port	(Optional) UDP port used for SIP messages. All SIP REQUESTS use voip_control_port as the UDP source port when nat_enable = 1. Range is from 1025 to 65535. Default is 5060.
Start Media Port	(Optional) Start RTP range for media. Range is from 16384 to 32766. Default is 16384.
End Media Port	(Optional) Configures the Real-Time Transport Protocol (RTP) end range for media. Valid values are 16384 to 32766. Default is 32766.
Backup Proxy	(Optional) IP address of the backup proxy server or gateway. Enter this address in IP dotted-decimal notation. **Note** You must specify at least one address or the phones cannot register.
Backup Proxy Port	(Optional) Port number of the backup proxy server. Default is 5060.
Emergency Proxy	(Optional) IP address of the emergency proxy server or gateway. Enter this address in IP dotted-decimal notation.
Emergency Proxy Port	(Optional) Port number of the emergency proxy server. Default is 5060.
Outbound Proxy	(Optional) IP address of the outbound proxy server. You can use either a dotted IP address or a DNS name.

Table D-4 Manual SIP Parameter Configuration (continued)

Parameter	Description
Outbound Proxy Port	(Optional) Port number of the outbound proxy server. Default is 5060. When an outbound proxy is enabled, all SIP requests are sent to the outbound proxy server instead of to the proxy*x*_address. All responses continue to reconcile the normal Via processing rules. The media stream is not routed through the outbound proxy. NAT and outbound proxy modes can be independently enabled or disabled. The received= tag is added to the Via header of all responses if there is no received= tag in the uppermost Via header and if the source IP address is different from the IP address in the uppermost Via header. Keep the following rules in mind: • If a received= tag is in the uppermost Via header, the response is sent back to the IP address contained in the received= tag. • If there is no received= tag and the IP address in the uppermost Via header is different from the source IP address, the response is sent back to the source IP address.

Table D-4 Manual SIP Parameter Configuration (continued)

Parameter	Description

NAT Enabled	(Optional) Enables NAT.
	
	Valid values are 0 (disable) and 1 (enable). Default is 0.
	
	• If NAT is enabled, the Contact header appears as follows:
	
	`Contact: sip:lineN_name@nat_address:voip_control_port`
	
	If the nat_address is invalid or UNPROVISIONED, the Contact header appears as follows:
	
	`Contact: sip:lineN_name@phone_ip_address:voip_control_port`
	
	and the Via header appears as follows:
	
	`Via: SIP/2.0/UDP phone_ip_address:voip_control_port`
	
	• If NAT is enabled, the Session Description Protocol (SDP) message uses the nat_address and an RTP port between the start_media_port and the end_media_port range in the C and M fields. All RTP traffic is sourced from the port advertised in the SDP message.

Table D-4 Manual SIP Parameter Configuration (continued)

Parameter	Description
NAT Address	(Optional) WAN IP address of the Network Address Translation (NAT) or firewall server. Value is either a dotted IP address or a DNS name.
Call Statistics	(Optional) Includes RTP statistics in BYE requests and responses. Valid values are 0 (disable) and 1 (enable). Default is 0. If this parameter is enabled, the phone inserts the headers RTP-RxStat and RTP-TxStat as follows: • RTP-RxStat: Dur=a,Pkt=b,Oct=c,LatePkt=d,LostPkt=e,AvgJit=f • RTP-TxStat: Dur=g,Pkt=h,Oct=i where the following apply: • Dur—Total number of seconds since the beginning of reception or transmission. • Pkt—Total number of RTP packets received or transmitted. • Oct—Total number of RTP payload octets received or transmitted (not including RTP header). • LatePkt—Total number of late RTP packets received. • LostPkt—Total number of lost RTP packets received (not including late RTP packets). • AvgJit—Average jitter, which is an estimate of the statistical variance of the RTP packet inter-arrival time, measured in timestamp unit and calculated according to RFC 1889. • a, b, c, d, e, f, g, h, and i—Integers.