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Basic Call Termination With Quintum Analog Tenors

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Introduction

Call termination is a very common type of application where a customer installs a VoIP Gateway at their location and they have this connected to the local PSTN lines. The customer then engages an international provider to send VoIP calls to his unit for termination within his country to take advantage of local rates. The customer charges a rate back to the call providers for this service.

This document will provide users information on how to perform a basic configuration on Quintum Tenor's to terminate VoIP calls to PSTN lines. We will provide specific information related to the termination of calls only. Other options for this application will be discussed.

For more information and questions, contact the QTAC at 1-877-435-7553 (Toll Free in the U.S.), 1-732-460-9399 (Internationally), or email at service@quintum.com.

Application Information

Description

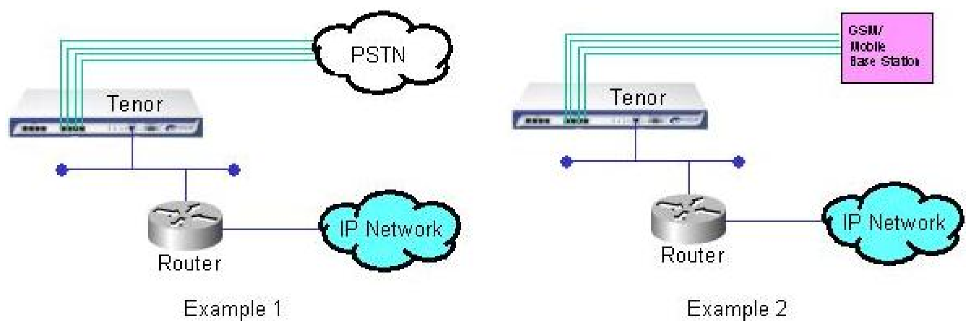


Figure 1

In a typical use, you would have a Quintum Tenor Analog Gateway installed in your location. This unit would then have a connection to the Internet (via a router) and then some number of connects to your local PSTN provider. In some cases, customers use mobile phone base stations instead of the actual analog lines as shown in example 2 above.

They will receive VoIP calls from providers that they have contracted with to provide traffic for a particular country or area. Typically, these providers will send the calls with a specific number format and the Quintum needs to be configured to accept the call and terminate it to the first available analog line and dial out the correct digits.

Considerations

When setting up for this type of application, the following issues should be taken into consideration and in some cases are necessary to know before configuring.

Type of connection to PSTN and Number of Lines

You must decide what is the best connection to have to the Tenor from the PSTN provider. Many things must be taken into consideration for this.

- **Price**. This is always the biggest consideration. Not so much as the initial installation, but the monthly charges and rates that you get will, many times, determine the type of connection. If you choose a T1 (or E1) connection, the monthly fee will be more expensive that that of several analog lines. Depending on the amount of traffic that you will receive, you may not be able to make enough money to pay for a T1 or E1 line, or you may need to charge a higher rate to your provider and at some point the provider may decide to switch to a less expensive competitor.

- **Number of calls to support**. You need to determine how many calls you want to support and see what type of PSTN connection and how many will support your requirements.
- **Availability**. Not all countries have all types of connections. Some may only have analog connections. You should check with your local PSTN provider about this.

Supervision – Answer & Disconnect

Both are important for billing. If you are using digital lines, such as ISDN signaled T1's/E1's, then answer and disconnect supervision are provided to you by the PSTN provider. However, if you are using analog lines, then you need to determine how you will handle both answer and disconnect supervision. It is recommended that you review our documentation on both topics located at:

http://www.quintum.com/support/xplatform/kb/telco/Answer_Supervision.pdf

http:// www.quintum.com /support/xplatform/kb/telco/Disconnect_Supervision.pdf

These two documents will describe how to set the analog Tenors for disconnect and answer supervision.

IP Bandwidth and Quality

You need to determine how much bandwidth you will need. You can do this through some simple math. You first determine how many VoIP calls you want active at the same time on your Tenor and multiply this by the bandwidth required based on your audio compression. If you plan to use G.729, then figure on about 19kb per call in each direction. If you plan to use G.723.1 @ 6.3kb, the bandwidth will be about 13.5kb per call in each direction. Of course you could reduce this further if you use Quintum's PacketSaver technology and all calls are between Quintum units, as PacketSaver is a proprietary technology to reduce bandwidth usage between 2 Quintum units.

For the quality, this will depend on the ISP that you use for your Internet connection. If it is a lower tier ISP, the quality may not be there in that you may experience high packet loss or long delays on your Internet connection. You should discuss this with your ISP.

Provider Information

At a minimum, you will need to know what the number/digit pattern is that your providers will send to you. Typical patterns are international prefix (like 00 or 011) + CountryCode + number, or Countrycode + number. In many cases, providers may add a special prefix to the front of the number. This is usually a 4 or 5 digit number (could be more or less) that gets added to the front of all numbers and is sent as part of the phone number. For example, if the prefix is 7894, then they may send you the number as 7894+00441234567890. It is very important to know what the digit pattern is that your provider will send to you as the Tenor needs to have these patterns configured in it to allow the call to terminate.

You may also want or need to know the provider's IP address for security. The Tenor allows you several ways restrict access to it. The easiest way is a simple access list of allowed IP addresses. More complex would be with a radius server for authentication. For more detailed information, you can review our document "End Point Authentication / IP Address Security" that is available on our web site at:

http://www.quintum.com/support/xplatform/ivr_acct/End_Point_Authentication.pdf

Load Balancing and Overflow

You may have several units installed at the same location to terminate calls. In these situations, most customers are looking for calls to be distributed evenly over all the units or in the least, if the first unit is filled, for the call to overflow to the next. This feature can only be used from the call origination point, not from the termination. Once the termination Tenor receives a call from IP, it cannot re-route it back to IP to another termination unit. You should discuss this with your provider/originator to have them perform this function at their side. If this is not possible, you may want to consider purchasing a Quintum Call Relay to install between your termination units and your originator. You would then have your originator send all calls to the Call Relay, and the Call Relay will be able to perform this load balancing and overflow to the termination units.

Sample Configurations

The following sample configurations are for Analog units only as specified in each section below. The configuration will only provide information specific to this application and the sample information provided.

Additionally, your specific application for termination may be different than our example, but this document should provide you a guide for configuring your unit as needed.

Application Information

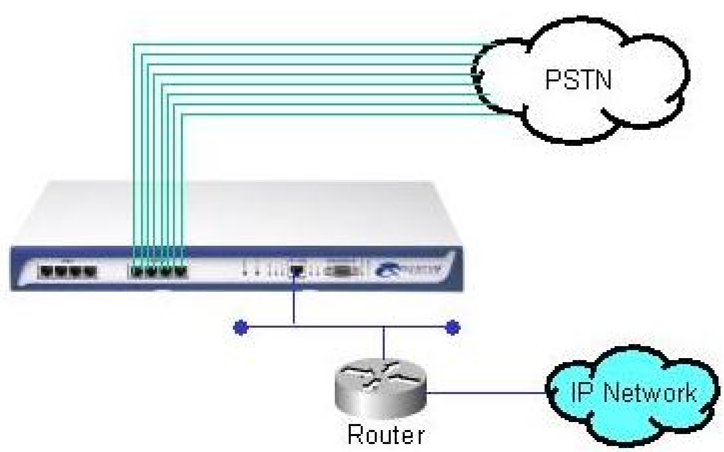


Figure 2

The above shows a Quintum Analog Tenor, in this case an A800, but for the sake of each section, this could be an A400, A800 or any of the AS or AX series Tenor that have FXO ports. The connection to the PSTN may be different on the model that you have and you should refer to the user guide for connection methods and requirements.

This unit is located/installed in Eatontown, NJ USA. The reason we have picked this is because we know all the correct dialing plans and will explain these for your information and to show how you might need this information for your country. All the information relates to how a person in Eatontown would make calls.

International dialing prefix = 011
National/Long Distance prefix = 1
Country code = 1
Local Area Code/City Code = 732

- To dial an international number, dial 011 + country code + number.
- To dial a Long Distance (out of the local calling area) number, dial 1 + area/city code + number.
- To dial a local number, dial 732 + number.
- To dial a mobile phone number in 732 area/city code, dial 732 + number.
- To dial a mobile phone number outside 732 area/city, dial 1 + city/area code + number.

Again, your dialing plan in your country may be different, but it is important to understand how you dial the different numbers from your phone.

This unit is on a public IP of 208.226.141.10 and this should be configured previously in the unit along with the default gateway and subnet mask.

No external Gatekeeper is used.

There is one provider that sends calls to this unit from an IP address of 64.251.32.18. All calls are presented with a prefix of 9955 + country code + number. The codec/compression to be used is G.729AB.

This Tenor is used to terminate all calls: local, national and international.

The analog lines in the US support forward disconnect (battery removal) for disconnect supervision. Keep in mind that typically only US and Canada support this type of disconnect supervision. Most other countries only support disconnect tone.

The Tenor is configured to use software-based answer supervision.

Configuration on Generation 1 Analog (A400/A800)

The following configuration uses the CLI (Command Line Interface) via Telnet. It is supposed that the customer has some knowledge of using the CLI. Prompts are provided with the commands, but we do not provide step-by-step directions to reach each prompt as that should be evident once in the CLI. This configuration also supposes that the unit has not been previously configured and is at factory default.

Prompt	Command	Syntax	Comments
config unit 1#	online	online 1	Sets the Tenor online, check alarms after submit.
config sys#	country	country 1	Sets the country to Canada. Can be set to anything other than US.[1]
	countrycode	countrycode	Removes the country code. See above note.
	areacode	areacode	Removes the area code. See above note 1.
	mindn	mindn 10	Sets the minimum number of digits to 10.
	maxdn	maxdn 20	Sets the maximum number of digits to 20.
	timeserver	timeserver p 208.184.49.9	Sets the primary timeserver to the configured IP address.
	timeserver	timeserver s 129.6.15.28	Sets the secondary timeserver to the configured IP address.
	utcoffset	utc – 4	Sets the UTC/GMT offset to – 4 hours.
config pstntg 1#	passthru	pass 0	Disables passthrough feature.

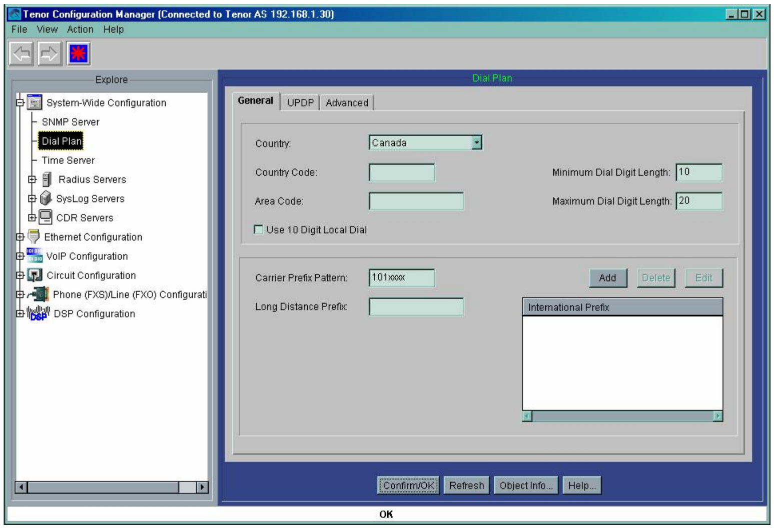
[1] Because the calls from IP are not in the standard E.164 format as defined by H.323, we will need to set the dial plan to a non-standard dial plan.

Prompt	Command	Syntax	Comments
	cassignaling	cassig 6	Sets the CAS signaling to Loop Start with Fwd Disconnect.
	supervision	super 2	Enables the answer supervision.
	answerdelay	answerdel 120	Sets the answer delay for answer supervision to 120 seconds.
	lampattern	lamp 199551732 lamp 2 99551 lamp 3 99552 lamp 4 99553 lamp 5 99554 lamp 6 99555 lamp 7 99556 lamp 8 99557 lamp 9 99558 lamp 10 99559	Configures the number patterns to terminate to the PSTN lines. These must match what is received from IP. Also, lampattern patterns, when the match a number, the matching digits will be deleted off of the number. See the CLI guide/help file for more information.
	lamreplacement	lamr 1 732 lamr 2 1 lamr 3 0112 lamr 4 0113 lamr 5 0114 lamr 6 0115 lamr 7 0116 lamr 8 0117 lamr 9 0118 lamr 10 0119	Replacements to add digits back on that were deleted from the lampattern. 1 is for local numbers so they are dialed out as 732+number, 2 is for all national/long distance calls where we need to dial 1+area/city code + number, and 3 through 10 are for all international calls.
config gksys#	borderelement	border 0 208.226.141.10	Sets this Tenor as its own Border Element.
config dsp#	voicecoding	voice 68	Sets the compression to G.729.
config#	submit	submit	submit changes.

Generation 2 Tenors (AS/AX) using Configuration Manager

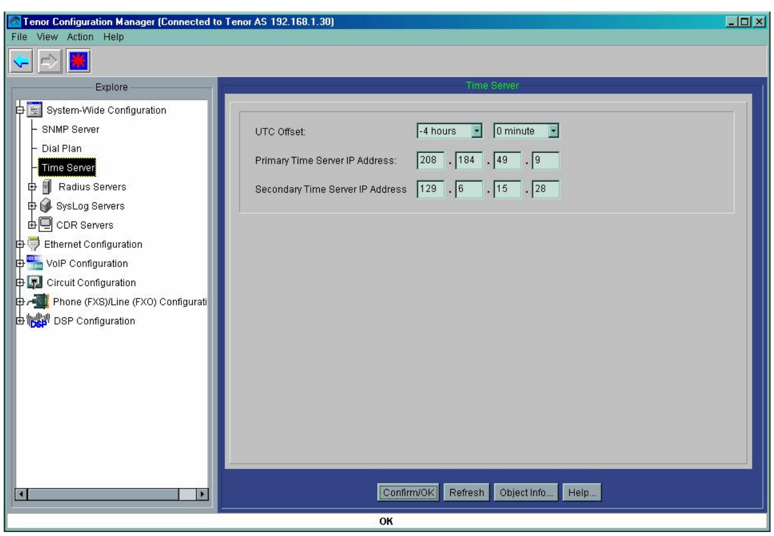
Please make sure that you have upgraded both your Tenor and the Configuration Manager to the latest version. The following will list the page or tree branch for the options and their settings. Please do not forget to press the Confirm/OK button.

System-Wide Configuration → Dial Plan



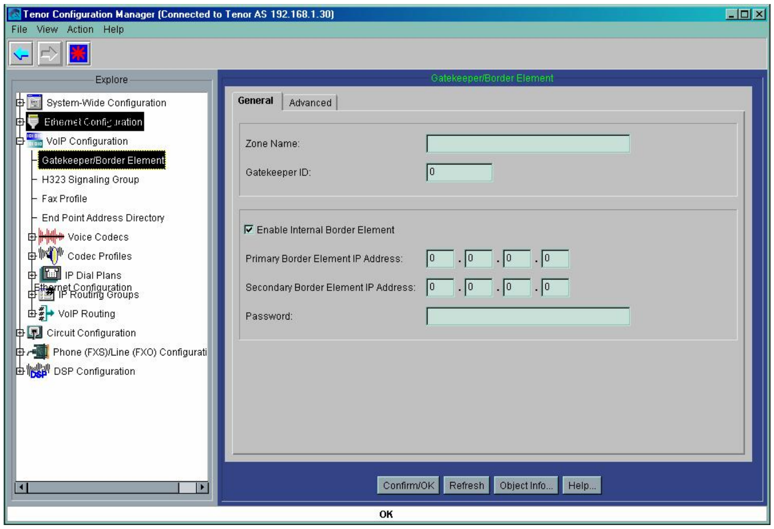
Country = Canada	Change to any non-US/non-standard dial plan.
Country Code = null	Remove country code.
Area Code = null	Remove area code
Minimum Dial Digit Length = 10	Sets the minimum digit length to 10 digits.
Maximum Dial Digit Length = 20	Sets the maximum digit length to 20 digits.
Long Distance Prefix = null	Remove long distance prefix.
International Prefix = null	Remove International prefix.

System-Wide Configuration → Time Server



UTC Offset = -4 hours 0 minutes	Sets the UTC/GMT offset for this location.
Primary Time Server IP Address = 208.184.49.9	Sets the primary time server.
Secondary Time Server IP Address = 129.6.15.28	Sets the secondary time server.

VoIP Configuration → Gatekeeper/Border Element

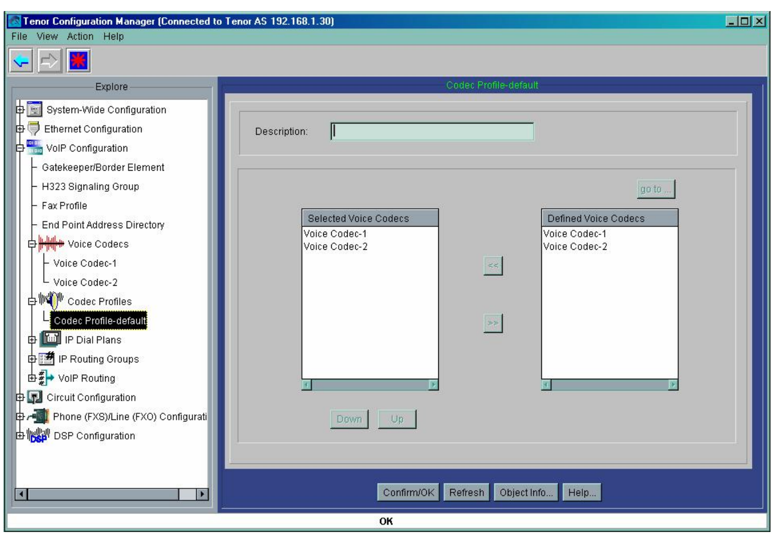


Enable Internal Border Element = checked	Enables the internal border element.

VoIP Configuration → Voice Codecs

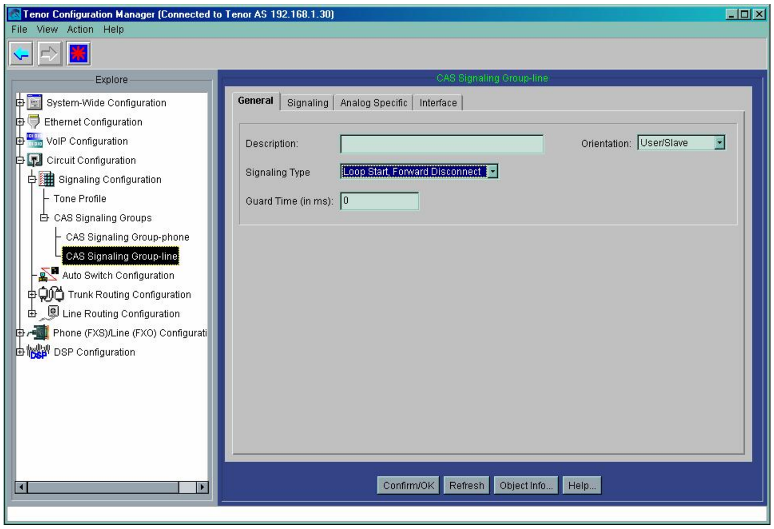
Make sure you have two Voice codecs, one set for G.723 and the other set for G.729.

VoIP Configuration → Codec Profiles → Codec Profile-default



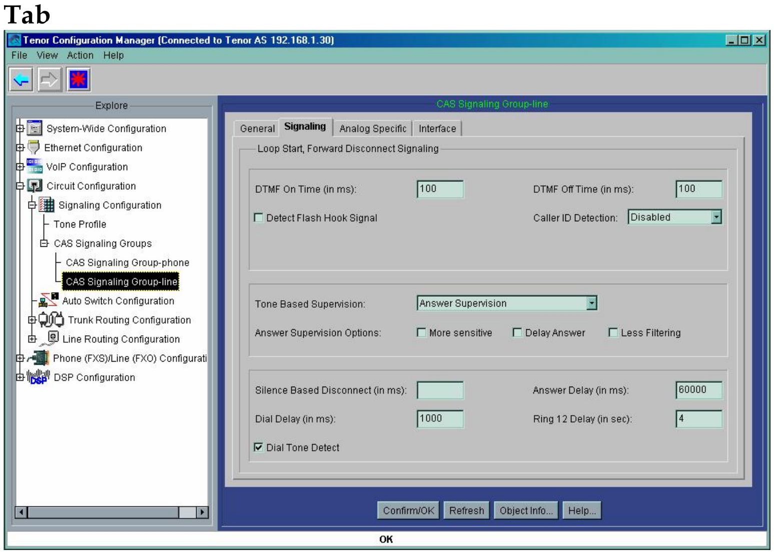
Make sure that Voice Codec-1 and Voice Codec-2 are shown in the Selected Voice Codecs list.

Circuit Configuration > CAS Signaling Groups > CAS Signaling Group-line > General Tab



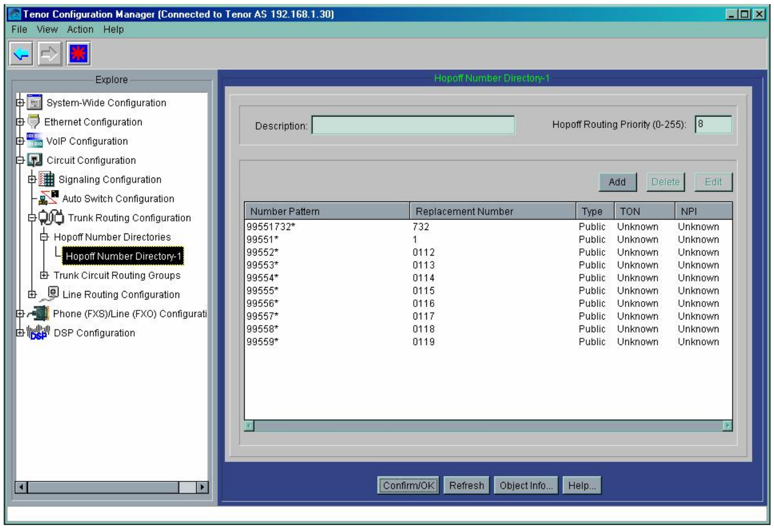
Signaling Type = Loop Start, Forward Disconnect	Enables the internal border element.

Circuit Configuration > CAS Signaling Groups > CAS Signaling Group-line > Signaling Tab



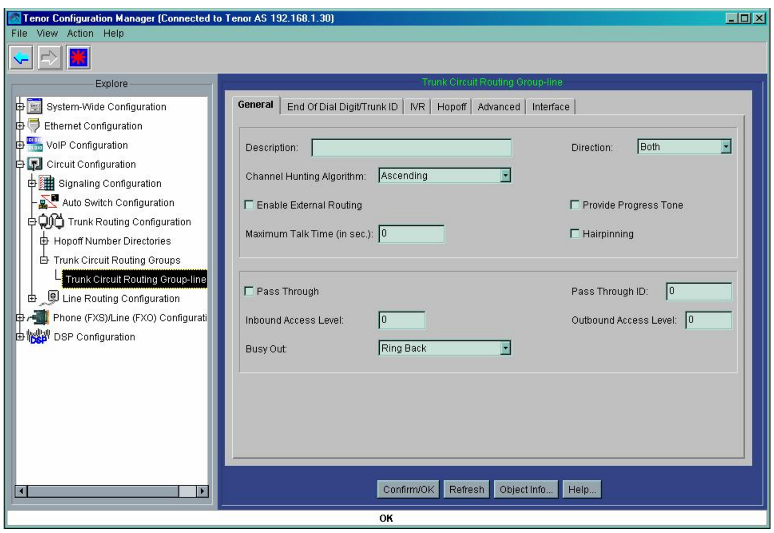
Tone Based Supervision = Answer Supervision	Enables software based answer supervision
Answer Delay = 60000	Sets the answer delay (for answer supervision) to 60000 ms (60 seconds).

Circuit Configuration > Trunk Routing Configuration > Hopoff Number Directories > Hopoff Number Directory-1



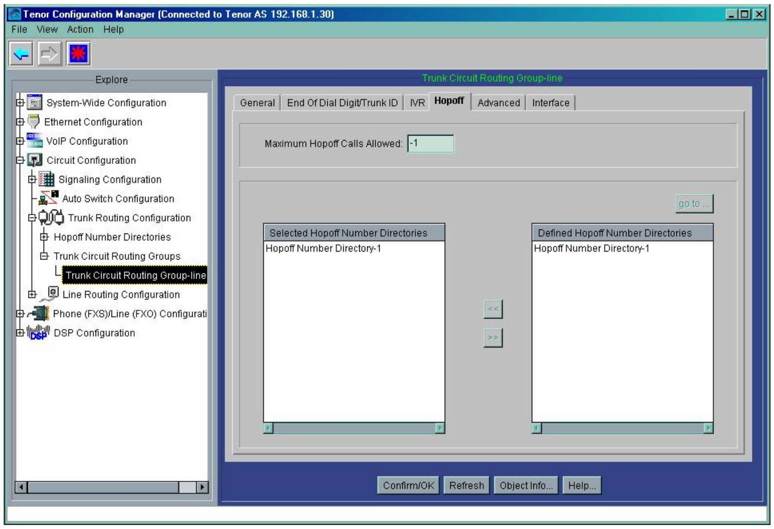
Add the shown patterns and replacements per the requirements for the digits received from IP and those needed to dial out to the PSTN.

Circuit Configuration > Trunk Routing Configuration > Trunk Circuit Routing Groups > Trunk Circuit Routing Group-line > General Tab



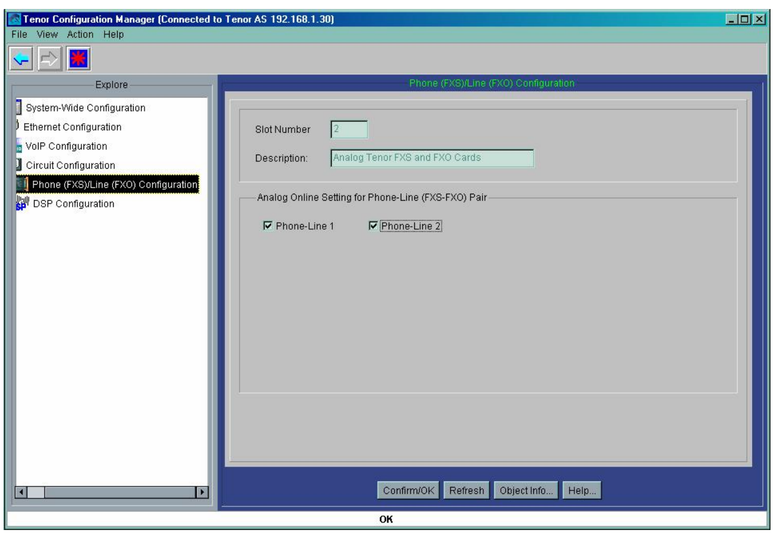
Channel Hunting Algorithm = Ascending	You can choose how you want the line hunting to be set.
Pass Through = uncheck	This disables passthrough.

Circuit Configuration > Trunk Routing Configuration > Trunk Circuit Routing Groups > Trunk Circuit Routing Group-line > Hopoff Tab



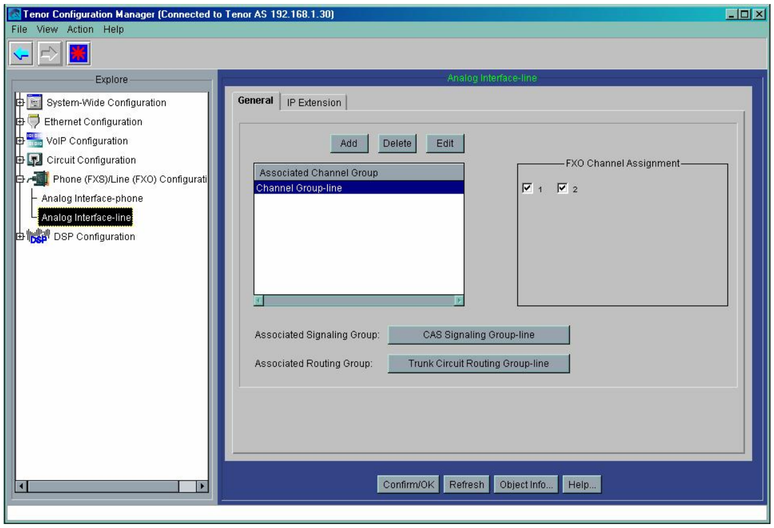
Make sure that the Hopoff Number Directory-1 is listed in the Selected Hopoff Number Directories.

Phone (FXS)/Line (FXO) Configuration



Make sure that all Phone-Line check boxes are checked allowing for all the lines to be put online.

Phone (FXS)/Line (FXO) Configuration → Analog Interface-line



Make sure that the channels are enabled in this group, that the CAS Signaling Group-Line is assigned to the Signaling Group, and the Trunk Circuit Routing Group-line is assigned to the Routing Group, as shown above.

Submit the changes at this point.

Check your active alarms (View > Tenor Alarm Status > Active Alarms and you should see no active alarms.

Generation 2 Tenors (AS/AX) using CLI (Command Line Interface)

The following shows how to configure the AS/AX for this application using the CLI interface from Telnet. Directly following this is a complete "set" list that you can edit with the correct hop-off patterns and replacements and any other information and then copy directly to the AS/AX, as long as it is running P100-19-xx. If you are running a higher version, these commands may work, but always check.

Straight CLI Information

Prompt	Command	Syntax	Comments
onfig-Site-1#	country	set c 1	Sets the country to Canada.
Config-TimeServer-1#	UTCOffset	set utco –4	sets the utc/gmt offset to –4 hours.
	PrimaryServerIPAddr	set psipa 208.184.49.9	Sets the primary time server.
	SecondaryServerIPAddr	set ssipa 129.6.15.28	Sets the secondary time server.
config-DialPlan-1#	MINDNlength	set mindn 10	Sets the minimum DN length to 10 digits.
	MAXDNlength	set maxdn 20	sets the maximum DN length to 20 digits.
	LongDistancePrefix	set ldp	Deletes the long distance prefix.
	INTernationaLPrefix	set intlp[1]	Deletes the international prefix.

Prompt	Command	Syntax	Comments
config-PUBlicNumberingPlan-1#	CountryCode	set cc	Deletes the country code.
	AreaCode	set ac	Deletes the area code
	slot 2	sl 2	Gets you to the slot 2 prompt.
config-SLot-SL2#	Online	set o[1] 1 set o[2] 1	Sets the lines, online. Repeat for all lines in the unit and listed.
config-HopoffNumberDirectory-1#	add	add 99551732 r 732 add 99551 r 1 add 99552 r 0112 add 99553 r 0113 add 99554 r 0114 add 99555 r 0115 add 99556 r 0116 add 99557 r 0117 add 99558 r 0118 add 99559 r 0119	Adds all the hop-offs and replacements that will terminate to this unit.
config-VoiceCodec-1#	CodecVoiceCoding	set cvc 81	Sets the first Voice codec to G.723.
config-VoiceCodec-2#	CodecVoiceCoding	set cvc 68	Sets the first Voice codec to G.729.
config-CASSignalingGroup-line#	SignalingType	set st 3	Sets the signaling to loop start with forward disconnect
	ToneBasedSupervision	set tbs 2	Enables software-based answer supervision.
	AnswerDelay	set ad 60000	Sets the answerdelay for answer supervision to 60000ms (60 seconds).
config-GateKeeperParam-1#	EnableInternalBE	set eibe 1	Enables the internal Border Element.
config-CodecProfile-default#	VoiceCodecAttached	set vca[1] vc-1 set vca[2] vc-2	This will attach both voice codecs (G.723 and G.729) to the codec profile.
config-TrunkCircuitRoutingGroup-line*	HUNTAlgorithm	set hunta 0	This sets the line hunting to Ascending. You can choose any hunting you prefer.
	PassThroughEnable	set pte 0	Disables passthrough.
	HopoffNumberDirAttach	set hnda[1] hnd-1	Attaches the hopoff directory to this trunk group.
config-ChannelGroup-line*	SignalingGroupAttached	set sga cassig-line	Attaches the cassignaling group to this channel group.
	RoutingGroupAttached	set rga tcrg-line	Attaches the TCRG to this channel group.
	submit	submit	Submits changes.

CLI Commands – Copy Directly to Tenor AS/AX

The following set commands are identical to the above, but you can make the necessary changes to the hop-off numbers, etc., copy the lines (CTRL-C) and paste them directly into a Telnet session with the Tenor at the Quintum prompt. By doing this you will "play" these commands and the Tenor will be configured just as if you typed them in separately. These commands are based on software version P100-19-xx and should only be copied to a system with this version or higher and only when at factory default.

```
config
table off
site 1
set c 1
ts 1
set utco –4
set psipa 208.184.49.9
set ssipa 129.6.15.28
dp 1
set mindn 10
set maxdn 20
set ldp
set intlp[1]
pubnp 1
set cc
set ac
sl 2
set o[1] 1
set o[2] 1
hnd 1
add 99551732 r 732
add 99551 r 1
add 99552 r 0112
add 99553 r 0113
add 99554 r 0114
add 99555 r 0115
add 99556 r 0116
add 99557 r 0117
add 99558 r 0118
add 99559 r 0119
vc-1
set cvc 81
vc-2
set cvc 68
cassig line
set st 3
set tbs 1
set ad 60000
gkp
set eibe 1
cp default
set vca[1] vc-1
set vca[2] vc-2
tcrg line
set hunta 0
set pte 0
set hnda[1] hnd-1
cg line
set rga tcrg-line
set sga cassig-line
table on
sub
```

Basic Troubleshooting for this Application

Because this sample configuration is simple in nature, as is most termination applications, there are few problems that can happen. Below is a list of the most common problems along with ways to diagnose this.

Event Logging

Capturing and review the event log built in to the Tenor is the best way to diagnose most all problems, however, there are many messages that come out from this and it may not be easy to understand. Below are some basics on this. This section is not intended to provide a description of all messages, but simply to show what a basic call looks like. In the following sections, you will see, from the event log, how to find other problems.

Enabling Event Log

To enable the event log from 1G Tenors (A400/A800), type the following:

ev 0	Clears all event log modules
ev +ch	Enables the Call Handler log module
ev c	Clears the event log buffer from previous events
q	Quits out of command mode and goes into log mode so all events are shown on the screen

To exit the event log from the 1G Tenor, press any key.

To enable the event log on a 2G Tenor (AS/AX), type the following:

ev l3 ch	Enables the Call Handler log at level 3
ev c	Clears the event log buffer
ev qu	Quits out of command mode and goes into log mode so all events are shown on the screen

To exit the event log from the 2G Tenor, press 'q'.

Standard Log Sample

CH :124225620:h323[0]: h323mgr:RcvIncomingCall Tenor receiving call from IP (H323)
CH :124225643:h323[0]: ocall:RcvSetup, my media type=16
CH :124225643:bandwidth info: max=-1 cur=12600.
CH :124225643:Fast start element present.
CH :124225643:Incompatible media type=9, calling->called.
CH :124225643:Incompatible media type=9, called->calling.
CH :124225643:Setting remote rtp port=192.168.1.30:10256. Call is coming from IP address of 192.168.1.30
CH :124225643:Remote side packet saver version = 3.
CH :124225643:CallInfo[0xd81030]: origCalled.digit(995517324609000) callingParty.digits() No Calling Number (ANI) Provided.
.
Incoming digits (Called Number).
CH :124225644:h323[6/59]: ocall:stackSendCallProc
CH :124225644:Sending rtp port=192.168.1.68:10240.
CH :124225647:Routing requested for: public(1) orig=995517324609000 public(1) normalized=995517324609000 route code= tg=0.
If there was a standardized dial plan, the normalized number would reflect this.
CH :124225647:1 match(es) found: 4 At least 1 match for this number has been found
CH :124225647:CasTG[7]: newTermCall: selected line=256 chan=256.
CH :124225647:Route response(6): result=1 cause=0.
CH :124225647:udp connect: 9 11
CH :124225647: c0a80144 10240 c0a8011e 10256
CH :124225647:TBCSM[6]: Setup from peer=0xd8100c NP=0x0 NT=0x0.
CH :124225647:OrigNum=995517324609000 NormNum=995517324609000 TranNum=7324609000 OrigDest=.
CH :124225647:[2: 1] sent message to cas: Setup Tenor has gone off-hook on PSTN interface, line 1 [2:1].
This is the number to dial out to the PSTN based on the hopoff/lam pattern and replacement.
CH :124225653:tsi connect: 001 202 01
CH :124225653:TsiConnXlate: 0:1, 2:2
CH :124225744:tsi disconnect: 001 202 01
CH :124225744:TsiDiscXlate: 0:1, 2:2
CH :124225744:[2: 1] received message from cas: Call-Proc Tenor has received dial tone from this PSTN line.
CH :124225751:tsi connect: 001 210 10
CH :124225751:TsiConnXlate: 2:10, 0:1
CH :124225971:tsi disconnect: 001 210 10
CH :124225971:TsiDiscXlate: 2:10, 0:1
CH :124225971:[2: 1] received message from cas: Alert Tenor has dialed all digits out to PSTN.
CH :124225971:h323[6/59]: ocall:stackSendProg
CH :124225971:Sending rtp port=192.168.1.68:10240.
CH :124225974:tsi connect: 001 209 01
CH :124225974:TsiConnXlate: 0:1, 2:9
CH :124225974:tsi connect: 001 209 10
CH :124225974:TsiConnXlate: 2:9, 0:1
CH :124225974:h323[6/59]: ocall:stackSendAlert
CH :124225974:Sending rtp port=192.168.1.68:10240.
CH :124226445:h323[6/59]: CasTermCall:ds0ToneCB(4)
CH :124226445:OBCSM[6]: Received peerRcvConnect. Call has been answered by destination.
CH :124226445:CallInfo[0xd81030]: sendConnected Event. leg(1)
CH :124226445:h323[6/59]: ocall:stackSendConnect
CH :124226863:ocall[6/59]:RcvRelComp, cause=16. Tenor has received a release from the originator (ocall) to hang up the call. The cause is a 16, Normal call clearing.
CH :124226863:CallInfo[0xd81030]: sendDisconnected Event.
CH :124226863:CallInfo[0xd81030]: cdr callid(6) rmtcallid(3b) calltype(1) destnumtype(1).
CH :124226863:CallInfo[0xd81030]: cdr dest(995517324609000) orig(0) remote() pin().
CH :124226863:CallInfo[0xd81030]: cdr icline(ffffffff) icport(c0a8011e) otline(1) otport(0).
CH :124226863:CdrManager::addCdr: 5 Return.
CH :124226863:tsi disconnect: 209 001 01
CH :124226863:TsiDiscXlate: 2:9, 0:1
CH :124226863:tsi disconnect: 209 001 10
CH :124226863:TsiDiscXlate: 0:1, 2:9
CH :124226863:udp disconnect: 9 11
CH :124226863: c0a80144 10240 c0a8011e 10256
CH :124226863:TBCSM[6]: Release from peer=0xd8100c cause=0x10.
CH :124226863:OBCSM[6]: Release complete from peer=0xd78c04.
CH :124226863:[2: 1] sent message to cas: Disc Tenor sends disconnect/hang-up to PSTN.

In 1G Tenors, the numbers on the right are internal time clicks representing .01 second. If you subtract the lower number from the higher, then divide by 100, you will get the number of seconds between the 2 messages. For example, you can see how long after dialing the digits till the call was connected by taking the connect time of 124226445 subtract the alert time of 124225971 (equals 474) then divide this by 100 and you will have 4.74 seconds elapse. This could be useful when determining the timing of events.

In gen 2 Tenors, the actual time is shown for each message.

Calls Not Completing

Obviously the most common issue is that calls are not connecting. There are many reasons for this. The most common is that the number received from IP does not match to the hop-off or lampatterns you have configured. You can tell this from the CDR records of the Tenor if you see many calls with a disconnect cause code of 34 (No channel available).

So the first step to trouble shoot this problem is to review your CDR records. You can get information on how to look at the CDR records on the Tenor from our web site, just search for CDR and you should find the correct documents for this.

When you look at the CDR, the first field that will give you information is the 7th field which represents the disconnect cause code. There should only be a value in this field if the call fails to connect.

Some examples are as follows.

8,,0,20040707204954,,20040707204954,34
9,995517324609000,0,20040707205120,,20040707205126,41
2,995517324609000,0,20040707183408,,20040707183434,16

If there is no disconnect code, then the call connected fine and disconnected normally (sometime after connect).

The following lists the common disconnect codes, the reason, and possible solutions.

Disconnect Cause Code 34

This code is defined as No Terminating Channel Available. In the Tenor, this generally means that the number received from IP does not have a match to terminate to. In other words, an invalid pattern of digits was received from IP. There are 2 ways to check this. The first is again through the CDR, the 2nd field is the Called Number field. This will be the digits that were received from IP. However, if there is no match for the digits received, this field will be blank as shown in the example below.

8,,0,20040707204954,,20040707204954,34

So you now need to review the event log to see what digits you have received. Follow the directions from above to setup the log and have the call made again as the log will only be able to show you the calls as they are made. Below is a sample of a failed call for cause 34.

```
CH    :125021689:h323[0]: h323mgr:RcvIncomingCall      Tenor receiving call from IP (H323)
CH    :125021712:h323[0]: ocall:RcvSetup, my media type=16
CH    :125021712:bandwidth info: max=-1 cur=12600.
CH    :125021713:Fast start element present.
CH    :125021713:Incompatible media type=9, calling->called.
CH    :125021713:Incompatible media type=9, called->calling.
CH    :125021713:Setting remote rtp port=192.168.1.30:10264.      Call is coming from IP address of 192.168.1.30
CH    :125021713:Remote side packet saver version = 3.
CH    :125021713:CallInfo[0xd81030]: origCalled.digit(552317324609000) callingParty.digits()
.                                                                 Incoming digits (Called Number).
CH    :125021713:h323[8/63]: ocall:stackSendCallProc
CH    :125021713:Sending rtp port=192.168.1.68:10240.
```

CH :125021716:Routing requested for: public(1) orig=552317324609000 public(1) normalized=552317324609000 route code= tg=0.
CH :125021716:0 match(es) found:
CH :125021716:Route response(8): result=0 cause=34.
CH :125021716:CallInfo[0xd81030]: cdr callid(8) rmtcallid(3f) calltype(1) destnumtype(1).
CH :125021716:CallInfo[0xd81030]: cdr dest(552317324609000) orig(0) remote() pin().
CH :125021716:CallInfo[0xd81030]: cdr icline(ffffffff) icport(c0a8011e) otline(ff) otport(80cc8891).
CH :125021716:CdrManager::addCdr: 7 Return.
CH :125021716:h323[8/63]: ocall:stackSendRelease

No matches to the dialed number were found.

If there was a standardized dial plan, the normalized number would reflect this.

Tenor returns cause code 34.

So, from the above, in the highlighted area, you can see that the number sent to us was 552317324609000, however, from our sample configurations, in the lampattern/Hopoff patterns, all of our numbers begin with 9955. The number patterns do not match and the Tenor will not accept this. You must either contact this provider and have them send the correct prefix, or you must enter additional lampatterns/hopoff patterns with the new prefix.

Disconnect Cause Code 41

A disconnect with a code of 41, temporary failure, is typical when the Tenor receives a call from IP and then goes off-hook on a PSTN line, but does not detect dial tone. The CDR record will look like;

9,995517324609000,0,20040707205120,,20040707205126,41

and a event log will show:

CH :125030324:h323[0]: h323mgr:RcvIncomingCall
CH :125030347:h323[0]: ocall:RcvSetup, my media type=16
CH :125030347:bandwidth info: max=-1 cur=12600.
CH :125030347:Fast start element present.
CH :125030347:Incompatible media type=9, calling->called.
CH :125030347:Incompatible media type=9, called->calling.
CH :125030347:Setting remote rtp port=192.168.1.30:10266.
CH :125030347:Remote side packet saver version = 3.
CH :125030347:CallInfo[0xd81030]: origCalled.digit(995517324609000) callingParty.digits()
.
CH :125030347:h323[9/64]: ocall:stackSendCallProc
CH :125030348:Sending rtp port=192.168.1.68:10240.
CH :125030351:Routing requested for: public(1) orig=995517324609000 public(1) normalized=995517324609000 route code= tg=0.
CH :125030351:1 match(es) found: 4
CH :125030351:CasTG[7]: newTermCall: selected line=256 chan=256.
CH :125030351:Route response(9): result=1 cause=0.
CH :125030351:udp connect: 1 11
CH :125030351: c0a80144 10240 c0a8011e 10266
CH :125030351:TBCSM[9]: Setup from peer=0xd8100c NP=0x0 NT=0x0.
CH :125030351:OrigNum=995517324609000 NormNum=995517324609000 TranNum=7324609000 OrigDest=.
CH :125030351:[2: 1] sent message to cas: Setup
CH :125030357:tsi connect: 001 209 01
CH :125030357:TsiConnXlate: 0:1, 2:9
CH :125030857:tsi disconnect: 001 209 01
CH :125030857:TsiDiscXlate: 0:1, 2:9
CH :125030937:[2: 1] received message from cas: RelComp
CH :125030937:OBCSM[9]: Release from peer=0xd78c04 cause=0x29 redir=.

Tenor receiving call from IP (H323)

Call is coming from IP address of 192.168.1.30

Incoming digits (Called Number).

At least 1 match for this number has been found

If there was a standardized dial plan, the normalized number would reflect this.

Tenor has gone off-hook on PSTN interface, line 1 [2:1].

This is the number to dial out to the PSTN based on the hopoff/lam pattern and replacement.

Tenor has issued a release from the PSTN side.

This is a hex value cause code (any number that has 0x in front is hex), if you convert this to decimal, it will be 41.

In the above log, you can see that the Tenor received the call from IP, the digits were correct and it went to route this call out the first available PSTN line; however, we never see the Call Proc coming from the CAS. This is a good indication that the Tenor did not detect dial tone and cannot dial out the digits if there is no dial tone.

There is a command on the Tenors to test each PSTN/FXO line to see if the Tenor can detect dial tone.

- For Gen 1 Tenors, the command to start the test is "test o x" (where x is the PSTN line number to test, the o stands for off-hook). To end this test you type "test e".
- For Gen 2 Tenors, the command to start the test is "debug test o x" where x is the FXO line to test starting at line 0). To end this test, type "debug test e".

Once you start a test you must always end the test. Even if it provides results, you must still end it. Once it is running, it will run for 500ms. If it detects a valid dial tone, it will show you a message indicating "Dial tone detected". If no dial tone is detected, it will state "Dial tone not detected".

There could be several reasons for no dial tone. The easiest is that there is no cable connected to the port, or the cable is broken. The line from the PSTN could be broken as well. Another reason is if there is voice mail on the line and the voice mail notification is a stutter dial tone, the Tenor will not recognize this as a valid dial tone.

Disconnect Cause Code 16

This cause code, Normal Call Clearing, generally is seen when the call originator hangs up the call before it connects. This may happen if they dial a number that is busy or an invalid number on the PSTN where a fast busy or re-order tone is heard. The Quintum Analog units do not report a busy connection or re-order; they will simply allow the tones to play through, but will not show a connection. Users will usually hang-up upon hearing these tones.

Another reason for this could be long Post Dial Delay. If the call takes too long to get ring back, users may hang up. There is little that can be done from the Tenor side to help this as there is no delay in the Tenor for processing the call from IP to the PSTN. There could be delay on the PSTN side however, and you would need to discuss this with your PSTN provider.