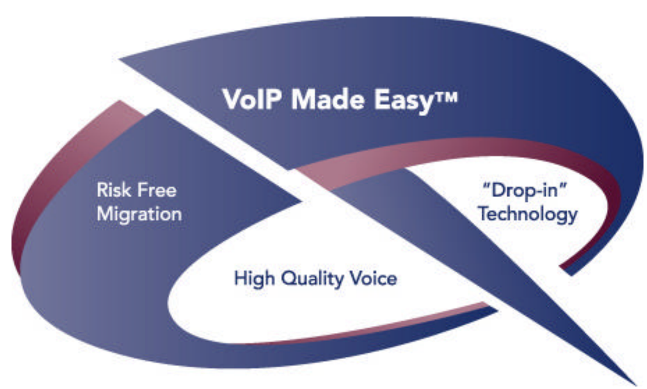
Section II: Tenor Features



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Chapter 1: Shared Features of the Digital and Analog Tenors

Unique Design

All of the Tenors pack lots of powerful VoIP features into one compact unit. The system's embedded design enables you to configure the unit directly without depending upon another operation system; in the Analog & Digital Tenors, they are all delivered in one 19" rack-mountable unit.

State-of-the-Art Configuration and Network Management

Just plug and go. Tenor's CAMWizard makes Tenor's configuration and management simple. Once you define an IP address via the RS-232 port, Tenor's CAMWizard will send you on your way with a default configuration. Further configuration through the Tenor's CAM is available as well as advanced configuration using Tenor's CLI. This allows you to set specific configuration options that will help you to further maximize Tenor's capabilities for a wide range of applications.

Both the Tenor CAMWizard and CAM are HTTP based interfaces making configuration easy. Further, many applications take as little as 10 minutes to configure the Tenor via the CAMWizard and be up and VoIPing immediately.

The CLI or Command Line Interface allows for a wide range of configuration parameters and options for specialized applications.

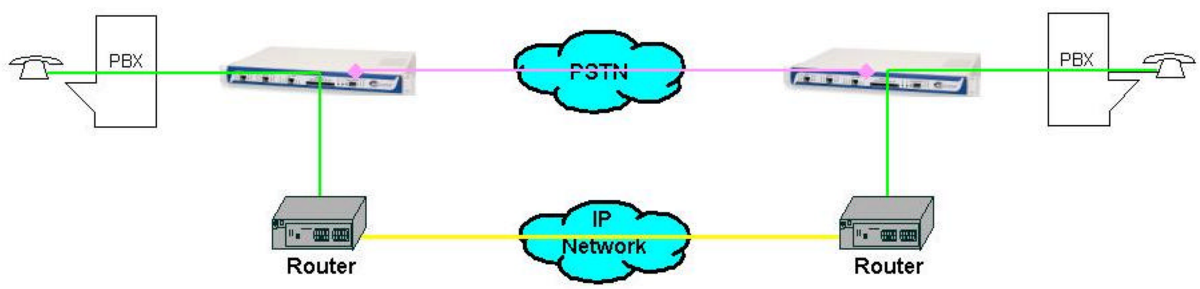


FIGURE II-1 SELECTNET/TASQ SAFETY NET (DIGITAL-TO-DIGITAL TENORS)

SelectNet™ Technology Safety Net (Auto-Switching)

Tenor's built in SelectNet™ (also known as TASQ™ - Transparent / Auto-Switch Quality) Technology safety net feature virtually guarantees that each call going VoIP will be not only routed successfully, but will deliver quality sound.

If the quality for a VoIP call becomes unacceptable - according to the amount of packet loss configured; combined with measured network Delay & Jitter – the Tenor will switch the call to the PSTN automatically, without you even knowing. The Digital Tenor is even able to complete this transaction in the event of a complete IP network failure.

So, even if you are using the public Internet, voice quality can be maintained.

Note

In order for SelectNet™ to work, there must be an actual connection from the Tenor to the PSTN.

SelectNet™ Technology Safety Net (continued)

Digital Tenor to Digital Tenor

(As diagrammed in Figure II-1 on previous page)

Most digital lines provide the DID or the dialed number when sending a call to the Tenor. So when the Digital Tenor receives an incoming call from the PSTN, digits are delivered; but possibly only the last 4 digits. The Tenor sees these digits and can determine how to route the call.

For our Auto-Switch Scenario involving a Digital Tenor, we assign a phone number as the Auto-Switch Number *(asnumber)* and set the *astype* as DID (0). Please be sure that this phone number is not used for anything on the PBX in the Digital/DID case, as the call will not be routed to the PBX. Whenever a call comes from the PSTN with these digits, the Tenor knows immediately that it is an Auto-Switch call from another Tenor, and goes through the Auto-Switch process immediately. With Digital Tenors and lines, auto-switching will work even if the IP cable is pulled out of the Tenor. *(See below to review the process in detail)*

1. The left side (Origination Side) places call to the right side (Termination Side).
2. Call goes over IP and connects to the destination (Through the "Termination Tenor").
3. When the call connects, the Termination Tenor sends, over IP, its Auto-Switch DN to the Origination Tenor, and tells the Origination Side that it expects to see DID <direct inward dialing> digits from the PSTN if an Auto-Switch call is necessary.
4. The Termination Tenor also sends the Origination Tenor a unique Auto-Switch ID number for this (and each) call.
5. While the users are conversing, the IP network degrades.
   a. Users may hear noise on call due to IP degradation.
   b. Tenor detects IP degradation.
6. When the Packet Loss exceeds the allowable threshold, the Call Origination Tenor starts to initiate a connection over the PSTN to Termination Tenor's Auto-Switch DN using a series of In-Band signals.
   a. The Termination Tenor sees the call come in from PSTN.
7. The Termination Tenor identifies the call as an Auto-Switch call by the digits that were delivered by the PSTN.
8. Termination and Origination Tenor synchronize on the PSTN channel to confirm the Auto-Switch and which call is to be switched.

The call is then switched seamlessly, without any User intervention!

Notes:

•Auto-Switched calls do not get switched back to IP.

•Calls get switched one at a time, not in a block.

•Calls get switched in age order; oldest first.

•The Tenor does not have to wait for first call to drop before switching next call.

•The Termination and Origination Tenors connect the PSTN channel to the active voice channel and disconnect the IP connection.

•Auto-Switch DN on Termination Tenor becomes available for next call.

SelectNet™ Technology Safety Net (continued)

Analog Tenor-to-Analog Tenor (or when the Termination Tenor is Analog)

(As diagrammed in Figure II-2 below)

Unlike Digital lines, Analog lines are not capable of delivering the dialed numbers and therefore, the Tenor cannot readily identify a call as an Auto-Switch, not by the digits anyway. So, as you will see the process is different than in our Digital Example above.

In most cases where there are multiple analog lines, there is only one hunt number that everyone dials into to call the office. When this hunt number is dialed, the call goes to the PBX and either to a receptionist or an auto-attendant where the caller is prompted for the person they wish to talk with and their call is transferred to the person. Because of all of this, you need to configure the hunt number as the Auto-Switch Number (*asnumber)* and set the type (*astype)* as No-DID (1), meaning that the Tenor will receive no digits on the inbound call.

When the Origination Tenor is prompted to Auto-Switch a call to the Destination Analog Tenor, it must first communicate, over IP, to the Analog unit that it will send an Auto-Switch Call to it. The Destination Analog Tenor will then set itself in Auto-Switch Mode. Any call that comes in to the Analog Tenor's PSTN ports while it is in Auto-Switch Mode will be run through the Auto-Switch process. If someone else calls in during this time period, the Analog Tenor will attempt to put the call through the Auto-Switch process and wait for a verification code from the Origination Tenor. Since the person calling is not able to send this verification code, they will hear about 1 second of silence. Since the unique Auto-Switch verification code was not received, the Destination Tenor knows that this is not an Auto-Switch call, and will route the call to the PBX, and the caller will hear the ringing of the PBX. When the appropriate call comes in from the Origination Tenor, and the unique Auto-Switch ID code is verified, the Tenors will complete the Auto-Switch. *(See next page to review the process in detail)*

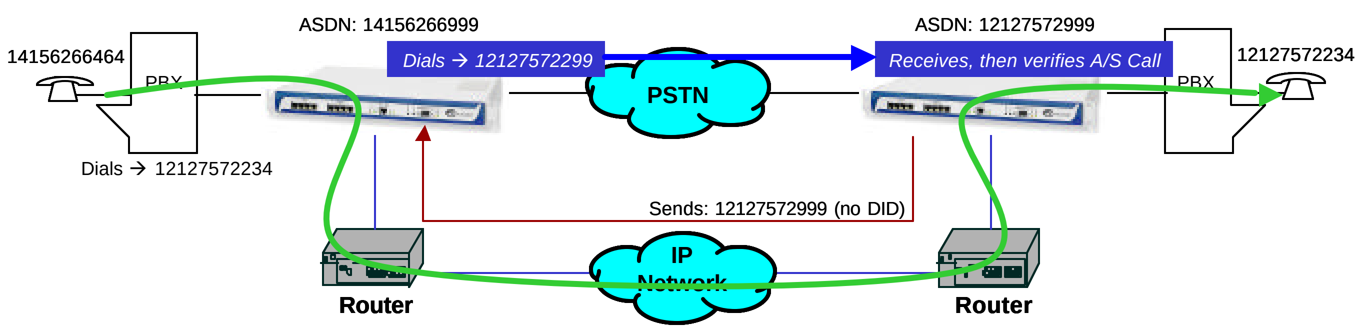


FIGURE II-2: SELECTNET/TASQ SAFETY NET (ANALOG-TO-ANALOG TENORS)

Analog Tenor Auto-Switch Process (continued)

1. The call goes over IP, from left to right, just as before, except that our Termination Tenor is Analog, and can't receive DID <Direct Inward Dialing> digits.
2. When the call connects, the Termination Tenor sends its unique Auto-Switch ID# (this time marked as "Non-DID") to the Origination Tenor.
3. Because the Termination Tenor cannot receive DID digits from the PSTN, this time the Tenors will negotiate the Auto-Switch connection, but it is dependant on the IP for a slightly longer amount of time.
    a. If the IP network TOTALLY and suddenly fails, the call will not be able to be Auto-Switched.
    b. While users are conversing, the IP network degrades.
    c. Users may hear noise on call due to IP degradation.
4. When the Packet Loss exceeds the allowable threshold, the Origination Tenor sends a message over IP to the Termination Tenor that an Auto-Switch call is coming.
5. The Termination Tenor then goes into "Auto-Switch mode" for a short period of time.
    a. The Termination Tenor sees **all** calls coming in from the PSTN as potential Auto-Switch calls.
6. The Origination Tenor dials the Auto-Switch Number.
7. When the Termination Tenor is expecting to receive an Auto Switch-Call, it attempts to verify each incoming call as an Auto-Switch call.
8. When the correct incoming call is verified, the Termination and Origination Tenors communicate with each other on the PSTN channel to confirm the "handoff" of the call, and that IP call has now been patched over PSTN.
9. If an incoming call is verified as a non-Auto-Switch call, the call is routed, as usual, to the PBX. The non-Auto-Switch Caller may hear one second of silence, and then ringing from the PBX.

The call is then switched seamlessly, without any User intervention!

Notes:

•Auto-Switched calls do not get switched back to IP.

•Calls get switched one at a time, not in a block.

•Calls get switched in age order, oldest call first.

•The Tenor does not have to wait for first call to drop before switching next call.

•The Termination and Origination Tenors connect the PSTN channel to the active voice channel and disconnect the IP connection.

•Auto-Switch DN on Termination Tenor becomes available for next call.

Uninterrupted Service

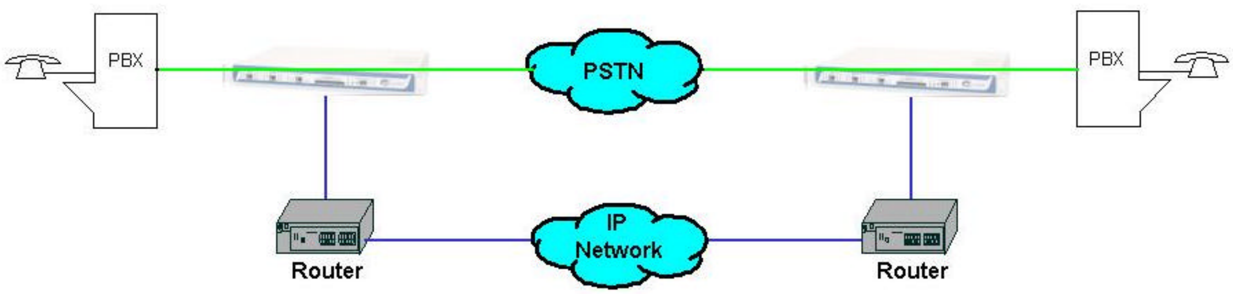


FIGURE II-3 UNINTERRUPTED SERVICE

If for any reason power is lost to the unit, or the unit is down, all call routing operations will bypass the Tenor unit. This means that the Tenor will be ignored and all calls will “bypass” the unit and route successfully to reach the PSTN or PBX. This feature can also be manually enabled/disabled from any of the configuration interfaces to allow the customer to test VoIP on their terms.

> **Note**
>
> For Bypass to work properly, both the PBX and PSTN connections/lines must have the same physical and signaling characteristics. For example, both lines are T1, B8ZS/ESF, E&M signaling, etc.

Dynamic Call Routing

Tenor’s call routing capabilities are state-of-the-art. Voice and fax calls are digitized and transmitted through the 10/100 Ethernet LAN and routed out to the corporate WAN. Tenor will first identify the call origination site - PBX, PST or IP Network - and then route it according to any parameters or defaults you configure in the routing database.

Dial Plans

The Tenor supports both Public and Private dial plans.

When set up for a public dial plan, the Tenor bases its routing on the E.164 standard using the full international number format. This includes the country code, area/city code and number. This does not mean that when you want to dial a long distance you must dial in the international format, unless that is a requirement of your PSTN, the Tenor will take care of building the call out, if needed, and route.

> **Note**
>
> In North America, the country code is 1, as is the long distance prefix. This may become confusing in later discussions.

When set up for a private dial plan, the Tenor uses the parameters that you configure in regards to the private plan. The Tenor does not make any adjustments to the number when setup in a private dial plan. This works best if you have a voice VPN that allows 4-digit dialing between locations.

Directory Number Tables

The Tenor has 3 tables that you configure with directory numbers (dialed numbers) depending on the function that you want for the number/number ranges. The tables are;

- Bypass Directory Number Table (BDN) - numbers and/or ranges configured here will always go from the PBX to the PSTN and never over IP of the local Tenor. The Tenor matches based on the number as it is received from the PBX.
- Local Directory Number Table (LDN) - numbers and/or ranges configured here on a local Tenor specify what numbers are accepted and routed out to the PBX from the IP network.
- Hop-off Table or LAM - numbers and/or number patterns configured here on a local Tenor specify what calls are allowed to come in to this Tenor from the IP network and “hop-off” to the PSTN.

PBX Calls

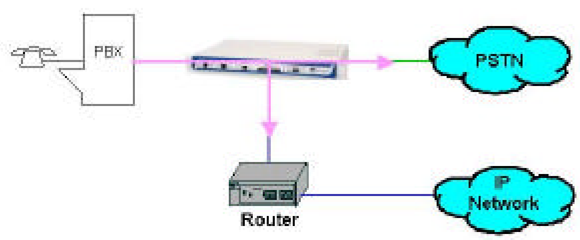


FIGURE II-4 PBX CALLS

Calls that originate from the PBX can be routed to either the IP network or the PSTN. The decision is made by Tenor based on the number that is dialed. For example, you can configure certain calls to always go to the PSTN. Any calls received from the PBX that match a LDN or Hop-Off DN of a Tenor on the IP network will be routed over the IP network.

If no match is found for a dialed number on the IP network, the call will be automatically routed to the PSTN. Also, if a call is routed over the IP network and for any reason the call does not connect, the Tenor will attempt to send the call out to the PSTN.

Note

When the Tenor receives a call from the PBX, it checks the BDN table first for a match to the number as received (prefix and number). If no match is found in the BDN table, then the Tenor checks for a match on the IP network based on the International format (no prefixes). If no match is found on the IP network, the call is routed to the PSTN as it was originally received from the PBX.

PSTN Calls

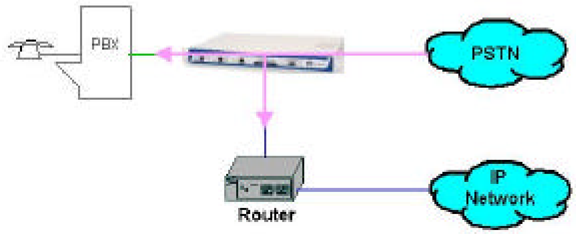


FIGURE II-5 PSTN CALLS

Calls that are received from the PSTN can be routed either to the PBX or over the IP network depending on your configuration.

Note

Calls can only be routed from the PSTN to the IP network in certain configurations / applications. Normally, any calls that come in from the PSTN are unconditionally routed to the PBX.

IP Network Calls

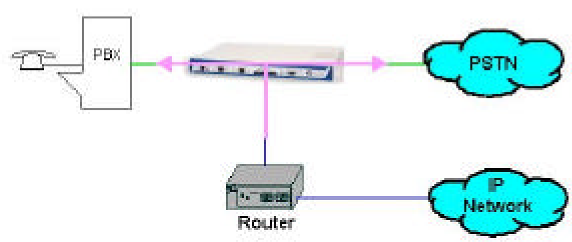


FIGURE II-6 IPNETWORK CALLS

Calls that are received from the IP network can be routed to either the PBX (based on matching an existing LDN) or to the PSTN (based on matching an existing Hop-Off DN).

Notes

- LDN's and Hop-Off DN's are matched based on the full International format (country code, city/area code, number) and are always configured at the destination Tenor end.
- LDN's and Hop-Off DN's have a set matching priority, 2 & 1 respectively (higher the number, the higher the priority). If there is a Hop-Off DN that is configured exactly as a LDN, then a call that matches these will be routed as a LDN to the destination PBX since LDN's have a higher priority.
- If two or more Tenors have the same LDN's and/or Hop-Off DN's configured, then the Tenors will perform load-balancing between themselves.

Bypass Calls (BDN)

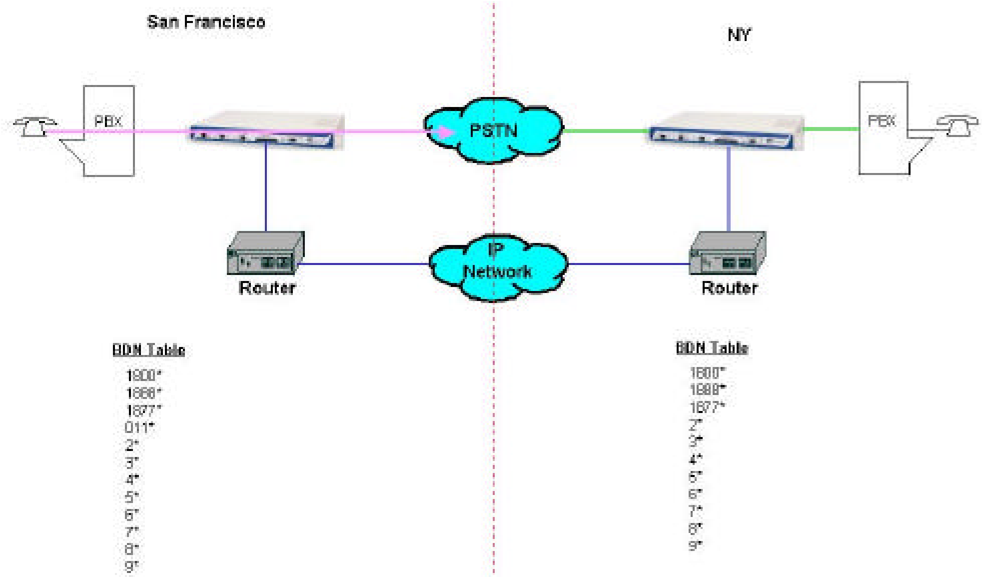


FIGURE II-7 BYPASS CALLS

Bypass calls are calls that when originated from the PBX are unconditionally sent out to the PSTN regardless of matching a number on the IP network. Typically, these are setup using a pattern instead of the entire number, but either can be done. Bypass DN's are matched based on the digits received from the PBX as they are sent to the Tenor.

Note

If no match is found on the IP network (like 911, 411, 800 calls, etc.) then the call will automatically be routed to the PSTN.

Note

The * can be used as a wild card to represent any digit, any number of placeholders. If used, it must always be the last character of the digit string. Example: 4609* represents the range of numbers from 4609000 through 4609999.

Hop-Off Calls (LAM)

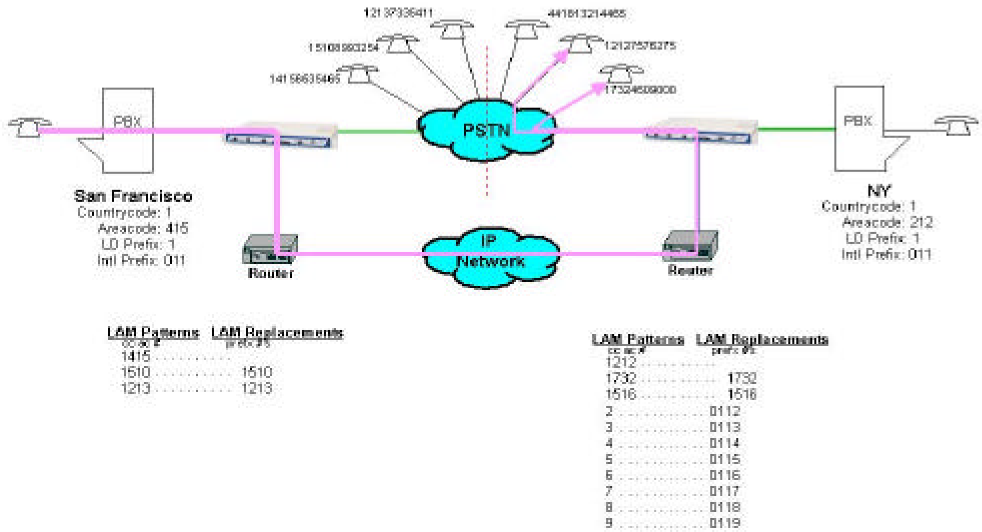


FIGURE II-8 HOP-OFF CALLS

Hop-Off calls, sometimes called Leaky Area Mapping or simply LAM, are calls that originate at one location, traverse the IP network and then get routed to the local PSTN thereby saving long distance and/or international toll charges. Matching and routing are based on the E.164 number standard using country code, area code and number. Not on the international dial prefix or the long distance dial prefix.

In the Tenor, these are setup using patterns and replacements. The pattern is what is used to match a dialed number and is configured on the Tenor that will send the call out of it's locally connected PSTN connection. When a match for the pattern is found, the destination Tenor deletes the matched pattern from the dialed number.

LAM Replacement Digits are those digits that you wish to append on to the number after the pattern has been deleted to satisfy the PSTN requirements. For example, in the above diagram, if a user behind the PBX dials 12127576275, the SF Tenor receives the digits (12127576275), sees that there is a pattern that matches this on the IP network (NY Tenor has a pattern of 1212) and routes the call over IP to the NY Tenor. When the NY Tenor receives the call, it matches the digits (12127576275) to the pattern of 1212 and deletes the 1212 off of the number. Since there is no replacement configured for this pattern, the number is sent to the local PSTN as a local 7-digit call, as it should be.

If that same user were to dial 17324609000, the SF Tenor receives the digits (17324609000), finds a match on the IP network (1732) and routes the call to the NY Tenor. The NY Tenor receives the call, matches the digits to the pattern of 1732 and deletes 1732 off of the number (due to the match). If the NY Tenor were to send the call out to the local PSTN it would only send 4609000 and the local PSTN would not routed it correctly, so there is a replacement of 1732 to add back on to the number. The replacement is added to the number and sent out to the local PSTN as 17324609000, which the PSTN routes correctly. This is done so that the NY PSTN sees the call originating from a 212 area code rather than a 415 area code.

> **Note**
>
> SelectNet™ is not typically used or enabled for this type of call, as it will use 2 channels on the PSTN line to accomplish the Auto-Switch.

Automatic Call Type Detection

The Tenor automatically detects voice, fax and modem calls and supports both voice and fax calls. It takes analog voice and fax data and converts it into IP packets, using the ITU H.323 standard, for transport over the IP network.

Modem Call Bypass

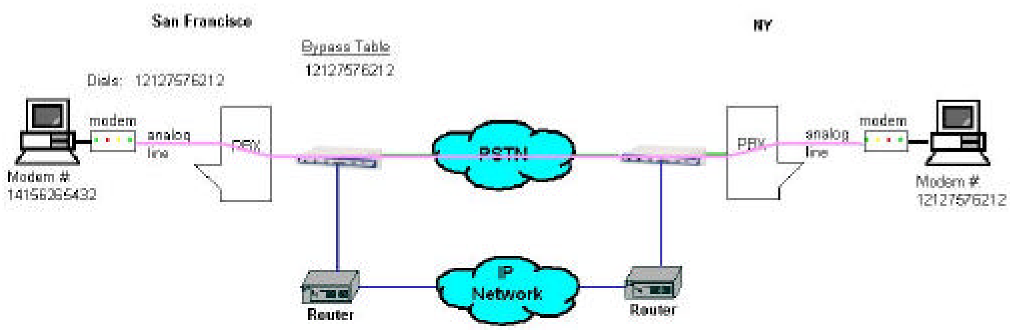


FIGURE II-9 MODEM CALL BYPASS

Since analog modem calls are not reliable over IP due to the voice compression that is utilized, the Tenor can actually detect a modem call and force a bypass to the PSTN for reliability. When a user launches a modem call from their PC, the Tenor routes the call over the IP network to the destination end. When the destination modem answers and tries to synchronize with the origination modem, the Tenor hears the modem tones. The Tenor will disconnect that modem call before full connection and memorize the number dialed in to its Bypass table automatically. The next time that number is dialed, the call is sent directly out to the PSTN.

H.323 Gatekeeper Call Control Management

The Tenor has a built-in H.323 Gatekeeper, so there is no need to purchase, install and configure a separate one. The built-in Gatekeeper performs IP call routing functions, which comply to the H.323 industry specifications for voice control and management. The Tenor's Gatekeeper performs call routing functions for calls entering and exiting a site.

In addition to the Gatekeeper, the Tenor utilizes a master Gatekeeper called the Border Element. The Border Element is the master Gatekeeper of a network of Tenors.

H.323 Overview

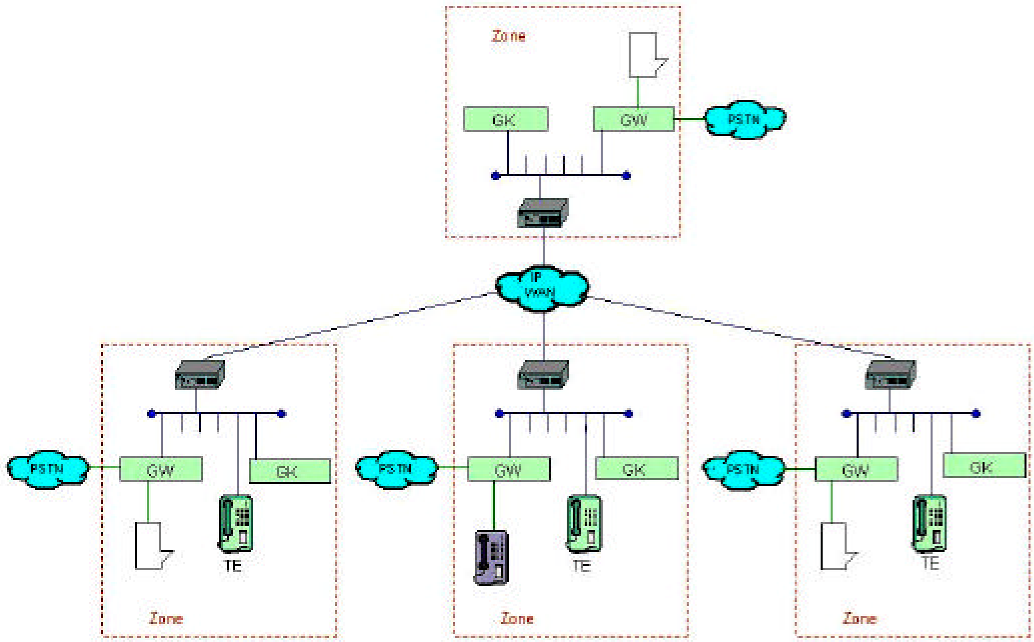


FIGURE II-10 H.323 HIERARCHY

The H.323 standard specifies the process to send and receive multi-media (voice, fax and video) over traditional IP networks. The standard defines 4 types of functions and they are;

- Terminals and Endpoints: IP based phones and PC based voice (NetMeeting) that terminates H.323 directly to the end user.
- Gateways: Units that provide the conversion between H.323 and traditional multimedia formats.
- Gatekeepers: Units that perform all call control functions for H.323 such as call routing.
- MCU (Multi-point Control Unit): Units that allow multiple video and audio streams to be "conferenced" together.

H.323 Call Control Communication

Gateways and terminals, when connected to an IP network, will "register" with a Gatekeeper. In the registration process, the gateway and terminal will tell the Gatekeeper what their IP address is, what type of unit they are (gateway or terminal) and any routing information such as phone numbers, number patterns, etc.

The routing information (with IP address) is continuously sent to the Gatekeeper by the Gateway and terminal as a "heart-beat" or keep alive signal.

The Gatekeeper maintains the master routing table of all the devices that have registered with it.

When a call is originated using either a gateway or terminal, a call request is sent to the Gatekeeper. The call request provides the Gatekeeper with the dialed number and requests the routing information.

The Gatekeeper will look-up in its master table to find a match for the number. When a match is found, the Gatekeeper sends the destination IP address corresponding with the number to the origination gateway or terminal. The gateway or terminal can now address the IP packet with the correct IP address to the destination end.

Gatekeeper

There can be a single Gatekeeper for the entire network, or each location can have a Gatekeeper. When each location has a Gatekeeper, that location is considered a **zone** on the network. If there is more than one Gatekeeper at a single location, then each Gatekeeper comprises a zone. Therefore, a zone is considered to be a single Gatekeeper and any Gateways and terminals that are registered with it.

In this scenario, where there are multiple Gatekeepers for a network, one Gatekeeper is designated the master Gatekeeper. Quintum calls this the Border Element. The Border Element maintains the master routing information for the entire network.

Quintum - H.323 Architecture

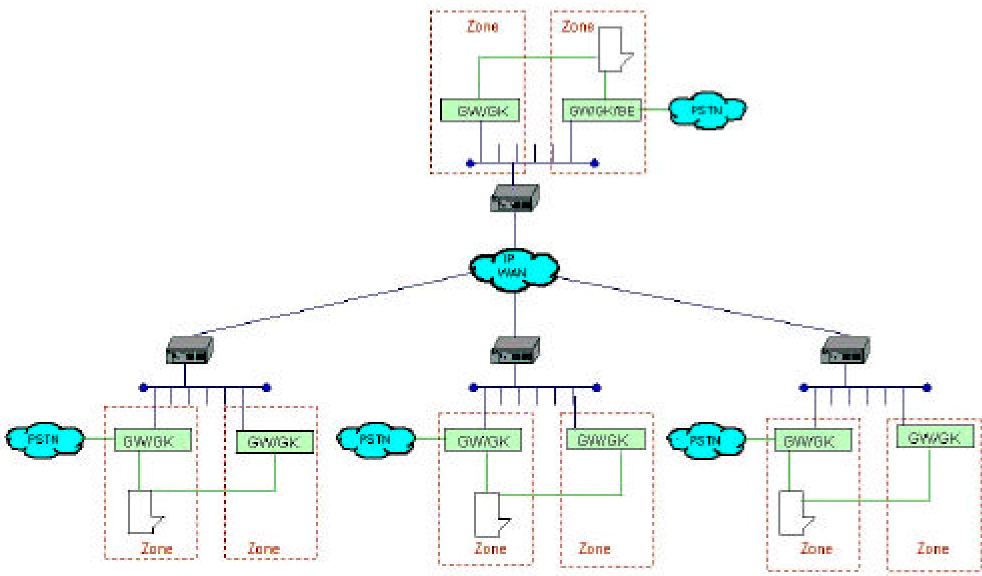


FIGURE II-11 H.323 ARCHITECTURE

The Quintum Tenor not only provides the Gateway function, but also can perform both the Gatekeeper and Border Element functions simultaneously. There is no need to dedicate a unit to any one function.

Currently, the Quintum Tenor operates in a standalone fashion. While you can “stack” the Tenors in one location, each unit will operate independently of any others in the same location. Also, each Tenor runs performs Gatekeeper function (along with the Gateway functions). So, as mentioned before, each Tenor is considered a zone. When installing a network of Tenors, you must decide which unit will be the Border Element and configure accordingly.

When Quintum introduces its multi-unit operation feature, later this year, at that time, multiple Tenor units in one location can be logically connected and configured like one large unit. Also, one unit will take on the responsibility of the Gatekeeper function automatically. If this unit should fail, then another unit within the “stack” will take over the Gatekeeper function. The units registering with the Gatekeeper as well as the Gatekeeper itself will be the zone. You will still need to manually assign the Border Element.

Quintum - H.323 Call Control Communications Hierarchy

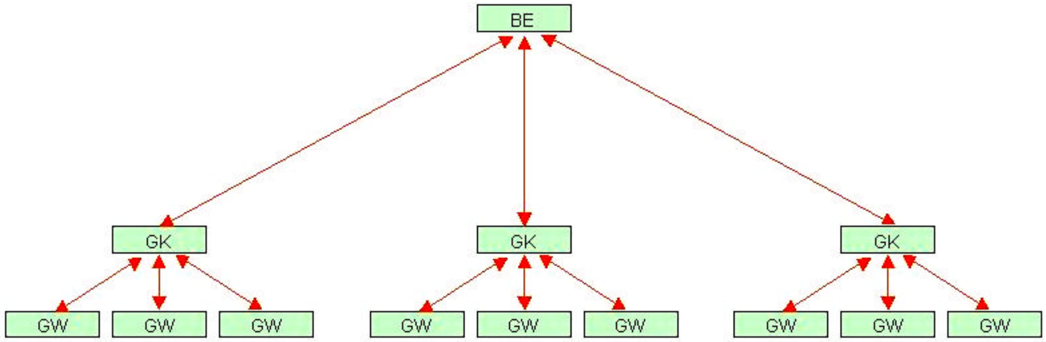


FIGURE II-12 H.323 CALL CONTROL

Within a network of Quintum Tenors where you have a Border Element, and Gatekeepers/ Gateways, the call control communications flow up to the Border Element. In the stand-alone operation that Quintum supports today, each unit registers internally to the Gatekeeper process in every unit. The Gatekeeper processes will register with the Border Element.

In the multi-unit operation, the Gateways will register with their local Gatekeepers and the Gatekeepers will register with the Border Element.

This Hierarchal flow keeps the call control traffic over the IP WAN to a minimum.

Powerful System Monitoring

There are many different ways to monitor the health of the unit, including LED’s and alarms.

LED's appear on the front of the unit and light up according to an internal alarm the system is experiencing. Through the Tenor CAM management system or a telnet session, you can view active alarms as well as the alarm history. Each alarm tells you specifically what problem the unit is encountering and Quintum uses (where applicable) standard telecommunication type alarms.

PacketSaver™

PacketSaver™ multiplexes multiple individual VoIP sessions (destined to the same termination Tenor) into consolidated IP packets, significantly reducing the total amount of bandwidth needed to support voice calls over IP networks. This more efficient approach to transporting VoIP results in three key benefits:

1. By minimizing the bandwidth required for VoIP traffic, PacketSaver™ lowers the likelihood that other congestion on the network will threaten voice quality.
2. Because PacketSaver™ reduces the total number of packets used for VoIP traffic; it also reduces the chances of packet loss - another factor that can affect voice quality.
3. In minimizing the amount of bandwidth required for VoIP, PacketSaver™ contributes to the overall efficiency of the IP network - reducing congestion and lowering overall infrastructure costs.

This last benefit is particularly appealing to VoIP service providers, whose profitability is largely contingent on their ability to deliver high-quality voice services over IP networks at the least possible cost.

Traditional VoIP

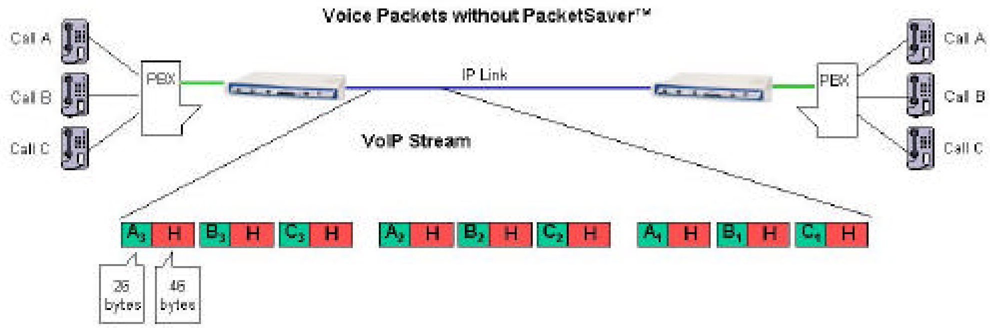


FIGURE II-13 TRADITIONAL VOIP PACKET

In the case of Voice over IP, packets are usually created as a VoIP gateway receives a voice stream. The gateway compresses the voice and digitizes it into a packet payload, adding a header with destination information. For example, the Tenor uses the G.723.1 codec for voice compression by default. This compresses the voice down to 6.3kbps. In terms of IP payload, this comes out to a 24byte payload. The packet header (containing timestamp, sequence number, destination, etc) is 46 bytes. Almost double that of the actual payload.

With conventional gateways, packets are created as each individual voice stream hits the gateway. This results in the creation of tremendous header overhead. The network - which usually has lots of data packets flowing over it already - can easily become saturated, causing congestion and even lost packets. This can comprise the voice quality, as well as the overall health of the network.

Quintum's PacketSaver™ Method

Quintum's PacketSaver™ technology is a packet assembly technique that queues up several voice and/or fax packets that are destined to the same termination address, and multiplexes them into one or more larger packets, without introducing unacceptable delays in packet processing. When the maximum payload size is reached (user configurable), or the pre-determined time limit for packet construction is reached (whichever occurs first), it is sent to the common destination point.

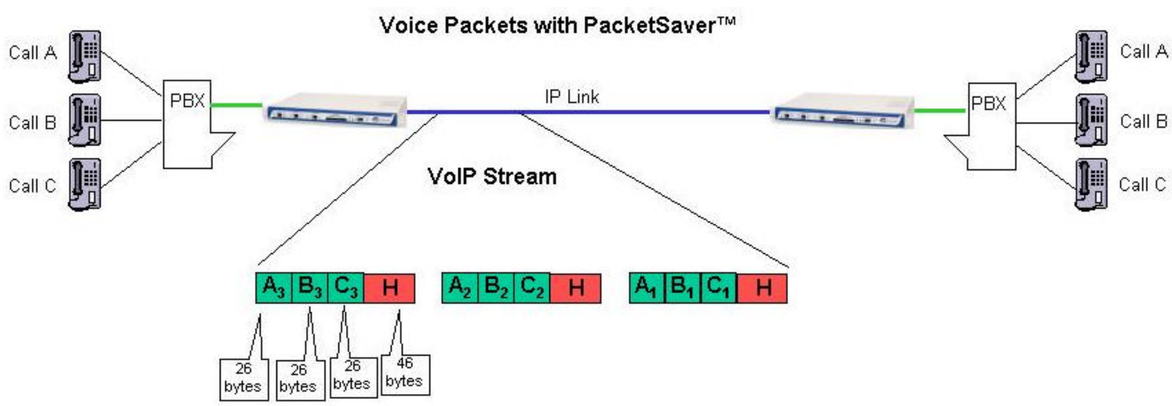


FIGURE II-14 VOIP PACKET WITH PACKETSAVER

When this multiplexed packet reaches the destination device, it is then de-multiplexed, allowing the packets for each original voice conversation and/or fax transmission to be distributed to their appropriate end-point destinations.

The result of this is that fewer, but larger packets are sent over the network, which means less header overhead is sent. PacketSaver™ also reduces bandwidth requirements and router processing loads while improving reliability. With just 3 calls, PacketSaver™ can reduce the overall bandwidth by 40%.

Note

If you plan to use the PacketSaver™ feature, then you must enable this feature on all Tenor units in your network. Further, this feature is not compatible with any other VoIP product.

Note

Quintum recommends that the maximum payload size be 1500 bytes and the maximum time to accumulate be 20ms. These settings allow for the maximum amount of packet muxing without sacrificing quality.

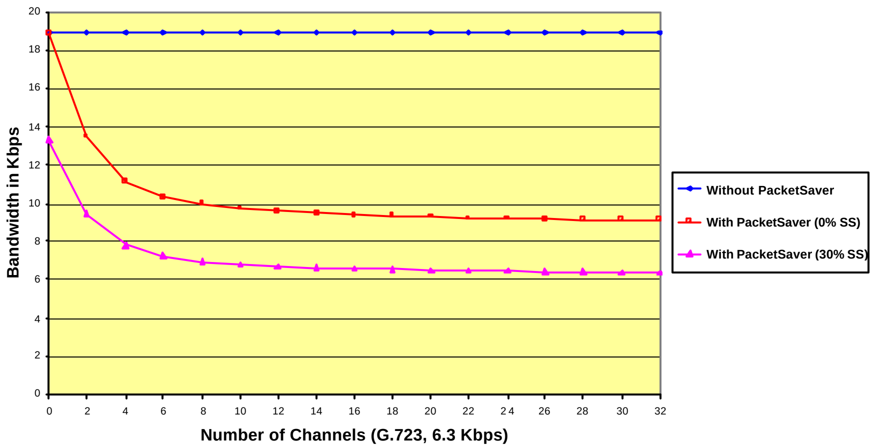


FIGURE II-15 PACKET CHART

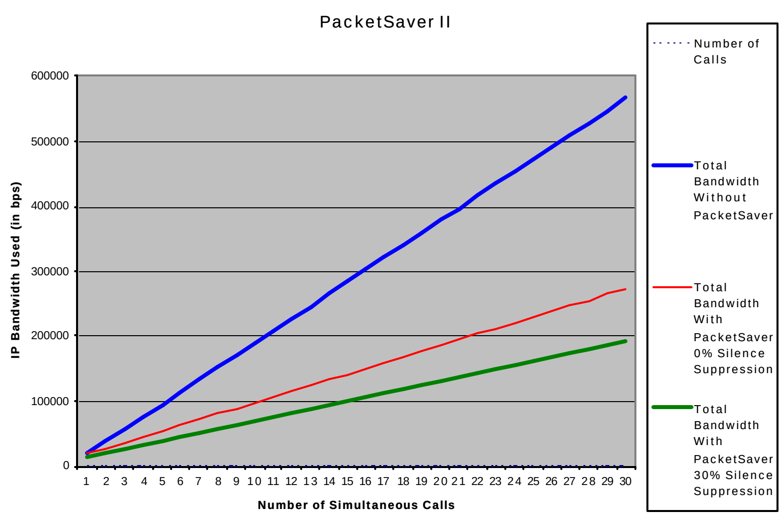
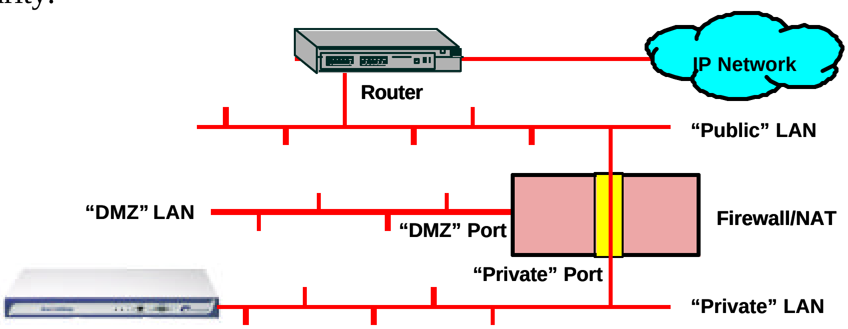


FIGURE II-16 PACKETSAVER BANDWIDTH COMPARISON

NATAccess™

NATAccess™, our Network Address Translation (NAT) feature is integrated within Tenor. The NAT capability enables the Tenor to match an Internal IP address with a public External IP Address, which facilitates routing Voice Packets around Firewalls and NAT Servers without compromising the integrity of IP Network Security.



Deploying the Quintum Tenor with NAT*Access* on the "Private" LAN

FIGURE II-17 NATACCESS™

Interoperability

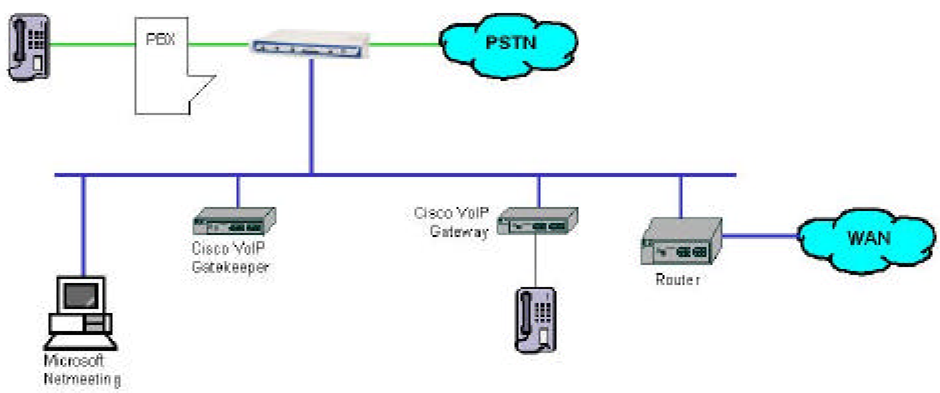


FIGURE II-18 INTEROPERABILITY

Interoperability is on everyone's mind, and at Quintum we proactively make efforts to support this idea and will continue to add features and functionality so that when you purchase and install a Quintum product, you do not become that "island" in the vast network.

Towards this end, the Quintum Tenor now provides interoperability between Cisco (both their Gateway and their Gatekeepers) and Microsoft's NetMeeting. There is detailed information regarding Interoperability with Cisco Products and NetMeeting clients in Section 9 of this guide.

Of course when interoperating with other vendors, some unique features of the Quintum Tenor may not be available since the other vendor(s) do not support these features like SelectNet™ and PacketSaver.

Call Detail Reporting Support

Through the Call Detail Reporting (CDR) feature, the Tenor unit is able to generate a CDR at the completion of each call. A CDR is a string of ASCII information that contains call information such as:

- Calling Number.
- Called number (if available).
- Call start and end date/time.
- Call duration.
- Line and channel (on the PSTN or PBX side) that the call used.

This information can be gathered by a billing system and transformed into real billing records to send to the customer. There is more detailed information regarding CDR data in Section 9 of this guide.

IVR and RADIUS

Interactive Voice Response (IVR) is a feature of the Tenor that enables you to offer services, such as Pre-paid calling cards and Post-paid accounts to your customers. The Tenor uses the **RADIUS (Remote Authentication Dial-In User Service)**, for authenticating and authorizing user access to the VoIP network. The RADIUS is a standard protocol that provides a series of standardized messages formats for transmitting and receiving dialed information, account data and authorization codes between the network access gateway and the billing server.

As a result, the RADIUS enables the Tenor to interoperate directly with billing server application software from a wide range of vendors. To provide redundancy, the Tenor supports two RADIUS servers: Primary and Secondary.

The IVR interface enables the Tenor to play back interactive pre-recorded voice messages to a customer calling in from the Public Switched Telephone Network (PSTN), requesting information such as account number, PIN number, calling number, and remaining credit. The caller is prompted for each piece of information and the digits are captured by the Tenor and transmitted into RADIUS format and sent over the IP network to the RADIUS server. The RADIUS will use the input data to identify the customer, verify the identity using the PIN code, check the account status, and then send back messages in RADIUS format to authorize the Tenor to proceed with the call. The call will then be routed over the VoIP network to the appropriate remote Tenor.

Through a IVR Prompt Server, you are able to pre-record and customize voice prompt files which lead the customer through the calling card procedure. You can pre-record messages to meet your network and customer needs. The IVR Prompt Server communicates with a Tenor via IP link using TFTP protocol. Tenor supports two IVR Prompt Servers: Primary and Secondary. The IVR Prompt server can be a separate PC on the same IP network as the Tenor.

Easy Upgrades

Upgrades to the Quintum Tenor are done using standard FTP functions.

Chapter 2: Tenor Digital Features

Multiple Channels / Signaling Supported

The Tenor Digital MultiPath product line can support from 8 channels (8 simultaneous VoIP connections) up to 30 Channels, depending on your needs.

Tenor connects to a PBX via upstream T1/E1 lines and to the PSTN via downstream T1/E1 lines. For each interface (T1/E1), there are two types of signaling supported; Channel Associated Signaling or CAS and Common Channel Signaling or CCS (mostly called ISDN).

Fractional T1/E1 Support

Tenor supports fractional T1/E1 bandwidth assigned to you (via carrier) in increments of 64kbps. this feature will benefit the small enterprise.

Tie Trunk Plus Capability

Tenor can replace any tie trunk. It provides all of the functionality of a tie trunk, including the considerable cost savings, but eliminates the need for a PBX trunk to be configured, or marked as a tie trunk.

A traditional tie trunk is a PBX configured direct connection between two PBX's in separate locations. The tie trunk bypasses the PSTN network, which results in considerable savings.

Your PBX does not need any additional configurations. Tenor treats all trunks the same without compromising voice quality.

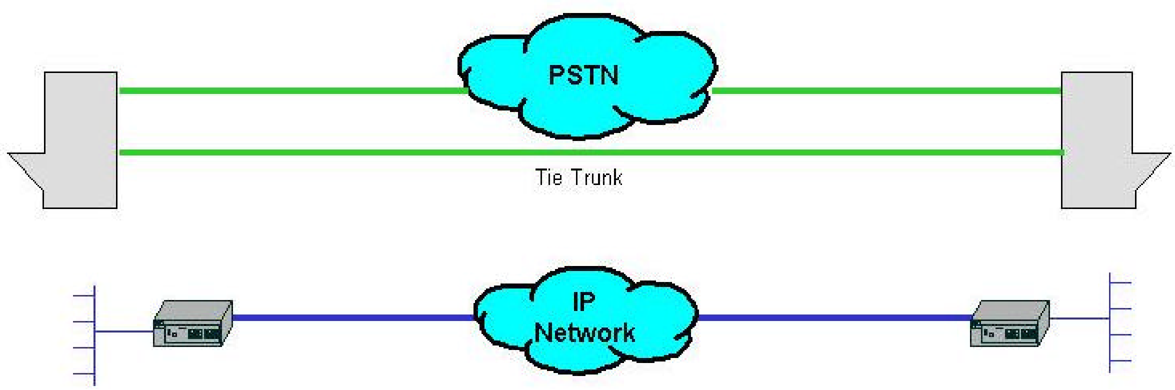


FIGURE II-19 TYPICAL TIE-TRUNK

Chapter 3: Tenor Analog Features

Multiple Lines / Signaling Supported

The Analog Tenor unit can support up to eight analog lines, which means you can support up to eight simultaneous VoIP calls, depending upon your need.

The PBX ports implement the FXS interface (connection to a telephone, key system or PBX); the PSTN ports implement the FXO interface (connection to the Central Office / PSTN).

The Analog Tenor supports Loop-Start signaling with 2 variations at this time; Loop Start Forward Disconnect (provides Disconnect Supervision) and Loop Start Reverse Battery* (provides Answer & Disconnect Supervision).

**Reverse Battery Signaling Type requires an additional Software Purchase and "Revision B" Hardware.*

Chapter 4: Hardware Specifications

Common Across Product Line

FIGURE II-20 TENOR DIGITAL AND ANALOG

- 19-inch rack mountable.
- Dimensions: 10.75" D x 17.25" W x 1.75" H.
- Weight: 7.2lbs.
- All network connections made in the front of unit.
- Air-cooled, no noisy fans.
- Power: AC 115-230 volts, 50-60 Hz.
- Operating Temperature: 40°-130° Fahrenheit, 5°-55° Celsius.
- Humidity: 20%-80%, non-condensing.
- Single printed circuit board design.

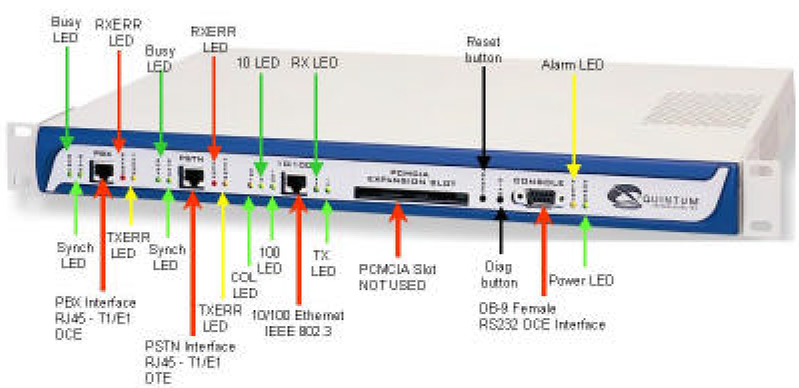


FIGURE II-21 DIGITAL TENOR - FRONT PANEL

Digital Tenor - Front Panel Overview

- 4 Models of Digital Tenor.
  - 800 - 8 simultaneous VoIP calls per unit.
  - 1600 - 16 simultaneous VoIP calls per unit.
  - 2400 - 24 simultaneous VoIP calls per unit.
  - 3000 - 30 simultaneous VoIP calls per unit.
    - Software upgradeable from one model to the next.
- Serial Interface.
  - DB-9 female RS232 DCE interface.
    - Initial speed is 38400.
    - Used to initial configure an IP address for the Tenor.

    - Connect using a PC and provided serial cable.
    - Use HyperTerminal to perform initial IP configuration.
  - Possible to connect a modem using a null modem cable.
- PCMCIA Slot - Not utilized.
- 10/100 Interface.
  - Provides standard RJ45 connection to Ethernet network.
  - Auto-senses either 10 or 100mbps network connection.
- PSTN Interface.
  - Provides standard RJ45 DTE interface.
  - T1 or E1 supported, standard T1 pin-out.
  - Straight cable is typically used to connect to PSTN.
  - Built-in CSU (software enabled).
- PBX Interface.
  - Provides standard RJ45 DCE interface.
  - T1 or E1 supported, standard T1 pin-out.
  - Straight cable is typically used to connect to PBX.
  - Built-in CSU (software enabled).
- Power LED.
  - Green.
  - When lit, indicates the presence of an external power source.
- Alarm LED.
  - Yellow.
  - When lit, one or more internal diagnostic test have failed and the unit may need to be replaced.
  - Will be lit briefly during power on self test (POST).
- TX & RX LED (10/100 interface).
  - Green (both).
  - Indicates the Ethernet transmit and receive status.
  - When flashing, data is being transmitted or received on the Tenor.
- 10 & 100 LED (10/100 interface).
  - Green (both).
  - When lit, indicates the speed of the LAN connection.
  - If off, the Tenor does not detect a LAN connection for that speed.
- COL LED (10/100 interface).
  - Yellow.
  - Indicates Ethernet collision status.
  - When flashing, indicates that some collisions are taking place on the Ethernet LAN.
- TXERR LED (PSTN & PBX interfaces).
  - Yellow.
  - Indicates transmit path errors.
  - When lit, the interface/line is in a Yellow alarm condition.
  - When flashing, other transmit errors are indicated.
- RXERR LED (PSTN & PBX interfaces).
  - Red.
  - Indicates receive path errors.
  - When lit, the interface/line is in a Red alarm condition.
  - When flashing, other errors are being received.
- SYNCH LED (PSTN & PBX interfaces).
  - Green.
  - Indicates the line status for each interface.
  - When lit, the line/interface is in an operational condition.

        - If line is ISDN, this LED will not light until the D-channel is active.
    - When off, indicates that the line is down due to a red alarm, yellow alarm or PRI layer 2 failure.
- BUSY LED (PSTN & PBX interfaces).
    - Green.
    - Indicates the channel status for each interface.
    - When lit, one or more channels on the interface are in use.
- DIAG button.
    - Recessed button for Quintum use only.
- RESET button.
    - Recessed button for resetting the Tenor.

Analog Tenor - Front Panel Overview

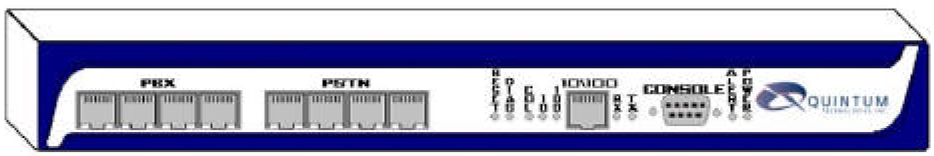


FIGURE II-22 ANALOG TENOR - FRONT PANEL

- 2 Models of Analog Tenor.
    - 400 - 4 simultaneous VoIP calls per unit.
        - 4 Analog PSTN & 4 analog PBX connections.
    - 800 - 8 simultaneous VoIP calls per unit.
- 8 analog PSTN & 8 analog PBX connections.
    - Not Software upgradeable from one model to the next.
- Serial Interface.
    - DB-9 female RS232 DCE interface.
        - Initial speed is 38400.
        - Used to initial configure an IP address for the Tenor.
        - Connect using a PC and provided serial cable.
        - Use HyperTerminal to perform initial IP configuration.
    - Possible to connect a modem using a null modem cable.
- 10/100 Interface.
    - Provides standard RJ45 connection to Ethernet network.
    - Auto-senses either 10 or 100mbps network connection.
- PSTN Interface.
    - Provides standard analog RJ45 FXO interface.
    - Model 400 uses pins 4 & 5 of each interface for lines 1 - 4.
    - Model 800 uses pins 4 & 6 of each interface for lines 5 - 8.
    - Straight cable is typically used to connect to PSTN.
    - FXO Loop-Start signaling support only. (With Forward Disconnect & Reverse Battery variants)
- PBX Interface.
    - Provides standard RJ45 FXS interface.
    - Model 400 uses pins 4 & 5 of each interface for lines 1 - 4.
    - Model 800 uses pins 4 & 6 of each interface for lines 5 - 8.
    - Straight cable is typically used to connect to PSTN.
    - FXO Loop-Start signaling support only. (With Forward Disconnect & Reverse Battery variants)

- Power LED.
  - Green.
  - When lit, indicates the presence of an external power source.
- Alarm LED.
  - Yellow.
  - When lit, one or more internal diagnostic test have failed and the unit may need to be replaced.
  - Will be lit briefly during power on self test (POST).
- TX & RX LED (10/100 interface).
  - Green (both).
  - Indicates the Ethernet transmit and receive status.
  - When flashing, data is being transmitted or received on the Tenor.
- 10 & 100 LED (10/100 interface).
  - Green (both).
  - When lit, indicates the speed of the LAN connection.
  - If off, the Tenor does not detect a LAN connection for that speed.
- COL LED (10/100 interface).
  - Yellow.
  - Indicates Ethernet collision status.
  - When flashing, indicates that some collisions are taking place on the Ethernet LAN.
- BUSY LED (PSTN & PBX interfaces).
  - Green.
  - 2 per interface.
  - Indicates the channel status for each interface.
  - When lit, line is in use.
- DIAG button.
  - Recessed button for Quintum use only.
- RESET button.
  - Recessed button for resetting the Tenor.

FIGURE II-23 TENOR REAR PANEL

Tenor Rear Panel Overview (all models)

- Power cord connection.
- Power Switch.
- Approvals Label.
- Serial Number & Ethernet/MAC address label.

Chapter 5: Standards and Specifications

Common Across all Platforms

Voice Coding Standards

Mu-Law: PCM Coding and companding law used widely in Japan and North America on T1.

A-Law: PCM coding and companding law used widely in Europe on E1.

Audio Compression Codecs

G.723: Supports compression rates of 5.3kbps and 6.3kbps.

- G.723 @ 6.3kbps is the default codec for the Tenor.

ADPCM: Adaptive Differential Pulse Code Modulation (G.721).

- Supports compression rates of 16kbps, 24kbps, 32kbps, & 40kbps.

G.711: Support 64kbps (no compression).

G.729: Supports a compression rate of 8kbps.

Digital Tenor Supported Standards

CAS Signaling (Robbed bit)

- E & M
  - Upstream (network side) & Downstream (user side)
  - Wink Start, Automatic Dial, Delay Dial, Immediate Dial
  - DTMF tones
- Loop-Start
  - Upstream (network side) & Downstream (user side)
  - Answer and Disconnect Supervision*
    - *Loop Start Forward Disconnect
  - DTMF tones
- Ground-Start
  - Upstream (network side) & Downstream (user side)
  - Disconnect Supervision
  - DTMF tones
- Feature Group D
  - Upstream (network side) & Downstream (user side)
  - ANI
  - DTMF tones
- MFC R2 (E1)
  - Upstream (network side) & Downstream (user side)
  - ANI
  - DTMF tones

CCS Signaling (ISDN)

- 4ESS, 5ESS, DMS, NI2, ETSI, INSNET-1500, QSIG
- Upstream (network side) & Downstream (user side)

Analog Tenor Supported Standards

- Loop-Start ("Plain" or with 2 variations)
  - Loop Start Forward Disconnect
    - Disconnect Supervision*
  - FXS & FXO

Section Key Points

- Management through CAM and/or CLI.
- SelectNet™ (or TASQ™) virtually guarantees that each call over IP will be routed successfully and with quality sound. If the IP network degrades, the call will be routed to the PSTN for back up, during the call, without the knowledge of either end-user.
- Online/Offline mode, sometimes referred to Bypass mode, provides backup protection in that if the Tenor loses power, your connections between the PBX and PSTN will be active.
  - This feature can also be implemented through the Tenor software for ease of testing.
- Dynamic Call Routing allows for calls to be routed in virtually any direction (PBX, PSTN and/or IP), as necessary based on the number.
  - The Tenor performs number matching based on the International number format (country code + city/area code + number).
  - If the Tenor cannot find a match for the number on the IP network, the call will be routed to the PSTN.
  - If, for any reason, the call does not get connected at the destination Tenor, the origination Tenor will try to re-route the call to other available resources, such as the PSTN.
  - Supports both public and private dial plans.
- Bypass Directory Numbers (BDN). Numbers and or patterns that you specify, that will never go over IP.
  - The * can be used as a wild card to indicate any digit, any number of placeholders.
- Local Directory Numbers (LDN). Numbers and or patterns that you specify, to say what calls/numbers are routed over IP to this unit and sent to the PBX. The PBX station range.
  - The * can be used as a wild card to indicate any digit, any number of placeholders.
- Hop-Off or Leaky Area Mapping Numbers (LAM). Numbers that are configured at a destination Tenor to allow VoIP calls to "hop-off" to the PSTN and thereby saving long distance and/or international toll charges.
  - The Hop-Off pattern is the number/digit pattern that is used to match those calls that will be allowed to "hop-off" to the PSTN using international number format.
  - Numbers that match to a Hop-Off pattern will have their digits that match to the pattern deleted off of the front of the number.
  - Replacements are the digit(s) that are to be added to the front of the number, after the Hop-Off pattern is deleted, to satisfy the routing requirements of the PSTN.
- Modem Bypass. Analog modem-analog-modem calls over IP are not reliable due to the compression algorithms.
  - Tenor has ability to detect these calls, disconnect them and automatically store the dialed number in the bypass table so that the next time the call is attempted, it will be sent directly to the PSTN.
- Border Element is the Master Gatekeeper for a Tenor Network.
  - All Gatekeepers report their routing information (IP address, LDN's and LAM's) to the BE. The BE builds a master routing table of all the GK information and sends this table back out to all of the GKs.
  - Gateways register to GKs and make routing requests to the GKs.

Section Key Points (continued)

- PacketSaver™ allows you to conserve IP bandwidth by multiplexing several voice streams that are headed to the same Tenor destination, in to a single IP Packet.
  - Save bandwidth by reducing the number of IP packets sent their associated headers.
- Call Detail Reporting (CDR) used to construct billing records.
- Software upgrades done using standard FTP.
- Multiple signaling supported.
  - Digital supports both T1 and E1 (CAS and ISDN).
  - Analog supports FXS & FXO (loop-start).
    - Tenor does provide Answer Supervision by way of Loop Start Reverse Battery Signaling, which requires a paid software upgrade and "Revision B" Analog Tenor Hardware.
- Different product levels to support every customers needs.
  - Digital comes in 8, 16, 24 and 30 port models.
  - Analog comes in 4 and 8 port models.
- Both Digital and Analog models come in "Mini Gatekeeper" and "Standard Gatekeeper" versions, all include SelectNet™.
  - PacketSaver™ is available for an additional cost.