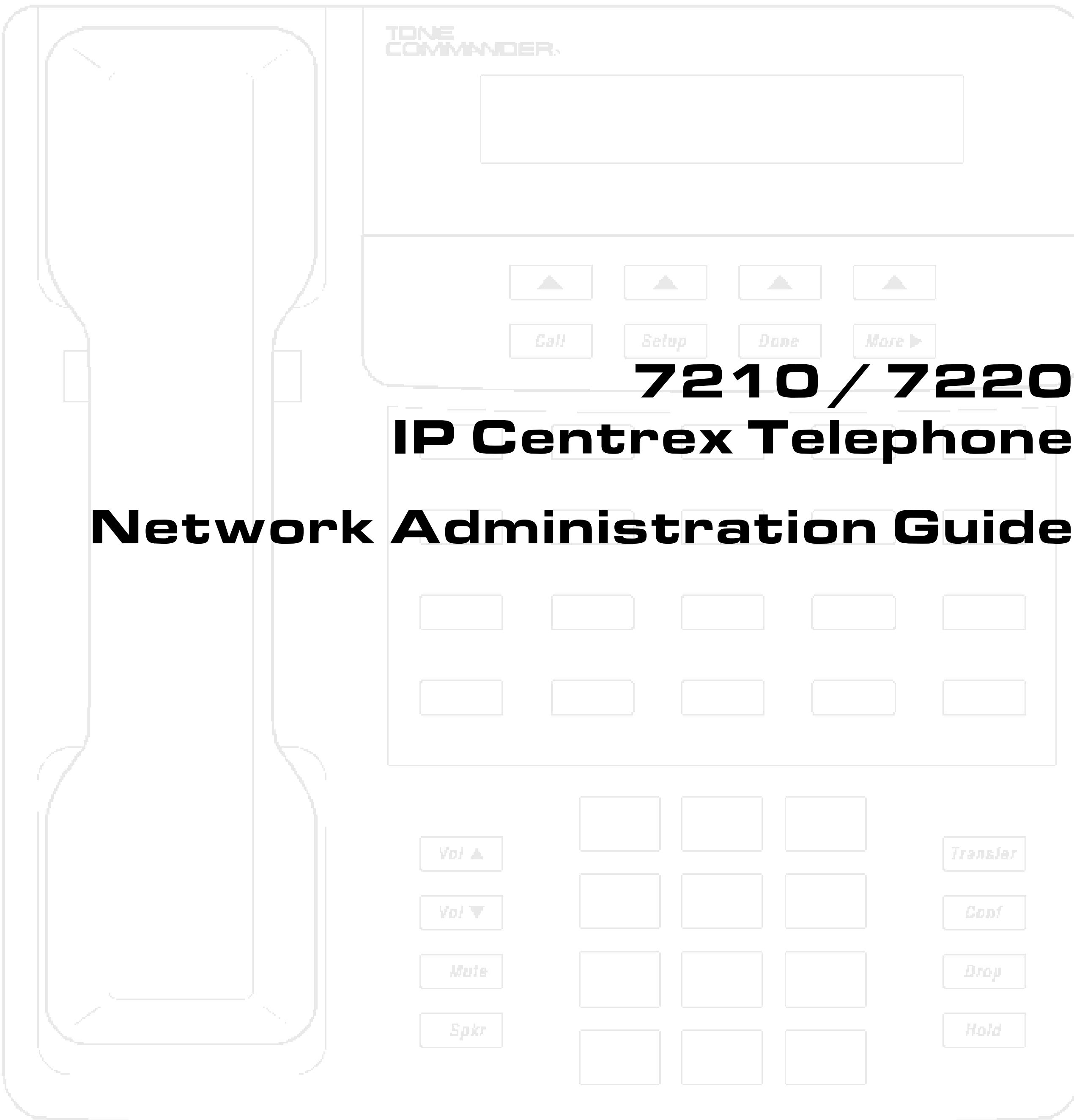
TONE COMMANDER®



7210 / 7220
IP Centrex Telephone

Network Administration Guide

13-102865 Rev. K
April 2003

*This document applies to telephone software version **01.02.18**.*

Contents

Overview

This manual is intended for use by network administrators.

Each telephone on the network must be assigned a unique IP address. The address can be assigned automatically by a DHCP server, or entered manually at the phone.

Telephone operating software can be updated from a TFTP server. The update process can be initiated manually at the phone, or set to occur daily at a preset time. Operating software is automatically updated during power-up or reset.

Procedures for setting up DHCP and TFTP servers, and updating the phone software, are explained below.

DHCP Server Configuration

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.

Examples for Windows® 2000 Server and Windows NT® Server follow.

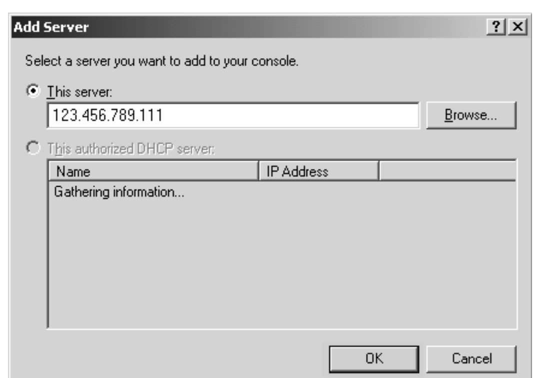
Windows 2000 Server

Run **DHCP** from the Administrative Tools menu.

Add Server

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

1. Select Add Server from the Action menu.
2. Enter the IP address of the DHCP server in the This server field.
3. Click OK.

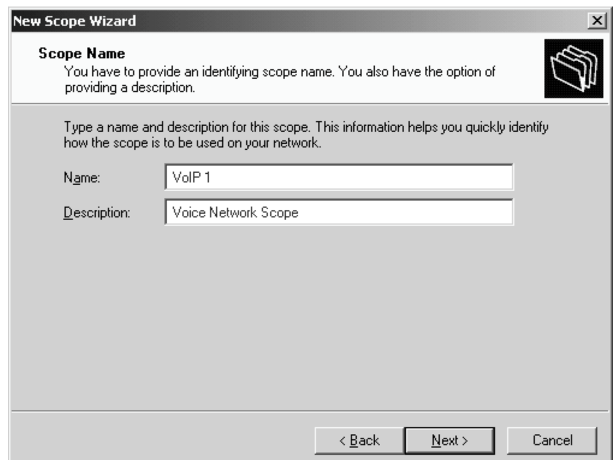


Add Scope

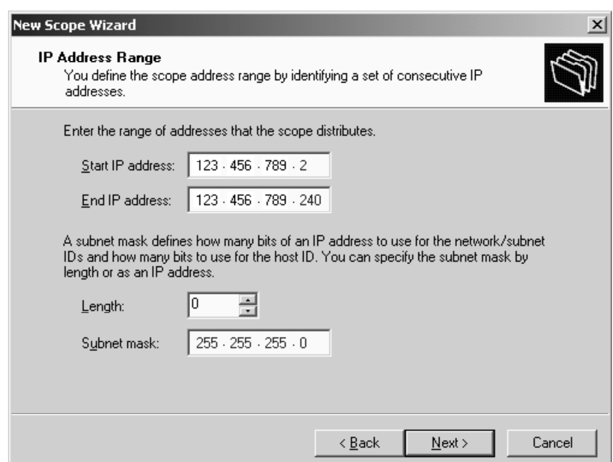
1. Select the DHCP server you will use for assigning IP addresses to the telephone.
2. Select **New Scope** from the Action menu to run the New Scope Wizard.

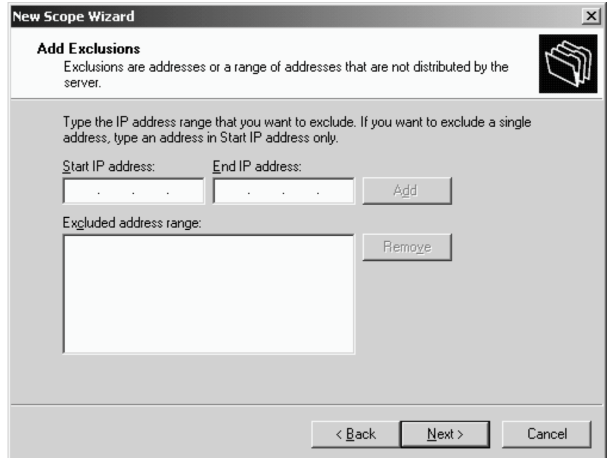
3. Click Next to show the Scope Name screen.

4. Enter a name and description for the scope.

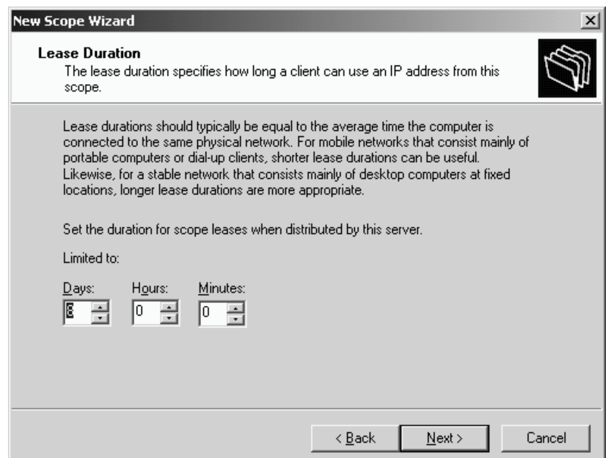


5. Enter the start and end of the IP address range that can be assigned to telephones.
6. Enter the appropriate subnet mask.



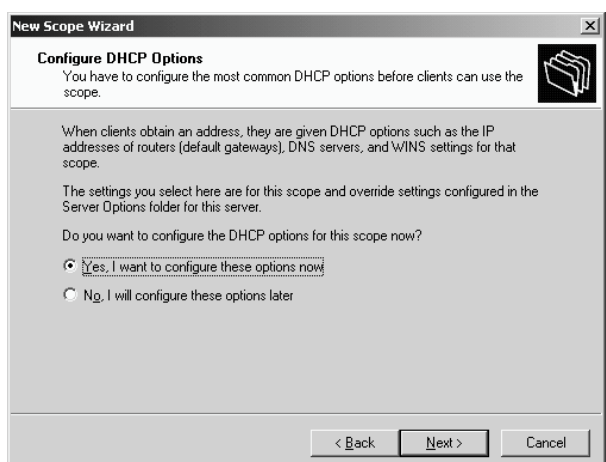


7. If you need to exclude some IP addresses from the range, enter them here, otherwise click **Next**.

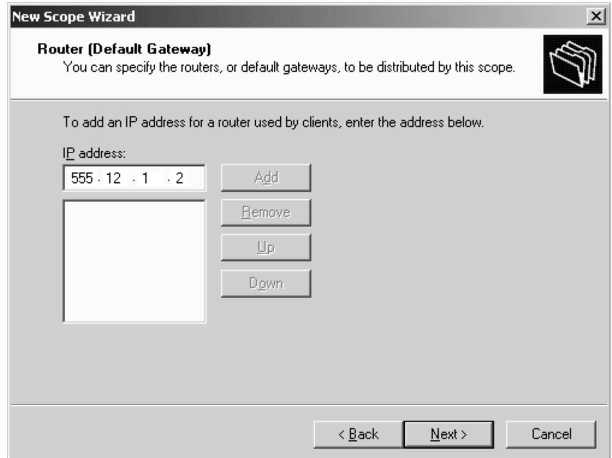


8. You can use the defaults, or enter a new lease duration for telephone IP addresses. Lease duration should be set to 7 days or longer.

> *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.*



9. Select **Yes** on the Configure IP Options screen.

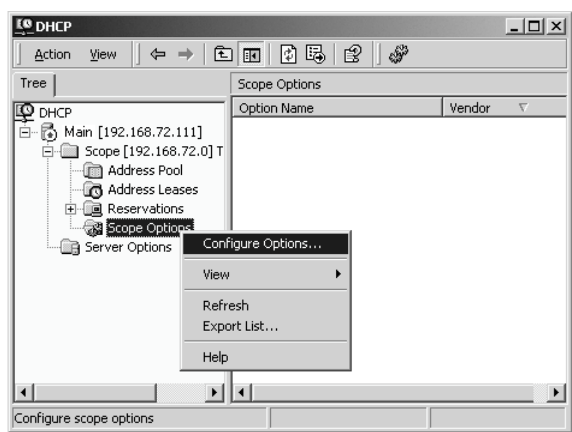


10. Enter the IP address of the router or default gateway.
11. If needed, enter the parent domain name, DNS servers, and WINS servers on the next two screens.

12. Select **Yes** to activate this scope.

13. Click **Finish** on the next screen.

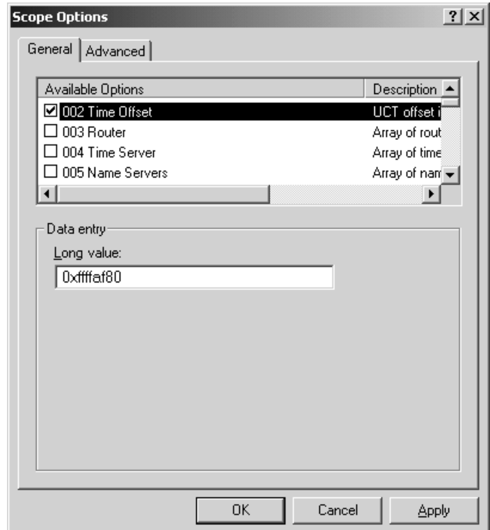
Scope Options

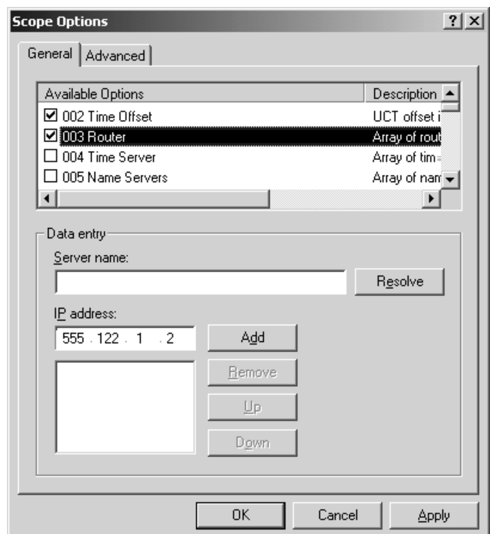


1. In the Tree pane, expand (double-click or '+') the VoIP DHCP server.
2. Expand the Scope entry for the telephone IP address range.
3. Right-click **Scope Options**, then select **Configure Options** from the menu.

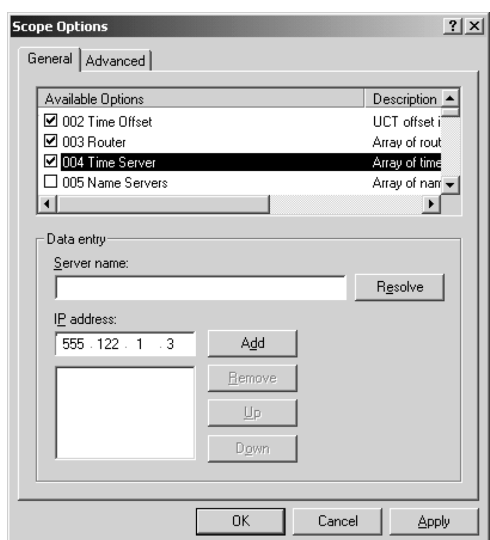
4. Check the box next to 002 Time Offset.
5. Select **002 Time Offset.** If your network time server is set to UTC time, enter the hex value for your location's offset from UTC time in seconds. If your network time server is set to local time, enter **0**.

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

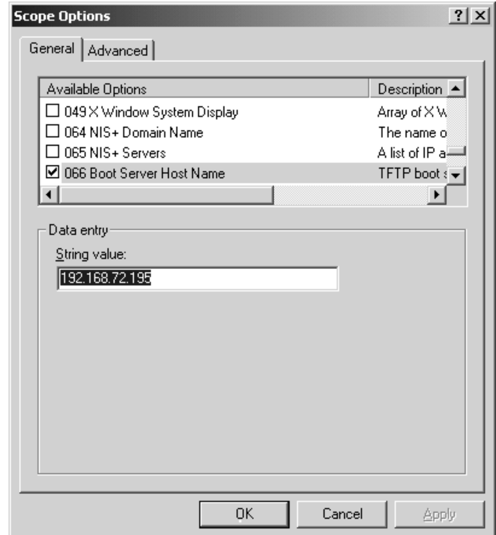




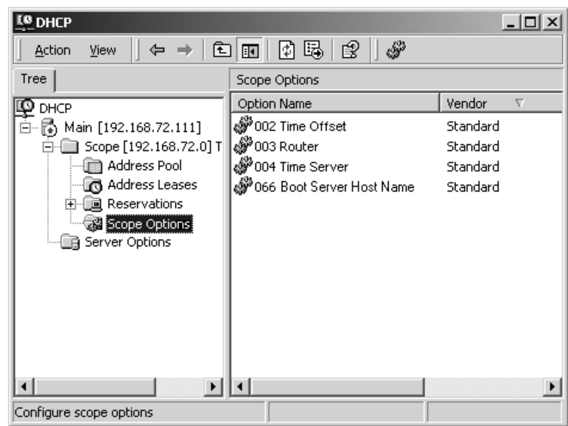
6. Make sure the box next to **003 Router** is checked.
7. The router IP address may have been entered from the New Scope Wizard. If not, select **003 Router**, enter the IP address, then click **Add**.



8. Check the box next to **004 Time Server**.
9. Select **004 Time Server**, enter the IP address of the SNTP time server for your network, then click **Add**.



10. Check the box next to **066 Boot Server Host Name**.
11. Select **066 Boot Server Host Name**, then enter the TFTP boot server's IP address in the **String value** field.
12. Click **OK**.



13. "002 Time Offset", "003 Router", "004 Time Server", and "066 Boot Server Host Name" should now appear in the Scope Options pane.

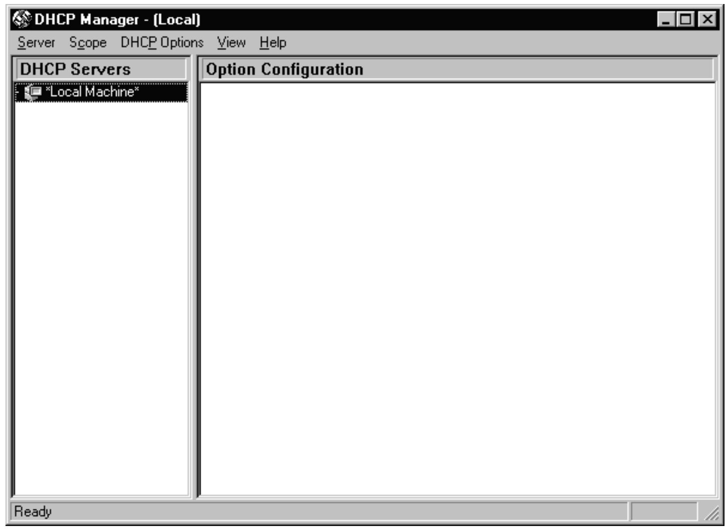
To preclude the issuance of a new IP address each time the phone reboots, it is recommended that ICMP ping prior address assignment be disabled at the DHCP server.

Windows NT Server

Run **DHCP Manager** from the Administrative Tools menu.

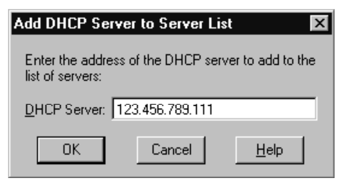
Add Server

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

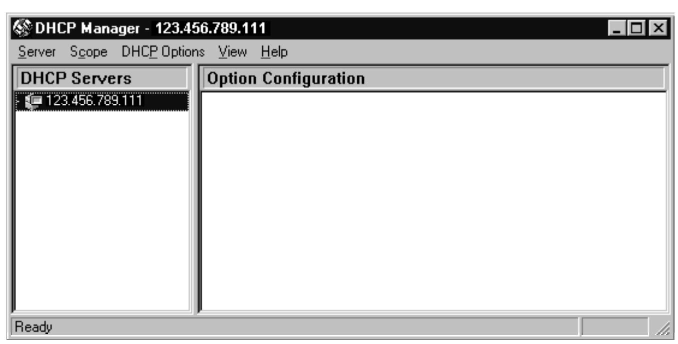


1. To add a server, select **Server** from the File menu, then select **Add**.

2. Enter the address of the new DHCP server.

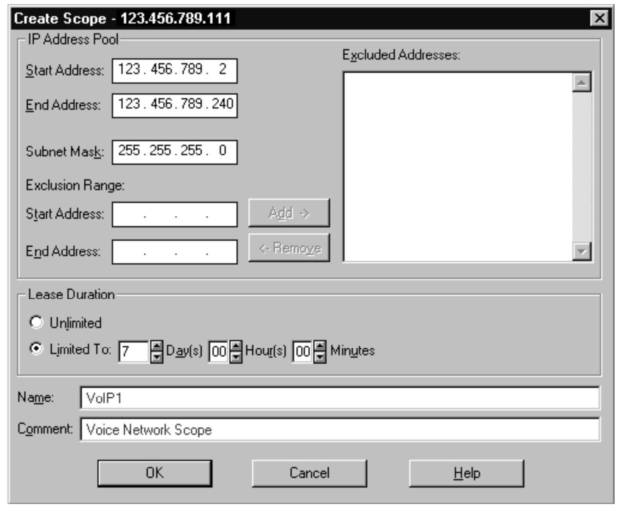


Add Scope



1. Select the DHCP server.
2. Select **Create** from the Scope menu.

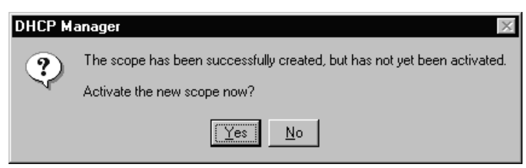
3. Enter the start and end of the IP address range that can be assigned to telephones.
4. Enter the appropriate subnet mask.
5. Enter a name for the scope in the Name field, and an optional description in the Comment field.
6. Exclude addresses from the range and change the lease duration if needed. Lease duration should be set to 7 days or longer.



> *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.*

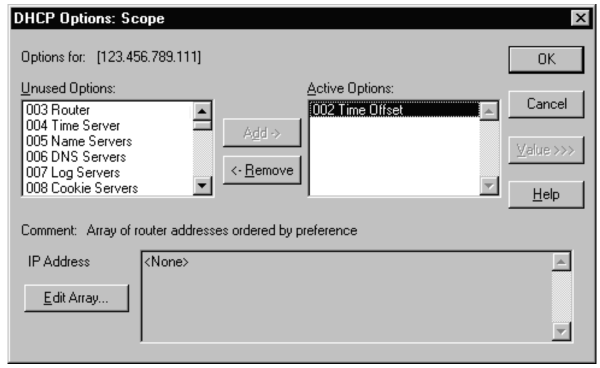
7. Click **OK**.

8. Click **Yes** to activate the scope.



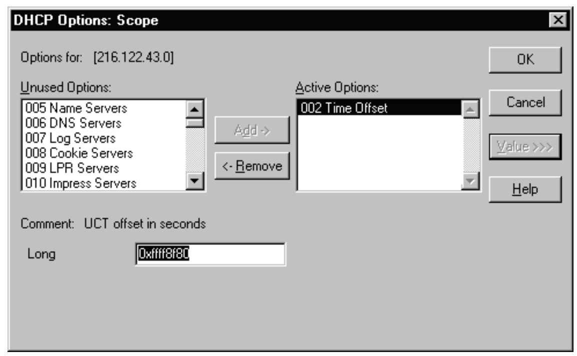
Scope Options

1. Select **002 Time Offset** from the Unused options list and click **Add** to add it to the Active Options list.
2. Click **Edit Array**.

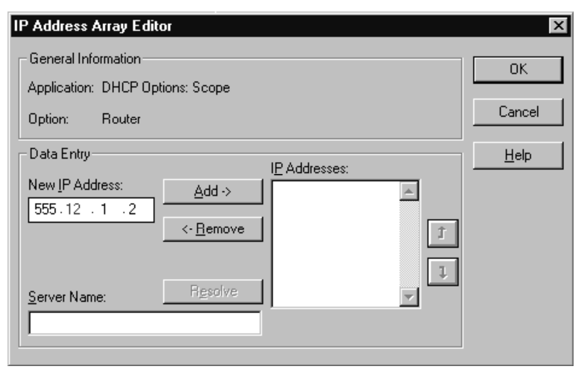


3. If your network time server is set to UTC time, enter the hex value for your location's offset from UTC time in seconds. If your network time server is set to local time, enter **0**. Click **Add**.
4. Click **OK**.

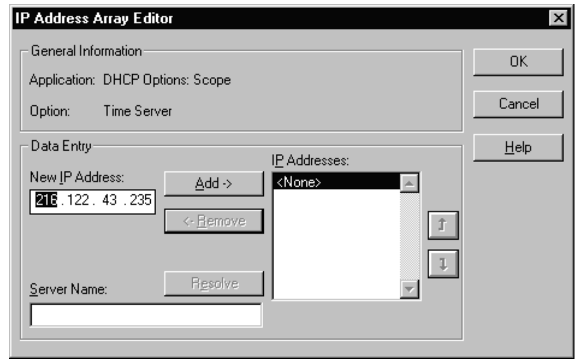
Time Zone	Offset
Pacific Standard Time	0xffff8f80
Pacific Daylight Time	0xffff9d90
Mountain Standard Time	0xffff9d90
Mountain Daylight Time	0xffffaba0
Central Standard Time	0xffffaba0
Central Daylight Time	0xffffb9b0
Eastern Standard Time	0xffffb9b0
Eastern Daylight Time	0xffffc7c0



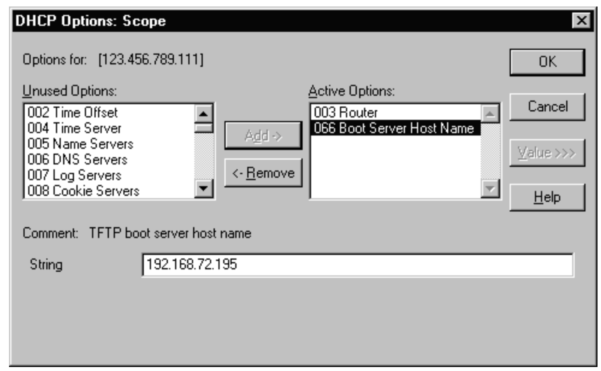
5. Add **003 Router** to the Active Options list.
6. Click **Edit Array**.
7. Enter the router IP address, then click **Add**.
8. Click **OK**.



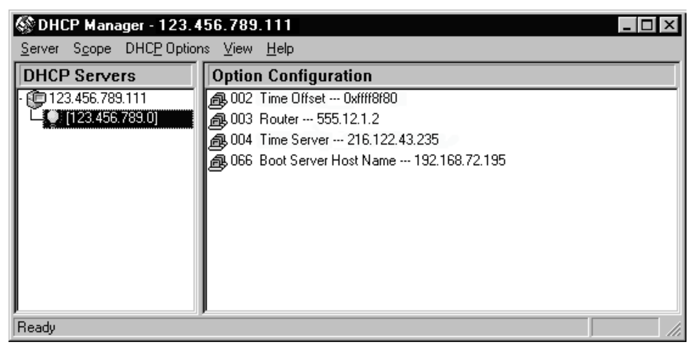
9. Add **004 Time Server** to the Active Options list.
10. Click **Edit Array**.
11. Enter the IP address of the SNTP time server for your network, then click **Add**.
12. Click **OK**.



13. Add **066 Boot Server Host Name** to the Active Options list.
14. Enter the TFTP boot server's IP address in the **String** field.
15. Click **OK**.



16. "002 Time Offset", "003 Router", "004 Time Server, and "066 Boot Server Host Name" should now appear in the Scope Options pane.



To preclude the issuance of a new IP address each time the phone reboots, it is recommended that ICMP ping prior address assignment be disabled at the DHCP server.

TFTP Server Configuration

Telephones download configuration information and software upgrades from a TFTP server. The TFTP server's IP address must be identified during DHCP server setup, or entered manually in the phone if DHCP is disabled.

Configuration packages distributed by Tone Commander include a phone boot ROM image file, application software image file, two configuration files, and a Readme text file.

All files must be located in the TFTP server's root folder .

The root folder should contain the following files:

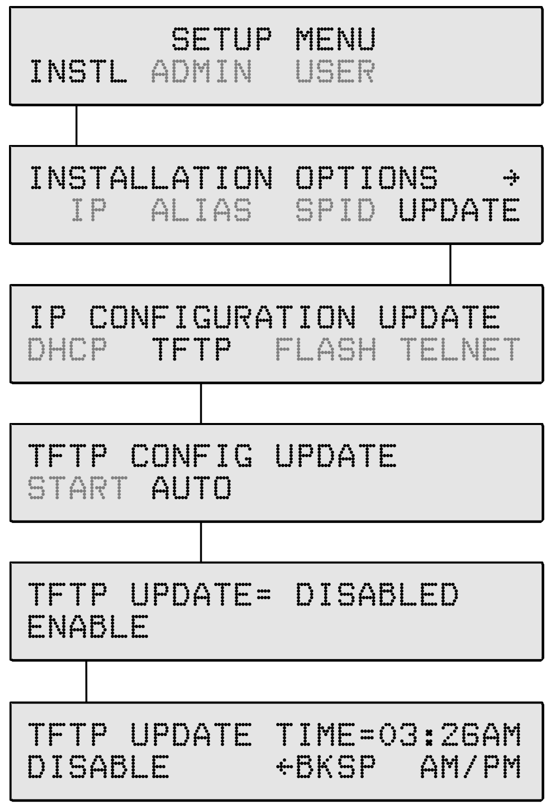
***filename*.bin**	Boot program image.
***filename*.z**	Compressed application image.
tcs7200a.txt	Configuration options not included in the standard DHCP options.
tcs7200b.txt	Names the application file and boot program to be downloaded to the phone. These files need to be in the TFTP server root folder. The phone will attempt to download parameters or programs only if the information in the above .txt files indicates that the phone is not at the desired current configuration.
readme.txt	information file
***xxxxxxxxxxxxxx*.txt**	Optional files allow customers to set up phones with specific configuration parameters or program versions based on the phones' alias (*xxxxxxxxxxxxxx* in the file name, e.g., 30947701840101). Any information that is different in this file overrides the information in the tcs7220a,b files. A separate file is required for each phone that differs from the standard configuration or programming as defined in tcs7200a.txt and tcs7200b.txt files.

Configuration files can be modified with any text editor. See page 29 for file format descriptions.

Telephone Configuration Update

Telephone operating software and configuration parameters can be automatically updated daily at a preset time, or manually updated.

Automatic Update



1. Select **AUTO** from the TFTP CONFIG UPDATE menu.

   (***Setup*** → INSTL → UPDATE→ TFTP→ AUTO)
2. **Enable** TFTP Update if necessary.
3. Enter the time you want the daily update to occur.

   The minutes value is calculated based on the MAC address of the phone, to minimize the possibility of multiple phones simultaneously requesting updates. You may manually enter the minutes digits, but it is recommended that you use the default calculated value.
4. Press the ***Done*** key.

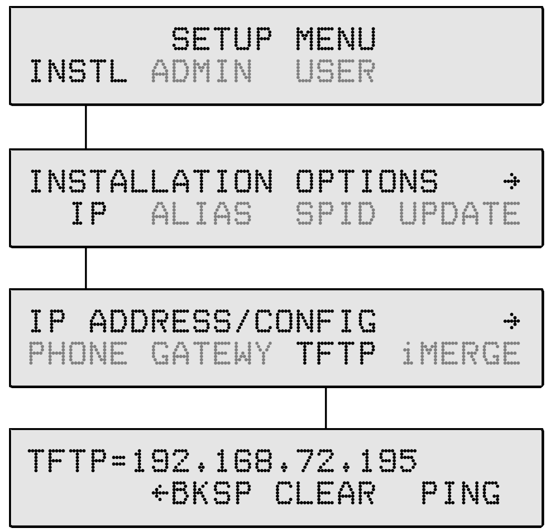
NOTE – For special instructions and information, please refer to the Upgrading Notes associated with a specific upgrade.

Manual Update

To perform a manual update, the phone must have the TFTP server's IP address entered manually or downloaded via DHCP.

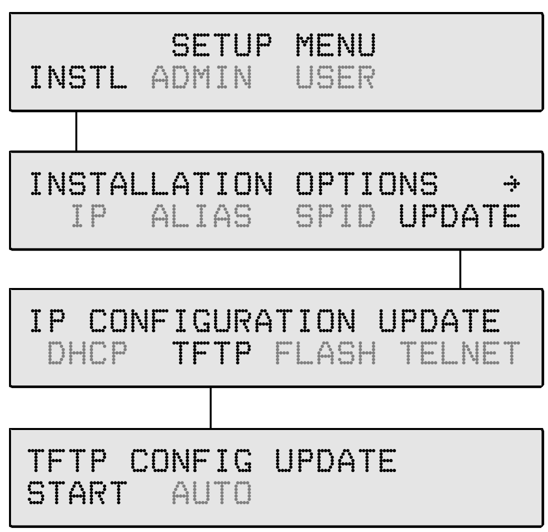
Manual TFTP Server Entry

1. Select **TFTP** from the **IP ADDRESS/CONFIG** menu.
2. Enter the TFTP server address with the dial pad.
3. Press the ***Done*** key.



Starting Manual Update

Use the **TFTP CONFIG UPDATE - START** option to initiate the manual update.



Quality of Service

Quality of Service (QoS) settings can improve voice performance over a network by prioritizing voice packets, and adjusting packet buffering and packetization rate.

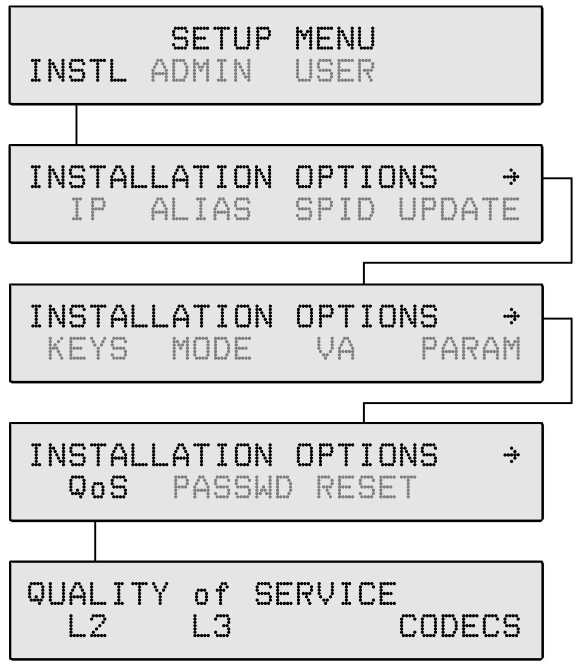
These settings affect network traffic, and should not be changed unless required to correct audio problems.

To view or change Quality of Service settings, select QoS from the Installation Options menu.

(***Setup*** → INSTL → ***More***▸ → ***More***▸ → QoS)

Layer 2, Layer 3, and Codecs can be selected from the Quality of Service menu.

When finished viewing or changing any setting, press the ***Done*** key to return to the previous menu or the ***Setup*** key to exit Setup Mode.

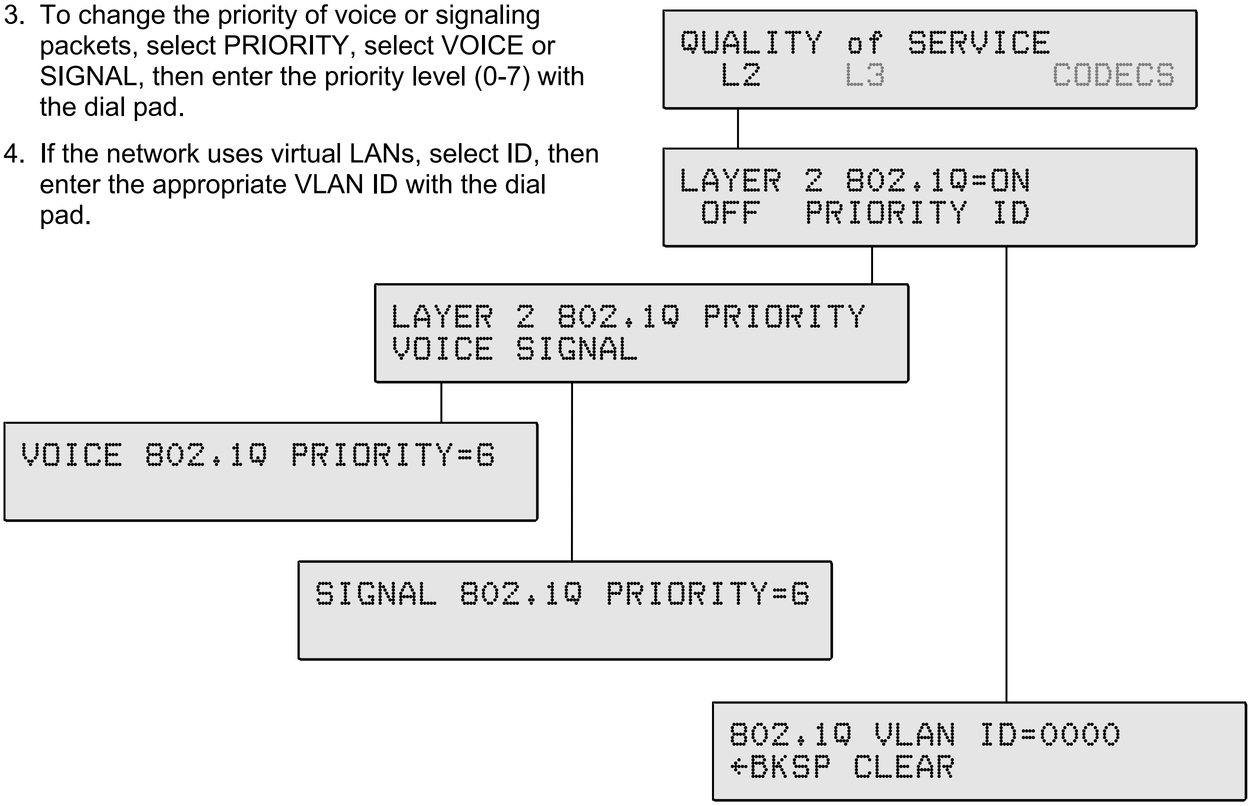


Ethernet Layer 2 802.1Q Options

IEEE 802.1Q allows packets to be assigned one of eight priority levels. Voice traffic with less than 10 ms of latency is normally assigned a priority level of *6* (phone default). Network switches must support 802.1Q for this setting to have an effect. If the LAN does not support 802.1Q, this parameter should be set to OFF.

NOTE – The Ethernet card in a PC connected to the phone must support 802.1Q, since the phone does not add 802.1Q tagging to packet headers not generated in the phone. PC packets are passed through without modification.

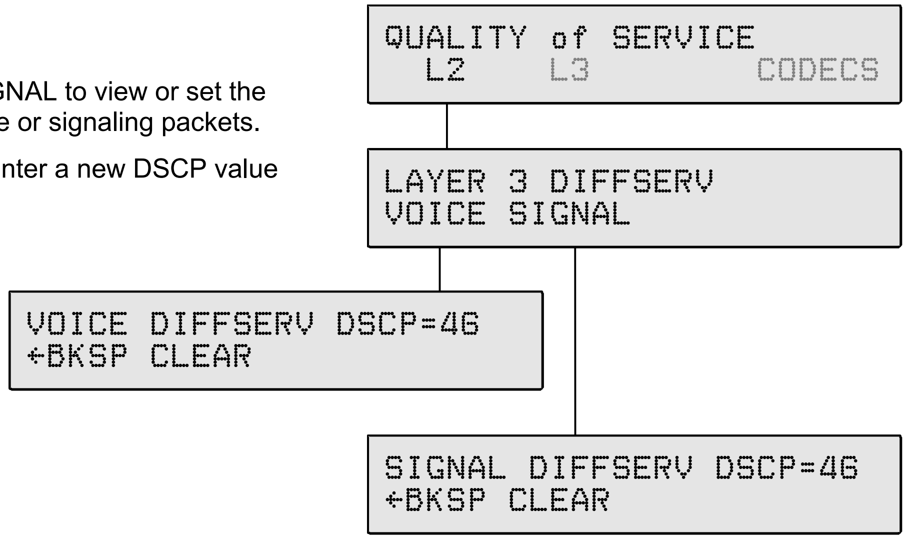
1. Select L2.
2. Select ON or OFF to enable/disable Layer 2 802.1Q support.
3. To change the priority of voice or signaling packets, select PRIORITY, select VOICE or SIGNAL, then enter the priority level (0-7) with the dial pad.
4. If the network uses virtual LANs, select ID, then enter the appropriate VLAN ID with the dial pad.



IP Layer 3 Differentiated Services (DiffServ)

The Quality of Service for voice and signaling packets is determined by each service type's Differentiated Services Code Point (DSCP) setting. This value must be matched to network router settings. The default setting for both voice and signaling is *46.*

1. Select L3.
2. Select VOICE or SIGNAL to view or set the DSCP value for voice or signaling packets.
3. Use the dial pad to enter a new DSCP value (0-63).



Codecs

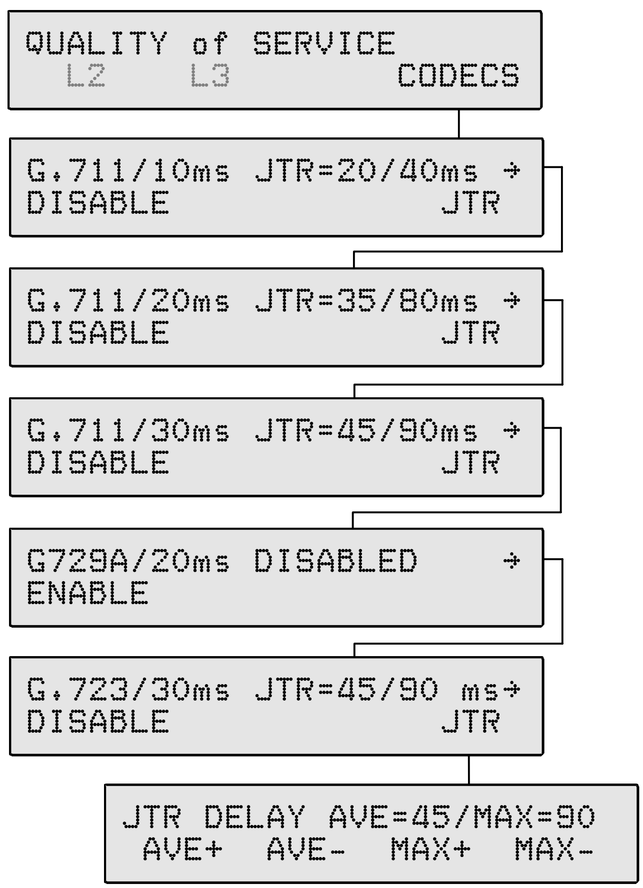
Five codec selections are available. Any codec/packet rate combination can be disabled.

1. Select CODEC.
2. Cycle through the codecs with the ***More*** ▸ key. The display will show the codec type (G.711, G.729A, or G.723), the packet rate (10, 20, or 30ms), and the average and maximum jitter delay for each enabled codec.
3. Enable or disable codecs as required.

   If more than one codec is enabled, the selected codec is negotiated on a per-call basis, between the list of codecs enabled on iMerge and codecs enabled on the phone. iMerge codecs take precedence.

   You can view the negotiated codec selection through the Packet Diagnostics menu (see below).

> ***CAUTION** – Make sure at least one codec that is supported by the iMerge CFG is enabled, otherwise a voice channel cannot be established.*



Codec Selections

G.711 – Uncompressed, 64Kbps data rate (10, 20, or 30 ms packet rate)
G.729A – Compressed, 8Kbps data rate (20 ms packet rate)
G.723 – Compressed, 6.3Kbps data rate (30 ms packet rate)

Uncompressed codecs with higher packet rates (e.g. G.711/10ms) may provide better voice performance with lower audio delay, but increase network traffic.

4. Average and maximum jitter delay can be set for each enabled codec. Select JTR, then change the delay values with AVE+, AVE-, MAX+, and MAX-. Note that the maximum jitter delay cannot be set to less than twice the average jitter delay setting.

The average jitter delay is the average amount of time that packets are received before they are played. Since IP networks have variable packet transmission delays, yet packets must be played at a constant rate, a local jitter buffer is required to “smooth out” the variations in packet arrival times. The larger the variance in packet delay through the network, the larger the average jitter delay setting must be to compensate. Audio dropouts may occur (due to delayed packets) if the average jitter delay is too small. Unacceptable audio delay may result if the jitter delay is too long.

Use the Packet Diagnostics menu (see below) to review packet statistics and jitter performance. If a significant number of packets (>5%) are concealed and a significant portion of packets are delayed longer than the average jitter buffer delay setting, the average jitter buffer setting should be increased to “capture” these dropped packets. If there are consistently very few concealed packets and almost all received packets are delayed less than the average jitter delay setting, this setting can likely be decreased without increasing the number of concealed packets to improve audio delay.

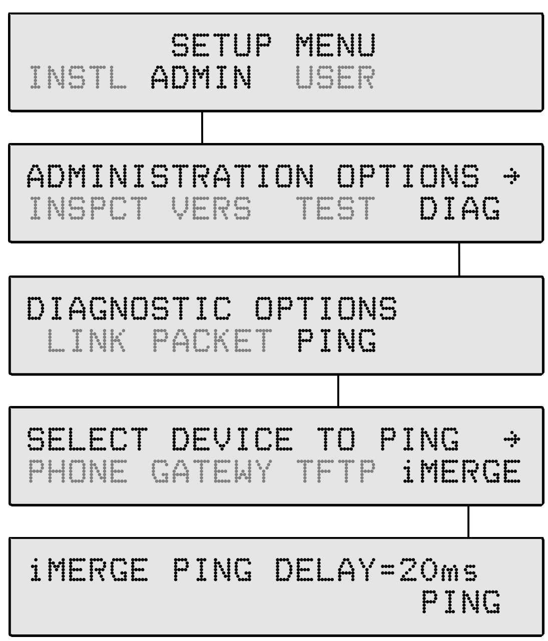
The maximum jitter delay is the longest delay allowed until playback for a packet that arrives early. This setting also affects dropped packets due to clock slips on long-duration calls. If the iMerge packet transmission rate is slightly faster than the phone playback rate, then the jitter buffer will gradually fill up until it reaches the maximum capacity and overflows (after an hour or two). When this happens, an audio “skip” will occur as the jitter buffer is reset to the average delay setting. Immediately before the overflow correction occurs, the additional packet delay due to the jitter buffer is as long as the maximum jitter delay. Therefore, the maximum jitter delay should be set long enough so that overflow events do not happen very often, but not so long that excessive audio delay occurs before a correction is made.

Diagnostic tools built into the 7210/7220 can assist you in determining the optimum jitter delay settings for your network. The ping test provides a quick method of measuring single packet network delays. For a more detailed picture of packet delay and loss during actual calls, use the Packet Diagnostics menu.

Ping Test

You can test network delays by pinging the iMerge server (***Setup*** → ADMIN → DIAG → PING → iMERGE). The ping delay will be shown in the display. Select PING several times to perform multiple ping tests and note the difference between delay measurements.

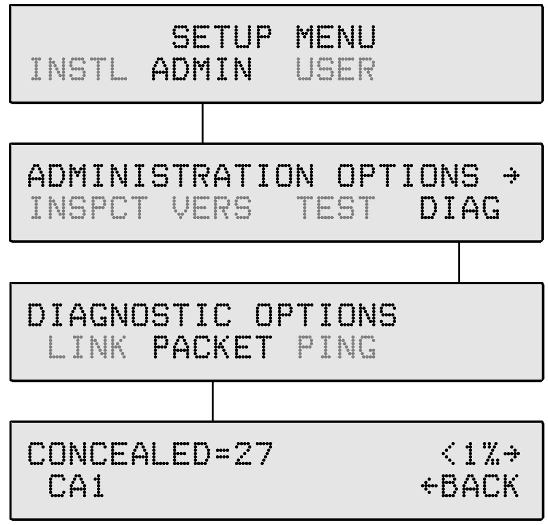
The <u>difference</u> between readings (not absolute delay values) gives an “order of magnitude” indication of the average jitter buffer setting needed to prevent dropped packets that result in audio interruptions.



Packet Diagnostics Menu

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.

Packet statistics are viewed through the Packet Diagnostics menu (***Setup*** → ADMIN → DIAG → PACKET). Concealed packet statistics are displayed first; press the ***More*** ▸ key to view additional packet types.



The following statistics are recorded and displayed:

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.

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).

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).

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.

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.

Codec/Jitter Buffer Settings – shows the negotiated codec and associated jitter buffer selections for the current call. These values are based on the codec and jitter buffer settings in the phone, as well as iMerge codec settings, and are negotiated on a per-call basis.

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.

Troubleshooting

7210 and 7220 telephones have built-in diagnostic, logging, and testing capabilities to quickly isolate problems affecting their operation.

Network 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.

Problem Observed	Remedial Action
No display information is shown.	Check power connections and source.
"NO ETHERNET CONNECTION" is shown continuously.	Check connections to the LAN or WAN.
"DHCP ERROR RETRYING" etc… is shown continuously.	Verify that the DHCP server is operating and accessible. If the LAN/WAN does not include a DHCP server, disable IP configuration via DHCP and enter the appropriate values (IP address, default gateway, TFTP address), using the INSTL→IP menu.
"iMERGE= "	There is no iMerge address (new phone). Enter the appropriate iMerge IP address.
"ALIAS= " is shown continuously.	Either there is no Alias (new phone) or the previously-entered Alias has been rejected by iMerge (Error Log of the phone reports "INVALID ALIAS"). Enter the appropriate Alias.
"H235 PW= " is shown continuously.	Either there is no H.235 password (new phone) or the previously-entered password has been rejected by iMerge (the Error Log of the phone reports "SECURITY DENIAL"). Enter the appropriate password or press the ***Done*** key without entering a password, if a password is not required at iMerge.
"RETRY INITIALIZATION?" is shown continuously.	The IP address, default gateway, or subnet mask have been changed from the Setup menu. Select 'YES' to initialize with new values.
"ENTER PRIMARY PHONE # - - " is shown continuously.	Auto-SPID is not supported by the telco network. The phone prompts the user to enter the ten digit DN of the phone. This entry will construct a generic format SPID (for National ISDN).
"ENTER PRIMARY PHONE # - " is shown continuously.	The phone prompts the user to enter the seven digit (no area code) primary DN of the phone. This entry will construct a Custom format SPID (for Lucent Custom ISDN).
"ID= " is shown continuously.	The SPID sent previously has been rejected by the telco network. The user must enter the actual SPID.
"IP:DHCP FAIL", etc. is shown continuously.	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.
"PHY:100MBPS IP:LINKED L3:DOWN SWITCH:UNKNOWN" is shown continuously.	ISDN Layer 3 of the phone is not initialized. Select RESTART from the ADMIN Menu. Entry of the actual SPID may be required.

Problem Observed	Remedial Action
"NO TFTP SERVER ADDRESS " " EDIT "	There is no TFTP address and a manual or automatic software version query was attempted by the phone. Enter the appropriate TFTP address.
"NO TFTP SERVER RESPONSE " " EDIT "	A manual or automatic software version query was attempted by the phone, but the addressed TFTP server did not respond. Using the ADMIN/DIAG/PING/TFTP menu, ping the programmed address. If there is no response to the ping: 1) verify that the TFTP server is operational; 2) verify that the programmed address is correct and edit if necessary.
"1= " The above is shown and no dial tone is received after an attempt to originate a call.	If iMerge does not respond to "keep alive" messages sent by the phone, recovery procedures are initiated by the phone to re-establish the link to the iMerge CFG. If an attempt to originate a call is made during these procedures, the "1= " display and no receipt of dial tone will be observed. Wait or initiate a restart via the ADMIN menu.
"PHY:100MBPS IP:LINK LOST " "L3:DOWN SWITCH=(switch type)"	If recovery procedures to re-establish a lost link are not successful after a predetermined interval, this display is shown. The phone will then attempt to re-register with the iMerge CFG. Wait or initiate a restart via the ADMIN menu.
"REGISTERING WITH iMERGE " "NO RESPONSE FROM iMERGE " *The above is shown for two to three minutes.*	Procedures initiated by the phone to register or re-establish a link to iMerge CFG have failed. Continue to wait or: 1) verify that the phone, default gateway, and iMerge IP addresses are correct; 2) ping the default gateway. If it responds, go to Step 3. If there is no response, verify the integrity between the phone and the default gateway; 3) ping the iMerge CFG. If there is no response to the ping, verify that pinging is enabled at the iMerge CFG. If it is enabled, investigate the integrity between the local gateway and the iMerge CFG. If pings are disabled at the iMerge CFG, set up a network sniffer application to trace messages coming to and from the phone for a period of at least two minutes. The trace should show H.225 REGISTRATION REQUESTS going to the iMerge CFG and H.225 REGISTRATION CONFIRMED messages returning. If the former is not shown, restart the phone via the ADMIN menu. If the latter is not shown, investigate the integrity between the local gateway and the iMerge CFG.
"REGISTERING WITH iMERGE " "D-CHAN SETUP FAILED "	This display may be shown temporarily whenever the phone is registering with the iMerge CFG. If the display persists, restart the phone.
"PING UNSUCCESSFUL " " PING" - or - "PING FAILED " "PRESS ANY KEY "	If all attempts to ping valid 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 IP addresses using either the INSTL/IP or the ADMIN/DIAG menus.
"FILE NOT FOUND ON TFTP " "TCS7200B.TXT "	This display is typically shown after a manual or automatic attempt to download new software and the TFTP server resides on a different subnet. Verify that all router/gateways are set for the "fast mode". Router/gateways set to this mode will immediately respond to ARPs from the phone, allowing it to find the TFTP server within the time restraints of the download process.

Call Control Troubleshooting

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

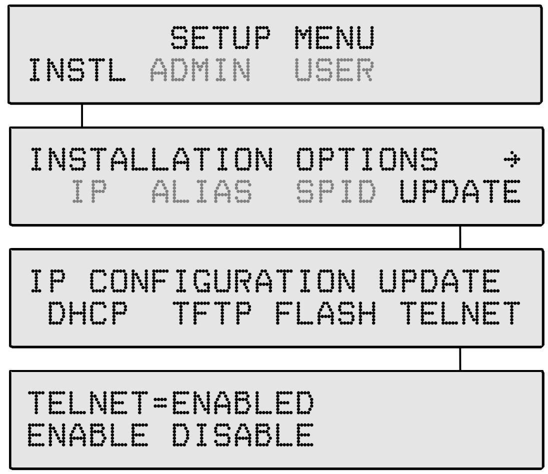
Problem Observed	Remedial Action
Going off-hook or pressing the ***Spkr*** key does not automatically select a CA/DN and draw dial tone. Selecting a CA/DN or hotkey dialing while on-hook does, however.	This is the behavior of a phone that is connected to a National ISDN network, with Call Preference set to 'NONE'. If this behavior is not desired, change the Call Preference value to either RING or IDLE, using the USER→PREF menu.
When on-hook, selecting any CA/DN does not return dial tone. The user must go off-hook or press the ***Spkr*** key first.	This is the behavior of a phone that is connected to a Lucent Custom ISDN network that is not optioned for 'One Touch' in network translations. If this behavior is not desired, request enabling of this attribute in network translations.
When selecting a Feature Activator key while on-hook, nothing happens.	Some features require an active call to function. If the phone is connected to a National ISDN network, select the FA(CALL) option using the INSTL→KEYS→FA menu. If the phone is connected to a Lucent Custom ISDN network, request enabling of the 'One Touch' attribute in network translations. In both cases, a call will be originated using the speakerphone.
When attempting to transfer a call, the LED for the ***Conf*** key illuminates rather than for the ***Tran*** key.	This is normal behavior of a phone that is connected to either a DMS100 or EWSD switch that is NI-1 compliant. If either switch is NI-2 compliant, this condition will not occur.
When originating calls either by going off-hook or pressing the ***Spkr*** key, not all CA/DNs are accessible.	Verify that the affected CA/DNs are optioned for 'ORIGINATING DN=YES' and not optioned for 'RESERVED=INCOMING ONLY', using the INSTL→KEYS→CA/DN menu.
When inspecting the DN List, there are no entries.	The DN List is filled automatically with the last ISDN parameter download (PDL). If the host switch does not support PDL, DNs must be entered manually using the INSTL→KEYS→CA/DN menu.
All CA/DNs indicate the arrival of inbound calls via LEDs; however, the phone does not ring for some of them.	Verify that the affected CA/DNs are not set for NEVER or an extended WAIT interval, using the USER→RING→CONTROL MENU. **Note!** When connected to a Lucent Custom ISDN network, individual CAs can be optioned in network translations not to ring.
All CA/DNs indicate the arrival of inbound calls via LEDs; however, the phone never rings.	Verify that "RINGER OFF" is not showing in the display. If it is, use the ***Vol***▲ key to set the ringer level to a value higher than OFF.
Going off-hook or on-hook with the handset does nothing. When a CA/DN or the ***Spkr*** key is pressed, a CA/DN is selected with the appropriate display, but dial tone cannot be heard via the speakerphone.	This is normal operation while in the HEADSET mode. If this behavior is not desired, change the Voice Mode to HANDSET, using the USER→VOICE menu.

Telnet

A telnet client can connect to the phone for troubleshooting purposes.

Use this option to view or clear the error log when you are not physically present at the phone location. The error log can also be accessed from the phone's Administration Options menu. Other options available through telnet should be used *only* under the direction of Tone Commander support personnel.

1. Telnet must be enabled at the phone before establishing a connection (***Setup*** → INSTL → UPDATE → TELNET → ENABLE).



2. Start the telnet application (Windows includes a telnet client; click Run in the Start menu, then enter **telnet**). Set the terminal type to **vt100**.
3. Connect to the phone's IP address. This address must be accessible from your PC location.
4. Enable logging to a file. This will allow you to print or save information that is displayed on your screen.

   A "`->`" prompt will appear when you connect to the phone.
5. Enter **menu** to view a list of options.

```
-> menu
<enter>  Exit
      1. TCS
      2. AGCS
      3. DUMP flash logs
      4. CLR flash logs
>
```

6. Enter **3** to view the phone's error log, or **4** to clear the log in the phone.

> *Options 1 and 2 can adversely affect the operation of the 7210/7220. Use these options only when directed to do so by Tone Commander.*

7. When finished, disconnect the telnet client.

> *DO NOT enter **exit** or **quit**. These commands will stop the telnet option in the phone until the next phone restart.*

Reference

DHCP Messages

7210/7220 Client Discover Options

The DHCP DISCOVER message broadcast by the 7210/7220 telephone uses the following options:

Parameter Request List	Option 0x37 (55 dec)	Info being requested from DHCP server is as follows: Subnet Mask — Option 0x01 Time Offset — Option 0x02 Router — Option 0x03 Time Server — Option 0x04 Broadcast Address — Option 0x1c (28 dec) Vendor Specific Info — Option 0x2b (43 dec) Requested IP Addr. — Option 0x32 (50 dec) Renewal (T1) Time — Option 0x3a (58 dec) Rebinding (T2) Time — Option 0x3b (59 dec) TFTP Server — Option 0x42 (66 dec)
Class Identifier	Option 0x3c (60 dec)	Setting to “TCS.7220” value indicates 7210/7220 phone
Client Identifier	Option 0x3d (61 dec)	MAC address of the 7210/7220 phone

DHCP Server Options

The DHCP server DHCPACK/DHCPOFFFER message should contain the following:

Phone IP Address	yiaddr fixed field	IP address for lease that server is offering phone
Subnet Mask	Option 0x01	
Time Offset	Option 0x02	Offset from UTC time
Default Gateway	Option 0x03	
Time Server	Option 0x04	SNTP Time Server IP address
Vendor Specific	Option 0x2b (43 dec)	TCS.7220 in field identifies server as capable of providing download information to phone
Renewal Time	Option 0x3a (58 dec)	Relative time to renew lease
Rebinding Time	Option 0x3b (59 dec)	Relative time to rebind lease
Lease Time	Option 0x33 (51 dec)	Relative time until lease expires for IP address
Server Identifier	Option 0x36 (54 dec)	Identifies DHCP server
TFTP Server	Option 0x42 (66 dec)	Location of configuration data/files for phone

TFTP Server Configuration File Formats

Keywords assign parameters/program versions as shown here:

<keyword>=<parameter>

All keywords that are valid for a file type are optional.

Do not include leading '0' characters in IP addresses; follow the examples shown here.

Comments can be put into the TFTP server files by preceding them with a semicolon. *Separate the semicolon from the preceding value with at least one space.*

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

Line length must be limited to 110 characters (80 characters for phone software version 01.02.05 and prior).

tcs7220a.txt

Configuration options not included in the standard DHCP options.

Valid keywords:

IMERGE=ip	IP address for iMerge CFG (ip = valid IP address xxx.xxx.xxx.xxx format, omit leading zeroes)
GATEWAY=ip	IP address for gateway (router); normally specified by DHCP (ip = valid IP address xxx.xxx.xxx.xxx format, omit leading zeroes)
SUBNET=m	Subnet mask; normally specified by DHCP (m = xxx.xxx.xxx.xxx format, omit leading zeroes).
8021Q_ENABLE=OFF	Ethernet Layer 2 802.1Q Support (ON or OFF). Default value = OFF
8021Q_VOICE_PRI=n	802.1Q Voice Packet Priority (n = 0-7). Default value = 6
8021Q_SIGNAL_PRI=n	802.1Q Signaling Packet Priority (n = 0-7). Default value = 6
8021Q_VLAN_ID=n	802.1Q VLAN ID (n = 0-4095). Default value = 0
DSCP_VOICE=n	Layer 3 DiffServ Voice Packet DSCP Value (n = 0-63). Default value = 46
DSCP_SIGNAL=n	Layer 3 DiffServ Signaling Packet DSCP Value (n = 0-63). Default value = 46
G711_10=a/b	G.711 codec with 10ms packetization rate (a = average jitter buffer delay in ms, valid range = 10-90 increments of 5; b = maximum jitter buffer depth in ms, valid range = 2*a to 180 increments of 10. Replace a/b with OFF to disable this codec option). Default value = 20/40
G711_20=a/b	G.711 codec with 20ms packetization rate (a = average jitter buffer delay in ms, valid range = 20-90 increments of 5; b = maximum jitter buffer depth in ms, valid range = 2*a to 180 increments of 20. Replace a/b with OFF to disable this codec option). Default value = 35/80
G711_30=a/b	G.711 codec with 30ms packetization rate (a = average jitter buffer delay in ms, valid range = 30-90 increments of 5; b = maximum jitter buffer depth in ms, valid range = 2*a to 180 increments of 30. Replace a/b with OFF to disable this codec option). Default value = 45/90

G729_20=a/b	G.729A codec with 20ms packetization rate (a = average jitter buffer delay in ms, valid range = 20-90 increments of 5; b = maximum jitter buffer depth in ms, valid range = 2*a to 180 increments of 20. Replace a/b with OFF to disable this codec option). Default value = OFF
G723_30=a/b	G.723.1 codec with 30ms packetization rate (a = average jitter buffer delay in ms, valid range = 30-90 increments of 5; b = maximum jitter buffer depth in ms, valid range = 2*a to 180 increments of 30. Replace a/b with OFF to disable this codec option). Default value = OFF
TFTP_UPDATE=n	TFTP update start hour (n=0-23). TFTP automatic update process begins at a pseudo-random interval after this time each day. Default value = 1
TFTP_WINDOW=n	TFTP update window hours (n=1-24). TFTP update time for each phone is calculated from TFTP_WINDOW, TFTP_UPDATE, and the phone's MAC address: Update Time = TFTP_UPDATE + ((MAC Address [23:0]) MOD (TFTP_WINDOW * 60)) 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). Recommended TFTP window duration is one hour per 60 phones on a single TFTP server. Default value = 3
TIME_SERVER=ip	IP address for SNTP time server; may also be specified by DHCP (ip = valid IP address xxx.xxx.xxx.xxx format, omit leading zeroes)
TIME_OFFSET=+n	Offset (in hours) from UTC or time server time; may also be specified by DHCP (n = +12 to –12) 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 Default value = 0
LDAP=filename	LDAP specification file, contains information for LDAP application
EOF	End of keyword list (Required)

Example File:

```
;"tcs7200a.txt" Configuration file for Tone Commander 7210 and 7220 IP Phones
;Do not edit without instructions! Refer to 7210/7220 Network Admin Guide
IMERGE=0.0.0.0           ;iMerge IP address, no leading zeros after dots
;GATEWAY=0.0.0.0         ;Gateway IP address, normally specified by DHCP
;SUBNET=255.255.255.0    ;Subnet mask, normally specified by DHCP
;G711_10=20/40           ;G.711, 10ms rate (ave delay/max delay or OFF)
;G711_20=35/80           ;G.711, 20ms rate (ave delay/max delay or OFF)
;G711_30=45/90           ;G.711, 30ms rate (ave delay/max delay or OFF)
;G729_20=OFF             ;G.729A, 20ms rate (ave delay/max delay or OFF)
;G723_30=OFF             ;G.723.1, 30ms rate (ave delay/max delay or OFF)
;8021Q_ENABLE=OFF        ;Ethernet Layer 2 802.1Q Support (ON or OFF)
;8021Q_VOICE_PRI=6       ;802.1Q Voice Packet Priority (0-7)
;8021Q_SIGNAL_PRI=6      ;802.1Q Signaling Packet Priority (0-7)
;8021Q_VLAN_ID=0         ;802.1Q VLAN ID (0-4095)
;DSCP_VOICE=46           ;Layer 3 DiffServ Voice Packet DSCP Value (0-63)
```

```
;DSCP_SIGNAL=46          ;Layer 3 DiffServ Signaling Packet DSCP Value (0-63)
;TFTP_UPDATE=1           ;TFTP update start hour (0-23)
;TFTP_WINDOW=3           ;TFTP update window hours (1-24)
;TIME_SERVER=0.0.0.0     ;SNTP time server IP address
;TIME_OFFSET=-8          ;Offset (in hours) from UTC (-12 to +12)
;LDAP=commnet.txt        ;LDAP Specification File
EOF                      ;Required End-of-File mark
```

tcs7220b.txt

Names the compressed application and boot program files, located in the TFTP server root folder, to be downloaded to the 7210/7220 phone.

Valid keywords:

APPLICATION	The name of the file that contains the application program that runs on the phone.
BOOTROM	The name of the file that contains the code for the boot program that runs on the phone.
EOF	Indicates the end of the keyword list.

NOTE – For phone software versions prior to 01.02.03, the boot program file is named in the file ***tcs7220c.txt***.

Example File:

```
; tcs7220b.txt file contains file names of compressed application and uncompressed
bootrom for TCS 7210/7220 phone.
; The application file must be on the TFTP server for the phone update to work.
APPLICATION=TCS_01_01_07.z
BOOTROM=TCS_BOOT_01_01_04.bin
EOF
```

xxxxxxxxxxxxxx.txt (*alias*.txt)

Custom configuration files for individual telephones.

All keywords used in the tcs7220a.txt and tcs7220b.txt files are valid in this custom file.

Example File:

```
;alias.txt  config file for ABC Widget Company,
;overrides for Billy Bob - alias = 30947701840101
IMERGE=130.131.190.81          ; gatekeeper IP address changed from tcs7220a.txt file
APPLICATION=TCS_ZZ_YY_XX.z     ; use alternate application program
EOF
```

NOTE – Keyword parameters not specified in a custom configuration file are set to the values specified in ***tcs7220a.txt***.

Port Usage

AGCS iMerge Port Numbers

iMerge Listening (receiving) UDP ports:	1719, 6000 – 6192 (A-chassis RTP), and 32768 – 65535 (C-chassis RTP)
iMerge Listening (receiving) TCP ports:	1720

Tone Commander CPE

CPE Listening (receiving) UDP Ports:	1719, 6000
CPE Listening (receiving) TCP Port:	1720, 57571

TONE
COMMANDER®

11609 49th Place West
Mukilteo, WA 98275-4255
(800) 524-0024   (425) 349-1000
Fax: (425) 349-1010
www.tonecommander.com