



Configuring Regional Parameters and Supplementary Services

Use the *Regional* tab to configure regional and local settings, such as Vertical Service Activation codes (star codes), Vertical Service Announcement codes, and local language and dictionary.

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Call progress tone pass-through lets a user hear call progress tones (such as ringing) that are generated by the far-end network.

Scripting for Cadences, Call Progress Tones, and Ring Tones

Cisco SPA IP phones have configurable call progress tones. Parameters for each type of tone can include a number of components, defining frequency, amplitude and cadence.

Cadence Script

A CadScript is a mini-script that specifies the cadence parameters of a signal up to 127 characters in length. Syntax:

```
S1[;S2]
```

where:

```
Si=Di(oni,1/offi,1[,oni,2/offi,2[,oni,3/offi,3[,oni,4/offi,4[,oni,5/offi,5[,oni,6/offi,6]]]])) and is known as a section
```

$on_{i,j}$ and $off_{i,j}$ are the on/off durations in seconds of a *segment*. The subscript ranges are: $i = 1$ or $2, j = 1$ to 6 .

D_i is the total duration of the section in seconds. All durations can have up to three decimal places to provide 1 ms resolution. The wildcard character "*" indicates infinite duration.

The segments within a section are played in order and repeated until the total duration is played.

Example: Normal Ring

```
60(2/4)
```

where:

- Number of Cadence Sections = 1
- Cadence Section 1: Section Length = 60 s
- Number of Segments = 1
- Segment 1: On=2s, Off=4s
- Total Ring Length = 60s

Example 2: Distinctive Ring (short, short, short, long)

```
60(.2/.2,.2/.2,.2/.2,1/4)
```

where:

- Number of Cadence Sections = 1
- Cadence Section 1: Section Length = 60s
- Number of Segments = 4
- Segment 1: On=0.2s, Off=0.2s
- Segment 2: On=0.2s, Off=0.2s
- Segment 3: On=0.2s, Off=0.2s
- Segment 4: On=1.0s, Off=4.0s
- Total Ring Length=60s

Tone Script

A ToneScript is a mini-script that specifies the frequency, level, and cadence of a call progress tone. It can contain up to 127 characters. Syntax:

```
FreqScript;Z1[;Z2]
```

Section Zi is similar to the Si section in a CadScript except that each on/off segment is followed by a frequency parameter: $Z_i = D_i(\text{on}_i, 1/\text{off}_i, 1/f_{i,1}[, \text{on}_i, 2/\text{off}_i, 2/f_{i,2} [, \text{on}_i, 3/\text{off}_i, 3/f_{i,3} [, \text{on}_i, 4/\text{off}_i, 4/f_{i,4} [, \text{on}_i, 5/\text{off}_i, 5/f_{i,5} [, \text{on}_i, 6/\text{off}_i, 6/f_{i,6}]]]])$

where $f_{i,j} = n_1[+n_2]+n_3[+n_4[+n_5[+n_6]]]$ $1 < n_k < 6$ indicates which of the frequency components given in the FreqScript are used in that segment; if more than one frequency component is used in a segment, the components are summed together.

Example: Dial Tone

```
350@-19,440@-19;10(*0/1+2)
```

- Number of Frequencies = 2
- Frequency 1 = 350 Hz at -19 dBm
- Frequency 2 = 440 Hz at -19 dBm
- Number of Cadence Sections = 1
- Cadence Section 1: Section Length = 10 s
- Number of Segments = 1
- Segment 1: On=forever, with Frequencies 1 and 2
- Total Tone Length = 10s
- Example 2: Stutter Tone
- $350@-19,440@-19;2(.1/.1/1+2);10(*0/1+2)$
- Number of Frequencies = 2
- Frequency 1 = 350 Hz at -19 dBm
- Frequency 2 = 440 Hz at -19 dBm
- Number of Cadence Sections = 2
- Cadence Section 1: Section Length = 2s
- Number of Segments = 1
- Segment 1: On=0.1s, Off=0.1s with Frequencies 1 and 2
- Cadence Section 2: Section Length = 10s
- Number of Segments = 1
- Segment 1: On=forever, with Frequencies 1 and 2
- Total Tone Length = 12s

Example: SIT Tone

```
985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
```

- Number of Frequencies = 3
- Frequency 1 = 985 Hz at -16 dBm
- Frequency 2 = 1428 Hz at -16 dBm
- Frequency 3 = 1777 Hz at -16 dBm
- Number of Cadence Sections = 1

- Cadence Section 1: Section Length = 20s
- Number of Segments = 4
- Segment 1: On=0.38s, Off=0s, with Frequency 1
- Segment 2: On=0.38s, Off=0s, with Frequency 2
- Segment 3: On=0.38s, Off=0s, with Frequency 3
- Segment 4: On=0s, Off=4s, with no frequency components
- Total Tone Length = 20s

Ring Script

A RingScript is a mini-script that describes a ring tone. Syntax:

```
n=ring-tone-name;w=waveform-id-or-path;c=cadence-id;b=break-time;t=total-time
```

`ring-tone-name` identifies this ring tone. This name appears on the Ring Tone menu of the phone. The same name can be used in a SIP Alert-Info header in an inbound INVITE request to tell the phone to play the corresponding ring tone. The name should contain the same characters allowed in a URL only.

`waveform-id-or-path` is the index of the desired waveform to use in this ring tone. The built-in waveforms are:

- 1 = Classic phone with mechanical bell
- 2 = Typical phone ring
- 3 = Classic ring tone
- 4 = Wide-band frequency sweep signal

This field can also be a network path (url) to download a ring tone data file from a server on-the-fly. In this case, the syntax of the field is:

```
w=[tftp://]hostname[:port]/path
```

`cadence-id` is the index of the desired cadence to play the given waveform. 8 cadences (1–8) as defined in <Cadence 1> through <Cadence 8>. Cadence-id can be 0 If w=3,4, or an url. Setting c=0 implies the on-time is the natural length of the ring tone file.

`break-time` specifies the number of seconds to break between two bursts of ring tone, such as b=2.5.

`total-time` specifies the total number of seconds to play the ring tone before it times out.

Call Progress Tones

For definitions of all call progress tones, see [Call Progress Tone Description](#).

Distinctive Ring Patterns

Ring cadence defines the ringing pattern that announces a telephone call. The Cisco WIP310 has only eight distinctive ring pattern fields.

The pattern is:

```
length(on/off)
```

where:

- Length: The total length of the ring
- On: The number of on seconds the ring is heard.
- Off: The number of seconds the ring is silent.

Example 1: Normal Ring

```
60(2/4)
```

where:

- Number of Cadence Sections = 1
- Cadence Section 1: Section Length = 60 s
- Number of Segments = 1
- Segment 1: On=2s, Off=4s
- Total Ring Length = 60s

Example 2: Distinctive Ring (short, short, short, long)

```
60(.2/.2,.2/.2,.2/.2,1/4)
```

where:

- Number of Cadence Sections = 1
- Cadence Section 1: Section Length = 60s
- Number of Segments = 4
- Segment 1: On=0.2s, Off=0.2s
- Segment 2: On=0.2s, Off=0.2s
- Segment 3: On=0.2s, Off=0.2s
- Segment 4: On=1.0s, Off=4.0s
- Total Ring Length=60s

Distinctive Call Waiting Tone

When the phone is off-hook on a call, the call waiting tone plays. Support for Distinctive Ring is based on the Alert-Info header that supports the Distinctive Call Waiting tone. The distinctive call waiting tone is generated based on the phone call waiting tone frequency and gain values, and the cadence value of the matched ring tone.

The cadence value (following the `c=` element) of the matched ring tone must be an integer from 1 to 9 that specifies the ring cadence under the **Regional** tab of the phone web user interface.

If there is no matching ring tone name, or an invalid cadence value is specified, the configured Call Waiting Tone is used.

Control Timer Values (sec)

The table describes Control Timer parameters.

Field	Description
Reorder Delay	Delay after far end hangs up before reorder (busy) tone is played. Ranges from 0 to 255 seconds. 0 = play immediately inf = never play 255 = return the phone immediately to on-hook status and do not play the tone. Defaults to 5.
Call Back Expires	Expiration time of a call back activation request. Ranges from 0 to 65535 seconds. Defaults to 1800.
Call Back Retry Intvl	The interval between call back retry requests. Ranges from 0 to 255 seconds. Defaults to 30.
Call Back Delay	Delay after receiving the first SIP 18x response before declaring the remote end is ringing. If a busy response is received during this time, the Cisco SPA IP phones still consider the call as failed and continues to retry. Defaults to 0.5.

Field	Description
Interdigit Long Timer	<p>Long timeout between entering digits when dialing. The interdigit timer values are used as defaults when dialing. The <i>Interdigit Long Timer</i> is used after any one digit, if all valid matching sequences in the dial plan are incomplete as dialed. Ranges from 0 to 64 seconds.</p> <p>Setting this value high can result in a longer post dialing delay (PDD), which is the time between the start of a call and the time the phone starts ringing. A value that is too low can result in dialed digits not being correctly recognized.</p> <p>Defaults to 10.</p>
Interdigit Short Timer	<p>Short timeout between entering digits when dialing. The <i>Interdigit Short Timer</i> is used after any one digit, if at least one matching sequence is complete as dialed, but more dialed digits would match other as yet incomplete sequences. Ranges from 0 to 64 seconds.</p> <p>Defaults to 3.</p>

Configuring Supplementary Services (Star Codes)

The Cisco IP phones provide native support of a large set of enhanced or supplementary services (also known as star codes). A user can enter star codes (such as *21 for call forward, followed by the target number) to perform call features such as call return, blind call transfers, call pickup, and so on. These codes can be handled locally by the phone or to be sent to the network as an INVITE to the service provider.

Some service providers choose to disable star codes. See [Configuring Supplementary Services \(Star Codes\)](#) for more information.

Entering Star Code Values

The phone provides default values for star codes. To change star code values, navigate to **Admin Login > advanced > Regional**. Under **Vertical Service Activation Codes**, enter the values you want to change for the codes.

The codes are:

- Call Return (*69)—Calls the last caller, regardless which extension.
- Blind Transfer (*98)—Allows the user to transfer a call to another number without waiting for the other party to pick up.
- Call Back Act (*66)—Periodically redials the last busy number (every 30 seconds by default) until it rings or until the attempt expires (30 min by default), regardless which extension. Only one call back operation can be ordered at a time. A new order automatically cancels the previous order.
- Call Back Deact (*86)—Cancels the last call back operation.
- Call Forward All Act (*72)—Call forwards all inbound calls. Applies to primary extension only.
- Call Forward All Deact (*73)—Cancels call forward all. Applies to primary extension only.
- Call Forward Busy Act (*90)—Call forwards on busy. Applies to primary extension only.
- Call Forward Busy Deact (*91)—Cancels call forward on busy. Applies to primary extension only.

- Call Forward No Answer Act (*92)—Call forwards if no answer. Applies to primary extension only.
- Call Forward No Answer Deact (*93)—Cancels call forward no answer. Applies to primary extension only.
- CW Act (*56)—Enables call waiting. For example, if call waiting is turned off globally, this star code will turn on call waiting until the CW Deact code is entered.
- CW Deact (*57)—Deactivates call waiting. For example, if call waiting is turned on globally, this star code deactivates call waiting until the CW Act code is entered.
- CW Per Call Act (*71)—Enables call waiting for a single call. For example, if call waiting is turned off globally, this star code will turn on call waiting for that call.
- CW Per Call Deact (*70)—Deactivates call waiting for a single call. For example, if call waiting is turned on globally, this star code deactivates call waiting for that call.
- Block CID Act (*67)—Blocks caller ID on all outbound calls. Applies to all extensions.
- Block CID Deact (*68)—Deactivates caller ID blocking on outbound calls. Applies to all extensions.
- Block CID Per Call (*81)—Blocks caller ID on the next outbound call (on the current call appearance only).
- Block CID Per Call Deact (*82)—Deactivates caller ID blocking on the next outbound call (on the current call appearance only).
- Block ANC Act—Blocks anonymous calls. Applies to all extensions.
- Block ANC Deact—Deactivates anonymous call blocking. Applies to all extensions.
- DND Act (*78)—Activates Do Not Disturb. Applies to all extensions.
- DND Deact (*79)—Deactivates Do Not Disturb. Applies to all extensions.
- Secure All Call Act (*16)—Defaults to prefer to use encrypted media (voice codecs).
- Secure No Call Act (*17)—Defaults to prefer to use unencrypted media for all outbound calls. Applies to all extensions.
- Secure One Call Act (*18)—Prefers to use encrypted media for the outbound call (on this call appearance only).
- Secure One Call Deact (*19)—Prefers to use unencrypted media for the outbound call (on this call appearance only).
- Paging (*96)—Pages the number called.
- Call Park (*38)—Parks a call on an entered line number.
- Call UnPark Code (*39)—Retrieves a call from an entered line number.
- Call Pickup (*36)—Picks up a call at an entered extension.
- Group Call Pickup (*37)—Picks up a ringing call at a group of extensions.
- Media Loopback Code (*03)—A service provider can set up a test call from an IP media loopback server (the source) to a subscriber VoIP device (the mirror). The test call provides statistical reporting on network performance and audio quality.

Depending on the source capabilities, a service provider can see packet jitter, loss, and delay (although Media Loopback cannot identify an offending hop). This helps the service provider identify an offending hop that could be causing issues in VoIP calls to a subscriber. The test results can also provide audio quality scoring, that lets a service provider better understand the subscriber's experience.

- Referral Services Codes—One or more * codes can be configured into this parameter, such as *98, or *97|*98|*123, and so forth. The maximum total length is 79 characters.

This parameter applies when the user places the current call on hold (by Hook Flash) and is listening to second dial tone. Each * code (and the following valid target number according to current dial plan) entered on the second dial-tone triggers the Cisco IP phone to perform a blind transfer to a target number that is prepended by the service * code. For example:

- After the user dials *98, the Cisco IP phone plays a special prompt tone while waiting for the user to enter a target number (which is validated according to the dial plan as in normal dialing).
- When a complete number is entered, the Cisco IP phone sends a blind REFER to the holding party with the Refer-To target equals to *98 *target_number*. This feature allows the Cisco IP phone to hand off a call to an application server to perform further processing, such as call park.

The * codes should not conflict with any of the other vertical service codes internally processed by the Cisco IP phone. You can delete any * code that you do not want the call server to process.

Feature Dial Services Codes: Tells the Cisco IP phone what to do when the user is listening to the first or second dial tone.

You can configure one or more * codes into this parameter, such as *72, or *72|*74|*67|*82, and so on. The maximum total length is 79 characters. When the user has a dial tone (first or second dial tone), they can enter a * code (and the following target number according to current dial plan) to trigger the Cisco IP phone to call the target number prepended by the * code. For example:

- After the user dials *72, the Cisco IP phone plays a special prompt tone while waiting for the user to enter a target number (which is validated according to the dial plan as in normal dialing).
- When a complete number is entered, the Cisco IP phone sends an INVITE to *72 *target_number* as in a normal call. This feature allows the proxy to process features such as call forward (*72) or BLock Caller ID (*67).

You can add a parameter to each * code in *Features Dial Services Codes* to indicate what tone to play after the * code is entered, such as *72'c'|*67'p'.

Following is a list of allowed dial tone parameters (note the use of back quotes surrounding the parameter without spaces).

- 'c' = Cwd dial tone
- 'd' = Dial tone
- 'm' = MWI dial tone
- 'o' = Outside dial tone
- 'p' = Prompt dial tone
- 's' = Second dial tone
- 'x' = No tones are place, x is any digit not used above

If no tone parameter is specified, the Cisco IP phone plays the prompt tone by default.

If the * code is not to be followed by a phone number, such as *73 to cancel call forwarding, do not include it in this parameter. In that case, add that * code in the dial plan.

Activating or Deactivating Supplementary Services

You can disable services handled locally by the phone in one of two ways:

- Delete the star code in the *Vertical Service Activation* section in the **Regional** tab.
- Disable the service in the **Phone** tab. See [Configuring Supplementary Services \(Star Codes\)](#), page 7-7.

If a service is enabled in the Phone tab but cleared in the Regional tab, the service can still be enabled/disabled from the IP phone screen or the phone web user interface. If a service is disabled, the soft button associated with that service is hidden on the IP phone screen. Also, any menu item associated with a disabled service is preceded with an exclamation mark (!).

A supplementary service should be disabled if

- the user has not subscribed to it
- the service provider intends to support similar service by using means other than relying on the Cisco IP phone.

Vertical Service Announcement Codes (Cisco SPA300 Series and Cisco SPA500 Series)

The Cisco SPA300 Series and Cisco SPA500 Series IP phones support all services that can be activated on a phone (call forward, do not disturb, and so on). Vertical service announcement codes apply only when the user dials the corresponding star code.

Following is an example of how you can use these fields:

```
<Service Annc Base Number> = 1234
<Service Annc Extension Codes>= "CWT:00;CWF:01;FAT:02;FAF:05;FBT:03;FBF:05;FNT:04;FNF:05;"
Here CWT: Call waiting service enabled;
CWF: Call waiting service disabled;
FAT: Call forward all service enabled;
FAF: Call forward all service disabled;
FBT: Call forward busy service enabled;
FBF: Call forward busy service disabled;
FNT: Call forward no answer enabled;
FNF: Call forward no answer disabled;
```

When the user enables call waiting service, the IP phone automatically calls 123400@\$proxy.

When the user *disables* the call waiting service, IP phone connects to 123401@\$proxy.

If the <Service Annc Extension Codes> do not define CWT/CWF extension codes, the IP phone defaults to normal.

Bonus Services Announcement Description

When the user enables the callback service using the *code, the IP phone automatically calls 123400@\$proxy.

When the user disables the callback service using the *code, the IP phone automatically connects to the 123401@\$proxy.

If the *Service Annc Extension Codes* do not define CBT/CBF extension codes, the IP phone does not use this feature.

```
[Line1/2]<Service Announcement Serv> = Yes
[Regional]<Service Annc Base Number> = {announcement server base number}
[Regional]<Service Annc Extension Codes> = {SAEC Script}
SAEC Script format:{SA_map;}* Here * means 0 or multiple
SA_map syntax:
    SA_serv=SA_extcode
    SA_serv is the name of service plus the current condition;
    SA_extcode is the extension code which the ANNC server will route to.
```

Appendix: SA_serv list

- 1) Call Back
 - CBT: Call back enabled
 - CBF: Call back disabled
 - CBB: Call back busy enabled
- 2) Call Forward
 - FAT: Call forward all enabled
 - FAF: Call forward all disabled
 - FBT: Call forward busy enabled
 - FBF: Call forward busy disabled
 - FNT: Call forward no answer enabled
 - FNF: Call forward no answer disabled
 - FLT: Call forward last enabled
 - FLF: Call forward last disabled
- 3) Call Waiting
 - CWT: Call waiting enabled
 - CWF: Call waiting disabled
- 4) Block Last Call
 - BLT: Block last call enabled
 - BLF: Block last call disabled
- 5) Accept Last Call
 - ALT: Accept last call enabled
 - ALF: Accept last call disabled
- 6) Block Caller ID
 - BCT: Block caller id enabled
 - BCF: Block caller id disabled
- 7) Distinctive Ringing
 - DRT: Distinctive ringing enabled
 - DRF: Distinctive ringing disabled
- 8) Speed Dial
 - SDT: Speed dial enabled
 - SDF: Speed dial disabled
- 9) Secure Call
 - SCT: Secure call enabled
 - SCF: Secure call disabled
- 10) Do Not Disturb
 - DDT: DND enabled
 - DDF: DND disabled
- 11) Caller ID
 - CDT: Caller ID enabled
 - CDF: Caller ID disabled
- 12) CW CID
 - WDT: CWCID enabled
 - WDF: CWCID disabled
- 13) Block Anonymous call
 - BAT: Block anonymous call enabled
 - BAF: Block anonymous call disabled

Outbound Call Codec Selection Codes

Codec call selection codes affect voice quality. For more information about voice codecs, see the [Configuring Voice Codecs](#). You can choose a *preferred* codec for a call or *force* a call to use a specific codec:

- Prefer *G.711u* (*017110) through *G.729a* (*01729)—Sets the preferred codec for next outbound call. If the preferred codec is unavailable, the second, then the third preferred codec is used, if specified.
- Force *G.711u* (*027110) through *G.729a* (*02729) —Forces the specified codec for next outbound call. If the specified codec is unavailable, the preferred codecs are used in order, if specified.

See [Configuring Voice Codecs](#) for more information.

Miscellaneous Parameters

This section describes Dual Tone Multi-Frequency (DTMF) and localization parameters:

DTMF Parameters

DTMF is used by touch-tone phones to assign a specific frequency (consisting of two separate tones) to each key so that it can easily be identified by a microprocessor.

In-Band and Out-of-Band (RFC-2833): IP phones can relay DTMF digits as out-of-band events to preserve the fidelity of the digits. This can enhance the reliability of DTMF transmission required by many IVR applications such as dial-up banking and airline information.

The following parameters can either reduce false detection or get better detection by the IVR. In general, the default values are recommended for both IVR functions.

- *DTMF Playback Level*: Local DTMF playback level in decibels per minute, up to one decimal place. Applicable locally when a user presses a digit or when the phone receives an out-of-band (OOB) DTMF signal from the network side. Does not affect DTMF transmission. Defaults to -16.
- *DTMF Playback Length*: Local DTMF playback duration in milliseconds. Affects only OOB. Defaults to .1.
- *Inband DTMF Boost*: Controls the amount of amplification applied to DTMF signals. Affects only tones sent by inband method. Choices are 0, 3, 6, 9, 12, 15, and 18 decibels. Defaults to 12 dB.

To support false detection, avoid inband and use OOB. With OOB, the DTMF Playback Length does not matter. If you use inband, use a smaller DTMF Boost value.

To get better detection by the IVR, avoid inband and use OOB. This way, the DTMF tone is reconstructed by the PSTN gateway or the remote endpoint, and the quality is not subject to distortion from the audio codec. If you use OOB, use a slightly longer DTMF Playback Length.

If you use inband, use a higher Inband DTMF boost.



Note

On the Cisco SPA525G2, when using the Mobile Link line (through the Bluetooth-enabled mobile phone), the local user can hear a double tone (echo) when pressing digits (DTMF tones) and engaged on a call. This can happen with certain mobile phones that have the option to play locally the local tone (which is also played by the Cisco SPA525G2). This does not affect operation with interactive voice

response applications, as the tone is audible only on the local device. See the Cisco support community at <http://www.cisco.com/go/smallbizsupport> for phone compatibility information, and also consult the latest Cisco SPA525G2 release notes, available at cisco.com.

Localizing Your IP Phone

The following table describes the localization parameters in the Miscellaneous section.

Field	Description
Set Local Date (mm/dd)	Enter the local date (<i>mm</i> represents the month and <i>dd</i> represents the day). The year is optional and uses two or four digits. For example, May 1, 2008, can be entered as: 05/01 or 05/01/08 or 05/01/2008 .
Set Local Time (HH/mm)	Enter the local time (<i>hh</i> represents hours and <i>mm</i> represents minutes). Seconds are optional.
Time Zone	Selects the number of hours to add to GMT to generate the local time for caller ID generation. Choices are GMT-12:00, GMT-11:00, ..., GMT, GMT+01:00, GMT+02:00, ..., GMT+13:00. Defaults to GMT-08:00.
Time Offset (HH/mm)	Enter the offset from GMT to use for the local system time.
Daylight Saving Time Rule	Enter the rule for calculating daylight saving time. See Configuring Daylight Saving Time .
Daylight Saving Enable	Select yes to enable or no to disable DST on the phone. This setting affects all lines (extensions) on the phone.
Dictionary Server Script	Defines the location of the dictionary server, the languages available, and the associated dictionary. See Creating a Dictionary Server Script .
Language Selection	Specifies the default language. The value must match one of the languages supported by the dictionary server. The script (dx value) is: <pre><Language_Selection ua="na"> </Language_Selection></pre> Defaults to blank; the maximum number of characters is 512. For example: <pre><Language_Selection ua="na"> Spanish </Language_Selection></pre>

Managing the Time and Date

Cisco IP phones obtain the time settings in one of three ways:

- **NTP Server**—When the phone boots up, it tries to contact the first Network Time Protocol (NTP) server to get the time. The phone periodically synchronizes its time with the NTP server. The synchronization period is fixed at 1 hour. Between updates the phone tracks time with its internal clock.
- **SIP Messages**—Each SIP message (request or response) sent to the phone could contain a Date header with the current time information. If the header is present, the phone uses it to set its clock.

- **Manual Setup**—The time and date can be entered manually by using the IP phone screen or the phone web user interface. However, this value is overwritten by the NTP time or SIP Message Date whenever they are available to the phone. Manual setup requires that you enter the time in 24-hour format only.

The time served by the NTP Server and the SIP Date Header are expressed in GMT time. The local time is obtained by offsetting the GMT according to the time zone of the region.

The *Time Zone* parameter can be configured by using the phone web user interface or through provisioning. This time can be further offset by the *Time Offset (HH/mm)* parameter. This parameter must be entered in 24-hour format and can also be configured from the IP phone screen.

The *Time Zone* and *Time Offset (HH/mm)* offset values are *not* applied to manual time and date setup.

Configuring Daylight Saving Time

The phone supports auto adjustment for daylight saving time. You must set *Daylight Savings Time Enable* to **yes** and enter the DST rule. This option affects the time stamp on the *CallerID*.

To enter the rule for calculating DST, include the start, end, and save values separated by semi-colons (;) as follows:

```
Start = start-time; end=end-time; save = save-time
```

For example, the default DST rule is:

```
start=4/1/7;end=10/-1/7;save=1.
```

The *start-time* and *end-time* values specify the start and end dates and times of daylight saving time. The format is:

```
month/day/weekday [ /HH:mm:ss]
```

The *month* value equals any value in the range 1-12 (January-December).

The *day* value equals any + or - value in the range 1-31. If value is -1, the time will change on the weekday on or before the end of the month; the last occurrence of a weekday in that month.

The *weekday* value equals any value in the range -7 to 7 (Monday to Sunday). If the weekday value is 0, the date to start or end daylight saving is exactly the *month* and *day*. If the weekday value is -7 to 7, daylight saving starts or ends on the weekday value on or *after* the month and day. If the weekday value is not 0 and the day value is negative, then daylight saving starts or ends on the weekday value on or *before* the month and day.

Optional time values: *HH* represents hours (0-23), *mm* represents minutes (0-59), and *ss* represents seconds (0-59). Optional values inside brackets [] are assumed to be 0 if not specified. Midnight is represented by 0:0:0.

The *save-time* value is the number of hours, minutes, and/or seconds to add to the current time during DST. The *save-time* value cannot be a minus (-) in version 7.5.1 and higher. Earlier firmware versions support a negative value.

Daylight Saving Time Examples

The following example configures daylight saving time for the U.S, adding one hour starting at midnight on the first Sunday in April and ending at midnight on the last Sunday of October; add 1 hour (USA, North America):

```
start=4/1/7/0:0:0;end=10/31/7/0:0:0;save=1
```

```
start=4/1/7;end=10/-1/7;save=1
start=4/1/7/0;end=10/-1/7/0;save=1
```

The following example configures daylight saving time for Egypt, starting at midnight on the last Sunday in April and ending at midnight on the last Sunday of September:

```
start=4/-1/7;end=9/-1/7;save=1 (Egypt)
```

The following example configures daylight saving time for New Zealand (in version 7.5.1 and higher), starting at midnight on the first Sunday of October and ending at midnight on the third Sunday of March.

```
start=10/1/7;end=3/22/7;save=1 (New Zealand)
```

The following example reflects the new change starting March 2007. DST starts on the second Sunday in March and ends on the first Sunday in November:

```
start=3/8/7/02:0:0;end=11/1/7/02:0:0;save=1
```

Selecting a Display Language

This section describes how to localize the Cisco SPA300 Series and Cisco SPA500 Series IP Phone display language. You can define up to nine languages, in addition to English, to be available and host the dictionaries for each of the languages on the HTTP or TFTP provisioning server. Language support follows Cisco dictionary principles.



Note

The Cisco WIP310 does not support localization.

Use the Language Selection parameter to select the phone default display language. The value must match one of the languages supported by the dictionary server. The script (dx value) is as follows:

- `<Language_Selection ua="na">`
- `</Language_Selection>`

Defaults to blank; the maximum number of characters is 512. For example:

```
<Language_Selection ua="na"> Spanish
</Language_Selection>
```

During startup, the phone checks the selected language and downloads the dictionary from the TFTP/HTTP provisioning server indicated in the phone configuration. The dictionaries are available at the support website. See [Appendix B, "Where to Go From Here,"](#) for the website location.

The end user can change the language of the phone on the phone by following these steps:

-
- Step 1** Press the **Setup** button.
 - Step 2** Select **Language**, then press the **Select** soft button.
 - Step 3** Select **Option** to change the language.
 - Step 4** With the desired language highlighted, press **Save**.
-

Creating a Dictionary Server Script

The Dictionary Server Script defines the location of the dictionary server, the languages available and the associated dictionary. The syntax is:

```
Dictionary_Server_Script ua="na"/Dictionary_Server_Script
```

**Note**

TFTP, HTTP, and HTTPS are supported for the dictionary download.

Defaults to blank; the maximum number of characters is 512. The detailed format is as follows:

```
serv={server ip port and root path};  
d0=language0;x0=dictionary0 filename;  
d1=language1;x1=dictionary1 filename;  
d2=language2;x2=dictionary2 filename;  
d3=language3;x3=dictionary3 filename;  
d4=language4;x4=dictionary4 filename;  
d5=language5;x5=dictionary5 filename;  
d6=language6;x6=dictionary6 filename;  
d7=language3;x7=dictionary7 filename;  
d8=language8;x8=dictionary8 filename;  
d9=language5;x9=dictionary9 filename;
```

For example:

```
Dictionary_Server_Script ua="na"  
serv=tftp://192.168.1.119/;d0=English;x0=enS_v101.xml;d1=Spanish;x1=esS_v101.xml  
/Dictionary_Server_Script
```