NetVanta IP Telephony

Certification course - ATSP/IP Telephony

Course Guide

Revision 7/2009

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Huntsville, AL 35814-4000
Phone: (256) 963-8000

Customer Service, Product Support Information, and Training

ADTRAN will replace or repair this product within five years from the date of shipment if the product does not meet its published specification, or if it fails while in service.

A return material authorization (RMA) is required prior to returning equipment to ADTRAN. For service, RMA requests, training, or more information, see the toll-free contact numbers given below.

Presales Inquiries and Applications Support

Please contact your local distributor, ADTRAN Applications Engineering, or ADTRAN Sales:

Applications Engineering	(800) 615-1176
Sales	(800) 827-0807

Post-Sale Support

Please contact your local distributor first. If your local distributor cannot help, please contact ADTRAN Technical Support and have the unit serial number available.

Technical Support	(888) 4ADTRAN

The Custom Extended Services (ACES) program offers multiple types and levels of service plans which allow you to choose the kind of assistance you need. For questions, call the ACES Help Desk.

ACES Help Desk	(888) 874-2237

Training

The Enterprise Network (EN) Technical Training offers training on our most popular products. These courses include overviews on product features and functions while covering applications of ADTRAN's product lines. ADTRAN provides a variety of training options, including customized training and courses taught at our facilities or at your site. For more information about training, please contact your Territory Manager or the Enterprise Training Coordinator.

Training Phone	(800) 615-1176, ext. 7500
Training Fax	(256) 963-6700
Training Email	training@adtran.com
Web Site	www.adtran.com/training

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July 2009

Revision date 7/6/09

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- Cover Key NetVanta IP Telephony Applications

ADTRAN, Inc.

ADTRAN, Inc.

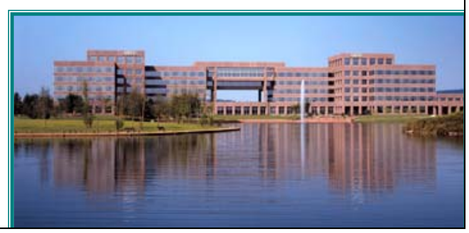
- Global provider of networking and communications equipment
- Widely deployed by carriers, distributed enterprises, and Small-and-Medium-sized businesses (SMBs)
- Headquartered in Huntsville, Alabama
- Product Distribution
  - Value Added Resellers
  - Distributors

ADTRAN, Inc. is a leading global supplier of networking and communications equipment with an innovative portfolio of more than 1,700 solutions for use in the last mile of today's telecommunications networks. Widely deployed by carriers, distributed enterprises and Small- to Medium-sized Businesses (SMB), ADTRAN solutions enable voice, data, video, and Internet communications across copper, fiber and wireless network infrastructures. Our solutions are currently in use by every major U.S. service provider and many global ones, as well as by thousands of public, private and governmental organizations worldwide.

ADTRAN Support

ADTRAN Support

- Free First-Class Telephone Support
  - Presales
    - Applications Engineering (800) 615-1176
  - Post-sales
    - Technical Support (888) 4ADTRAN
- Industry Leading 5 or 10 year Warranty
- No Cost Software Updates
- ACES Installation and Maintenance Services
  - Guaranteed response time
  - Onsite or phone installation
  - Guaranteed replacement plans



Every product is backed by an industry-leading five-year warranty, best-in-class telephone technical support from our team of degreed engineers, and is eligible for free firmware upgrades.

The ADTRAN product warranty includes a return-to-factory repair and replacement program and free technical phone support. Technical support engineers are accessible for both pre- and post-sales support. ADTRAN Custom Extended Services (ACES) is also available for an extended guarantee and rapid response time. Priority access to technical and installation support is guaranteed with a 30-minute call back and on-site product replacement in as few as four hours, depending on the service plan selected.

NetVanta Series

NetVanta Series

- ADTRAN IP Telephony Solutions built on successful router/switch platform
  - Industry-leading LAN/WAN infrastructure
  - Feature-rich Router and PoE Switch
  - Full suite of QoS for delay sensitive VOIP traffic
  - Built-in security
  - Excellent Service/Support

ADTRAN, the #1 Value!

#2 in router marketshare

NetVanta Series Overview

NetVanta Series Overview

- **Ethernet Switch**
  - Fast Ethernet and Gigabit Switches
  - Managed
  - Auto-Rate, Auto Duplex
  - Auto-MDI/MDI-X
  - 802.1D Spanning Tree
  - VLAN
  - 802.1p CoS
  - 802.3af Power over Ethernet
  - 15.4 watts for each of the 24 ports
- **VPN, Firewall**
  - Stateful Inspection Firewall
  - NAT (1:1), NAPT (Many:1)
  - DoS Protection
  - Access Control Lists
  - IPSec
  - DES/3DES/AES Encryption
- **IP Router**
  - 56/64K, T1, Multi-T1, T3
  - RIP V1/V2, OSPF,BGP
  - PPP, PPPoE, Frame Relay, HDLC
  - MLPPP/MLFR
  - DHCP Client/Server
  - Class-based Weighted Fair Queuing, Low Latency Queuing
  - Diffserv aware/mark

ADTRAN IP Telephony Solutions

ADTRAN IP Telephony Solutions

- IP Communications Platform
  - NetVanta 7100
- IP PBX
  - NetVanta 7060
- IP Business Gateways
  - NetVanta 6355
  - Total Access 900 Series
- IP Phones
  - ADTRAN 700 Series
  - ADTRAN/Polycom IP Phones

IP Communication Platform

The NetVanta 7100 represents a break through in next-generation communication systems. This unique Office in a Box contains everything businesses need to deploy a converged IP voice and data network for small- to medium-sized offices with up to 100 stations, including a full-function IP PBX for voice. It includes an integrated24-port Power over Ethernet (PoE) switch-router for data, a stateful inspection firewall for security, Virtual Private Network (VPN) for secure Internet tunneling, and a DSU/CSU for network termination. The only other requirements for deploying your VoIP network are connections from the service provider and cables to the desktop.

IP PBX

The NetVanta 7060 simplifies the implementation of VoIP for businesses that already have an IP data network established. The NetVanta 7060 complements the existing network, quickly enabling VoIP by providing IP PBX functionality which includes SIP-based telephony features, voice mail (3000 messages, eight ports), multilevel auto attendant, caller ID name/number and all the other features a business needs for a complete VoIP network.

IP Business Gateways

ADTRAN IP Business Gateways are purpose built devices that include a variety of advanced routing, security, and voice functionality for Hosted IP applications.

Ideally suited for SMB and distributed enterprise networks, this category of products includes the Total Access 900 and 900e Series of dual and multi-T1 platforms that include analog and SIP gateway, robust IP router, firewall and VPN functionality.

The NetVanta 6355 platform provides a unique, all-in-one solution for Hosted VoIP. This product combines all of the IP voice functionality, SIP gateway, router, firewall/VPN features of the Total Access 900/900e Series with a managed 24-port PoE switch into a single 1U chassis.

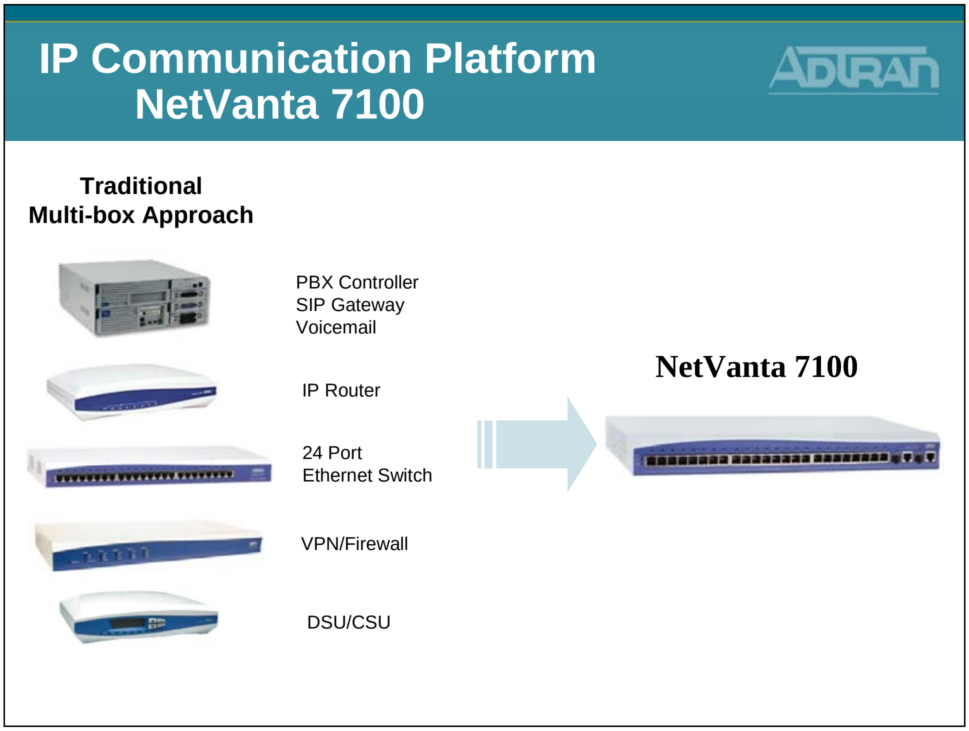
ADTRAN IP Phones

ADTRAN offers SIP phones designed to address the growing converged VoIP and IP telephony marketplace. The new ADTRANIP 700 Series of phones includes the IP 706, a six-line version and the IP 712, a 12-line version and both phones are designed with a large backlit display. ADTRAN IP phones offer an affordable, feature-rich VoIP solution that delivers unsurpassed quality and performance.

ADTRAN-Polycom IP Phones

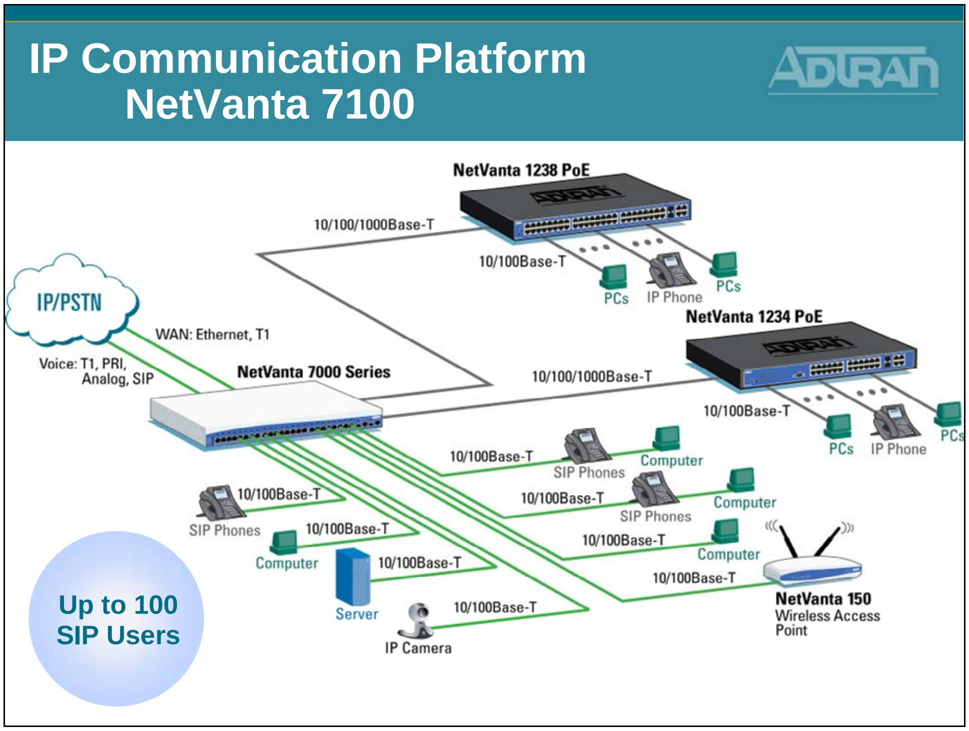
Working together, ADTRAN and Polycom have partnered to deliver a full line of IP telephones. The phones integrate seamlessly with ADTRAN’s NetVanta and Total Access® 900 Series VoIP products. The ADTRAN-enabled IP stations include the IP 430 (two-line), IP 650 (six-lines) and IP 650 Expansion Module. The combination of ADTRAN’s award-winning VoIP equipment with a broad line of ADTRAN-Polycom IP phones and accessories offers a cost-effective, simplified VoIP internetworking solution.

IP Communication Platform – NetVanta 7100



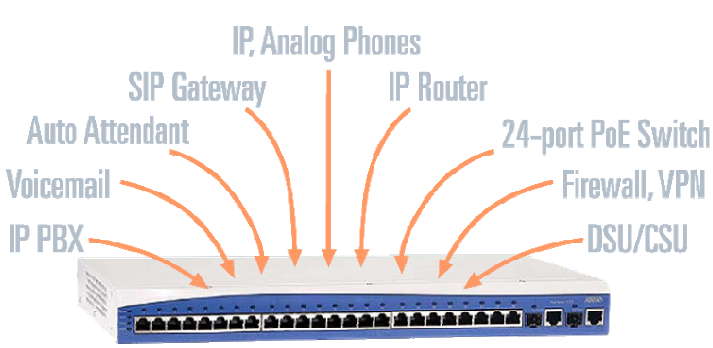
Multiple Functions in a Single Box

The NetVanta 7000 Series offers all the business-class functionality a Small-to-Medium sized Business (SMB) requires, at an affordable price. The all-in-one platform consolidates multiple functions in a single, easy-to-manage platform. Both the NetVanta 7100 and 7060 include multiple levels of auto-attendant function and a system scheduler. This allows the customization of auto-attendant functions based on the time or day settings programmed. The NetVanta 7000 Series also works in key system mode and PBX mode for increased flexibility and ease of use.

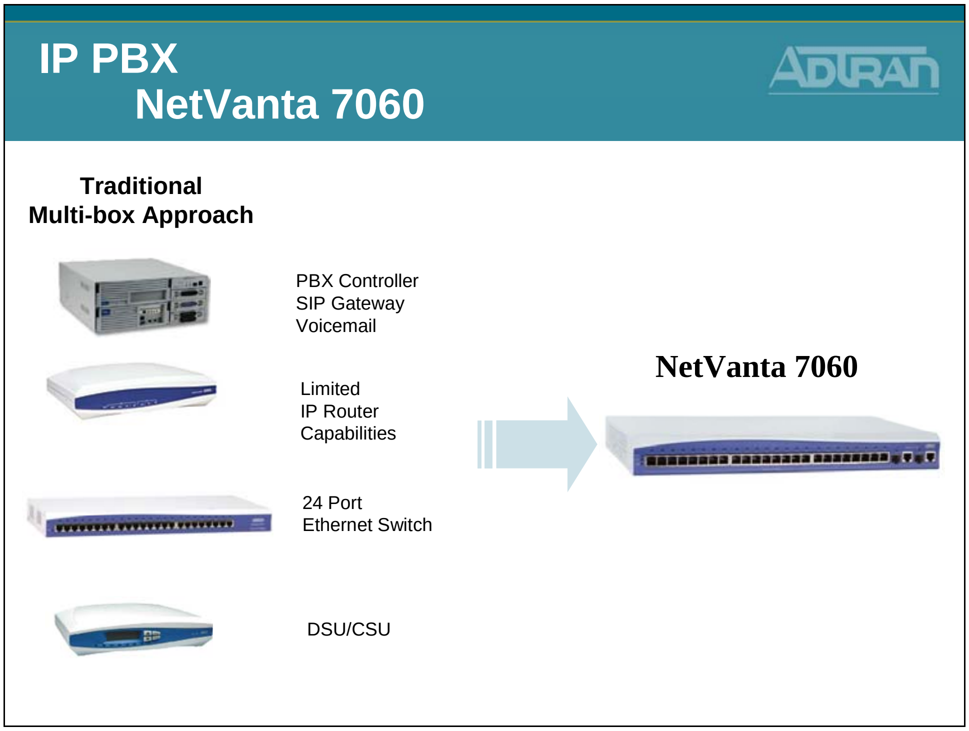


Office-in-a-Box

The NetVanta 7100 is a complete voice and data networking solution for business locations of up to 100 SIP Users. This innovative platform includes an IP PBX, voice mail, multilevel auto attendant, full-featured IP router, firewall, Virtual Private Network (VPN), 24-port Power over Ethernet (PoE) (802.3af) Fast Ethernet switch with Gigabit uplinks, and two expansion slots for Network Interface Modules (NIMs) and Voice Interface Modules (VIMs). The NetVanta 7100 IP PBX functionality includes SIP-based telephony features such as voice mail (12 hours, eight ports), multilevel auto attendant (eight ports), caller ID name/number, Shared Line Appearances (SLA), Busy Lamp Field (BLF), Class of Service (CoS), trunk groups, music on hold, overhead paging and a number of call options including call coverage lists, forwarding of calls to a cell phone and email notification of voice mail.



IP PBX – NetVanta 7060

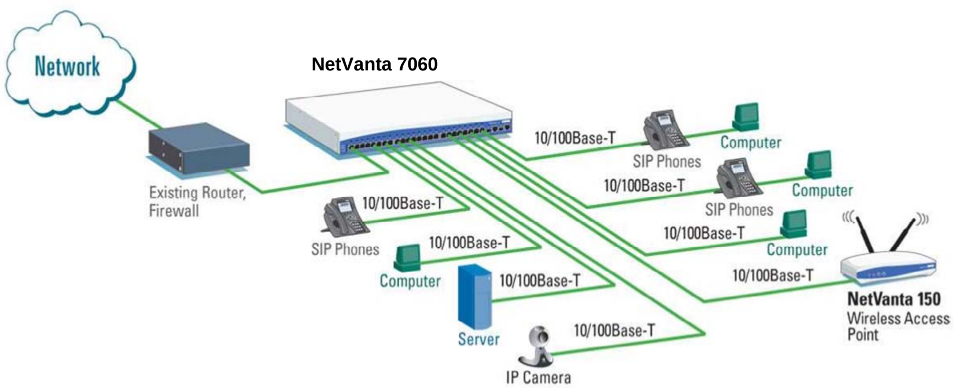


NetVanta 7060

The NetVanta 7060 is an IP telephony solution ideal for business locations that already have an IP data network established with routing and VPN functionality. The NetVanta 7060 is an unbundled solution providing IP PBX functionality which includes SIP-based telephony features, voice mail (3000 messages, eight ports), multilevel auto attendant, caller ID name/number, COS, trunk groups, music on hold, overhead paging, and a number of call options including call coverage lists, forwarding of calls to a cell phone, and email notification of voice mail.

IP PBX
NetVanta 7060

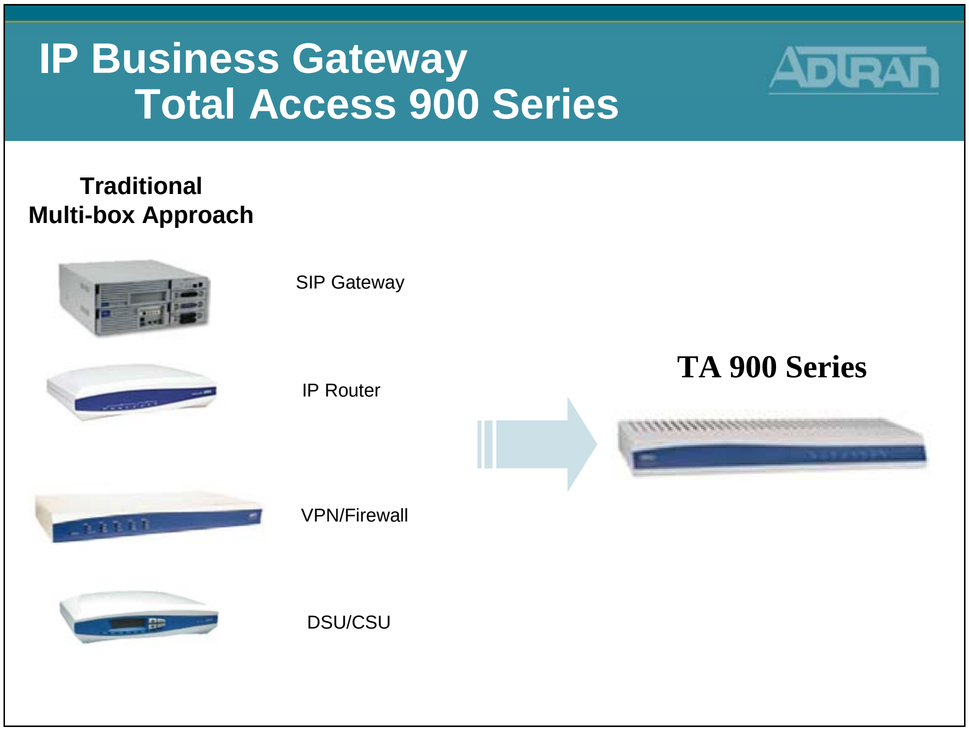
- Interoperates with external data equipment



ADTRAN's new NetVanta® 7060 IP PBX is designed to work in a multi-vendor environment so businesses that already have modern robust data networking equipment can add the NetVanta 7060 as their phone system. The NetVanta 7060 includes the phone system capabilities businesses need and can interoperate with external routers, firewall and Virtual Private Networking (VPN) devices.

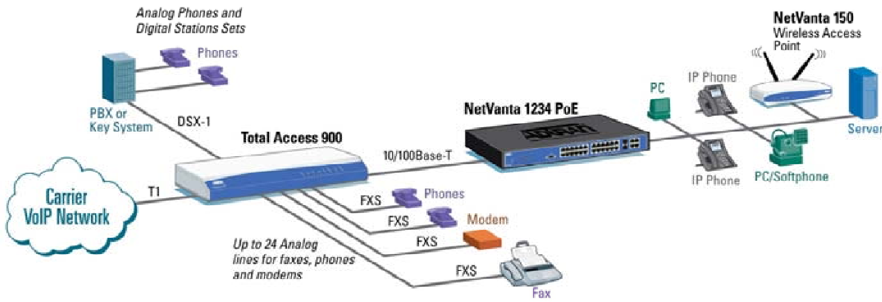
- Uses existing IP data equipment
- Provides PBX phone system, including voice mail and auto attendant
- Provides integrated 24 port Power over Ethernet (PoE) switch

IP Business Gateway – Total Access 900 Series



The Total Access 900 Series of IP Business Gateways combine the functionality of ADTRAN's industry-leading integrated access devices with a SIP and analog gateway to provide Incumbent Local Exchange Carriers (ILECs), Competitive Local Exchange Carriers (CLECs), and Internet Service Providers (ISPs) a cost-effective IP network strategy for VoIP deployment, with support for legacy equipment. The Total Access 900 and 900e Series allow carriers to deliver SIP trunks, hosted PBX, and other voice and data services such as Dedicated Internet Access (DIA) to small and medium businesses, quickly and cost-effectively.

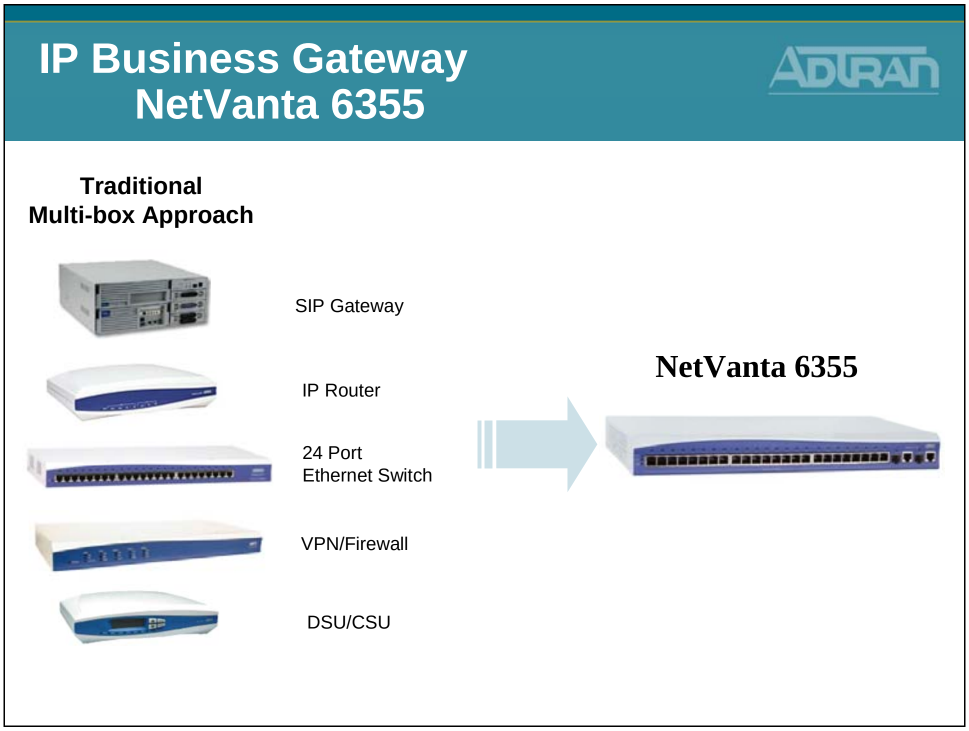
IP Business Gateway
Total Access 900 Series



Total Access 900 Series Features and Benefits

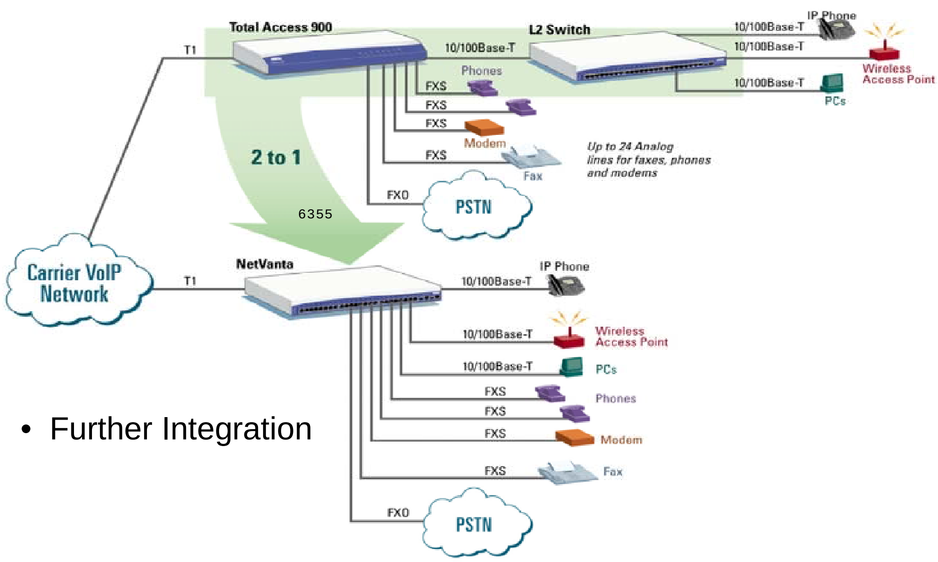
- Carrier-class, cost-effective multi-T1/dual Ethernet IP Business Gateway for integrated services such as VoIP
- Supports up to 24 analog interfaces for legacy equipment
- Integral DSX-1 PRI/CAS for PBX connectivity
- Transparent proxy with survivability for network outages
- Voice Quality Monitoring (VQM) for enhanced Quality of Service (QoS)
- Compatible with industry leading softswitches and call agents
- Integral full-featured IP router for data support and Internet access
- Stateful inspection firewall for network security
- Quality of Service (QoS) for delay sensitive traffic like VoIP
- Command Line Interface (CLI) mimics industry de facto standard
- Feature-rich ADTRAN Operating System (AOS)
- Industry-leading 10-year North American warranty
- Four T1 WAN interfaces/two Ethernet interfaces/24 FXS analog interfaces
- Visit www.adtran.com for Alternate Configurations with part numbers for chassis with some number of FXS and some number of FXO interfaces for mixed mode analog environments

IP Business Gateway – NetVanta 6355



The NetVanta 6355 IP Business Gateway is a unique, all-in-one solution for Hosted VoIP PBX services, Internet access, and business connectivity. This powerful platform combines the voice functionality of ADTRAN’s industry leading Total Access 900e Multiservice Access Device and the widely deployed NetVanta Power over Ethernet (PoE) Switch-Router into a compact 1U chassis. This all-in-one product includes a robust SIP-Gateway, a full-featured IP router, stateful inspection firewall, VPN, 24-port powered (802.3af) Fast Ethernet switch with Gigabit uplinks, and two expansion slots for Network and Voice Interface Modules (NIM/VIMs).

IP Business Gateway NetVanta 6355



- Further Integration

NetVanta 6355 Features and Benefits

- All-in-one Hosted IP voice and data solution
- Integral SIP gateway, router, PoE switch, and security
- Full-featured IP router supporting up to three T1s for data and Internet access
- Managed, 24-port PoE (802.3af) switch
- Stateful inspection firewall for network security
- IPSec VPN for secure corporate connectivity across the Internet
- Compatible with industry-leading softswitches and call agents
- Up to 10 analog POTS interfaces with remote survivability
- Supports IP amd analog phones/phone systems; fax machines, modems, and Wireless Access Points (WAPs)
- Dynamic bandwidth allocation enables more efficient utilization
- Standardized G.729a voice compression requires less bandwidth per voice call
- Industry-leading warranty

Visit www.adtran.com for additional information on the NetVanta 6355.

ADTRAN IP 700 Series Phones

ADTRAN IP 700 Series Phones

ADTRAN

- ADTRAN IP Phones:
  - IP 706: 6 lines
  - IP 712: 12 lines
- Supports multiple SIP registrations
- Busy Lamp Field and Shared Line Appearance Support
- High Quality Full Duplex Speaker Phone
- Dual 10/100 Switched Ethernet Ports
- Large Backlit Display
- 802.3af Power over Ethernet
- Adjustable base stand
- Wall mountable
- Headset jack with Electronic Hook Switch Detection

ADTRAN® offers SIP-enabled phones designed to address the growing converged Voice over IP (VoIP) and IP telephony marketplace for small businesses and multi-site enterprises. The ADTRAN IP 706, a six-button programmable phone and the IP 712, a 12-button expanded version offer an affordable and standards-based solution that delivers unsurpassed quality and performance.

Ease of Use, Style and Productivity

The ADTRAN IP 700 Series of telephones delivers an attractive and functional business-class telephone for today's businesses, all at affordable and cost-effective prices. In addition to the appealing desktop style for business offices of any type, users will appreciate the large, backlit, easy-to-read LCD screens and well-designed layout of frequently used buttons and functions. On screen menus and navigation keys work together in an intuitive, user-friendly manner. ADTRAN's IP phones are designed to provide enhanced efficiency and convenience for the user.

Enhanced Functionality

ADTRAN IP phones are available in either six- or 12-line versions, supporting multiple call functions. Dedicated keys are available for the most common user functions with additional programmable soft keys. On-screen menus enable users to quickly change directory information and phone settings, as well as view a history of internal/external and missed calls, and program distinctive ring tones for specific calls. The phones include an adjustable desk stand or can be wall mounted and feature high-quality, full-duplex speakers engineered for clear, hands-free communication. An integrated headset jack with electronic hook-switch eliminates the need for mechanical handset lifter. The overall enhanced functionality for the price makes ADTRAN IP phones among the most cost-efficient business-class IP phones.

Quick, Easy Set-up

The ADTRAN 700 Series features an intuitive, Graphical User Interface (GUI) for easy set-up and installation. The phones can be directly powered from the NetVanta 7000 Series or a Power over Ethernet (PoE) switch, providing inline power and eliminating the need for a separate power supply. ADTRAN phones can be locally powered, allowing for multiple options for worry-free installation and ease of use. The phones also have two Ethernet ports to connect to a PC for converged voice and data across a single wiring infrastructure.

IP 700 Series Product Features

- Fully interoperable with NetVanta 7000 Series
- Six or 12 programmable buttons
- Large backlit display
- Message waiting indicator
- Integrated headset jack
- Web-based management
- Distinctive ring tones
- Multiple call appearances
- Three-way conferencing
- Busy Lamp Field (BLF)
- Shared Line Appearance (SLA)
- Hands-free auto-answer intercom
- High-quality full-duplex speaker phone

Interoperability with Polycom IP Phones

To complement the new ADTRAN 700 Series of IP phones, Polycom IP phones offer additional VoIP solutions for an extended range of business applications.

Some of the Supported Polycom Phones Include:

- IP 601 Three-line IP Phone
- IP 650 Six-line High Definition IP Phone
- IP Expansion Module Attendant Console
- IP 6000 Conference Phone

ADTRAN/Phone Features

ADTRAN/Phone Features

- Call Drop
- Call Forward (All, Busy, No Answer)
- Call Forward to Outside Line (Cell Phone)
- Call Hold
- Caller ID Name/Number
- Call Logs
- Call Park
- Call Park Retrieve
- Call Transfer
- Call Waiting
- Conferencing (3-person)
- Do Not Disturb
- Handsfree Auto Answer Intercom
- Headset Jack
- Message Waiting Light
- Missed Call Indicator
- Multiple Call Appearances
- Music on Hold
- Mute
- Overhead Paging
- Redial
- Speakerphone
- Volume Control

ADTRAN Analog Door Phone

ADTRAN Analog Door Phone

- Single-gang Wall box
- Analog Speakerphone
- Line powered
- Weather-resistant
- Stainless steel finish
- System supports multiple door phones
- 1 Year Warranty

The ADTRAN® ADP-40 is an analog speakerphone primarily used for entry applications such as door or gate communication, business delivery entrances, and residential, commercial, or industrial door security. The ADP-40 complements the NetVanta 7000 line by providing a rugged communication endpoint to any entry way. Once a person's identity is announced through the door phone, a phone user enters a special code which allows the door to open.

The ADP-40 offers a weather-resistant design that is easy to install in new or existing construction. It fits flush in any single gang electrical box and receives power directly from the telephone line. With weather and vandal resistant features that include an 18-guage stainless steel faceplate, Mylar speaker, hex drive mounting screws, a stainless steel speaker screen, and gaskets for the faceplate, microphone, and speaker, the ADP-40 can be installed inside or outside.

This full featured entry phone supports auto answer to enable remote communications of the area immediately around the speakerphone, intelligent call progress detection for automatic hang-up when a call is completed, and microphone and speaker volume controls.

The ADP-40 conveniently connects directly to one of the analog station (FXS) ports on the NetVanta 7100. The user account for the station port can be configured as a hotline phone to allow the ADP-40 to call a specific extension or a ring group when the Call button is pressed. Once off hook, a phone user dials a code that controls the relay latch to open the door.

ADTRAN IP Softphone

ADTRAN IP Softphone

- PC SoftPhone
- SIP-based
- Requires headset / microphone
- Familiar functions
  - VoiceMail Indicator
  - Transfer, Conf, Hold, etc.

The ADTRAN IP SoftPhone is an intuitive software application designed to enable Voice over Internet Protocol (VoIP) communication from your laptop or desktop PC and works seamlessly with ADTRAN's IP telephony product lines.

The IP SoftPhone is easy to use and offers a built-in audio tuning wizard that helps simplify setup. Any audio devices available to the host PC such as USB headsets or PC speakers can be used with the SoftPhone. The "Speaker" button offers single-button selection to switch between headset or speaker phone devices.

The IP SoftPhone offers six call appearances with conferencing capability and other familiar features like transfer, hold, do-not disturb, and a message waiting indicator. These features offer mobile employees many of the same convenient capabilities they enjoy when in the office.

The ADTRAN IP SoftPhone improves productivity by enabling users to have quick access to their address book and call logs to identify recently received calls, missed calls, and dialed calls. The ADTRAN IP SoftPhone can be configured using the same extension as the user's office phone or as a completely separate extension.

By using Virtual Private Networks (VPNs), remote and mobile workers can use the ADTRAN IP SoftPhone with any Internet connection and be confident that the voice and data traffic is secure and private. VPNs provide encryption and ensure the security of the data and voice traffic between the corporate network and a remote office Internet connection or wireless hotspot or hotel broadband connection.

PC-based Phone Manager

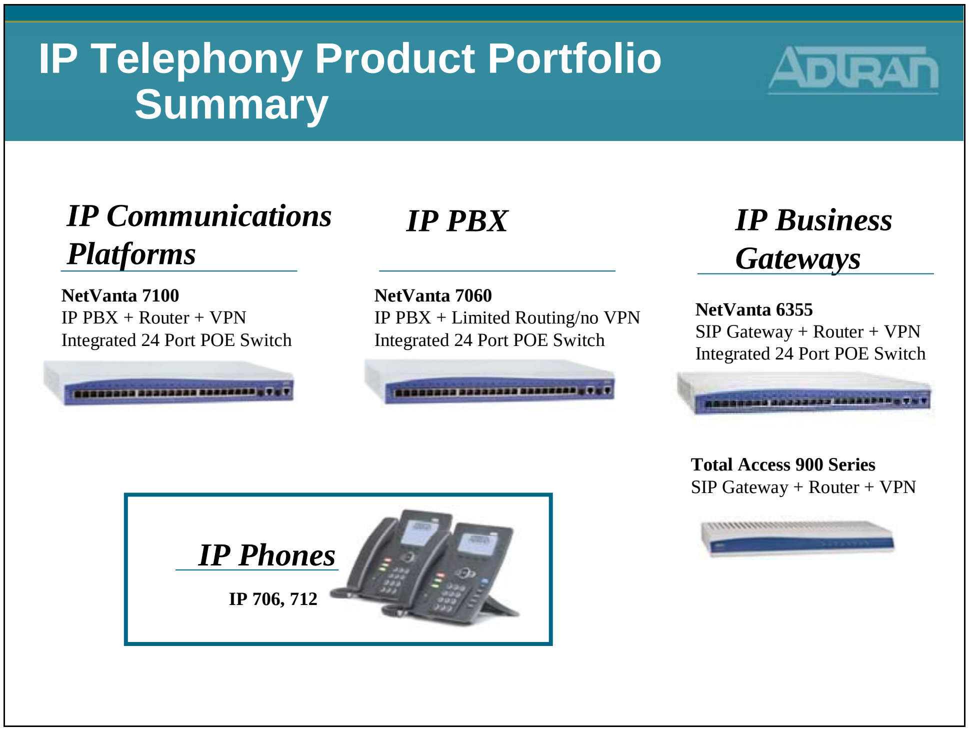
PC-based Phone Manager

ADTRAN

- Web-based utility
- User can customize phone settings
  - System Directory
  - Speed Dial
  - Click to dial
  - Call Coverage
  - Call Forwarding

The Personal Phone Manager is an easy-to-use Web-based utility browser provided by NetVanta 7000 Series platforms that is designed so each user can customize phone settings. These settings include speed dial, call coverage, and view directory and include the click-to-dial feature for quick-and-easy phone number dialing.

IP Telephony Product Portfolio - Summary

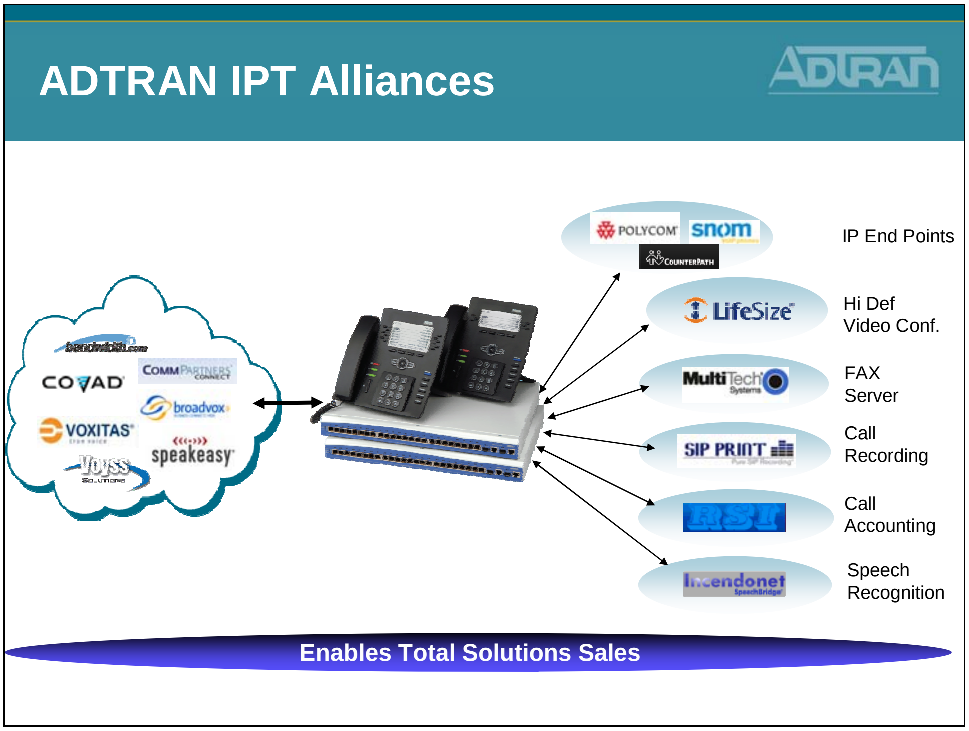


Each IP telephony solution simplifies the migration to VoIP and resolves complicated network assessments and equipment interoperability issues. Our products offer significantly lower initial costs and ongoing maintenance expenses, when compared to traditional systems. Cost savings are achieved by consolidating voice and data networks, reducing monthly service charges and eliminating expensive add-on phone and voicemail licenses. From our integrated VoIP and data communication platforms to our IP PBX Systems and IP Phones, our IP telephony solutions deliver years of reliable service.

ADTRAN IP telephony Solutions:

- Are Ideal for small to medium businesses
- Make your communication network flexible and affordable
- Provide feature-rich, standards-based solutions that scale
- Resolve complicated network assessments and interoperability issues
- Reduce TCO, significantly lowering initial and ongoing costs

ADTRAN IPT Alliances

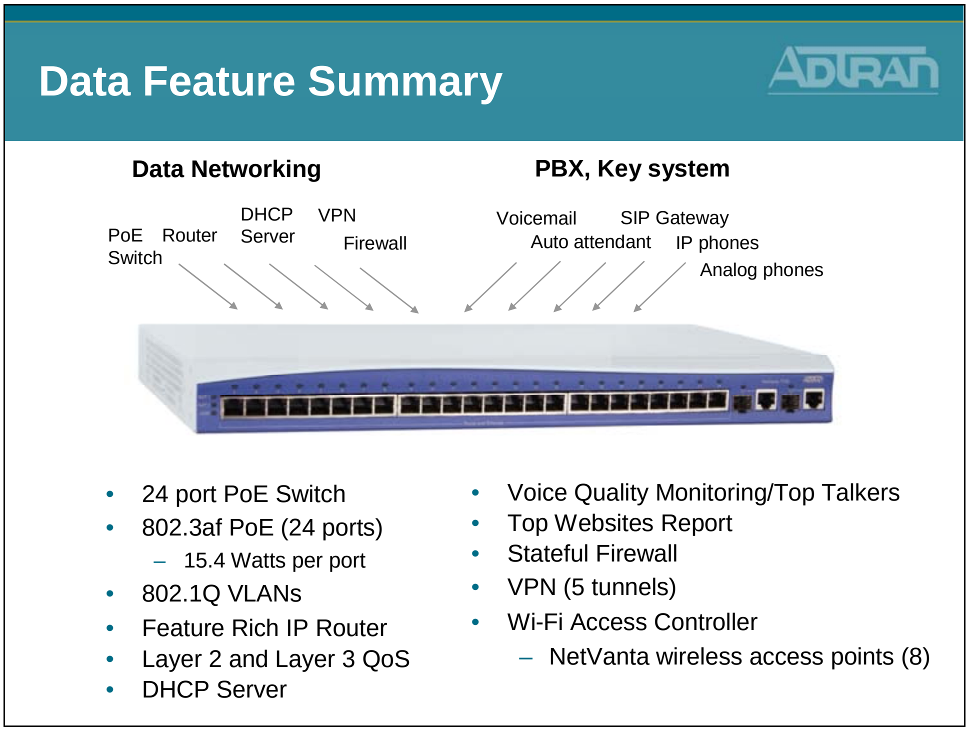


The ADTRAN Alliance Program expands the reach of IP communications solutions to small- and medium-sized businesses. The ADTRAN Alliance Program is collaboration with best-in-breed technology and service providers that complement the NetVanta 7000 converged IP PBX Series and enable ADTRAN solutions providers to deliver world class integrated network solutions. Visit www.adtran.com/alliance for additional information.

SIP Trunking Service Provider Alliances
SIP Trunking Service Providers offer IP telephony service offerings that are certified to be fully interoperable with the NetVanta 7000 Series. The combination of the NetVanta 7000 Series with these services offers SMB customers proven ways to cost-effectively transition to converged voice and data networking.

IP Telephony Technology Partners
Innovative solutions that have been strategically chosen to address specific applications in conjunction with the NetVanta 7000 Series. These best-of-breed partners include Polycom, CounterPath, SNOM, Incendonet, LifeSize, SIP Print, MultiTech, and RSI. The combination of the NetVanta 7000 Series and the complementary partner solutions now enable service and solutions providers to offer a broader, more comprehensive solution with the added benefit of proven interoperability to meet the growing SMB and Enterprise market needs for IP Telephony solutions.

Data Feature Summary



The NetVanta® 7100 is an integrated IP data networking and telephony solution designed to simplify Voice over IP (VoIP) and IP telephony for business locations of up to 100 employees. This one-box solution combines multiple data and voice functions into a single, affordable platform. The ADTRAN® NetVanta 7100 IP Communication Platform includes a router, 24 port Power over Ethernet (PoE) switch, firewall, Virtual Private Network (VPN), Wireless LAN controller, SIP Gateway, and business-class phone system with integrated voice mail and automated attendant.

Voice Feature Summary

Voice Feature Summary

ADTRAN

- PBX and key system modes
- No phone or voicemail licenses
- Supports up to 100 SIP stations,
- Supports up to 10 Analog stations
- Supports SIP, T1/PRI and Analog Trunks
- Supports ADTRAN IP 706/712 and certified Polycom phones
- SIP/PSTN Gateway
- Zone Paging
- Internal voice mail (3000 messages, 8 ports)
- Multilevel auto attendant (8 ports)
- Shared Line Appearance (SLA)
- Shared Call Appearance (SCA)
- Dial by name directory
- System Scheduler
- Voice Quality Monitoring (VQM) and Mean Opinion Score (MOS)
- Music-on-hold input, paging output, door relay

The NetVanta 7100 is a complete voice and data networking solution for business locations of up to 100 stations. This innovative platform includes an IP PBX, voice mail, multilevel auto attendant, full-featured IP router, firewall, Virtual Private Network (VPN), 24-port Power over Ethernet (PoE) (802.3af) Fast Ethernet switch with Gigabit uplinks, and two expansion slots for Network Interface Modules (NIMs) and Voice Interface Modules (VIMs).

The NetVanta 7100 IP PBX functionality includes SIP-based telephony features such as voice mail (store up to 3000 messages, eight ports), multilevel auto attendant (eight ports), caller ID name/number, Shared Line Appearances (SLA), Busy Lamp Field (BLF), Class of Service (CoS), trunk groups, music on hold, overhead paging and a number of call options including call coverage lists, forwarding of calls to a cell phone and email notification of voice mail.

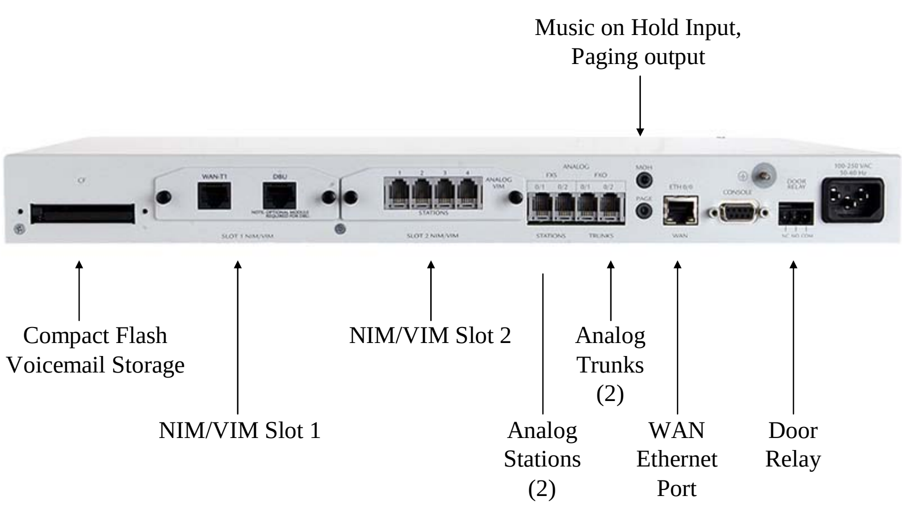
NetVanta 7000 Series – Front Panel

NetVanta 7000 Series Front Panel

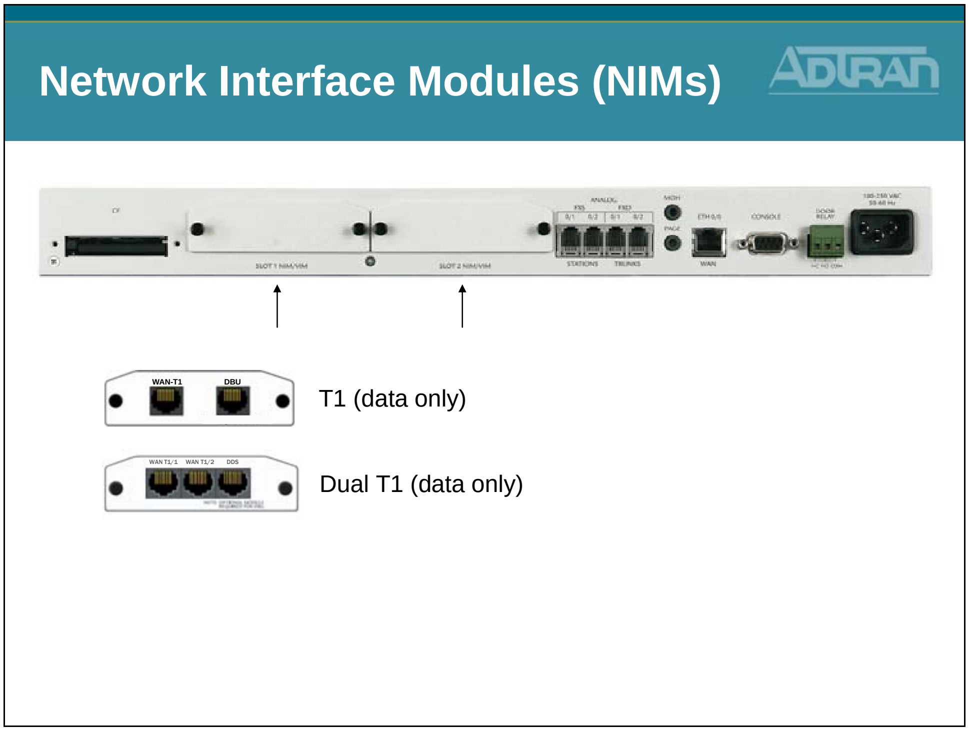
- 24 10/100 PoE ports
  - Ethernet 0/1 - 0/24
  - 802.3af (15.4 watts per port)
  - Auto-Rate /Auto-Duplex / Auto-MDI/MDI-X
- 2 10/100/1000 ports
  - Gigabit 0/1 - 0/2
  - Copper or Fiber (SFP)

NetVanta 7000 Series - Rear Panel

NetVanta 7000 Series Rear Panel



Network Interface Modules (NIMs)

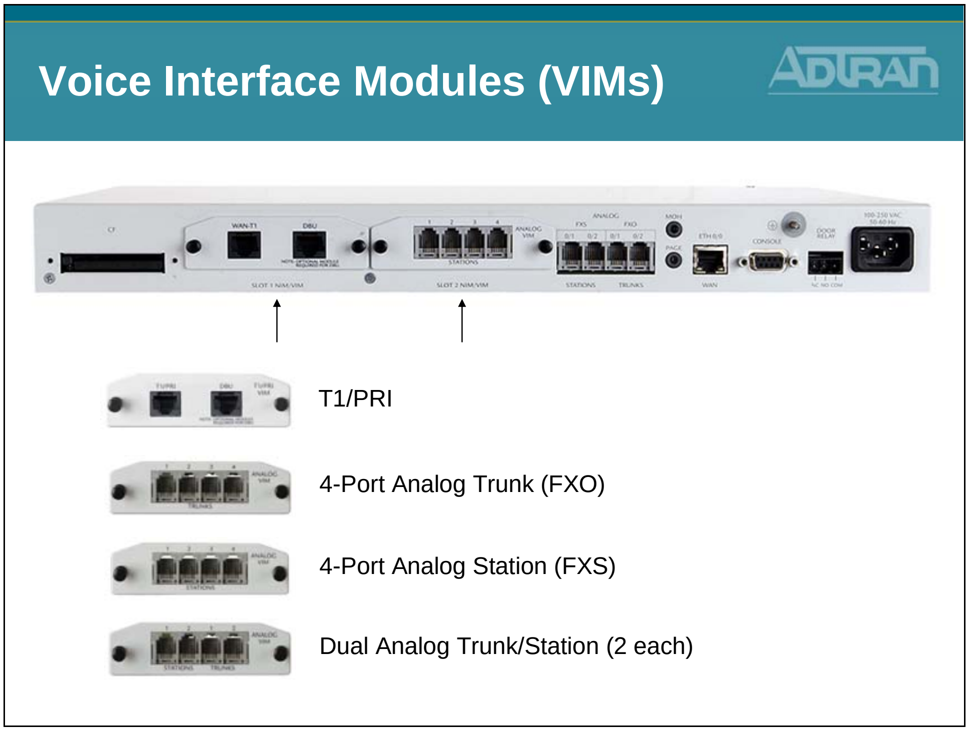


T1/FT1 NIM
Provides a network interface for a fractional or full T1 for NetVanta 1000, 3000, 4000, and 7000 series products

Dual T1 NIM

Terminates two full or fractional T1s or two T1s aggregated together / Integral DSU/CSU

Voice Interface Modules (VIMs)



NetVanta T1/PRI Voice Interface Module
Provides one RBS T1 or one PRI (5E, DMS100, or National) interface for termination of TDM voice trunks

NetVanta Analog 4-Port Trunk Voice Interface Module
Provides four analog RJ-11 trunk (FXO) ports for termination of PSTN circuits / Supports loop-start and ground-start and captures Caller ID name/number using FSK / Part 68 compliant

NetVanta Analog 4-Port Station Voice Interface Module
Provides four analog RJ-11 station (FXS) ports for connection to analog devices such as POTS phones, FAX machines, and/or modems / Delivers Caller ID name/number using FSK / Loop-start/DTMF / Includes ring generator

NetVanta Analog 2-Trunk/2-Station Voice Interface Module
Provides two analog RJ-11 trunk (FXO) ports for termination of PSTN circuits / Supports loop-start and ground-start and captures Caller ID name/number using FSK / Part 68 compliant / Provides two analog RJ-11 station (FXS) ports for connection to analog devices such as POTS phones, FAX machines, and/or modems / Delivers Caller ID name/number using FSK / Loop-start/DTMF / Includes ring generator

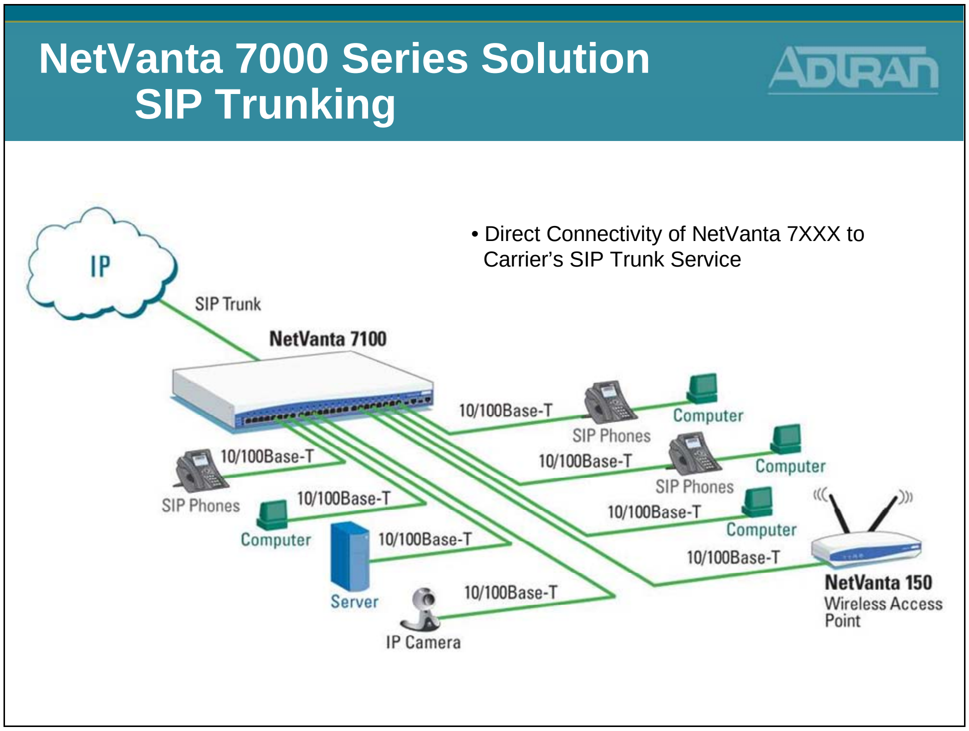
NetVanta 7000 Series - Port Configurations

NetVanta 7000 Series Port Configurations

- 4 Analog Trunk, 4 Analog Station, 24 PoE
- 6 Analog Trunk, 6 Analog Station, 24 PoE
- 8 Analog Trunk, 4 Analog Station, 24 PoE
- 10 Analog Trunk, 2 Analog Station, 24 PoE
- 1 T1/PRI, 6 Analog Station, 24 PoE

The NetVanta 7100 chassis provides two analog trunk and station interfaces and two expansion slots. For additional trunk and station connectivity, the NetVanta 7100 offers several Voice Interface Modules (VIMs). These include a four-port analog (FXO) trunk module, TI/PRI trunk module which supports voice or integrated voice and data, and a four-port analog (FXS) station module. A combination module which provides two analog stations and two analog trunks is also available.

SIP Trunking

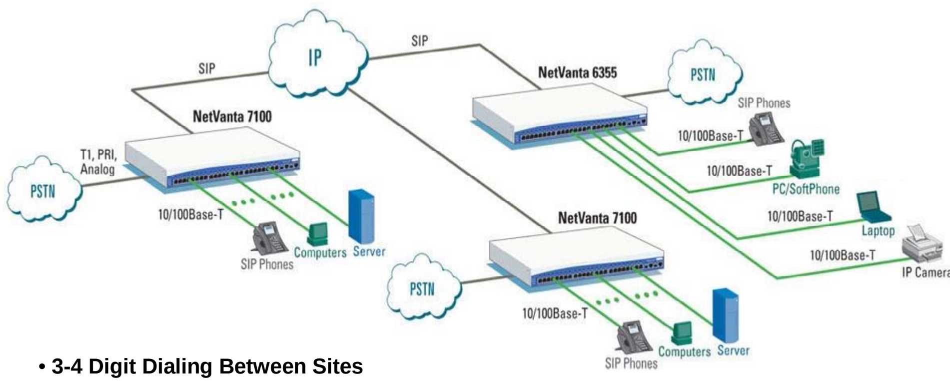


For businesses that want to make full use of their new generation IP communication solution, the NetVanta® 7100 and 7060 provide SIP Trunking capabilities between the business and the local Service Provider. SIP trucking is a dynamic and efficient IP link that can carry voice and data traffic, replace the traditional TDM trunks and lower monthly service costs for the business.

- Converge voice and data across single trunk
- Dynamic bandwidth allocation for voice and data traffic
- Can support local, long distance and Internet
- Interoperable with a variety of carrier SIP Trunking services
- Direct Connectivity of NV 7100 to Carrier's SIP Trunk Service

SIP Networking

NetVanta 7000 Series Solution SIP Networking

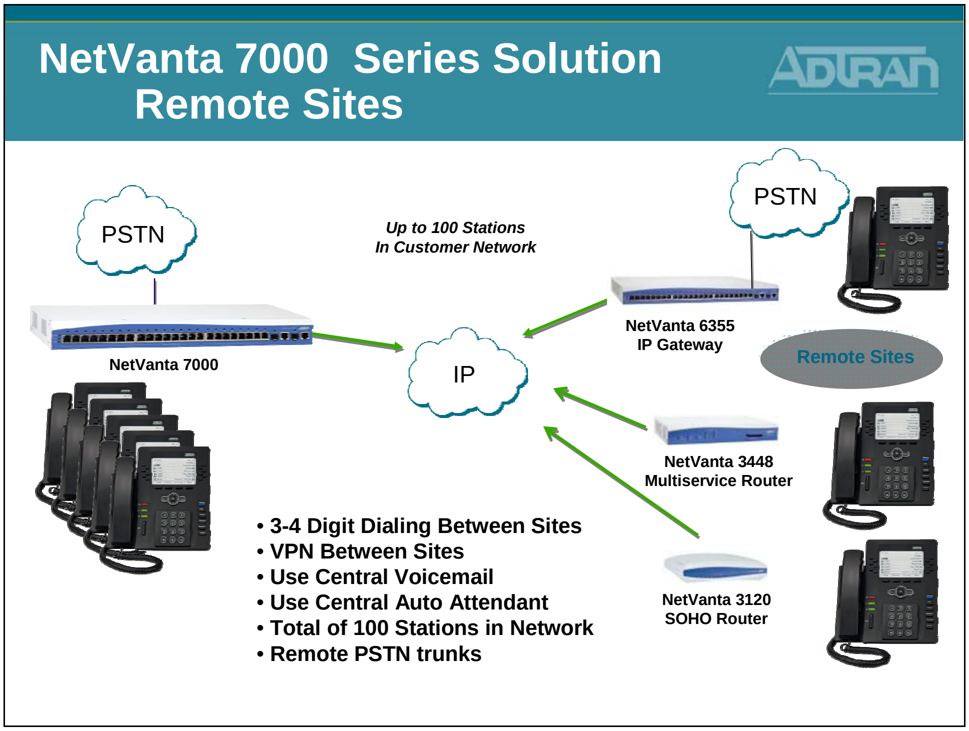


- **3-4 Digit Dialing Between Sites**
- **VPN Between Sites**
- **Toll Bypass**
- **Access Remote Site Local Trunks**

The ADTRAN NetVanta 7000 Series will support SIP networking between multiple locations. With SIP Networking, businesses will be able to connect multiple sites and have three- to four-digit dialing, local call routing and survivability, and on-net calls for toll bypass. The NetVanta 7100 and 7060 are best for locations that need local voice mail; while ADTRAN's NetVanta 6355 IP Business Gateway provides the ideal solution for locations that will use a central NetVanta 7000 voice mail.

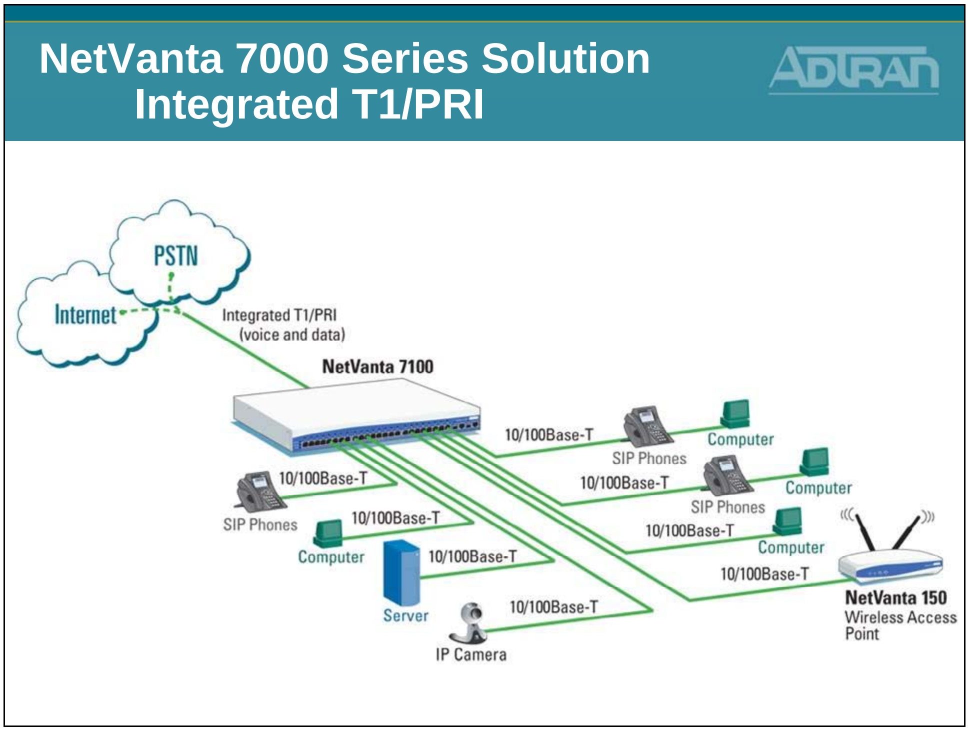
- Links multiple sites together
- Supports inter-office, three- to four-digit dialing
- Provides local PSTN access

Remote Site Solutions



The ADTRAN NetVanta 7000 Series will support SIP networking between multiple locations. The NetVanta 7100 and 7060 are best for locations that need local voice mail; while ADTRAN's NetVanta 6355 IP Business Gateway provides the ideal solution for locations that will use a central NetVanta 7000 voice mail. The remote site NetVanta 3448 router or 6355 can provide local survivability as well by continuity to route intra-office calls, or where provisioned, directly to a local PSTN for guaranteeing phone service. The NetVanta 7100 and NetVanta 3120 enable secure, always-on, voice, data and high-speed data access to business resources from a remote home office.

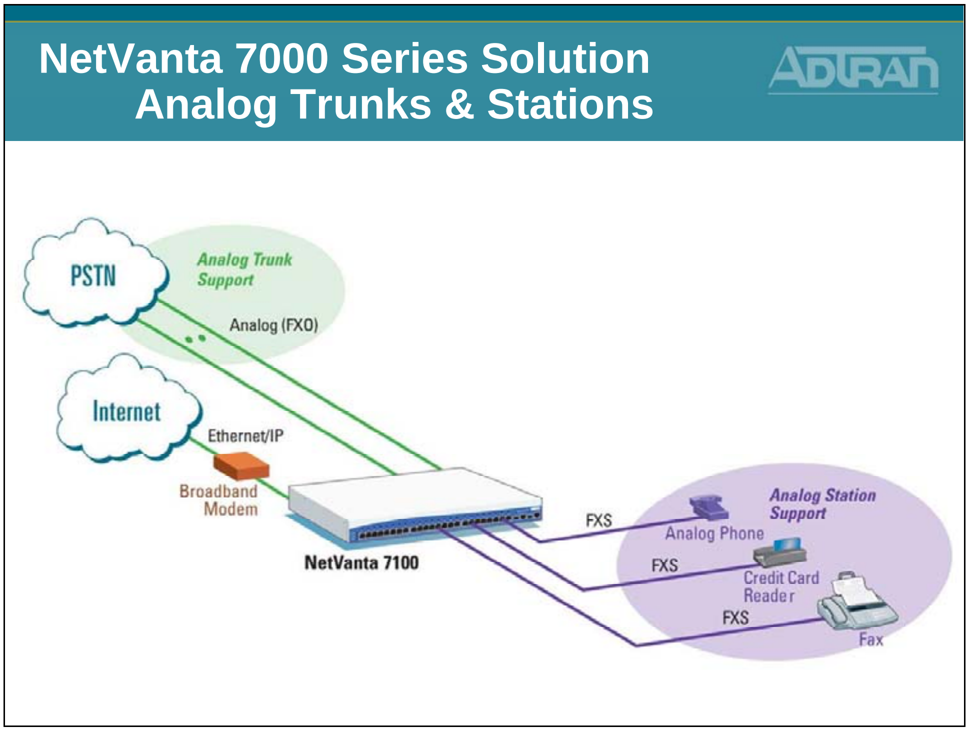
Integrated T1 /PRI



Using the NetVanta 7000 Series T1/PRIVoice Interface Module (VIM), customers can consolidate separate voice lines and Internet access onto a single T1 or PRI trunk. Small-to Medium-sized Business (SMB) locations with analog business lines and high-speed Internet access can benefit from lower monthly costs, higher reliability, and added capacity for growth through T1/PRIconsolidation. Check with your service provider for attractive offers on integrated T1/PRI circuits and terminate the service with the NetVanta 7000 Series T1/PRI VIM for an ideal business-grade Voice over IP (VoIP) solution.

- Supports up to 24 T1 channels
- Supports up to 23 PRI channels
- Consolidates voice and data
- Reduces monthly service costs

Analog Trunks & Stations



ADTRAN's NetVanta® 7100 is ideal for businesses that need a combination of IP and analog communications. Along with IP interfaces, the NetVanta 7100 can support analog trunks, analog phones, fax machines and credit card readers without the need for analog telephone adapters.

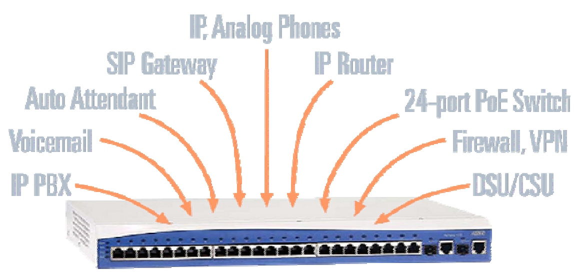
- Eliminates the need for additional analog telephone adaptors
- Supports up to 10 analog ports
- Enables analog data devices to achieve higher-speed performance

Module Objectives

Module Summary

At the end of this module, you should be able to:

- Discuss ADTRAN's IP Telephony Solutions
- Discuss ADTRAN's IP Telephony Features
- Recognize Key NetVanta IP Telephony Applications



All in One "Office in a Box"

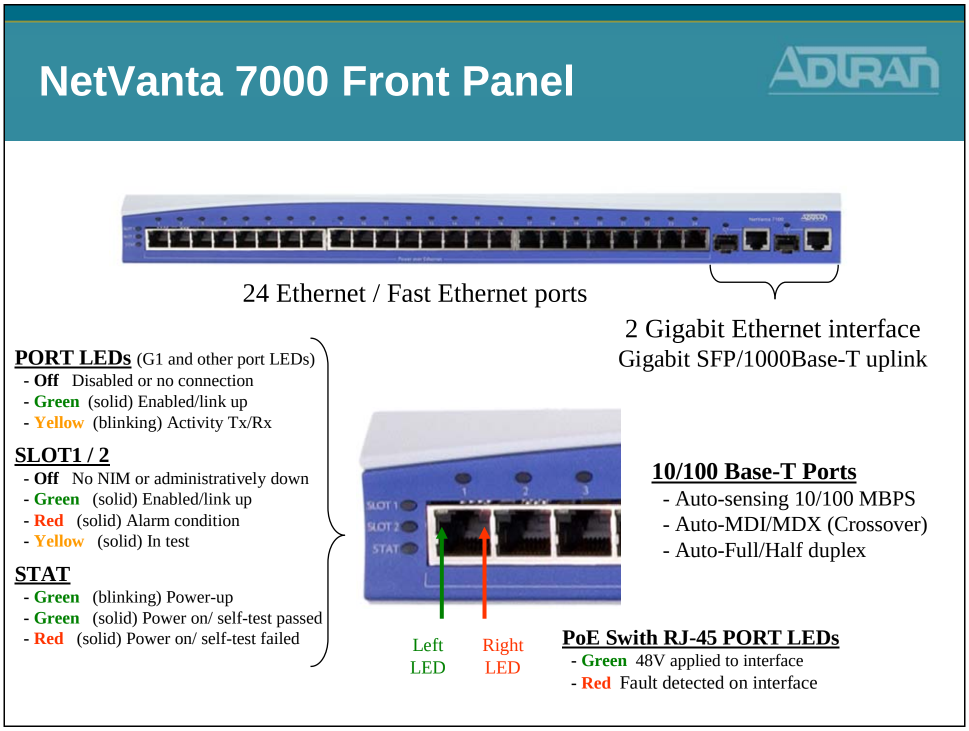
Module 2: Introduction to NetVanta 7000 Series Data Configuration

Module Objectives

Objectives

- Introduce the ADTRAN Operating System (AOS)
- Introduce the NetVanta 7000 Web-Based GUI
- Understand System Factory Defaults
- Understand Switch Factory Defaults
- Understand Router Factory Defaults
- Understand Firewall Factory Defaults

NetVanta 7000 Front Panel



Front Panel RJ-45 Ports and LEDS

The NetVanta front panels contain twenty-four 10/100BaseT Ethernet ports (RJ-45). These ports are consecutively numbered one through twenty-four, from left to right, with the numbers screened directly above each port. Status LEDs for each of these ports are located directly over these numbers.

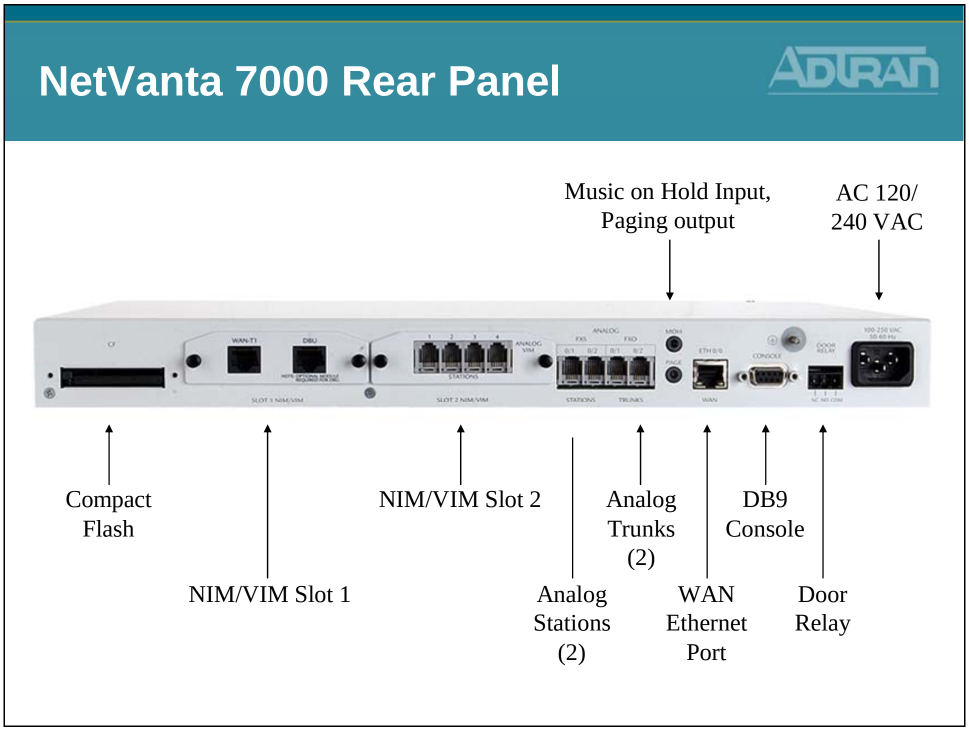
Front Panel Gigabit Ethernet Interfaces and LEDs

The NetVanta 7000 front panel also contains two Gigabit Ethernet interfaces. These interfaces are provided as RJ-45 jacks or SFP slots and are labeled G1 and G2.

Power Over Ethernet

The NetVanta 7000 Power over Ethernet (PoE) interfaces provide the ability to detect attached powered devices (PD) and deliver 48 VDC to the PD via existing CAT5 cabling. The PoE interfaces are fully compliant with the IEEE 802.3af power over Ethernet standard. By default, the PoE ports automatically discover and provide power to IEEE-compliant PDs.

NetVanta 7000 Rear Panel



The NetVanta 7000 rear panel contains a power connection and a single DB-9 (female) interface (labeled CONSOLE) used for connecting to a VT100 terminal or a PC running VT100 terminal emulation software. The rear panel also includes the Ethernet port (labeled ETH 0/0) for WAN and/or administration connectivity, dual analog stations and trunks, compact flash (CF), message on hold (MOH), PAGE, and alarm contacts (DOOR RELAY). In addition, the NetVanta 7000 contains modular network interfaces that accept a variety of modules.

NetVanta 7000 Memory

NetVanta 7000 Memory

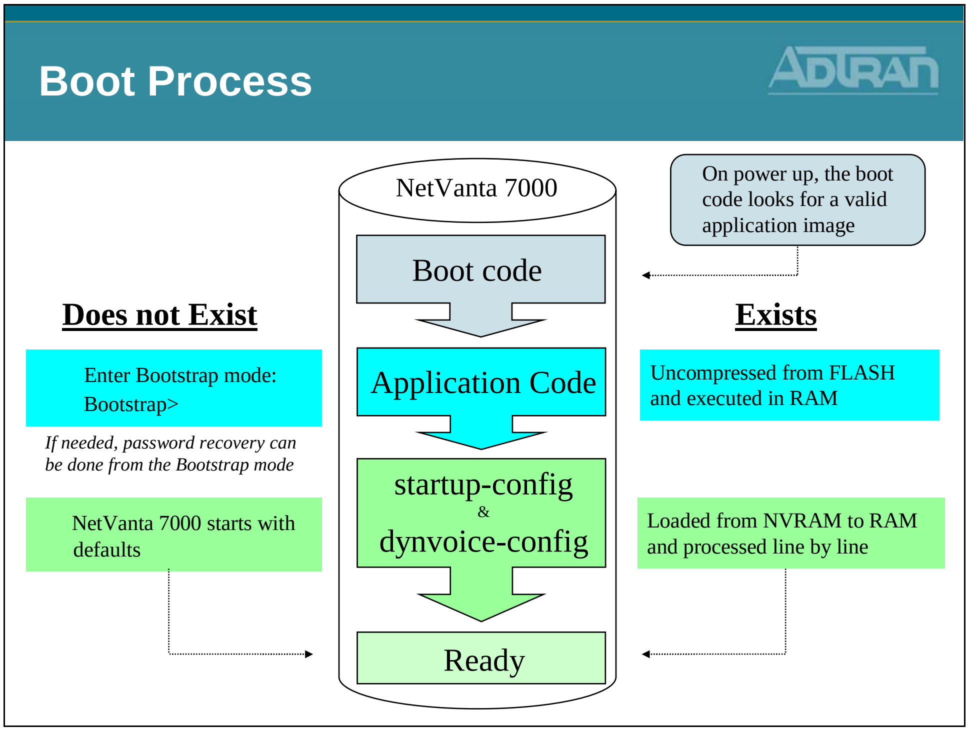
- FLASH (32 Mbytes)
  - boot code storage / compressed application code storage
  - store non-volatile configuration data (**startup-config**)
  - store non-volatile dynamic voice config (**dynvoice-config**)
  - retains contents when NetVanta is powered down
- CFLASH (256 Mbytes)
  - Non-volatile storage of Voicemail and User prompts
  - Firmware and configs can be stored here
  - Can store up to 3000 voicemail messages
  - retains contents when NetVanta is powered down
- RAM (128 Mbytes)
  - running copy of the application code
  - running copy of the configuration file (**running-config**)

Flash memory is non-volatile memory and is where the boot code, compressed application code, saved configurations, and startup-configurations are stored. Everything in Flash is saved when the NetVanta is powered down. The NetVanta has the ability to save different user defined configurations that may be loaded into the running-configuration in RAM. The number of configuration files that can be saved is only limited by the amount of Flash memory used.

RAM (Random Access Memory) is the main memory and contains a running copy of the application code, a running copy of the configuration file, and is considered volatile memory. Therefore, it is cleared when the NetVanta AOS device is powered down. The actual compressed application code is stored in Flash, but is uncompressed and stored in RAM upon device bootup. Changes to the running-configuration are also stored here. This is why it is important to save or write your configuration changes to FLASH and therefore include them in your startup-configuration file. The type of RAM typically incorporated in the AOS devices is dynamic RAM (DRAM).

The CF (CompactFlash) slot supports a small flash memory module. The memory chips are enclosed in a case and retain data after they are removed from the system. The CompactFlash card may be used to store configuration files and AOS images.

Boot Process



Unit Boot Up

Plug the unit into the wall and turn on the power. The unit begins the boot up process, which includes the following:

- The Power-On Self Test runs.
  - This test checks the unit hardware for normal operation. The hardware includes the central processing unit (CPU), the memory, and the interfaces.
- The Bootstrap Startup Program (factory set in the ROM) runs.
  - The Bootstrap Startup Program is read by the unit to discover the proper source for the operating system image.
- The operating system image is loaded into RAM.
- The configuration files startup-config and dynvoice-config saved in NVRAM are loaded into RAM, where they are accessed by the unit and then executed one line at a time.

Configuration Methods

Configuration Methods

Two Configuration methods

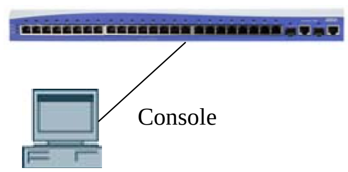
- ADTRAN Operating System (CLI)
  - Connect a PC's VT100 Terminal
  - Console port, telnet, secure shell
- Web-Based GUI
  - PC with installed web browser
  - HTTP (port 80) or HTTPS (port 443)
  - Internet Explorer 5 or Higher; Firefox 1.5 or Higher

The NetVanta products can be configured through the Command Line Interface (CLI) or the Web-based Graphical Interface (GUI). Both are enabled from the factory.

Console Port Connection

Console Port Connection

- Use a DB-9 (male) to DB-9 (female) straight-through serial cable
- Open a VT-100 session with the NetVanta 7100
- Configure the COM port with the following parameters:
  – Data Rate: 9600
  – Data Bits: 8
  – Parity Bits: None
  – Stop Bits: 1
  – Flow Control: None



ACCESSING THE CLI

Access the AOS CLI via the CONSOLE port or a Telnet session. To establish a connection to the NetVanta unit's CONSOLE port, you need the following items:

- VT100 terminal or PC (with VT100 terminal emulation software)
- Straight-through serial cable with a DB-9 (male) connector on one end and the appropriate interface for your terminal or PC communication port on the other end

a. Connect the DB-9 (male) connector of your serial cable to the CONSOLE port on the rear panel of the unit.

b. Connect the other end of the serial cable to the terminal or PC.

c. Insert the connector of the provided power cord into the power interface on the rear panel of the unit, and plug the cord into a standard electrical outlet.

d. Once the unit is powered up, open a VT100 terminal session using the following settings: 9600 baud, 8 data bits, no parity bits, and 1 stop bit.

e. Press <Enter> to activate the AOS CLI.

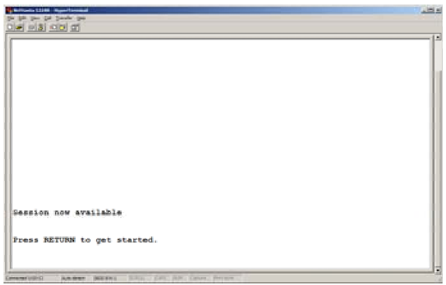
f. Enter "enable" at the > prompt and then the enable password when prompted

   - The default enable password is **password**

ADTRAN Operating System

ADTRAN Operating System

- ADTRAN Operating System
  - Command Line Interface (CLI)



  - Press RETURN to access the basic ADTRAN OS security level

When you first log into the unit, or if your session has timed out, you will see the screen above. Simply press <Return> or <enter> to log back into the NetVanta.

Note: This allows you to access the NetVanta's Command Line Interface.

Command Security Levels

Command Security Levels

Two Command Security Levels

– Each security level supports a specific set of commands

- Basic Level (Initial Level) **NV7100>**
  – display system information with show command
  – perform traceroute, ping, and telnet
  – enter the enable (privileged) level

```
NV7100> enable
  Password : ********
NV7100#
```

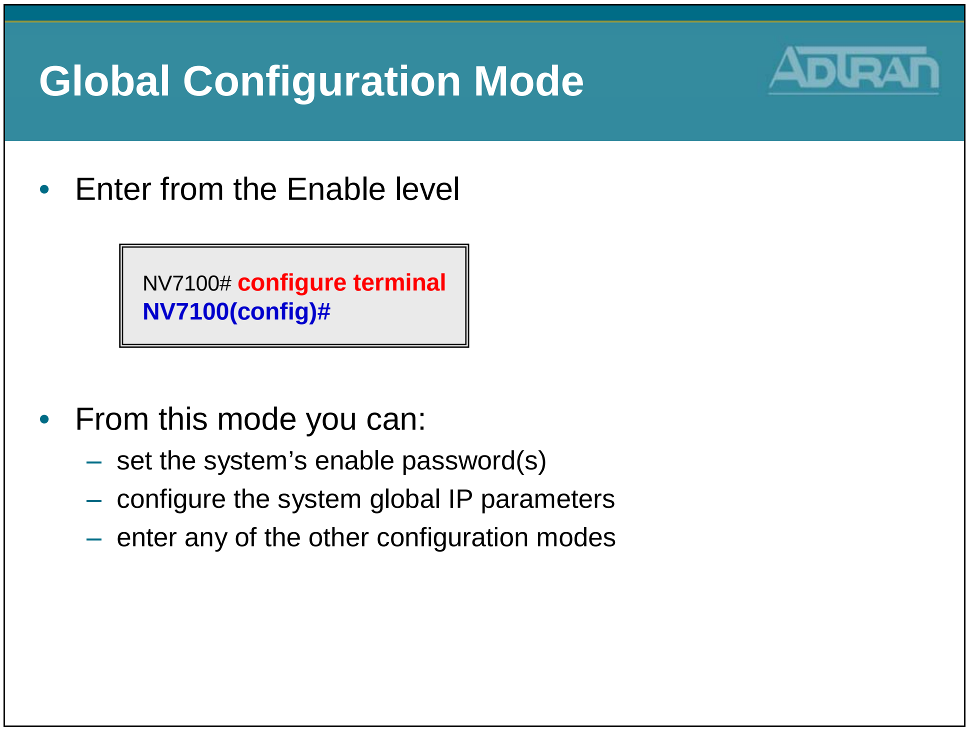
- Enable Level **NV7100#**
  – manage the startup and running configurations
  – use the debug commands
  – enter the Global Configuration mode

There are two command security modes, each one supporting a specific set of commands. When first logging into the NetVanta via the Command Line Interface (CLI), you are in Basic mode.

Basic Mode
Interaction with your unit begins at the Basic mode. The commands supported at this command tier are limited, as is interaction with the unit itself. The Basic mode is to keep users without access to the higher tiered commands from changing the preferred configurations of the unit.

Enable Mode
Enable mode is the privileged mode in the command hierarchy, one step up from the Basic mode. ADTRAN suggests that a password be required to access the Enable mode. From the Enable mode, you can access the configurations of your product as well as handle the boot settings and running configuration, among other things.

Global Configuration Mode



Global configuration mode allows the user to set the system's enable passwords, configure the global IP parameters, and enter into any of the other configuration modes.

To see the commands available to the Global configuration mode, type a question mark at the prompt. A list of commands and brief description of their function will be displayed.

Specific Configuration Modes

Specific Configuration Modes

- Global Configuration Mode
  - NV7100(config)#
- Line Configuration Mode
  - NV7100(config-con0)# (config)# **line con 0**
  - NV7100(config-telnet0)# (config)# **line tel 0**
- Router Configuration Mode
  - NV7100(config-rip)# (config)# **router rip**
  - NV7100(config-ospf)# (config)# **router ospf**
- Interface Configuration Mode
  - NV7100(config-eth 0/1)# (config)# **int eth 0/1**
  - NV7100(config-t1 1/1)# (config)# **int t1 1/1**
  - Type **exit** to return to Global Config mode
  - Type **<ctrl> "z"** to exit out of Config mode

```
NV7100(config-rip)# exit
NV7100(config)#
```

```
NV7100(config-rip)# <ctrl> z
NV7100#
```

The Global configuration mode allows the user to make changes that are 'global' to the unit, and not specific to one interface. A configuration change made in Global configuration mode would affect all the enabled interfaces in the device.

Examples of the various configuration modes are displayed below:

Mode	Access by…	Sample Prompt	Operation
Global	Entering **config** while at the Enable command security level prompt. Example: >enable #config t	(config)#	• Set the system's Enable-level password(s) • Configure the system global IP parameters • Configure the SNMP parameters • Enter any of the other configuration modes

Line	Specifying a line (console or Telnet) while at the Global Configuration mode prompt. For example: >enable #config t (config)#line console 0	(config-con0)#	• Configure the console terminal settings (data rate, login password, etc.) • Create Telnet login and specify parameters (login password, etc.)
Router	Enter **router rip** or **router ospf** while at the Global Configuration mode prompt. For example: >enable #config t (config)#router rip	(config-rip)#	• Configure RIP or OSPF parameters • Suppress route updates • Redistribute information from outside routing sources (protocols)
Interface	Specify an interface (T1, Ethernet, Switchport, Frame Relay, PPP, etc.) while in the Global Configuration mode. For example: >enable #config t (config)#int eth 0/1	(config-eth 0/1)# (The above prompt is for the first Ethernet switchport interface located on the front panel of the unit.)	• Configure parameters for the available LAN and WAN interfaces

Help Tools

Help Tools

- "?" Command
  - List available commands

    ```
    NV7100# ?
    ```

  - List options available to command

    ```
    NV7100# show ?
    ```

  - Auto finish

    ```
    NV7100# tr <Tab>
    NV7100# traceroute
    ```

Arguably, the ? is the most used command in the CLI. No matter if one is a novice or expert the ? is a valuable resource. There are thousands of commands and parameters in the AOS and the ? allows one to search for the elusive directive.

To aid in the execution and at times the correction of commands the AOS includes shortcut keys. These shortcuts move the cursor forward and backward on the command line.

Further information regarding these Help tools is available on the following pages.

The following shortcut keys are available from the CLI configuration:

Shortcut	Description
Up arrow key	To re-display a previously entered command, use the up arrow key. Continuing to press the up arrow key cycles through all commands entered starting with the most recent command.
<**Tab**> key	Pressing the <**Tab**> key after entering a partial (but unique) command will complete the command, display it on the command prompt line, and wait for further input.
?	The ADTRAN CLI contains help to guide you through the configuration process. Using the question mark, do any of the following: Display a list of all subcommands in the current mode. For example: (config-t1 1/1)#**coding ?** ami - Alternate Mark Inversion b8zs - Bipolar Eight Zero Substitution Display a list of available commands beginning with certain letter(s). For example: (config)#**ip d?** default-gateway dhcp-server domain-lookup domain-name domain-proxy Obtain syntax help for a specific command by entering the command, a space, and then a question mark (?). The ADTRAN CLI displays the range of values and a brief description of the next parameter expected for that particular command. For example: (config-eth 0/1)#**mtu ?** <64-1500> - MTU (bytes)
<**Ctrl + A**>	Jump to the beginning of the displayed command line. This shortcut is helpful when using the **no** form of commands (when available). For example, pressing **<Ctrl + A>** at the following prompt will place the cursor directly after the **#**: config(eth-0/1)#ip address 192.168.10.1 255.255.255.0
<**Ctrl + E**>	Jump to the end of the displayed command line. For example, pressing <**Ctrl + E**> at the following prompt will place the cursor directly after the **1**: config(eth-0/1)#ip address 192.168.10.**1**
<**Ctrl + U**>	Clears the current displayed command line. The following provides an example of the <**Ctrl + U**> feature: config(eth-0/1)#ip address 192.168.10.1 255.255.255.0 (Press <**Ctrl + U**> here) config(eth-0/1)#
auto finish	You need only enter enough letters to identify a command as unique. For example, entering **int t1 1/1** at the Global configuration prompt provides you access to the configuration parameters for the specified T1 interface. Entering **interface t1 1/1** would work as well, but is not necessary.

General Command Introduction

General Command Introduction

- Basic security level
  - show version
  - enable
- Enable security level
  - show flash
  - show cflash
  - show startup-config
  - show running-config
  - copy running-config startup-confi

show version command

show version command

- Displays system hardware and software info

```
NV7100> show version
ADTRAN, Inc. OS version A2.03.00.E
  Mainline Version: M00
  Checksum: 8A30C916, built on Fri Mar 27 08:22:35 2009
  Upgrade key: fb9ab213c71d061d002d70615ed80777
Boot ROM version 15.01.00
  Checksum: 0F45, built on: Thu Apr 26 10:28:09 2007
Copyright (c) 1999-2007, ADTRAN, Inc.
Platform: NetVanta 7100, part number 1200796E1
Serial number G17A8905
Flash: 33554432 bytes  DRAM: 134217728 bytes
NV7100 uptime is 3 days, 19 hours, 8 minutes, 13 seconds
slot 0, DSP 1
  DSP software version: G1.A2.02.17
  DSP hardware version: Freescale MSC7116
  Total channels: 20
System returned to ROM by Software Watchdog
Current system image file is "NONVOL:/NV7100A-A2-03-00-E.biz"
Boot system image file is "NONVOL:/NV7100A-A2-03-00-E.biz"
Primary system configuration file is "startup-config"
NV7100#
```

Use the show version command to display the current AOS version information.

Other key information that appears from the **show version** output is the NetVanta unit information including the *part number* and *serial number*.

show flash command

show flash command

- List files stored in FLASH

```
NV7100# show flash
       158    000000000000-directory.xml
       462    000000000000.cfg
     20280    adtran-sip.cfg
       939    adtran_000000000000.txt
       163    adtran_boot.txt
                    :
       119     customer-sip.cfg
      7890    defaultpolycom.cfg
      2512    dynvoice-config
     12164    startup-config
       837    polycomboot.cfg
       739    polycomboot_remote.cfg
       779    polycomConfigDefaults.cfg
    130729    sip.cfg
        87    sip.ver
  14540095    NV7100A-A2-03-00-E.biz
15055905 bytes used, 15684030 available, 30739935 total
NV7100#
```

startup-config / dynvoice-config
text files that are read and executed line by line at startup

.biz file
the NetVanta 7100 application image

The **show flash** command may be executed from the Enable security mode and shows what is currently stored in the Flash portion of NVRAM. In this output, the **".biz"** file is the application image, or the firmware. Generally, any application images will have a .biz extension. There may be multiple image files stored in flash with a .biz extension. The sizes of each of the files are listed in front of the file names. The total space used and available is also shown.

To view the image that was loaded upon startup, type the **show version** command.

Other files listed in flash include the **startup-configuration** file, the **startup-config.bak** file, and any other configuration files that have been created.

The **startup-config** file is a text file that is read and executed line by line at startup. If no **startup-config** file exists and no other file is specified to be used at startup, the router will load with factory default settings. If a **startup-config** file does exist and no other file is specified to be used at startup, the NetVanta will always use this file named **startup-config** to load the initial configuration. The **startup-config.bak** file is a backup file that is automatically created and updated as changes are made to the **startup-config** file.

show cflash command

show cflash command

- List files stored in Compact FLASH

```
NV7100# show cflash
 373 000000000000.cfg
(dir)        0   AA
(dir)        0   ADTRAN
(dir)        0   Polycom
           845   polycomboot.cfg
           739   polycomboot_remote.cfg
(dir)        0   SystemDefaultPrompts
(dir)        0   VoiceMail
29847552 bytes used, 225980416 available, 255827968 total
NV7100#
```

AA - Stores Auto Attendant Files
ADTRAN - Stores ADTRAN phone config files
Polycom - Stores Polycom phone config files

- System Default Prompts
  - General System prompts
  - Used with Auto Attendant, Voicemail, etc..
- Voicemail
  - Name, temporary, external, and internal greetings
  - Messages

Use the show cflash command to display a list of all files currently stored in CompactFlash® memory or details about a specific file stored in CompactFlash memory.

show startup-config

show startup-config

- Display the startup configuration

```
NV7100# show startup-config
```

  – startup-config is located in NVRAM
  – startup-config is loaded from NVRAM to RAM and processed line by line at startup

To show the contents of the startup-config file, use the command **show startup-config** at the Enable security mode. The startup-config file is stored in the Flash portion of NVRAM and will be displayed line by line to the screen output when executing this command. If no startup-config file exists, the router will show a message stating that **“**File does not exist.”

show running-config

show running-config

- Display the running configuration

```
NV7100# show running-config
```

  - running-config is located in RAM
  - Cleared when the NetVanta 7000 is powered down

Use the show running-config command to display all the non-default parameters contained in the current running configuration file. Specific portions of the running configuration may be displayed, based on the command entered. Variations of this command can be seen by issuing **“show run ?”** .

show running-config verbose

Default Settings

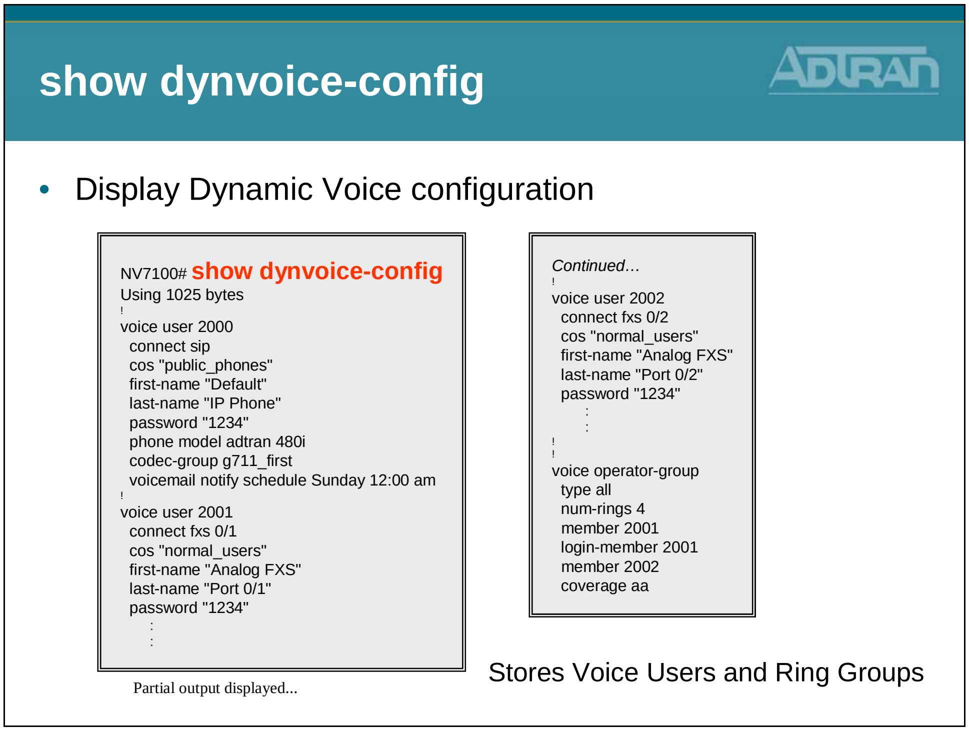
- Examine the running configuration along with the NetVanta 7000 default settings

```
NV7100# show running-config verbose
        :
line con 0
 no login
 password ""
 line-timeout 15
 databits 8
 parity none
 stopbits 1
 speed 9600
 no flowcontrol software in
        :
```

Partial output displayed...

The **show running-configuration** output only displays the basic configuration settings and any changes made from the default configuration settings. The **show running-configuration verbose** command displays all of the default and non-default configured parameters in the NetVanta device.

show dynvoice-config



Use the show dynvoice-config command to display the dynamic voice configuration. This file stores voice user and ring group configuration.

The dynvoice-config file is stored in the Flash portion of NVRAM and will be displayed line by line to the screen when executing this command.

Saving Configuration

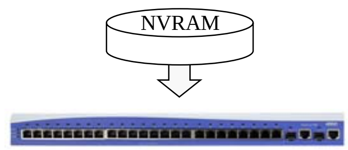
Saving Configuration

- Save current configuration to startup-config

NV7100# **copy running-config startup-config**

or

NV7100# **write memory**



- startup-config is located in NVRAM
- NVRAM retains contents when the NetVanta 7000 is powered down
- startup-config is read and executed line by line at startup

In order to save any changes that were made to the configuration since the unit was powered on, you must copy the running configuration into the startup configuration file in NVRAM.

The following commands may be used to save the configuration:

NV7100# **copy running-config startup-config**

NV7100# **write memory**

Factory-default Command

Factory-default Command

- Restore unit to factory-default settings

```
NV7100# factory-default
WARNING - Restoring the factory default settings will erase
the current startup configuration and will reboot the unit.
Restore factory default settings?[y/n]
```

  – This command erases the current startup-config and dynvoice-config files, and then creates the factory delivered startup-config and dynvoice-config files before rebooting

Use the factory-default command to reset the unit to the factory default settings.

After you issue this command, the system responds by first warning you that restoring the factory default settings will erase the current configurations. It then asks if you would like to proceed. Choose n to return to the command prompt (no configuration changes are made). Choose y to erase the startup-configuration, replace it with the factory-default configuration, and reboot the unit. After reboot, the new configuration takes effect.

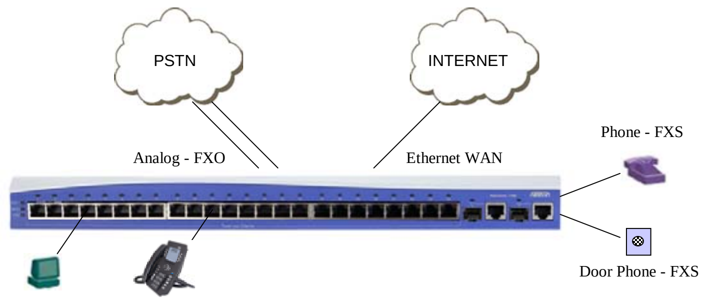
The only files that are affected by the factory-default are startup-config and dynvoice-config. No other files are removed or modified.

- IP phones look for configuration files from the boot server at boot. If you wish to default the unit and phones, the phone configuration files must be removed also.
- Phone configuration files are created by the NetVanta 7000 when creating new voice users for ADTRAN and Polycom phones. These files will be covered in a later module.

NetVanta 7000 - Factory Default Configuration

NetVanta 7000
Factory Default Configuration

ADTRAN



The NetVanta 7000 is delivered from the factory with a default configuration that will allow you to quickly deploy a complete IP telephony and Data networking Solution.

NetVanta 7000 - Data (VLAN) Factory Defaults

NetVanta 7000
Data (VLAN) Factory Defaults

ADTRAN

DATA VLAN
VLAN 1
IP Address: 10.10.10.1/24

NetVanta 7100

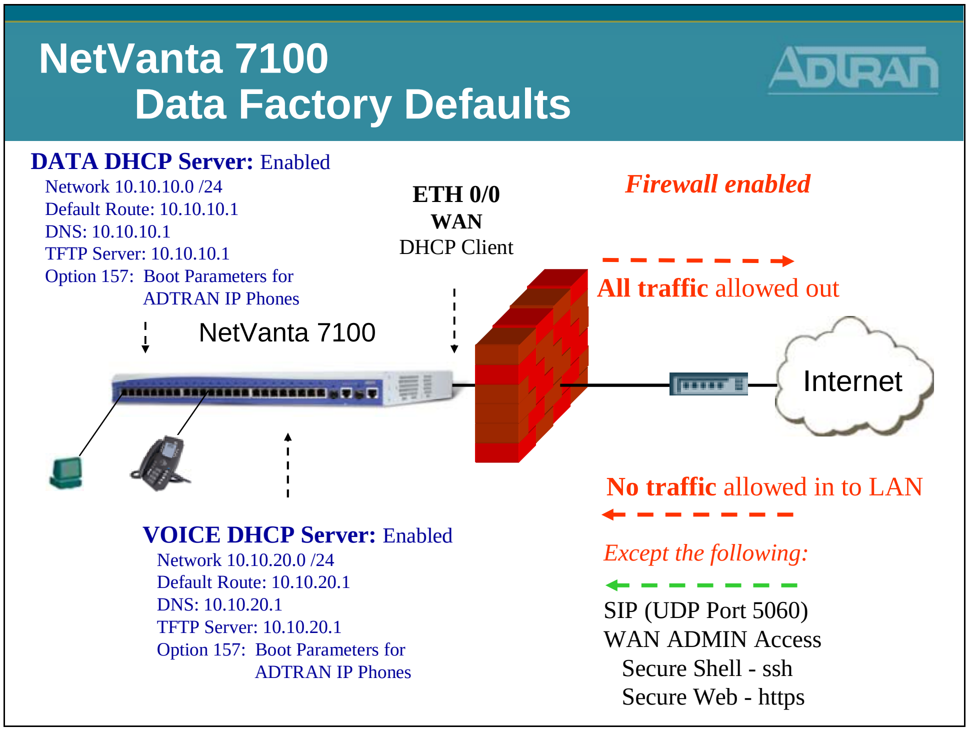
VOICE VLAN
VLAN 2
IP Address: 10.10.20.1/24

Native VLAN: 1

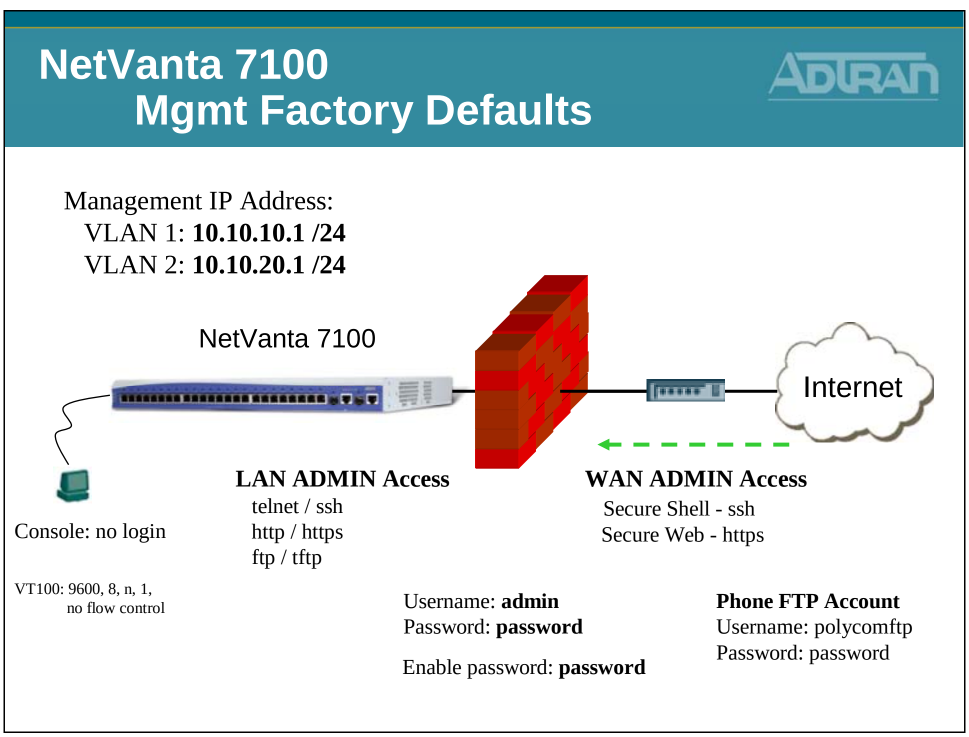
Ethernet 0/1-24
Enabled (activated with no shutdown)
Speed/Duplex/Cable Type: **auto**
Switchport mode: **trunk**
Allowed VLANs: **all**
Native VLAN: **1**
Spanning Tree Mode: **edgeport**

Gigabit 0/1-2
Enabled (activated with no shutdown)
Speed/Duplex/Cable Type: **auto**
VLAN membership: **trunk**

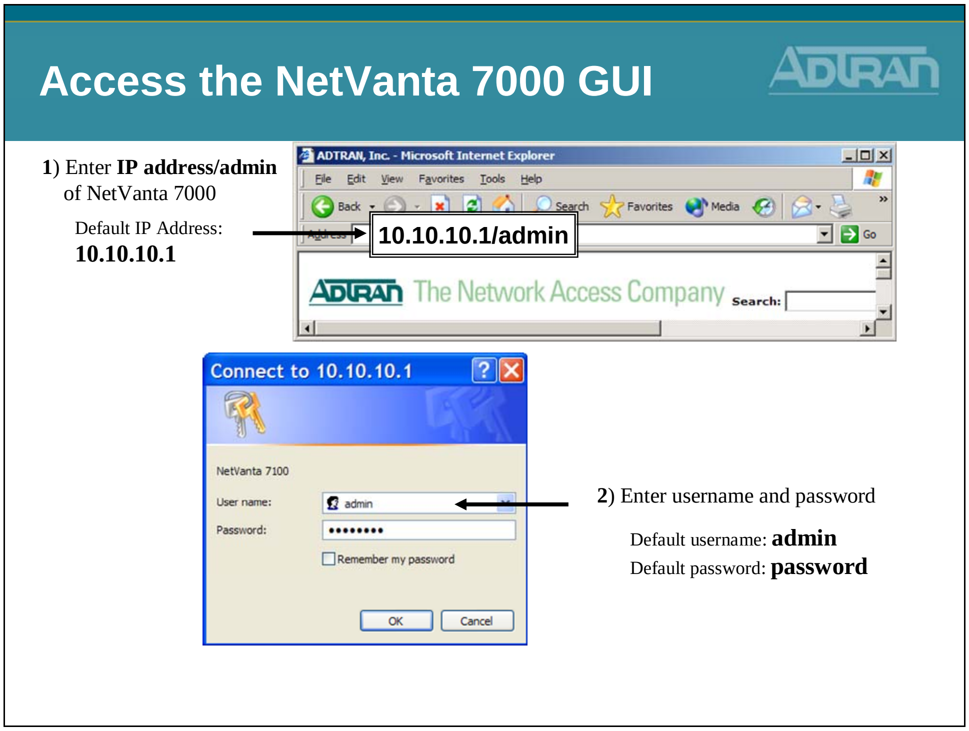
NetVanta 7100 - Data Factory Defaults



NetVanta 7100 - Mgmt Factory Defaults



Access the NetVanta 7000 GUI



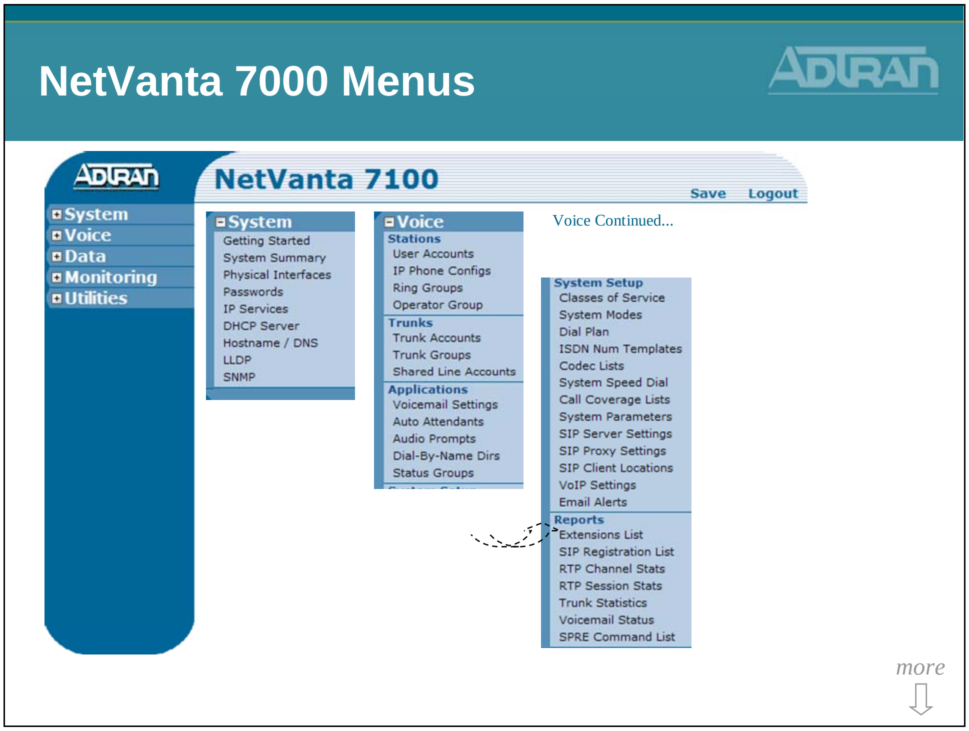
The Web-based Graphical User Interface (GUI) is enabled from the factory. If the web interface has been disabled or you wish to enable it with another NetVanta product, the minimum configuration would be:

- Turn on web server (ip http server)
- Add username and password (username admin password password)
- Assign IP address to VLAN or router interface

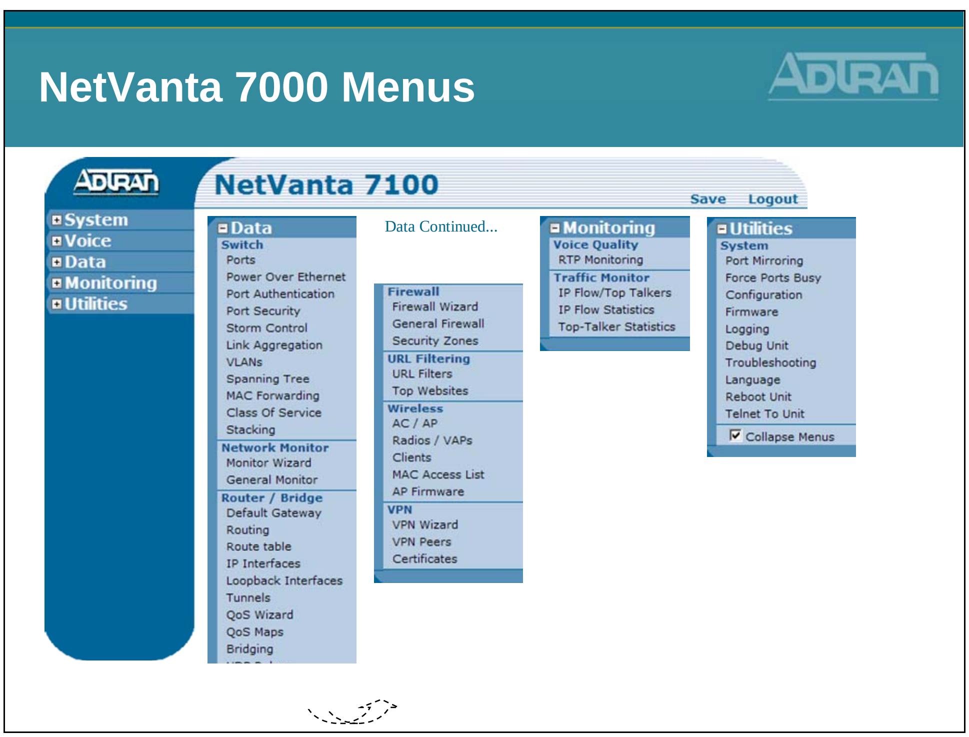
ACCESSING THE GUI

1. Connect the unit to your PC using the first Ethernet (eth 0/1) port on the front of the unit
2. Set your PC to obtain an IP address automatically via Dynamic Host Configuration Protocol (DHCP) or change your PC to a fixed IP address of 10.10.10.2
   - If you cannot change the PC's IP address, you will need to change the unit's IP address using the CLI
3. Enter the unit's IP address/admin in your browser address line
   - The default IP address is **10.10.10.1/admin**
4. You will then be prompted for the username and password
   - The default settings are **admin** and **password**

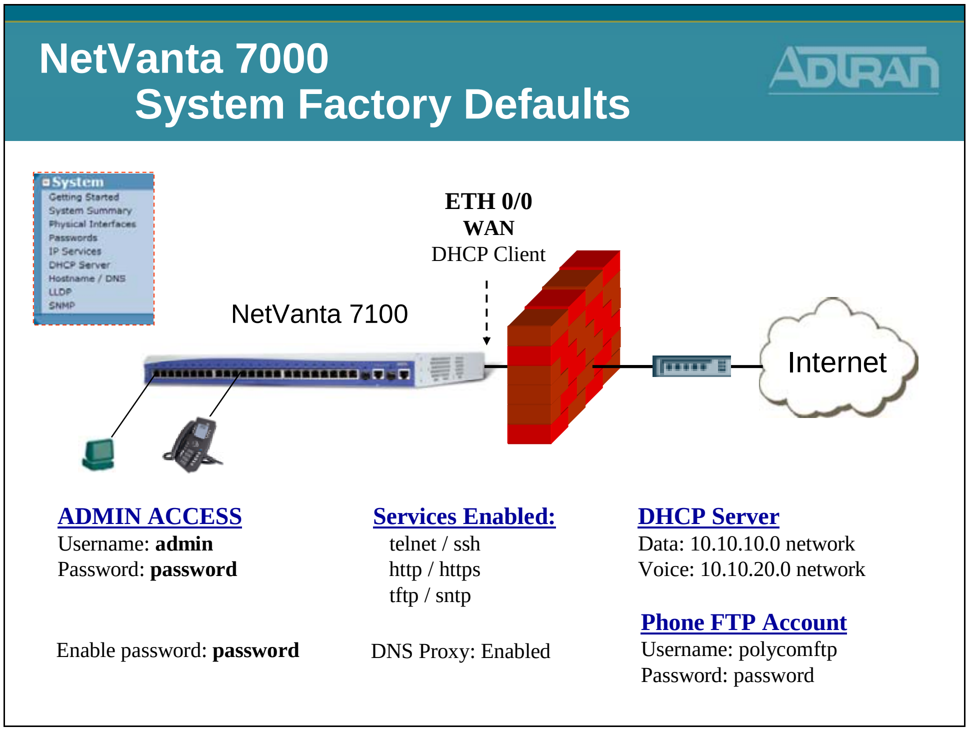
NetVanta 7000 Menus



NetVanta 7000 Menus



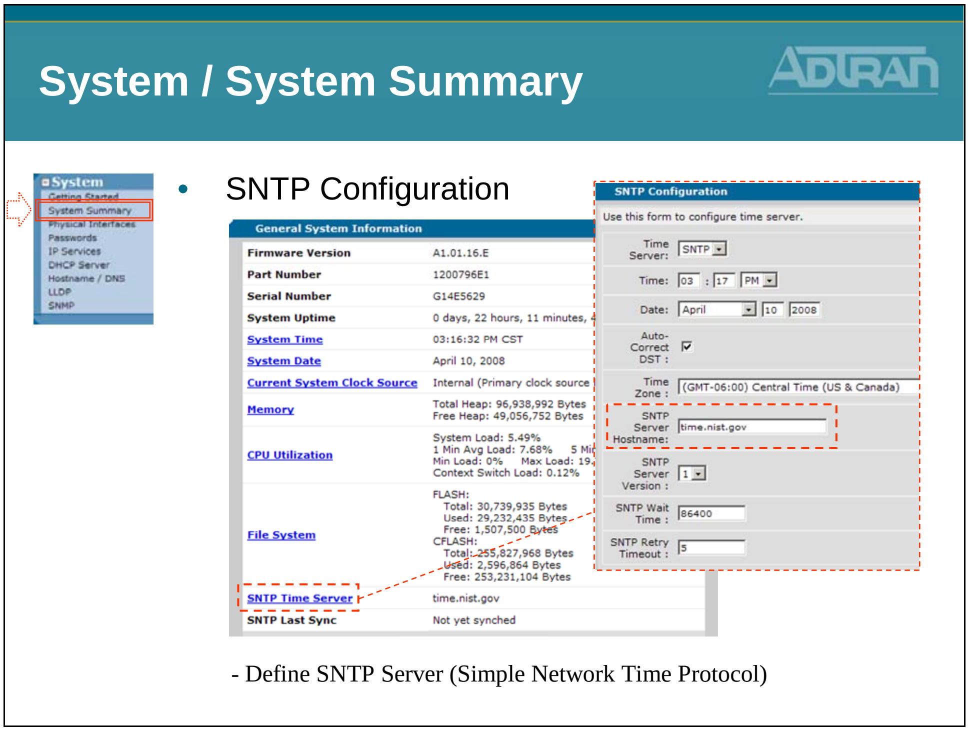
NetVanta 7000 - System Factory Defaults



From the factory, the NetVanta Web-Based GUI is enabled and ready to be accessed. The NetVanta is shipped from the factory with the services shown above.

ADTRAN strongly recommends that you change the default passwords shown above.

System/System Summary



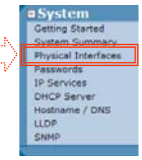
The System Summary screen allows the user to view general system information regarding the NetVanta 7000. This includes the firmware version, the part number, serial number, and system uptime. System time and date may also be viewed (and set) on this screen.

Current System Clock Source
The preferred timing source for the system is defined here. The NetVanta 7000 can have up to two independent T1 clock sources when a PRI is used. Select the T1/PRI interface to configure the system timing source for the voice subsystem.

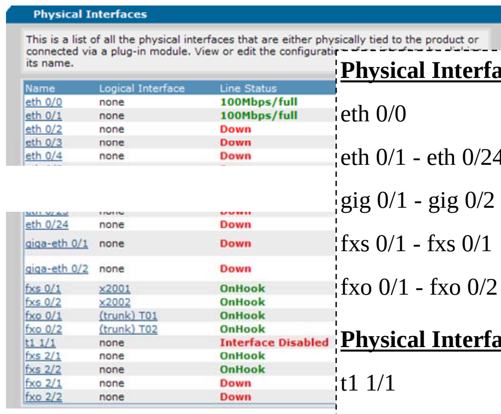
Configurable menu items such as system time and date are indicated by blue underlined text. The user may click on these items to make changes. Non-configurable items are shown in black text and are read-only status fields that may not be configured through this menu.

System/Physical Interfaces

System / Physical Interfaces



- List available Physical Interfaces

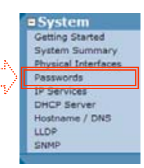


- *Includes NIM/VIM interfaces*

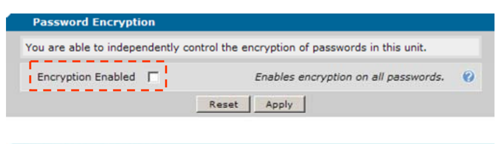
Physical Interfaces: Built In	
eth 0/0	WAN port
eth 0/1 - eth 0/24	Switch ports
gig 0/1 - gig 0/2	Gig switch ports
fxs 0/1 - fxs 0/1	Station Ports
fxo 0/1 - fxo 0/2	Trunk Ports
Physical Interfaces: Modular	
t1 1/1	T1/ PRI port
Fxs 2/1 - 2/2	Station ports
Fxo 2/1 - 2/2	Trunk port

System/Passwords

System / Passwords



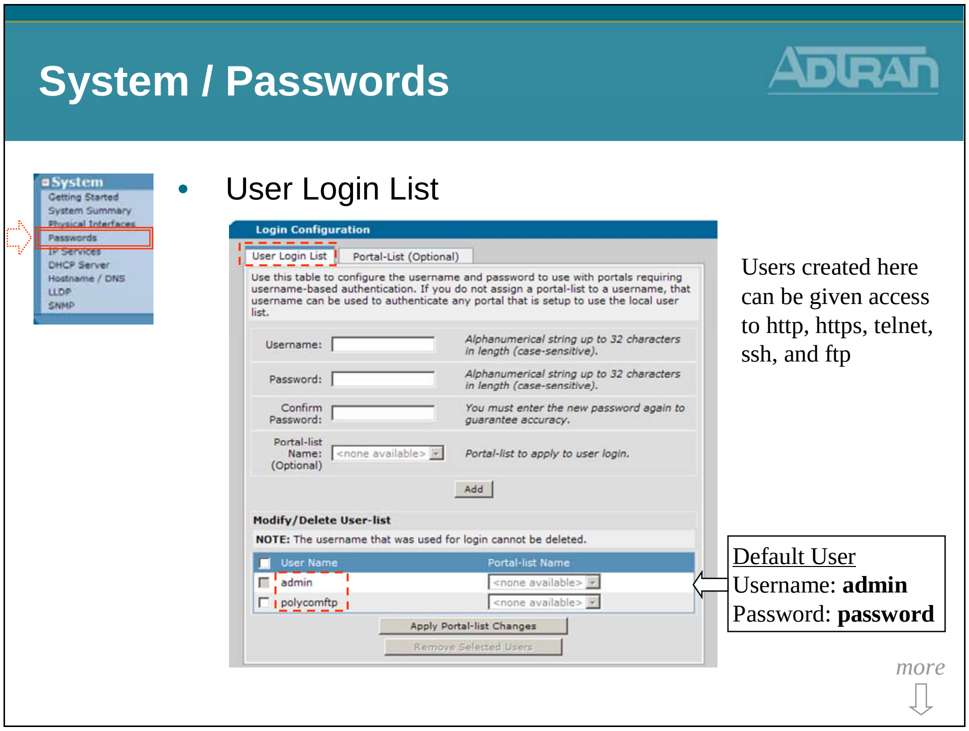
- Password Encryption



  - Enabling this feature will encrypt all existing passwords and any passwords entered in the future
  - When disabled, all passwords entered will be clear text
  - Example:
    - Clear text: username admin password password
    - Encrypted: username "admin" password encrypted "171fa669387f868ae7438c2154f6ae69bcb2"

more

System/Passwords



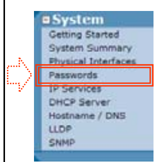
User Login List

Use this table to configure the username and password to use for all protocols requiring a user name-based authentication system, including FTP server authentication, line (login local-user list), HTTP, HTTPS, SSH, and Telnet access.

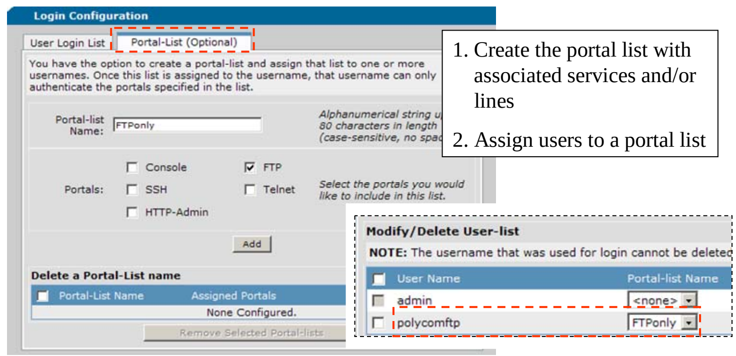
The username can be assigned a Portal List defining the specific application that this user will have access to. If you do not assign a portal-list to a username, that username can be used to authenticate any portal that is setup to use the local user list.

System/Passwords

System / Passwords



- Portal-List allow users to be created with restricted access modes



- Does not enable privilege levels
- Simply allows a user to access the router using only the specified services or lines

more

Portal-List (Optional)

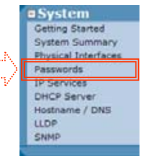
You have the option to create a portal-list and assign that list to one or more usernames. Once this list is assigned to the username, that username can only authenticate the portals specified in the list as shown below:

- Console
- FTP
- SSH
- Telnet
- HTTP-Admin

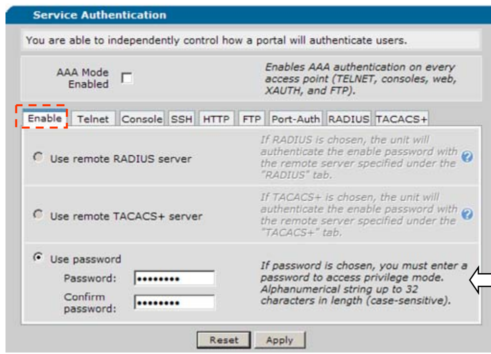
System/Passwords

System / Passwords

ADTRAN



- Enable Password



Default Password
password

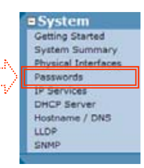
- The enable password is required to access the privileged "enable" mode from the command line of the ADTRAN Operating System

more

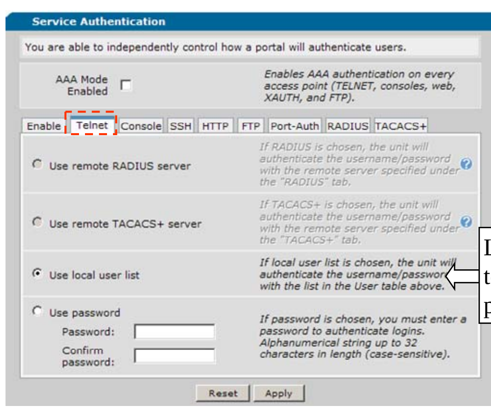
System/Passwords

System / Passwords

ADTRAN



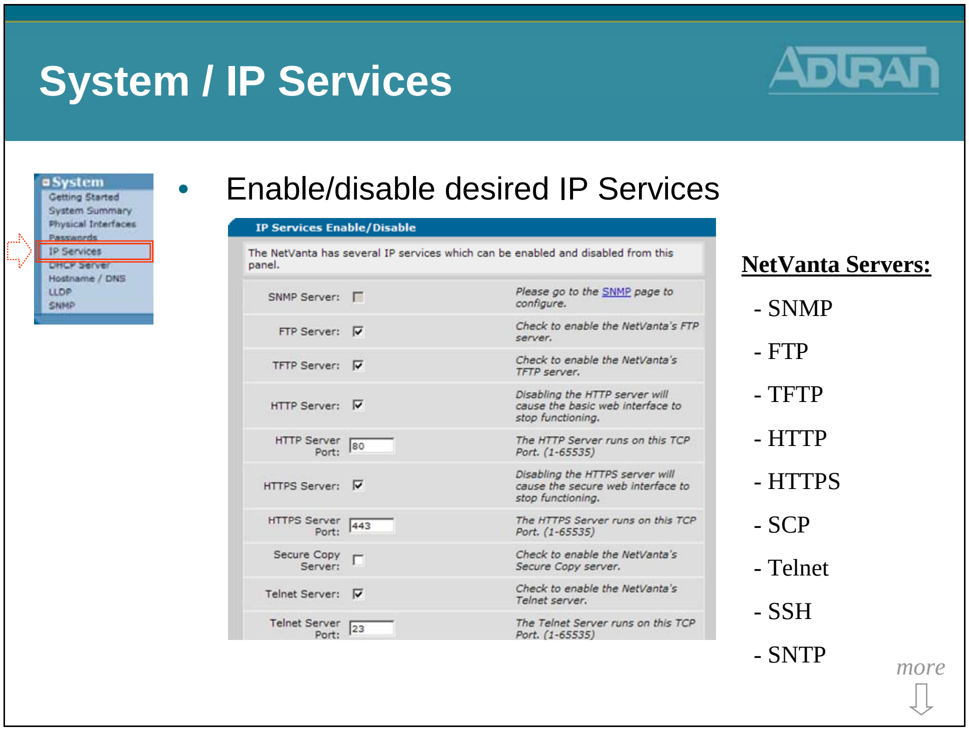
- Telnet Password



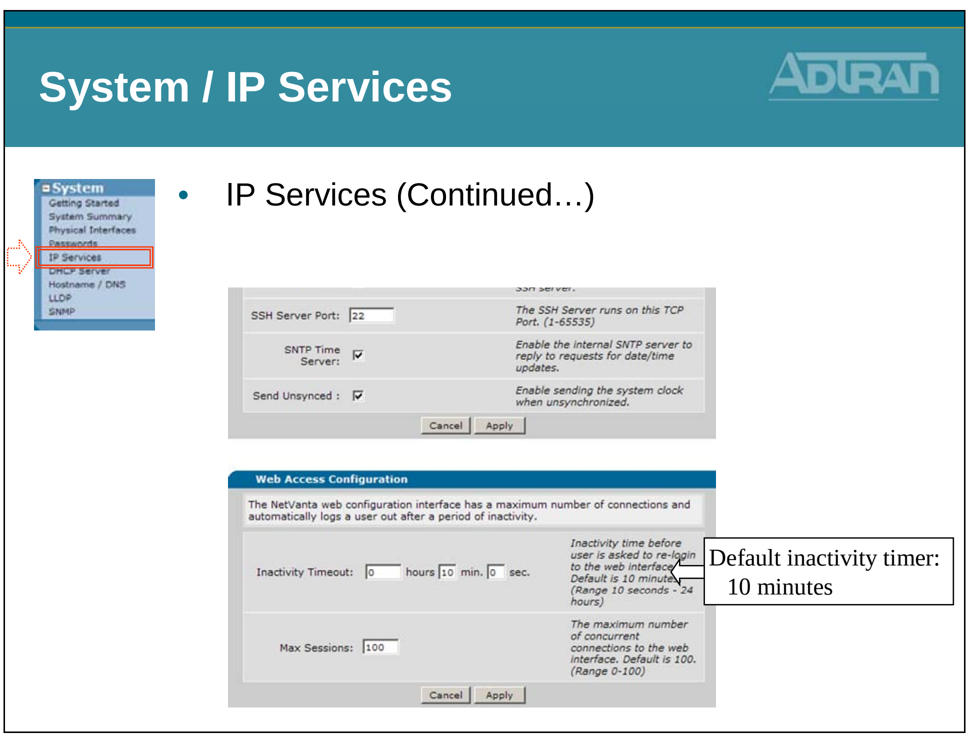
Local user list is default to username: **admin** and password: **password**

- The telnet password is required to remotely login to the command line of the ADTRAN Operating System

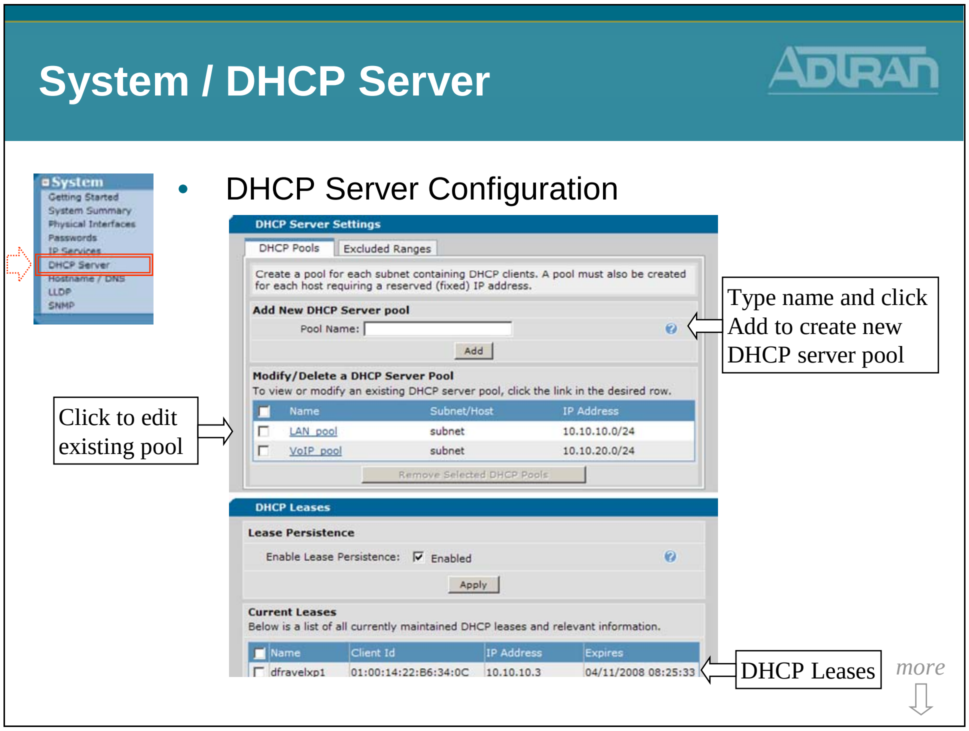
System/IP Services



System/IP Services



System/DHCP Server



The DHCP Server is enabled by default for both VLAN 1 and VLAN 2. The DHCP Server pool for VLAN 1, the data network, provides IP addresses from the 10.10.10.0 /24 network. The DHCP Server pool for VLAN 2, the voice network, provides IP addresses from the 10.10.20.0 /24 network.

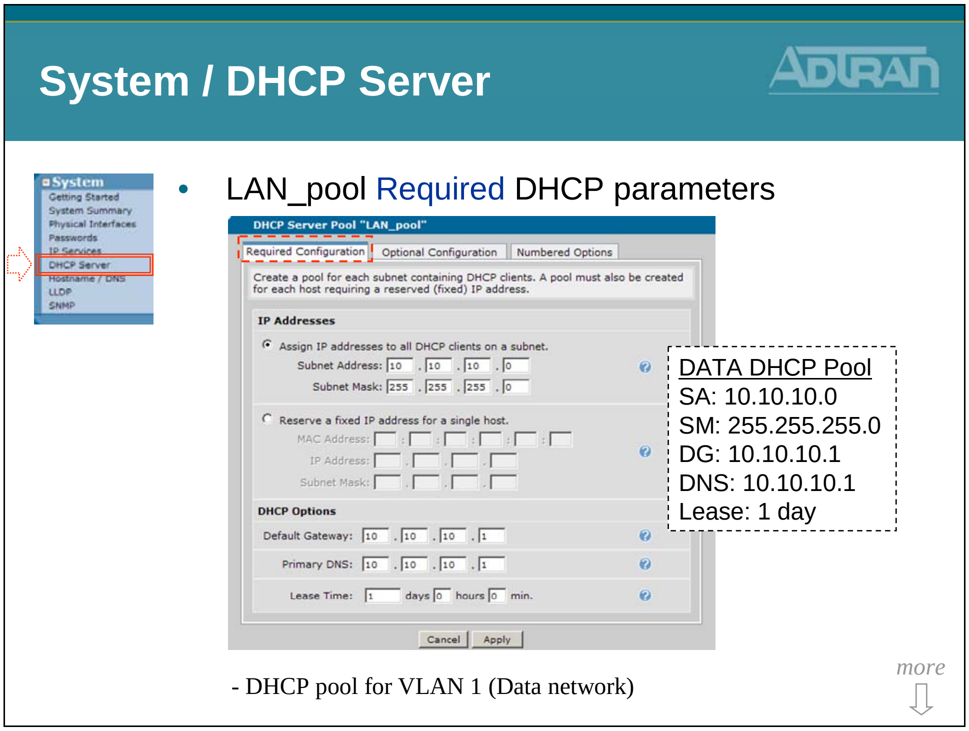
If there is an existing DHCP server that you wish to use, there are a couple of options:

a. Remove the default DHCP server for VLAN 1 (typically for PCs on the LAN) and leave the default DHCP server for VLAN 2 (used by IP Phones)

b. Remove the DHCP Server pools for both VLAN 1 and VLAN 2 and allow the existing DHCP server to service both VLANs

   Note: If the NetVanta 7060/7100 DHCP server is not used, DHCP Options (66 and 157) will need to be configured on the existing DHCP server. Review the default configuration of the DHCP server pools for details and syntax.

DHCP Server Pool – Required Configuration



The DHCP Server pool for VLAN 1, the data network, provides IP addresses from the 10.10.10.0 /24 network. Untagged traffic that enters a Switchport will be assigned to the native VLAN, VLAN 1 by default. Since the IP address assigned to interface VLAN 1 falls in the subnet 10.10.10.0 /24, it uses the DHCP pool LAN_Pool.

REQUIRED DHCP CONFIGURATION

IP Address Subnet

The IP addresses on the assigned subnet that are NOT excluded will be assigned to clients.

A Pool can be created to reserve a fixed IP address for a specific host. Host will always be assigned this IP address and network mask. Typically the MAC address is set to the host's Ethernet adapter MAC address.

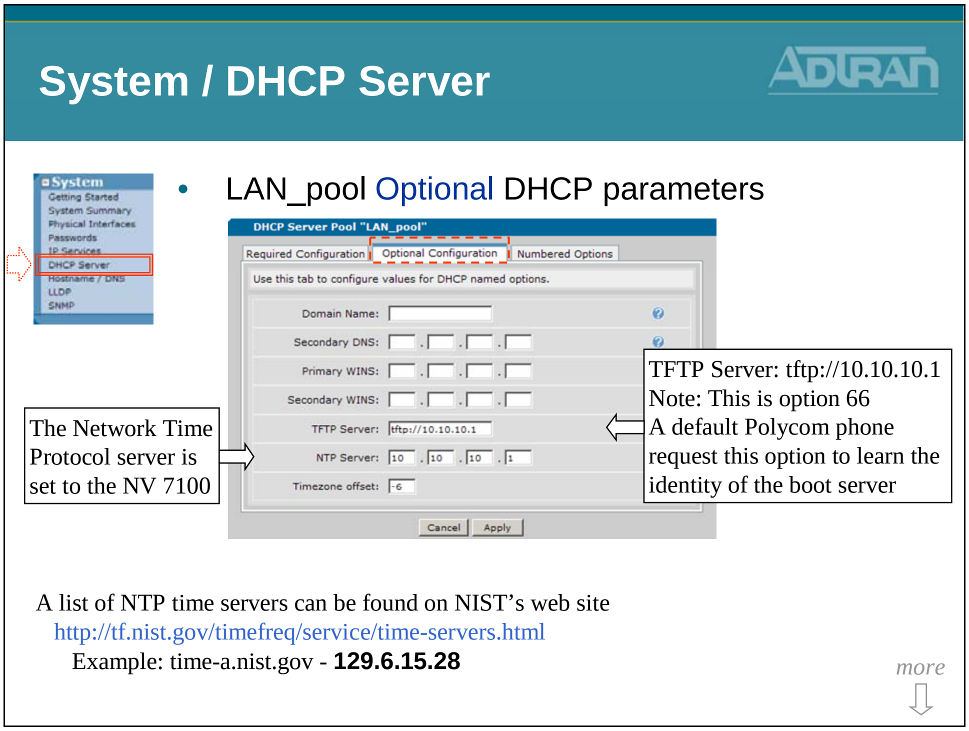
Default Gateway

The default-gateway IP address that the DHCP server will assign to clients. When specifying a router to use, verify that the router is on the same subnet as the DHCP client. Typically, the default-gateway should be set to the IP address of an interface on the unit you are configuring.

Primary DNS

If DNS proxy is enabled, the unit will forward DNS requests sent to any of its interface IP addresses to the DNS servers. These servers can be obtained dynamically from an ISP or configured statically on the Hostname/DNS page.

DHCP Server Pool – Optional Configuration



Domain Name
The Domain that the DHCP Clients will be a member of.

Secondary DNS
Clients will use secondary DNS if name resolution with primary fails.

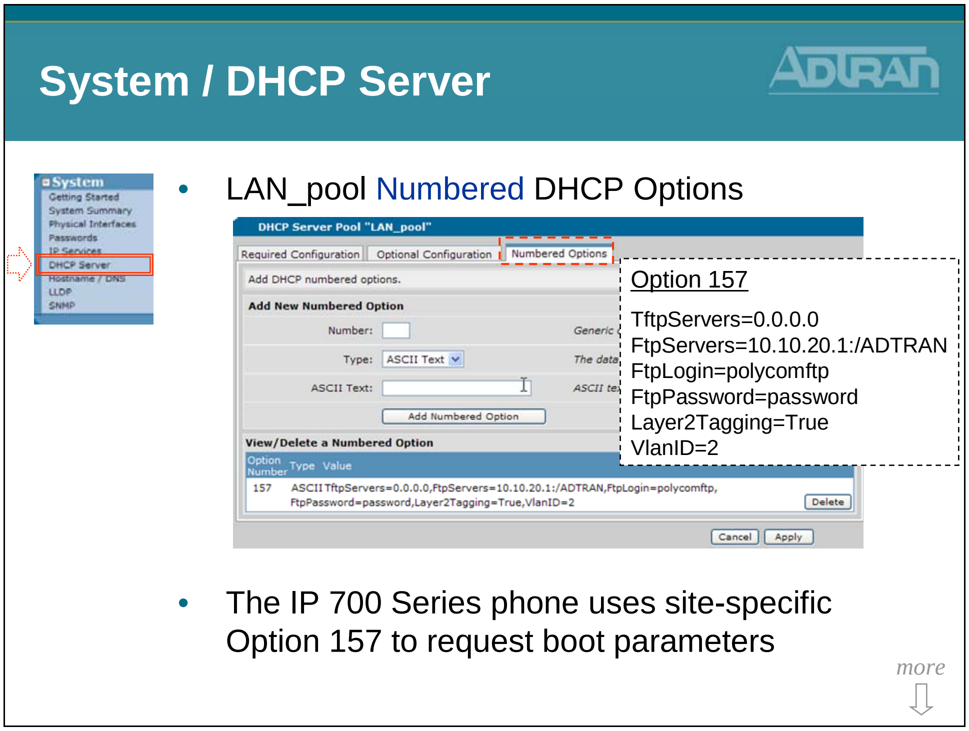
Primary WINS
Needed for Microsoft Networking so clients can resolve NetBIOS names. Clients will typically use secondary WINS if NetBIOS name resolution fails with primary.

TFTP Server
Host name (or address) of the TFTP server given to any requesting DHCP client. The default value of tftp://10.10.10.1 is used by factory default Polycom phones during the initial boot. A boot files tell the Polycom phone to use FTP after initial boot.

NTP Server
Network Time Protocol IP address served to a DHCP client. By default, the NetVanta 7XXX is the NTP server for LAN clients. The public time server used by the NetVanta 7XXX is configured from the System/Summary menu.

Timezone offset
Timezone offset in hours (-12 to 12). There are 25 integer World Time Zones from -12 through 0 (GMT) to +12. Each one is 15° of Longitude as measured East and West from the Prime Meridian of the World at Greenwich, England. Set for your region.

DHCP Server Pool – Numbered Options



DHCP numbered options describe a generic DHCP option to be published to the DHCP client. The admin may specify any number of generic options to be published to the client.

Number
Generic option number. Valid values are 0-255.

Type
The data type for the numbered option:

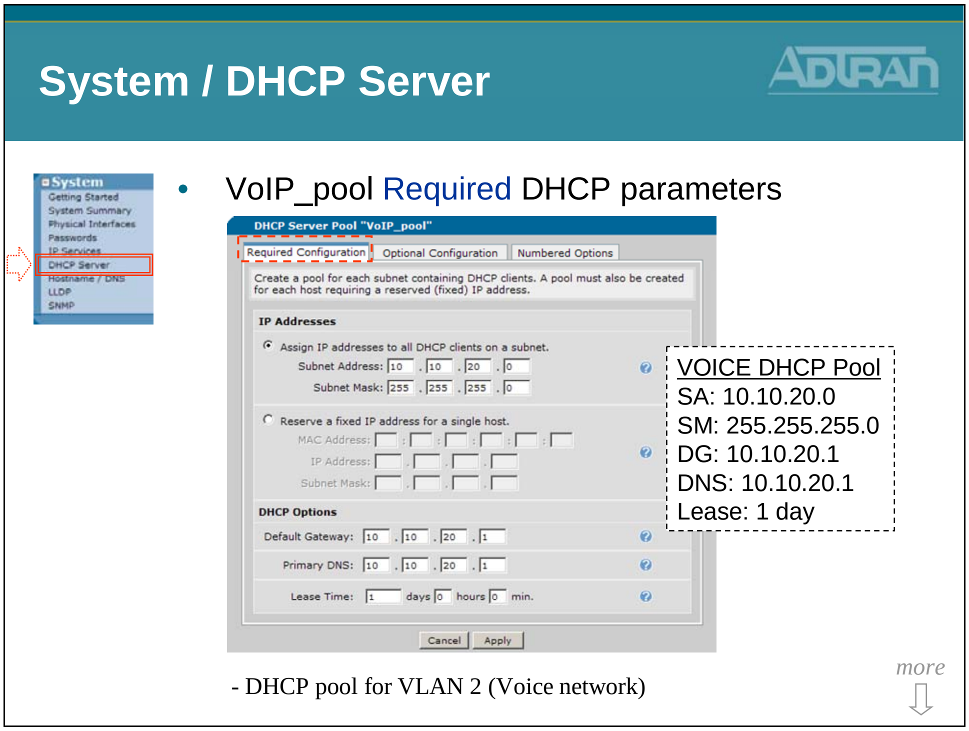
- Ascii Text
- Hex
- IP Address

ASCII Text
ASCII text data for the option.

The IP 700 Series phone uses site-specific Option 157 to provide the following information to the phones: TftpServers=0.0.0.0, FtpServers=10.10.20.1:/ADTRAN, FtpLogin=polycomftp, FtpPassword=password, Layer2Tagging=True, VlanID=2

** Option 157 must be set on both the LAN_pool and the VoIP_pool to direct the phones to the correct boot server.*

System/DHCP Server

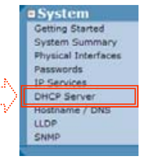


The DHCP Server pool for VLAN 2, the voice network, provides IP addresses from the 10.10.20.0 /24 network. Generally, IP phones will learn and tag voice traffic with a VLAN ID of 2. Since the IP address assigned to interface VLAN 2 falls in the subnet 10.10.20.0 /24, it uses the DHCP pool VoIP_Pool.

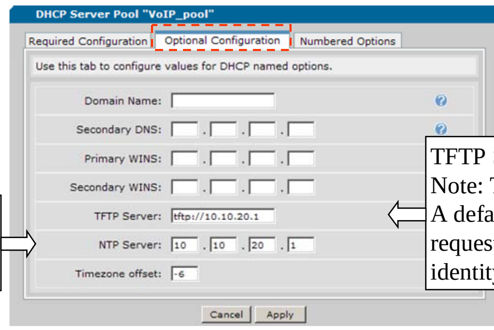
Other than IP addresses, the DHCP server pools LAN_Pool and VoIP_Pool are identical.

System/DHCP Server

System / DHCP Server



- VoIP_pool Optional DHCP parameters



The Network Time Protocol server is set to the NV 7100

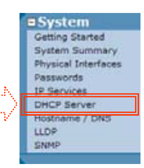
TFTP Server: tftp://10.10.20.1
Note: This is option 66
A default Polycom phone request this option to learn the identity of the boot server

A list of NTP time servers can be found on NIST's web site
http://tf.nist.gov/timefreq/service/time-servers.html
Example: time-a.nist.gov - **129.6.15.28**

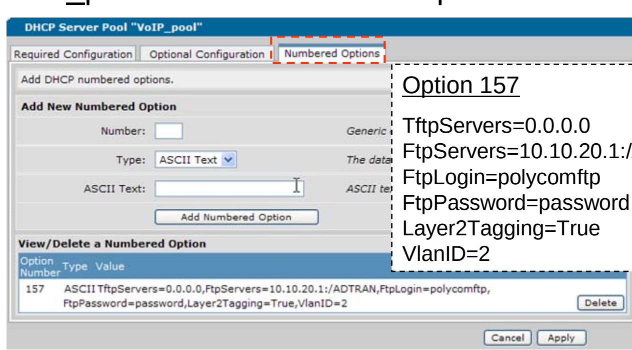
more

System/DHCP Server

System / DHCP Server



- VoIP_pool Numbered DHCP Options

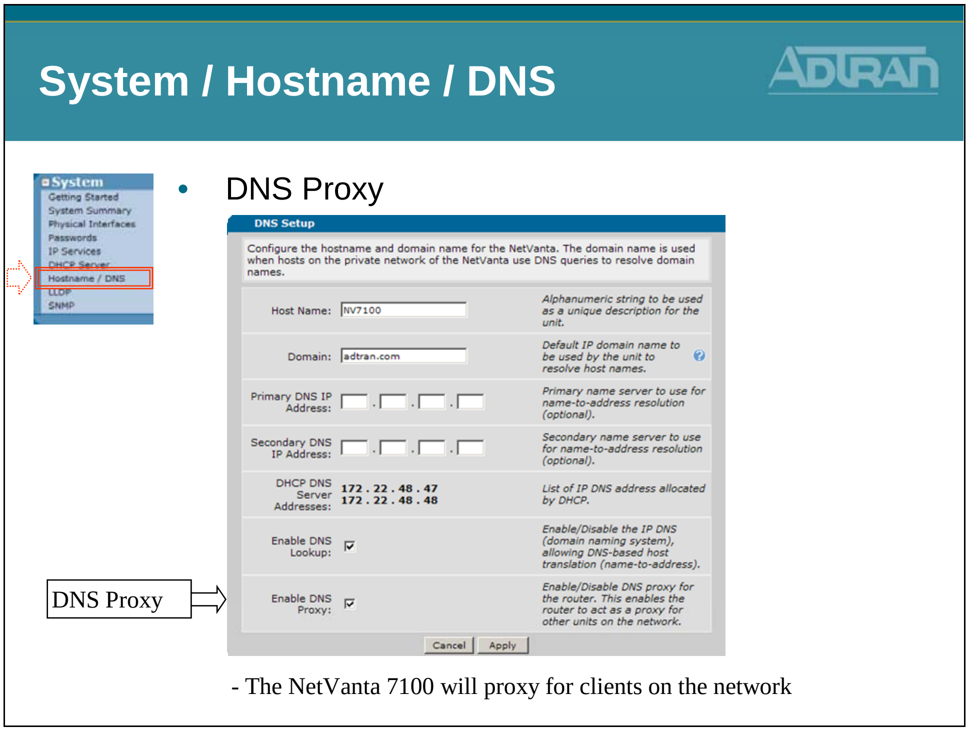


Option 157

TftpServers=0.0.0.0
FtpServers=10.10.20.1:/ADTRAN
FtpLogin=polycomftp
FtpPassword=password
Layer2Tagging=True
VlanID=2

- The IP 700 Series phone uses site-specific Option 157 to request boot parameters

System/Hostname/DNS



Host Name
Alphanumeric string to be used as a unique description for the unit.

Domain
Default IP domain name to be used by the unit to resolve host names.

Primary /Seconday DNS IP Address:
Primary/Seconday name server to use for name-to-address resolution (optional).

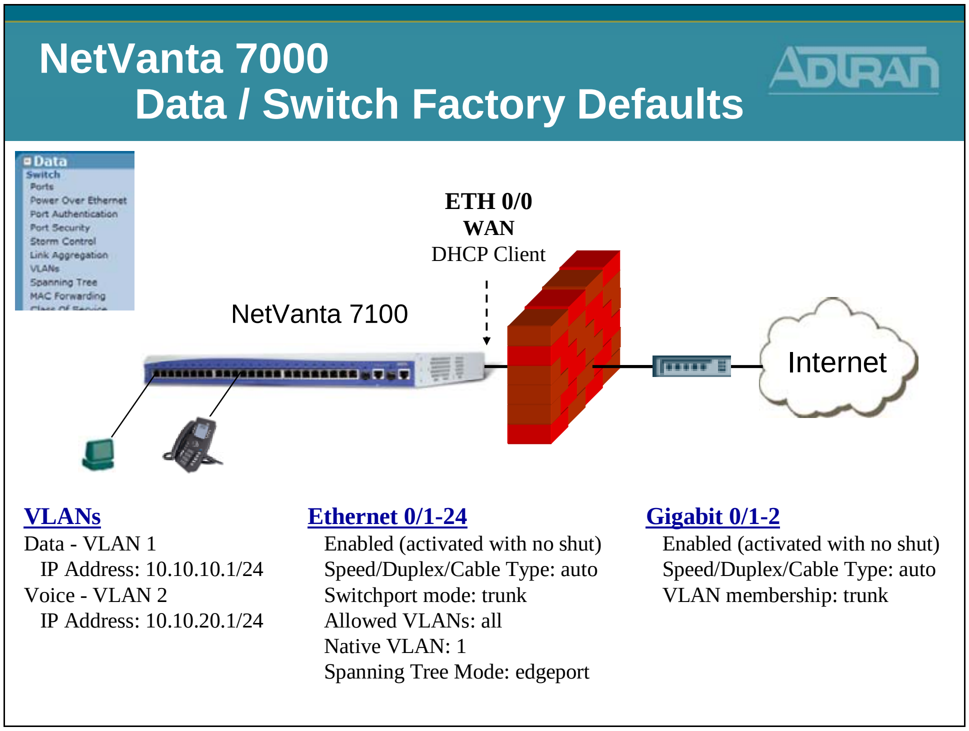
DHCP DNS Server Addresses:
List of IP DNS address allocated by DHCP.

Enable DNS Lookup:
Enable/Disable the IP DNS (domain naming system), allowing DNS-based host translation (name-to-address).

Enable DNS Proxy
By default, DHCP clients send DNS request to the NetVanta 7XXX. With DNS Proxy enabled, The NetVanta 7XXX will forward the DNS request to the DNS server it learned on it WAN. The Ethernet 0/0 WAN interface is configured as a DHCP client by default.

If the NetVanta 7XXX DHCP pools are configured with the ISPs DNS server IP address, DNS Proxy can be disabled.

NetVanta 7000 Data/Switch Factory Defaults



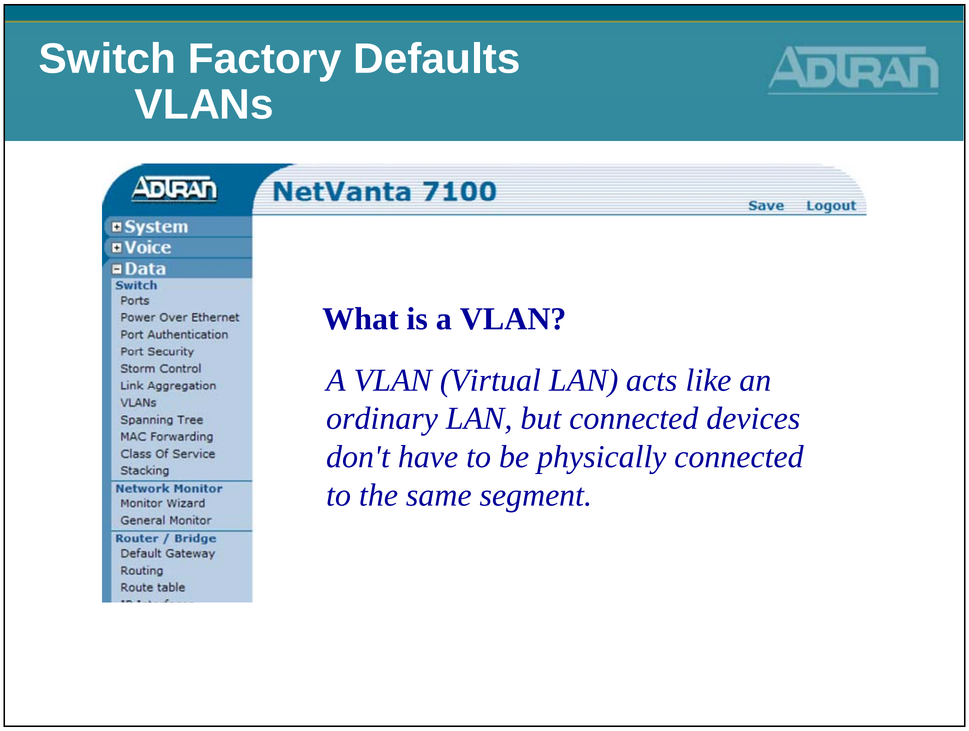
VLAN 1 is defined with an IP address of 10.10.10.1 255.255.255.0
VLAN 2 is defined with an IP address of 10.10.20.1 255.255.255.0

It is often necessary to change the VLAN IP address scheme on a NetVanta 7100 from its factory default settings. This is usually done at the request of the customer so that the NetVanta 7100 can reside in an existing network without requiring changes to devices currently running on that network.

If changing the current IP scheme, additional settings will need to be applied in order to have proper phone operation when VLAN subnet changes have been applied. Include the following areas when making your IP changes:

- DHCP Pools
- IP Phone Config – Boot Settings tab
- IP Phone Configs – Default Settings tab
- Firewall Policies

Switch Factory Defaults VLANs

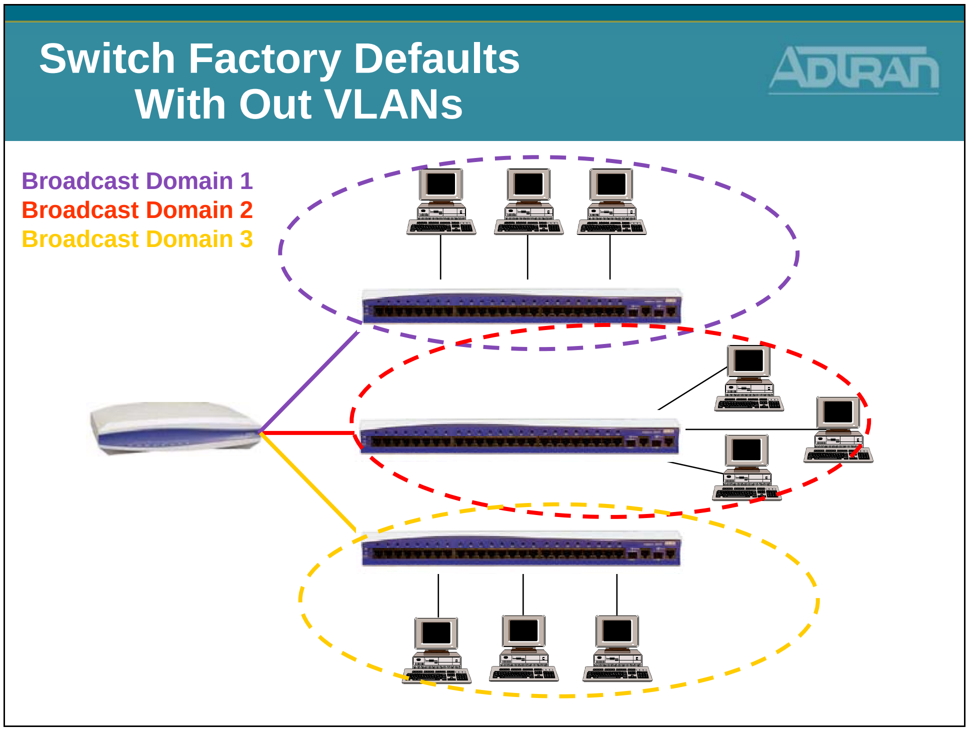


Virtual Local Area Network (VLAN)

Routers, computers and other data devices have the ability to send a type of message known as a “broadcast message”. Broadcast messages are sent to every device or node within a given network or subnetwork. Common functions of broadcast messages are to identify when network devices are enabled and available, to advertise services, and to request address resolution. Many of these types of messages are vital to network operation. Yet, the frequency of these messages and the number of devices on a network transmitting these messages could cause network congestion. Unlike collision domains, which may be divided based on Layer 2 MAC Addresses, broadcast domains typically exist at the logical or network layer of the OSI model. An example of this is when a broadcast message is defined for the broadcast address (10.10.10.255) of the (10.10.10.0/24) network. A Layer 2 switch would forward this message (or IP packet) out all switch ports, as it does not know which end devices are members of the 10.10.10.0/24 network. A router is the device that recognizes this.

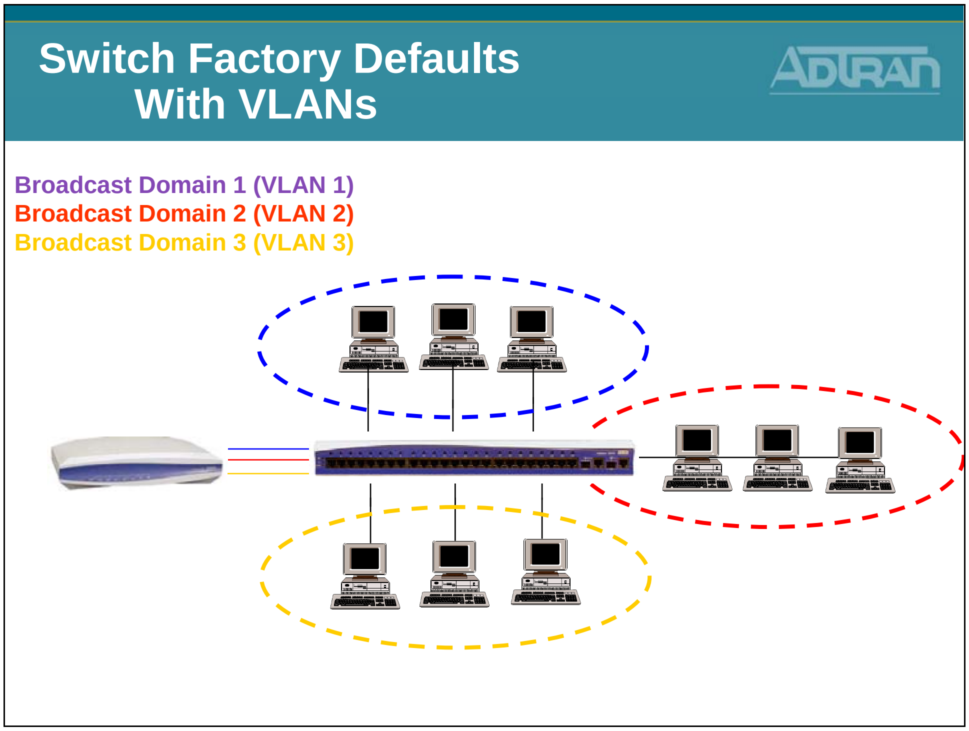
So, the question exists, how would a switch break up broadcast domains? Or, is this function only available in a Layer 3 device such as a router?

Switch Factory Defaults With Out VLANs



In a single Layer 2 switch, without the use of virtual local area networks (VLANs), this function is not possible. Separate switches create separate broadcast domains so that broadcast messages from attached devices do not get sent to devices attached to the other switches, unless sent through the router. Every device connected to a single switch will receive all broadcast traffic generated by any end device connected to that same switch. This is not the most streamlined or cost-effective approach to designing a network. Purchasing switches simply to break up a broadcast domain, and not based on port density and performance, may lead to wasted switch ports and underutilized resources. An alternative solution is the use of VLANs in a single switch.

Switch Factory Defaults With VLANs



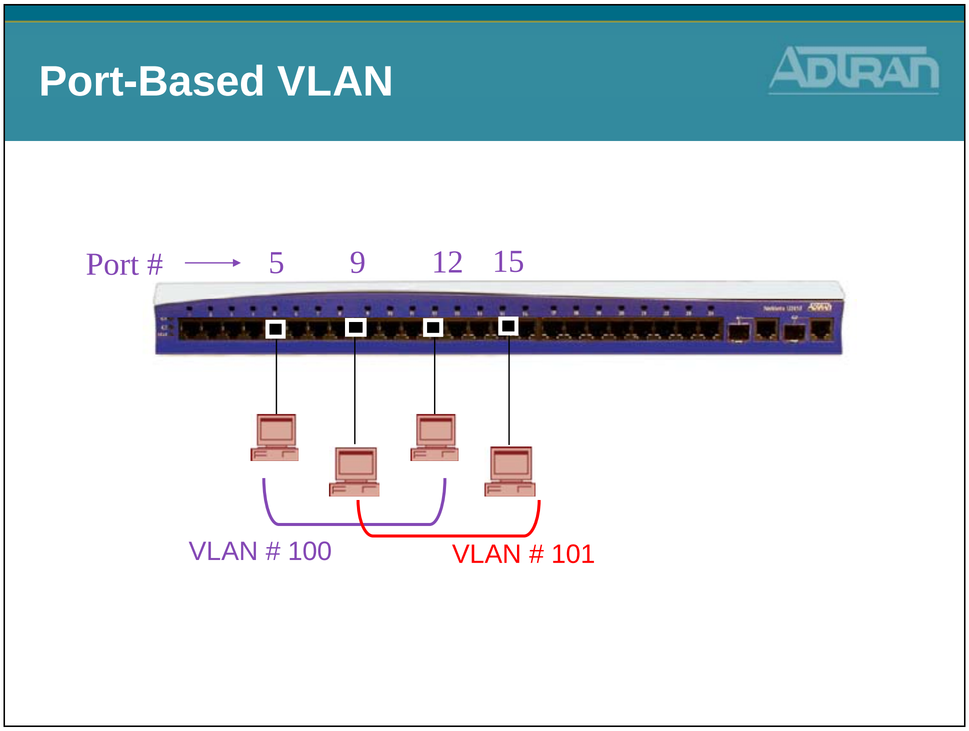
Incorporating VLANs

Basic components of VLANs: A VLAN or Virtual Local Area Network is designed to provide a logical segmentation of devices which may be based on function or application, rather than physical location. VLANs provide the ability to break up broadcast domains in a switch by segmenting the ports of the switch based on their VLAN ID.

Incorporating VLANs into a typical network allows for control and segmentation of that network. By using VLANs, a single switch may accomplish the same task as the previous diagram by creating separate broadcast domains but still allowing inter-vlan routing to occur (provided each switch and VLAN has a connection to the router). Multiple end-user devices may be connected to a single switch but belong to different numbered VLANs. Even though the devices are physically connected to the same switch, they would not be able to communicate without the aid of the router or other layer 3 device. (The router has the ability to route or talk between VLANs.) In essence, a VLAN breaks up a broadcast domain by allowing broadcast messages transmitted by devices that are connected to switchports with a specific VLAN membership ID to only be received by devices connected to switchports with that same VLAN membership ID.

VLANs are able to span devices. Therefore, if trunk communication exists between two switches, devices connected to switchports that have the same VLAN membership ID on both switches are able to transmit and receive traffic within that VLAN without a router present.

Port-Based VLAN

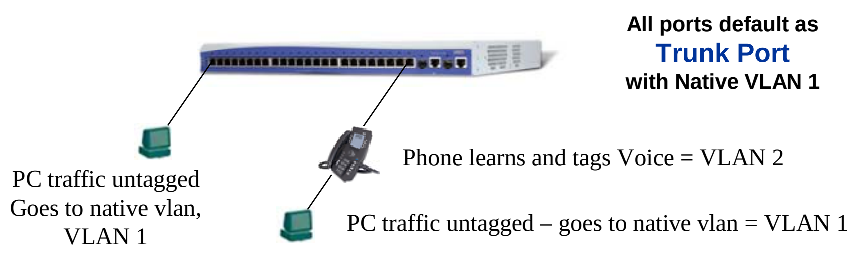


The NetVanta switchports support port-based or static VLANs. Static VLANs are created by manually assigning a VLAN number to a specific interface in configuration. The end-device attached to that interface does not know the VLAN exists, as the switch is responsible for determining which VLAN the traffic came from and then forwarding broadcasts to other members of the same VLAN. Therefore, any device attached to a switchport defined with a specific VLAN ID would be able to transmit messages to other devices that are attached to switchports with the same VLAN ID.

Types of VLAN Ports

Types of VLAN Ports

- Access Port
  - Only a member of 1 VLAN
- Trunk Port
  - Allows multiple VLANs (VLAN Trunking)
  - 802.1Q is the standard supported for VLAN Trunking
  - By default, all active VLANs are allowed to transmit and receive traffic on a trunk port



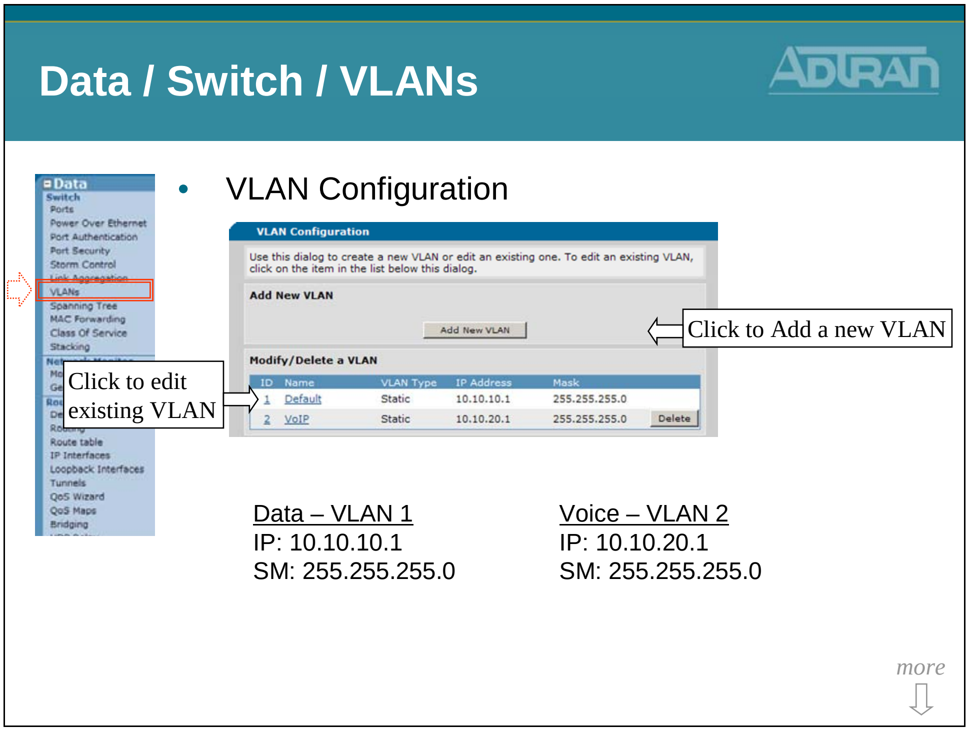
There are two types of VLAN ports that may be configured on the NetVanta: access ports and trunk ports.

Access ports may only be a member of one VLAN. Each switchport may be assigned a single access VLAN. Therefore, if connecting between devices a separate port is needed for each VLAN in access mode. This is a valid application, but will quickly use up available physical interfaces. However, a port may be used to transport multiple VLANs, typically in between switchports of different units or to a Layer 3 device such as a router. This port is known as a “trunk port”.

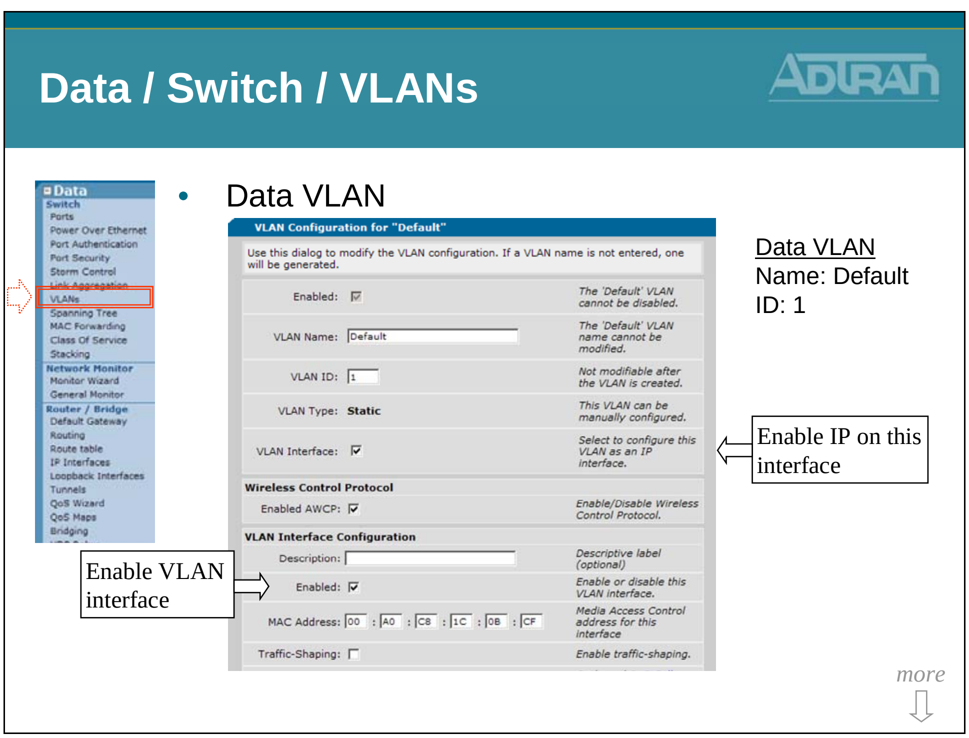
Trunk ports are the other type of VLAN ports that may be configured in a NetVanta switch. A trunk port may carry multiple VLANs across a single interface. Trunk ports are used to connect to other devices that may also need to communicate with those VLANs, or to allow inter-vlan routing.

A trunk is a point to point link that transmits and receives traffic between switches or between switches and routers. Trunks can carry traffic from multiple VLANs and can extend VLANs across an entire network. On a NetVanta unit, any switchport may be used for trunking. The standard for VLAN trunking is defined by the IEEE 802.1Q standard. This is the method that is supported in the NetVanta AOS devices.

Data/Switch/VLANs



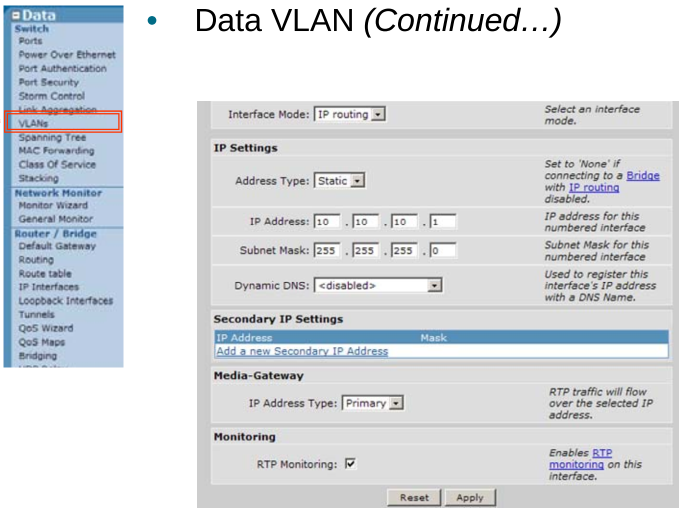
Data/Switch/VLANs



Data/Switch/VLANs

Data / Switch / VLANs

- Data VLAN *(Continued…)*



Data VLAN
Name: Default
ID: 1

Address Type set to Static

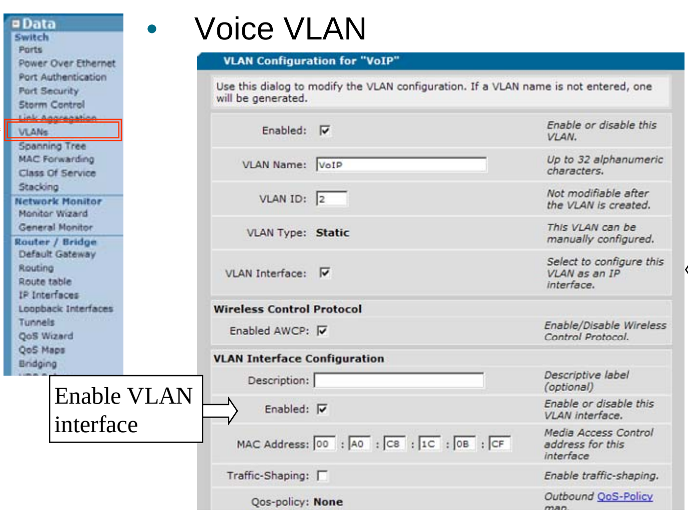
VLAN IP address and subnet mask

Media-Gateway set to Primary

Data/Switch/VLANs

Data / Switch / VLANs

- Voice VLAN



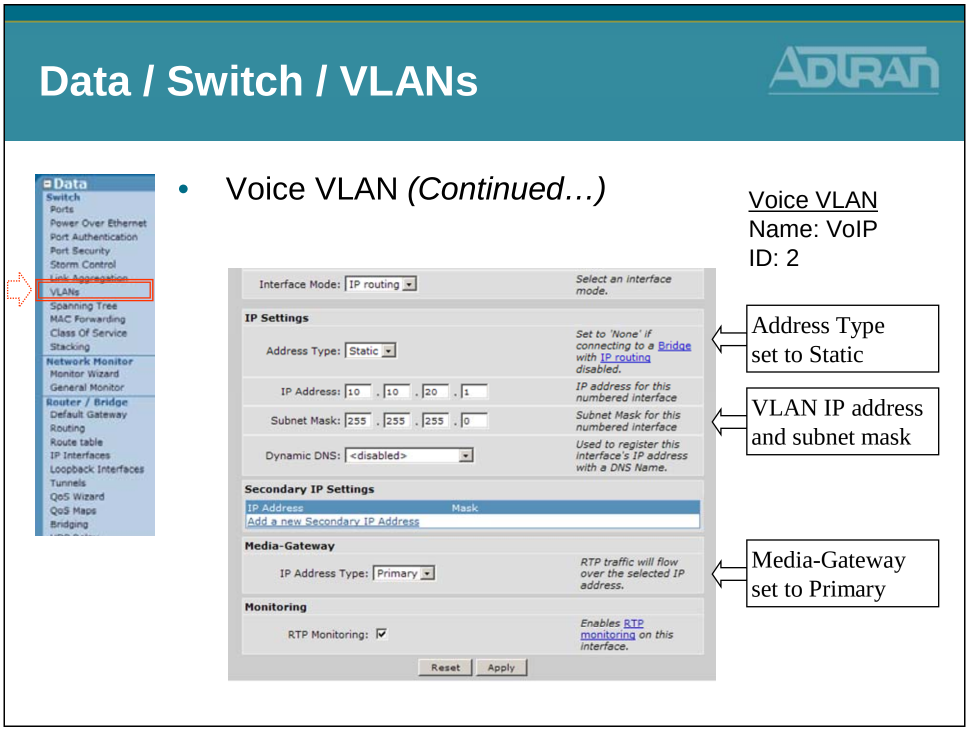
Voice VLAN
Name: VoIP
ID: 2

Enable IP on this interface

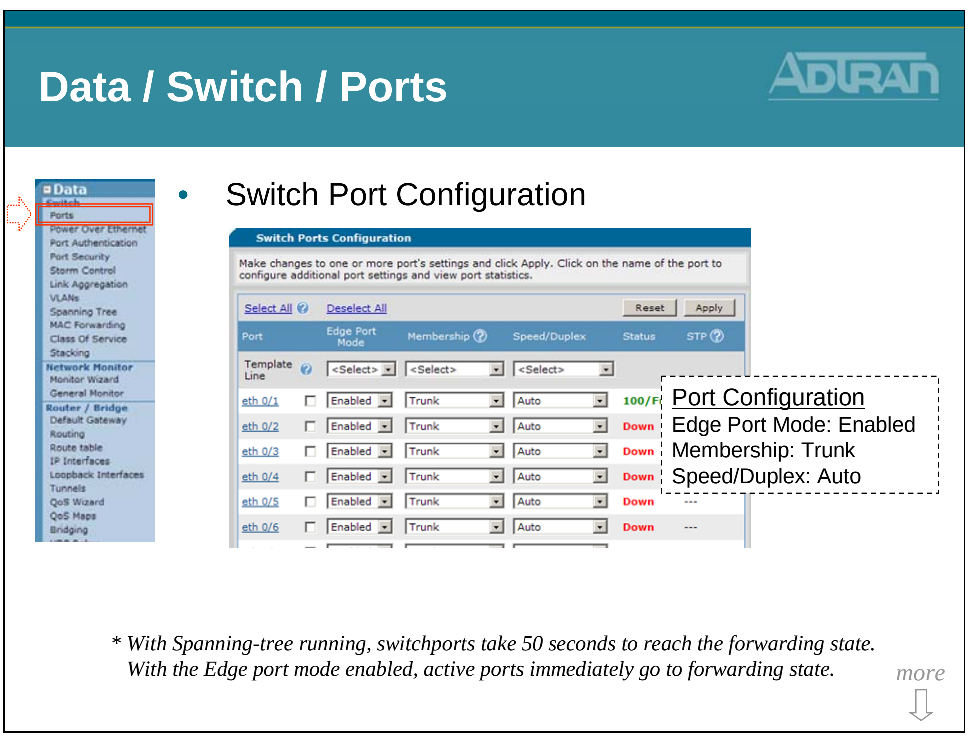
Enable VLAN interface

more

Data/Switch/VLANs



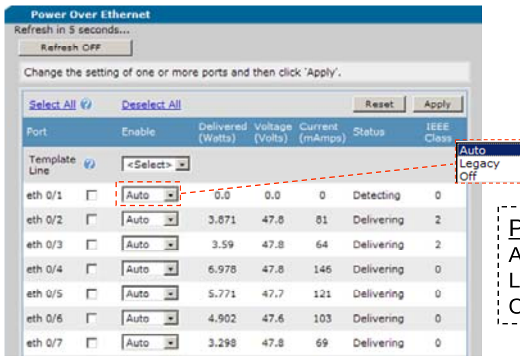
Data/Switch/Ports



Data/Switch/Power Over Ethernet

Data / Switch / Power Over Ethernet

- Power Over Ethernet



Power Options
Auto: Detect 802.3af
Legacy: Non 802.3af
Off: Power disabled

  - The default setting of Auto will detect attached Powered Devices (PDs) and deliver 48 VDC, compliant with the IEEE 802.3af power-over-Ethernet standard

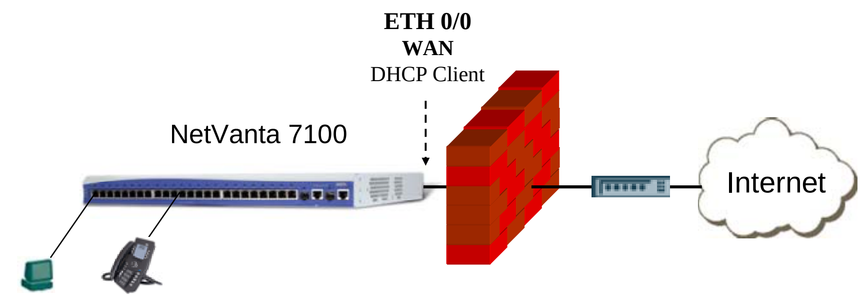
Power over Ethernet (PoE) technology provides the ability to detect attached Powered Devices (PDs) and deliver 48 VDC to the PD via existing CAT5 cabling. The NetVanta 7000 units are fully compliant with the power delivery options called out in the IEEE 802.3af Power over Ethernet specification. By default, the PoE interfaces discover and provide power to IEEE compliant PDs.

To disable power detection and supply, change the PoE port setting to Off. This can also be used as a quick toggle to power cycle phones. Remove power, click apply to remove power. Then change setting back to Auto and then click Apply to restore power to phone.

The Legacy option, enables power detection and supply of legacy non-IEEE 802.3af compliant PDs.

NetVanta 7000 - Router Factory Defaults

NetVanta 7000 Router Factory Defaults

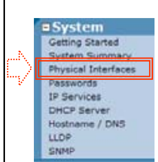


VLANs
Data - VLAN 1
IP Address: 10.10.10.1/24
Voice - VLAN 2
IP Address: 10.10.20.1/24

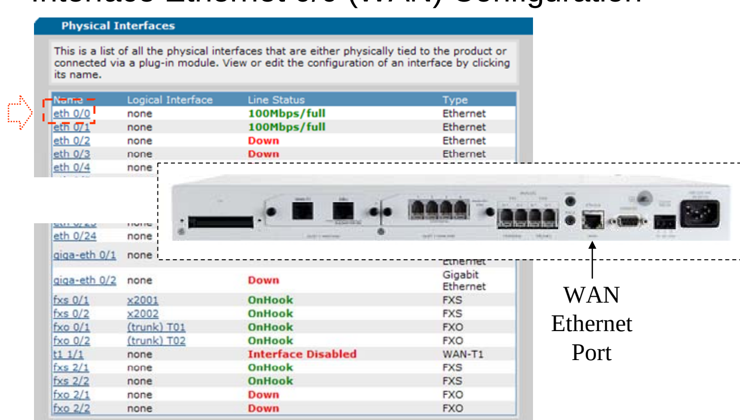
Ethernet 0/1-24
Switchport mode: trunk
Allowed VLANs: all
Native VLAN: 1

WAN Ethernet 0/0

WAN Ethernet 0/0



- Interface Ethernet 0/0 (WAN) Configuration



- *Click to edit interface eth 0/0*

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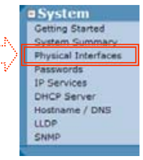
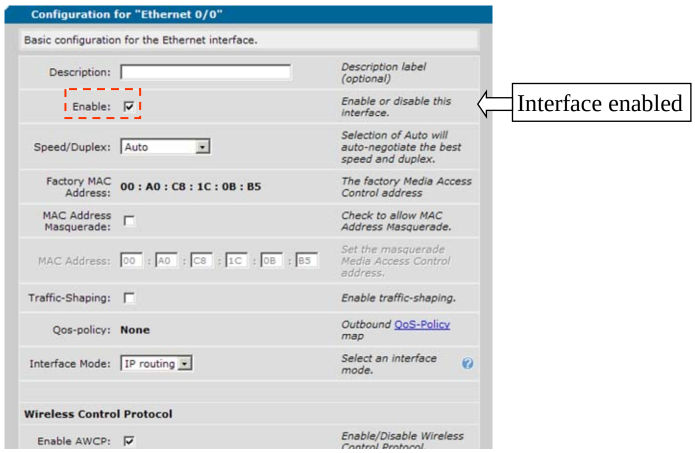
WAN Ethernet 0/0

WAN Ethernet 0/0



- Interface Ethernet 0/0 (WAN) Configuration



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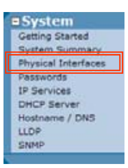
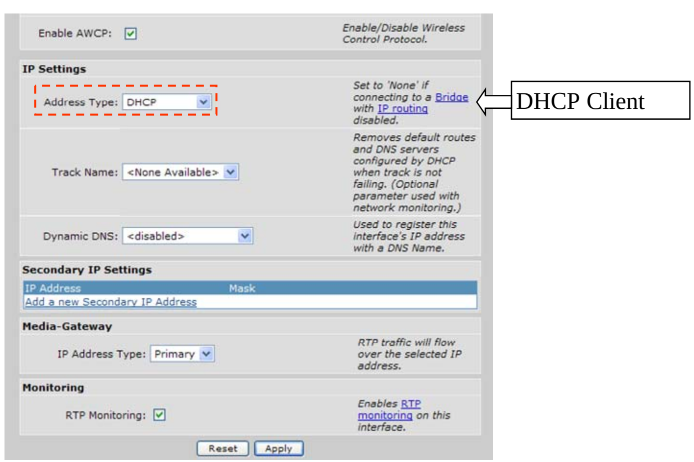
WAN Ethernet 0/0

WAN Ethernet 0/0

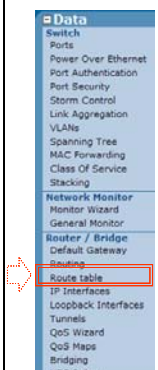


- Interface Ethernet 0/0 (WAN) Configuration

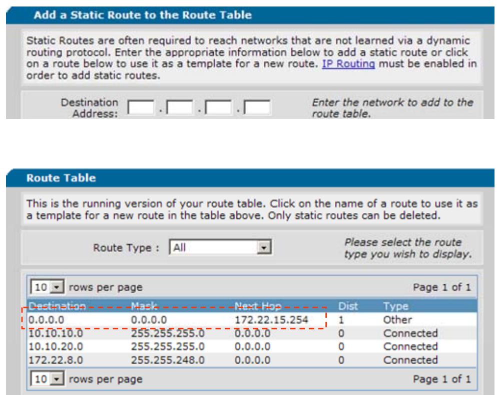


Data / Router / Route table

Data / Router / Route table



- Default Route to ISP



** From the factory, interface Ethernet 0/0 is configured as a DHCP client.*
The default route is learned from the ISP by default.

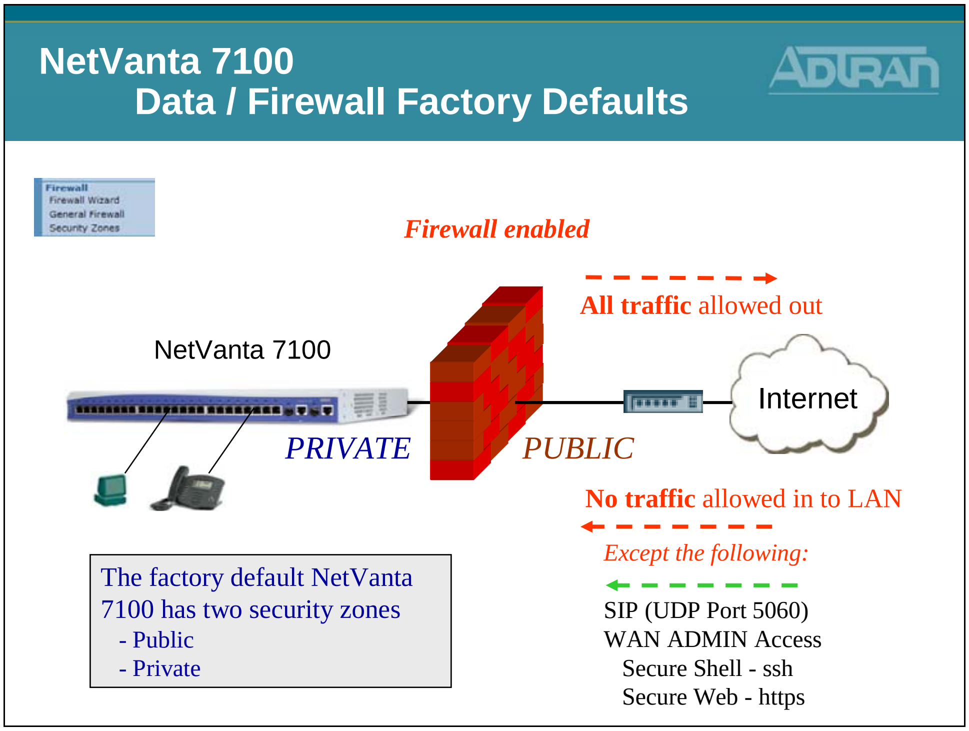
From the factory, interface Ethernet 0/0 is configured as a DHCP client. Not only does the interface get assigned an IP address, it also receives a default route and the primary DNS server.

If interface Ethernet 0/0 is being assigned a static IP address, you must manually define the default route used by the NetVanta 7XXX.

To Configure a Default Route, set following:

Destination Address:	0.0.0.0
Destination Mask:	0.0.0.0
Gateway:	Enter next hop (gateway) or Select WAN interface

NetVanta 7100 - Data/Firewall Factory Defaults



The factory default NetVanta 7100 allows (and NATs) all traffic going to the internet. UDP port 5060 SIP traffic, secure shell, and secure web traffic are the only traffic allowed in the PUBLIC interface by default.

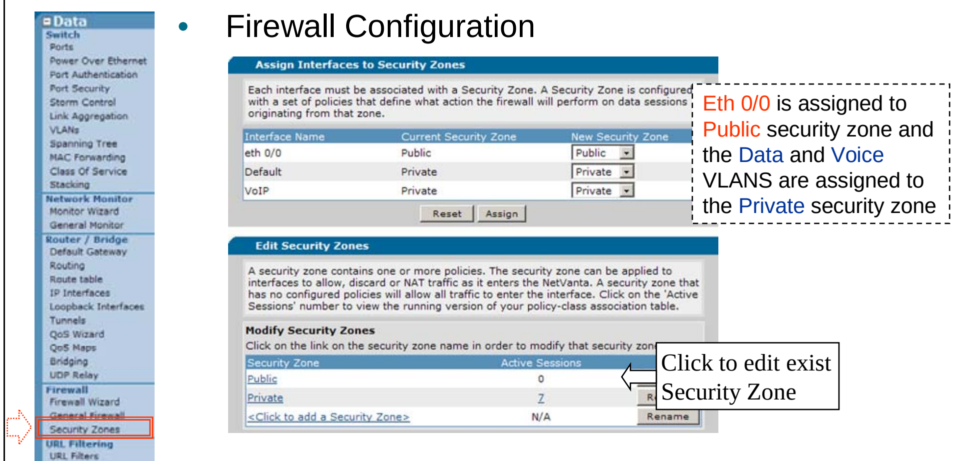
The NetVanta 7100 is equipped with a stateful inspection firewall. A stateful inspection firewall operates by monitoring traffic passing through it. It only allows traffic it is specifically configured to allow as well as return traffic matching traffic that was specifically allowed.

For example, if a computer sends a request to a web site, through the firewall, it is only necessary to configure an allow (NAT) for the outbound traffic, the traffic from the requesting computer to the web server. The response traffic from the website will be automatically allowed. All traffic that has not been initiated from within the network will be automatically blocked unless otherwise specified.

Data/Firewall - Security Zones

Data / Firewall
Security Zones

- Firewall Configuration



- The Factory Default NetVanta 7100 has two security zones (Public and Private)

more

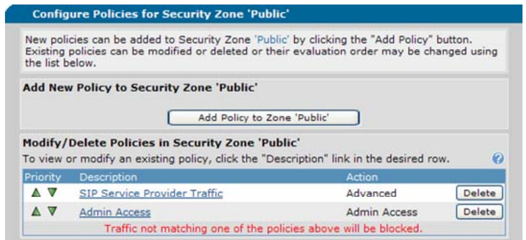
Each interface should be associated with a Security Zone. A Security Zone is configured with a set of policies that define what action the firewall will perform on data sessions originating from that zone. A security zone that has no configured policies will allow all traffic to enter the interface.

The Public and Private Security Zone listed above are present with the factory delivered NetVanta 7100. The firewall inspects traffic inbound. To control traffic coming from the Internet, modify the Public Security Zone. To control traffic coming from VLAN 1 or VLAN 2, modify the Private Security Zone.

Data/Firewall - Public Security Zone

Data / Firewall
Public Security Zone

- Access from Outside the NetVanta 7100

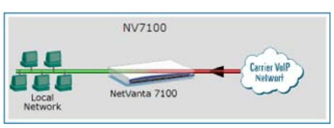


Top down processing

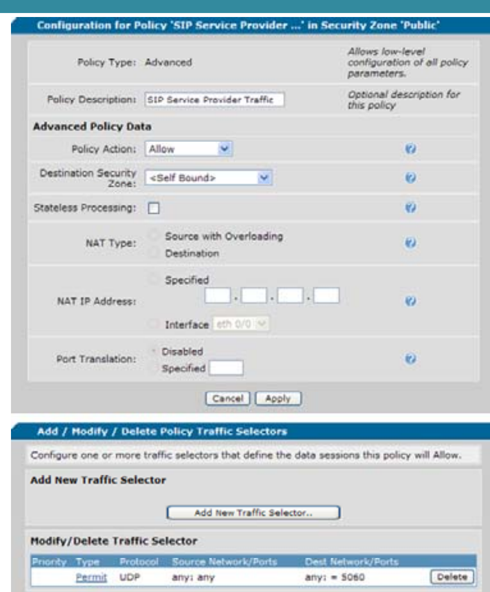
- – SIP Service Provider Traffic
  - Allow SIP (UDP 5060) traffic in
- – Admin Access
  - Allow allows https and ssh access from the Public security zone

Public Security Zone - SIP Service Provider Traffic

Public Security Zone
SIP Service Provider Traffic

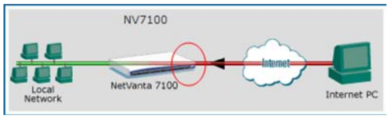


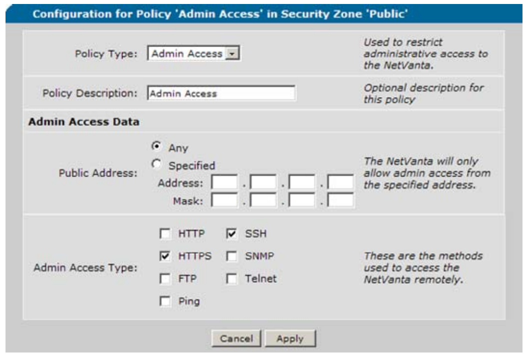
- The SIP Service Provider policy allows SIP – UDP port 5060 from anywhere in to the NetVanta 7100
  - – If this is truly from a SIP service provider, the traffic selector should be modified defining the source as the specific IP address of service provider



Public Security Zone – Admin Access

Public Security Zone
Admin Access

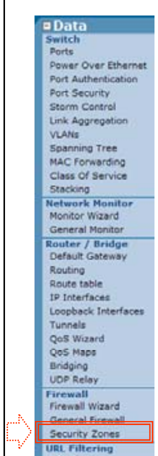




- The Admin Access policy allows https and ssh access from the Public security zone by default
  - Can be more specific
  - Could also allow other admin traffic such as:
    - HTTP, FTP, SNMP, Telnet, and Ping

Data/Firewall - Private Security Zone

Data / Firewall
Private Security Zone



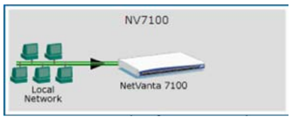
- Access from LAN

Top down processing

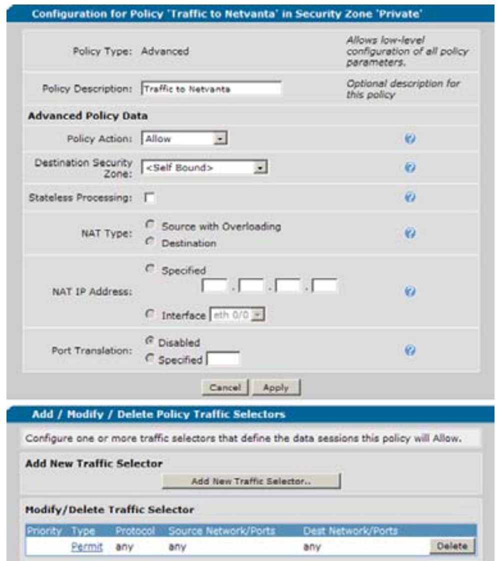
  - Traffic to NetVanta
    - Inside traffic with destination of NetVanta 7100 allowed
  - Voice / Data VLAN Traffic
    - Allow VLAN to VLAN traffic
  - NAT list NAT
    - Allow all traffic going to Internet

Private Security Zone – Traffic to NetVanta

Private Security Zone
Traffic to NetVanta

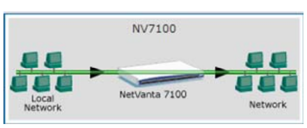


- Inside traffic with a destination of the NetVanta 7100 is allowed
  - Examples:
    - SIP
    - RTP
    - DHCP
    - TFTP
    - FTP

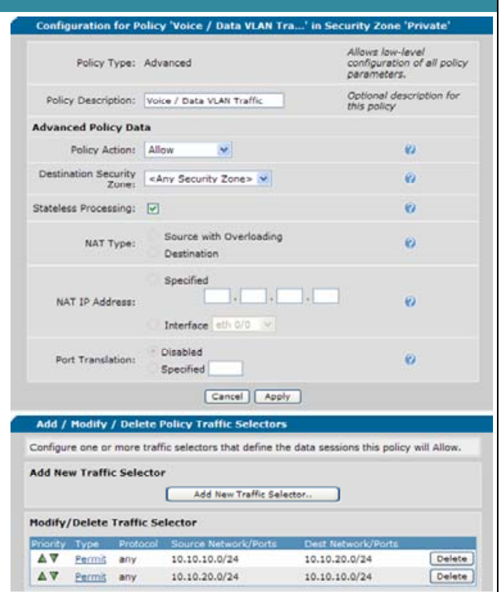


Private Security Zone – Voice / Data VLAN Traffic

Private Security Zone
Voice / Data VLAN Traffic

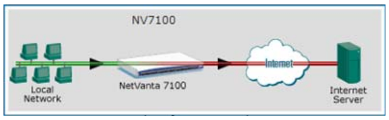


- Allow VLAN to VLAN traffic
- Required if you want to allow the following:
  - PC with Softphone to call a SIP hard phone
  - PC to access WEB GUI of an IP phone

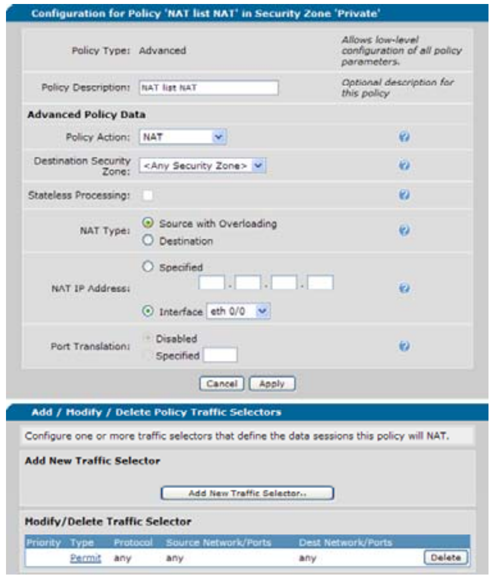


Private Security Zone – NAT list NAT

Private Security Zone
NAT list NAT



- Allow all traffic going to Internet
  - Traffic selectors matches all traffic
  - Outbound traffic is translated from the private inside IP address to the public IP address assigned to the outgoing interface



Module Objectives

Module Summary

At the end of this module, you should be able to:

- Perform basic navigation in the ADTRAN OS
- Navigate the NetVanta 7000 Web-Based GUI
- Understand System Factory Defaults
- Understand Switch Factory Defaults
- Understand Router Factory Defaults
- Understand Firewall Factory Defaults

Module 3: Introduction to NetVanta 7000 Series Voice Configuration

Module Objectives

Module Objectives

- Introduce the NetVanta 7000 Switchboard
- Voice Settings
  - Dial Plan
  - Classes of Service
- Voice Stations
  - User Accounts
  - Ring Group
  - Operator Group
- Voice Trunks
  - Trunk Introduction
  - Analog Voice Trunk Configuration

NetVanta 7000 Switchboard – Call Routing

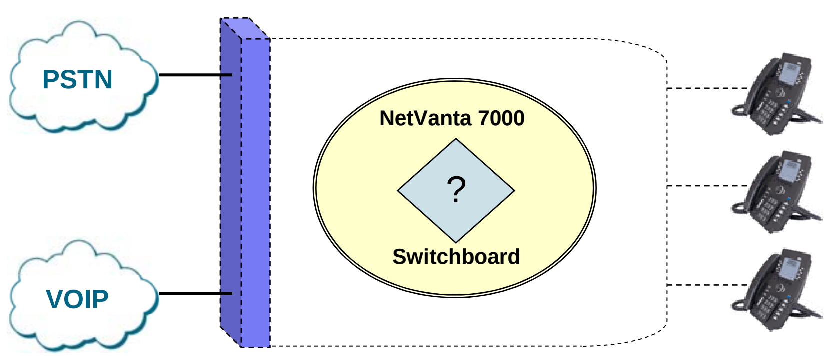
NetVanta IP Telephony Course

NetVanta 7000 Switchboard
Call Routing

NetVanta 7000 Switchboard

NetVanta 7000 Switchboard

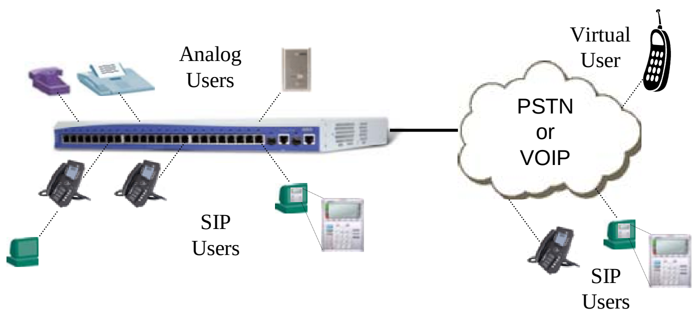
- The switchboard in AOS voice products ensures that inbound and outbound phone calls are routed to the proper voice station, trunk, or other extension



Voice - Stations

Voice – Stations

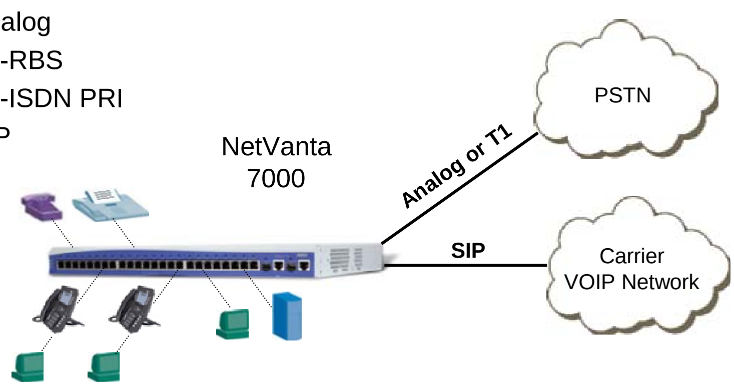
- The Voice Station “User Accounts” are the individual voice users on the system
- Supported Voice Station Phone Types:
  - SIP
  - Analog
  - Virtual



Voice - Trunks

Voice - Trunks

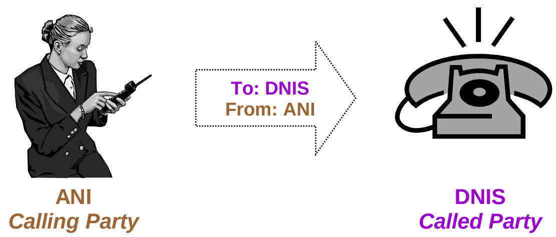
- **Trunk** lines connect the NetVanta 7000 to the outside world. They are delivered from the carrier and may be digital or analog.
- Supported Voice Trunk Types
  - Analog
  - T1-RBS
  - T1-ISDN PRI
  - SIP



ANI/DNIS

NetVanta 7000 Switchboard ANI/DNIS

- By default, the NetVanta 7000 makes call routing decisions based on DNIS
  - ANI - Automatic Number Identification
    - the calling party's information
    - typically represents the caller's number
  - DNIS - Dialed Number Identification Service
    - the called party's information
    - typically represents the number that the originating caller dialed



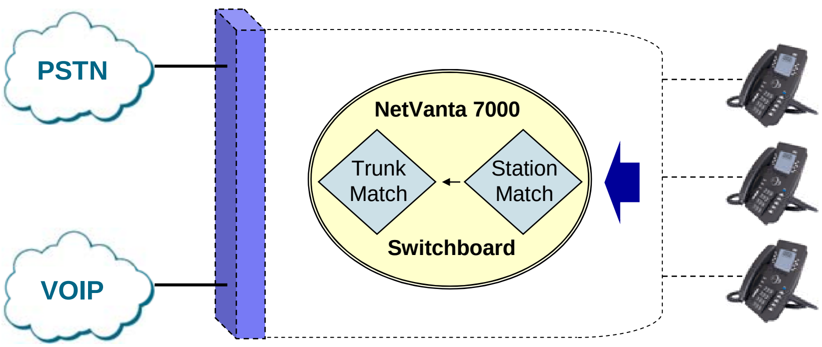
ANI – Automatic Number Identification
ANI is a service that provides the receiver of a telephone call with the number of the calling phone. For example, ANI is used by emergency dispatchers to quickly respond to an emergency when the caller is unable to report their location. The emergency dispatchers are able to use the two parts of ANI to locate the caller and retrieve the caller's telephone number. The two parts of ANI are its information digits and the calling party's telephone number. The information digits designate class of service (CoS) and are transmitted by dual tone multi-frequency (DTMF) tones or in-band multi-frequency (MF) signaling. This information may sound like caller ID, but it is a separate entity that is transmitted with the phone call, even if caller ID blocking is activated, allowing receivers of the information to determine the calling party's phone number and in some cases location.

DNIS – Dial Number Identification Service
Most call routing is based on DNIS. The DNIS system routes calls either locally or through the network based on DNIS matching. In this method of call routing, calls are routed to voice stations based on whether the DNIS of the call matches a call account number, an alias to the call account, or the Session Initiation Protocol (SIP) identity of the call account. If there is a match, the call is routed to that account. DNIS call routing employs weighted DNIS matching, meaning calls with the most exact DNIS match or the lowest cost are routed first.

NetVanta 7000 Switchboard - Call Routing

NetVanta 7000 Switchboard Call Routing

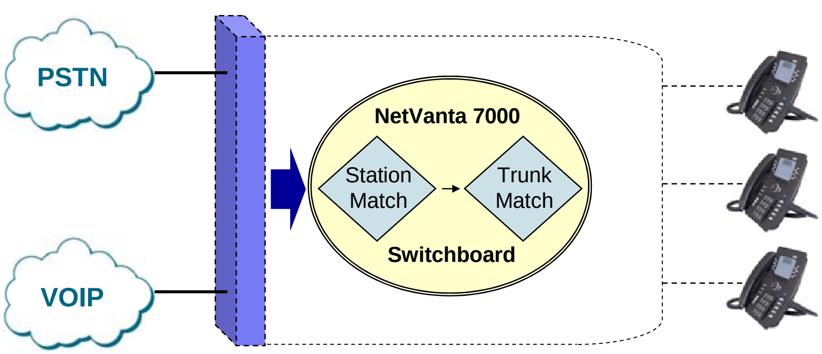
- Switchboard call handling from **Station**
  - SB attempts to send calls to voice stations first, then voice trunk groups



NetVanta 7000 Switchboard - Call Routing

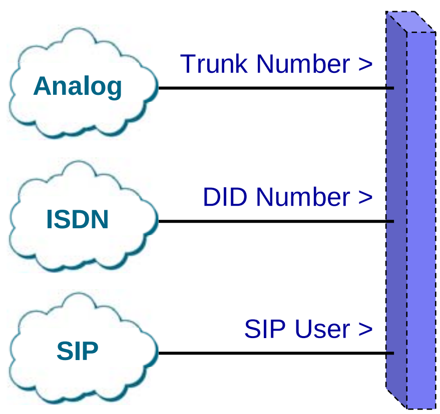
NetVanta 7000 Switchboard Call Routing

- Switchboard call handling from **Trunk**
  - SB attempts to send calls to voice stations first, then voice trunk groups



NetVanta 7000 Switchboard Call Routing

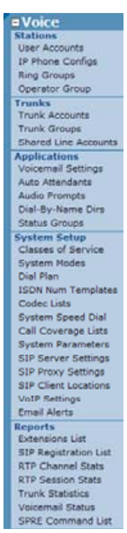
NetVanta 7000 Switchboard Call Routing



- Voice Users
- Ring Groups
- Operator Group (0)
- Mailbox
- Auto Attendant (x8200)
- Voicemail (x8500)
- Leave Voicemail (x8504)
- Overhead Paging (x8000)
- Special Prefix Codes
- Speed Dial
- Alias
- DID

NetVanta 7000 Series – Voice Menus

NetVanta 7000 Series Voice Menus



Voice configuration is primarily accomplished from the NetVanta 7000 Voice menus

The following **topics** are covered in this module:

- Stations - phone users, phone configs, ring and operator groups
  - **Voice Users**
  - **Ring/Operator Group**
- Trunks - Define circuit connected to your carrier's trunk lines
  - **Analog Trunks**
- Applications - Voice applications such as Auto Attendant
- System Setup - Define system voice parameters
  - **Dial Plan**
  - **Classes of Service**
- Reports - Voice statistics

Voice/System Setup – Dial Plan

NetVanta IP Telephony Course

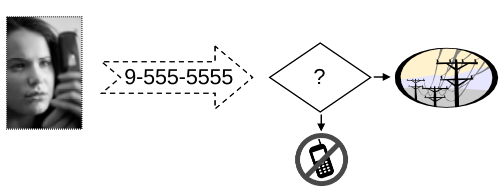
Voice System Setup

Dial Plan

Voice/System Setup - Dial Plan

Voice System Setup
Dial Plan

- The Dial Plan tells the NetVanta 7000 how to route calls and assigns a dialing type to a given number template
  - For example, what is an extension, local number, long distance number, etc..
- The defined dialing type and number template works in conjunction with Classes of Service
  - Determines whether a user has permission to dial a given number



Voice/System Setup - Dial Plan

Voice System Setup
Dial Plan

ADTRAN

- Valid template characters
  - 0-9 - any single digit matches only itself
  - X - any single digit 0-9
  - N - any single digit 2-9
  - M - any single digit 1-8
  - [ ] - any single digit defined within bracket
  - $ - any number string dialed
  - - () , - punctuation characters that are ignored

Valid Template Characters

The valid template characters are: 0-9 , () - M N X [] $

0-9	- any single digit matches only itself
X	- any single digit 0-9
N	- any single digit 2-9
M	- any single digit 1-8
[ ]	- any single digit of those within the bracket
$	- any number string dialed
- () ,	- punctuation characters that are ignored

Examples:	MXXX	- match digits 1000 to 8999
	963-81XX	- match 963-8100 to 963-8199
	963-812[0,1,2]	- match 963-8120 to 963-8122
	963-$	- match all numbers that start with 963

Voice/System Setup - Dial Plan

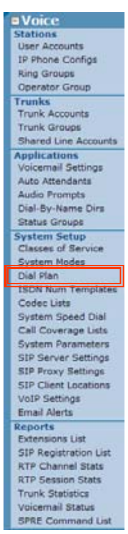
Voice System Setup
Dial Plan

Default Templates

911, 9-911	- Always Permitted
0	- Internal Operator
MXXX	- Extensions
9-NXX-XXXX	- Local
9-1-NXX-NXX-XXXX	- Long Distance
9-1-800-NXX-XXXX (also 866/877/888)	- Toll Free
9-0-NXX-NXX-XXXX	- Operator Assistance
9-011-$	- International
Undefined	- Specified Carrier
Undefined	- 900 Calls

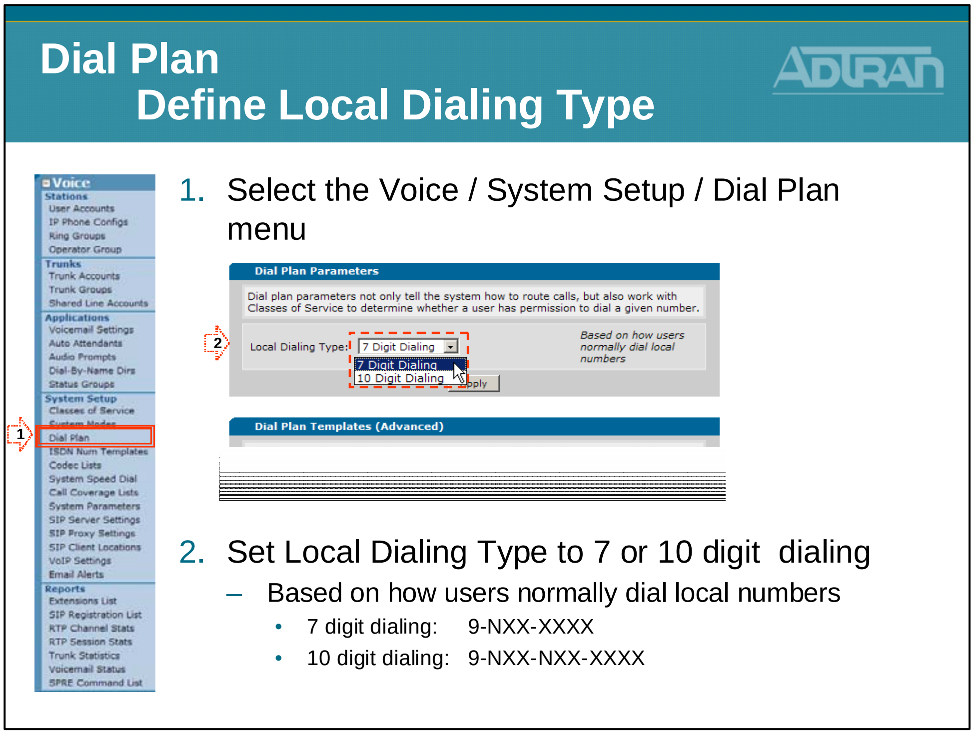
Dial Plan - Configuration

Dial Plan
Configuration



1. Select Voice / System Setup / Dial Plan from the NetVanta 7100 menus
2. Select 7-Digit or 10-Digit dialing based on how users normally dial local numbers
   – 7 digit 9-NXX-XXXX
   – 10 digit 9-NXX-NXX-XXXX
3. In the Dial Plan Template field, enter valid characters for desired number pattern

Dial Plan - Define Local Dialing Type



Setting Local Dialing Type

The Local Dialing Type is default to 7-digits but can easily be changed to 10-digits if required for your area.

1) Select Voice / System Setup / Dial Plan from the NetVanta 7100 menus.

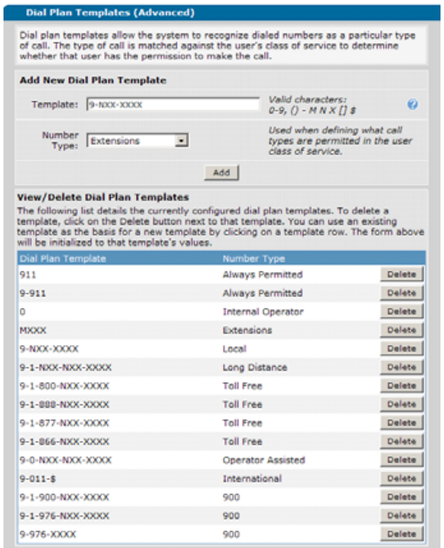
2) Select 7-Digit or 10-Digit dialing based on how users normally dial local numbers.

- If 7 Digit Dialing is selected, the "Local" dial plan number type template is defined as 9-NXX-XXXX

- If 10 Digit Dialing is selected, the "Local" dial plan number type template is defined as 9-NXX-NXX-XXXX

Dial Plan - Define Dial Plan Template

Dial Plan
Define Dial Plan Template



Dial Plan Template

- The default Dial Plan can be modified to fit your calling plan
- To Create a new Dial Plan template
  1. Type new number pattern in the Template field
  2. Specify the Number Type
  3. Click Add
- To modify an existing template, delete the existing template and add a new one

Dial Plan Templates

Dial plan templates allow the system to recognize dialed numbers as a particular type of call. The type of call is matched against the user's class of service to determine whether that user has the permission to make the call.

Create or Modify Dial Plan Template

The dial plan template is used when defining what call types are permitted in the user class of service. It is also used as a number complete match when dialing from analog phones.

1) Select Voice / System Setup / Dial Plan from the NetVanta 7100 menus.

2) In the Dial Plan Template field, enter valid characters for desired number pattern.

3) Select the Number Type that the entered pattern will be associated with. If there is an existing template that matches this number type, and is no longer needed, it can be deleted.

Voice/System Setup - Classes of Service

NetVanta IP Telephony Course

Voice System Setup
Classes of Service

Voice/System Setup - Classes of Service

Voice System Setup
Classes of Service

- A Class of Service (CoS) defines a set of user permissions for making voice calls
- A CoS is required before a user can make calls (other than to the operator and 911)
- The permissions include the types of calls and actions a voice user can perform

Some call types restricted for this group

All call types allowed for this group

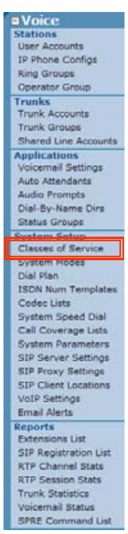
Voice/System Setup - Classes of Service

Voice System Setup
Classes of Service

- There are four default Classes of Service
  - normal_users
    - allow all call types except international and 900 calls
    - can not unlock the door by default
  - public_phones
    - allow internal, local, and toll-free calls
    - many CoS voice features disabled
  - executive_users
    - allow all call types
    - all CoS voice features enabled
  - door_phone
    - allow only internal calls
    - all CoS voice features disabled
  - You can also create your own Class of Service

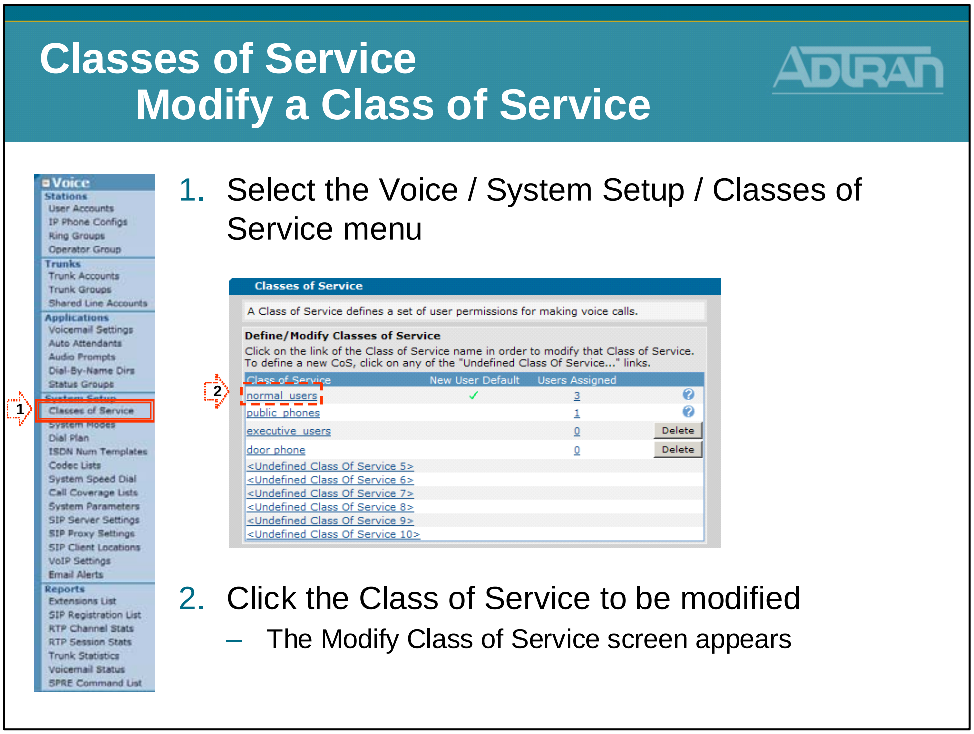
Classes of Service - Basic Configuration Steps

Classes of Service
Basic Configuration Steps



1. Select Voice / System Setup / Classes of Service from the NetVanta 7100 menus
2. Click an existing Class of Service to modify the permissions for users assigned to that Class of Service or select one of the Undefined Classes of Service to create your own
3. Specify permitted call types and desired voice actions for users assigned to this Class of Service

Classes of Service – Modify a Class of Service



Modify a Classes of Service

1) Select Voice / System Setup / Classes of Service from the NetVanta 7100 menus.

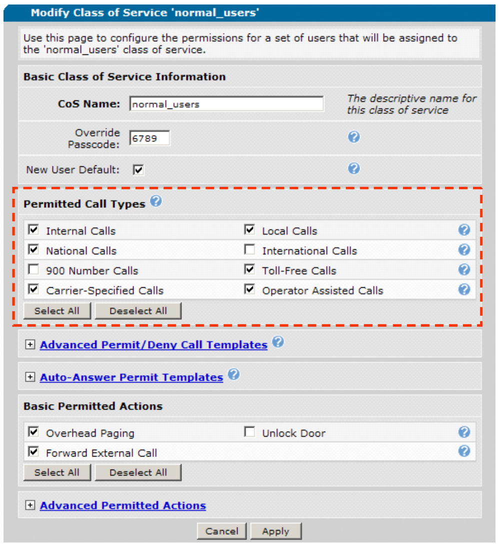
2) Click an existing Class of Service to modify the permissions for users assigned to that Class of Service or select one of the Undefined Classes of Service to create your own.

3) Specify permitted call types and desired voice actions for users assigned to this Class of Service.

Classes of Service - Permitted Call Types

Classes of Service
Permitted Call Types

ADTRAN



<u>Permitted Call Types</u>

- Determines the type of calls a user is permitted to make as a member of this class of service
- The pattern for the different call types was defined in the Voice / System Setup / Dial Plan menu
- Customized Call types can be added by selecting Advanced Permit/Deny Call Templates

Class of Service - Permitted Call Types

Permitted Call Types determine what type of calls that a user is permitted to make as a member of this class of service. Note that ranges of phone numbers are assigned to the call types (e.g. 9-NXX-XXXX = Local Calls) from the Dial Plan menu.

Internal Calls
Members of this CoS are permitted to make internal extension-to-extension calls (2XXX through 8XXX).

Local Calls
Members of this CoS are permitted to make local calls 9-NXX-XXXX.

National Calls
Members of this CoS are permitted to make national long distance calls 9-1-NXX-NXX-XXXX.

International Calls
Members of this CoS are permitted to make international long distance calls 9-011-$.

900-Number Calls
Members of this CoS are permitted to make national 1-900-NXX-XXXX and local 976-XXXX pay-per-service calls.

Toll-Free Calls
Members of this CoS are permitted to make national toll-free calls 9-1-800-NXX-XXXX including those to area codes 800, 888, 877, 866, and 855.

Carrier-Specified Calls
Members of this CoS are permitted to specify the long distance service provider for each call using a 'PIC' code 1010XXX-NXX-NXX-XXXX.

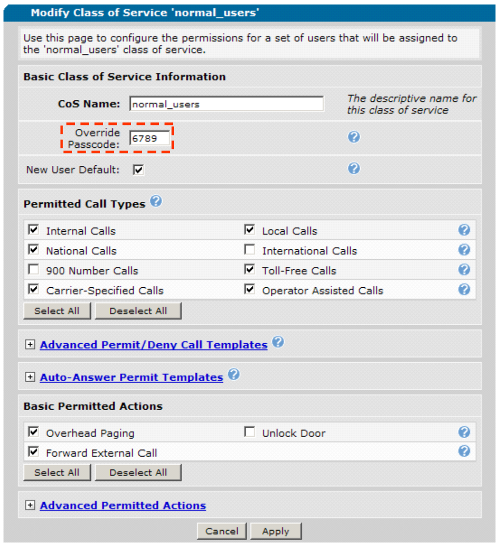
Operator Assisted Calls
Members of this CoS are permitted to dial an outside operator for assistance with making calls 9-0-NXX-NXX-XXXX.

Advanced Permit/Deny Call Templates
Click the 'Configure Advanced Templates' button to configure templates that require more detail such as area codes, etc.

- **Permit Templates** - Use this section to add and delete specific call templates that users in this Class of Service can dial. All calls matching the specified pattern will be permitted.
- **Deny Template** - Use this section to add and delete specific call templates that users in this Class of Service can NOT dial. All calls matching the specified pattern will be denied.

Classes of Service - Override Passcode

Classes of Service
Override Passcode



<u>Override Passcode</u>

- A 4-digit code used with the SPRE code *90 to override a phones configured (CoS).
  - Format *90xxxx
    (x = passcode of CoS)
- For example, if a voice user (assigned to the normal_users Class of Service) wishes to place a call from a phone assigned to the Public CoS, the user would enter *906789

Basic Class of Service Information

CoS Name
The descriptive name for this class of service.

Override Passcode
This 4-digit code is used in conjunction with the Class of Service (CoS) Override feature (*90), enabling a user to override an extension's configured CoS with the ‘this users' CoS as represented by this Override Passcode.

New User Default
When creating a new user, apply this Class of Service (CoS) automatically.

Hand Free Auto-Answer

Hands Free Auto-Answer is an intercom like feature. A user initiates a call to a SIP phone. Instead of requiring the recipient to answer the call, the (speaker) phone automatically answers and users are able to start a conversation.

Auto-Answer Permit Templates

Only voice users assigned to a Class of Service with an Auto-Answer permit template are allowed to place hands free auto answer calls.

Hands Free Auto-Answer Configuration

1) Select Voice / System Setup / Classes of Service from the NetVanta 7100 menus.

2) Edit the Class of Service that contains the voice users you wish to allow to place hands free calls.

3) Define the auto-answer permit template that users in this Class of Service can dial hands free.

Optional - Give voice users permission to block incoming auto-answer calls
- Configured per Class of Service

Optional - Block incoming auto-answer calls for specific voice user
- Configured per specific voice user extension

Placing Hands Free Auto-Answer Calls

To place an Auto-Answer call, the digits ** must precede the number. The prefix can be dialed before or with the extension.

For example, a user could place two calls: ****** and then **2004**,

Or a user could dial ****2004**

Blocking Auto-Answer Calls

Users with the Class of Service option ‘Auto Answer Do Not Disturb’ enabled can block incoming auto-answer calls with a SPRE code.

- When a user does not want to receive an Auto-Answer call, they can dial *971
- When user wishes to receive Auto-Answer calls again, they can dial *970

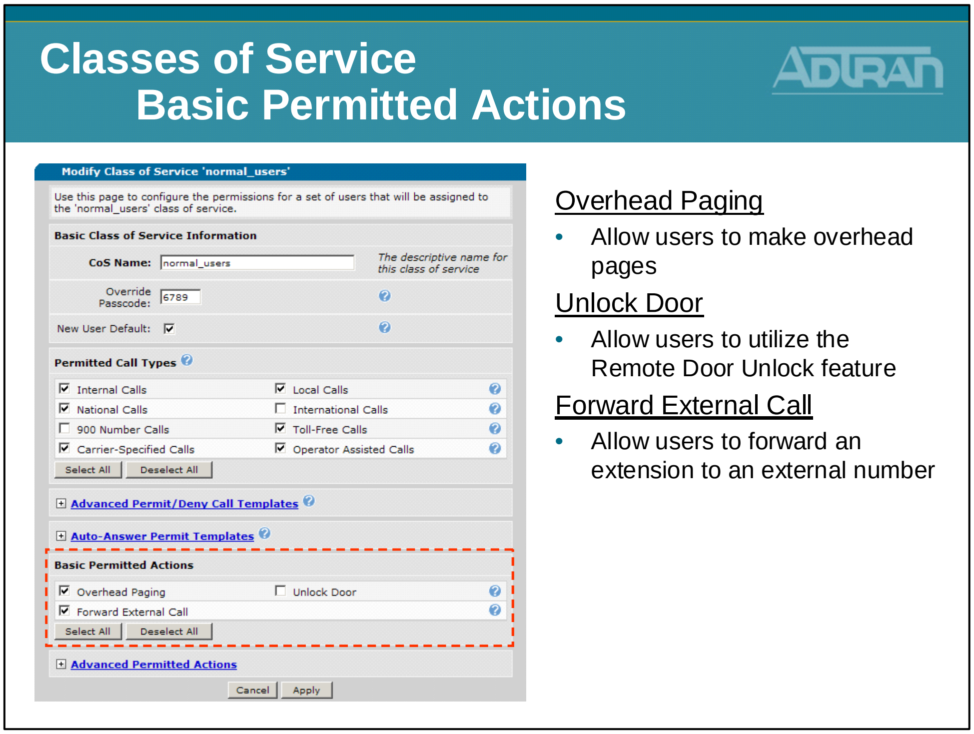
Note: There is also a per user Auto-Answer Do Not Disturb option. If enabled, any incoming Auto-Answer calls will ring normally instead of being automatically answered by the phone.

Hands Free Auto-Answer - No Permission or Blocked

If an Auto-Answer call is initiated by a user that “does not” have permission to do so, a normal call is placed. (No Auto-Answer functionality)

If an Auto-Answer call is received by a user that has blocked the functionality a normal call is placed. (No Auto-Answer functionality)

Classes of Service - Basic Permitted Actions



Class of Service - Basic Permitted Actions

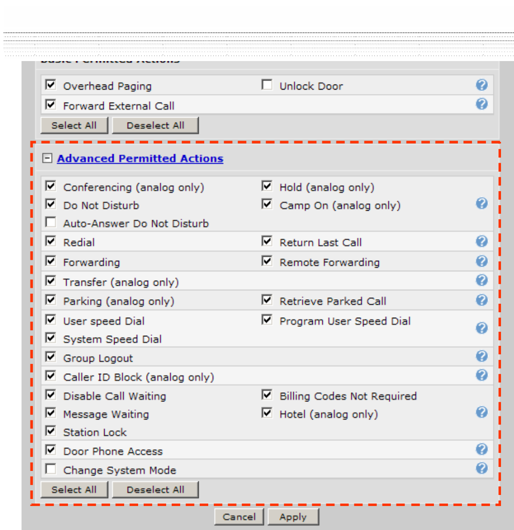
Overhead Paging
Select to allow users to make overhead pages.

Forward External Call
Select to allow users to forward an extension to an external number.

Unlock Door
Select to allow users to utilize the Remote Door Unlock feature.

Classes of Service - Advanced Permitted Actions

Classes of Service
Advanced Permitted Actions



Do Not Disturb

- Allow user to place an extension in Do Not Disturb mode

Group Logout

- Allow a user to logout of a call group

Station Lock

- Allow a user to place their extension in a locked mode

Door Phone Access

- Allow a user to make calls to the intercom designated as the 'door phone'

Change System Mode

- Allow a user to change the current system mode

Class of Service - Advanced Permitted Actions

Conferencing
Select to allow a user to establish conference calls.

Hold
Select to allow a user to put calls on hold.

Do Not Disturb
Select to allow a user to place an extension in Do Not Disturb mode.

Camp On
Select to allow a user to request a callback when a busy number becomes idle.

Auto-Answer Do Not Disturb
Select to allow a user to force incoming Auto-Answer calls to ring the phone instead.

Redial
Select to allow a user to use the redial functionality of the system to redial the last dialed number.

Return Last Call
Select to allow a user to return the call of the last incoming caller.

Forwarding
Select to allow a user to enable call forwarding.

Remote Forwarding
Select to allow a user to enable call forwarding from a remote location.

Transfer
Select to allow a user to transfer calls to an internal user.

Parking
Select to allow a user to park calls to a public hold zone.

Retrieve Parked Call
Select to allow a user to retrieve parked calls from a public hold zone.

User Speed Dial
Select to allow a user to have personal speed dial numbers.

Program User Speed Dial
Select to allow a user to modify his personal speed dial numbers.

System Speed Dial
Select to allow a user to utilize the system speed dial numbers.

Group Logout
Select to allow a user to logout of a call group.

Caller ID Block
Select to allow a user to block caller ID for outbound calls.

Disable Call Waiting
Select to allow a user to disable the shared call appearance known as call waiting (if available).

Billing Codes Not Required
If selected, the user does not have to enter a billing code prior to dialing a number.

Message Waiting
Select to allow a user to change the manner in which message notification takes place.

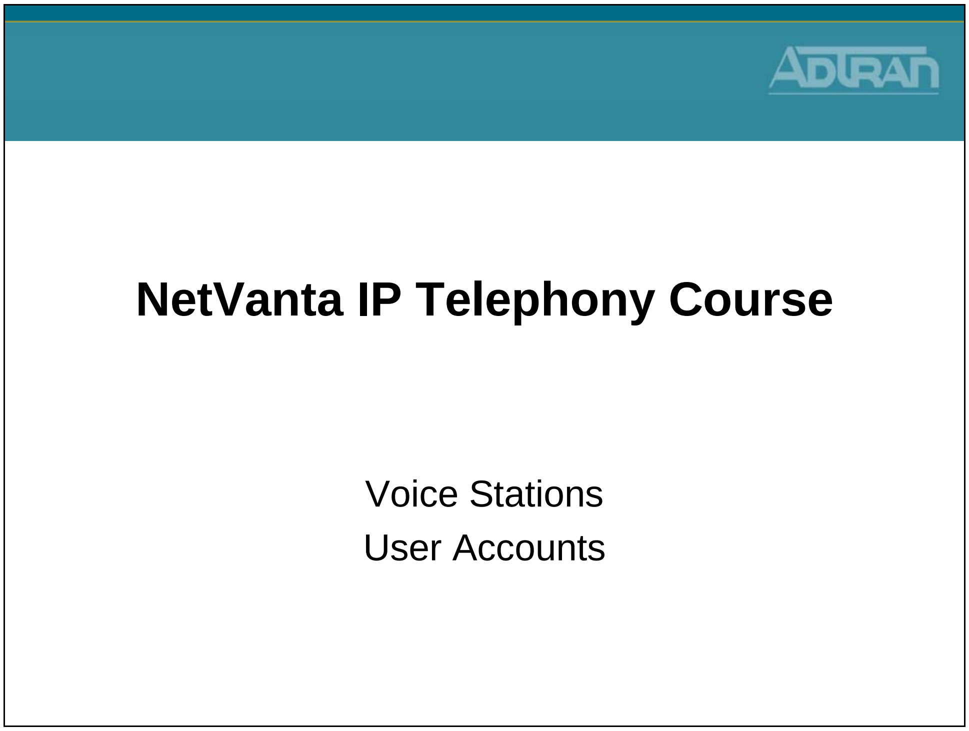
Hotel
Select to allow a user to login to a phone designated for hotelling or hotdesking.

Station Lock
Select to allow a user to place his extension in a locked mode.

Door Phone Access
Select to allow a user to make calls to the intercom designated as the 'door phone'.

Change System Mode
Select to allow a user to change the current system mode of the unit.

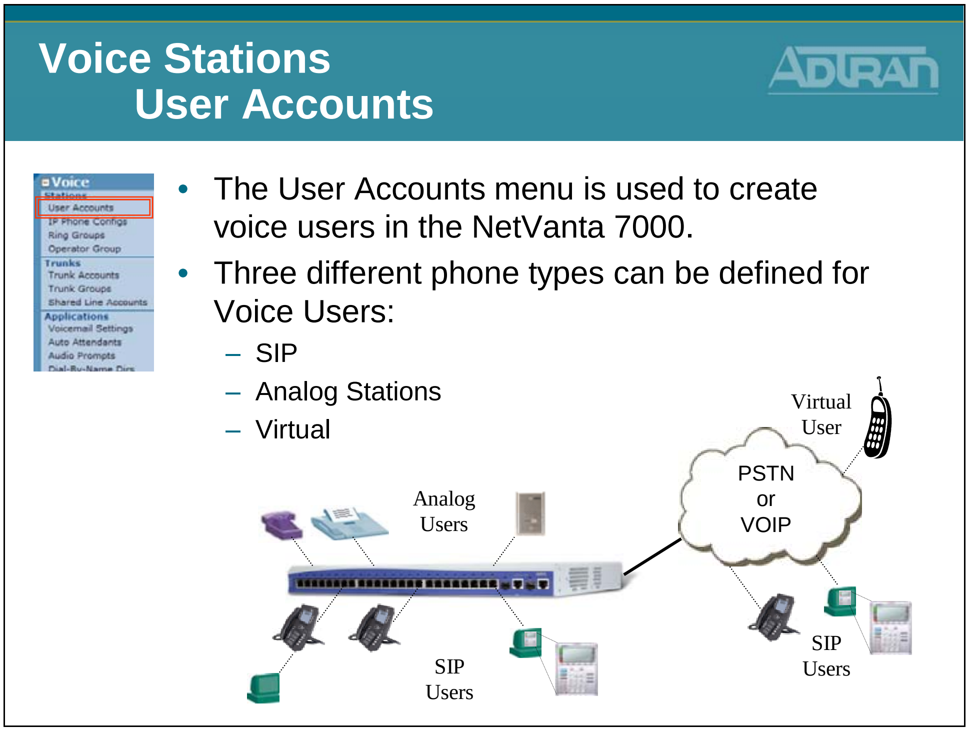
Voice Stations - User Accounts



Voice Stations

The Voice Station menus include User Accounts, Ring Groups, and Operator Group. The User Accounts configuration screen allows you to create a user account for every telephone user in the NetVanta 7000 Series system. The Ring Group menu allows you to define a group of user accounts that can be called in a coordinated way with a single extension. From the Operator Group menu, you define the members of the Operator Group.

Voice Stations - User Accounts



User Stations Accounts define phone users in the NetVanta 7100. The three different phone types that can be defined for Voice Users are:

SIP - user account is associated with a SIP port

Analog Stations – user is associated with a physical FXS interface

Virtual - user account is not associated with a physical port

User Accounts - Analog Users

User Accounts
Analog Users

- An Analog Voice user associates a physical Analog FXS port with a voice user
- Analog Station accounts are required with the following:
  – traditional analog telephones
  – door phones
  – faxes
  – modems
  – credit card readers

Analog Station Voice Users

Voice users with a Phone Type of Analog Station associate a physical Analog FXS port with a voice user. The selection of the phone type Analog Station is required when creating voice users for traditional analog telephones, door phones, faxes, modems, or credit card readers.

To Create an Analog User Account

1. Select Voice / Stations / User Accounts from the NetVanta 7000 menus
2. Assign users extension and name
3. Select Phone Type Analog Station
4. Select the physical FXS Port
5. Define user parameters such as Classes of Service and Voicemail settings

User Accounts - SIP Voice User

User Accounts
SIP Voice User

- A SIP voice user is a voice user that communicates using the SIP standard
  - The NetVanta 7000 is designed to meet SIP standards and is interoperable with many SIP-compliant telephones
  - The SIP voice user could be associated with a SIP-compliant telephone or an IP SoftPhone running on your laptop or desktop PC
  - When creating a SIP Voice User, phone configuration files can be created for recognized phone models
  - The phone configuration files are created in the NetVanta 7000’s flash memory (CFLASH by default)

SIP Voice User

A SIP voice user is a voice user that communicates using the SIP standard. The NetVanta 7100 is designed to meet SIP standards and is interoperable with many SIP-compliant telephones. The SIP voice user could be associated with a SIP-compliant telephone or an IP SoftPhone running on your laptop or desktop PC.

To Create a SIP User Account

1. Select Voice / Stations / User Accounts from the NetVanta 7000 menus
2. Assign users extension and name
3. Select Phone Type SIP
4. Choose New Address then type phones MAC Address
   - Phone configuration files are created for recognized phone models and stored in 7000 CFLASH by default
   - If MAC address “Not Set” is selected, no configuration files are created
5. Define user parameters such as Classes of Service and Voicemail settings

User Accounts - Virtual User

User Accounts
Virtual User

- A Virtual User is a voice user that is not tied to a physical interface
  - Creating virtual users may be useful for employees who do not need a permanent phone in an office
  - Virtual users can be given Voicemail ability and call forwarding capabilities
  - When in the office, the virtual users can login into an analog phone that has the hotel feature enabled (shared-desk application)

Virtual Voice Users

A Virtual User is a voice user that is not tied to a physical interface. Creating virtual users may be useful for employees who do not need a permanent phone in an office. Virtual users can be given Voicemail ability and call forwarding capabilities. When in the office, the virtual users can login into an analog phone that has the hotel feature enabled. (shared-desk application)

To Create a Virtual User Account

1. Select Voice / Stations / User Accounts from the NetVanta 7000 menus
2. Assign users extension and name
3. Select Phone Type Virtual
4. Define user parameters such as Classes of Service and Voicemail settings

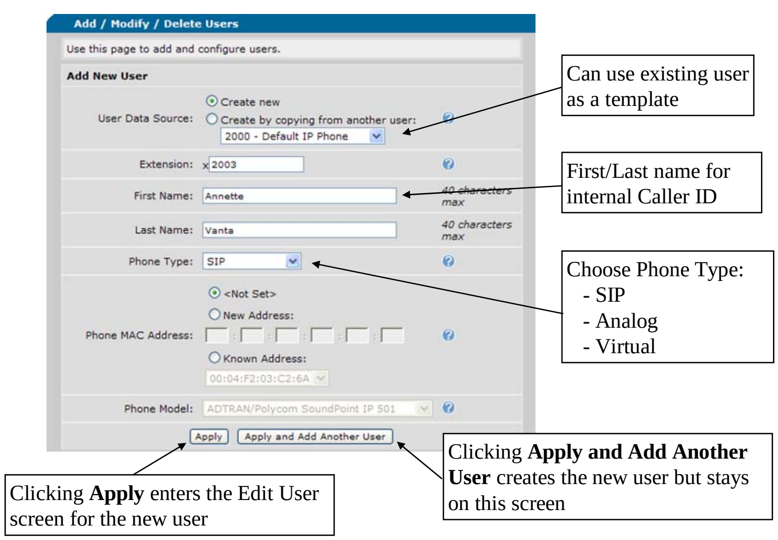
Creating New User Accounts

Creating New User Accounts

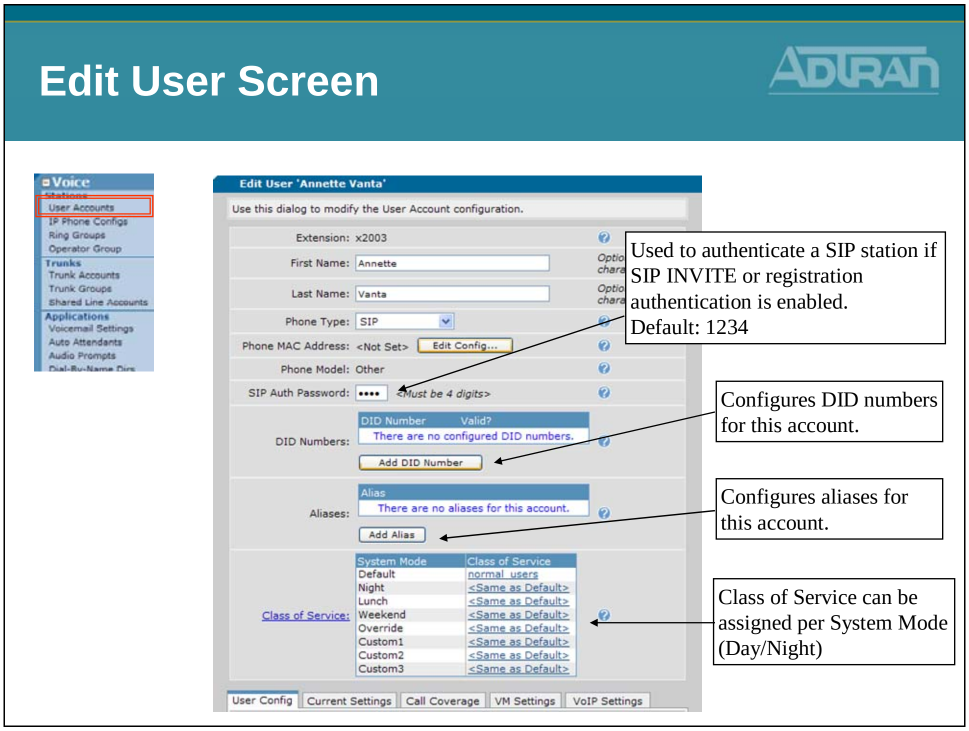
1. Select Voice / Stations / User Accounts from the NetVanta 7000 menus
2. Assign users extension and name
3. Select Phone Type
   - Analog, SIP, or Virtual
     - If analog, select available FXS port
     - If known SIP phone model, enter MAC address
4. Define user parameters such as Classes of Service and Voicemail settings

New User Screen

New User Screen



Edit User Screen



Voice User Settings

The Voice User Settings are the settings that can be seen or modified while editing a voice user. When a new voice user is created, you are placed in the Edit <voice user> screen where the settings below display.

Editing Voice Users - Initial Screen

Extension
Assigned when a voice user is created and can not be modified

First Name
(Optional) 40 characters max

Last Name
(Optional) 40 characters max

Phone Type - Analog Station
User account is associated with an Analog FXS port. If Analog Station is not displayed as an option, it is because there are no available FXS ports.

Phone Type – SIP
User account is associated with a SIP port

Phone Type – Virtual
User account is not associated with a physical port

Phone MAC Address (SIP ONLY)
Optionally enter the MAC Address of this user's SIP phone. Note that a phone configuration file can be created for this phone only if a complete MAC Address is entered.

SIP Auth Password (SIP ONLY)
The SIP Auth Password is used to authenticate a SIP station if SIP INVITE or registration authentication is enabled.

Phone Port: (ANALOG ONLY)
If Phone Type is Analog Station: The physical Analog FXS port to associate with this user account. If Phone Type is Virtual: Not used

Login PIN (SIP)
The SIP Auth Password is used to authenticate a SIP station if SIP INVITE or registration authentication is enabled.

Login PIN (ANALOG or VIRTUAL)
The Login PIN is used to log into and out of analog phones. This allows a user to "take over" another person's phone or for "hotdesking"

DID Numbers
Configures DID numbers for this account. The table shows all existing DID numbers (you may have to scroll to see all of them) and whether each number is currently valid. A number is considered valid if it matches any trunk's DID prefix and digit count. If no DID information has been configured in trunks, then all numbers are considered valid.

- To add a new DID number, click the Add DID Number button just below the DID Number table and enter the DID number in the popup box.
- To delete a DID number, click the Delete button next to the number you want to delete.

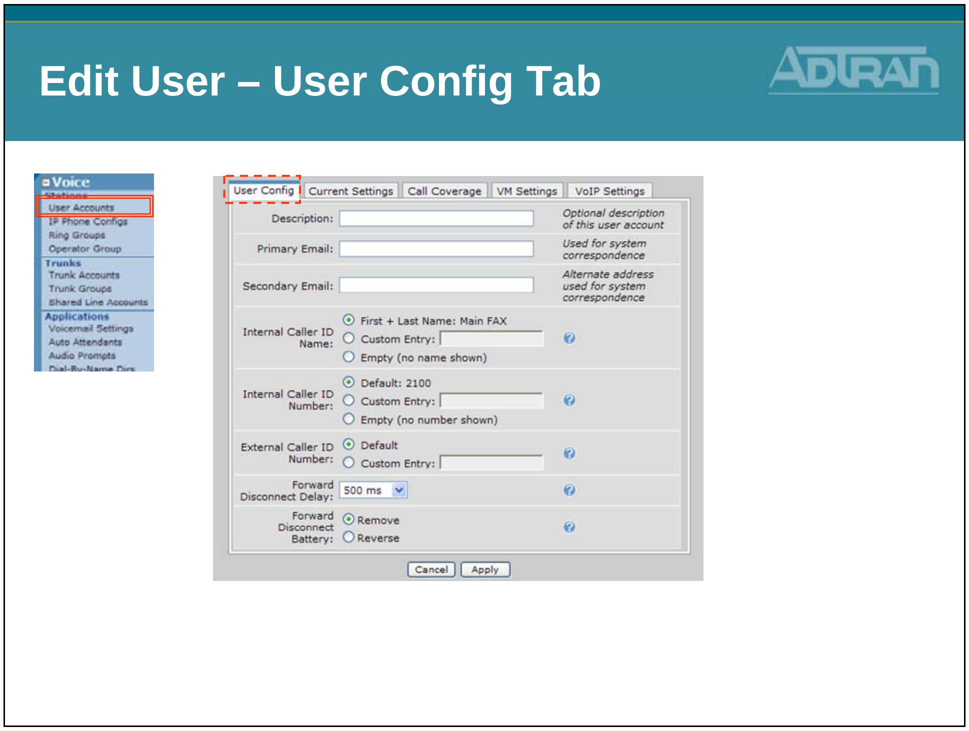
Aliases
Configures aliases for this account. The table shows all existing aliases (you may have to scroll to see all of them).

- To add a new alias, click the Add Alias button just below the Alias table and enter the new alias for this account in the popup box.
- To delete an alias, click the Delete button next to the alias you want to delete.

Class of Service

Configures this user's Class of Service.

Edit User – User Config Tab



The **User Config** tab allows you to configure the user's email address, caller ID settings, and Forward Disconnect for analog users.

Description
Optional description of this user account

Primary Email
Used for system correspondence

Secondary Email
Alternate address used for system correspondence

Internal Caller ID – Name
Configures the name portion of the Caller ID display for internal calls made by this user.

- **First + Last Name** - Sets Caller ID Name to be the configured first and last name.
- **Custom Entry** - Sets the Caller ID Name to be the value entered in the adjacent text box.
- **Empty** - Sets the Caller ID Name to be empty.
- Note: The system has no control over Caller ID Name display for external calls.

Internal Caller ID – Number
Configures the number portion of the Caller ID display for internal calls made by this user.

- **Default** - Sets the internal Caller ID Number to be the extension of this user account.
- **Custom Entry** - Sets the internal Caller ID Number to the the value entered in text box.
- **Empty** - Sets the internal Caller ID Number to be empty.

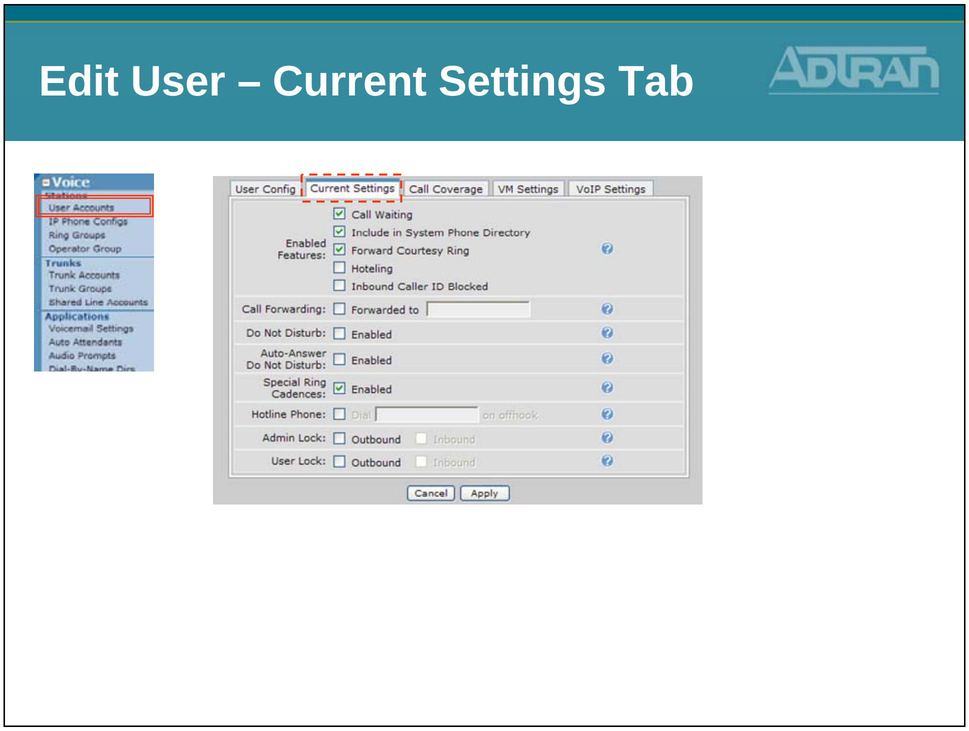
External Caller ID – Number
Configures the number portion of the Caller ID display for external calls made by this user. Note that external Caller ID info is only sent if delivered out particular T1 interfaces such as Feature Group D or PRI.

- **Default** - Automatically sets the external Caller ID Number to be the first DID entry if one exists, otherwise it’s set to nothing.
- **Custom Entry** - Sets the external Caller ID Number to the value entered in the adjacent text box.

Forward Disconnect Delay
Setting Forward disconnect delay enables the removal or reversal of battery for the specified amount of time. When the unit removes/reverses the battery current, the connected equipment will acknowledge this condition by dropping the line.

Forward Disconnect Battery
Select whether the connected equipment expects battery removal or reversal.

Edit User – Current Settings Tab



The **Current Settings** tab allows you to change voice settings for this user.

Call Waiting
If checked, call waiting is enabled on this user account.

Include in System Phone Directory
If checked, the user will be included in the dial-by-name directory.

Forward Courtesy Ring
If checked, the user's phone will issue a short "blip" ring when a call comes in as a reminder that the phone is forwarded.

Hotelling
If checked, another user can log into this user's phone without logging this user out first. Useful for shared-desk applications.

Inbound Caller ID Blocked
If checked, no inbound Caller ID information will be delivered to this user's phone.

Call Forwarding
If checked, this user's extension is forwared to the number displayed

Do Not Disturb
If checked, Do Not Disturb is enabled and all calls will go directly to the user's call coverage list.

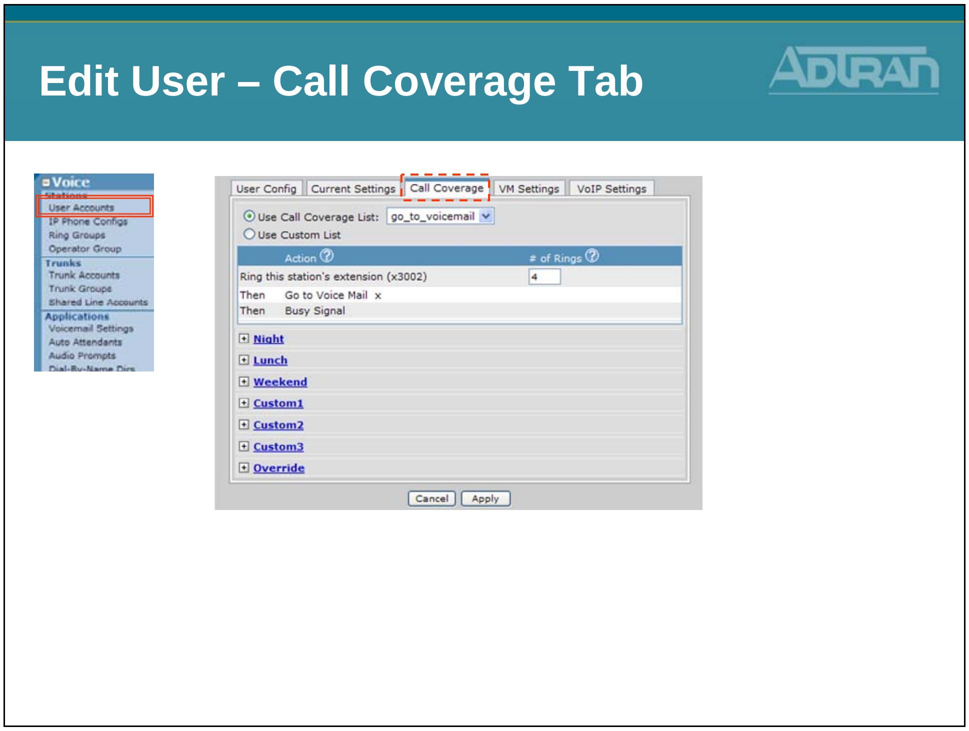
Special Ring Cadences
If checked, the phone will ring with a different cadence depending on the call type, such as internal, external, or priority calls. If unchecked, the phone will always ring with the default cadence.

Hotline Phone
If checked, a call will be immediately placed to the configured number when this user goes offhook

Admin Lock
Displays current administrative lock

User Lock
Displays current user lock

Edit User – Call Coverage Tab



The Call Coverage tab displays the call coverage settings for this user. If the user has been assigned to a Call Coverage List, you can view the settings on this page. You can also create a custom Call Coverage List only for this User Account. Use the question mark symbol to assist with the configuration settings.

Use Call Coverage List
Used to make a copy of the selected global Call Coverage List for this specific extension. Modifications made to this copy do not affect the original global list.

Use Custom List
Create a custom list of how to handle a call when no one answers the phone.

Action

Actions on a list are evaluated in the order displayed.

# of Rings
If there is no response after this many rings (or the extension is busy), the next item in the call coverage list will be tried.

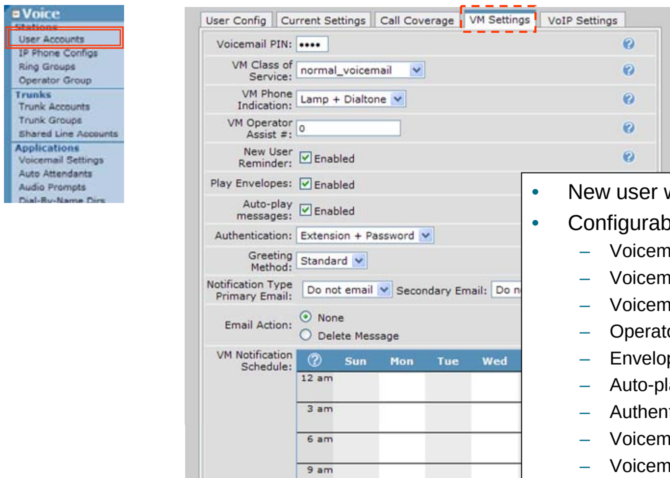
If a value of 0 is used, the call coverage list will only be processed if the station is busy. Otherwise, the phone will ring indefinitely.

System Modes

Call coverage can be configured per system mode. The number of rings between call coverage choices can also be set per system mode. Voice users, ring groups, and operator group, shared line accounts can use the global or custom call coverage list.

Edit User – VM Settings Tab

Edit User – VM Settings Tab



- New user wizard for mailbox setup
- Configurable voicemail options
  - Voicemail Pin
  - Voicemail Class of Service
  - Voicemail Phone Indication
  - Operator Assist Number
  - Envelope playback
  - Auto-play of messages
  - Authentication options
  - Voicemail Greeting Method
  - Voicemail Notification Email
    - Text File
    - WAV File

The VM Settings tab allows you to edit the user's voicemail settings such as VM Phone Indication and VM Notification Schedule.

Voicemail PIN
Sets the password the user must enter to access the voicemail system

VM Class of Service
The voicemail class of service assigned to this user account

VM Phone Indication
Lamp + Dialtone - use both the message waiting lamp and stutter dial tone to indicate new voicemail

- **Lamp Only** - use the message waiting lamp to indicate new voicemail
- **Dialtone Only** - use a stutter dialtone to indicate new voicemail
- **Off** - no indication of new voicemail

VM Operator Assist #
This number will be dialed if a caller requests to speak with the operator while leaving a voicemail.

New User Reminder
Checking this box alerts the Voicemail system to prompt the user to record their name. The recorded name is subsequently used for playback within the system.

Play Envelopes
When enabled, envelopes preceding voice messages will be played. An envelope includes the Calling party and the Date/Time information about a message.

Auto-play messages
When enabled, voice message playback will begin automatically after logging into your voice mailbox.

Authentication
Choose the authentication method to be used when logging into your voice mailbox. From valid phones, authenticate using:

- mailbox/password
- password only
- no authentication

WARNING: Selecting “None” will allow anyone who knows your extension to hear your messages.

Greeting Method
Choose the greeting that will be heard by callers leaving voice messages. The Default greeting is your recorded name. To record Standard and Alternate greetings, login to your voicemail via your phone and follow the instructions under the Greetings menu.

Voicemail Notification Schedule
The Voicemail Notification Schedule configures when and how the system will notify this user when they receive a voicemail message. To configure the schedule:

1. Click the Add Range button below the schedule detail.
2. Enter the start and end times for the range. A 'range' is a range of time during the week that will have the same notification type.
3. Select the notification type to use. The available options are to send an email to the primary email address or the secondary email address. These addresses are configured in the User Config tab on this page.
4. Click the Apply button just below the Enabled Actions selection.

This will add a schedule range to both the graphic schedule display as well as the schedule detail table. You can edit an existing range by clicking on the Start Day/Time text link in the detail table. You can delete an existing range by clicking the Delete button next to the range in the detail table that you want to remove.

Remember to click the Apply button at the bottom of the page to save the schedule changes. You will lose your changes if you do not click the Apply button.

New User Wizard for Mailbox

Voicemail Feature
New User Wizard for Mailbox Setup

- User prompted to record name/greeting and setup password first time through

"Welcome to the Voicemail Setup Wizard. As a new user you will need to configure various aspects of your voicemail prior to using it."

*"**To setup your mailbox press 1,** to skip setup of your voicemail account and run the wizard again the next time you log in press 2, for help press 0."*

New User Wizard for Mailbox

Voicemail Feature
New User Wizard for Mailbox Setup

- *To setup your mailbox press **1**….*
  - If new user presses 1 they will be prompted to complete the following:

*"Record your **name** after the tone and press # when you're finished."*

:

*"Record your **standard greeting** after the tone and press # when you're finished."*

:

*"Your password is used to provide security for your messages. You should set it to something different than your extension and it must be 4 digits long. After the tone enter your **4 digit password** followed by the # sign.*

:

Configurable VM Authentication

Voicemail Feature
Configurable VM Authentication

- From valid phones, authenticate using:
  - mailbox/password
  - password only
  - no authentication
- Authentication options function with the SPRE code ***98** or when pressing the **Messages** key on the ADTRAN and Polycom phones
  - When dialing voicemail extension you will always be asked for your mailbox/password combination

Voicemail Notification Email

Voicemail Feature
Voicemail Notification Email

- Email notification when voicemail has been left
- Select between
  - NOT being notified via email
  - an email that contains only **text**
  - email with voicemail message attached in **WAV** format
- Optional "Delete Message" after sending a voicemail as an email attachment
- Voicemail Notification schedule

Enabling Email Notification of Voicemail Messages

Configuring voicemail notification consists of selecting the time of day and specifying email addresses the system will use to notify users when they receive a new voicemail message. When this feature is configured, the system sends an email alert to the specified email address.

1. To allow email notification, the system administrator must first configure the outgoing mail server settings under Utilities > Logging > Email Forwarding menu. The minimum configuration required is to configure the Email Server (IP address) and the Email Sender (email address).

2. Navigate to the Voice > Stations > User Accounts menu and edit voice user.

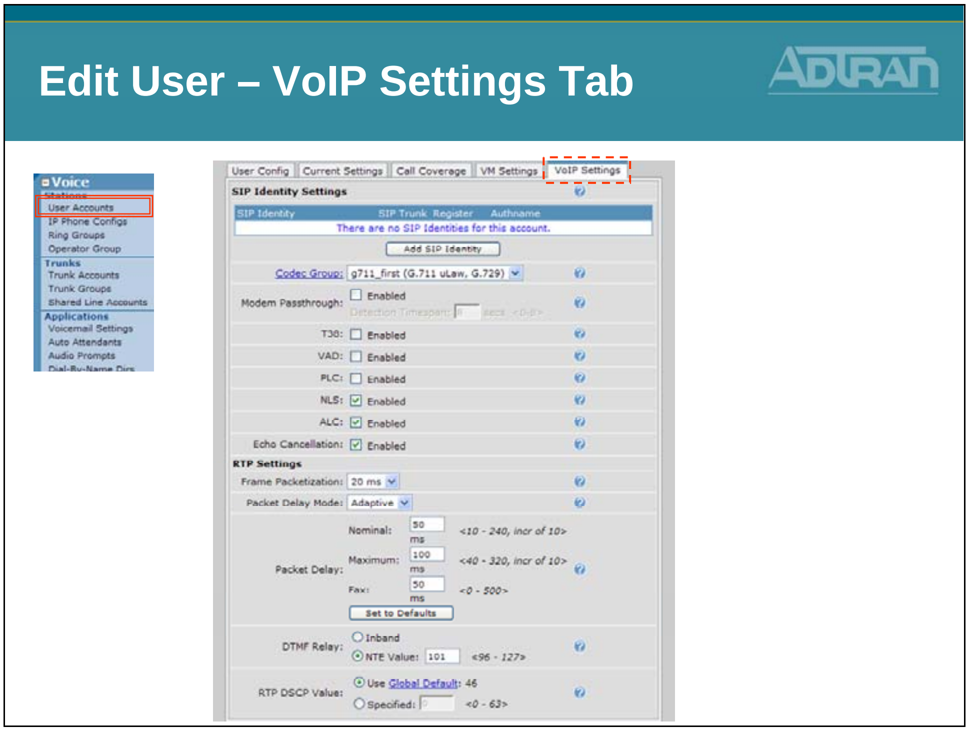
3. On the Edit User screen, scroll down to the User Config tab and set (or edit) the email address(es) to use for voicemail notification.

4. From the VM Settings tab, set the Notification Type for the Primary or Secondary Email to None, Text, or attach WAV.

5. From the VM Setting tab, select the Add Range button.

- The Add Notification Schedule Range menu appears. Enter the Start Day/Time and End Day/Time times for the voicemail notification range. A range is the period of time during the week that will have the same notification type. The schedule range added here will appear in the VM Notification Schedule graph.

Edit User – VoIP Settings Tab



The VoIP Settings tab allows you to edit the user’s voice over IP settings like codec group, VAD, and RTP settings.

SIP Identity Settings

Configures SIP Identities for this account. The table shows all existing SIP Id's (you may have to scroll to see all of them).

- To add a new SIP Id, click the Add SIP Identity button just below the SIP Identity table and enter the new SIP Id for this account in the popup box.
- Click the Delete button next to the SIP Id entry if you wish to remove it.

SIP Identity
Enter this user's SIP Identity. Currently, this value must be equal to the user's extension.

Associated SIP Trunk
Select the SIP Trunk this station will use for registration purposes.

Trunk Registration
Select whether or not this user should register with selected the SIP Trunk.

Trunk Authentication
Optionally, set the authentication information for this station. If 'Not Set' is chosen, the unit will use the registration trunk authentication data if it exists. Otherwise, no authentication data will be sent.

Codec Group
Select the codec group to use for this station account.

Modem Passthrough
When Modem Passthrough is enabled and an existing call detects a modem or fax tone, the unit will automatically renegotiate with the far end to be modem-compatible (switch to G.711, all voice improvements turned off, packet delay set to Fax).

T38
When T.38 is enabled and an existing call detects a fax tone, the unit will automatically renegotiate with the far end to be T.38.

VAD
When Voice Activity Detection is enabled, silence is not transmitted over the network, only audible speech. When VAD is enabled, the sound quality is slightly degraded but the connection monopolizes much less bandwidth.

PLC
Enables/disables Packet Loss Concealment. When enabled, the unit will try to reconstruct sound lost from dropped packets.

NLS
Enables/disables the echo canceller's Non-Linear Suppression. When enabled, acoustic echo should be reduced.

ALC
Enables/disables the Automatic Leveling Control. When enabled, reduces received RTP signals to a predefined level.

Echo Cancellation
When enabled, reflected noise is cancelled from the transmitted voice signal. Echo cancellation should normally only be disabled if the voice station is connected to a fax machine or modem.

RTP Settings

Frame Packetization
Select the number of audio samples in ms (1 frame/sample is 10 ms) included in a single RTP packet.

Packet Delay Mode
Configures the operation mode of the jitter buffer for VoIP calls involving this account.

- **Adaptive -** The buffer's delay starts at the nominal delay setting but will increase up to the delay setting if it detects that an intolerable number of packets are being discarded due to jitter. Conversely, the buffer will decrease the amount of delay if it can afford to.
- **Fixed** - The buffer's delay stays at the nominal setting at all times.

Packet Delay
Configures various packet delay settings for this account.

- **Nominal** - For voice calls, the nominal delay value represents the desired amount of packet delay. In adaptive mode, the buffer may increase this value up to the maximum delay. In fixed mode, the delay is constantly set at this value.
- **Maximum** - For voice calls, the maximum delay value represents the maximum delay to which the adaptive jitter buffer can grow.
- **Fax -** If Modem Passthrough is enabled and modem/fax tones are detected, the packet delay setting will be switched to this value.

DTMF Relay
Select how DTMF tones are to be transmitted over RTP. If out of band (NTE), also enter the NTE value.

RTP DSCP Value
Select the DiffServe code point for this station's RTP packets. Either use the global default (which will change as the global default changes) or specify a value for this station only.

Creating Voice User Account Examples

NetVanta IP Telephony Course

Voice Stations

Creating Voice User Examples

ADTRAN ADP-40 Door phone

Analog Voice User
ADTRAN ADP-40 Door phone

The ADTRAN ADP-40 is a brushed stainless steel, compact, weather and vandal resistant, telephone line powered speakerphone designed to provide two-way hands free communication.

Door Phone Configuration Summary

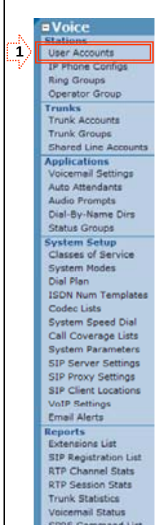
ADTRAN ADP-40 Door Phone Configuration Summary

1. After installation of the door phone, attach the phones RJ-11 connector to one of the NetVanta 7000 FXS ports
2. Create a Ring Group to define the phones that will ring and the call flow when someone presses the call button on the door phone
3. Create an analog station account for the door phone
4. Configure the analog station account to dial the Ring Group when button is pressed

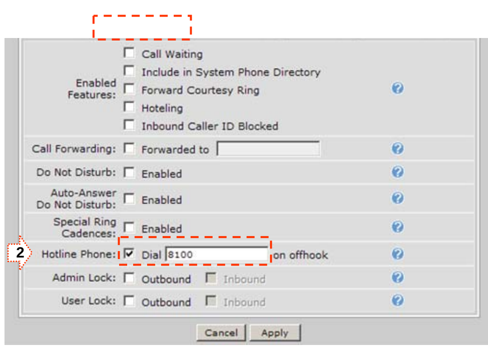
The user's forward disconnect should be left at the default of 500ms.

Door phone Configuration

Door Phone Configuration Enable Hotline



1. From the Voice / Stations / User Accounts menu, create Analog Voice user for door phone



Set Class of Service to **door phone**

Other settings should be disabled

When someone presses door phone, it will dial extension entered here

2. From the Current Setting tab, enable Hotline and set Dial to number to automatically call

ADTRAN IP SoftPhone

ADTRAN IP SoftPhone

- Optional IP SoftPhone
- Turns PC or laptop into easy to use telephone
- Ideal for remote office or access while traveling
- Compatible with USB headsets for hands-free operation

The ADTRAN IP SoftPhone is an intuitive software application designed to enable VoIP communication from your laptop or desktop PC. It offers many business features including transfer, conference, forward, hold, do-not-disturb and quick access to the address book and call logs such as recently received calls, missed calls and placed calls. The SoftPhone is ideal for business, home office, or mobile communications.

NOTE: The SoftPhone must be registered with the serial number that was received with the SoftPhone order. You can download the ADTRAN IP SoftPhone at any time by going to www.adtran.com/softphone.

License Key: When starting the ADTRAN IP SoftPhone for the first time, you will be prompted for a product-specific license key. Copy and paste this vendor-provided key into the on-screen field labeled License Key. *You must have an active connection to the Internet when this is done.*

The ADTRAN IP SoftPhone User Manual and additional information can be found at www.adtran.com/softphone.

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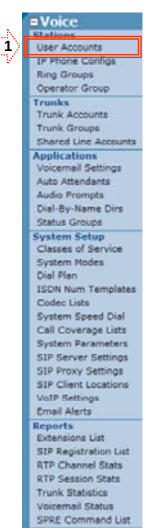
IP SoftPhone - Configuration Summary

ADTRAN IP SoftPhone Configuration Summary

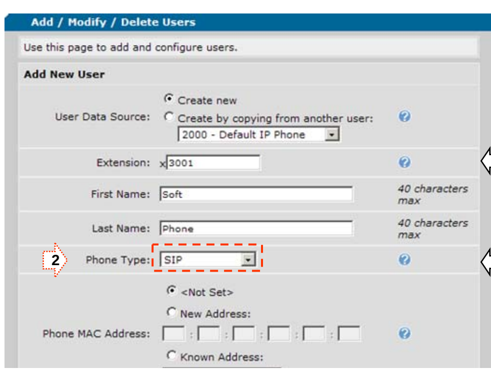
1. From the NetVanta 7000, create a SIP user for this client
2. Install the ADTRAN IP SoftPhone on the user’s PC
3. License the ADTRAN IP Softphone using the instructions provided at purchase
4. Add a SIP account to the Softphone that matches a user on the NetVanta 7000

Creating a SIP User for the IP SoftPhone

Creating a SIP User for the IP SoftPhone



1. From the Voice / Stations / User Accounts menu, create a SIP User Account



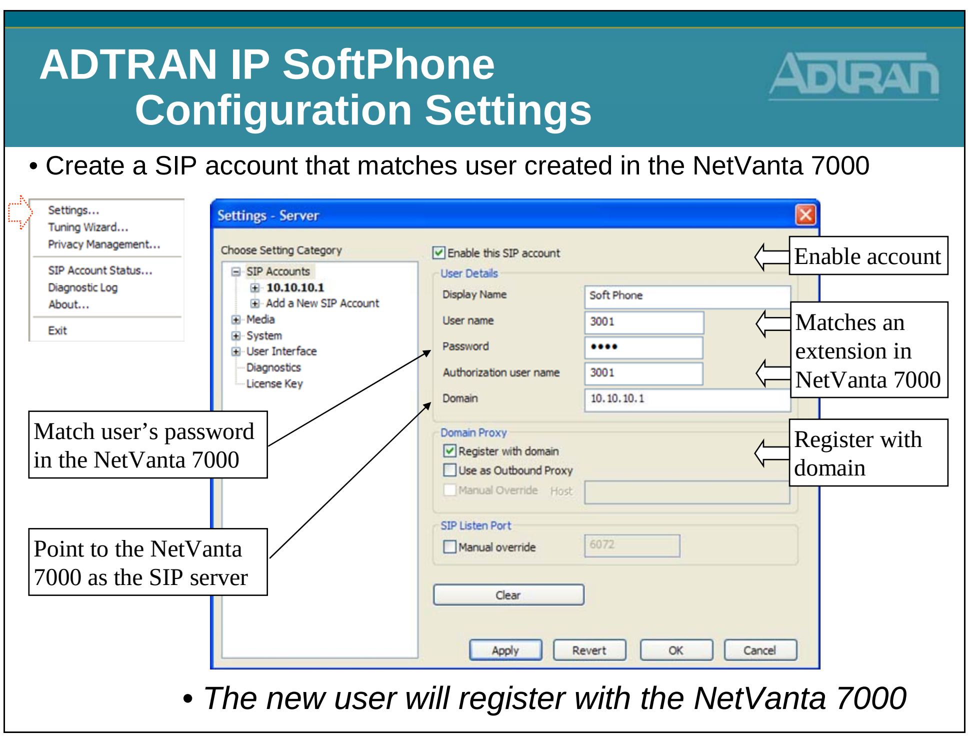
Enter extension and name

Select Phone Type **SIP** and Phone MAC Address to **<Not Set>**

– Create a SIP user account that will match a user created in SoftPhone

IP SoftPhone – Configure SoftPhone

IP SoftPhone – Configuration Settings



SIP User Account – Known Phone Models

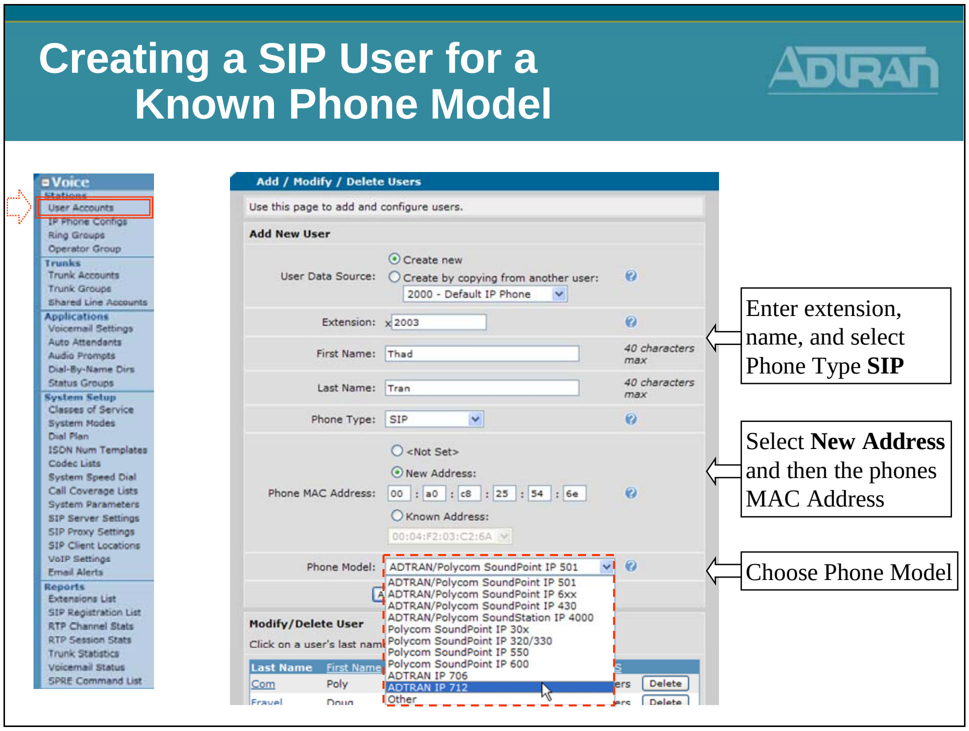
SIP User Account Known Phone Models

- Phone configuration files are created for recognized phone models and stored in 7000 CFLASH by default
  - The phone will load this configuration file at boot
  - Phone configuration files define phone features, user information, SIP Server, etc…

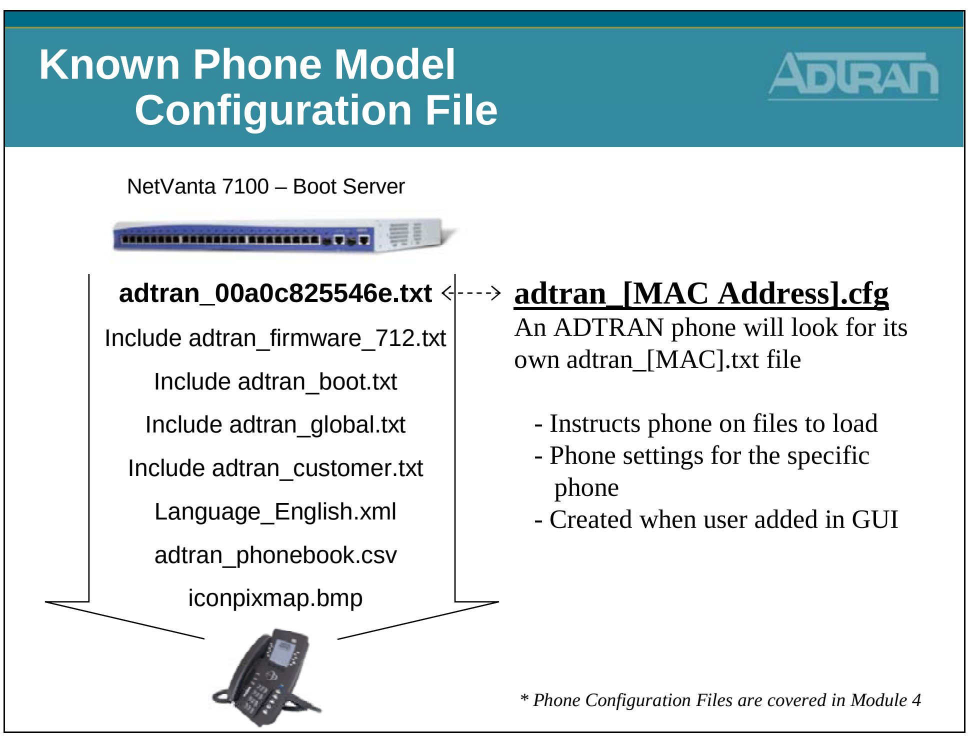
Standard SIP phones load configuration files that define most of the IP phone features and configuration parameters. When the phone boots, it loads configuration files based on its MAC address. The NetVanta 7000 stores phone configuration files in CFLASH.

- ADTRAN phone configuration files are stored in the CFLASH ADTRAN folder
- Polycom phone configuration files are stored in the CFLASH Polycom folder

Creating a SIP User for a Known Phone Model



Known Phone Model – Configuration File



SIP User Status

The status of a SIP user can be seen from the Voice / Stations / User Accounts screen.

Voice / Stations / User Accounts

Modify/Delete User

Click on a user's last name to edit their configuration.

Last Name	First Name	Extension	Port	Station CoS	
Doe	Jane	2006	~~SIP~~ ?	normal_users	Delete
IP Phone	Default	2000	~~SIP~~ ?	public_phones	Delete
Lobby	South	2003	fxs 2/1	normal_users	Delete
Port 0/1	Analog FXS	2001	fxs 0/1	normal_users	Delete
Port 0/2	Analog FXS	2002	fxs 0/2	normal_users	Delete
Smith	John	2004	SIP ?	normal_users	Delete
Tran	Thad	2005	SIP ?	normal_users	Delete

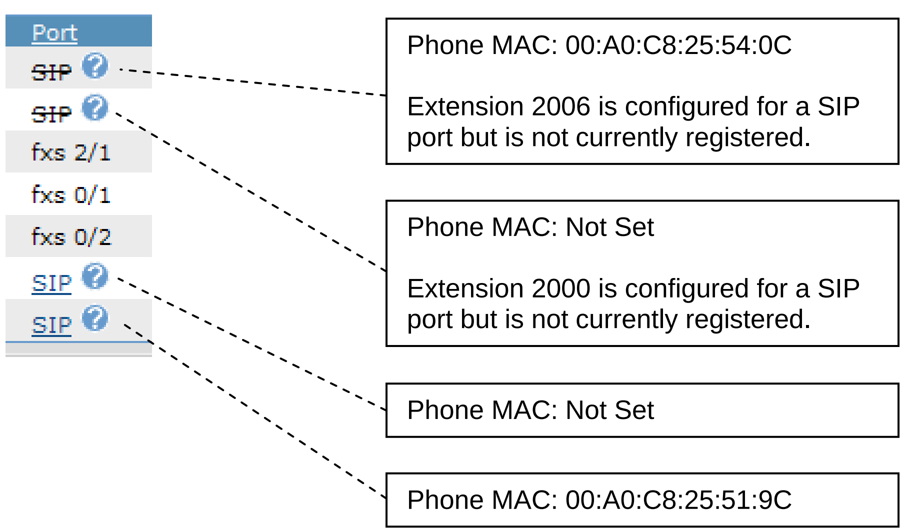
Registration Status

If the SIP user has **registered** with the NetVanta 7100, a line displays below the word <u>**SIP**</u>.

If the SIP user has **not registered** with the NetVanta 7100, a line displays though the word ~~**SIP**~~.

A bubble displays next to the SIP user's port. If you place your cursor over the bubble, information about the SIP user will display.

SIP Status Information Examples



Hotelling *(Analog Only)*

The Hotelling option allows users to log into a hotel enabled phone. When hotelling is enabled, a user can log into a user's phone without logging the current user out first. Useful for shared-desk applications.

Hotteling must be enabled for for both the Voice User of the analog phone and the Virtual voice user that will have permission to log into a hotel enabled phone.

To Enable Hotelling

1) From Voice / Stations / User Accounts, create or edit the analog Voice User that will allow hotelling.

2) From the voice User's Current Settings tab, enable the Hotelling option and then click Apply.

Hotelling must be enabled on the phone that will allow Hotelling and it must be enabled for the users that will be allowed to log into a Hotel enabled phone.

Note: The User will also need to be assigned to a Class of Service that permits the use of the Hotel feature.

Logging into a Hotel enabled phone

From the hotel enabled analog phone issue the following SPRE codes to login or logout:

Hotel Login: ***46xxxx#pppp#** (*HO)

xxxx: Virtual user's account number
pppp: Virtual user's password

Hotel Logout: ***47pppp#**   (*HQ)

Virtual User Status

When a virtual user is logged into a hotel enabled phone, a ? bubble will display next to the users port. If you place your cursor over the bubble, the login status of the virtual user or hotel enabled phone will display.

Voice Stations - Ring Groups/Operator Group

NetVanta IP Telephony Course

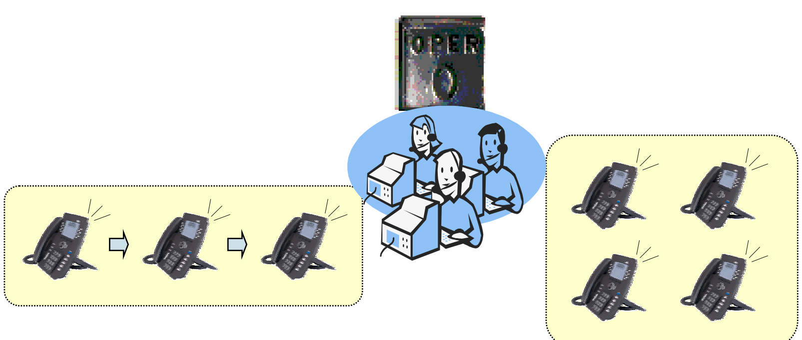
Voice Stations

Ring Groups/Operator Group

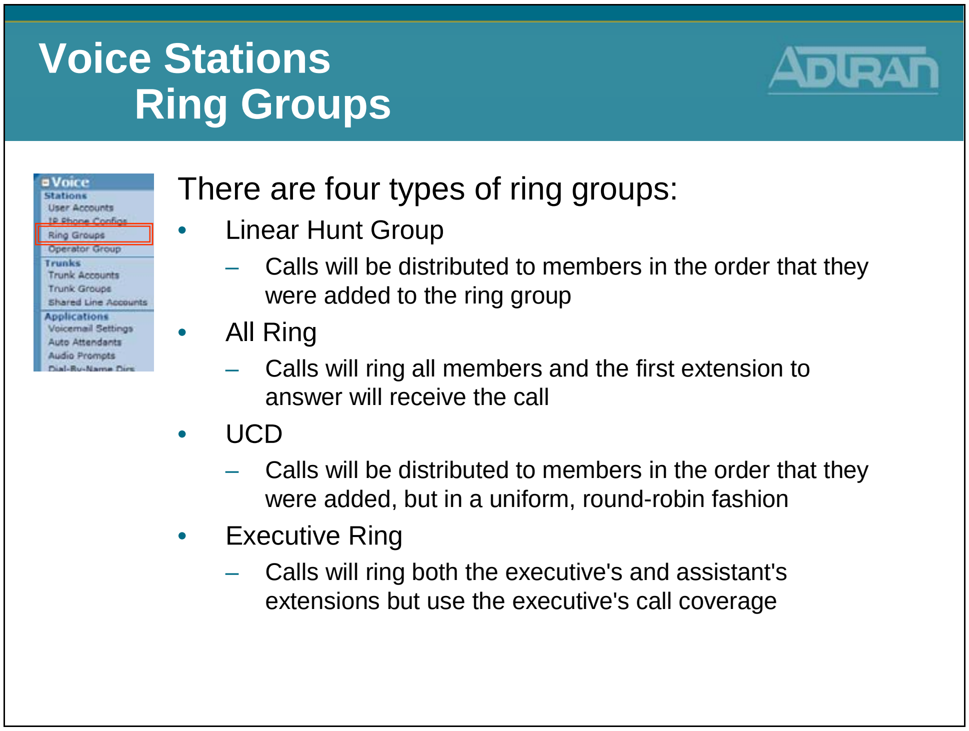
Voice Stations - Ring Groups

Voice – Stations
Ring/Operator Group Menu

- Define a group of user accounts that can be called in a coordinated way with a single extension
  - The incoming caller ID from a group member denotes a group call with a "GRP:" prefix



Voice Stations - Ring Groups



Ring All Ring Group

- Rings all members simultaneously
- Members can login or logout
- Group call coverage; single voice mail box for the group
- Call-waiting disabled while on a group call and receive a group call

Linear Ring Group

- Rings members one at a time, always starting with the first member in the group
- Members can login or logout
- Group call coverage; single voice mail box for the group
- Call-waiting disabled while on a group call and receive a group call

Uniform Call Distribution (UCD) Ring Group

- Rings members one at a time, starting with the next member
- Members can login or logout
- Group call coverage; single voice mail box for the group
- Call-waiting disabled while on a group call and receive a group call

Executive Ring Group

- Members include executive and assistant extension
- Rings both members
- Uses executive's call coverage for voice mail

Ring Group Configuration

1. Select Voice / Stations / Ring Group from the NetVanta 7000 menus
2. Assign extension and description
3. Select Ring Group Type (All, Linear, UCD, Executive)
4. Add members (voice users) to ring group
5. Define max calls allowed into ring group
6. Configure Call Coverage and Voicemail settings for ring group

Ring Group - Logging in and out of group

Ring Group
Logging in and out of group

- The Admin can log users in and out from the Voice / Stations/ Ring Group screen
- Members can login to a Group with a SPRE code when they want to receive calls to the group and logout using a SPRE code when they do not want to receive calls to the group
  - From the desired phone:
    - Group Login: *55xxxx* (*LL)
      xxxx: Group number
    - Group Logout: *56xxxx* (*LO)

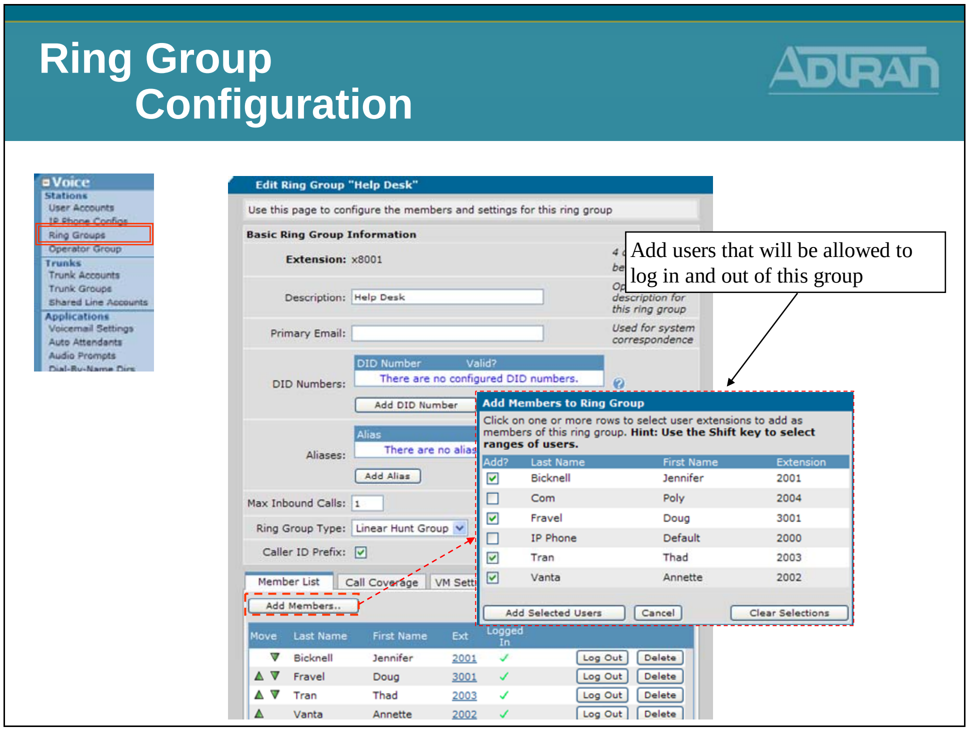
Logging in and out of Groups

Members can login to a Group with a SPRE code when they want to receive calls to the group and logout using a SPRE code when they do not want to receive calls to the group.

From the desired phone, enter the following SPRE codes to login or logout:

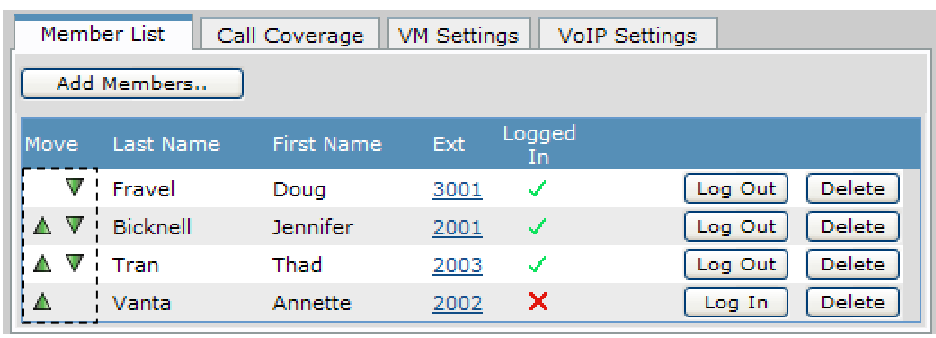
Group Login:	***55xxxx*** (*LL)
	xxxx: Group number
Group Logout:	***56xxxx*** (*LO)

Ring Group - Configuration



If the Caller-ID Prefix option is selected, when a call comes into the group, incoming Caller_ID displays "GRP:" along with the originating Caller-ID.

Voice / Stations / Ring Groups

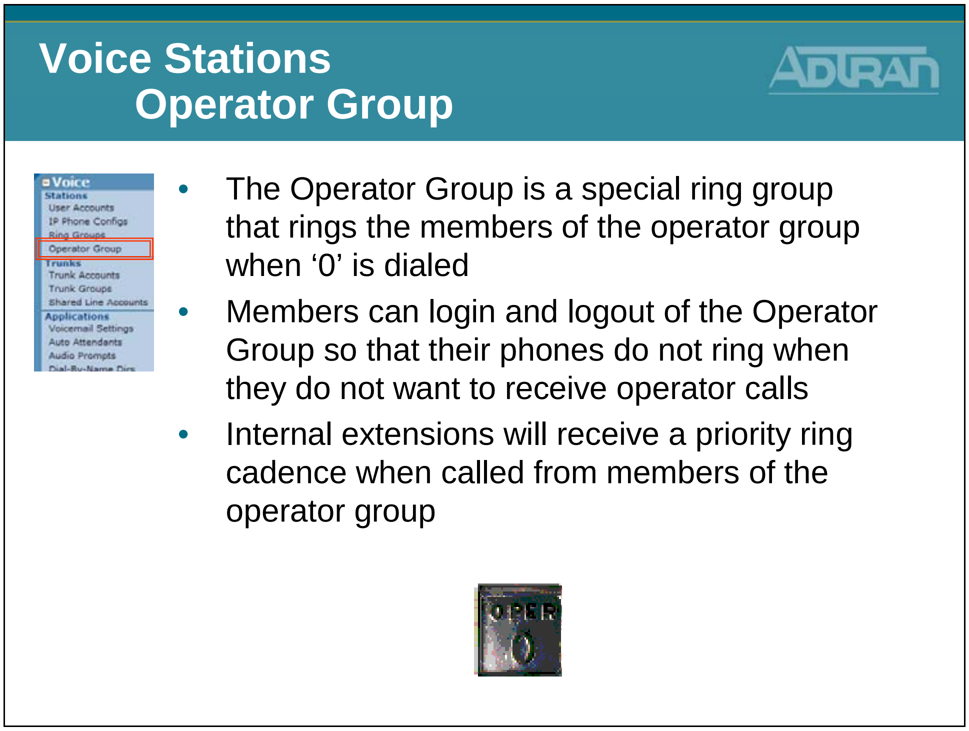


The up and down arrows can be used to change the call progression order of the group's voice users

Member status

Members can be logged in or out of a group

Voice Stations - Operator Group



Operator Group

- Rings all members simultaneously
- Members can login or logout
- Group call coverage; single voice mail box for the group
- Internal extensions receive priority ring cadence when called from operator extensions
- Configured to use Linear Ring, UCD Ring, or All Ring
- Optional Operator Calling-Party ID

Operator Group Configuration

1. Select Voice / Stations / Operator Group from the NetVanta 7000 menus
2. Select Group Type (All, Linear, UCD)
3. Add members (voice users) to group
4. Define max calls allowed into group
5. Configure Call Coverage and Voicemail settings for Operator group

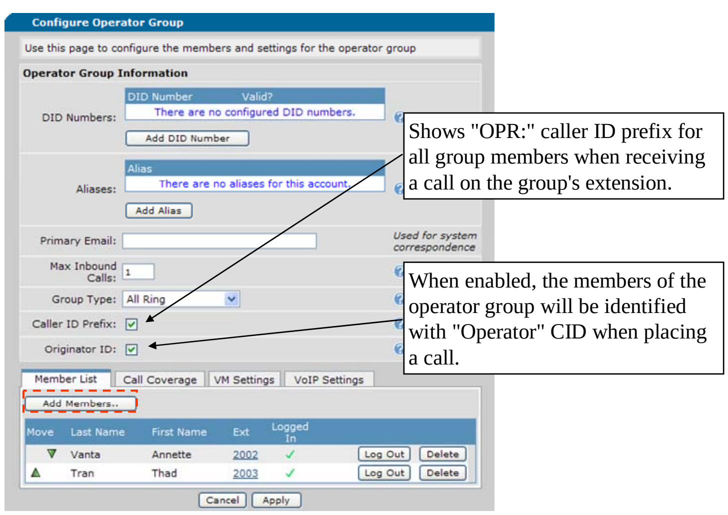
Operator Group - Logging in and out of group

Operator Group
Logging in and out of group

- The **Admin** can log users in and out from the Voice / Stations/ Operator Group screen
- **Members** can login to Operator Group with a SPRE code when they want to receive calls to the group and logout using a SPRE code when they do not want to receive calls to the group
  - From the desired phone:
    - Operator Group Login: ***550*** (*LL
      0 is Group #
    - Operator Group Logout: ***560*** (*LO)

Operator Group - Configuration

Operator Group
Configuration



Voice Ring Group / Operator Group Settings

The Ring Group / Operator Group Settings are the settings that can be seen or modified while editing a ring group. When a new ring group is created, you are placed in the Edit <Ring Group> screen where the settings below display.

Editing Ring Group / Operator Group - Initial Screen

Extension
The extension associated with this ring group

Description
Optional description for this ring group

Primary Email
Used for system correspondence

DID Numbers
Configures DID numbers for this account. The table shows all existing DID numbers (you may have to scroll to see all of them) and whether each number is currently valid. A number is considered valid if it matches any trunk's DID prefix and digit count. If no DID information has been configured in trunks, then all numbers are considered valid.

- To add a new DID number, click the Add DID Number button just below the DID Number table and enter the DID number in the popup box.
- To delete a DID number, click the Delete button next to the number you want to delete.

Aliases
Configures aliases for this account. The table shows all existing aliases (you may have to scroll to see all of them).

- To add a new alias, click the Add Alias button just below the Alias table and enter the new alias for this account in the popup box.
- To delete an alias, click the Delete button next to the alias you want to delete.

Max Inbound Calls
Enter the number of concurrent inbound calls allowed into this group (1-9). Any further concurrent calls will go directly to call coverage.

Ring Group Type

- **Linear Hunt Group** - Calls will be distributed to members in the order that they were added to the ring group.
- **All Ring** - Calls will ring all members and the first extension to answer will receive the call.
- **UCD** - Calls will be distributed to members in the order that they were added, but in a uniform, round-robin fashion.

- **Executive Ring -** Calls will ring both the executive's and assistant's extensions but use the executive's call coverage.

Caller ID Prefix (Ring Group)
Shows "GRP:" caller ID prefix for all group members when receiving a call on the group's extension.

Caller ID Prefix (Operator Group)

Shows "OPR:" caller ID prefix for all group members when receiving a call on the group's extension.

Originator ID (Operator Group)
When enabled, the members of the operator group will be identified with "Operator" CID when placing a call.

Editing Ring Group / Operator Group – Members List Tab

The Members List tab displays all the users that are in this ring group. Once a member has been added, the move arrows can be used to change the order of the member in the group. It also displays the status of which members are currently logged into the group.

Add Members Button
Click on one or more rows to select user extensions to add as members of this ring group. Hint: Use the Shift key to select ranges of users.

Log In / Log Out Button
Members can be logged in or out of a group by the admin

Editing Ring Group / Operator Group – Call Coverage Tab

Define what happens when a call is not answered by members of this ring group. A call will always follow the ring group’s call coverage, not the individual members call coverage.

Editing Ring Group / Operator Group – VM Settings Tab

The VM Settings tab allows you to edit the user’s voicemail settings such as VM Phone Indication and VM Notification Schedule.

Voicemail PIN

Sets the password the user must enter to access the voicemail system. Password must be 4 digits.

VM Class of Service

The voicemail class of service assigned to this ring grop

VM Operator Assist #
This number will be dialed if a caller requests to speak with the operator while leaving a voicemail.

New User Reminder
Checking this box alerts the Voicemail system to prompt the user to record their name. The recorded name is subsequently used for playback within the system.

Play Envelopes
When enabled, envelopes preceding voice messages will be played. An envelope includes the Calling party and the Date/Time information about a message.

Auto-play messages
When enabled, voice message playback will begin automatically after logging into your voice mailbox.

Authentication
Choose the authentication method to be used when logging into your voice mailbox. From valid phones, authenticate using:

- mailbox/password
- password only
- no authentication

WARNING: Selecting "None" will allow anyone who knows your extension to hear your messages.

Notification Type Primary Email (future)
When being notified that a voicemail has been left, the type of notification may be chosen.

- Select between NOT being notified via email, an email that contains only text, or an email that has the voicemail message attached in WAV format.
- The Operator Group and Ring Groups simply need to have their email addresses configured to begin receiving voicemail notifications.
- User Accounts, however, must define a notification schedule for this setting to have an effect.

Greeting Method
Choose the greeting that will be heard by callers leaving voice messages. The Default greeting is your recorded name. To record Standard and Alternate greetings, login to your voicemail via your phone and follow the instructions under the Greetings menu.

Editing Ring Group / Operator Group – VoIP Settings Tab

The VoIP Settings tab allows you to configure SIP Identities for this ring group.

SIP Identity Settings

Configures SIP Identities for this account. The table shows all existing SIP Id's (you may have to scroll to see all of them).

- To add a new SIP Id, click the Add SIP Identity button just below the SIP Identity table and enter the new SIP Id for this account in the popup box.
- Click the Delete button next to the SIP Id entry if you wish to remove it.

SIP Identity
Enter this user's SIP Identity. Currently, this value must be equal to the user's extension.

Associated SIP Trunk
Select the SIP Trunk this station will use for registration purposes.

Trunk Registration
Select whether or not this user should register with selected the SIP Trunk.

Trunk Authentication
Optionally, set the authentication information for this station. If 'Not Set' is chosen, the unit will use the registration trunk authentication data if it exists. Otherwise, no authentication data will be sent.

Trunks

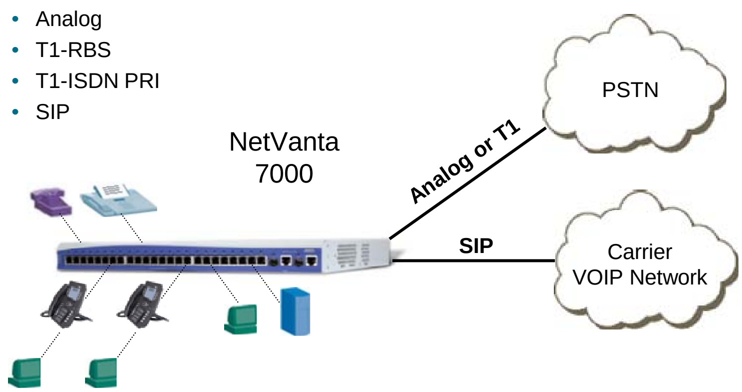
NetVanta IP Telephony Course

Voice Trunks

Voice - Trunks

Voice - Trunks

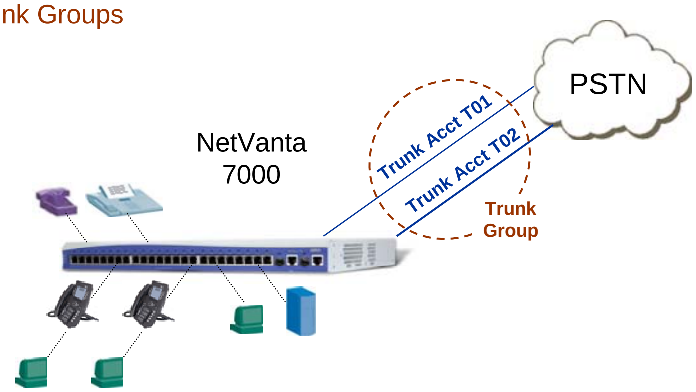
- **Trunk** lines connect the NetVanta 7000 to the outside world. They are delivered from the carrier and may be digital or analog.
  - NetVanta 7000 Supported Trunk Types
    - Analog
    - T1-RBS
    - T1-ISDN PRI
    - SIP



NetVanta 7100 Voice Trunks

NetVanta 7000 Voice Trunks

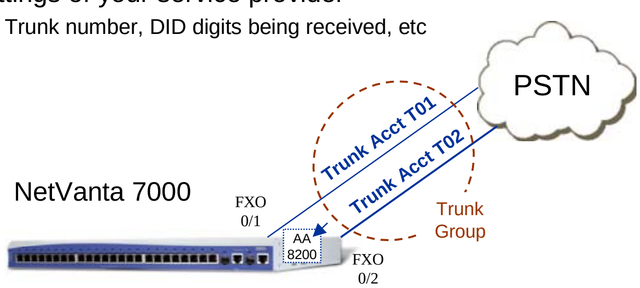
- NetVanta 7000 Trunk Components
  - Trunk Accounts
  - Trunk Groups



NetVanta 7100 Voice Trunks - Trunk Accounts

NetVanta 7000 Voice Trunks Trunk Accounts

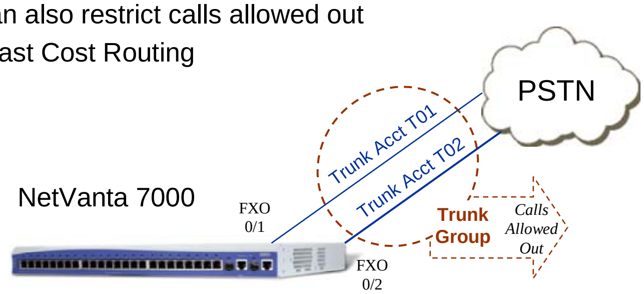
- Trunk Accounts are created to define the following trunk line parameters:
  - Type of Trunk
    - Analog, RBS, PRI, or SIP
  - Physical interface
    - FXO or T1
  - Settings of your service provider
    - Trunk number, DID digits being received, etc



NetVanta 7100 Voice Trunks - Trunk Groups

NetVanta 7000 Voice Trunks Trunk Groups

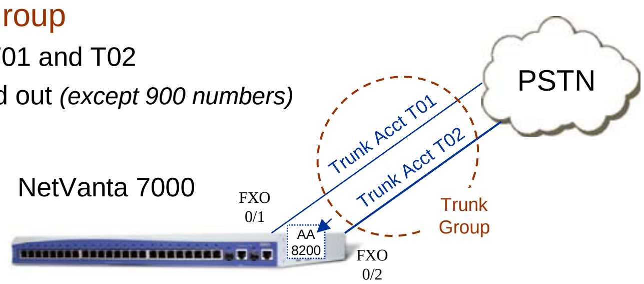
- Trunk Groups combine one or more Trunk Accounts
- Trunk Groups control the following:
  – Resources used for outbound calls
  – Outbound Call Templates are assigned to the Trunk Group to define calls allowed out this Trunk Group
  – Can also restrict calls allowed out
  – Least Cost Routing



NV 7100 Voice Trunks - Factory Default Config

NetVanta 7000 Voice Trunks Factory Default Config

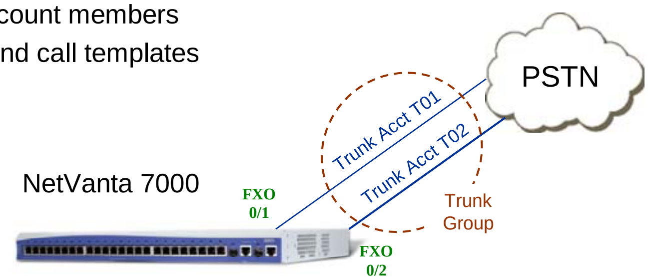
- Trunk Account T01
  – Physical Interface = FXO 0/1
  – Inbound call goes to Auto Attendant (8200)
- Trunk Account T02
  – Physical Interface = FXO 0/2
  – Inbound call goes to Auto Attendant (8200)
- Analog Trunk Group
  – Includes both T01 and T02
  – All calls allowed out *(except 900 numbers)*



Analog Trunk - Basic Configuration Steps

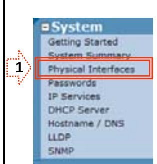
Analog Trunk
Basic Configuration Steps

1. Configure Trunk Physical Interface
   – FXO interfaces enabled by default
2. Create Trunk Account
   – Configure trunk number, caller-id, etc..
   – Assign FXO port(s)
3. Create Trunk Group
   – Add Trunk Account members
   – Define outbound call templates

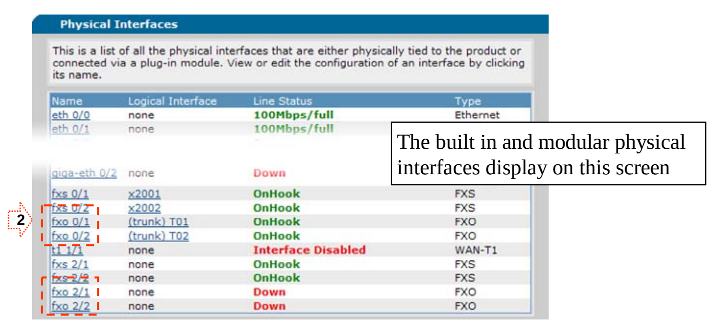


1) Configure Physical Interface

Analog Trunk Configuration
1) Configure Physical Interface



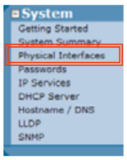
1. Select the System / Physical Interfaces menu



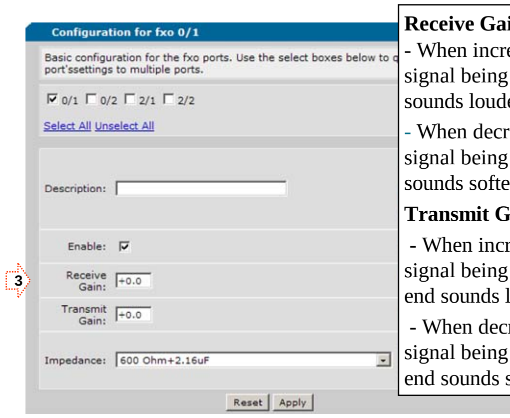
2. Click the FXO interface to be edited
   – *FXO interfaces are enabled by default*

1) Configure Physical Interface

Analog Trunk Configuration
1) Configure Physical Interface



3. *Optional*: Interface Gain and Impedance can be adjusted if needed



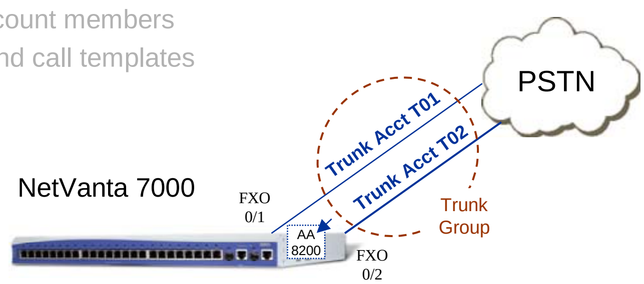
Receive Gain:
- When increasing this value, the signal being received on this port sounds louder
- When decreasing this value, the signal being received on this port sounds softer

Transmit Gain:
- When increasing this value, the signal being transmitted to the far end sounds louder
- When decreasing this value, the signal being transmitted to the far end sounds softer

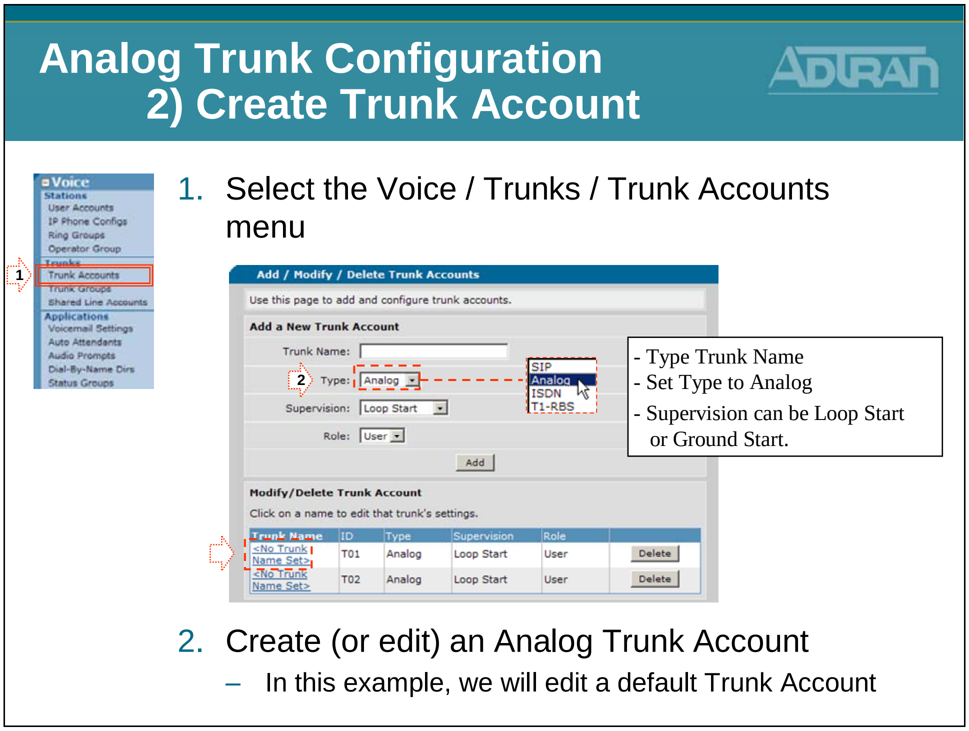
Analog Trunk - Basic Configuration Steps

Analog Trunk
Basic Configuration Steps

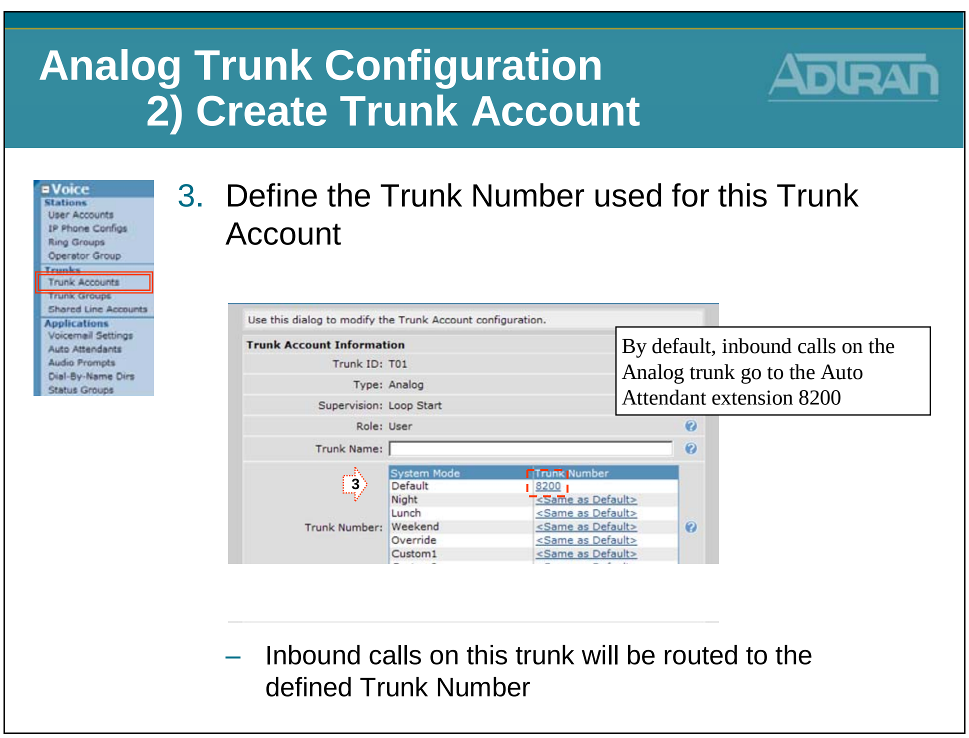
1. Configure Trunk Physical Interface
   – FXO interfaces enabled by default
2. Create Trunk Account
   – Configure trunk number, caller-id, etc..
   – Assign FXO port(s)
3. Create Trunk Group
   – Add Trunk Account members
   – Define outbound call templates



2) Create Trunk Account

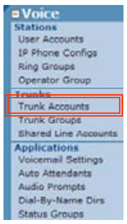


2) Create Trunk Account

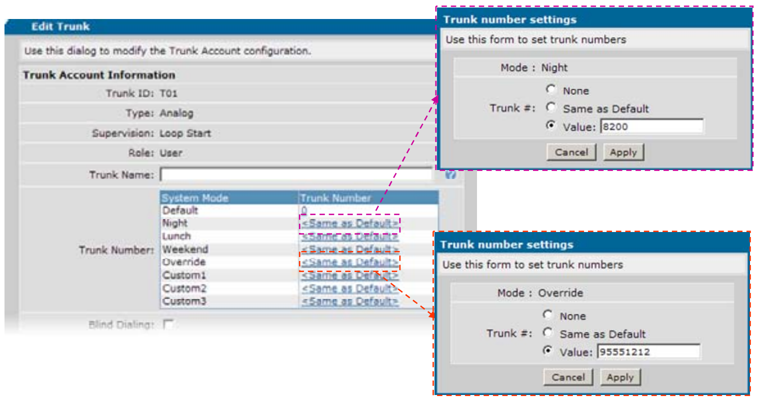


2) Create Trunk Account

Analog Trunk Configuration
2) Create Trunk Account



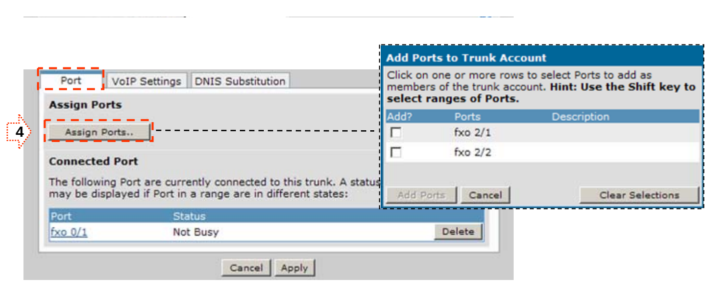
- Optional - Define a Trunk Number per System Mode
  – None / Same as Default / Value – *extension/number*



2) Create Trunk Account

Analog Trunk Configuration
2) Create Trunk Account

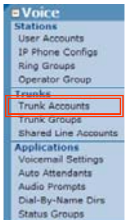
4. Specify the physical interface(s) this trunk account will use for voice calls



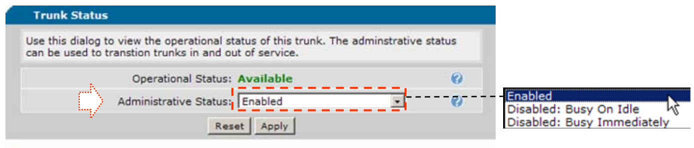
– Click one or more FXO interfaces to be added

2) Create Trunk Account

Analog Trunk Configuration
2) Create Trunk Account



- *Optional*: Administrative Status
  - Enabled by default

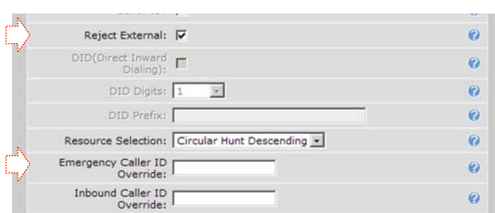


  - Configurable Trunk status Options:
    - **Enabled** - Trunk operates as normal
    - **Disabled: Busy On Idle** - Current calls stay active, but no new calls are accepted
    - **Disabled: Busy Immediately** - All current calls are torn down, and no new calls are accepted

2) Create Trunk Account

Analog Trunk Configuration
2) Create Trunk Account

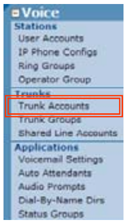
- *Optional*: Reject External
  - Unselect to allow trunk to trunk calls



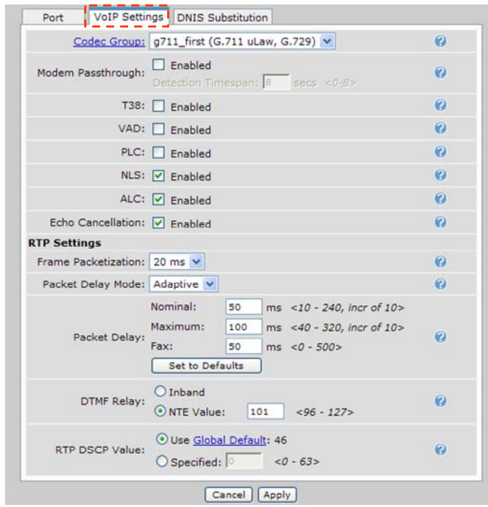
- *Optional*: Emergency Caller ID Override
  - Specify the calling party number presented on outbound emergency calls

2) Create Trunk Account

Analog Trunk Configuration
2) Create Trunk Account



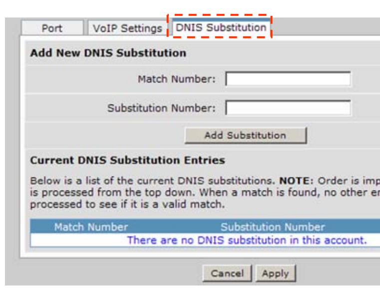
- *Optional*: Adjust VoIP settings for this interface



2) Create Trunk Account

Analog Trunk Configuration
2) Create Trunk Account

- *Optional*: Add DNIS substitution



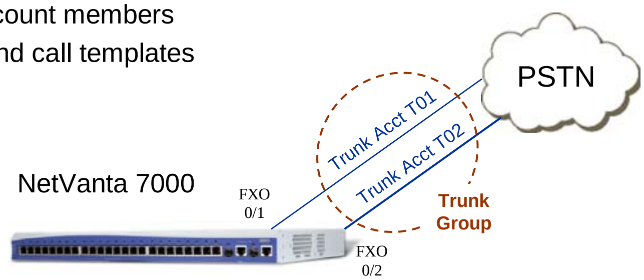
Order is important:
- Multiple match statements can be entered per trunk account
- The first valid match that is found for outbound numbers will be used

  – Examples:
    - Match: NXX-XXXX Subst: 256-NXX-XXXX
    - Match: 1-NXX-XXX-XXXX Subst: NXX-XXX-XXXX
    - Match: 1-NXX-NXX-XXXX Subst: 10-10-220-NXX-NXX-XXXX

Analog Trunk - Basic Configuration Steps

Analog Trunk
Basic Configuration Steps

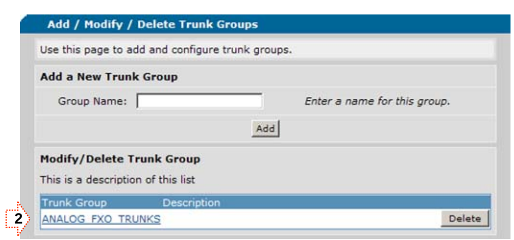
1. Configure Trunk Physical Interface
   - FXO interfaces enabled by default
2. Create Trunk Account
   - Configure trunk number, caller-id, etc..
   - Assign FXO port(s)
3. Create Trunk Group
   - Add Trunk Account members
   - Define outbound call templates



3) Create Trunk Group

Analog Trunk Configuration
3) Create Trunk Group

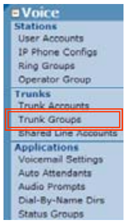
1. Select the Voice / Trunks / Trunk Groups menu



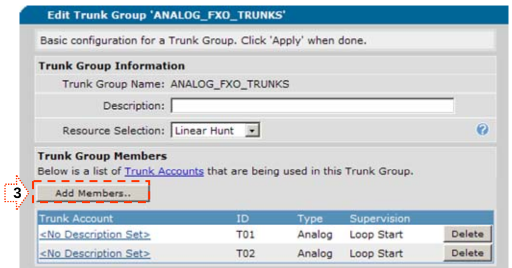
2. Create (or edit) a Trunk Group
   - In this example, we will edit the default Trunk Group

3) Create Trunk Group

Analog Trunk Configuration
3) Create Trunk Group



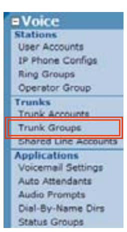
3. Click Add Members to add existing Trunk Accounts to this Trunk Group



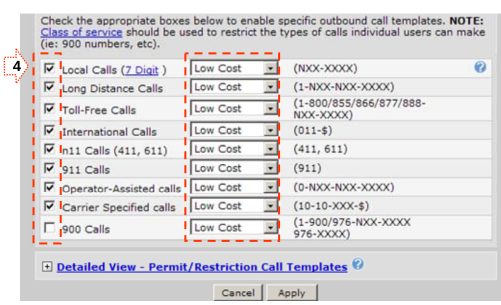
– Multiple Trunk Accounts can belong to a Trunk Group as long as they share the same calling plan

3) Create Trunk Group

Analog Trunk Configuration
3) Create Trunk Group



4. Outbound Call Template
   – Define call types allowed out this Trunk Group



– *Optional*: Define cost for each type of call
  • Least cost routing

Introduction to Voice Troubleshooting

NetVanta IP Telephony Course

Introduction to Voice Troubleshooting

Introduction to Voice Troubleshooting

Introduction to Voice Troubleshooting

- A Few Components to consider in a Converged Voice and Data network
- Endpoints
  - SIP phones, Softphone, Analog phones, FAX, etc…
- Interfaces
  - Ethernet, FXS, FXO, T1, PRI, etc…
- Services
  - DHCP, Boot Server, Registration Server, SIP Server, etc…
- Data network
  - Switching, Routing, Firewall, VPN, QoS, Data Services, etc...
- Voice network
  - Voice User, Dial Plan, CoS, Trunks, QoS, Voicemail, etc…

Introduction to Voice Troubleshooting

Introduction to Voice Troubleshooting

Digits gathered – number type match?	***Dial Plan***
Permission to call number?	***Class of Service***
Local Station match?	***Voice Users***
Trunk match? Resource available?	***Trunk Groups***
Dialed user's location known? Available?	*SIP Proxy* *SIP Registration*
User's Phone functional?	*DHCP Server,* *Boot Server,* *SIP Registration*

Voice Troubleshooting

Voice Troubleshooting

- The NetVanta 7000 provides numerous voice troubleshooting commands. Below are a few that we will introduce in this module:
  - show run voice
  - show run voice user
  - show run voice verbose
  - show voice users
  - debug voice summary
  - undebug all

show run voice

show run voice

- Display only voice running configuration

```
NV7000# show run voice
Building configuration...
!
voice feature-mode local
voice forward-mode local
!
voice dial-plan 0 always-permitted 911
voice dial-plan 1 always-permitted 9-911
voice dial-plan 2 internal-operator 0
voice dial-plan 3 extensions MXXX
voice dial-plan 4 local 9-NXX-XXXX
voice dial-plan 5 long-distance 9-1-NXX-NXX-XXXX
voice dial-plan 6 toll-free 9-1-800-NXX-XXXX
           :
!
voice class-of-service normal_users
 override-passcode 6789
 default-level
 block-caller-id
           :
```

* Partial output displayed

show run voice user

show run voice user

- Display voice user configuration

```
NV7000# show run voice user
Building configuration...
!
voice user 2000
 connect sip
 cos "public_phones"
 first-name "Default"
 last-name "IP Phone"
 password "1234"
     :
     :
voice user 2001
 connect fxs 0/1
 cos "normal_users“
     :
```

* Partial output displayed

show run voice verbose

show run voice verbose

- Display detailed voice running configurations

```
NV7000# show run voice verbose
Building configuration...
!
voice prompt-language English
!
voice country-code 1
voice international-prefix 011
no voice international-prefix abbreviated
voice transfer unattended
!
voice overhead-paging extension 8000
!
voice feature-mode local
voice flashhook threshold 300 1000
voice timeouts interdigit 4
voice timeouts connected 12
voice timeouts alerting 5
voice hold-reminder 10 30
voice park-return 60
        :       :
```

* Partial output displayed

show voice Commands

show voice commands

```
NV7000# show voice ?
alias              - display voice alias configuration
ani                - ani substitution parameters
available          - list fxs ports that are not associated with a user
dial-plan          - number complete templates
did                - direct inward dialing
directory          - show directory(s) and included users
door-phone         - display the door-phone account
extensions         - current voice extensions and status
grouped-trunk      - voice trunk groups
line               - voice line stations
loopback           - Show status on loopback accounts
mail               - display voicemail information
operator-group     - ring groups
phone-files        - files required for sip phone configuration
quality-stats      - display voice quality stats for all calls
ring-group         - ring groups
service-mode       - current voice service mode
speed-dial         - system speed dial
spre               - view spre (special prefix) codes
status-group       - status groups
switchboard        - voice switchboard extensions
system-mode        - Current voice system mode
trunk              - voice trunks
users              - voice user stations
```

show voice users

show voice users

- Display all voice stations

```
NV7000# show voice users

First           Last       Ext    Interface  Description
--------------------------------------------------------------------------------
Default         IP Phone   2000   ip
Analog FXS      Port 0/1   2001   fxs 0/1
Analog FXS      Port 0/2   2002   fxs 0/2
South           Lobby      2003   fxs 2/1
John            Smith      2004   virtual
Thad            Tran       2005   ip
Annette         Vanta      3001   virtual

Total number of configured voice users:  7
```

debug voice Commands

debug voice commands

```
NV7000# debug voice ?
<cr>
account-status      - station account-status events
autoattendant       - autoattendant events
dsp                 - DSP events
lineaccount         - line account events
linemanager         - line manager events
loopback            - Loopback events
mail                - voicemail events
phoneconfig         - ip phone config utility events
phonemanager        - phone manager events
promptstudio        - prompt-studio events
proxydial           - proxy dial events
rtp                 - rtp events
smdr                - smdr events
stationaccount      - station account events
statusgroups        - status group events
summary             - simple voice events
switchboard         - switchboard events
toneservices        - tone services events
trunkaccount        - trunk account events
trunkmanager        - trunk manager events
trunkport           - trunkport events
verbose             - detailed voice events
```

debug voice summary

debug voice summary

- Summarize voice events

```
NV7000# debug voice summary
16:55:22 VOICE.SUMMARY voice user 2005 cos allowed the call to Extensions
16:55:22 VOICE.SUMMARY 2005 is calling 2006 (2006).
16:55:24 VOICE.SUMMARY 2005 is connected to 2006 (2006)
16:55:28 VOICE.SUMMARY Call from 2005 to 2006 (2006) ended by 2006: normal clearing

17:01:54 VOICE.SUMMARY voice user 2006 cos allowed the call to Extensions
17:01:54 VOICE.SUMMARY 2006 is calling T01 (911).
17:01:56 VOICE.SUMMARY 2006 is connected to T01 (911)
17:02:04 VOICE.SUMMARY Call from 2006 to T01 (911) ended by T01: normal clearing
```

Turning off Debug

Turning off Debug

- Turn off one debug command

```
NV7000# no debug <specific debug command>
```

- Turn off all active debug commands

```
NV7000# undebug all
```

- Show active debug commands

```
NV7000# show debug
```

Module Summary

Module Summary

- At the end of this module, you should be able to:
- Understand basic call routing
- Modify the NetVanta 7000 Dial Plan
- Create and modify Voice Classes of Service
- Create and modify Voice User Accounts
- Create and modify Ring Groups/Operator Group
- Configure Analog Voice Trunks
- Perform basic Voice Troubleshooting

Module 4: ADTRAN Phone Configuration Files

Module Objectives

Module Objectives

- Introduce the ADTRAN/Polycom IP Phones
- Introduce the ADTRAN/Polycom phone config files
- Modify phone configuration files
- Troubleshoot the boot process of the ADTRAN IP 700 Series phone

ADTRAN IP 700 Series Phones

ADTRAN IP 700 Series Phones

- Two Models:
  - IP 706: 6 lines
  - IP 712: 12 lines
- Supports Multiple SIP registrations
- Busy Lamp Field and Shared Line Appearance Support
- High Quality Full Duplex Speaker Phone
- Dual 10/100 Switched Ethernet Ports
- Large Backlit Display
- 802.3af Power over Ethernet
- Adjustable base stand
- Wall mountable
- Headset jack with Electronic Hook Switch Detection

The ADTRAN IP phones are available in either 6 line or 12 line versions, supporting multiple call functions. Dedicated keys are available for the most common user functions with additional programmable soft keys. On-screen menus enable users to quickly change directory information and phone settings, as well as view a history of internal/external and missed calls, and program distinctive ring tones for specific calls. The phones include an adjustable desk stand or can be wall mounted and feature high-quality, full duplex speakers engineered for clear, hands-free communication. An integrated headset jack with electronic hook-switch eliminates the need for a mechanical handset lifter. The overall enhanced functionality for the price makes ADTRAN IP phones among the most cost-efficient business-class IP phones.

The ADTRAN 700 Series features an intuitive, Graphical User Interface (GUI) for easy set-up and installation. The phones can be directly powered from the NetVanta® 7000 Series or a Power over Ethernet (PoE) switch, providing inline power and eliminating the need for a separate power supply. The phones also have two Ethernet ports to connect to a PC for converged voice and data across a single wiring infrastructure. ADTRAN phones can be locally powered, allowing for multiple options for worry-free installation and ease of use.

Polycom Phones - Supported by ADTRAN

Polycom Phones Supported by ADTRAN

ADTRAN

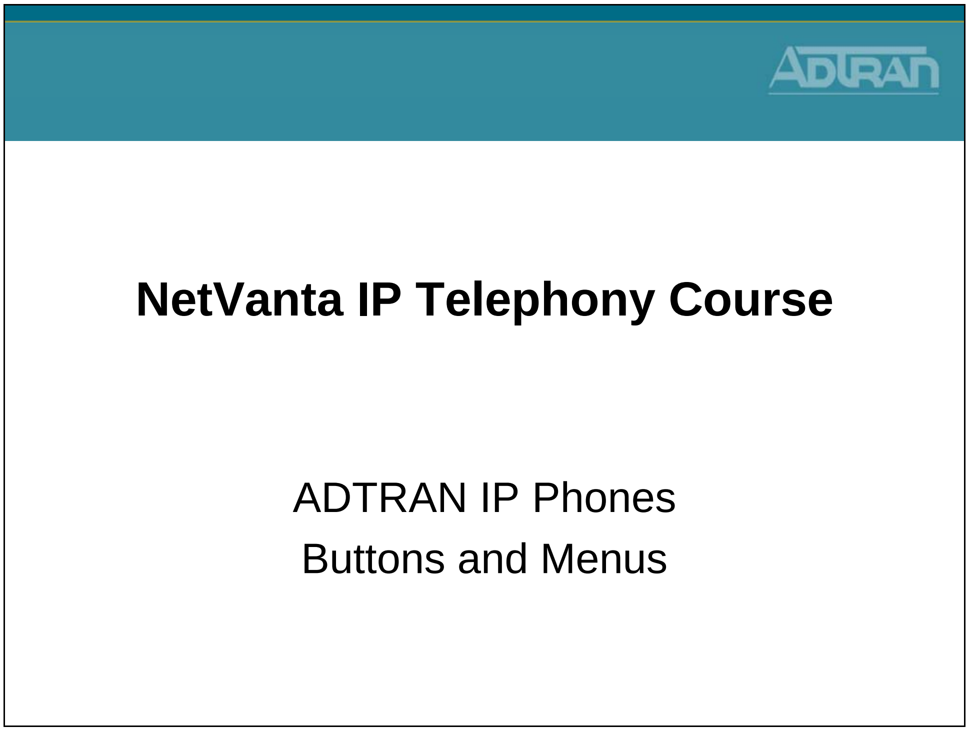
- **IP 430** – two lines
- **IP 650 –** High Definition Audio, six lines
- **IP 650 with Expansion Modules** – ideal for “power user” or attendant console, up to 48 lines
- **IP 6000** – conference room speaker phone
- Plus more…

ADTRAN and Polycom have joined forces to deliver a best of breed solution for the VoIP market

ADTRAN and Polycom have worked together to ensure interoperability of the Polycom SoundPoint IP 300, 400, 500, 600, 4000, and 6000 series of SIP phones with the ADTRAN IP Telephony solutions.

ADTRAN's NetVanta 7000 series also supports the Polycom SoundPoint 650 IP phone for multiline attendant applications or high definition voice clarity. The Polycom SoundPoint IP 650 incorporates Polycom's HD Voice Technology and wideband audio for over twice the voice quality and clarity. The IP 650 can also be equipped with up to three Expansion Modules for attendant console applications delivering up to 48 buttons.

Buttons and Menus



The next few pages are a basic guide to using the ADTRAN IP 700 series phone. For more detailed information, refer to the IP 700 Series Phone User Manual, as well as other resources available at: www.adtran.com/phones.

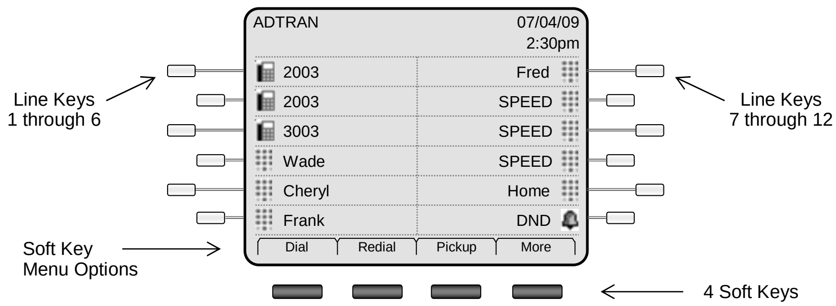
IP 706 Phone Diagram

IP 712 Phone Diagram

Line Keys

Line Keys

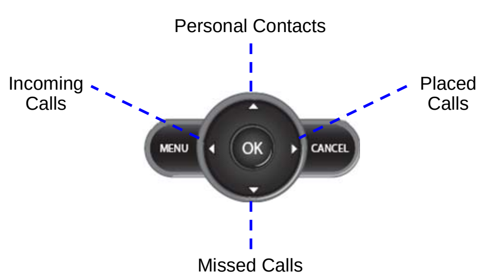
- Line keys can be programmed as one of the following:
  - secondary extension, SLA, or BLF/DSS
  - speed dial entry
  - a shortcut to the Do Not Disturb (DND) feature



Menu Navigation Bar

Menu Navigation Bar

- The navigation bar enables you to scroll through the menus presented on the LCD display, as well as make selections



ADTRAN IP 700 Series Phone Icons

Icon	Icon Name	Description
	On Hook/Idle	The line has registered with the SIP server and is available for use.
	Not Registered	The line has not registered with the SIP server and is not available for use.
	Alarm Bell	The line is receiving an incoming call.
	In Use	A call is active on the line.
	Speed Dial Entry	The line is set to speed dial.
	DND DND-Enabled	The first icon indicates that the line key is dedicated to the Do Not Disturb (DND) feature, but is not activated. Once the icon appears with an X through it, DND is activated and incoming lines will not ring.
	Hold	A call is on hold.
	Calls Forwarded	The line is forwarded to another extension or number.
	Call Conferenced	A three-way conference call is in progress on the line.
	Speaker with Volume	The plus (+) end of the volume control bar has been pressed to increase volume.
	Speaker with No Volume	The minus (-) end of the volume control bar has been pressed to decrease volume.
	Voice Mail	Indicates the user has voice mail.
	Line Seized	The line has been seized by another member of a ring group. This icon only displays to the members that did not answer the call. This icon displays for approximately 5 seconds before being replaced with the in-use icon.
	Progressing Ringback	The line is currently making a call.
	Busy Lamp Field	The line is set as a Busy Lamp Field (BLF) and is monitoring another phone that is not in use.
	Line Is In Use	The line is set as a Busy Lamp Field (BLF) and is monitoring another phone that is in use.

ADTRAN IP 700 Series Phone Function Keys

Function Key	Icon Name	Description
Messages	Messages LED illuminates Blue to indicate message waiting	The LED can be configured to illuminate solid, flash, or blink to indicate the message count. It can be set to directly access voice mail by pressing the message indicator key. Contact your system administrator for more information.
Hold	Hold	Press to place the current call on hold.
Transfer	Transfer	Press to initiate a call transfer.
Conference	Conference	A call is active on the line.
	Speed Dial Entry	Press to add a third party to an active call.
Directories	Directories	Press to access the System and Personal Contacts directories, as well as display placed, missed, and incoming call histories.
Goodbye	Goodbye	Press to disconnect from the current call.
Mute	LED flashes Red when active	Press to silence the speaker, handset, or headset microphone. Press the mute key again to reactivate audio.
Headset	LED illuminates Green when in use	Indicates that the headset is active. You must have a headset connected to your phone to use this function.
Speaker	LED illuminates Green when active	Press to enable the speaker.
Volume	Voice Mail + –	The ringer volume is adjusted using this function key while the phone is idle. The call volume is adjusted using this function key during an active call. Press the + (plus) end of the key to increase the volume or press the – (minus) end of the key to decrease the volume.

ADTRAN IP 700 Series Phone Functions

Forwarding Calls

To forward calls to another extension:

1. Press the **More** soft key on the idel screen.
2. Press the **Forward** soft key.
3. Enter the extension to which calls will be forwarded.
4. Using the navigation arrows, highlight **All** and press the **Enable** soft key. Press **Ok**.
5. To cancel call forwarding, select the **Forward** soft key and then select **Disable**.

Enabling Do Not Disturb (DND)

The DND feature prevents the phone from ringing or paging over the speaker when incoming calls are received. To enable:

1. Press **Menu**.
2. Press **3** for **Features**.
3. Press **2** for **DND Off**.
4. Select the **DND On** soft key.
5. Press the **Exit** soft key until the idle screen appears, or press **CANCEL** on the navigation bar to return directly to the idle screen.

Conferencing a Call

To conference a third party into the active call:

1. Press the Conference function key during an active call. The active call will be placed on hold, and the exclusive hold icon appears.

2. The next available line displays the ringback icon.

3. At the prompt, enter the phone number of the third party to add.
4. When the second call is connected, press the Conference key again to add the call to the conference. The conference icon will display.

Only three parties can be conferenced at a time. If one party disconnects, another party can be added.

Making a Call

To make a call using the handset, headset, or speaker:

1. Pick up the handset, or press the speaker key, or if using the headset, press the **Headset** key.
2. Listen for the dial tone.
3. Dial the desired number.

Answering a Call

To answer a call using the handset, headset, or speaker:

1. Pick up the handset, or press the headset key, or press the **Speaker** key.
2. If you have multiple incoming phone lines, press the key next to the extension receiving the call.

Ending a Call

To disconnect from a call, use one of the following:

- Press the **Goodbye** function key.
- Return the handset to the cradle.
- Press the headset key (if using the headset).
- Press the speaker key (if using speaker).

Adjusting LCD Contrast

1. Press **Menu**.
2. Press **2** for **Phone Settings**.
3. Press **5** for **Contrast**.
4. Press the **+** (plus) or - (minus) soft keys until the desired contrast is reached.
5. Press the **Ok** soft key or **OK** on the navigation bar.
6. Press the **Exit** soft key until the idle screen appears, or press **CANCEL** on the navigation bar to return directly to the idle screen.

Redialing a Number

To dial the last number called, press the **Redial** soft key on the idle screen. If the **Redial** soft key is not displayed, press the **More** soft key. The redial history screen will display. Use the navigation arrows to scroll to a previously dialed number, then press the **Dial** soft key.

Transferring a Call

To use unattended transfer:

1. During an active call, select the **Transfer** function key.
2. Dial the extension to which to transfer the call.
3. Press the **Transfer** key again when you hear the extension ring. This will disconnect you from the call.

To use attended transfer:

1. During an active call, select the **Transfer** function key.
2. Dial the extension to which to transfer the call.
3. Listen for the second call to connect.
4. Press the **Transfer** key to transfer the call.
5. If the party does not answer, press the **Cancel** soft key to disconnect the new call and return to the original call.

Directory and Call History Shortcuts

Use the arrows on the Navigation Bar to quickly access the Personal Contacts Directory, Placed Calls List, Missed Call List, or Incoming Calls List.

Phone Feature Quick Reference

Place a Call Pick up handset or press the **Speakerphone** button. Enter the desired number or enter the number on the keypad. Then press the **Dial** soft key.

Answer a Call Pick up the handset, press the **Answer** soft key, or the **Speakerphone** button.

Hold Once a call is established, press the **Hold** button (or **Hold** soft key) to place the caller on hold. To retrieve a call on hold, press the **Hold** button, **Resume** soft key, or the **Line** key.

Mute While a call is active, press the **Mute** button to mute the audio you are sending to the other party. Press the **Mute** button again to un-mute.

Unattended Transfer Once a call is established, press the **Transfer** key or **Transfer** soft key and enter the target's extension. Once the phone starts ringing, press the **Transfer** key (or **Transfer** soft key) again to complete the transfer, or simply hang up to complete the transfer.

Attended Transfer Once a call is established, press the **Transfer** key or **Transfer** soft key and enter the target's extension. Once the target has answered, announce the caller then press the **Transfer** key (or **Transfer** soft key) to complete the transfer or hang up.

Disable Forwarding Press the **Forward** soft key, and then select **Disable**.

Do Not Disturb Press the **Do Not Disturb** button to enable or disable Do-Not-Disturb mode. Disable by pressing the **Do Not Disturb** button again.

Hands-free Auto-Answer Intercom Dial ** in front of any IP phone extension number to invoke hands-free auto-answer intercom.

Hands-free Auto-Answer Intercom Do not Disturb To Block hands-free intercom calls to your extension, Dial ***97x (where x = 1-Block, 0-Unblock**. (This feature is dependant upon users Class of Service.)

Access Call Lists To access the call lists, press the **Call Lists** (IP501) or the **Directories** (IP601) button. Use the up/down arrows to scroll through the call lists. Press the **Select** soft key to select a call list. Press the **Exit** soft key to exit the call lists.

Blind Transfer Once a call is established, press the **Transfer** key or **Trnsfer** soft key, then the **Blind** soft key and enter the target extension.

Park Call Once a call is established, press the **More** soft key, then press **Park**, enter a Park Zone number (0 to 9), then press the **Park** button again or use the Park Zone Busy Lamp Field (BLF).

Retrieve Parked Call Obtain dial tone. Press the **Pickup** soft key, enter the Park Zone number (0-9), and then press the **Retrieve** soft key to pickup the call.

Page Obtain dial tone. Dial overhead paging extension or SPRE code (________). Page the party, then hang up.

Conference (Three-Way) While on a call, press the **Conference** button (or select the **More** soft key, then press the **Conference** soft key), and dial the third-party's extension. Once the party has answered, press the **Conference** button (or the **Conference** soft key) again to connect the parties.

Forward Call Press the **Forward** soft key. Enter the destination extension (or outside number), and then press the **Enable** soft key. When enabled, all incoming calls will be re-directed to the forwarded extension or number.

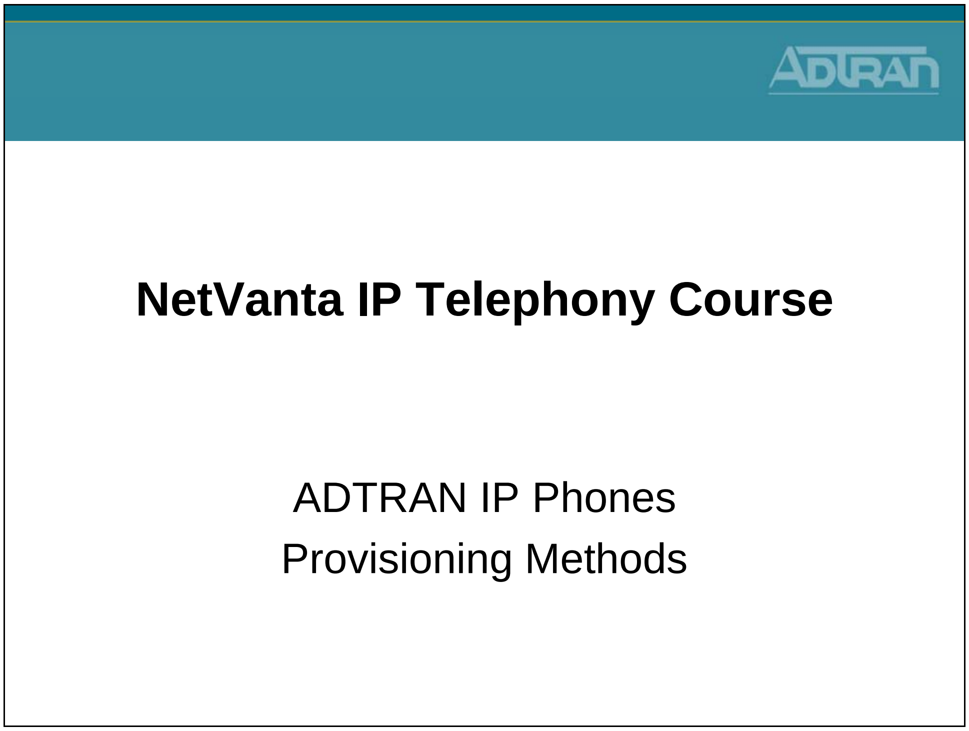
System Speed Dial Dial ***25** plus the two digit system speed dial number (00 to 99).

Speed Dial Programming Press the **Directories** button. Select **Contact Directory** from Directories menu. Press the **More** soft key, then press **Add**. Using the keypad, enter the First name, Last Name and Phone Number (contact). Press the **Save** soft key to save. Press the **Exit** soft key to exit the directory.

Speed Dialing Press the line key button that corresponds to the number you wish to dial.

Last Number Redial Press the **Redial** button to dial the last number that was dialed from the phone.

Provisioning Methods



Provisioning Method - Order of Precedence

- Parameters manually entered using either the phone’s LCD Menus (Phone Settings) or via the administrator’s Web interface (Phone Manager) have the highest priority and override parameters received from all other sources.
- Parameters received in a configuration file override those received from DHCP and defaults.
- Parameters returned by DHCP (if it is enabled) override default settings.
- Default parameters are used if no other source is available.

ADTRAN IP Phones - Provisioning Methods

ADTRAN IP Phones
Provisioning Methods

- Local phone based configuration
  - Local phone LCD Menus (Phone Settings)
    - Password = 1234
  - Web interface (Phone Manager - user)
    - <ip address>
    - Username = user
    - Password = password
  - Web interface (Phone Manager - admin)
    - <ip address>/admin
    - Username = admin
    - Password = password
- **Centrally Provisioned from Boot Server (NetVanta 7000)**
  - Consist of Global and per-phone configuration files
- DHCP
  - Can set a limited number of parameters
    - including the location of configuration files

MANAGING IP 700 SERIES PHONES

There are multiple ways to manage ADTRAN IP 700 Series phones, each providing a different management approach.

- Password-protected administrator's Web interface (Phone Manager) to view and change current settings on a single phone.
  - o <ip address>/admin or <ip address> (for user interface)
  - o Username = admin Username = user
  - o Password = password Password = password

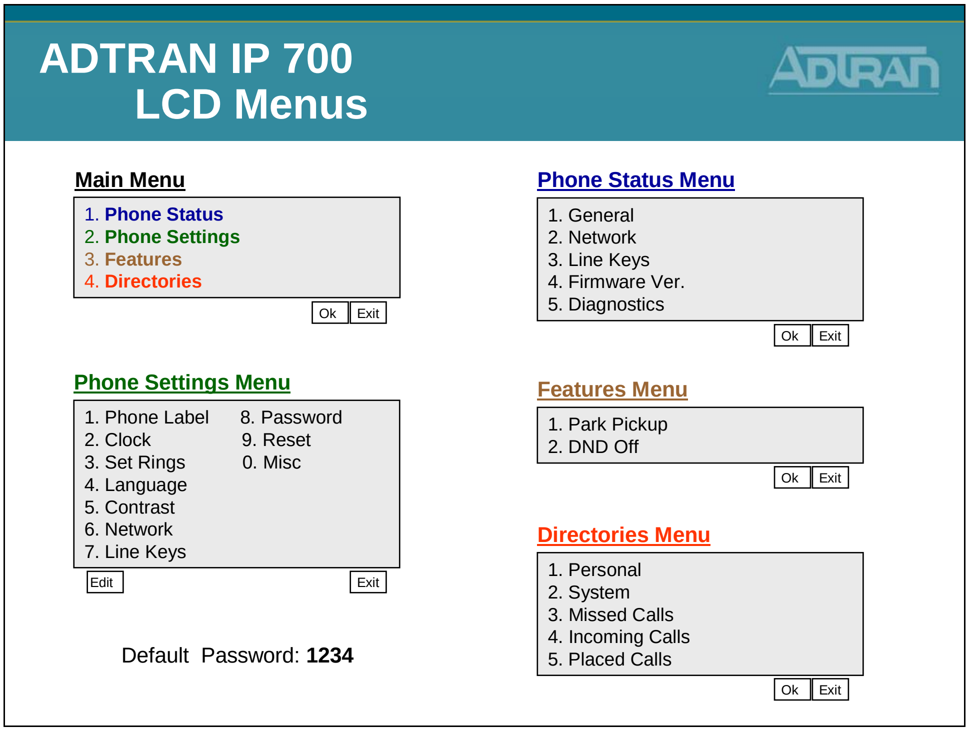
- Phone's LCD Menu to view and modify current settings locally.
  - o Password = 1234 (Changed to 456 after connected to NetVanta 7000)

- Configuration files to automatically download parameters upon phone startup and update firmware. (These files are created by the NetVanta 7000)

- Dynamic Host Configuration Protocol (DHCP) to set a limited number of parameters including the location of configuration files.

In this class, most configuration changes of the IP phones will be done from the NetVanta 7000 web interface. Visit www.adtran.com/phones to download the IP 700 Series Phone Administrator Guide for additional phone information.

ADTRAN IP 700 - User Interface Menus



The LCD menus provide another method for controlling and interfacing with the IP phone. Many programmable features of the phone can be accessed using the LCD menu. All keys, whether line, soft, or function keys, interact with the LCD menus.

Example Phones Settings that can be changed (See User Guide for others)

To change the **time/date format**, use the following steps:

1. Press Menu, then 2 for Phone Settings
2. Press 2 for Clock
3. Press 2 for Time Format or 3 for Date Format
4. Using the arrow keys on the navigation bar, scroll to the desired time format
5. Press the Select soft key to select the highlighted option
6. Press the Ok soft key or OK on the navigation bar
7. Press the Exit soft key until the idle screen appears, or press CANCEL on the navigation bar to return directly to the idle screen

To adjust the **LCD display contrast**, use the following steps:

1. Press Menu, then 2 for Phone Settings
2. Press 5 for Contrast
3. Press the + (plus) or - (minus) soft keys until the desired contrast is reached
4. Press the Ok soft key or OK on the navigation bar
5. Press the Exit soft key until the idle screen appears, or press CANCEL on the navigation bar to return directly to the idle screen

ADTRAN IP Phone - DHCP Provisioning Method

- The IP 700 Series phone uses site-specific Option 157 to provide the following information to the phones:
  - TftpServers=0.0.0.0
  - FtpServers=10.10.20.1:/ADTRAN
  - FtpLogin=polycomftp
  - FtpPassword=password
  - Layer2Tagging=True
  - VlanID=2

  * Option 157 must be set on both the LAN_pool and the VoIP_pool to direct the phones to the correct boot server.

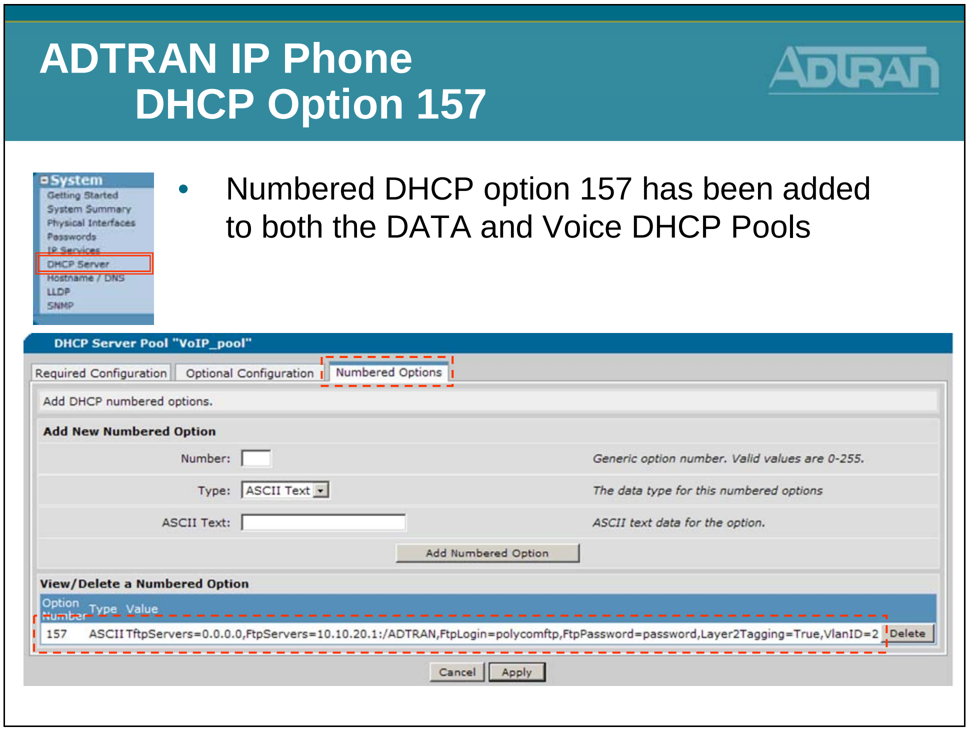
The NetVanta 7000 Series Product ships with the following default configuration regarding phones:

- DHCP Server
  - Enabled
  - Option 157 defines the boot server as ftp://10.10.20.1/ADTRAN, FTP Username and Password, and VLAN ID

- FTP Server
  - Enabled
  - Pointing to CFLASH filesystem
  - Default FTP Username and Password defined

- ADTRAN IP 7xx Phones
  - The IP 7xx phones depend on DHCP Option 157 to program their boot parameters during the DHCP process

ADTRAN IP Phone - DCHP Option 157

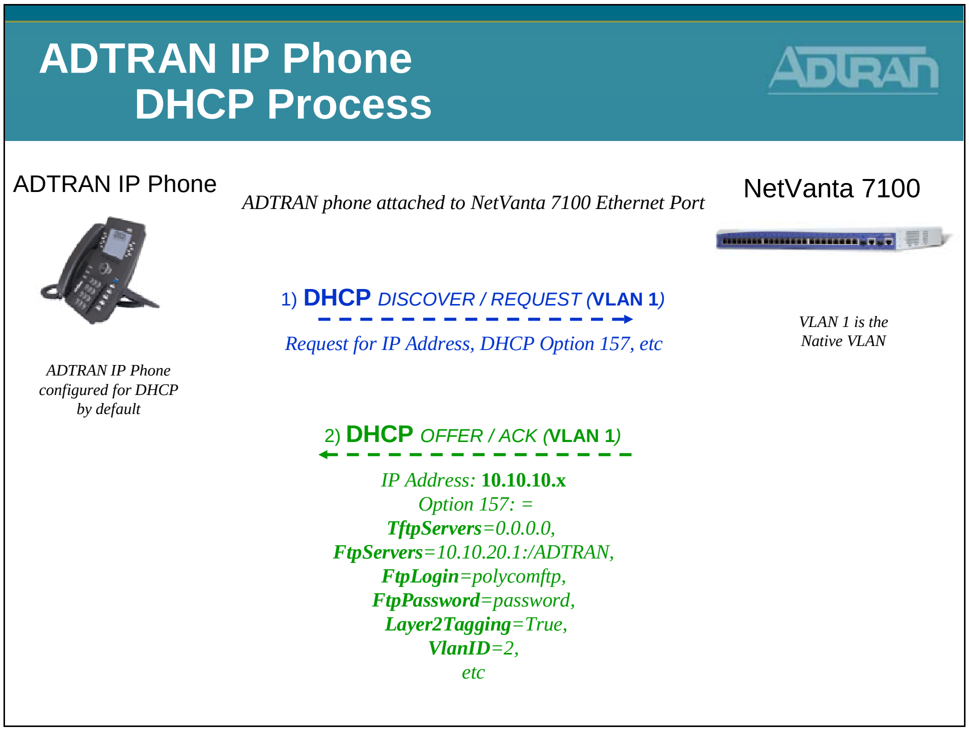


Below is the default configuration for the two DHCP Server Pools

ip dhcp-server pool "LAN_pool"
network 10.10.10.0 255.255.255.0
dns-server 10.10.10.1
default-router 10.10.10.1
tftp-server tftp://10.10.10.1
ntp-server 10.10.10.1
timezone-offset -6:00
option 157 ascii TftpServers=0.0.0.0,FtpServers=10.10.20.1:/ADTRAN,
FtpLogin=polycomftp,FtpPassword=password,Layer2Tagging=True,VlanID=2

ip dhcp-server pool "VoIP_pool"
network 10.10.20.0 255.255.255.0
dns-server 10.10.20.1
default-router 10.10.20.1
tftp-server tftp://10.10.20.1
ntp-server 10.10.20.1
timezone-offset -6:00
option 157 ascii TftpServers=0.0.0.0,FtpServers=10.10.20.1:/ADTRAN,
FtpLogin=polycomftp,FtpPassword=password,Layer2Tagging=True,VlanID=2

ADTRAN IP Phone - DHCP Process



DHCP Request Process

A default IP 700 series phone is programmed to request DHCP parameters at boot. The first time the phone boots, the request comes in on the Native VLAN. (VLAN 1 by default) Besides for boot server information, the phone is assigned a Voice VLAN. (VLAN 2 by default) At that point, the phone releases the IP address from the Native VLAN and then does a new DHCP request on VLAN 2.

DHCP Debug Output (debug ip dhcp-server)

2009.07.01 18:49:59 DHCP.SERVER Processing **Discover** Message (Xid = e1ea0b59) on **10.10.10.0**/255.255.255.0 from 00:A0:C8:25:55:50
2009.07.01 18:49:59 DHCP.SERVER Offering IP Address **10.10.10.5** to 00:A0:C8:25:55:50
2009.07.01 18:50:04 DHCP.SERVER Processing Request Message (Xid = e1ea0b59) on 10.10.10.0/255.255.255.0 from 00:A0:C8:25:55:50
2009.07.01 18:50:04 DHCP.SERVER Server sent an Ack to the client

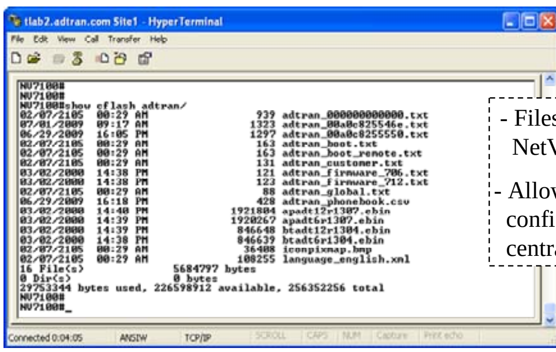
2009.07.01 18:50:04 DHCP.SERVER Processing **Release** Message (Xid = e1ea0b50) on 10.10.10.0/255.255.255.0 from 00:A0:C8:25:55:50
2009.07.01 18:50:04 DHCP.SERVER No Reply required

2009.07.01 18:50:31 DHCP.SERVER Processing **Discover** Message (Xid = e1ea26cb) on **10.10.20.0**/255.255.255.0 from 00:A0:C8:25:55:50
2009.07.01 18:50:31 DHCP.SERVER Offering IP Address **10.10.20.2** to 00:A0:C8:25:55:50
2009.07.01 18:50:36 DHCP.SERVER Processing Request Message (Xid = e1ea26cb) on 10.10.20.0/255.255.255.0 from 00:A0:C8:25:55:50
2009.07.01 18:50:36 DHCP.SERVER Server sent an Ack to the client

NetVanta 7000 - Boot Server

NetVanta 7000 - Boot Server

- Numerous files are created by the NetVanta 7000 and the ADTRAN/Polycom phones
- The NetVanta 7000 functions as the boot server for IP phones

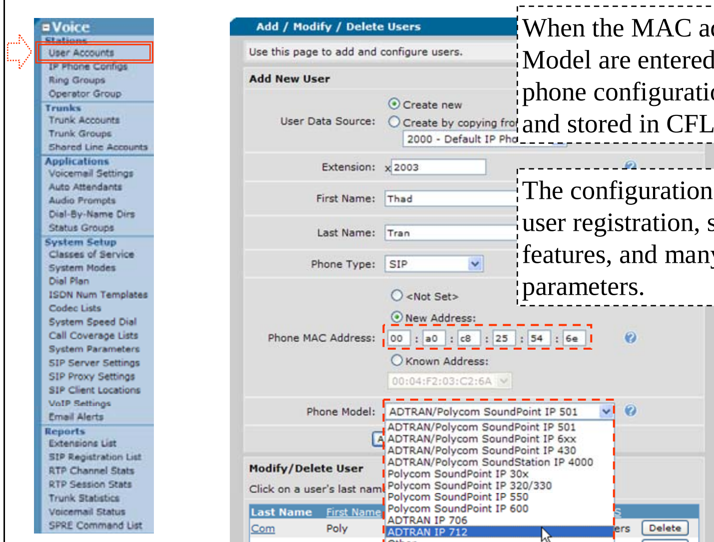


- Files are stored in CFLASH of the NetVanta 7000

- Allows global and per-phone configuration to be managed centrally

Creation of Phone Config Files

Creation of Phone Config Files



When the MAC address and Phone Model are entered for a new user, phone configuration files are created and stored in CFLASH

The configuration files define SIP user registration, server, phone features, and many other phone parameters.

ADTRAN IP Phone Configuration Files

ADTRAN IP Phone Configuration Files

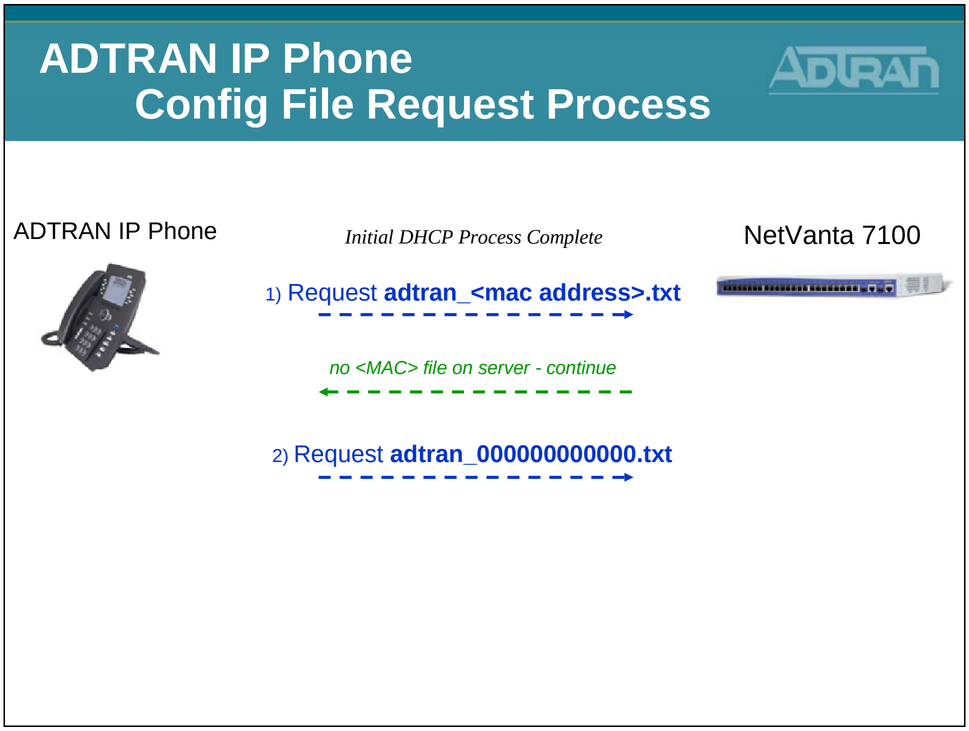
- A unique configuration file is required for each phone. The MAC address of the phone is used to identify the appropriate file for downloading.
- When the phone boots up, it checks the FTP/TFTP server for its specific configuration file.
- The file must be stored on the FTP/TFTP server in the following format:
  - **adtran_<mac address>.txt**
    - Lowercase letters only
- If the phone cannot find its MAC address-based configuration file, it will download the file **adtran_000000000000.txt** and use it as the main configuration file

NOTE: Most configuration file changes can be done from the GUI.

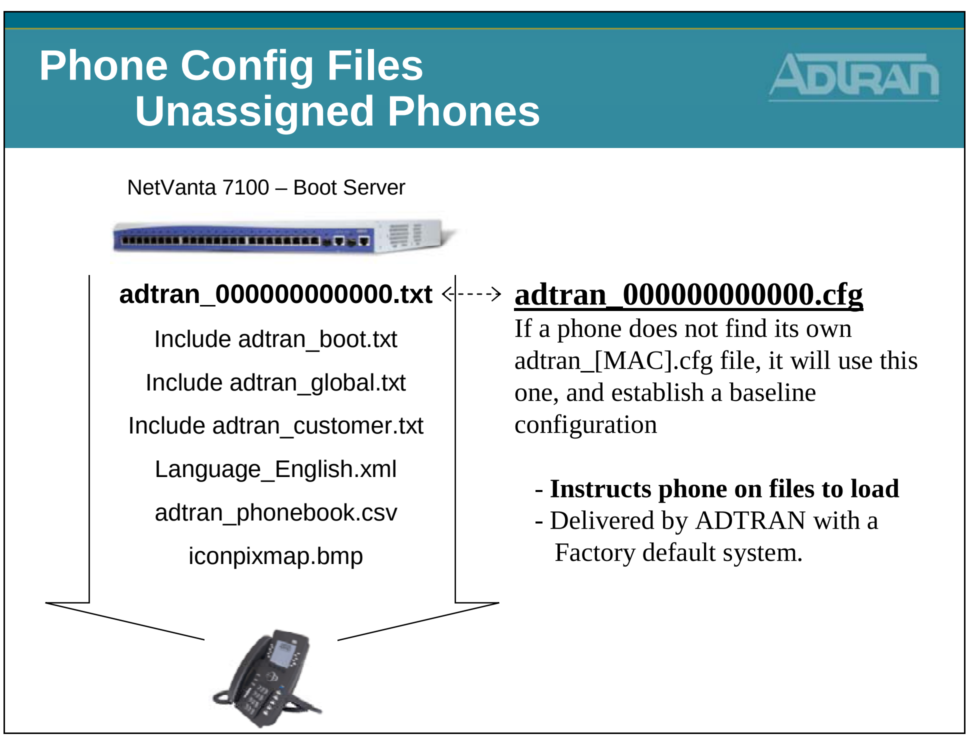
Configuration Files Rules

- Each parameter must appear on its own line
- A <name> <value> pair is entered for each parameter
- The <name> <value> may be separated by an arbitrary number of spaces or tabs
- Any combination of uppercase or lowercase letters can be used within the configuration file because it is not case sensitive
- Spaces are not permitted in any of the configuration values unless quote marks are used
- Comments may be included in a configuration file by starting the comment line with the # character

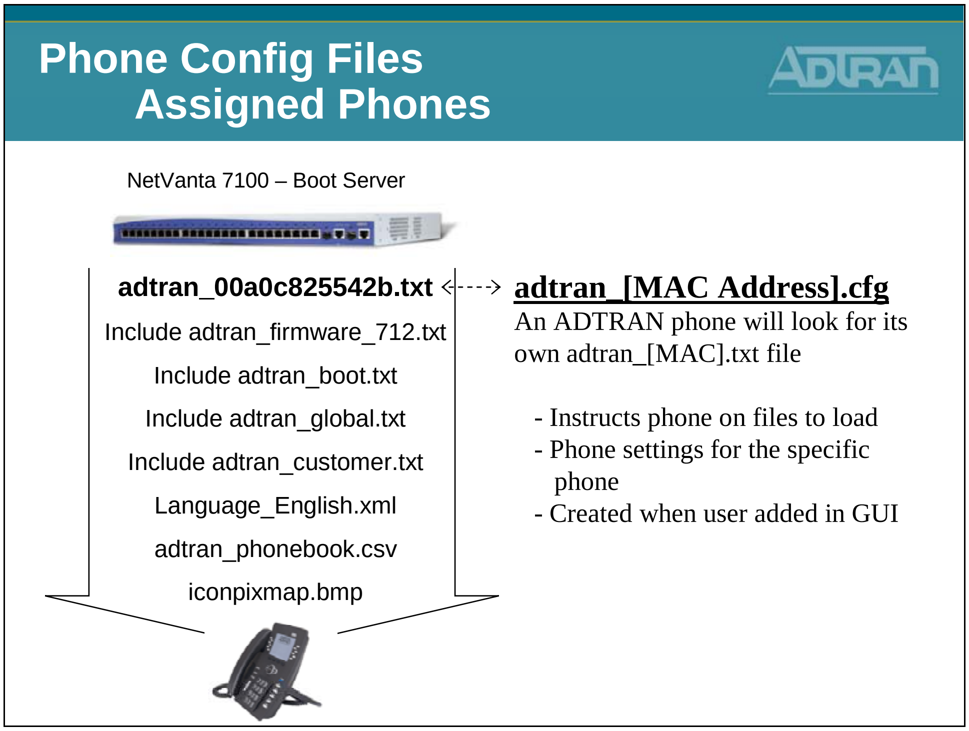
ADTRAN IP Phone - Config File Request Process



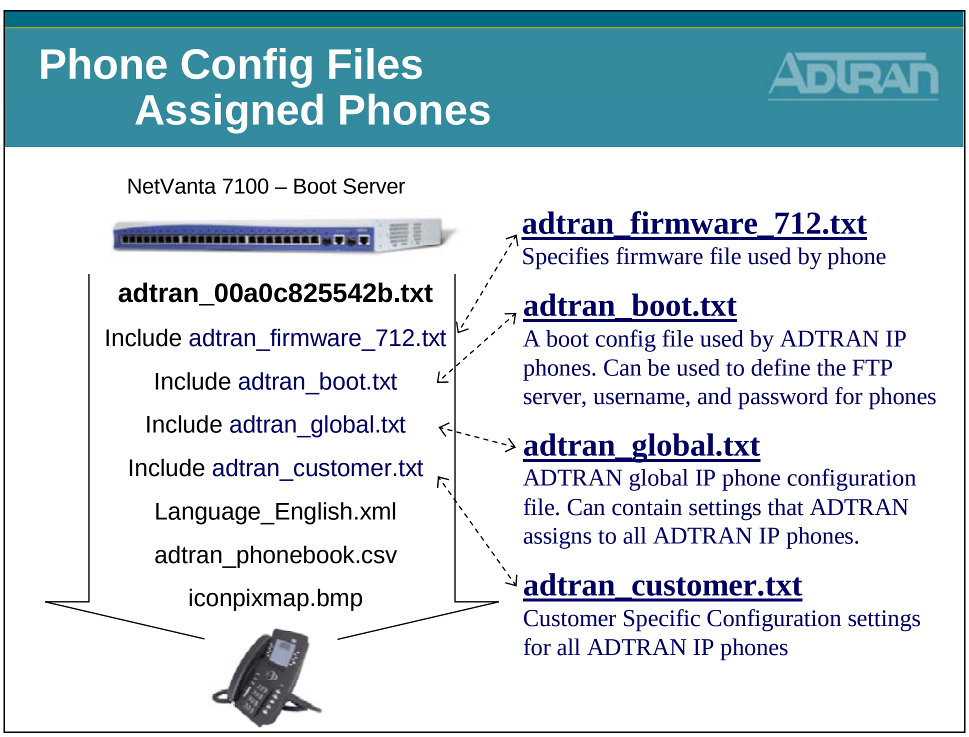
Phone Config Files - Unassigned Phones



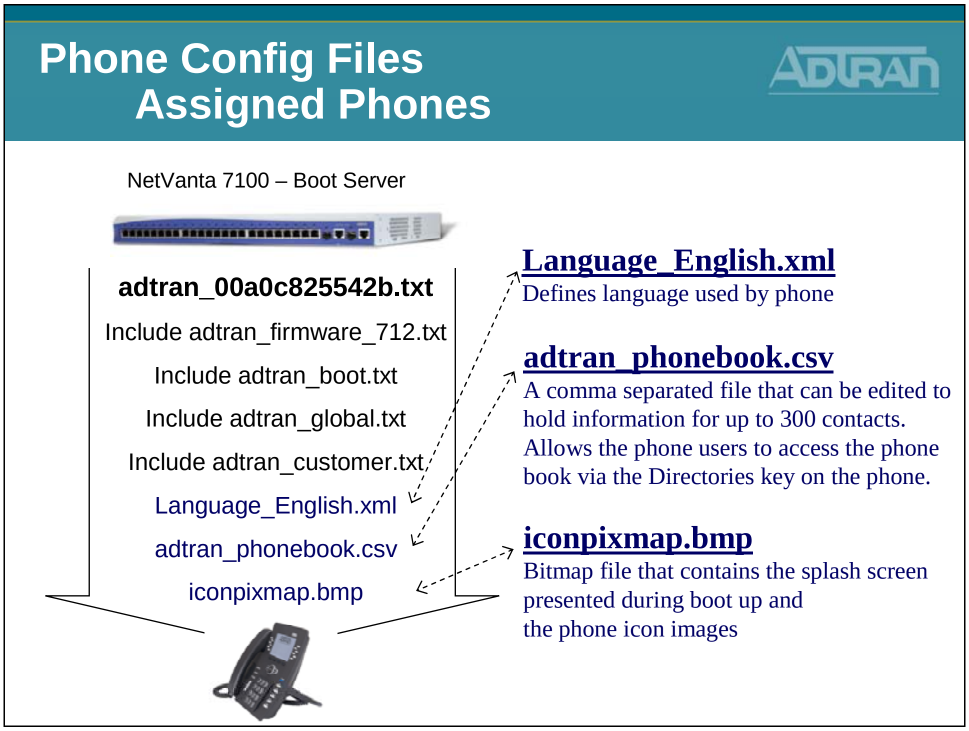
Phone Config Files - Assigned Phones



Phone Config Files - Assigned Phones



Phone Config Files - Assigned Phones



IP 700 Series Phone Boot Process

1. Phone boots and requests DHCP parameters
2. NetVanta 7000 Series Product responds with these parameters:
    a. IP address, subnet mask, and gateway in VLAN 1 (10.10.10.0)
    b. DHCP Option 157 defines the boot server as ftp://10.10.20.1/ADTRAN, FTP Username and Password, and VLAN ID of 2.
3. The phone then reboots and requests DHCP parameters in VLAN 2 (10.10.20.0)
4. The phone attempts to download the following files via FTP:
    a. adtran_mac.txt (where "mac" is the MAC address of the phone)
    b. adtran_firmware_7xx.txt (7xx will be specific to configured phone model)
    c. adtran_boot.txt
    d. adtran_global.txt
    e. adtran_customer.txt
    f. Language_English.xml
    g. adtran_phonebook.csv
    h. iconpixmap.bmp
5. Once the files are downloaded, the phone will attempt to register to the NetVanta 7000 Series Product based on he information in adtran_mac.txt.

IP 700 Series Phone Configuration Files

adtran_<MAC ADDRESS>.txt (MAC address of the phone)
An ADTRAN phone will look for its own adtran_[MAC].txt file based on its MAC address. This file contains SIP Registration information, phone settings for the specific phone, and pointers to other files to be loaded.

adtran_firmware_706.txt
Specifies firmware file used by the ADTRAN IP 706 phone

adtran_firmware_712.txt
Specifies firmware file used by the ADTRAN IP 706 phone

adtran_boot.txt
A boot config file used by local ADTRAN IP phones. Of all settings in file today, it uses the phone password to change the LCD password from the default “1234” to “456”.

adtran_boot_remote.txt
A boot config file used by remote ADTRAN IP phones.

adtran_global.txt
ADTRAN global IP phone configuration file. Contains settings that ADTRAN assigns to all ADTRAN IP phones.

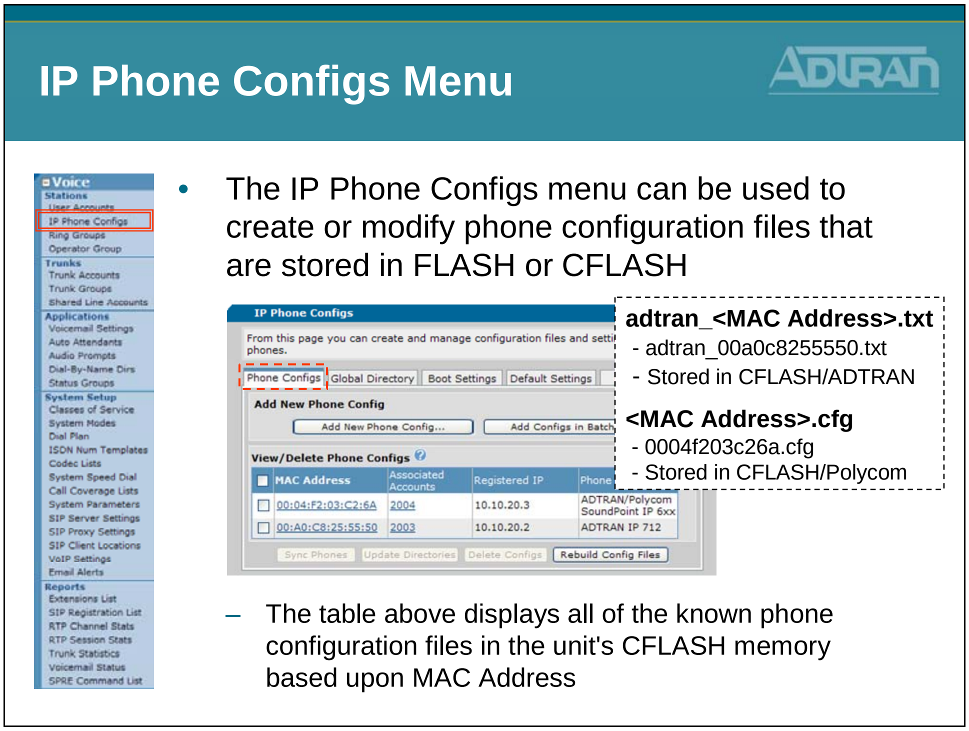
adtran_customer.txt
This file is where customizations for all IP 700 phones on the system would be configured.

Language_English.xml
Defines the phone language file used by phone

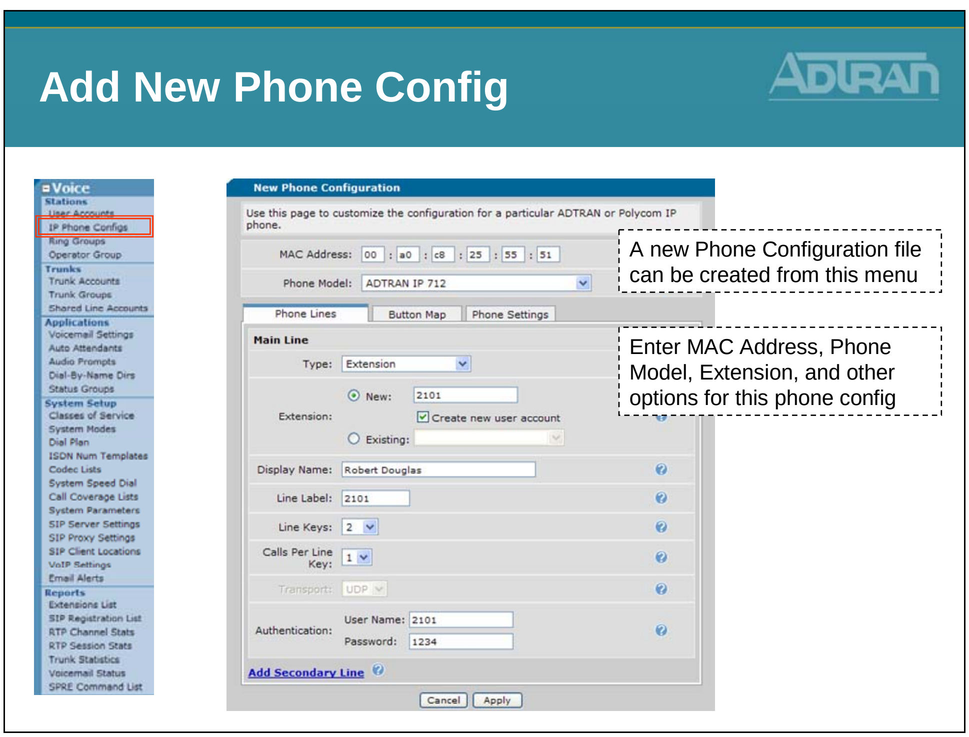
adtran_phonebook.csv
This is the System Directory for the IP 700 phones stored in Comma-Separated Value (CSV) format. Can be edited to hold information for up to 300 contacts. Allows the phone users to access the phone book via the Directories key on the phone.

iconpixmap.bmp
Bitmap file that contains the splash screen presented during boot up and the phone icon images.

IP Phone Configs Menu

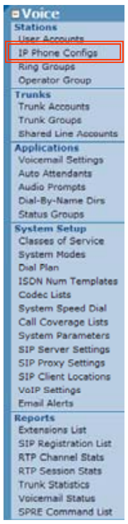


Add New Phone Config

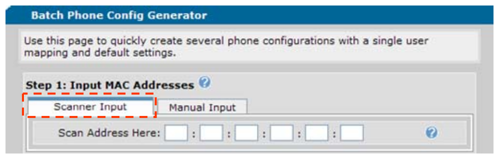


Batch Phone Config Generator - Scanner

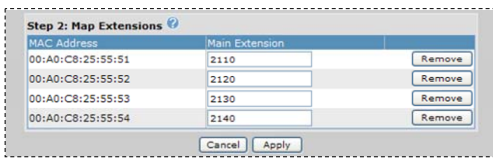
Batch Phone Config Generator



- Handheld Scanner Input
  - Make sure the input carat is on the first textbox, then use a handheld scanner to scan the address

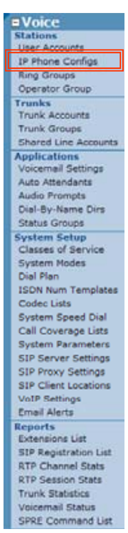


  - enter the main extension that should be associated with each MAC address

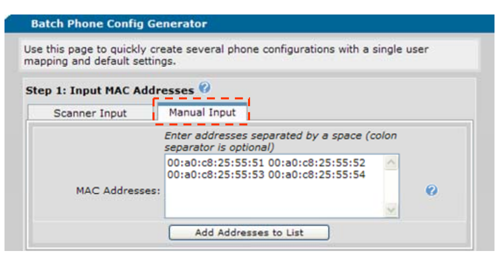


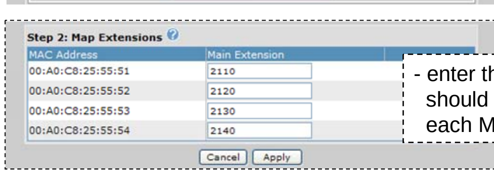
Add New Phone Config – Manual Input

Batch Phone Config Generator



- Manual Input
  - Enter multiple MAC Addresses by hand in the text box, then click the Add Addresses to List button



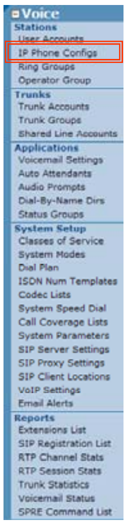


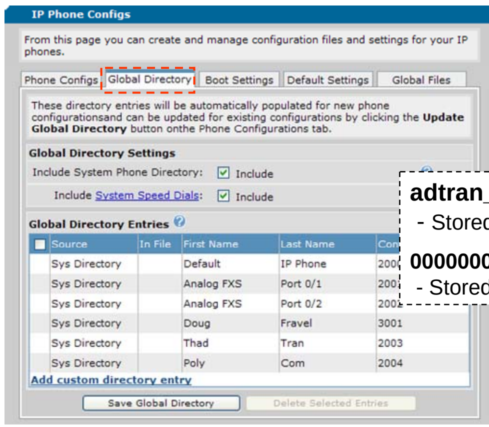
- enter the main extension that should be associated with each MAC address

Global Directory

Global Directory

- Creation and modification of the Global directory is done from this tab





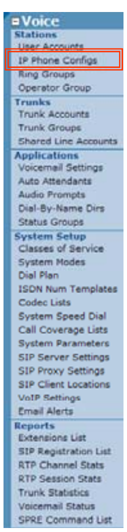
adtran_phonebook.csv
- Stored in CFLASH/ADTRAN

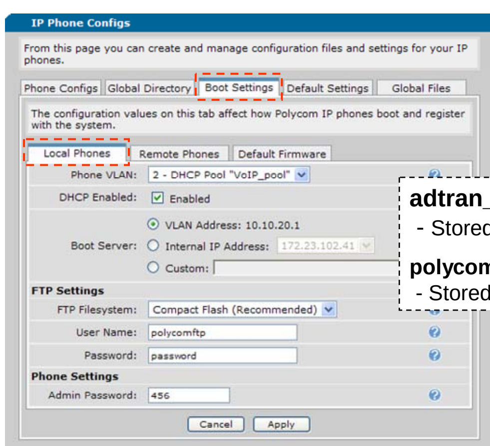
000000000000-directory.xml
- Stored in CFLASH/Polycom

Boot Settings – Local Phones

Boot Settings
Local Phones

- Boot parameters for Polycom phones
  - Voice VLAN / Boot Server FTP Username / Password



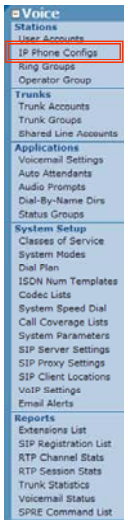


adtran_boot.txt
- Stored in CFLASH/ADTRAN

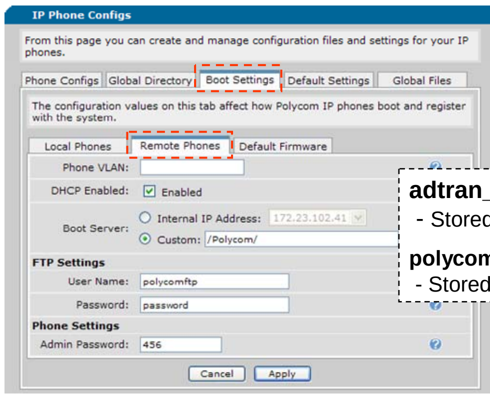
polycomboot.cfg
- Stored in CFLASH/Polycom

Boot Settings – Remote Phones

Boot Settings
Remote Phones



- Boot parameters for Remote Polycom phones
  - Voice VLAN / Boot Server FTP Username / Password

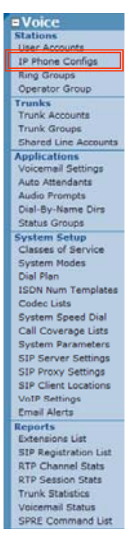


adtran_boot_remote.txt
- Stored in CFLASH/ADTRAN

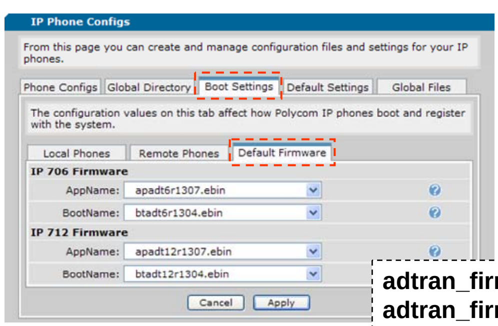
polycomboot_remote.cfg
- Stored in CFLASH/Polycom

Boot Settings – Default Firmware

Boot Settings
Default Firmware



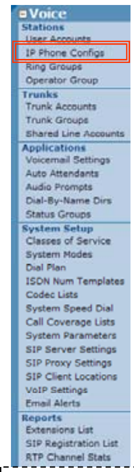
- Specify current Application and Bootcode to be used by ADTRAN IP 700 phones



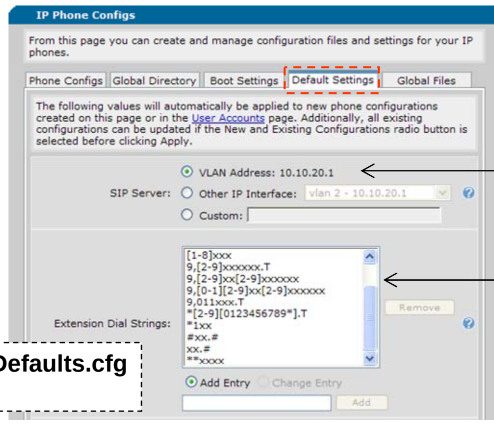
adtran_firmware_706.txt
adtran_firmware_712.txt
- Stored in CFLASH/ADTRAN

Default Settings

Default Settings



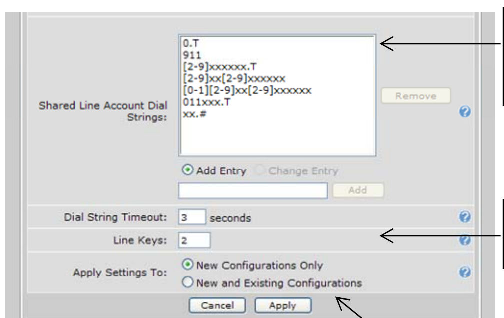
- The values here will automatically be applied to new phone configurations created on this page or in the User Accounts page



polycomConfigDefaults.cfg
- Stored in FLASH

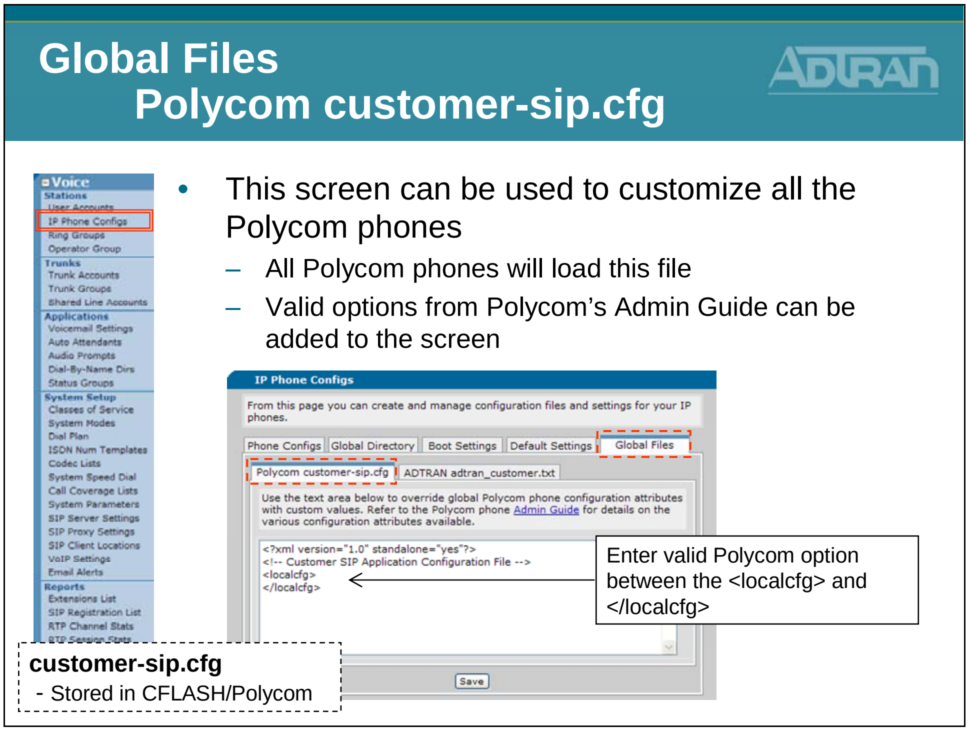
Default Settings

Default Settings (Continued…)



polycomConfigDefaults.cfg
- Stored in FLASH

Global Files – Polycom customer-sip.cfg



Polycom Customization Examples:

To Disable the Call Waiting Beep

The Call Waiting beep is enabled by default on the Polycom phones. To disable it, the following could be entered on the Global Files screen

```
<CALLWAITING se.pat.callProg.6.inst.1.type="silence" se.pat.callProg.6.inst.1.value="10"/>
<CALLWAITINGLONG se.pat.callProg.7.name="long call waiting"
se.pat.callProg.7.inst.1.type="silence" se.pat.callProg.7.inst.1.value="10"
se.pat.callProg.7.inst.2.type="silence" se.pat.callProg.7.inst.2.value="150"
se.pat.callProg.7.inst.3.type="silence" se.pat.callProg.7.inst.3.value="10"/>
```

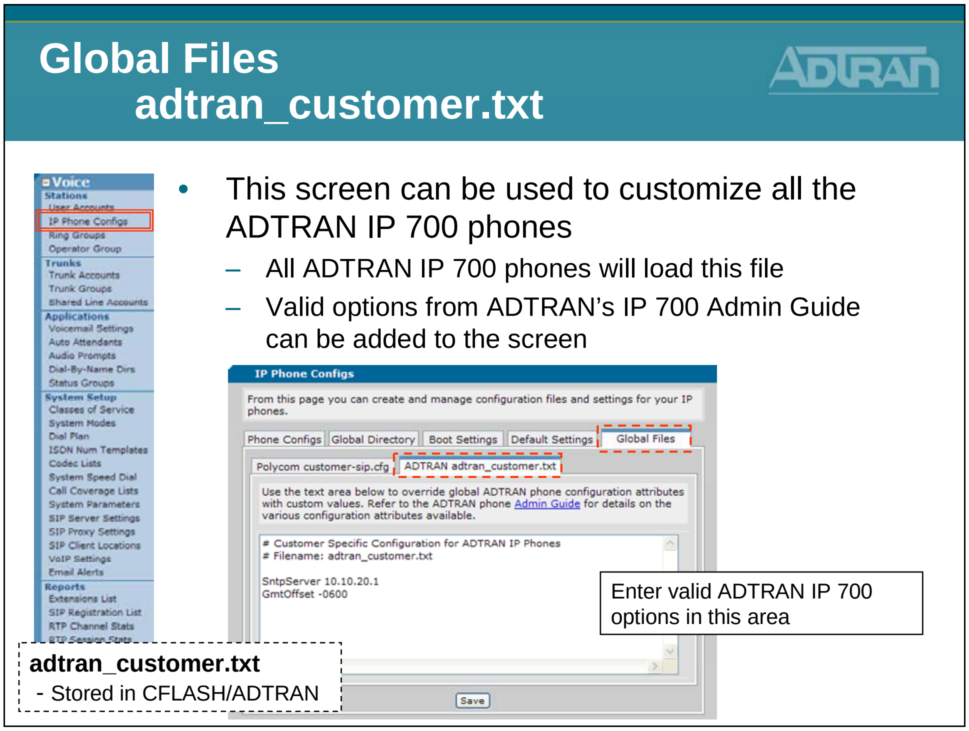
Hold reminder on Polycom Phones

By default Polycom phones do not beep every so often to let you know that you have a call on hold. Add the following line on the Global Files screen to enable this feature:

```
<localReminder call.hold.localReminder.enabled="1"/>
```

The above examples and other can be found in the Knowledge Base at kb.adtran.com.

Global Files – adtran_customer.txt



ADTRAN Customization Examples:

To Disable the Call Waiting Beep

By default ADTRAN IP 700 Series beep when there is a call waiting. To disable it, the following could be entered on the Global Files screen. *Verify that it is entered exactly as shown below.*

ToneDefine 1,0,0,0,0,0,0,0,0,1,0,0x0000,10,1,0,10,0,0
ToneMap Wait,1

Hold reminder on ADTRAN IP 700 Phones

By default ADTRAN IP 700 Series phones do not beep every so often to let you know that you have a call on hold.. Add the following line on the Global Files screen to enable this feature:

HoldReminder XX

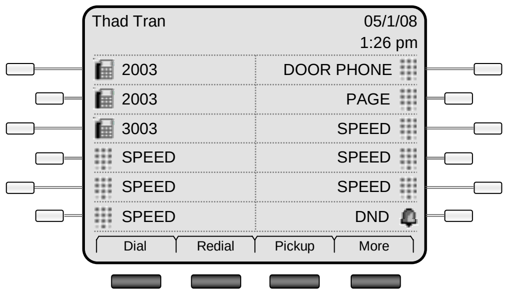
where XX is the frequency in seconds that you would like the phone to play the hold reminder

The above examples and other can be found in the Knowledge Base at kb.adtran.com.

ADTRAN IP - Registrations/Line Keys

ADTRAN IP Phone Registrations/Line Keys

- The ADTRAN IP phones support multiple registrations and programmable line keys



Registration: a SIP alias or phone number

Line key: a button on the phone beside the display

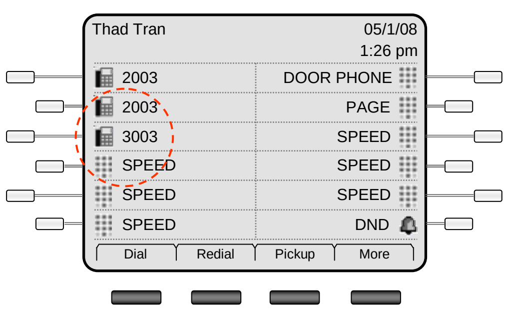
Call: a SIP session

- Registrations can appear on multiple line keys
- Speed dial can be configured by admin or user

ADTRAN IP Phones - Adding Line Registrations

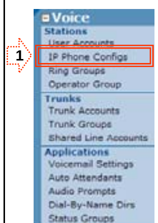
ADTRAN IP Phone Configuration Adding Line Registrations

1. Modify Phone Configuration file in GUI
2. Add secondary line key
3. Reboot/sync phone to load new configuration

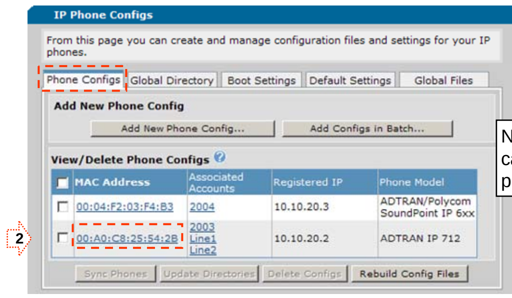


ADTRAN IP Phones - Adding Line Registrations

ADTRAN IP Phone Configuration Adding Line Registrations



1. Select the Voice / Stations / IP Phone Configs menu



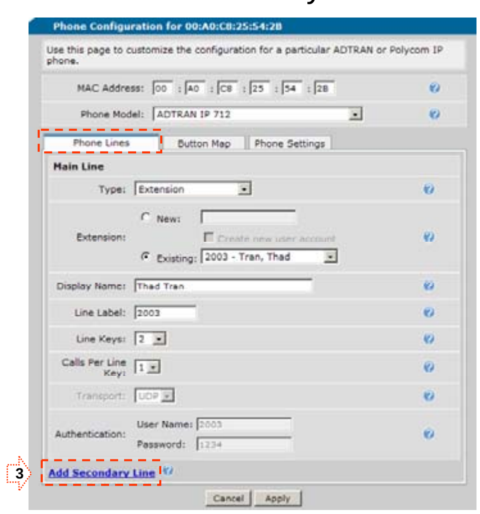
New line keys (registrations) can be added by editing the phone config file

2. Select the MAC address of the phone that you wish to add a line key (registration) to

ADTRAN IP Phones - Adding Line Registrations

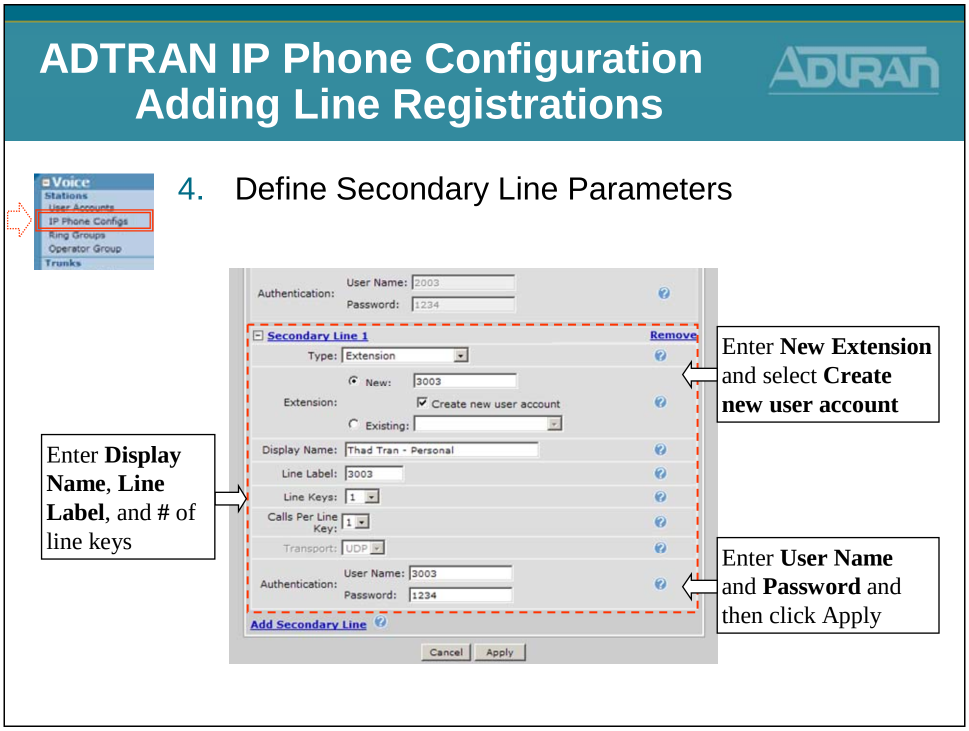
ADTRAN IP Phone Configuration Adding Line Registrations

3. Click Add Secondary Line

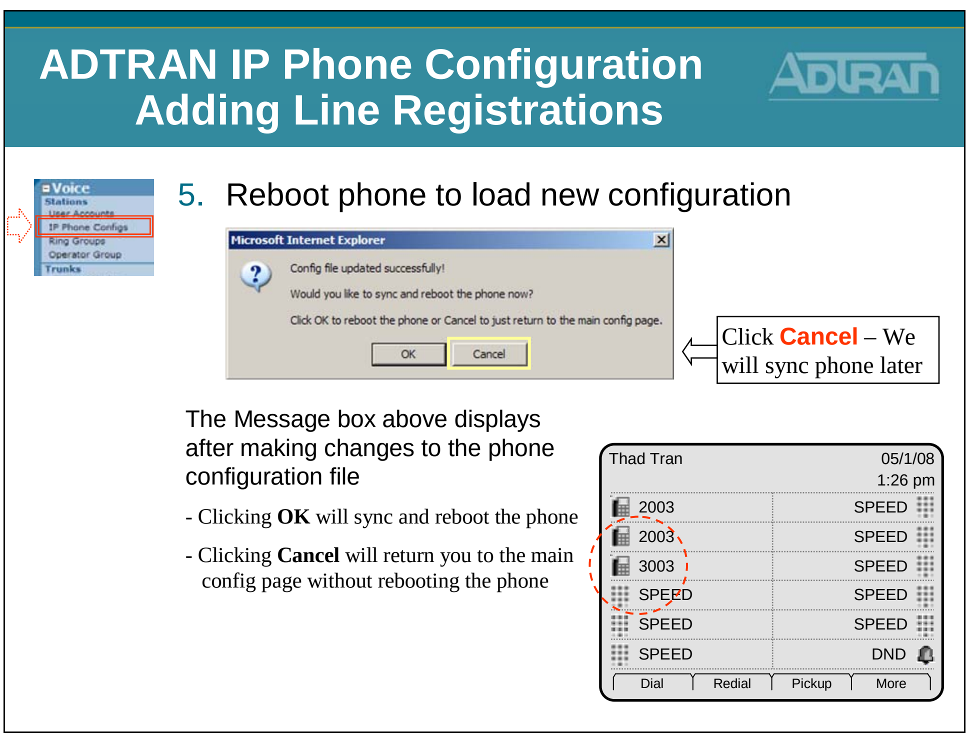


The Line Label and number of Line Keys for the first registration can be changed from this first menu

ADTRAN IP Phones - Adding Line Registrations



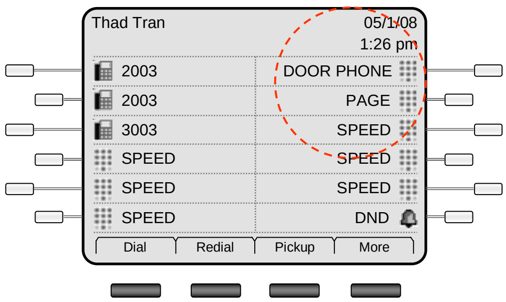
ADTRAN IP Phones - Adding Line Registrations



ADTRAN IP Phones - Map Line Key as Speed Dial

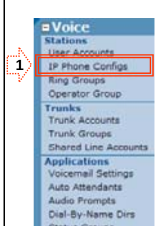
ADTRAN IP Phone Configuration
Map Line Key as Speed Dial

1. Modify phone configuration file in GUI
2. Map line key as speed dial
3. Reboot/sync phone to load new configuration

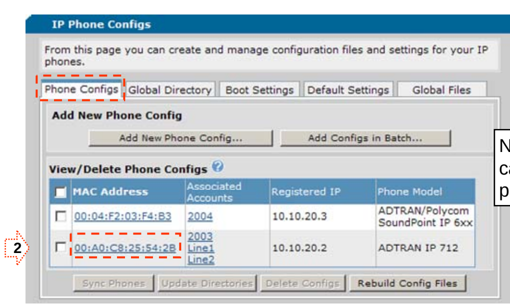


ADTRAN IP Phone - Map Line Key as Speed Dial

ADTRAN IP Phone Configuration
Map Line Key as Speed Dial



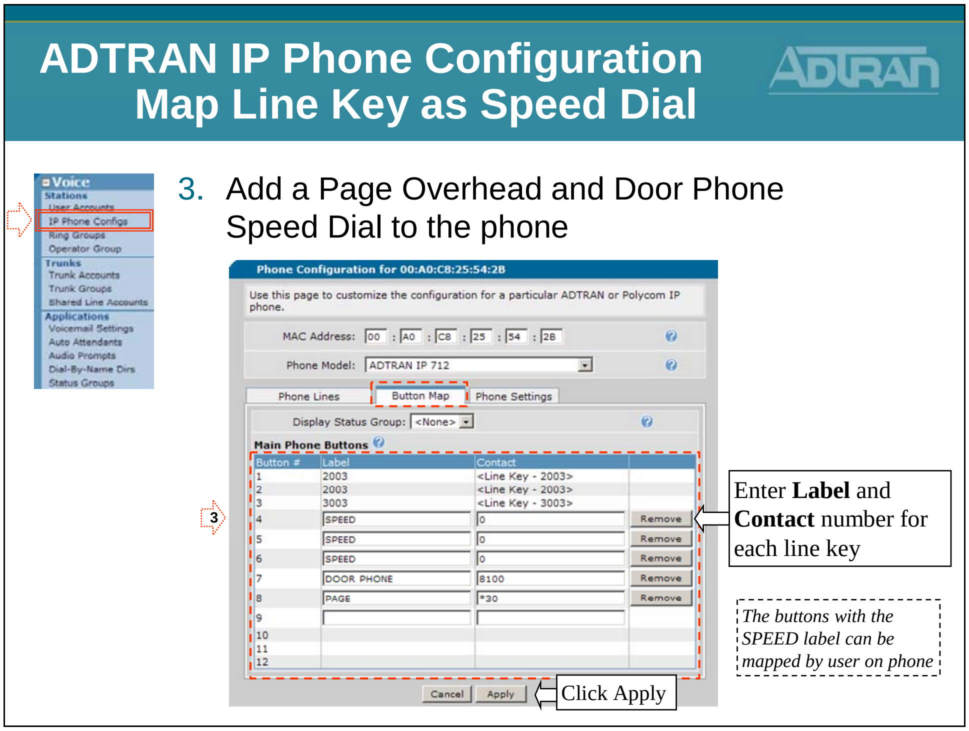
1. Select the Voice / Stations / IP Phone Configs menu



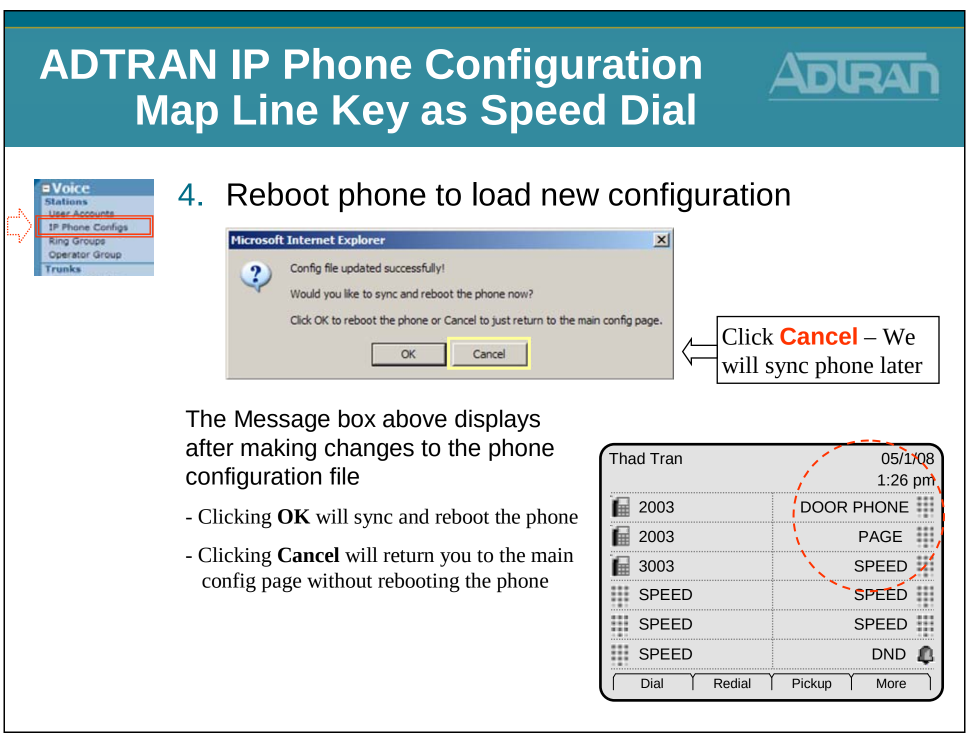
New line keys (registrations) can be added by editing the phone config file

2. Select the MAC address of the phone that you wish to add a line key (registration) to

ADTRAN IP Phones - Map Line Key as Speed Dial

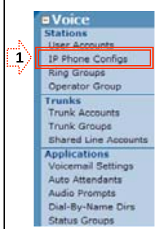


ADTRAN IP Phones - Map Line Key as Speed Dial

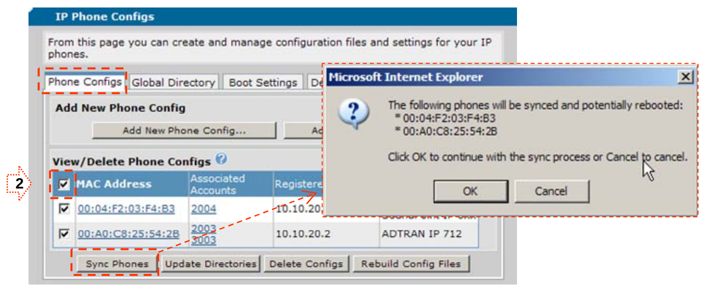


ADTRAN IP Phones - Syncing IP Phones

ADTRAN IP Phone Configuration
Syncing IP Phones



1. Select the Voice / Stations / IP Phone Configs menu



2. Select the MAC address for specific/all phones and then click Sync Phones
   – Attempts to sync/reboot selected phones. This action is only possible if the phone is either registered with the unit or if it is directly connected to the unit with inline power.

Troubleshooting IP Phones

ADTRAN

NetVanta IP Telephony Course

Troubleshooting IP Phones

ADTRAN IP Phones – Boot Process

ADTRAN IP Phone Boot Process

ADTRAN

ADTRAN IP Phone

1) DHCP Request Process

Phone obtains IP and learns the boot server IP address

debug ip dhcp-server

show ip dhcp-server binding

NetVanta 7100

2) File Request Process

Load phone config files, learn user identity and registrar SIP server

debug ip ftp-server

debug ip tftp server events *(Polycom)*

3) SIP Registration

Register location with SIP server

debug sip stack messages summary

show sip user-registration

debug ip dhcp-server

debug ip dhcp-server

- Display real-time messages associated with Dynamic Host Configuration Protocol (DHCP) server operation

```
NV7000# debug ip dhcp-server
… DHCP.SERVER Processing Discover Message (Xid = e1ea0b59) on
10.10.10.0/255.255.255.0 from 00:A0:C8:25:55:50
… DHCP.SERVER Offering IP Address 10.10.10.5 to 00:A0:C8:25:55:50
… DHCP.SERVER Processing Request Message (Xid = e1ea0b59) on
10.10.10.0/255.255.255.0 from 00:A0:C8:25:55:50
… DHCP.SERVER Server sent an Ack to the client

… DHCP.SERVER Processing Release Message (Xid = e1ea0b50) on
10.10.10.0/255.255.255.0 from 00:A0:C8:25:55:50

… DHCP.SERVER Processing Discover Message (Xid = e1ea26cb) on
10.10.20.0/255.255.255.0 from 00:A0:C8:25:55:50
… DHCP.SERVER Offering IP Address 10.10.20.2 to 00:A0:C8:25:55:50
… DHCP.SERVER Processing Request Message (Xid = e1ea26cb) on
10.10.20.0/255.255.255.0 from 00:A0:C8:25:55:50
… DHCP.SERVER Server sent an Ack to the client
```

show ip dhcp-server binding

show ip dhcp-server binding

- Display the Dynamic Host Client Protocol (DHCP) server client table with associated information

```
NV7000# show ip dhcp-server binding

IP Address     Client Id              Lease Expiration       Client Name

10.10.20.3     01:00:04:f2:03:c2:04   Jul 02 2009 4:08 PM    HunterPC
10.10.20.2     01:00:a0:c8:25:55:26   Jul 02 2009 6:50 PM    ShanePC
10.10.20.4     01:00:a0:c8:25:55:28   Jul 02 2009 6:50 PM    MathewPC
```

debug ip ftp-server

debug ip ftp-server

- Display real-time messages associated with File Transfer Protocol (FTP) server events in the NetVanta 7000

```
NV7000# debug ip ftp-server
FTP: NLST command - 'adtran_00a0c8255550.txt' transfer complete.
FTP: NLST command - 'adtran_firmware_712.txt' transfer complete.
FTP: NLST command - 'adtran_boot.txt' transfer complete.
FTP: NLST command - 'adtran_global.txt' transfer complete.
FTP: NLST command - 'adtran_customer.txt' transfer complete.
FTP: NLST command - 'Language_English.xml' transfer complete.
FTP: NLST command - 'adtran_phonebook.csv' transfer complete.
FTP: NLST command - 'iconpixmap.bmp' transfer complete.
```

debug sip stack messages summary

debug sip stack messages summary

- Activate SIP debug messages in real-time and displays only a summary (first line) of the available messages

```
NV7000# debug sip stack messages summary
18:51:24 SIP.STACK MSGSUM  Rx: REGISTER sip:10.10.20.1:5060 IP/2.0
18:51:24 SIP.STACK MSGSUM  Tx: SIP/2.0 401 Unauthorized
18:51:24 SIP.STACK MSGSUM  Rx: REGISTER sip:10.10.20.1:5060 IP/2.0
18:51:24 SIP.STACK MSGSUM  Tx: SIP/2.0 200 OK

18:51:24 SIP.STACK MSGSUM  Tx: NOTIFY sip:2003@10.10.20.2 SIP/2.0
18:51:24 SIP.STACK MSGSUM  Rx: SIP/2.0 200 OK
```

show sip user-registration

show sip user-registration

- Display local SIP server registration information

```
NV7000# show sip user-registration

EXTENSION  TYPE                          IP ADD        PORT  PROT  EXPIRES
---------  ----------------------------  ------------  ----  ----  -------

2003       Adtran-SIP-IP712/v1.3.7       10.10.20.2    5060  UDP   3559

2004       PolycomSoundPointIP601        10.10.20.3    5060  UDP   2838

Total phones registered: 2
```

ADTRAN IP 700 Phone Boot Process - SAMPLE DEBUG OUT

debug ip dhcp-server
debug ip ftp-server
debug sip stack messages summary

2009.07.01 18:49:59 DHCP.SERVER Processing Discover Message (Xid = e1ea0b59) on 10.10.10.0/255.255.255.0 from 00:A0:C8:25:55:50
2009.07.01 18:49:59 DHCP.SERVER Offering IP Address **10.10.10.5** to 00:A0:C8:25:55:50
2009.07.01 18:49:59 DHCP.SERVER Server sent an Offer to the client
2009.07.01 18:50:04 DHCP.SERVER Processing Request Message (Xid = e1ea0b59) on 10.10.10.0/255.255.255.0 from 00:A0:C8:25:55:50
2009.07.01 18:50:04 DHCP.SERVER Server sent an Ack to the client

2009.07.01 18:50:04 DHCP.SERVER Processing **Release** Message (Xid = e1ea0b50) on 10.10.10.0/255.255.255.0 from 00:A0:C8:25:55:50
2009.07.01 18:50:04 DHCP.SERVER No Reply required

2009.07.01 18:50:31 DHCP.SERVER Processing Discover Message (Xid = e1ea26cb) on 10.10.20.0/255.255.255.0 from 00:A0:C8:25:55:50
2009.07.01 18:50:31 DHCP.SERVER Offering IP Address **10.10.20.2** to 00:A0:C8:25:55:50
2009.07.01 18:50:31 DHCP.SERVER Server sent an Offer to the client
2009.07.01 18:50:36 DHCP.SERVER Processing Request Message (Xid = e1ea26cb) on 10.10.20.0/255.255.255.0 from 00:A0:C8:25:55:50
2009.07.01 18:50:36 DHCP.SERVER Server sent an Ack to the client

FTP: USER command - Password required for 'polycomftp'.
FTP: USER command - User 'polycomftp' logged in .
FTP: TYPE command - Type is set to I.
FTP: CWD command - directory changed to '/ADTRAN'.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for ls (10.10.20.2,1025).
FTP: NLST command - '**adtran_00a0c8255550.txt**' transfer complete.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for adtran_00a0c8255550.txt (10.10.20.2,1026).
FTP: RETR command - BINARY transfer complete.
2009.07.01 18:50:47 IP.FTP SERVER (RETR) Transfer of file '/ADTRAN/adtran_00a0c8255550.txt' complete for remote host '10.10.20.2'.

FTP: USER command - Password required for 'polycomftp'.
FTP: USER command - User 'polycomftp' logged in .
FTP: TYPE command - Type is set to I.
FTP: CWD command - directory changed to '/ADTRAN'.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for ls (10.10.20.2,1028).
FTP: NLST command - **'adtran_firmware_712.txt'** transfer complete.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for adtran_firmware_712.txt (10.10.20.2,1029).
FTP: RETR command - BINARY transfer complete.
2009.07.01 18:50:50 IP.FTP SERVER (RETR) Transfer of file '/ADTRAN/adtran_firmware_712.txt' complete for remote host '10.10.20.2'.FTP: USER command - Password required for 'polycomftp'.

FTP: USER command - User 'polycomftp' logged in .
FTP: TYPE command - Type is set to I.
FTP: CWD command - directory changed to '/ADTRAN'.

FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for ls (10.10.20.2,1031).
FTP: NLST command - **'adtran_boot.txt'** transfer complete.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for adtran_boot.txt (10.10.20.2,1032).
FTP: RETR command - BINARY transfer complete.
2009.07.01 18:50:51 IP.FTP SERVER (RETR) Transfer of file '/ADTRAN/adtran_boot.txt' complete for remote host '10.10.20.2'.NV7100#FTP: USER command - Password required for 'polycomftp'.

FTP: USER command - User 'polycomftp' logged in .
FTP: TYPE command - Type is set to I.
FTP: CWD command - directory changed to '/ADTRAN'.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for ls (10.10.20.2,1034).
FTP: NLST command - **'adtran_global.txt'** transfer complete.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for adtran_global.txt (10.10.20.2,1035).
FTP: RETR command - BINARY transfer complete.
2009.07.01 18:50:53 IP.FTP SERVER (RETR) Transfer of file '/ADTRAN/adtran_global.txt' complete for remote host '10.10.20.2'.FTP: USER command - Password required for 'polycomftp'.

FTP: USER command - User 'polycomftp' logged in .
FTP: TYPE command - Type is set to I.
FTP: CWD command - directory changed to '/ADTRAN'.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for ls (10.10.20.2,1037).
FTP: NLST command - **'adtran_customer.txt'** transfer complete.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for adtran_customer.txt (10.10.20.2,1038).
FTP: RETR command - BINARY transfer complete.
2009.07.01 18:50:54 IP.FTP SERVER (RETR) Transfer of file '/ADTRAN/adtran_customer.txt' complete for remote host '10.10.20.2'.

FTP: USER command - Password required for 'polycomftp'.
FTP: USER command - User 'polycomftp' logged in .
FTP: TYPE command - Type is set to I.
FTP: CWD command - directory changed to '/ADTRAN'.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for ls (10.10.20.2,1040).
FTP: NLST command - **'Language_English.xml'** transfer complete.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for Language_English.xml (10.10.20.2,1041).
FTP: RETR command - BINARY transfer complete.
2009.07.01 18:51:00 IP.FTP SERVER (RETR) Transfer of file '/ADTRAN/Language_English.xml' complete for remote host '10.10.20.2'.

FTP: USER command - Password required for 'polycomftp'.
FTP: USER command - User 'polycomftp' logged in .
FTP: TYPE command - Type is set to I.
FTP: CWD command - directory changed to '/ADTRAN'.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for ls (10.10.20.2,1043).
FTP: NLST command - **'adtran_phonebook.csv'** transfer complete.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for adtran_phonebook.csv (10.10.20.2,1044).
FTP: RETR command - BINARY transfer complete.

2009.07.01 18:51:05 IP.FTP SERVER (RETR) Transfer of file '/ADTRAN/adtran_phonebook.csv' complete for remote host '10.10.20.2'.

FTP: USER command - Password required for 'polycomftp'.
FTP: USER command - User 'polycomftp' logged in .
FTP: TYPE command - Type is set to I.
FTP: CWD command - directory changed to '/ADTRAN'.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for ls (10.10.20.2,1046).
FTP: NLST command - **'iconpixmap.bmp'** transfer complete.
FTP: PORT command - opening port from 10.10.20.2.
FTP: BINARY data connection for iconpixmap.bmp (10.10.20.2,1047).
FTP: RETR command - BINARY transfer complete.
2009.07.01 18:51:09 IP.FTP SERVER (RETR) Transfer of file '/ADTRAN/iconpixmap.bmp' complete for remote host '10.10.20.2'.

18:51:24 SIP.STACK MSGSUM Rx: REGISTER sip:10.10.20.1:5060 SIP/2.0
18:51:24 SIP.STACK MSGSUM Tx: SIP/2.0 401 Unauthorized
18:51:24 SIP.STACK MSGSUM Rx: REGISTER sip:10.10.20.1:5060 SIP/2.0
18:51:24 SIP.STACK MSGSUM Tx: SIP/2.0 200 OK

18:51:24 SIP.STACK MSGSUM Tx: NOTIFY sip:2003@10.10.20.2 SIP/2.0
18:51:24 SIP.STACK MSGSUM Rx: SIP/2.0 200 OK

NV7100# **show debug**
debug ip dhcp-server
debug ip ftp-server
debug sip stack messages summary

NV7100# **undebug all**

NV7100# **show ip dhcp-server binding**

IP Address	Client Id	Lease Expiration	Client Name
10.10.20.3	01:00:04:f2:03:c2:6a	Jul 02 2009 4:08 PM	
10.10.20.2	01:00:a0:c8:25:55:50	Jul 02 2009 6:50 PM	

NV7100# **show sip user-registraiontion**

EXTENSION	TYPE	IP ADDRESS	PORT	PROT	EXPIRES
----------	------------------------------	----------------	-----	----	-------
2003	Adtran-SIP-IP712/v1.3.7	10.10.20.2	5060	UDP	3559
2004	PolycomSoundPointIP-SPIP_601..	10.10.20.3	5060	UDP	2838

Total phones registered: 2

Module Objectives

Module Objectives

At the end of this module, you should:

- Have a basic understanding of ADTRAN phone configuration files
- Be able to modify phone configuration files in the GUI
- Troubleshoot the boot process of the ADTRAN IP 700 series phone

Polycom IP Phones - Provisioning Methods

Polycom IP Phones - Provisioning Methods

ADTRAN/Polycom IP Phone Provisioning Methods

- Local phone based configuration
  - Local phone user interface (Phone Settings)
  - Web interface (Phone Manager)
- Centrally Provisioned from Boot Server
  - Consist of Global and per-phone configuration files
- DHCP
  - Can set a limited number of parameters
    - including the location of configuration files

Polycom Default Passwords

Polycom Default Passwords

- Web interface
  - Username = Polycom
  - Password = 456
- Admin interface on the hard phone
  - Password = 456
  - User password - 123 (not used much)

Polycom Phones - Installation Process

Polycom Phones Installation Process

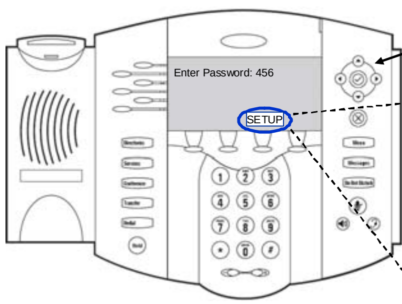
- Regardless of whether or not you will be installing a centrally provisioned system, there are two steps required to get your phones up and running
  - 1) Basic TCP/IP Network Setup
    - IP address and subnet mask
    - Local setup on phone
  - 2) Application Configuration
    - SIP application specific parameters
    - Setup done with Configuration files that are stored on boot server

Polycom Phones - TCP/IP Network Setup

Polycom Phones
TCP/IP Network Setup

- When phone first boots, press the **SETUP** softkey
- Enter password: *default = 456*

Navigate with arrow keys



DHCP Client:	Enabled
DHCP Menu:	**...**
Phone IP Addr:	000.000.000.000
Subnet Mask:	000.000.000.000
IP Gateway:	000.000.000.000
Server Menu:	**...**
SNTP Address:	000.000.000.000
GMT Offset:	0
DNS Server:	000.000.000.000
DNS Alt. Server:	000.000.000.000
DNS Domain:	
Ethernet Menu:	…
Syslog Menu:	…
EM Power:	Enabled

Exit Edit

"PLUG IN PHONES"

Polycom Phones - Default Setup Menu

Polycom Phones
Default Setup Menu

SETUP Menu

DHCP Client:	Enabled
DHCP Menu:	**...**
Phone IP Addr:	000.000.000.000
Subnet Mask:	000.000.000.000
IP Gateway:	000.000.000.000
Server Menu:	**...**
SNTP Address:	000.000.000.000
GMT Offset:	0
DNS Server:	000.000.000.000
DNS Alt. Server:	000.000.000.000
DNS Domain:	
Ethernet Menu:	…
EM Power:	Enabled

Exit Edit

Ethernet Menu

CDP:	Enabled
VLAN Id:	
LAN Port Mode:	Auto
PC Port Mode:	Auto

EDIT EXIT

DHCP Menu

Timeout:	3
Boot Server:	Option 66
BootSrv Opt:	150
BootSrv Type:	IP Address
VLAN Disc:	Disabled
VLAN Disc Opt:	129

EDIT EXIT

Server Menu

Server Type:	FTP
Server Address:	0.0.0.0
Server User:	PlcmSpIp
Server Password:	****
File Tx Tries:	3
Retry Wait:	1
Prov. String:	Default

EDIT EXIT

Default **FTP** Username and Password:
PlcmSpIp **PlcmSpIp**

Polycom Phones - DHCP Process

Polycom Phones DHCP Process

ADTRAN

Polycom Phone

Polycom Phone configured for DHCP by default

Polycom phone attached to NetVanta 7100 Ethernet Port

NetVanta 7100

VLAN 1 is the Native VLAN

1) **DHCP** *DISCOVER / REQUEST (**VLAN 1**)*

Request for IP Address, Boot Server(66), NTP Server(42) etc

2) **DHCP** *OFFER / ACK (**VLAN 1**)*

IP Address: **10.10.10.x**
Boot Server(66): tftp://10.10.10.1
NTP Server(42): **10.10.10.1**, *etc*

Boot Server

Boot Server

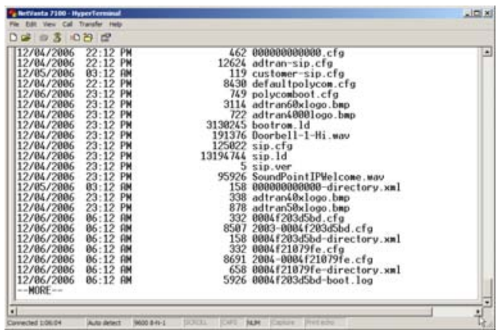
ADTRAN

- A boot server allows global and per-phone configuration to be managed centrally
- Configuration files are text XML-format files
  - Most configuration files are created automatically by the NetVanta 7100 when a new voice user is created
  - Phone configuration files could also be created manually and edited using an XML editor
  - Downloaded by the phones at boot time
- The boot server also facilitates automated application upgrades, diagnostics, and a measure of fault tolerance

Boot Server Files

Boot Server Files

- Numerous files are created by the NetVanta 7100 and the Polycom phones
- The NetVanta 7100 can function as the boot server for IP phones
  - Files are stored in Flash or CFLASH of the NetVanta 7100



Configuration Files

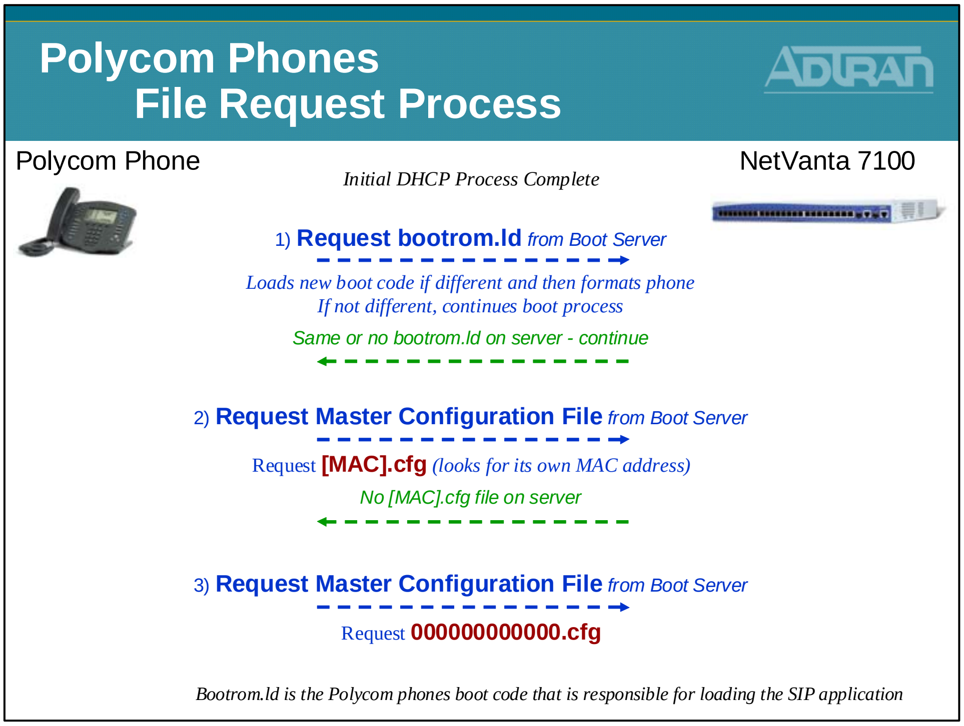
Configuration Files

- The phone configuration files consist of master configuration files and application configuration files.

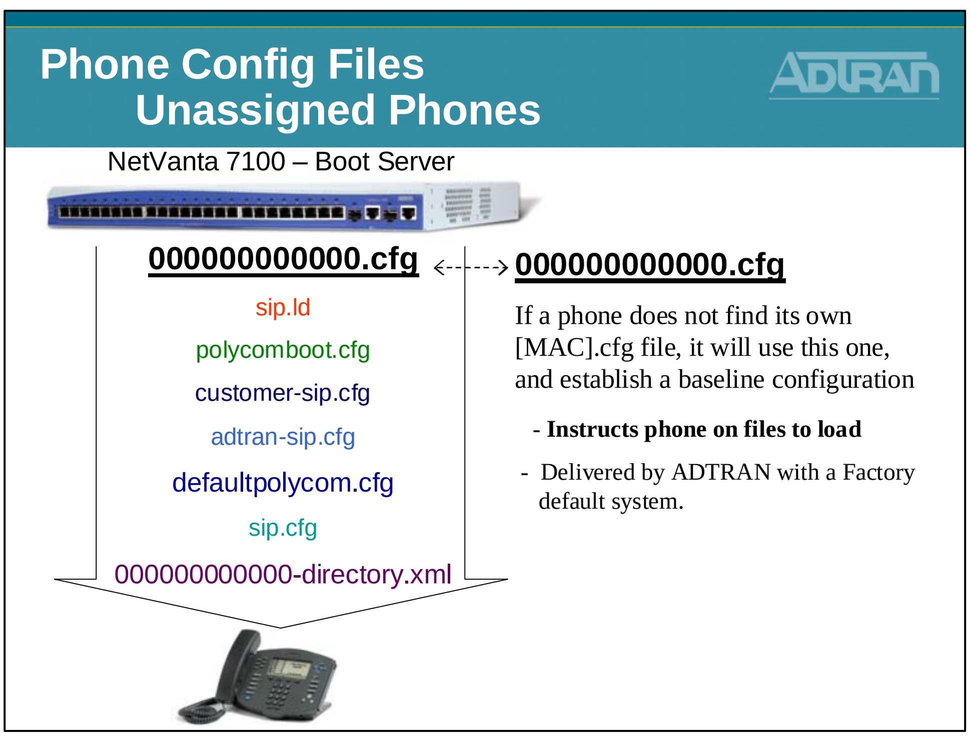
- Master Configuration Files
  - Instructs phone on files to load
  - 000000000000.cfg or [MAC].cfg

- Application Configuration Files
  - Application configuration files dictate the behavior of the phone once it is running the executable specified in the master configuration file
  - defaultpolycom.cfg, Ext-[MAC].cfg
    customer-sip.cfg, adtran-sip.cfg, sip.cfg

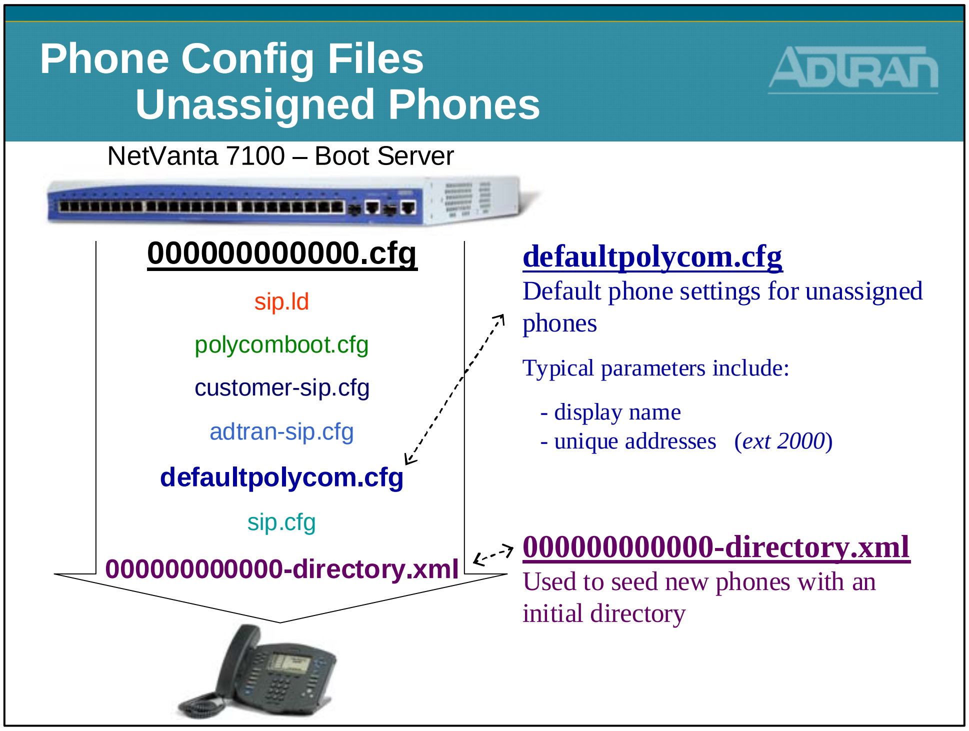
Polycom Phones - File Request Process



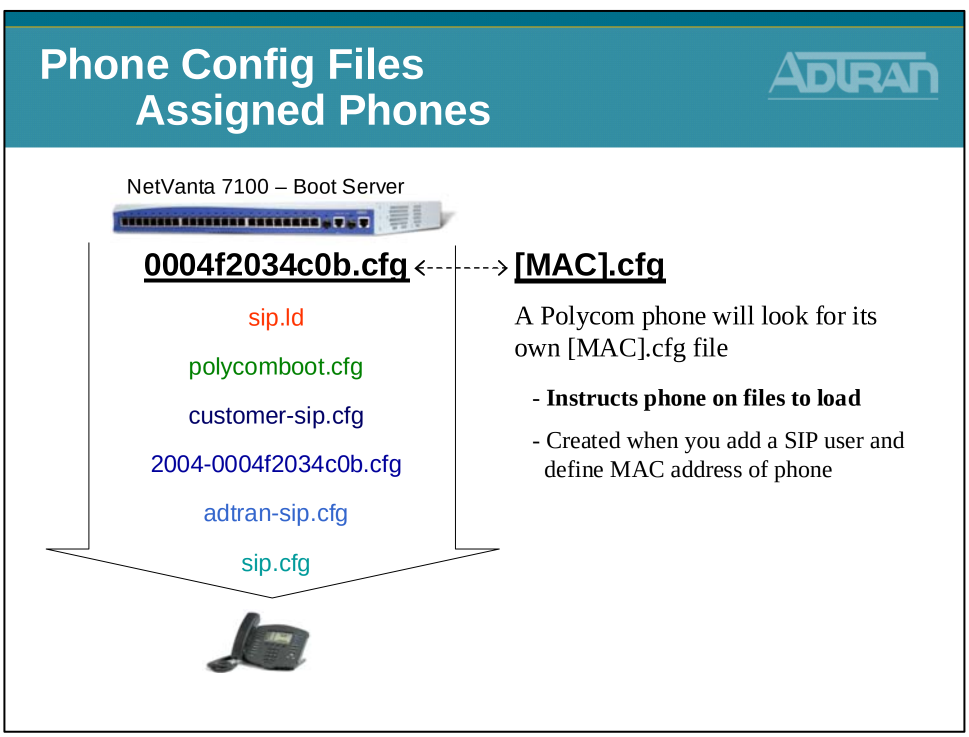
Phone Config Files - Unassigned Phones



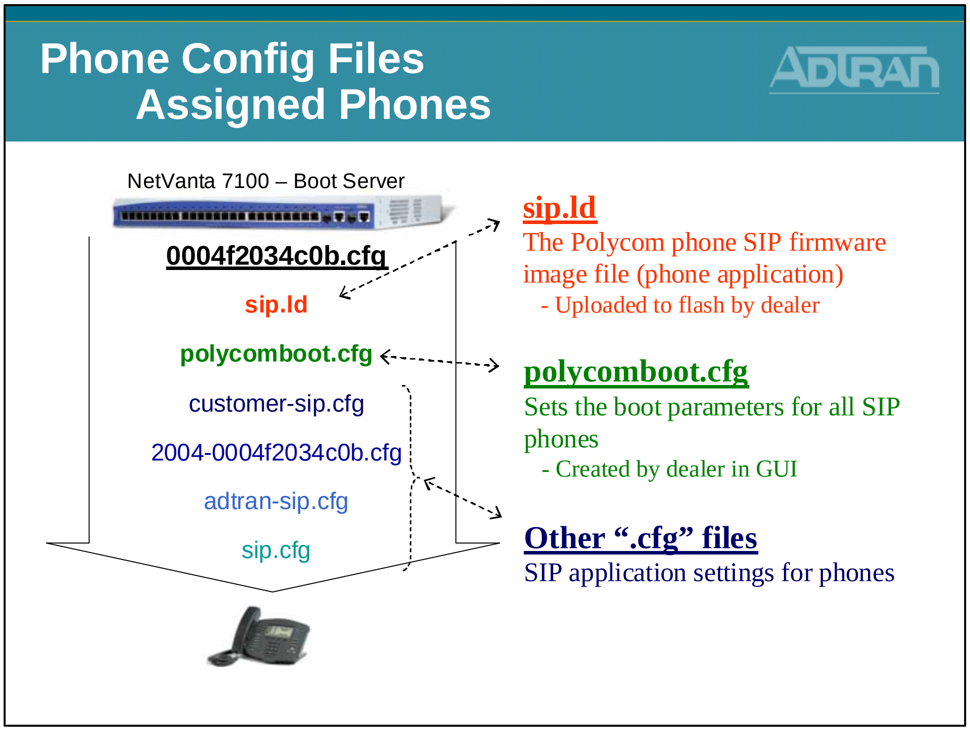
Phone Config Files - Unassigned Phones



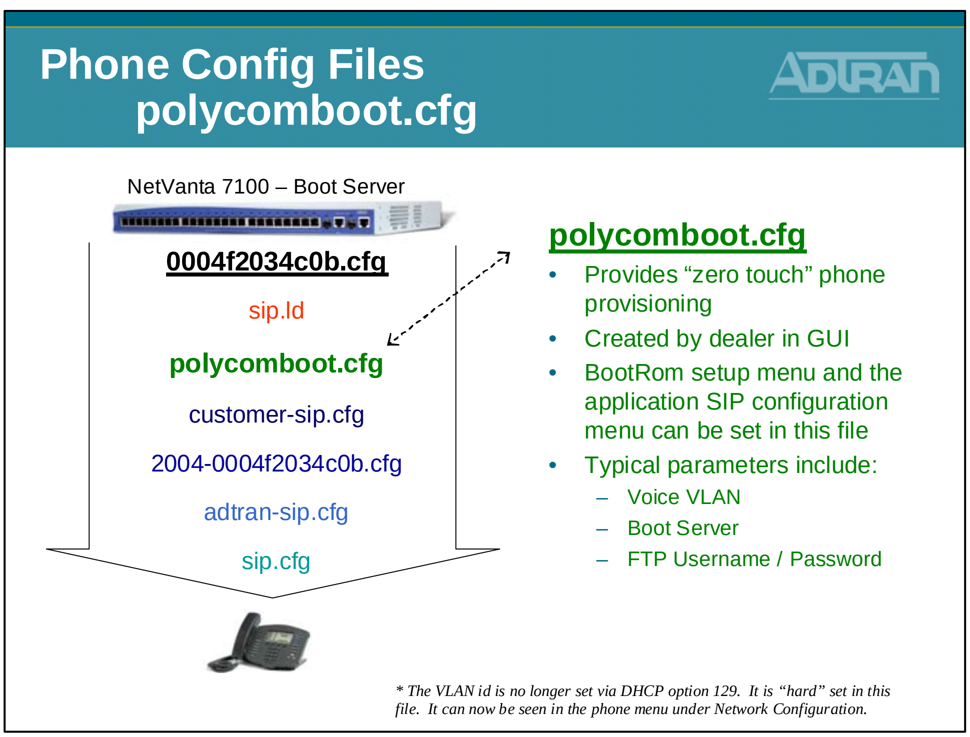
Phone Config Files - Assigned Phones



Phone Config Files - Assigned Phones



Phone Config Files - polycomboot.cfg

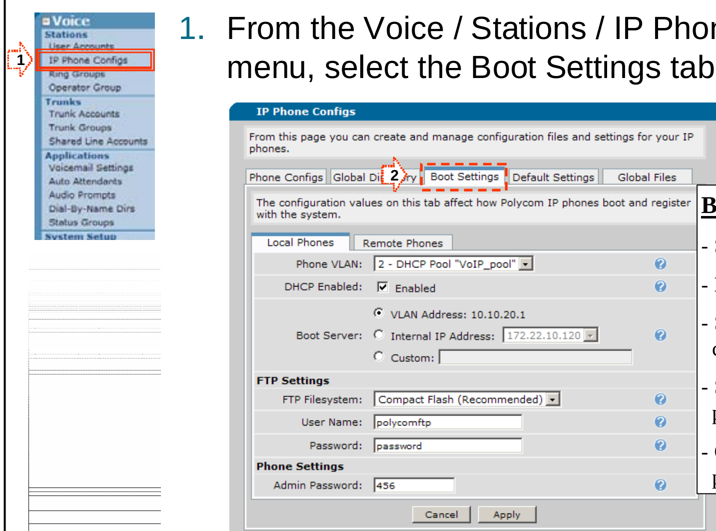


Phone Config Files - Editing polycomboot.cfg

Phone Config Files
Editing polycomboot.cfg

ADTRAN

1. From the Voice / Stations / IP Phone Configs menu, select the Boot Settings tab



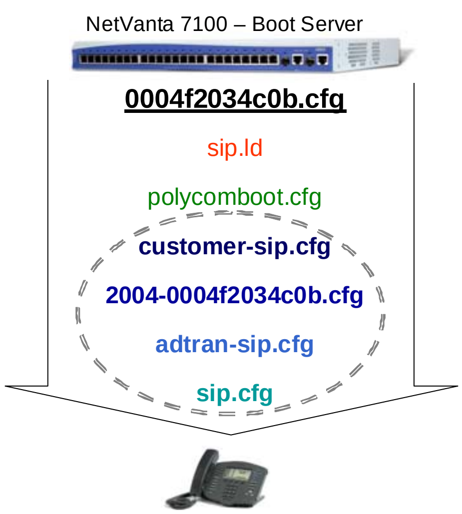
Boot Settings

- Set Phone VLAN
- Define Boot Server
- Specify where phone config files are stored
- Set FTP username and password
- Change phone Admin password

Phone Config - SIP App. Settings for Phones

Phone Config Files
SIP Application Settings for Phones

ADTRAN



customer-sip.cfg
Dealer/User can customize this file
- Examples: logos, wav files, etc.

2004-0004f2034c0b.cfg
Phone settings for the specific phone
- Created when user added in GUI

adtran-sip.cfg
Contains the default settings that ADTRAN chooses to employ on the Polycom phones
- Examples: ADTRAN logo, ringtones, etc

sip.cfg
Default settings from Polycom. This file should not be modified.

Phone Config - SIP App. Settings for Phones

Phone Config Files
SIP Application Settings for Phones

ADTRAN

customer-sip.cfg, adtran-sip.cfg, sip.cfg

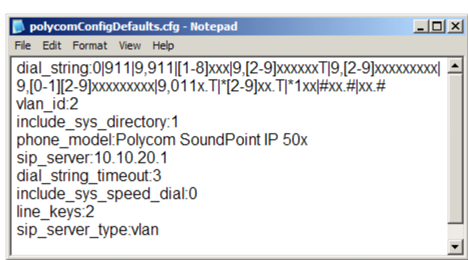
- Reasons for the 3 separate SIP config files
  - Polycom modifies the **sip.cfg** file occasionally
    - ADTRAN has needed this ability also
  - If customers have modified **sip.cfg** and made it unique, then it will be difficult for them to upgrade to new Polycom revisions that require an updated **sip.cfg** file.
- The 3 files allow…
  - Polycom to have their own SIP configurations that can be updated.
  - ADTRAN to have our own SIP configurations that can be updated.
  - The customer still has the ability to customize the SIP settings, and update ADTRAN and/or Polycom SIP configs.

polycomConfigDefaults.cfg

Default Polycom Configuration File
polycomConfigDefaults.cfg

ADTRAN

- A text file that contains default Polycom configuration values that will automatically be applied to new phone configurations created by the NetVanta 7100
  - This file is defined and created from the Default Settings tab of the NetVanta 7100 IP Phone Configs screen
- Included parameters:
  - Default Dial Plan Digit Map
  - Default Phone Model
  - Number of Line Keys

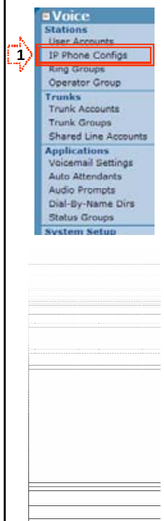

polycomConfigDefaults.cfg - Notepad

File Edit Format View Help

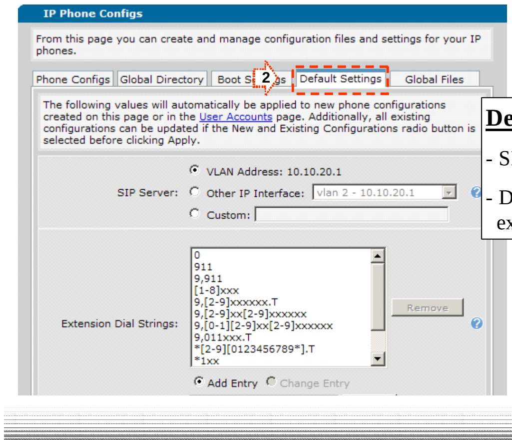
```
dial_string:0|911|9,911|[1-8]xxx|9,[2-9]xxxxxxT|9,[2-9]xxxxxxxxx|
9,[0-1][2-9]xxxxxxxxx|9,011x.T|*[2-9]xx.T|*1xx|#xx.#|xx.#
vlan_id:2
include_sys_directory:1
phone_model:Polycom SoundPoint IP 50x
sip_server:10.10.20.1
dial_string_timeout:3
include_sys_speed_dial:0
line_keys:2
sip_server_type:vlan
```

Phone Config Files - polycomConfigDefaults.cfg

Phone Config Files
polycomConfigDefaults.cfg



1. From the Voice / Stations / IP Phone Configs menu, select the Default Settings tab

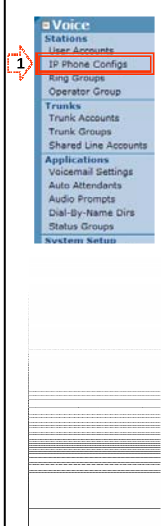


Default Phone Settings

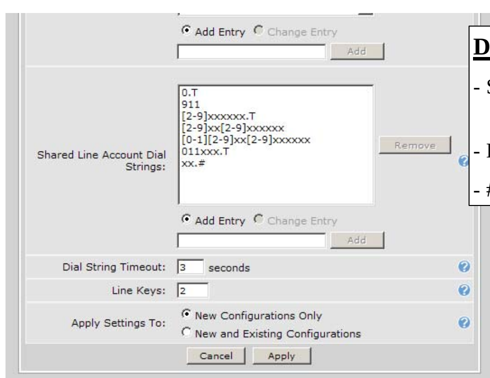
- SIP Server
- Dial String pattern for extensions

Phone Config Files - polycomConfigDefaults.cfg

Phone Config Files
polycomConfigDefaults.cfg



- Boot Settings tab continued…



Default Phone Settings

- Shared Line Account Dial String patterns
- Dial String Timeout
- # of Line Keys

Other Polycom Files

Other Polycom Files

- bootrom.ld
  - Generic program designed to load the application (SIP). The bootROM application uses the network to query the boot server for upgrades or configuration changes.
- sip.ld
  - The SIP firmware image file (phone application)
- sip.ver
  - Version number of SIP image
- [MAC]-phone.cfg
  - Local configuration changes made on the phone and uploaded to the boot server if allowed
- [MAC]-directory.xml
  - Per Phone Directory

Other Polycom Files (Continued)

Other Polycom Files (Continued)…

- SoundPointIPWelcome.wav
  - Wav file that's played when phone boots
- Doorbell-1-Hi.wav
  - Doorbell Wav file that can be used for door phone
- adtran40xlogo.bmp
  - Bitmap for IP430 phone (monochrome – 94x23) pixels)
- adtran50xlogo.bmp
  - Bitmap for IP500 phone (4 bit grayscale - 114x51 pixels)
- adtran60xlogo.bmp
  - Bitmap for IP600 phone (4 bit grayscale – 209x109 pixels)
- adtran4000logo.bmp
  - Bitmap for IP4000 phone (4 bit grayscale – 150x33) pixels)

Other Polycom Files (Continued)

Other Polycom Files (Continued)…

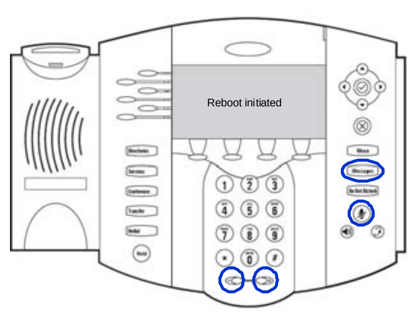
- [MAC].app.log
  - The application log file is uploaded periodically or when the local copy reaches a predetermined size.
- [MAC].boot.log
  - The boot log file is uploaded to the boot server after every reboot. A ~now-boot.log or ~now-app.log version of these files may also be seen.

Manual Reboot of Phone

Manual Reboot of Phone

- Rebooting from phone menu
  - The menu option is called Restart Phone and it is found in the Settings menu
- Rebooting with phone key combination
  - For the key combination, press and hold the following keys simultaneously until a confirmation tone is heard or for about three seconds:

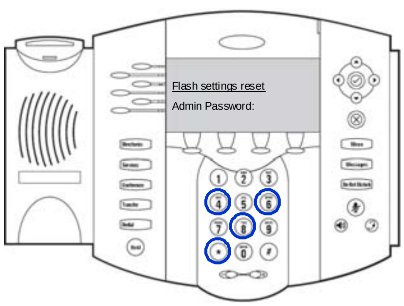
SoundPoint® IP 300 and 301:	Volume-, Volume+, Hold, Do Not Disturb
SoundPoint® IP 500 and 501:	Volume-, Volume+, Hold, Messages
SoundPoint® IP 600 and 601:	Volume-, Volume+, Mute, Messages
SoundStation® IP 4000:	*, #, Volume+, Select



Reset to Factory Default

Reset to Factory Default

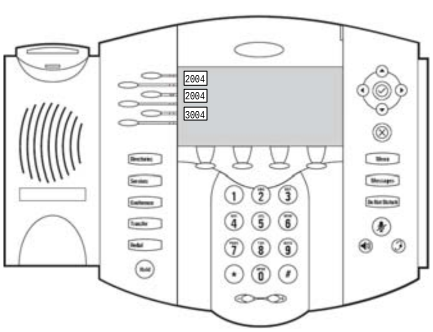
- The basic network configuration can be reset to factory defaults
  - Simultaneously press and hold the 4, 6, 8 and * dial pad keys until the password prompt appears
  - Enter the administrator password to initiate the reset



Registration, Line and Call Appearance

Registration, Line, and Call Appearance

Polycom phones can support multiple registrations, line appearances, and call appearances



Registration: a SIP alias or phone number

Line key: a button on the phone beside the display

Call: a SIP session

- Registrations can appear on multiple line keys

- Line keys can have multiple calls

Module Objectives

Module Objectives

ADTRAN

At the end of this module, you should:

- Have a basic understanding of ADTRAN phone configuration files
- Be able to modify phone configuration files in the GUI

Module 5: NetVanta 7000 Key System Application

Module Objectives

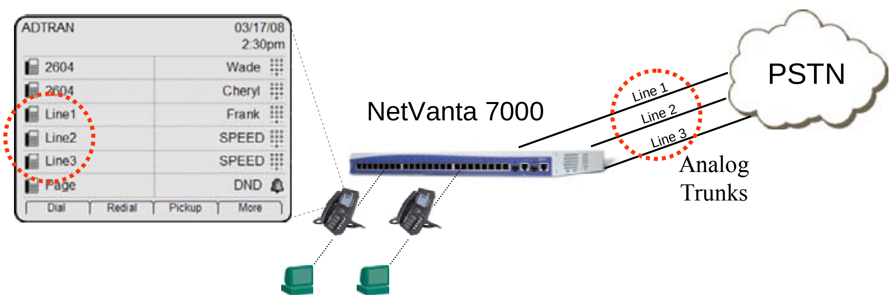
Module Objectives

- Introduce NetVanta 7000 Key System Applications
- Voice Trunk Review
- Configure Shared Line Accounts
- Enable Hands Free Auto-Answer
- Introduce and Configure System Modes
- Conduct Voice Troubleshooting in a NetVanta 7000 Key System Application

NetVanta 7000 - Key System Application

NetVanta 7000 Key System Application

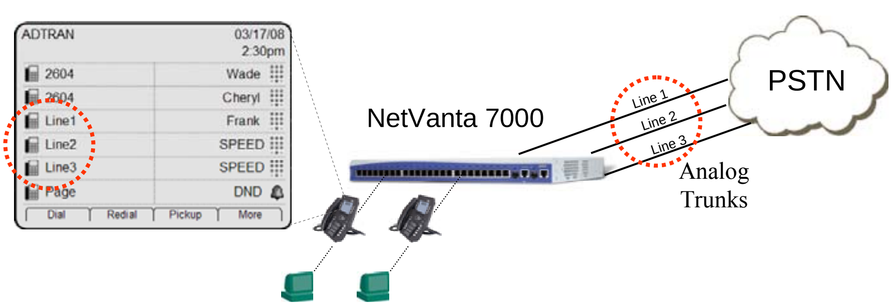
- What is a Key System?
  - Key Systems are typically used in small office environments
  - Allow a group of telephones to access a number of individual telephone lines
  - Characterized by telephones with buttons used to access, or answer, lines



NetVanta 7000 - Key System Features

NetVanta 7000 Key System Features

- Configuration of the following features are introduced in this section:
  - Shared Line Appearance (SLA)
  - Hands-free Auto-Answer
  - System Scheduler (Day/Night)



Voice Trunk Review

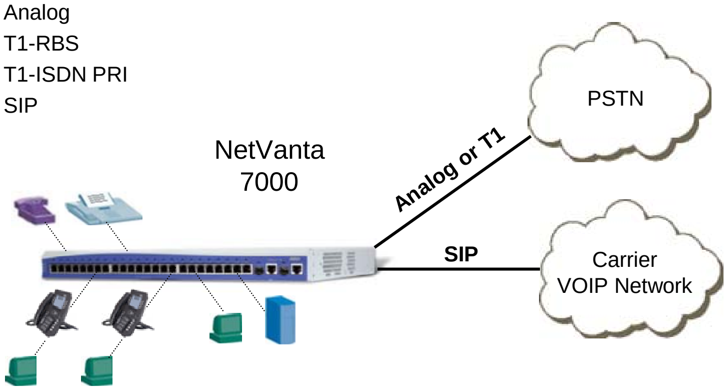
NetVanta IP Telephony Course

Voice Trunk Review

Voice Trunk Review

Voice Trunk Review

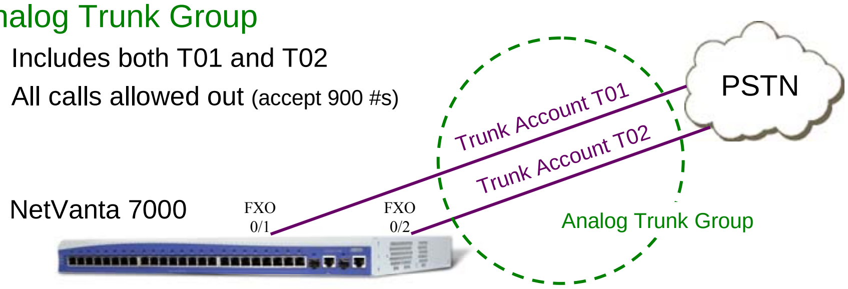
- **Trunk** lines connect the NetVanta 7000 to the outside world. They are delivered from the carrier and may be digital or analog.
  - Supported Voice Trunk Types
    - Analog
    - T1-RBS
    - T1-ISDN PRI
    - SIP



Factory Default Review

NetVanta 7000 Voice Trunks Factory Default Review

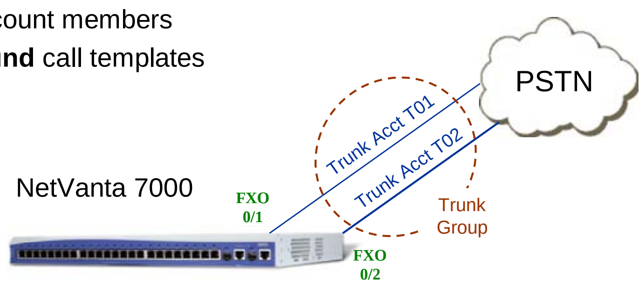
- Trunk Account T01
  - Physical Interface = FXO 0/1
  - Inbound call goes to Auto Attendant (8200)
- Trunk Account T02
  - Physical Interface = FXO 0/2
  - Inbound call goes to Auto Attendant (8200)
- Analog Trunk Group
  - Includes both T01 and T02
  - All calls allowed out (accept 900 #s)



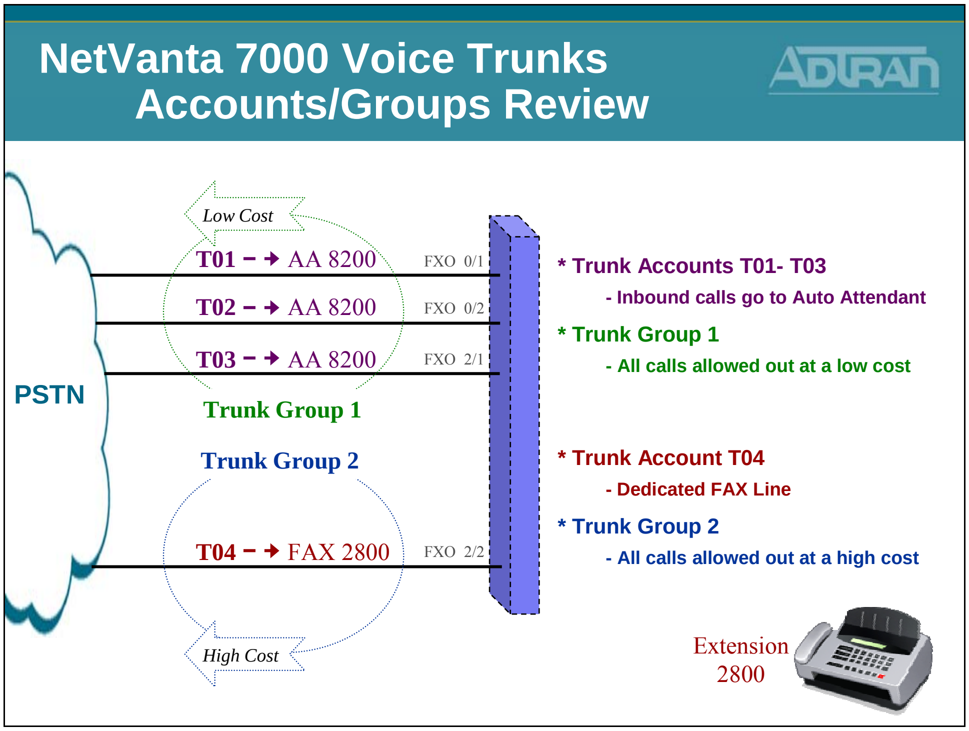
Basic Configuration Steps

Analog Trunk Review Basic Configuration Steps

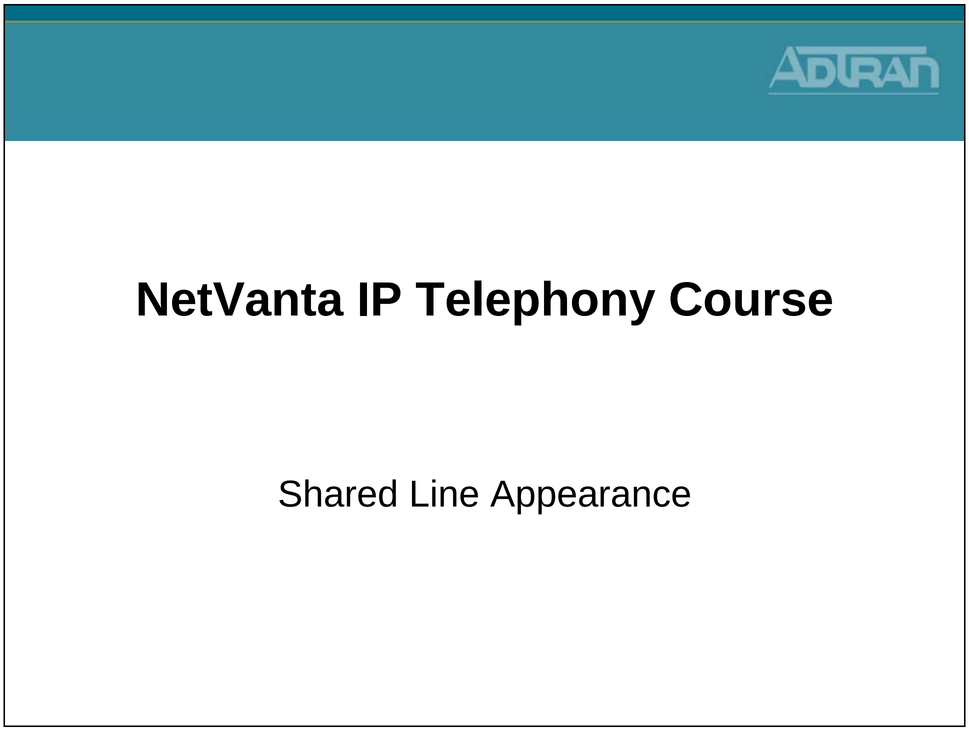
1. Configure Trunk Physical Interface
   - FXO interfaces enabled by default
2. Create Trunk Account
   - Configure **trunk number**, caller-id, etc..
   - Assign physical **FXO** port(s)
3. Create Trunk Group
   - Add Trunk Account members
   - Define **outbound** call templates



Trunk Accounts/Groups Review



Shared Line Appearance



A Shared Line Appearance (SLA) is a configurable portion of the NetVanta 7000 that allows system administrators to enable the key system mode on the unit. A Shared Line Account is created and then linked to the IP phone’s line keys and functions similar to a key system where the system enables multiple phone users to share the same analog trunk lines. SLAs allow businesses to cut the cost of providing individual analog phone lines from the carrier to each analog phone station in their facility.

For example, company XYZ has 16 sales employees that need a secondary extension on their ADTRAN desktop IP phones. Instead of ordering an additional 16 trunk lines, company XYZ orders three analog trunk lines and shares the lines between the sales team phone users.

SLAs offer features, such as public hold/retrieve, line status display on subscribers' phones, and the ability to place an outbound call out of a selected trunk.

Outlined below are key aspects of SLAs:

- SLAs can only be associated with analog trunks.
- Inbound calls on an SLA notify every SIP-based IP phone that registers to it.
- SLAs can be seized by pressing the appropriate line key(s) on the phone.
- The status of an SLA will be updated on all other phones registered to that line. Status conditions include idle, ringing, busy, and hold.
- A busy SLA cannot be seized by other SIP-based IP phones. Barge or monitoring of the SLA is NOT supported.
- SLAs can have coverage to auto attendant (AA), voicemail (VM), operator, extension, and an external number.

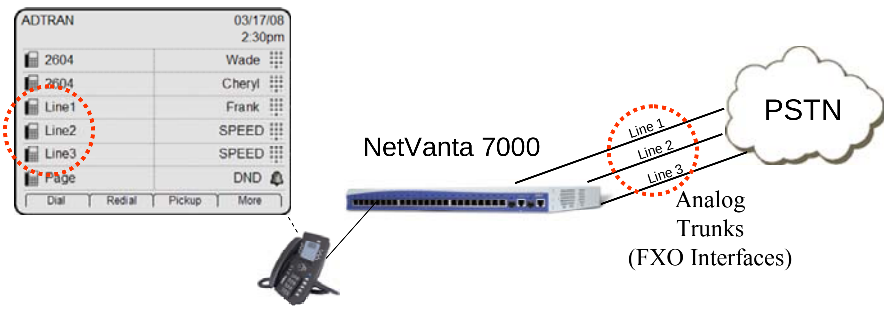
Using Public Hold

When using SLAs, calls that are put on hold are referred to as being on Public Hold, which means that every user on that particular SLA has access to the call placed on hold. Also, any phone registered to that SLA will be able to see the hold status and retrieve the call by pressing the line key corresponding to the call on their phone.

Shared Line Appearance

Shared Line Appearance

- Description
  - The key system functionality for SLA is targeted to make the IP PBX experience similar to existing key systems that some end users are accustomed to
  - A “Shared Line Appearance” refers to a line key on a phone that maps to an analog trunk



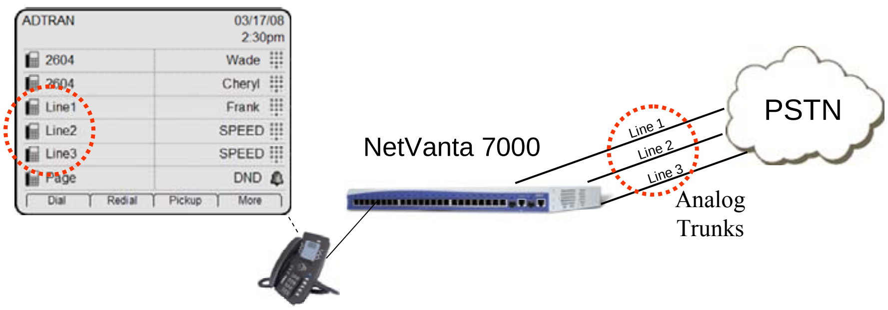
SLAs and Analog Trunk Lines

In order for analog trunk lines to appear and be accessible for incoming and outgoing calls for multiple IP phone users, the trunk line(s) must be configured and linked to an SLA. Once the SLA is associated with the trunk line, the SLA can be linked to individual phones. Prior to the introduction of SLAs, all calls were routed out the trunk groups and were only subjected to the permit/deny templates assigned under the trunk group settings. Now, SLAs contain their own set of permit/deny templates. This application allows trunk accounts that are used as SLAs to also be included in an outbound trunk group. Therefore, when a user presses the corresponding line key on a phone that only has SLA’s extensions programmed, they are subjected to the SLA’s Accept/Reject Templates. When running AOS A1 firmware or later, the SLA Accept/Reject Templates are not applied to inbound and outbound calls if the phone has a private line (user account) programmed. Instead, the individual user assigned class of service permission settings are applied to all outbound calls.

Shared Line Appearance - Features

Shared Line Appearance Features

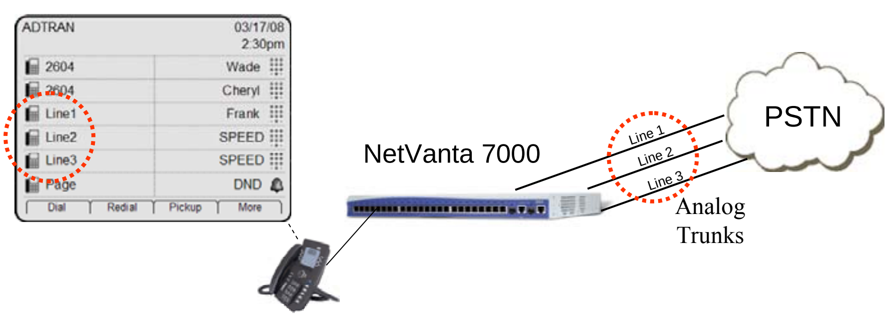
- Inbound calls on a LINE notify every SIP phone that registers to the line
- A LINE can be seized by selecting the LINE button on the phone
- The status of a LINE will be updated on all other phones registered to that line
  - Status will include IDLE, RINGING, BUSY, and HOLD



Shared Line Appearance - Features (Continued)

Shared Line Appearance Features (Continued…)

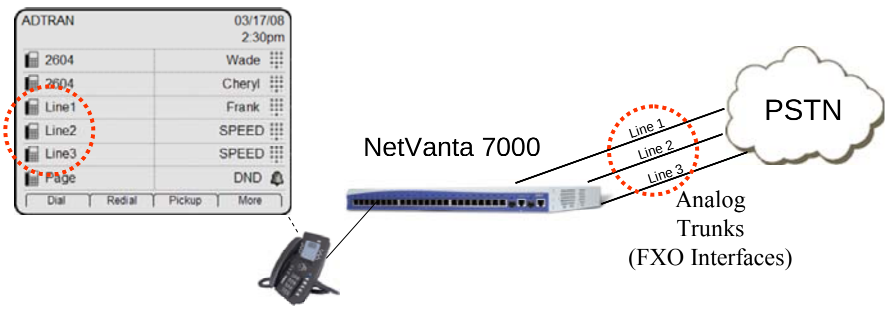
- A LINE on hold may be retrieved by any other SIP Phone that registers on the LINE
- A busy LINE cannot be seized by other SIP Phones
  - Barge or Monitoring of the LINE is NOT supported
- A LINE can have coverage to AA, VM, Operator, extension, and an external number



Shared Line Appearance - Basic Configuration

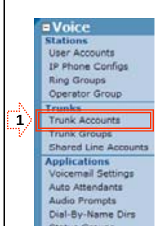
Shared Line Appearance Basic Configuration Steps

1. Create Analog Trunk Accounts
2. Create Shared Line Accounts (SLA)
   - Specify the Trunk Account that will be associated to this SLA
   - Configure Accept/Reject template for this SLA
3. Configure key on phone as a SLA

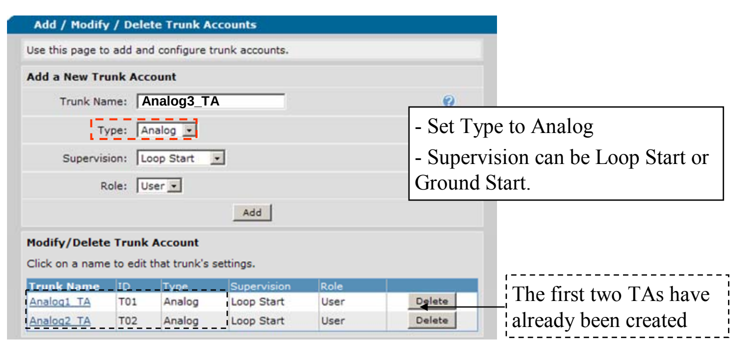


1) Create Trunk Account

SLA Configuration 1) Create Trunk Account



1) Create Analog Trunk Accounts (*as done in past*)

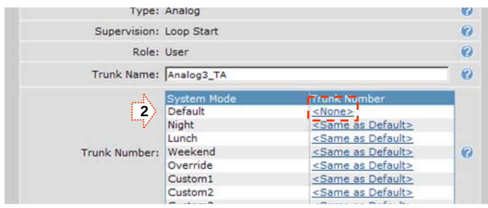


- Create one Analog Trunk Account per incoming analog line (FXO interface)

1) Create Trunk Account

SLA Configuration
1) Create Trunk Account

2) Remove the existing extension number
   - Inbound calls will go to all users registered to the Shared Line Account (SLA)

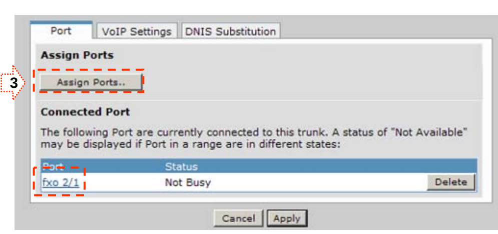


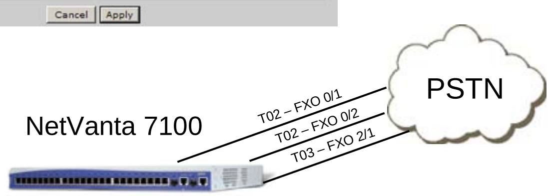
   - A different Trunk Number could be assigned for each of the System Modes

1) Create Trunk Account

SLA Configuration
1) Create Trunk Account

3) Assign one FXO interface per Trunk Account

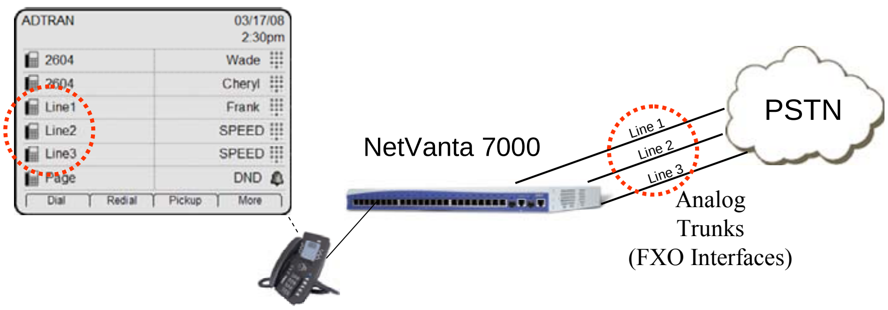




Shared Line Appearance - Basic Configuration

Shared Line Appearance Basic Configuration Steps

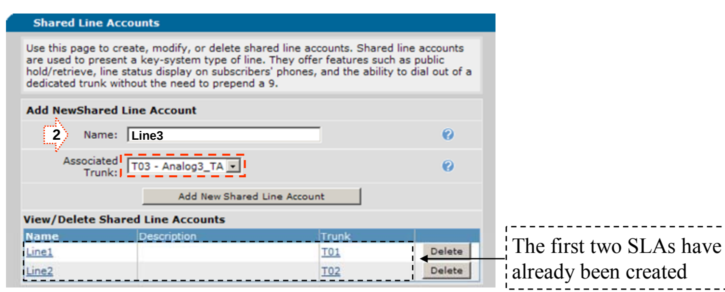
1. Create Analog Trunk Accounts
2. Create Shared Line Accounts (SLA)
   - Specify the Trunk Account that will be associated to this SLA
   - Configure Accept/Reject template for this SLA
3. Configure key on phone as a SLA



2) Shared Line Accounts

SLA Configuration 2) Shared Line Accounts

1) Select the Voice / Trunks / Shared Line Accounts menu

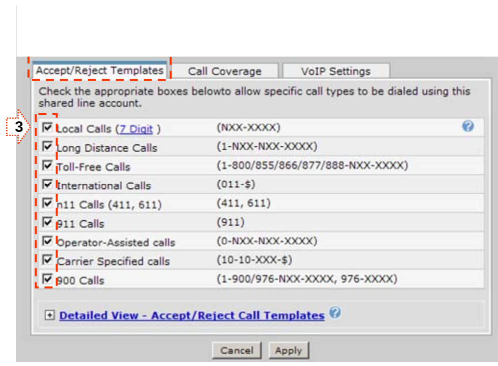


2) Create one Shared Line Account per Analog Trunk Account
   - Select the Analog Trunk Account that this shared line will be associated with

2) Shared Line Accounts

SLA Configuration
2) Shared Line Accounts

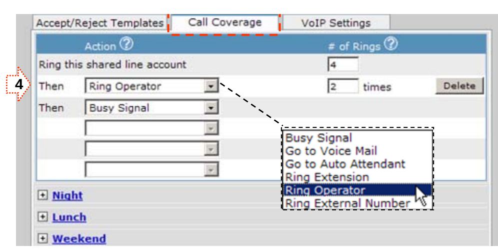
3) Specify the type of calls that will be allowed out with this Shared Line Account



2) Shared Line Accounts

SLA Configuration
2) Shared Line Accounts

4) Define the Call Coverage that will be used for this Shared Line Account

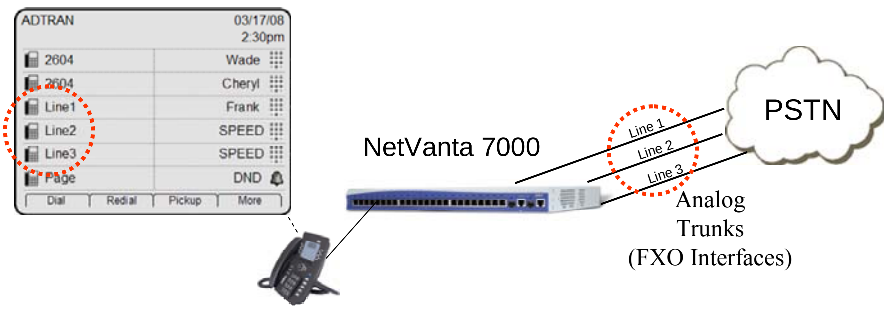


– A different Call Coverage could be defined for each of the System Modes

Shared Line Appearance - Basic Configuration

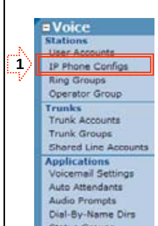
Shared Line Appearance
Basic Configuration Steps

1. Create Analog Trunk Accounts
2. Create Shared Line Accounts (SLA)
   - Specify the Trunk Account that will be associated to this SLA
   - Configure Accept/Reject template for this SLA
3. Configure key on phone as a SLA

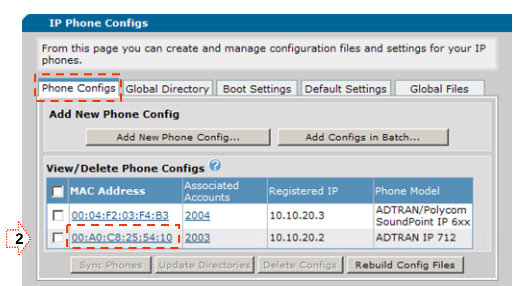


3) Line Key on Phone

SLA Configuration
3) Line Key on Phone

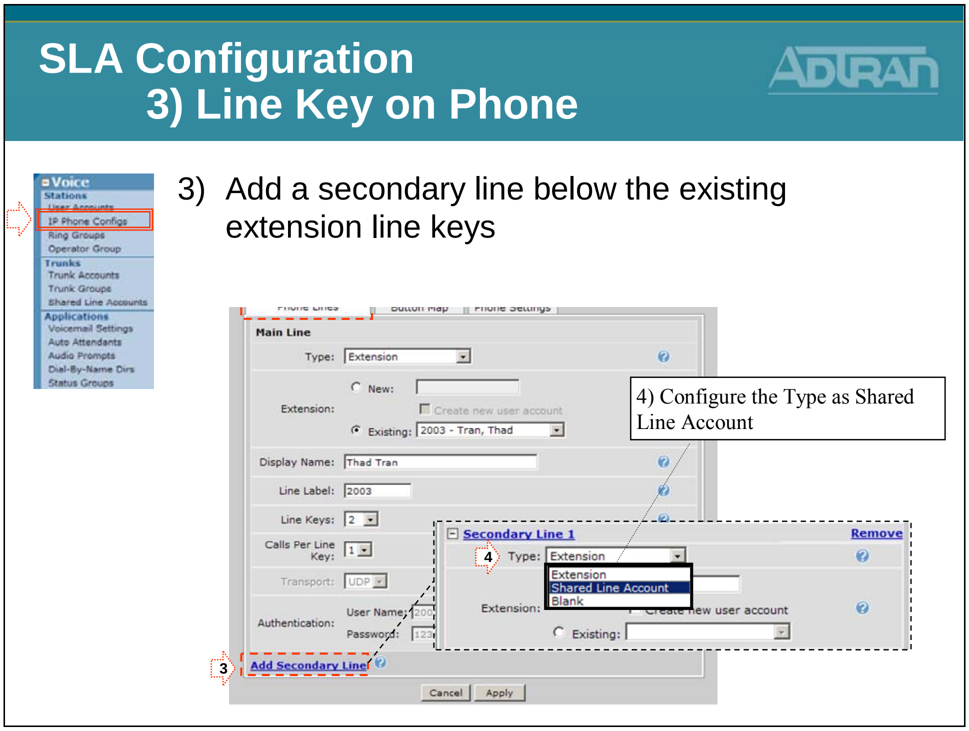


1) Select the Voice / Stations / IP Phone Configs menu

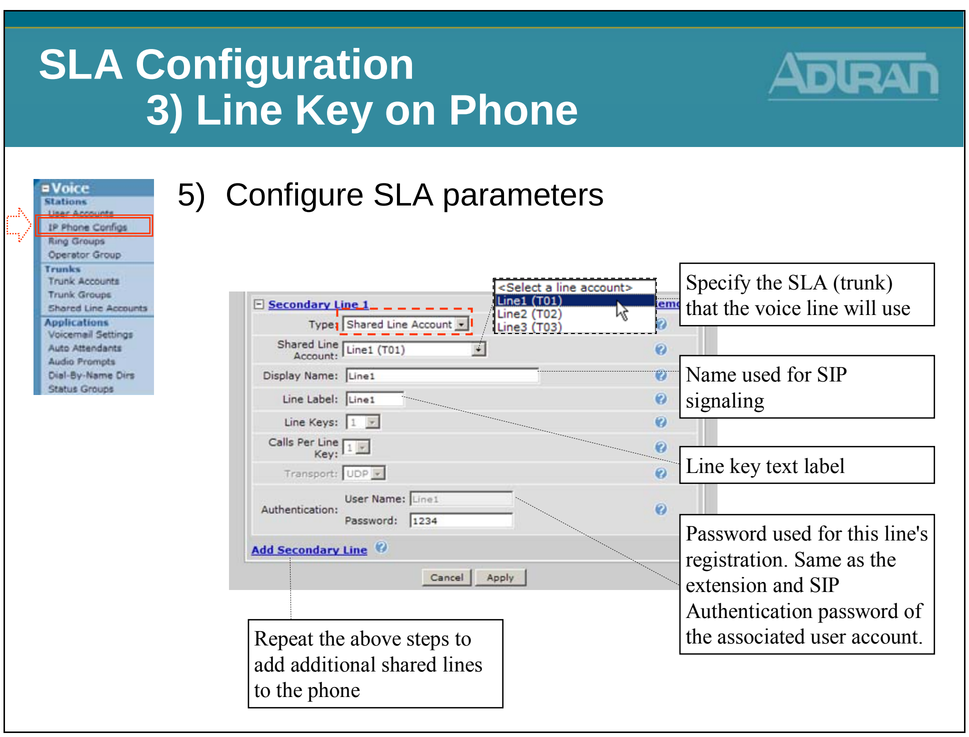


2) Select the MAC address of the phone you wish to add a shared line to

3) Line Key on Phone

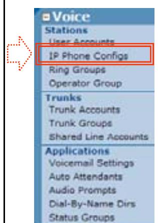


3) Line Key on Phone

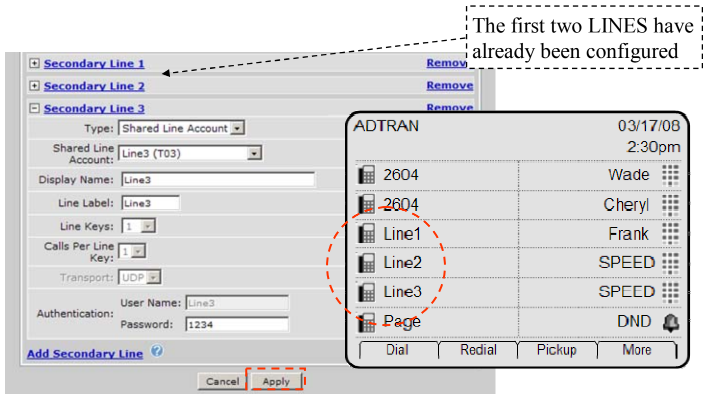


3) Line Key on Phone

SLA Configuration
3) Line Key on Phone



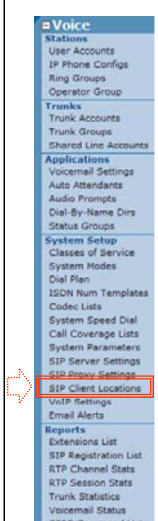
6) Click Apply and then reboot the phone



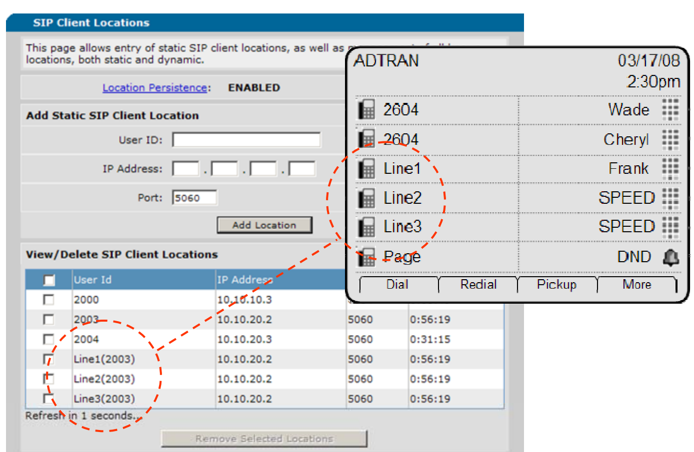
– After phone reboots, the new Shared Line Accounts appear on phone

Shared Line Account - View Registration

Shared Line Account
View Registration



- Shared Line Accounts are registered with the NetVanta 7000



Hands Free Auto Answer

NetVanta IP Telephony Course

Handsfree Auto Answer

Hands Free Auto Answer

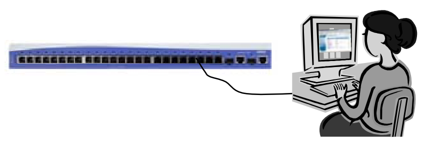
Handsfree Auto Answer

- Hands Free Auto-Answer is an intercom like feature
  - User initiates a call to a SIP phone
  - Instead of requiring the recipient to answer the call, the phone automatically answers and starts a conversation
- Supported with:
  - ADTRAN IP 706 and IP 712
  - Many Polycom Phones - (IP 650, IP 601, IP 501, IP 430)

Hands Free Auto Answer - Basic Configuration

Handsfree Auto Answer Basic Configuration Steps

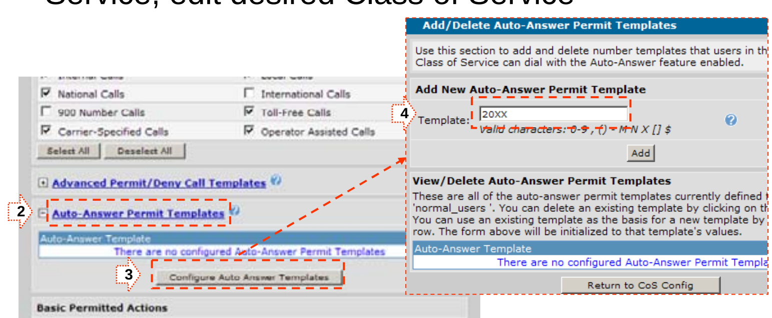
1. Configure auto-answer permit template
   – Template defined per voice Class of Service
2. Optional – Give voice users permission to block incoming auto-answer calls
   – Configured per voice Class of Service
3. Optional – Block incoming auto-answer calls for specific voice user
   – Configured per specific voice user extension



1) Configure AA Permit Template

Hands-Free AA Configuration 1) Configure AA Permit template

1) From Voice / System Setup / Classes of Service, edit desired Class of Service



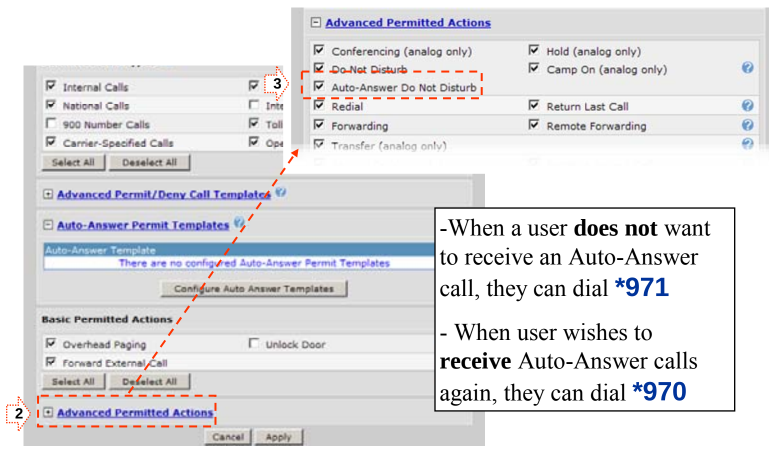
2) Auto-Answer permit templates area created per Class of Service

   – Configuration is very similar to the call accept template

2) Block Incoming AA calls

Hands-Free AA Configuration
2) Block Incoming AA calls

- Optional – Per CoS, allow users to block incoming auto-answer calls

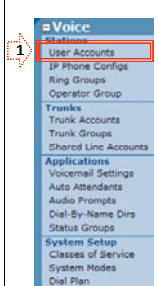


-When a user **does not** want to receive an Auto-Answer call, they can dial ***971**

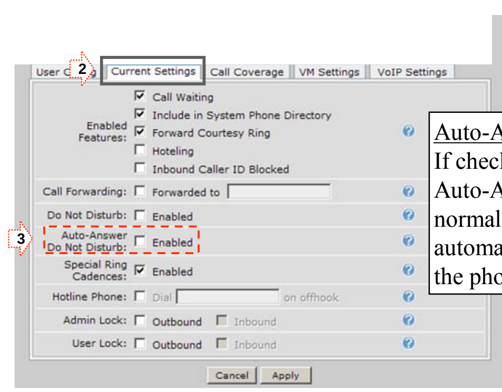
- When user wishes to **receive** Auto-Answer calls again, they can dial ***970**

3) Block Incoming AA calls

Hands-Free AA Configuration
3) Block Incoming AA calls



- Optional – Block incoming auto-answer calls for specific user



Auto-Answer Do Not Disturb
If checked, any incoming Auto-Answer calls will ring normally instead of being automatically answered by the phone

– While editing a Voice User, select the Current Settings tab

Hands Free Auto-Answer - Placing Call

Hands Free Auto-Answer
Placing Call

- To place an Auto-Answer call, the digits ****** must precede the number
  - The prefix can be dialed before or with the extension
  - For example, a user could place two calls:
    - ****** and then **MXXX**, or a user could dial ****MXXX**

Hands Free AA - No Permission or Blocked

Hands Free Auto-Answer
No Permission or Blocked

- Auto-Answer call initiated by a user that "does not have permission" to do so
  - Normal call is placed
  - No Auto-Answer functionality
- Auto-Answer call is received by a user that has blocked the functionality
  - A normal call is placed
  - No Auto-Answer functionality

System Scheduler

NetVanta IP Telephony Course

System Scheduler

System Scheduler - What is it?

Voice System Scheduler What is it?

- System Scheduler allows a user to define different configuration parameters based on day and time
  - Change voice users call privileges
  - Redirect incoming calls to a different number
  - Schedule different Auto Attendant greetings
  - Define different call coverage

System Scheduler - Modes of Operation

Voice System Scheduler Modes of Operation

System Scheduler has two modes of operation:

- Automatic Mode
  - A schedule can be defined to transition system modes at predetermined days and times, automatically
  - When a schedule is defined and active, no manual modes can be enabled, except for the override mode
    - Once the override mode is enabled, it stays in effect until manually disabled
- Manual Mode
  - Allows users to set the mode of operation manually
  - Once a mode is selected, the mode stays in effect until manually changed
  - All manual changes to system mode can occur via BLF and SPRE codes

System Scheduler - Predefined Modes

Voice System Scheduler Predefined Modes

- The are several predefined modes and custom modes where the schedule can be modified:
  - Default
  - Night
  - Lunch
  - Weekend
  - Custom1
  - Custom2
  - Custom3
  - Override

System Mode Feature

System Mode Operation is a feature in the NetVanta 7000 that allows a user to define different configuration parameters, such as User Class of Service, Trunk Account Number, and Call Coverage, based on the current mode. There are 7 configurable System Modes with one Override option. The System Mode can be configured to change on a schedule at a specific transition time or can be manually switched by the user without a schedule.

The 7 System Modes are:

- Default
- Night
- Lunch
- Weekend
- Custom1
- Custom2
- Custom3
- Override (enable/disable; stays in effect until disabled)

System Modes can be enabled by schedule, web interface, Auto Attendant digit action, or SPRE code. They can be monitored by a BLF key in a Status Group (IP 601, IP 706, IP712 phones).

Configuration Overview

1. Determine if scheduled or manual operation is desired and define a schedule for each System Mode if required.

2. Determine inbound call flow for Trunk Accounts and configure appropriately for each System Mode.

3. Determine Call Coverage for User Accounts, Operator Group, and any other Ring Groups and configure appropriately for each System Mode.

4. Determine Class of Service for User Accounts and configure appropriately for each System Mode

Allowing the Change of System Mode

In order to allow a phone to dial a SPRE code or use a BLF key to change the System Mode, this action must be enabled in the Advanced Permitted Actions for the Class of Service applied to the desired User Account. This applies to changing to any System Mode in Manual Operation, or to enabling Override in Scheduled Operation.

System Scheduler Override Mode

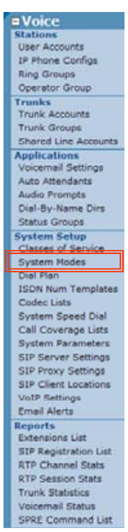
Voice System Scheduler Override Mode

- When a schedule is defined and active, no manual modes can be enabled, except for the override mode
- Once the override mode is enabled, it stays in effect until manually disabled
- If override mode is enabled and a scheduled transition occurs, the override remains and the transition is ignored
- When the override is disabled, the currently scheduled mode of operation takes effect

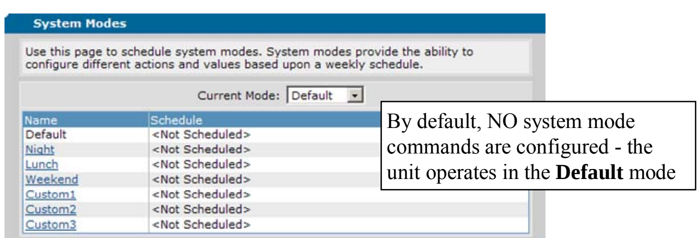
- The user/admin defines the action taken when in the override mode

System Mode Configuration

Voice System Scheduler System Mode Configuration



1. Select Voice / System Setup / System Modes from the NetVanta 7XXX menus

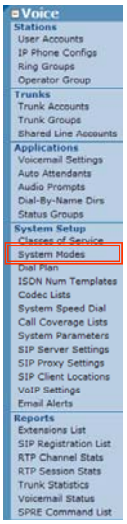


2. Click one of the existing System modes
3. Define schedule for the selected System Mode

NOTE: *Only define a schedule if you want the system mode to automatically change based on time of day/day of week*

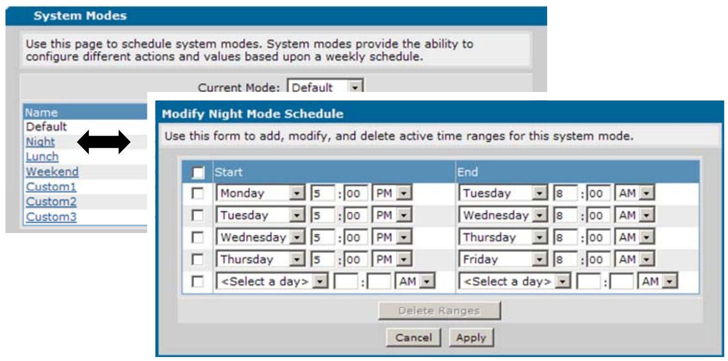
System Modes - Example

System Modes Example



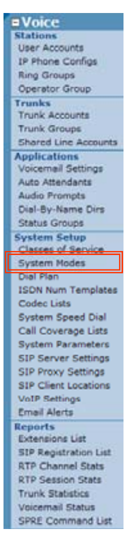
Night System Mode Example

- System will transition into the Night system mode at **5 PM** and back to the Default system mode at **8 AM**



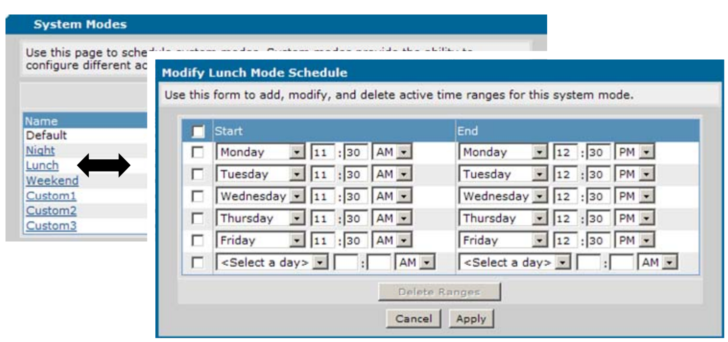
System Modes - Example

System Modes Example



Lunch System Mode Example

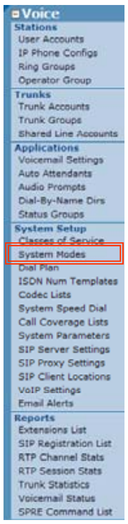
- System will transition into the Lunch system mode at **11:30 AM** and back to the Default system mode at **12:30 PM** each week day



System Modes - Example

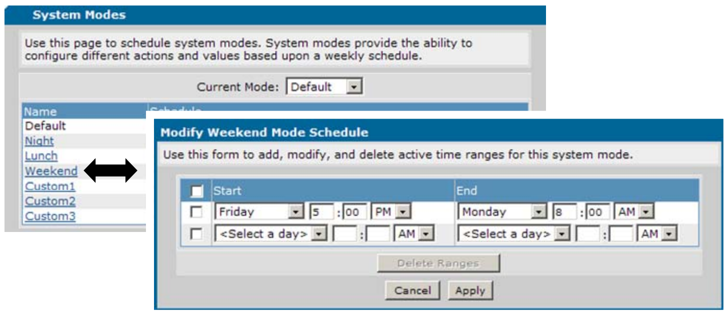
System Modes Example

ADTRAN



Weekend System Mode Example

- System will transition into the Weekend system mode at **5 PM Friday** and back to the Default system mode at **8 AM Monday**



System Modes - Where can they be applied?

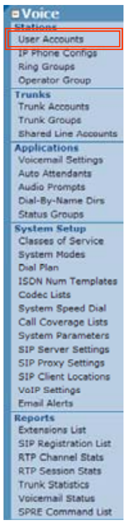
Voice System Modes Where can they be applied?

ADTRAN

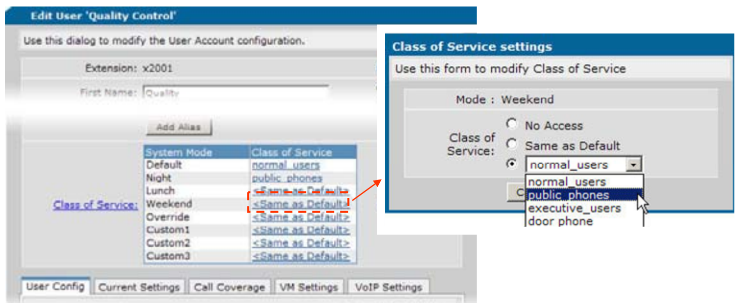
- Voice User - Class of Service
  - The CoS can be set to change for the user based on the current system mode
  - Defines the types of phone service that will be available to the user during the time period
- Voice Account – Trunk Number
  - Activate different system modes of operation that redirect incoming calls to a different number depending on the specified mode
    - Could have different Auto Attendants based on time and day
- Call Coverage
  - Call coverage can be configured per system mode
  - The number of rings between call coverage choices can also be set per system mode
    - Voice users, ring groups, and operator group, shared line accounts

Assigning System Modes - User Account CoS

Assigning System Modes
User Account Class of Service

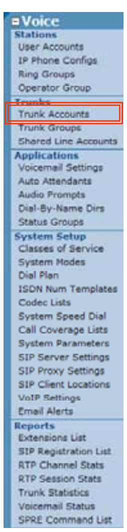


1. Edit an existing voice user
2. Define Class of Service per System Mode
   - No Access
   - Same as Default
   - An existing Class of Service

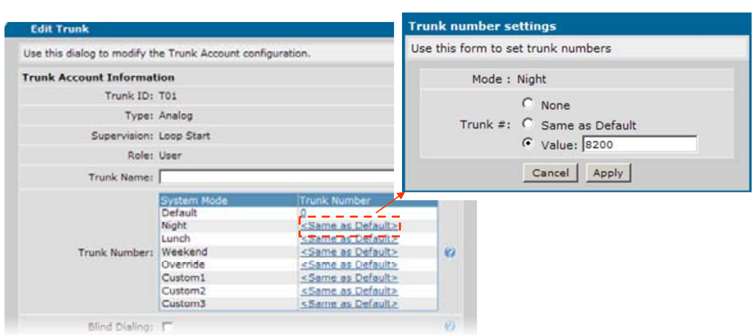


Assigning System Modes - Trunk Number

Assigning System Modes
Voice Account – Trunk Number



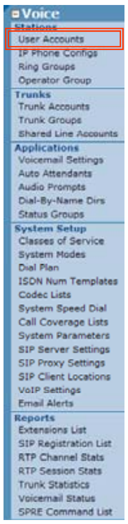
1. Edit an existing Trunk Account
2. Define Trunk Number per System Mode
   - None / Same as Default / Value - *extension*



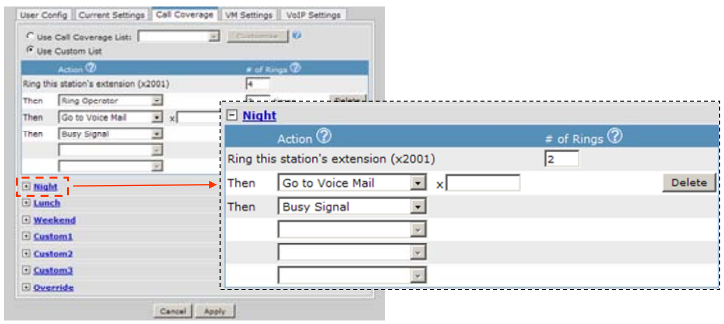
Assigning System Modes - Call Coverage

Assigning System Modes
Call Coverage

ADTRAN



1. Edit an existing voice user or ring group
2. Define Call Coverage per System Mode
   – Voice User Accounts / Ring Group / Operator Group



System Modes - Methods to Change

Voice System Modes
Methods to Change

ADTRAN

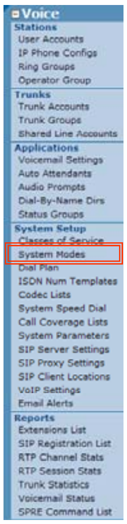
- Automatically switch based on schedule
- Web
- Auto Attendant
- SPRE Code
- BLF/SPRE Code

* If the system is in override, the unit will ignore any schedule that exists

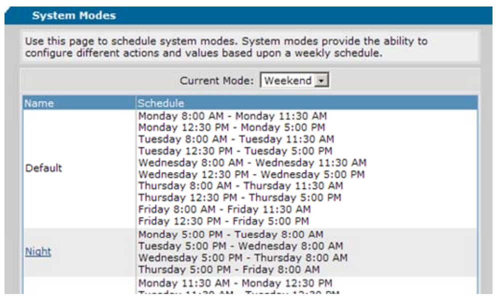
- The unit will stay in override until manually changed
  - This command is saved into the dynvoice-config file to preserve the state of the unit in case of power failure

Changing System Mode - Switch as Scheduled

Changing System Mode
Switch at Scheduled Time



- The System Mode will automatically change at scheduled time
  - If placed in the Override mode, it will no longer change until taken out of the Override mode

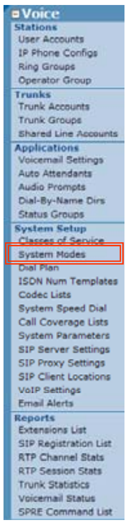


Scheduled Operation

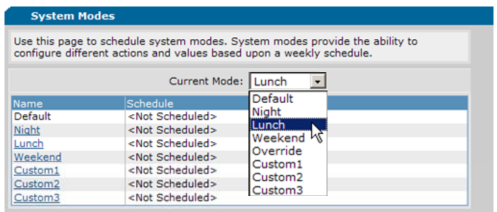
The NetVanta 7000 can be configured to automatically switch System Modes based on a schedule defined for each System Mode. When a schedule is defined for System Modes, the only option to disable the schedule is via the Override mode. Override is an enable/disable function. Once the NetVanta 7000 is in Override mode, it will remain there until Override is disabled (via BLF, SPRE, Auto Attendant or web interface). Override functions as a Toggle; to disable Override mode from the Auto Attendant, you must select Override mode from the choices given.

Changing System Mode - Manually Change in GUI

Changing System Mode
Manually Change in GUI



- If no day/time schedule has been configured, the admin can manually set the mode of operation in the GUI



- Once a mode is selected, the mode stays in effect until manually changed

NOTE*: If a day/time schedule has been defined, the only mode that can be selected from this menu is Override.*

The voice current-mode command can also be used to manually activate a particular system mode on the unit from the command line.

```
NV7000 (config)# voice current-mode default
                 voice current-mode lunch
                 voice current-mode night
                 voice current-mode override
                 voice current-mode weekend
                 voice current-mode custom1
                 voice current-mode custom2
                 voice current-mode custom3
```

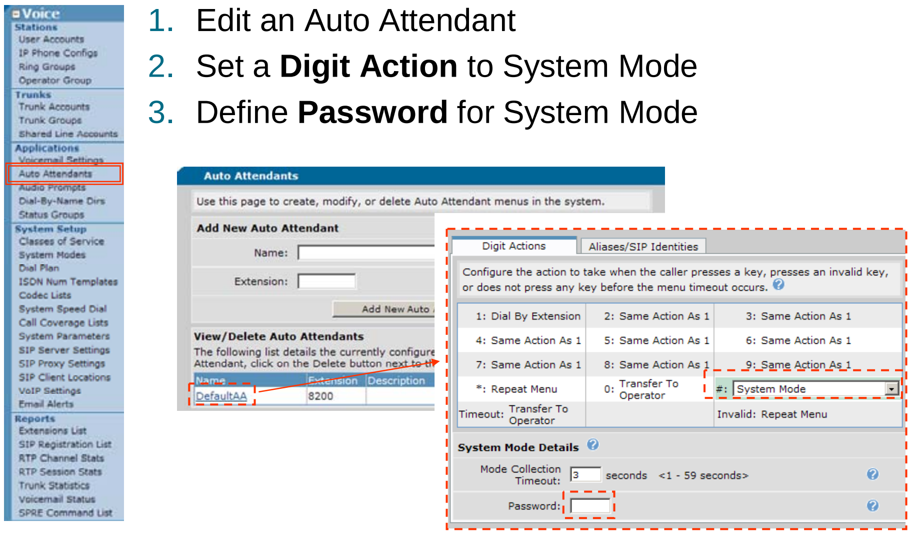
This command is used to put the unit into a specific system mode. The unit remains in the activated system mode until it is changed manually.

* If a day/time schedule has been defined, the only mode that can be set here is override.

Changing System Mode - Auto Attendant

Changing System Mode
Auto Attendant

1. Edit an Auto Attendant
2. Set a **Digit Action** to System Mode
3. Define **Password** for System Mode



Changing System Mode - Auto Attendant

Changing System Mode
Auto Attendant

- A voice user who presses the digit action for System Mode will hear the following:

PASSWORD?
The Current System Mode is Default
To Transition to Night Mode Press 2
To Transition to Lunch Mode Press 3
To Transition to Weekend Mode Press 4
To Transition to Custom1 Mode Press 5
To Transition to Custom2 Mode Press 6
To Transition to Custom3 Mode Press 7
To Transition to Override Mode Press 8
To Cancel Press *

Changing System Mode - SPRE Mode

SPRE Codes used to Change System Mode

Dial the SPRE code for the desired System Mode from any phone.

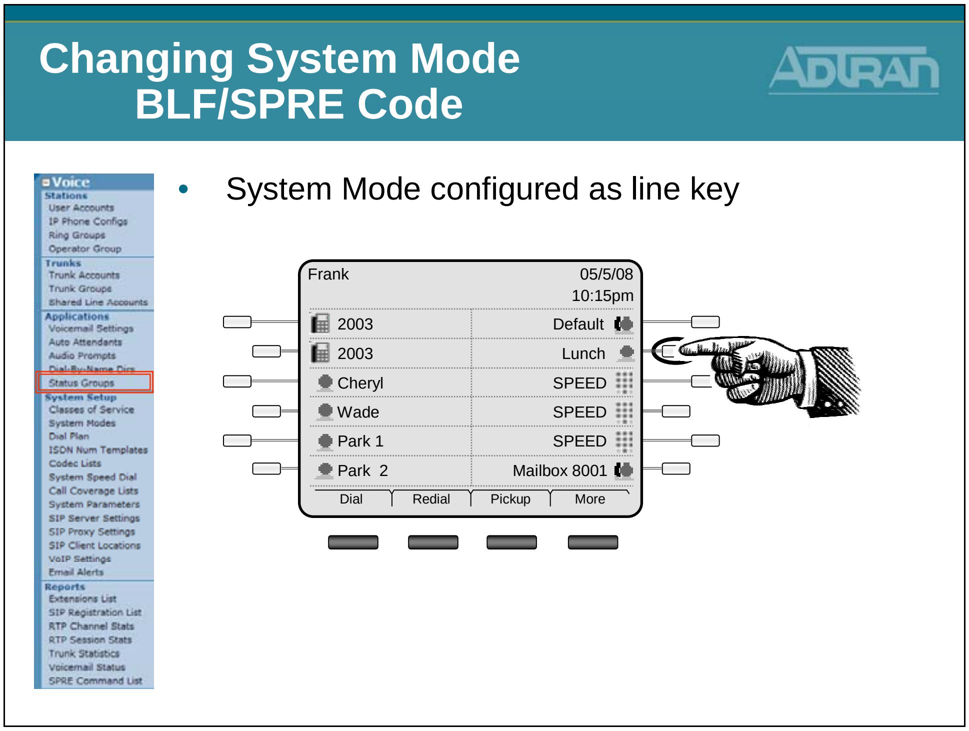
The SPRE Codes to enable/disable System Modes are these:

- *200 – Default
- *201 – Night
- *202 – Lunch
- *203 – Weekend
- *204 – Custom1
- *205 – Custom2
- *206 – Custom 3
- *207 – Override

Allowing the Change of System Mode

In order to allow a phone to dial a SPRE code or use a BLF key to change the System Mode, this action must be enabled in the Advanced Permitted Actions for the Class of Service applied to the desired User Account. This applies to changing to any System Mode in Manual Operation, or to enabling Override in Scheduled Operation.

Changing System Mode - BLF/SPRE Code



BLF Key in Status Group

A Status Group can be created to use a BLF to switch enable the different System Modes. A user would press the key associated with the desired System Mode to enable it.

For example, when the customer leaves the office for the day, they would press the Night key to enable Night mode. When they return to the office in the morning, they would press the Default mode to return to Default ("Day") mode. In Manual Operation, the Override option functions as just another Custom System Mode.

Allowing the Change of System Mode

In order to allow a phone to dial a SPRE code or use a BLF key to change the System Mode, this action must be enabled in the Advanced Permitted Actions for the Class of Service applied to the desired User Account. This applies to changing to any System Mode in Manual Operation, or to enabling Override in Scheduled Operation.

Key System Applications - Troubleshooting

NetVanta IP Telephony Course

Key System Applications
Troubleshooting

Troubleshooting SLAs

Troubleshooting SLAs

Shared Line Account Review:

- Configuration:
  - Is the proper trunk account associated with the SLA?
  - Are the proper call types checked in the Accept/Reject Templates in the SLA Config?
  - Is the proper call coverage defined in the SLA Config?
- Check the SLA registration with the phone
  - Is SIP password correct in the SLA Config?

Troubleshooting System Mode

Troubleshooting System Mode

System Mode Review:

- If "any" schedule has been defined in the Voice Settings / System Mode screen, the NetVanta 7000 is operating in the Automatic mode
  - System will automatically change based on time and day
  - Override Mode is used to take the system out of the Automatic mode and perform defined Override Mode settings
  - Stays in Override Mode until taken out of Override Mode
- Changing System mode with SPRE code or BLF
  - The Class of Service setting "Change System Mode" must be enabled for the admin user that needs this ability

Troubleshooting Auto Answer

Troubleshooting Auto Answer

Hands Free Auto Answer Review:

- No Auto Answer Calls are allowed by default
- Class of Service setting
  - Define which users can place Auto Answer Calls
  - Template defines who Auto Answer calls can be placed to
- ****** is used to place Auto Answer call
- Is feature disabled for user? (SPRE code *971)
  - Use SPRE code *970 to enable

show interface fxo 0/1

show interface fxo 0/1

- View the statistics for the specified interface

```
NV7000# show int fxo 0/1
fxo 0/1 is UP
  Two-wire Status:    Onhook
  Test Status:        INACTIVE
    No Tests
  Impedance:          600 ohms +2.16uF
  Transmit Gain:      +0.0dB
  Receive Gain:       +0.0dB
  Measured ERL:       not available - run 'test erl'
```

The default Impedance setting is 600 Ω + 2.16 µF. The unit may require a different setting to correct echo issues. For assistance, refer to the Echo Return Loss Measurement Guide on ADTRAN's Knowledge Base at kb.adtran.com.

show interface fxs 0/1 realtime

show interface fxs 0/1 realtime

- View interface statistics real time

```
NV7000# show int fxs 0/1 realtime
-----------------------------------------------------------------------------
fxs 0/1 is UP
  Two-wire Status is: Onhook
  Test Status is INACTIVE
    No Tests
  Impedance is:  600 ohms +2.16uF
  Transmit Gain is:  -6.0dB
  Receive Gain is:  -3.0dB
  Ring voltage is:  50 Vrms
  Signal Mode:  Loop-Start
  Caller ID Format is:  Multiple Data Message Format

-----------------------------------------------------------------------------
 Exit  -  'Ctrl-C',    Freeze  -  'f',    Resume  -  'r'
```

- Onhook
- Offhook
- Ringing

debug voice summary

debug voice summary

- View call routing summary real time
  - Can confirm proper trunk is being used

NV7000# **debug voice summary**
16:55:22 VOICE.SUMMARY voice user 2005 **cos allowed the call to Extensions**
16:55:22 VOICE.SUMMARY 2005 is **calling** 2006 (2006).
16:55:24 VOICE.SUMMARY 2005 is **connected** to 2006 (2006)
16:55:28 VOICE.SUMMARY Call from 2005 to 2006 (2006) **ended** by 2006: normal clearing

17:01:54 VOICE.SUMMARY voice user 2006 cos allowed the call to Extensions
17:01:54 VOICE.SUMMARY 2006 is calling **T01** (911).
17:01:56 VOICE.SUMMARY 2006 is connected to **T01** (911)
17:02:04 VOICE.SUMMARY Call from 2006 to T01 (911) ended by T01: normal clearing

Voice Trunk ID

debug interface fxo

debug interface fxo

- View interface events real time

Incoming Call

NV7000# **debug interface fxo**
2009.07.03 10:24:10 FXO.0/1 **Ringing Detected** 670041432 ms
2009.07.03 10:24:12 FXO.0/1 Ringing Removed 670043432 ms
2009.07.03 10:24:12 FXO.0/1 Normal Battery Detected 670043432 ms
2009.07.03 10:24:13 FXO.0/1 **Offhook** 670044481 ms
2009.07.03 10:24:13 FXO.0/1 Loop Current found - Battery detected, reset debounce 670044532 ms
2009.07.03 10:24:13 FXO.0/1 Normal Battery Detected 670044532 ms
2009.07.03 10:24:30 FXO.0/1 **Loop Current not present - Battery removed**, debounce 670061842 ms
2009.07.03 10:24:30 FXO.0/1 No Battery Detected 670061872 ms
2009.07.03 10:24:31 FXO.0/1 **Onhook** 670062372 ms
2009.07.03 10:24:31 FXO.0/1 Reverse Battery Detected 670062382 ms
2009.07.03 10:24:31 FXO.0/1 Normal Battery Detected 670062472 ms
2009.07.03 10:24:31 FXO.0/1 Onhook 670062972 ms

Disconnect

- The output above displays an incoming call from the PSTN on trunk FXO 0/1

debug voice phonemanager

debug voice phonemanager

- Display all phone manager event messages real time

```
NV7000# debug voice phonemanager

11:25:44:832 PM.0:1 Idle                 Processed OFFHOOK
11:25:44:832 PM.0:1 State change       >> Idle->Requesting Dialtone
11:25:44:832 PM.0:1 Requesting Dialtone  CACHG:ReqDigits on primary CA
11:25:44:832 PM.0:1 State change       >> Requesting Dialtone->SendingDigits
11:25:46:973 PM.0:1 SendingDigits        Digit 2 processed
11:25:48:033 PM.0:1 SendingDigits        Digit 0 processed
11:25:49:194 PM.0:1 SendingDigits        Digit 0 processed
11:25:50:454 PM.0:1 SendingDigits        Digit 3 processed
11:25:50:455 PM.0:1 State change       >> SendingDigits->Call Pending
11:25:50:457 PM.2003 Ca:0 SipPM_Idle         rcvd: CAS_Ringing

* Partial output displayed
```

View digits as entered from Analog phone

- Could also use “debug voice toneservices” to see tone events

Module Summary

Module Summary

- At the end of this module, you should be able to:
- Recognize NetVanta 7000 Key System Applications
- Configure Shared Line Accounts
- Enable Hands Free Auto-Answer
- Configure System Modes
- Conduct Voice Troubleshooting in a NetVanta 7000 Key System Application

Module 6: NetVanta 7000 IP PBX Application

Module Objectives

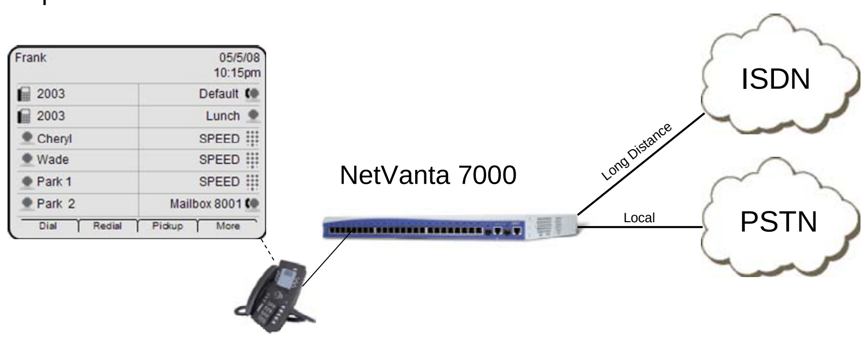
Module Objectives

- Introduce NetVanta 7000 IP PBX Applications
- Configure Voice Trunks – T1-RBS/ISDN PRI
- Create and Configure a Multi-level Auto Attendant
- Create and Configure Dial by Name Directories
- Configure Busy Lamp Field/ Public Park Zones
- Log Calls – Station Message Detail Recording (SMDR)
- Conduct Voice Troubleshooting in a NetVanta 7000 IP PBX Application

NetVanta 7000 - IP PBX Application

NetVanta 7000 IP PBX Application

- What is a PBX?
  - Private Branch eXchange
  - Designed for larger businesses
  - Offer more features/functionality than key systems
  - Outside lines are selected dynamically based on dialed phone number

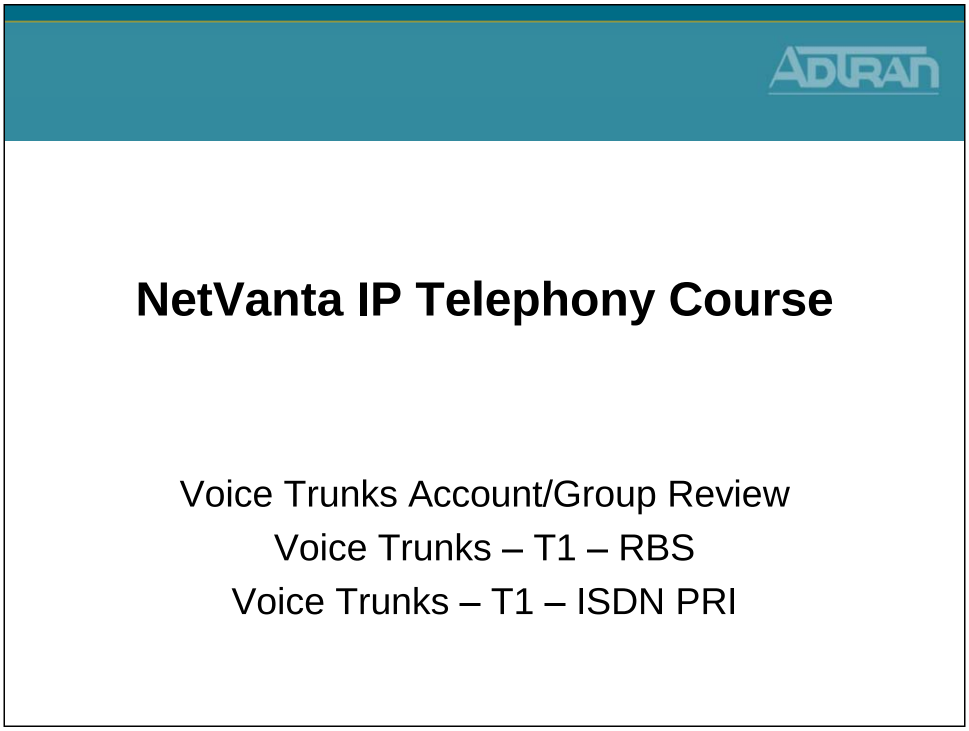


NetVanta 7000 - IP PBX Application

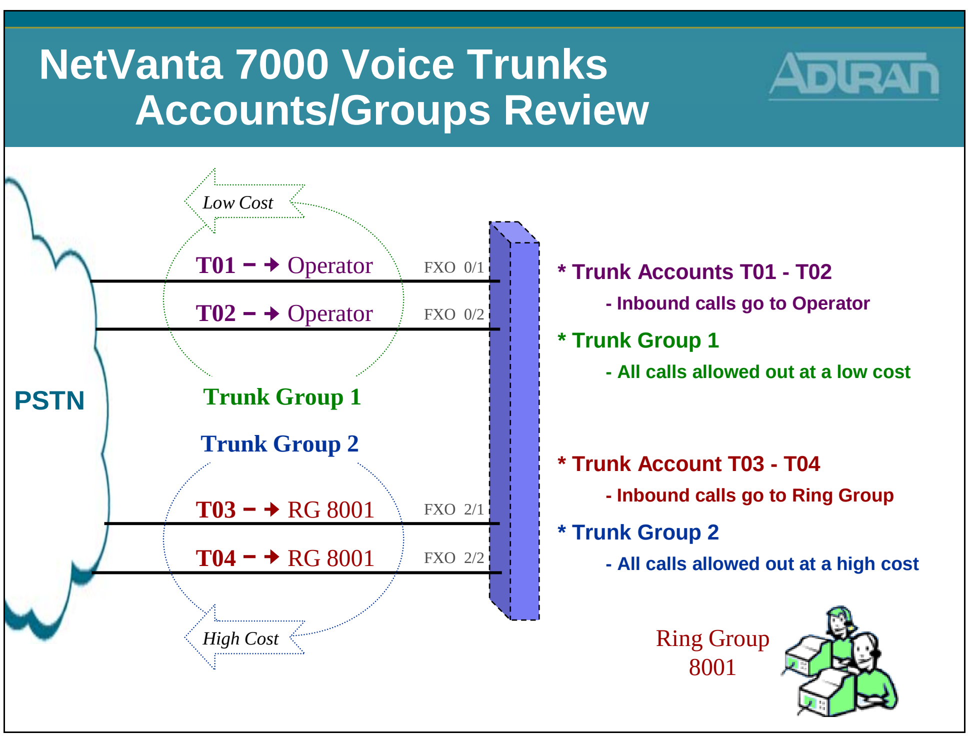
NetVanta 7000 IP PBX Application

- Configuration of the following IP PBX Application features are introduced in this section:
  - Voice Trunks – T1-RBS/ISDN PRI
  - Multi-level Auto Attendant
  - Dial by Name Directories
  - Busy Lamp Field/ Public Park Zones
  - Logging Calls - SMDR

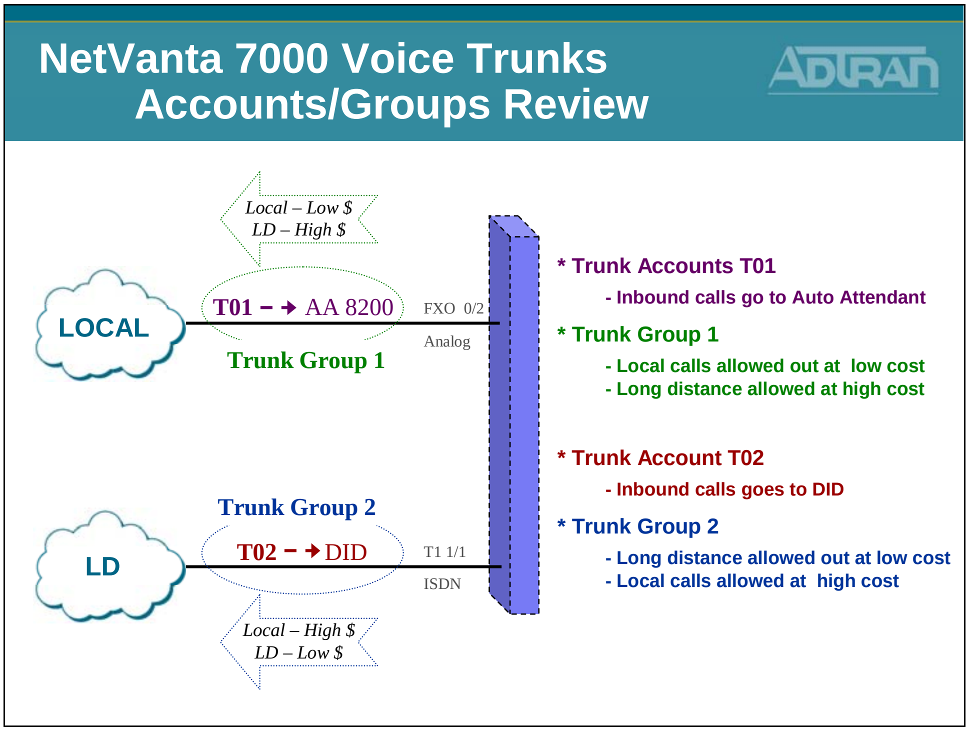
Voice Trunks



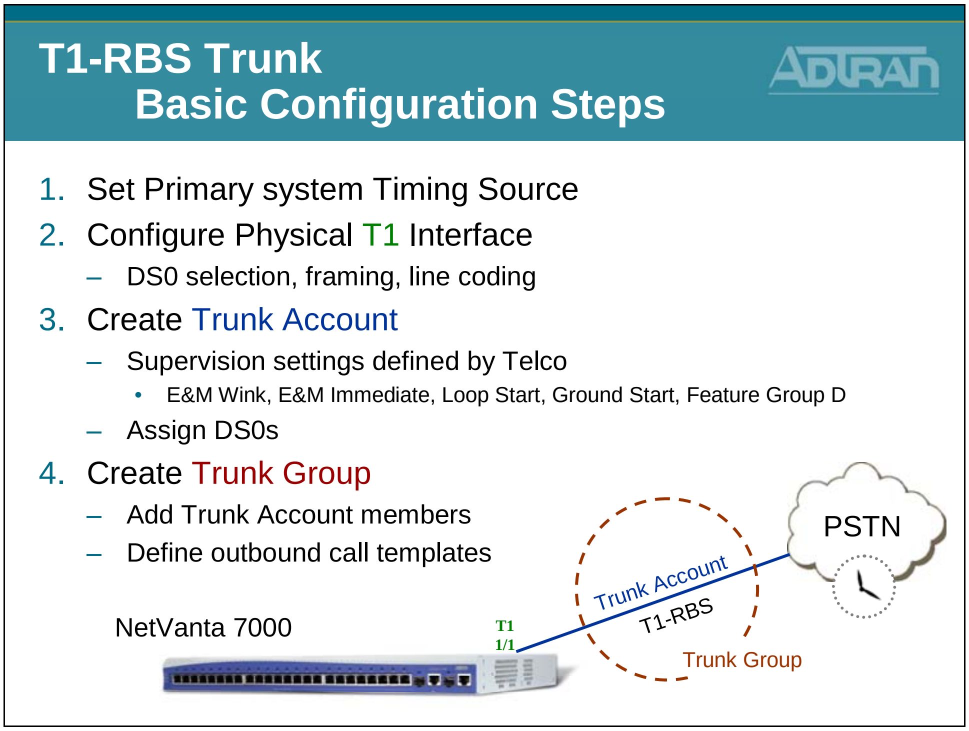
Voice Trunks - Accounts/Groups Review



Voice Trunks - Accounts/Groups Review



T1-RBS Trunk - Basic Configuration Steps



The term T1 circuit is commonly used to identify a multiplexed 24 channel, 1.544 Mbps digital data circuit providing communications between two facilities or from a local service provider. T1 refers to the transport of a DS-1 formatted signal onto a copper, fiber or wireless medium for deploying voice, data or video-conferencing services. T1 connections provide up to 24 64 kbps DS0 channels and use the RBS scheme to pass call signaling status information.

Robbed Bit Signaling: The process where the least significant bit in the 6th and 12th frame (of a SF T1) and the 16 & 24th frame (of an ESF T1) is "robbed" for voice A,B,C,and D signaling bits. These signaling bits indicate on/off-hook conditions etc.

The T1-RBS trunk can terminate a line from the provider (Telco) or be a termination point acting as the network to a PBX or key system requiring a T1 circuit.

T1-RBS Trunk Configuration

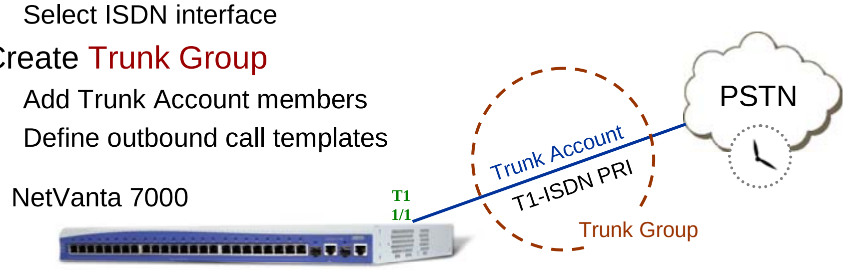
1. Set primary system timing source
2. Configure physical T1 interface and DSO selection
3. Create a T1-RBS Trunk Account
4. Create a Trunk Group

T1-ISDN PRI Trunk - Basic Configuration Steps

T1-ISDN PRI Trunk Basic Configuration Steps

ADTRAN

1. Set Primary system Timing Source
2. Configure Physical T1 Interface
   – DS0 selection, framing, line coding
3. Configure logical ISDN PRI Interface
   – ISDN switch type, emulation mode, digits transferred, etc…
4. Create Trunk Account
   – Select ISDN interface
5. Create Trunk Group
   – Add Trunk Account members
   – Define outbound call templates



The Integrated Digital Service Network (ISDN) Primary Rate Interface (PRI) is a circuit composed of 23 bearer (B) channels and 1 data (D) channel. ISDN PRI is an international standard for digital communications, allowing a full range of enhanced services supporting voice and data. The 23 B channels are used to transmit voice and/or data over an all-digital public switched telephone network. The D channel is used to transmit out-of-band signaling for the B channels that controls dialing numbers and features like call waiting.

The NetVanta 7000 can support the following ISDN PRI switch types: 1. National ISDN 2. AT&T 4ESS, Lucent 5ESS. Nortel DMS-100, and Euro ISDN.

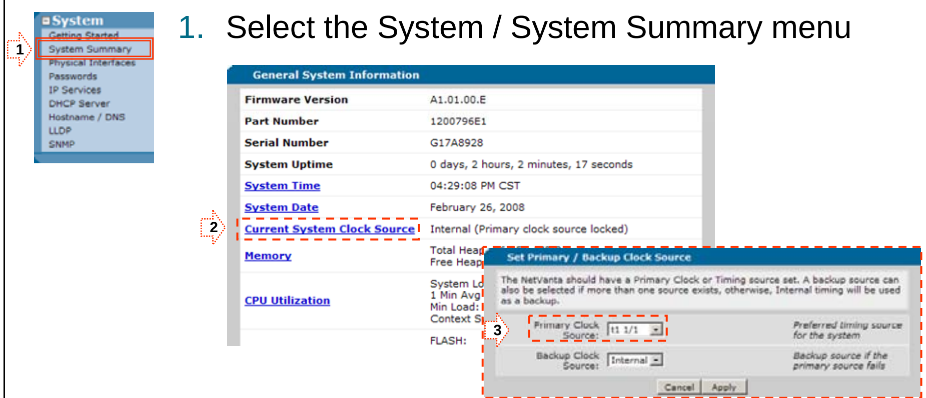
ISDN Trunk Configuration

1. Set primary system timing source
2. Configure physical T1 interface and DSO selection
3. Configure logical PRI interface
4. Create an ISDN Trunk Account
5. Create an ISDN Trunk Group

1) Set System Timing

T1 Trunk Configuration
1) Set System Timing

1. Select the System / System Summary menu

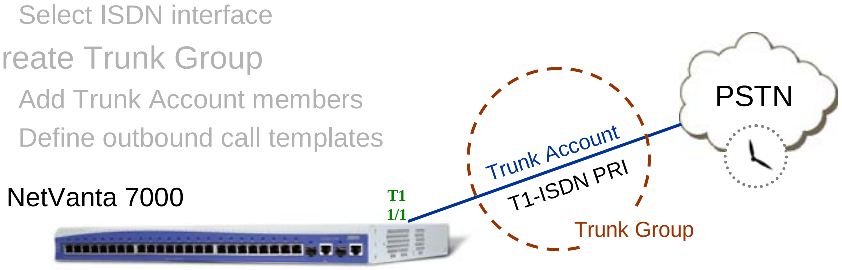


2. Click Current System Clock Source
3. Define the preferred system timing source

T1-ISDN PRI Trunk - Basic Configuration Steps

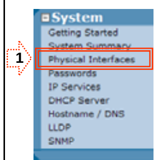
T1-ISDN PRI Trunk
Basic Configuration Steps

1. Set Primary system Timing Source
2. Configure Physical T1 Interface
   – DS0 selection, framing, line coding
3. Configure logical ISDN PRI Interface
   – ISDN switch type, emulation mode, digits transferred, etc…
4. Create Trunk Account
   – Select ISDN interface
5. Create Trunk Group
   – Add Trunk Account members
   – Define outbound call templates

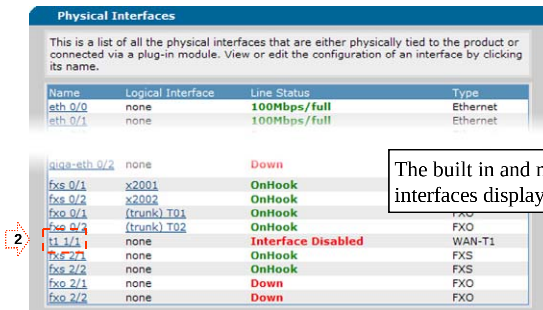


2) Configure Physical T1

T1 Trunk Configuration
2) Configure Physical T1



1. Select the System / Physical Interfaces menu



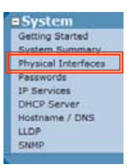
The built in and modular physical interfaces display on this screen

2. Click the T1 interface to be edited

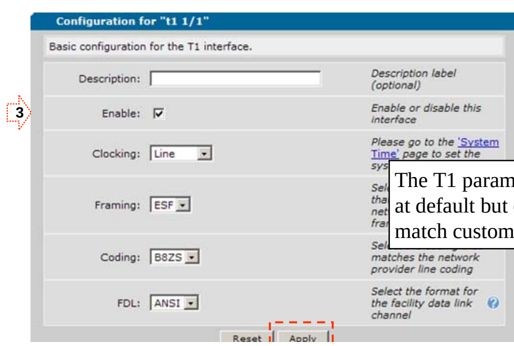
more

2) Configure Physical T1

T1 Trunk Configuration
2) Configure Physical T1



3. Enable the T1 interface



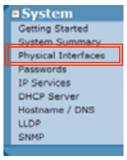
The T1 parameters are usually left at default but can be changed to match customers network

– You must click the Apply button before continuing to the next step (DS0 configuration)

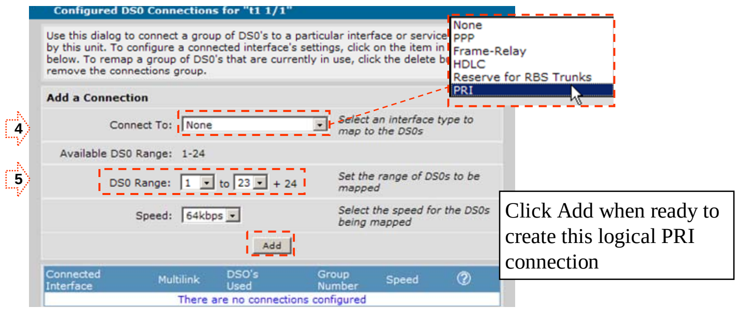
more

2) Configure Physical T1

T1-PRI Trunk Configuration
2) Configure Physical T1



4. Add a PRI Connection

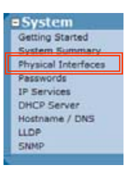


5. Define the DS0 Range for this connection

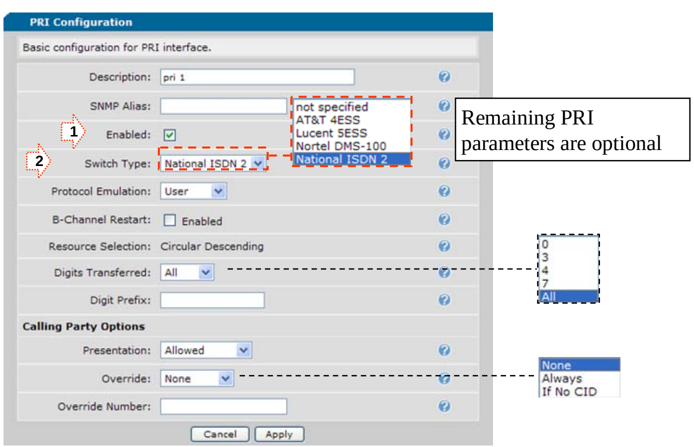
more

3) Configure PRI Interface

T1-PRI Trunk Configuration
3) Configure PRI Interface



1. Enable the Logical PRI Connection



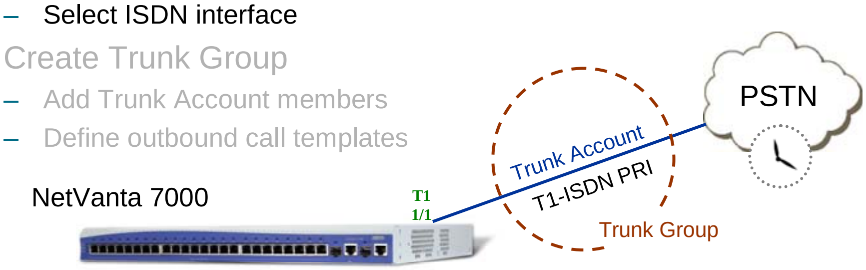
2. Define the ISDN Switch Type

more

T1-ISDN PRI Trunk - Basic Configuration Steps

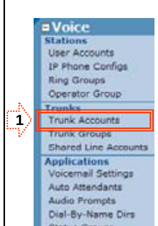
T1-ISDN PRI Trunk Basic Configuration Steps

1. Set Primary system Timing Source
2. Configure Physical T1 Interface
   – DS0 selection, framing, line coding
3. Configure logical ISDN PRI Interface
   – ISDN switch type, emulation mode, digits transferred, etc…
4. Create Trunk Account
   – Select ISDN interface
5. Create Trunk Group
   – Add Trunk Account members
   – Define outbound call templates

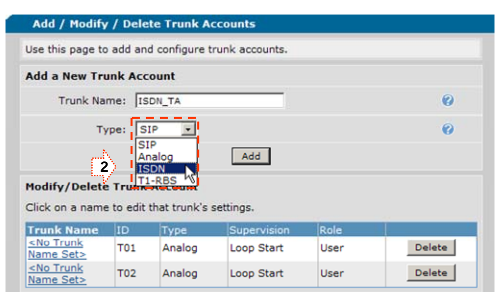


4) Create Trunk Account

T1-PRI Trunk Configuration 4) Create Trunk Account



1. Select the Voice / Trunks / Trunk Accounts menu

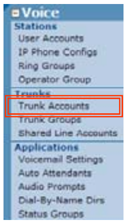


2. Create an ISDN Trunk Account
   – Enter name, set type to ISDN, and then click ADD

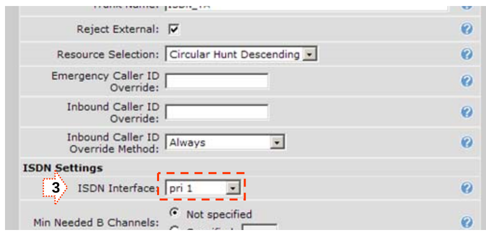
more

4) Create Trunk Account

T1-PRI Trunk Configuration
4) Create Trunk Account



3. Select the ISDN interface "PRI 1"



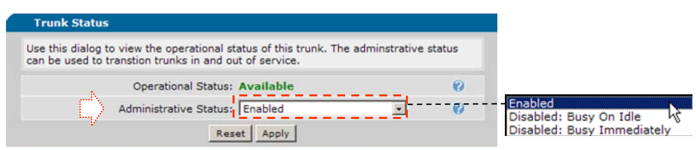
– The logical "PRI 1" interface was created in the "Connect To" step of the T1 configuration

more

4) Create Trunk Account

T1-PRI Trunk Configuration
4) Create Trunk Account

- *Optional*: Administrative Status
  - Enabled by default

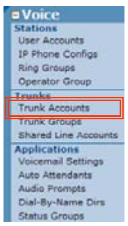


  - Configurable Trunk status Options:
    - **Enabled** - Trunk operates as normal
    - **Disabled: Busy On Idle** - Current calls stay active, but no new calls are accepted
    - **Disabled: Busy Immediately** - All current calls are torn down, and no new calls are accepted

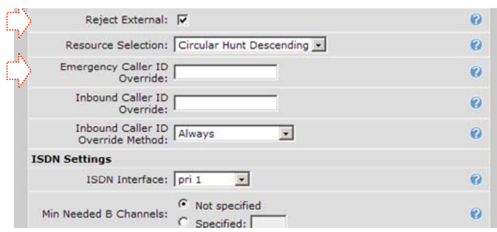
more

4) Create Trunk Account

T1-PRI Trunk Configuration
4) Create Trunk Account



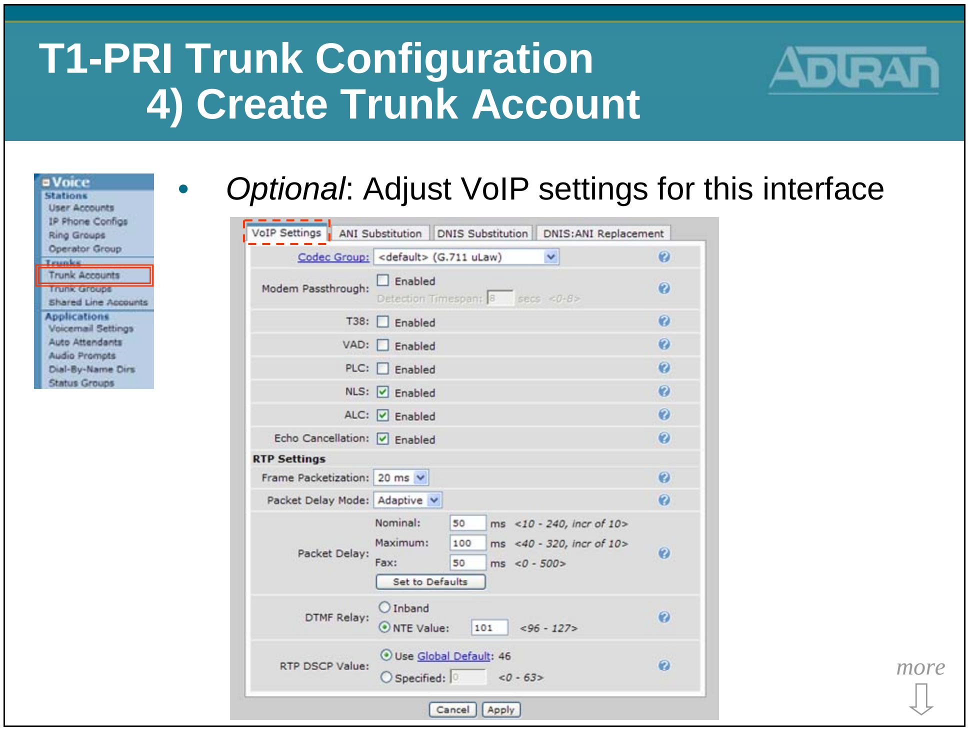
- *Optional*: Reject External
  - Unselect to allow trunk to trunk calls



- *Optional*: Emergency Caller ID Override
  - Specify the calling party number presented on outbound emergency calls

more

ISDN TA – VoIP Settings Tab



The VoIP Settings tab allows you to edit the trunk's voice over IP settings like codec group, VAD, and RTP settings.

Codec Group
Select the codec group to use for this station account.

Modem Passthrough
When Modem Passthrough is enabled and an existing call detects a modem or fax tone, the unit will automatically renegotiate with the far end to be modem-compatible (switch to G.711, all voice improvements turned off, packet delay set to Fax).

T38
When T.38 is enabled and an existing call detects a fax tone, the unit will automatically renegotiate with the far end to be T.38.

VAD
When Voice Activity Detection is enabled, silence is not transmitted over the network, only audible speech. When VAD is enabled, the sound quality is slightly degraded but the connection monopolizes much less bandwidth.

PLC
Enables/disables Packet Loss Concealment. When enabled, the unit will try to reconstruct sound lost from dropped packets.

NLS
Enables/disables the echo canceller's Non-Linear Suppression. When enabled, acoustic echo should be reduced.

ALC
Enables/disables the Automatic Leveling Control. When enabled, reduces received RTP signals to a predefined level.

Echo Cancellation
When enabled, reflected noise is cancelled from the transmitted voice signal. Echo cancellation should normally only be disabled if the voice station is connected to a fax machine or modem.

RTP Settings

Frame Packetization
Select the number of audio samples in ms (1 frame/sample is 10 ms) included in a single RTP packet.

Packet Delay Mode
Configures the operation mode of the jitter buffer for VoIP calls involving this account.

- **Adaptive** - The buffer's delay starts at the nominal delay setting but will increase up to the delay setting if it detects that an intolerable number of packets are being discarded due to jitter. Conversely, the buffer will decrease the amount of delay if it can afford to.
- **Fixed** - The buffer's delay stays at the nominal setting at all times.

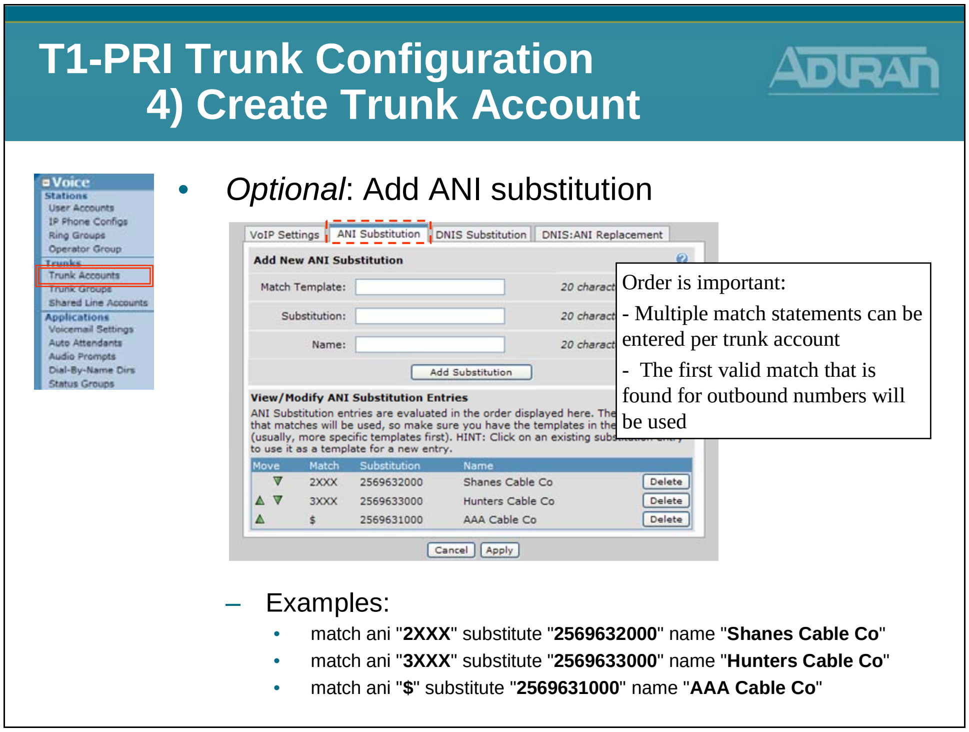
Packet Delay
Configures various packet delay settings for this account.

- **Nominal** - For voice calls, the nominal delay value represents the desired amount of packet delay. In adaptive mode, the buffer may increase this value up to the maximum delay. In fixed mode, the delay is constantly set at this value.
- **Maximum** - For voice calls, the maximum delay value represents the maximum delay to which the adaptive jitter buffer can grow.
- **Fax** - If Modem Passthrough is enabled and modem/fax tones are detected, the packet delay setting will be switched to this value.

DTMF Relay
Select how DTMF tones are to be transmitted over RTP. If out of band (NTE), also enter the NTE value.

RTP DSCP Value
Select the DiffServe code point for this station's RTP packets. Either use the global default (which will change as the global default changes) or specify a value for this station only.

ISDN TA – ANI Substitution Tab



Use ANI Substitution on this trunk to convert out-going Caller ID digits. Additionally, if supported by this device, the name of the calling party may be defined. Example uses are shown below:

ANI Substitution Examples:

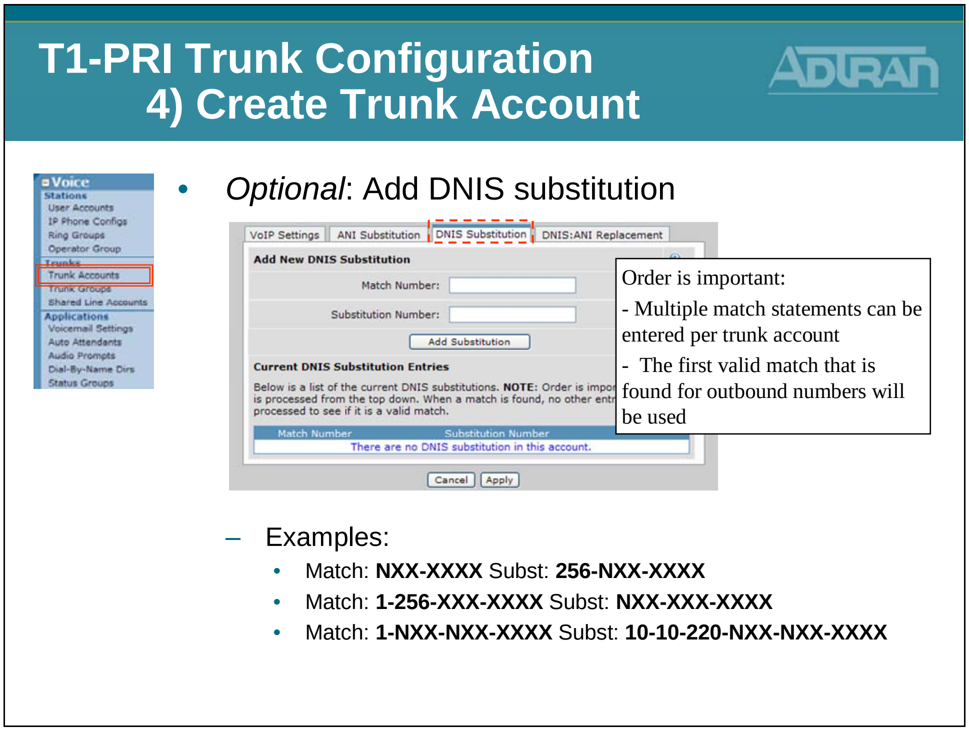
Match: 2XXX Subst: 2569632000 name Shanes Cable Co
- Calls from 2XXX extensions will have an outbound Caller-id number of 256962000 and Caller-ID name Shanes Cable Co

Match: 3XXX Subst: 2569633000 name Hunters Cable Co
- Calls from 3XXX extensions will have an outbound Caller-id number of 256963000 and Caller-ID name Hunters Cable Co

Match: $ Subst: 2569631000 name AAA Cable Co
- Calls from all other extensions will have an outbound Caller-id number of 256961000 and Caller-ID name AAA Cable Co

Multiple ANI substitution entries can be added to each trunk. The first valid match that is found for outbound numbers will be used. Order of input is important.

ISDN TA – DNSI Substitution Tab



Use DNIS Substitution if a dialed number should be replaced with a specific number of your choosing.

Match Number
Specifies the dialed number that you are trying to match

Substitution Number
Specifies the number that will be sent in place of the number that was matched

Wildcard Characters:

0-9	Match exact digit only
X	Match any single digit 0-9
N	Match any single digit 2-9
[ ]	Match any digit in the list. For example [1,4,6] matches 1, 4, and 6 only, while [1-3,5] matches 1 through 3 and 5
$	Match any number, must occur at end of pattern
- ( ) ,	Punctuation characters ignored unless used within [ ]

DNIS Substitution Examples:

1. Match: NXX-XXXX Subst: 256-NXX-XXXX
   Format a call for 10 digit dialing

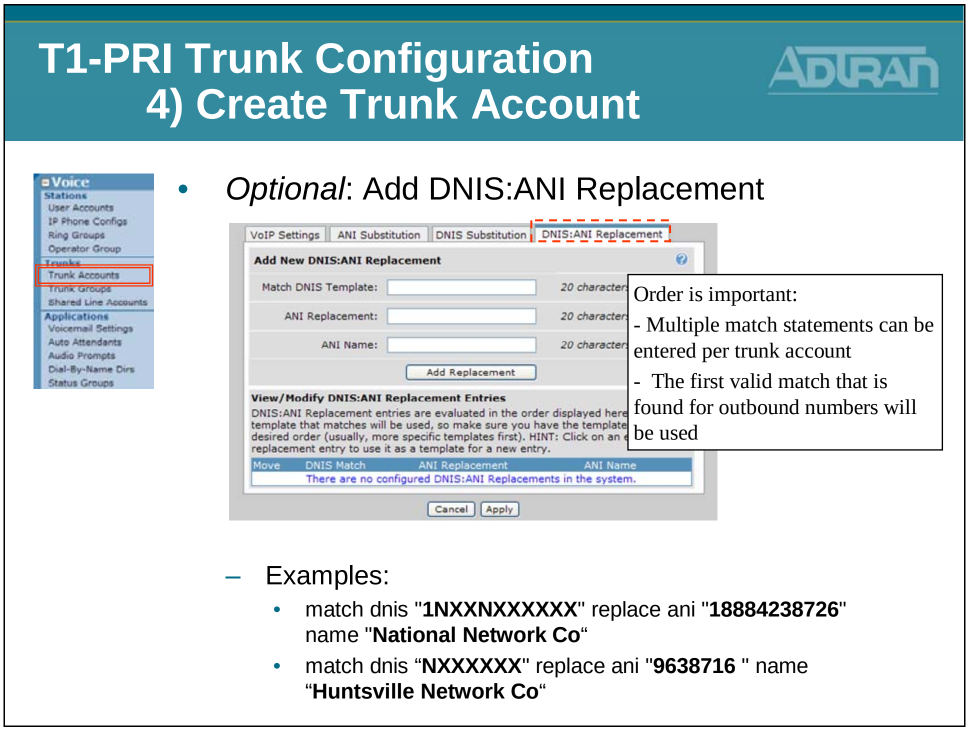
2. Match: 1-NXX-XXX-XXXX Subst: NXX-XXX-XXXX
   Format LD call for 10 digit dialing

3. Match: 1-NXX-NXX-XXXX Subst: 10-10-220-NXX-NXX-XXXX
   Insert a LD call Pick code for a particular carrier

4. Match: 411 Subst: 256-555-1212
   Redirect 411 information calls

Multiple DNIS substitution entries can be added to each trunk. The first valid match that is found for outbound numbers will be used. Order of input is important.

ISDN TA – DNIS:ANI Replacement Tab



Use DNIS:ANI Replacement on this trunk to convert out-going Caller ID digits (ANI) based on the digits dialed(DNIS). Additionally, if supported by this device, the name of the calling party may be defined. Example uses are shown below:

<u>DNIS:ANI Replacement Examples:</u>

match dnis "1NXXNXXXXXX" replace ani "18884238726" name "National Network Co“
- If a long distance number is dialed, set ANI digits to an 888 number

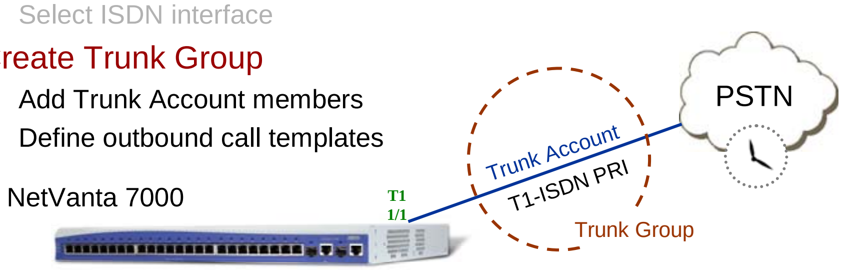
match dnis “NXXXXXX" replace ani "9638716 " name “Huntsville Network Co“
- If a local number is dialed, set ANI digits to a local number

Multiple DNIS:ANI replacement entries can be added to each trunk. The first valid match that is found for outbound numbers will be used. Order of input is important.

T1-ISDN PRI Trunk - Basic Configuration Steps

T1-ISDN PRI Trunk Basic Configuration Steps

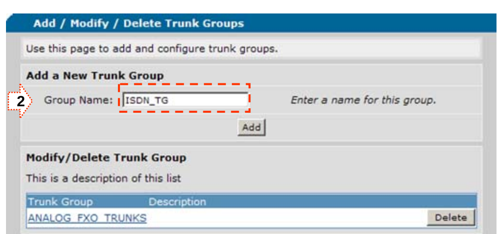
1. Set Primary system Timing Source
2. Configure Physical T1 Interface
   - DS0 selection, framing, line coding
3. Configure logical ISDN PRI Interface
   - ISDN switch type, emulation mode, digits transferred, etc…
4. Create Trunk Account
   - Select ISDN interface
5. Create Trunk Group
   - Add Trunk Account members
   - Define outbound call templates



5) Create Trunk Group

T1-PRI Trunk Configuration 5) Create Trunk Group

1. Select the Voice / Trunks / Trunk Groups menu

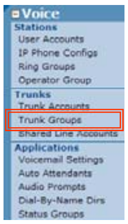


2. Create (or edit) a Trunk Group
   - Enter new Group Name then click Add

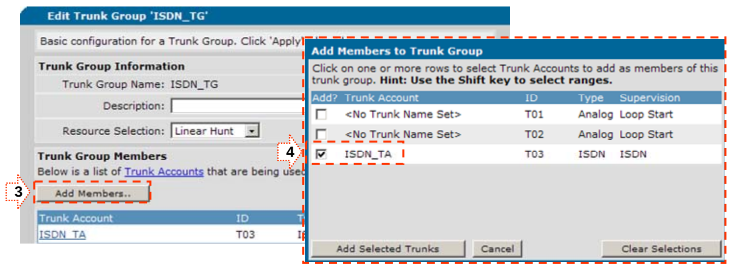
more

5) Create Trunk Group

T1-PRI Trunk Configuration
5) Create Trunk Group



3. Click Add Members to add existing Trunk Accounts to this trunk group

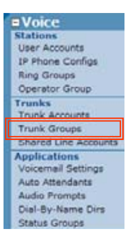


4. Add the ISDN Trunk Account that was just created to this Trunk Group

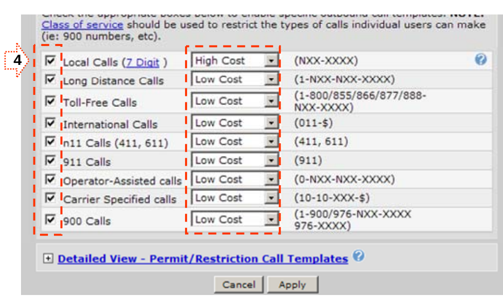
more

5) Create Trunk Group

T1-PRI Trunk Configuration
5) Create Trunk Group



4. Outbound Call Template
   – Define call types allowed out this Trunk Group

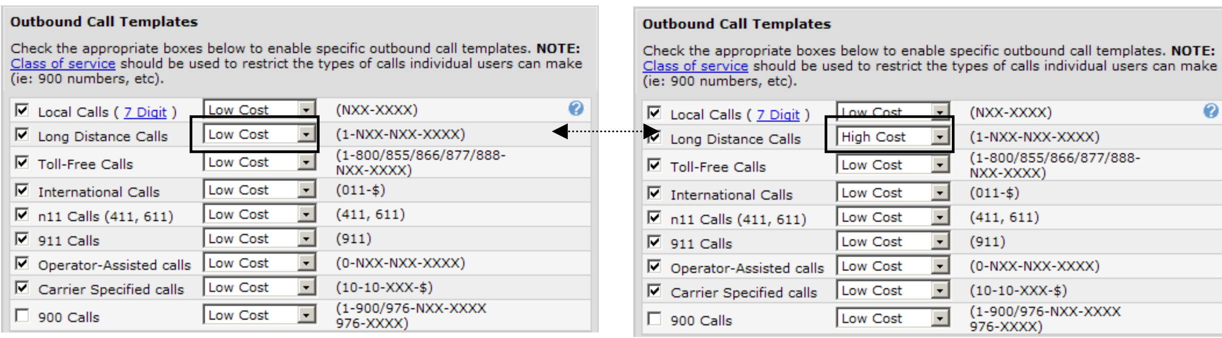


   – *Optional*: Define cost for each type of call
     • Least cost routing

Emergency 911, Redundancy, and Least Cost Routing

E911 calling is a priority as well as Redundancy. The NetVanta 7000 addresses both of these issues under Trunk Accounts. For example, an application with multiple analog trunks will enable E911 dialing on every trunk. No single trunk failure will prohibit E911 access.

Additionally, each of these Trunk Accounts may be placed in separate Trunk Groups. This will allow each Outbound Call attribute to be assigned a Cost on every trunk. Long Distance may be less expensive on a particular trunk, so it may be given a lower cost than long distance dialing on the other trunks. This provides Least Cost Routing.



Long Distance calls will go out the trunk on the left first because it has a lower cost. If there are no available channels on it then LD calls will go out the trunk on the right.

Auto Attendant

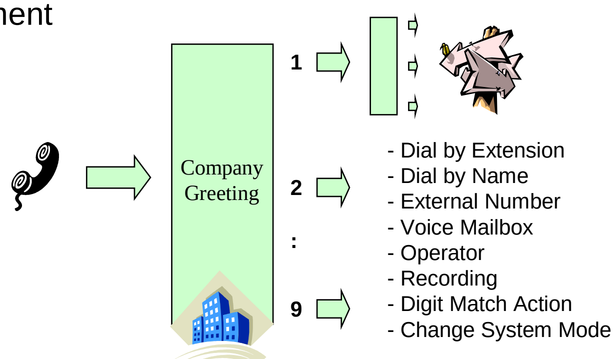
NetVanta IP Telephony Course

Auto Attendant

Multilevel Auto Attendant

Multilevel Auto Attendant

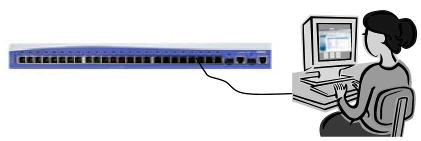
- Multiple Attendants
- Multiple Levels per Attendant
- Several different actions available for each pressed digit
- Prompt Management



Auto Attendant - Basic Configuration Steps

Auto Attendant
Basic Configuration Steps

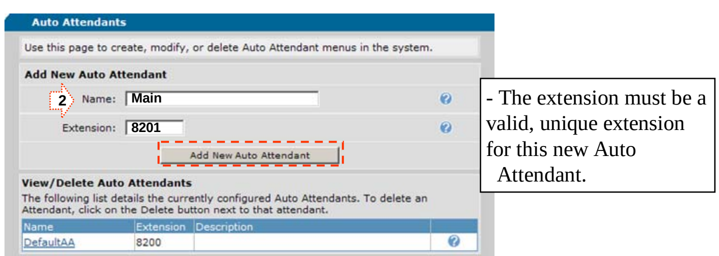
1. Create Auto Attendant Menu
2. Record the audio greeting for Menu prompt
3. Define at least one Digit Action



1) Create AA Menu

Auto Attendant Configuration
1) Create AA Menu

1. Select the Voice / Applications / Auto Attendants menu

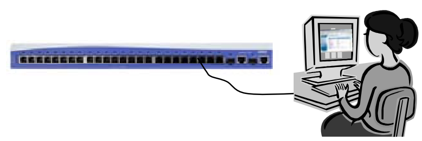


2. Create the Main Auto Attendant Menu
   - Assign name, AA extension, then click Add

Auto Attendant - Basic Configuration Steps

Auto Attendant Basic Configuration Steps

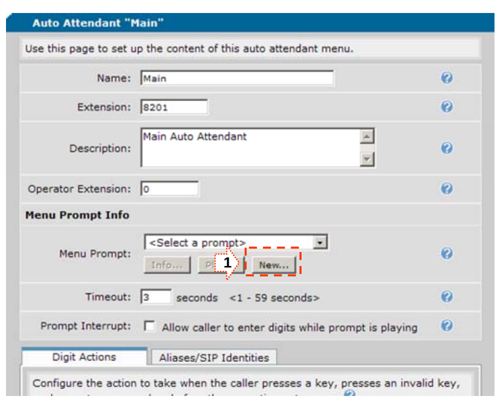
1. Create Auto Attendant Menu
2. Record the audio greeting for Menu prompt
3. Define at least one Digit Action



2) Record Audio Greeting

Auto Attendant Configuration 2) Record Audio Greeting

1. Record the audio greeting for the Menu prompt

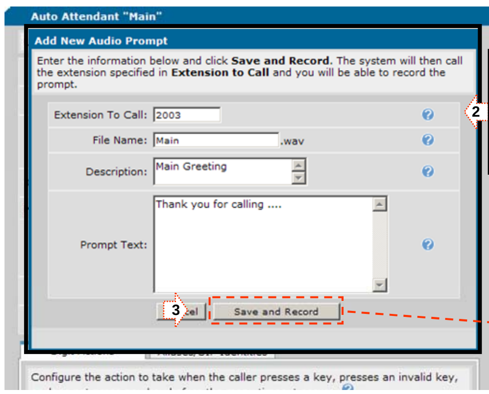


more

2) Record Audio Greeting

Auto Attendant Configuration
2) Record Audio Greeting

1. Record the audio greeting for Menu prompt *(Continued…)*



2 Enter extension to call, File name of WAV file, optional description and prompt text

3

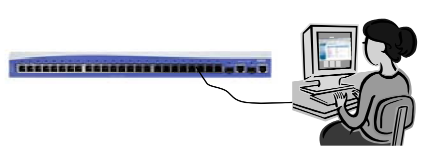
4 *Answer phone*
Record Greeting

x2003

Auto Attendant - Basic Configuration Steps

Auto Attendant
Basic Configuration Steps

1. Create Auto Attendant Menu
2. Record the audio greeting for Menu prompt
3. Define at least one Digit Action

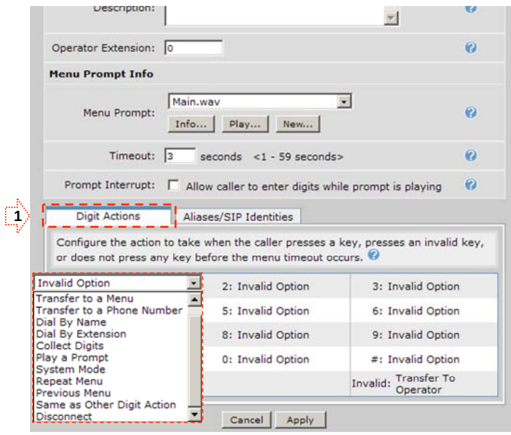


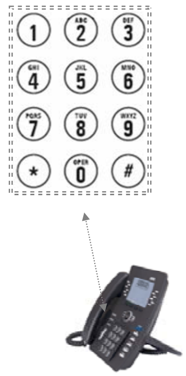
3) Define Digit Actions

Auto Attendant Configuration 3) Define Digit Actions

ADTRAN

1. Define at least one Digit Action



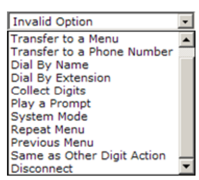


Auto Attendant - Digit Actions

Auto Attendant Digit Actions

ADTRAN

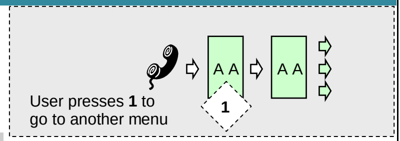
- Different actions available for each pressed digit:
  - Transfer to a Menu
  - Transfer to a Phone Number
  - Dial By Name
  - Dial By Extension
  - Collect Digits
  - Play a Prompt
  - System Mode
  - Repeat Menu
  - Previous Menu
  - Same as Other Digit Action
  - Disconnect

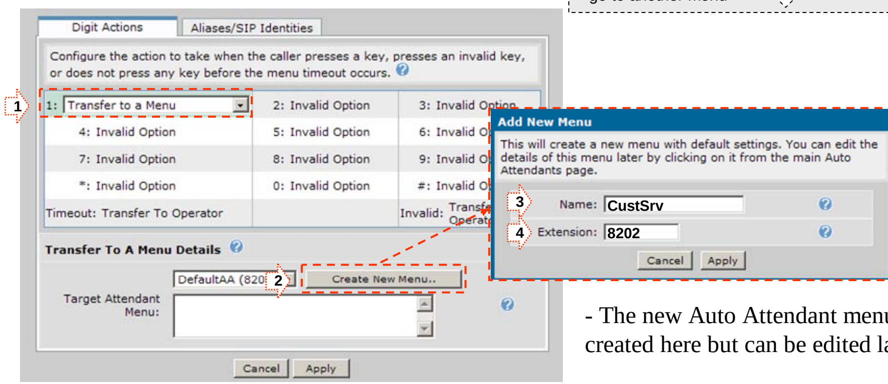


AA Digit Actions - Transfer to a Menu

Auto Attendant Digit Actions
Transfer to a Menu

- Control of the call is passed to a different Auto Attendant



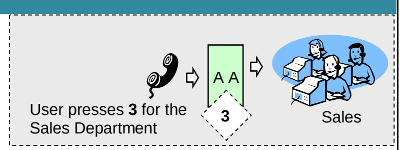


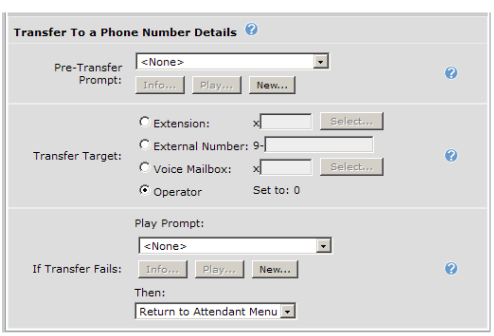
- The new Auto Attendant menu is created here but can be edited later

AA Digit Actions - Transfer to a Phone

Auto Attendant Digit Actions
Transfer to a Phone Number

- The caller is transferred to a specified number or voicemail box



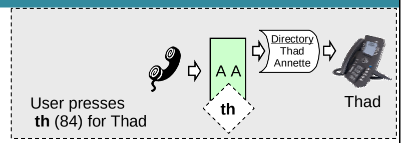


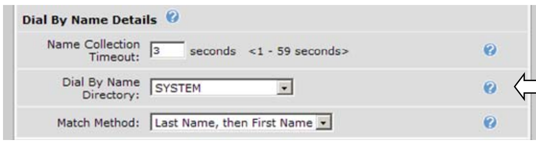
<u>Valid Transfer Targets</u>
- Extension
- External Number
- Voice Mailbox
- Operator

AA Digit Actions - Dial By Name

Auto Attendant Digit Actions Dial By Name

- Matches the caller's input against a defined set of names





Select Existing Dial By Name Directory

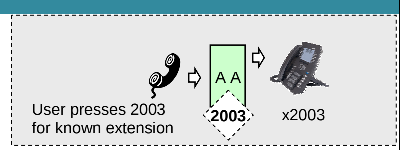
Match Methods
- Last Name, then First Name
- First Name, then Last Name
- Either method

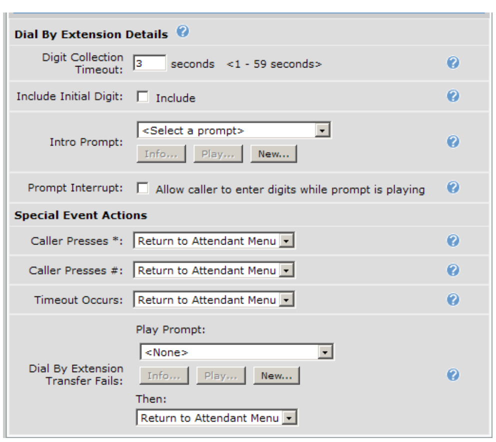
- The default **SYSTEM** dial by name directory contains all users that are included in the System directory

AA Digit Actions - Dial By Extension

Auto Attendant Digit Actions Dial By Extension

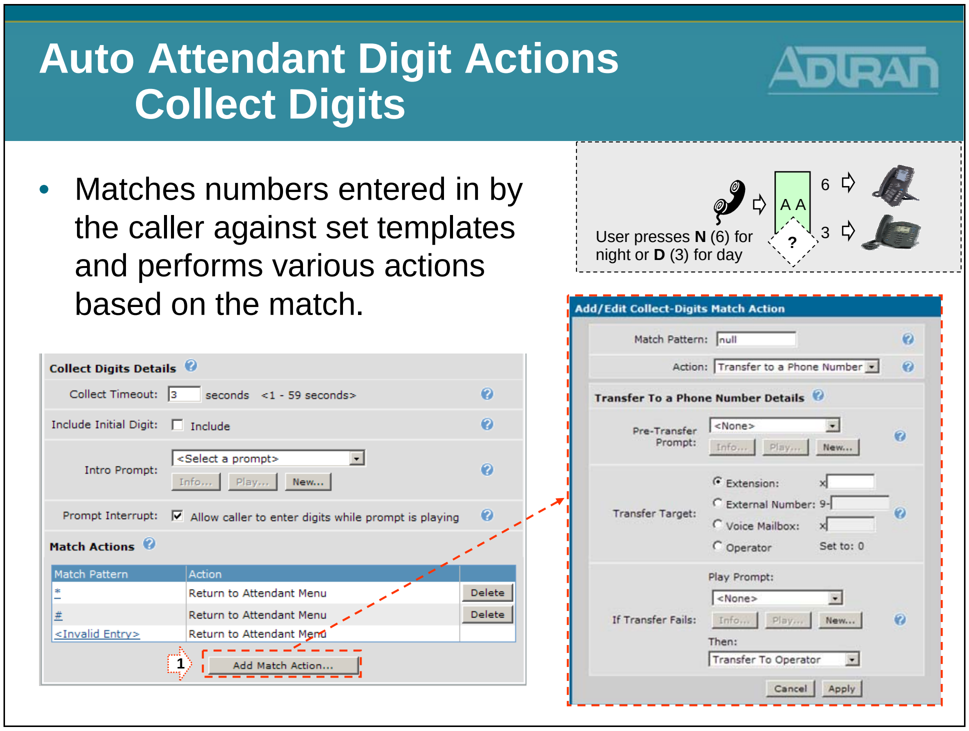
- Transfers the call to an extension entered in by the caller



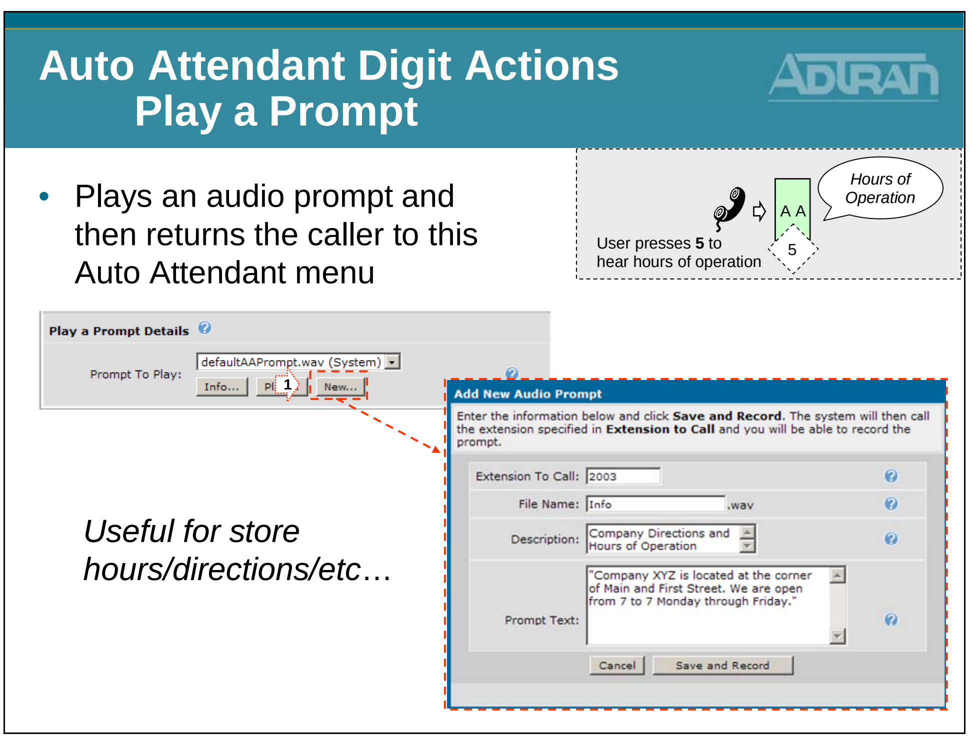


"If you know your party's extension..."

AA Digit Actions - Collect Digits



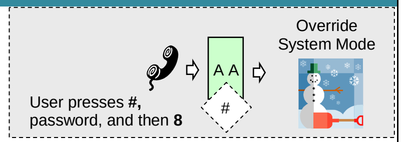
AA Digit Actions - Play a Prompt

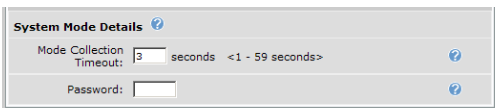


AA Digit Actions - System Mode

Auto Attendant Digit Actions System Mode

- The System Mode action provides a way to change the current system mode of the unit





A voice user who presses the digit action for System Mode will hear the following:

PASSWORD?

The Current System Mode is Default

To Transition to Night Mode Press 2

To Transition to Lunch Mode Press 3

To Transition to Weekend Mode Press 4

To Transition to Custom1 Mode Press 5

To Transition to Custom2 Mode Press 6

To Transition to Custom3 Mode Press 7

To Transition to Override Mode Press 8

To Cancel Press *

-The password assigned above must be entered before the caller can change the current system mode

AA Digit Actions - Other Digit Actions

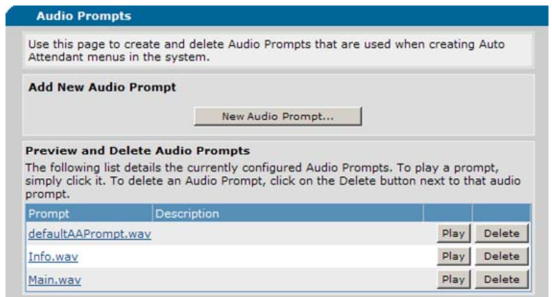
Auto Attendant Digit Actions Other Digit Actions

- Repeat Menu
  - Starts this Auto Attendant's menu prompt over and waits for caller input
- Previous Menu
  - Returns control of the call to the previous Auto Attendant
  - Previous Menu works in conjunction with Transfer To a Menu to allow the caller to navigate through the various Auto Attendant menus
- Same as Other Digit Action
  - Sets this event's action to be exactly the same as another event's action
- Disconnect
  - Terminates the call after optionally playing a prompt

Auto Attendant - Prompt Management

Auto Attendant Prompt Management

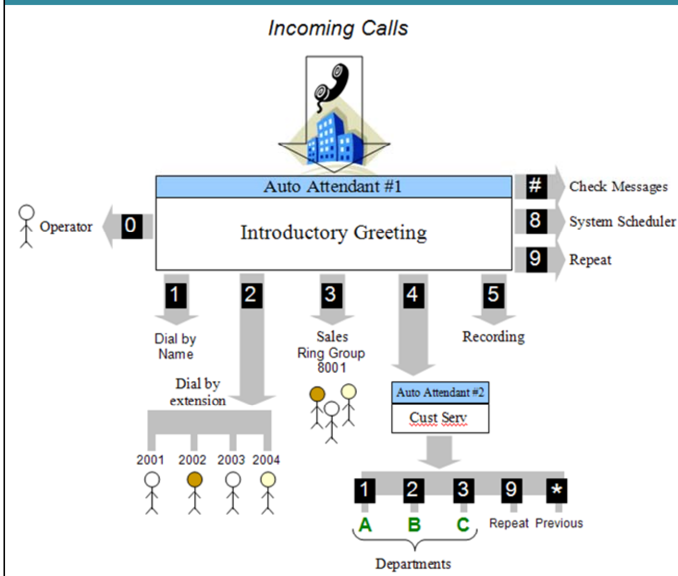
- Prompts can easily be recorded and played via the Web GUI



- Prompts can be recorded on the fly while creating and editing an attendant menu
- The Audio Prompts menu allows for creation, viewing, editing of all known prompts in the system

Auto Attendant Example

Auto Attendant Example



Introductory Greeting

"Thank you for calling Company XYZ. If you know the extension number of the party you are trying to reach, please dial it at any time; if the number is not known, press'1' to dial by name; if you are calling for the Sales Department press '3'; for Customer Service press '4'; for company hours of operation and location information press '5'; To repeat the selections, press '9'; Press '0' or stay on line for the operator."

Dial-by-Name Directory

NetVanta IP Telephony Course

Dial-by-Name Directory

Dial-by-Name Directory

Dial-by-Name Directory

- A default System directory that adds/removes users automatically
- Admin-definable directories containing customizable set of users/ring groups
- Users can be added both with the system name as well as aliases to handle different spellings or nicknames
- Name Searching
  - Can search by first name, last name, or either

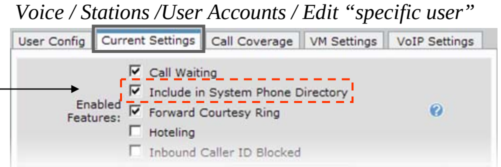
Dial-by-Name Directory - Default SYSTEM Directory

Dial-by-Name Directory
Default SYSTEM Directory

- The SYSTEM Directory contains all users that have the "Include in System Phone Directory" option enabled

Voice / Stations /User Accounts / Edit "specific user"

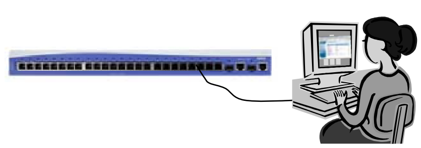
This option is configured in the Current Settings tab of the specific voice user



Dial-by-Name Directory - Basic Configuration

Dial-by-Name Directory
Basic Configuration Steps

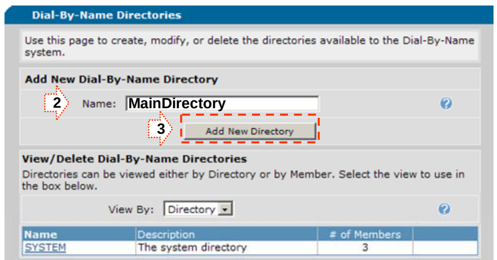
1. Create a new Dial-By-Name Directory
2. Add Users to the Directory
3. Optional – Add Alias to Directory
4. Configure Directory as a Digit Action in an Auto Attendant



1) Create new Dial-By-Name Directory

Dial-by-Name Directory Configuration
1) Create new Dial-By-Name Dir.

1. Select the Voice / Applications / Dial-By-Name Dirs menu



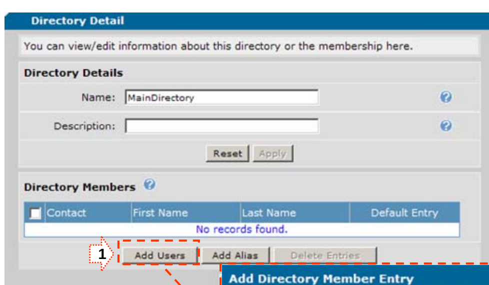
2. Type name and then click Add New Directory

more

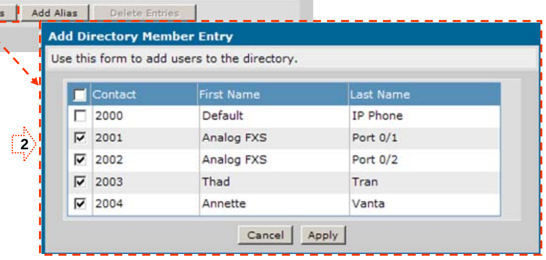
2) Add Users to the Directory

Dial-by-Name Directory Configuration
2) Add Users to the Directory

1. Click Add Users



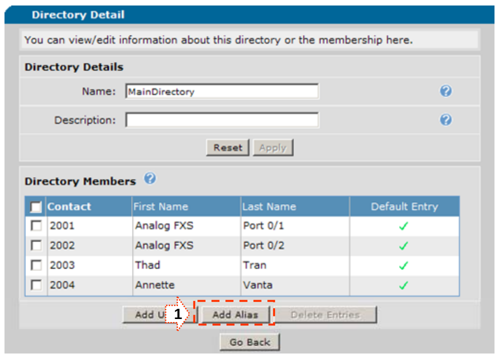
2. Select from the list of available voice users



3) Optional – Add Alias to Directory

Dial-by-Name Directory Configuration
3) Optional – Add Alias to Dir.

1. Click Add Alias



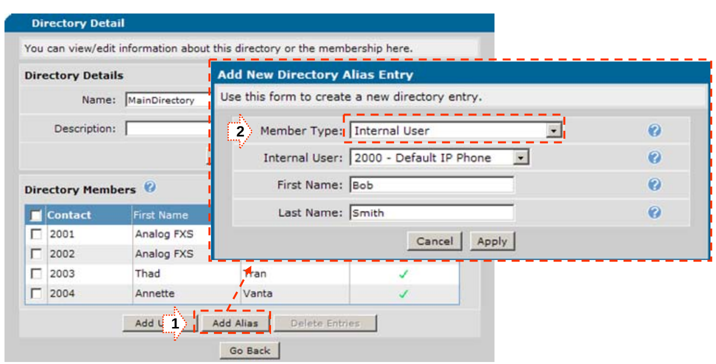
- An Alias can be added for an internal system user or a phone number such as a ring group

more

3) Optional – Add Alias to Directory

Dial-by-Name Directory Configuration
3) Optional – Add Alias to Dir.

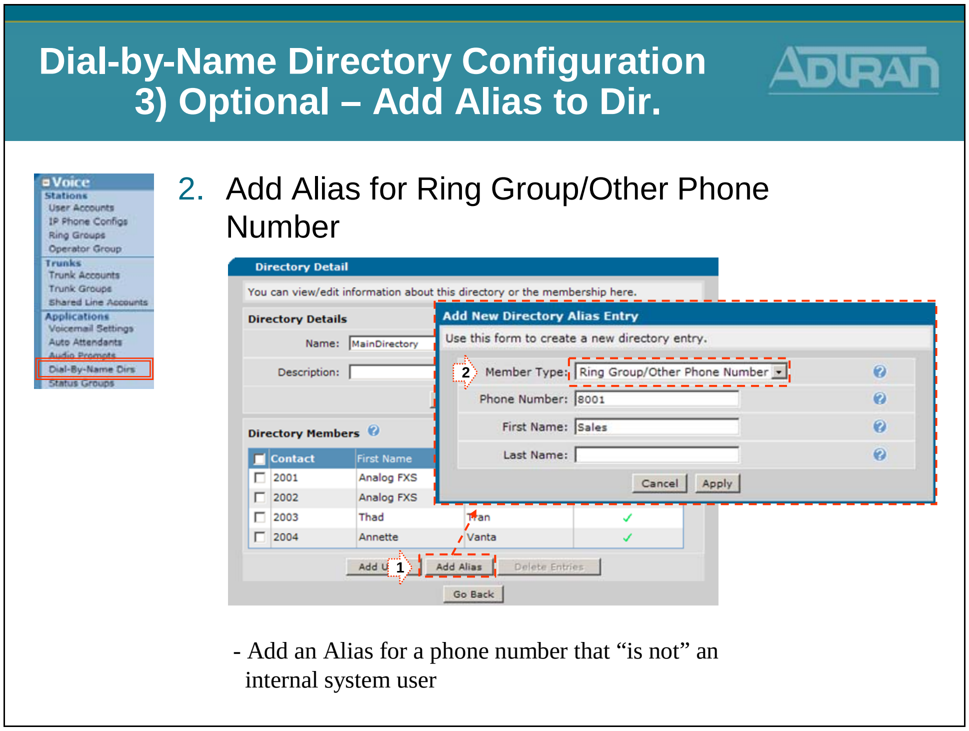
2. Add Alias for an Internal User



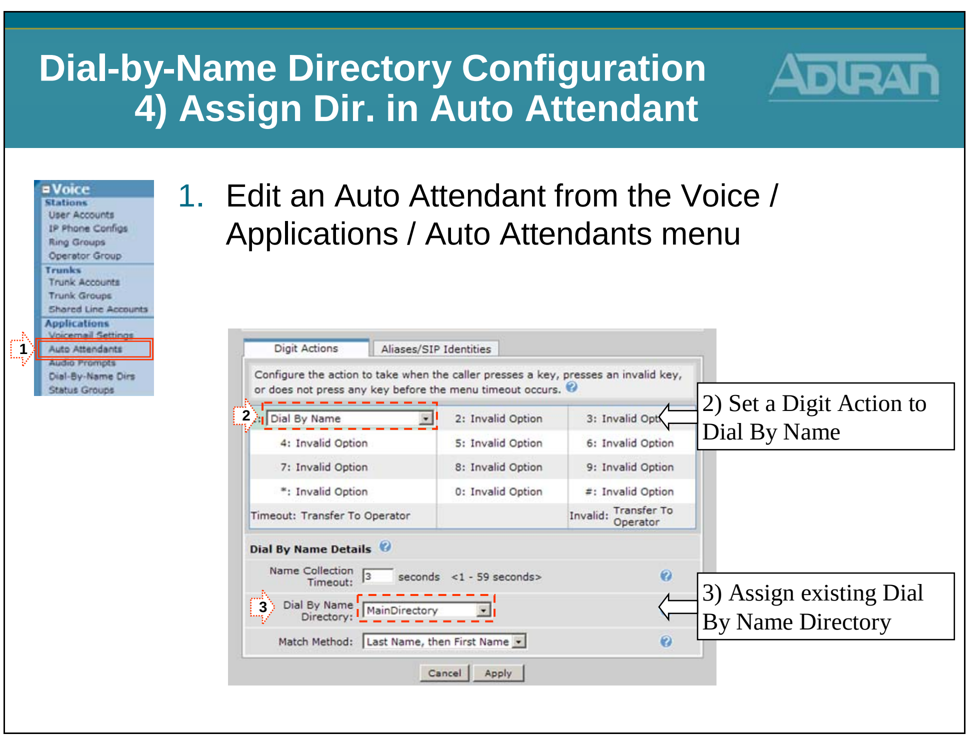
- Add an alternate name for an internal system user

more

3) Optional – Add Alias to Directory



4) Assign Directory in Auto Attendant



Busy Lamp Field/Public Park Zones

NetVanta IP Telephony Course

Busy Lamp Field/ Public Park Zones

Busy Lamp Field

Busy Lamp Field

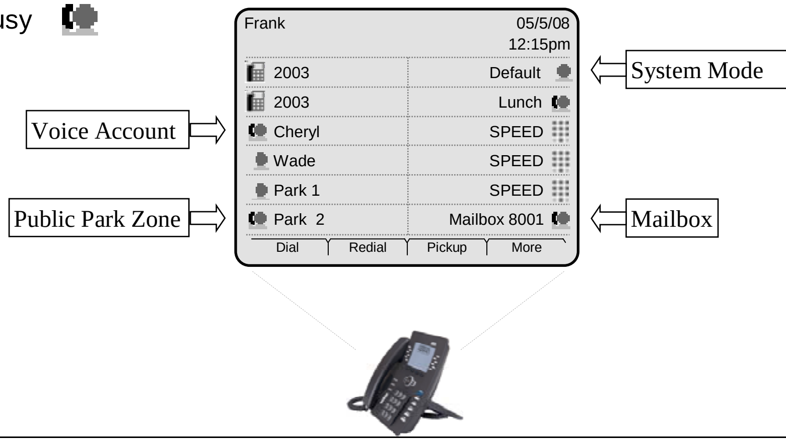
- Monitor busy or idle status
  - Another phone
  - Public park zone
  - System mode
  - Mailbox
- DSS function also supported on the same phone
- Supported with the ADTRAN 700 Series and SoundPoint IP 601/650 only
  - Other SIP and analog phones can be monitored using BLF

DSS: Direct Station Select
BLF: Busy Lamp Field

Busy Lamp Field

Busy Lamp Field

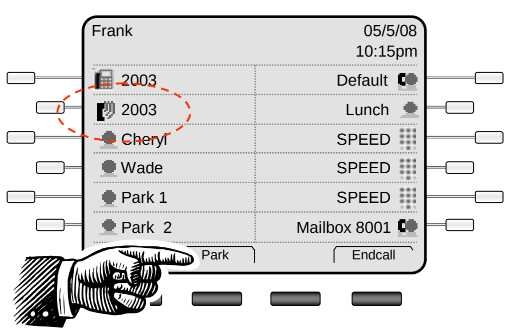
- Status Indicator
  - Idle
  - Busy



Public Park Zones - Parking Active Call

Public Park Zones Parking Active Call

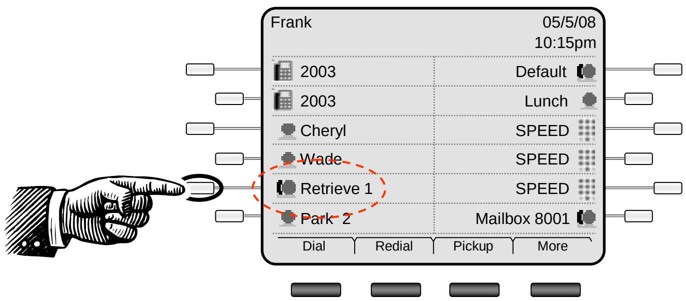
- Call is answered and parked in “Park 1” zone



Public Park Zones - Parking Active Call

Public Park Zones
Parking Active Call

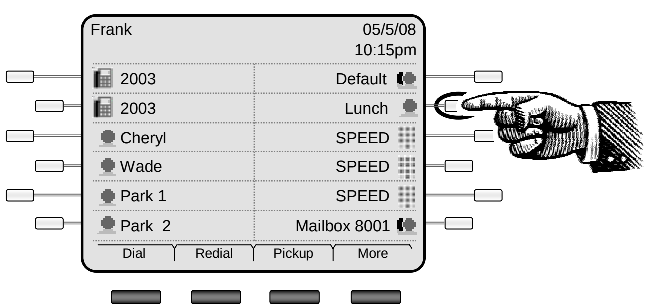
- Call is available for Retrieval



Busy Lamp Field - System Scheduler

Busy Lamp Field
System Scheduler

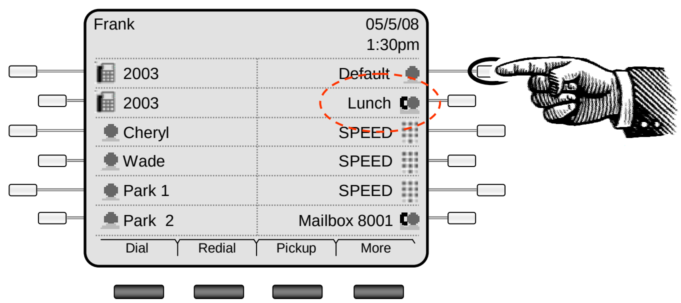
- Early Lunch – the NetVanta 7000 is manually changed from the Default System Mode to the Lunch System Mode



Busy Lamp Field - System Scheduler

Busy Lamp Field
System Scheduler

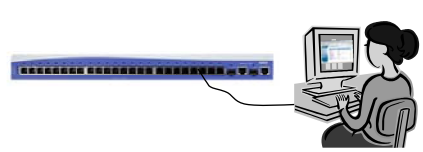
- The NetVanta 7000 will stay in the Lunch System Mode until manually changed back to the Default or other System Modes



Busy Lamp Field - Basic Configuration Steps

Busy Lamp Field/Public Park Zones
Basic Configuration Steps

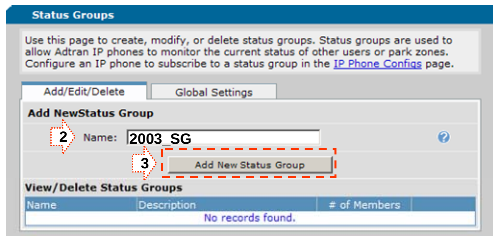
1. Create a Status Group
2. Add members to the Status Group
   – Members can be users, park zones, system modes, or a mailbox to be monitored
3. Configure SIP phone to subscribe to Status group
   – Subscribe to Status Group in the IP Phone Configs Menu



1) Create Status Group

Busy Lamp Field Configuration 1) Create Status Group

1. Select the Voice / Applications / Status Groups menu



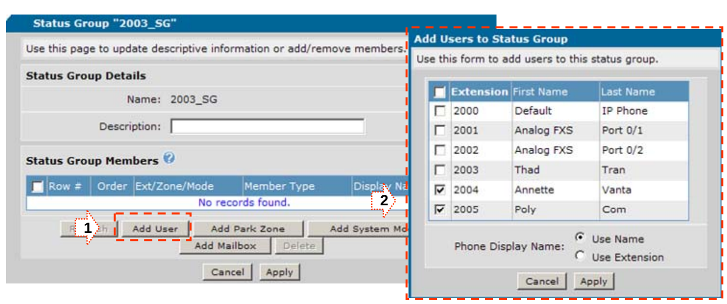
2. Type name for new Status Group
3. Click Add New Status Group

- A Status Group is created to define the voice users, park zones, system modes, or a mailbox to be monitored

2) Add Members to Status Group

Busy Lamp Field Configuration 2) Add Members to Status Group

1. Click Add Users



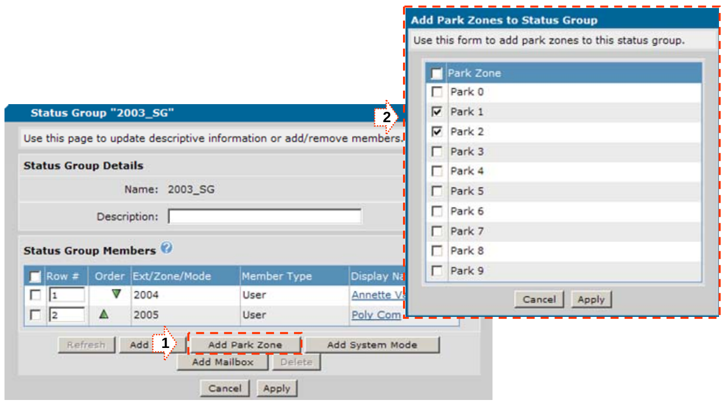
2. Select from existing voice users

- BLF buttons can also be used as a speed dial or direct station select (DSS) for that user
- BLF buttons may also be used when invoking transfers

2) Add Members to Status Group

Busy Lamp Field Configuration
2) Add Members to Status Group

1. Click Add Park Zones

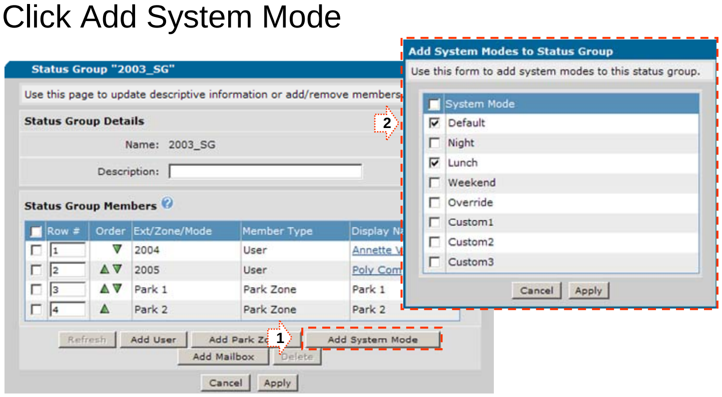


2. Add Parks Zones to be monitored

2) Add Members to Status Group

Busy Lamp Field Configuration
2) Add Members to Status Group

1. Click Add System Mode

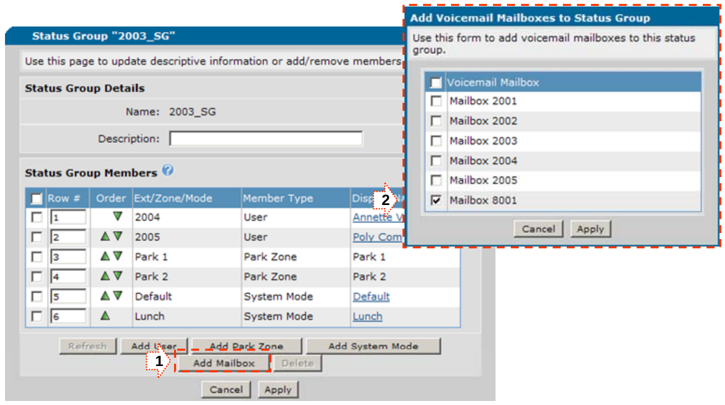


2. Add System Modes to be monitored

2) Add Members to Status Group

Busy Lamp Field Configuration
2) Add Members to Status Group

1. Click Add Mailbox

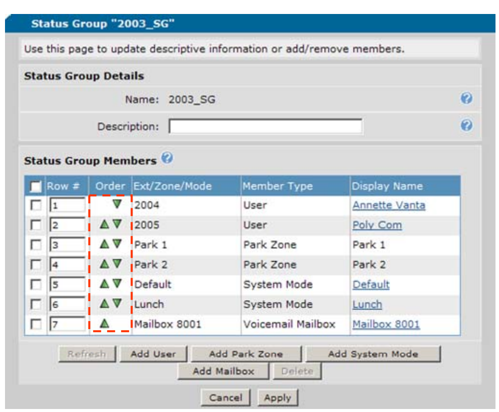


2. Add Mailbox to be monitored

2) Add Members to Status Group

Busy Lamp Field Configuration
2) Add Members to Status Group

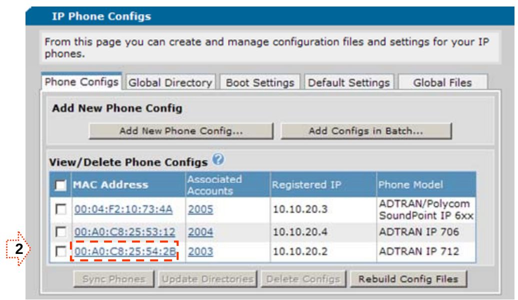
- Optional – Change order of Status Group members



3) Subscribe to Status Group

Busy Lamp Field Configuration
3) Subscribe to Status Group

1. Select the Voice / Stations / IP Phone Configs menu

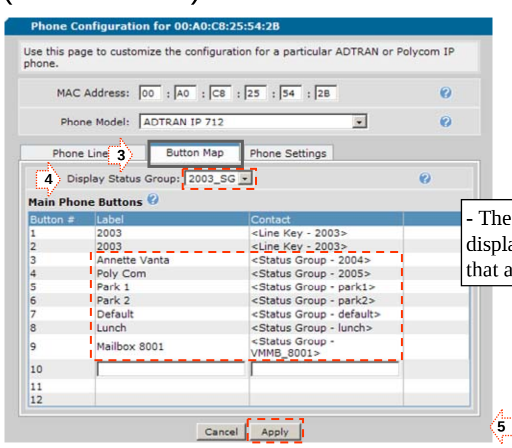


2. Configure phone to subscribe to Status Group
   – Select the MAC Address of phone that will monitor the Status Group

3) Subscribe to Status Group

Busy Lamp Field Configuration
3) Subscribe to Status Group

2. Configure phone to subscribe to Status Group (Continued…)



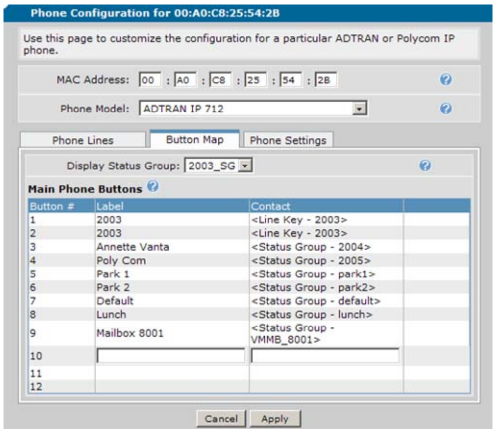
- The Status Group will display below the line keys that are currently configured

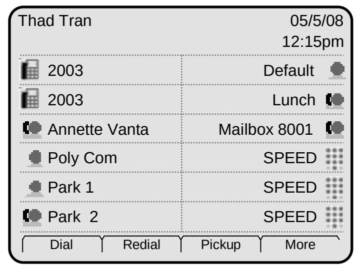
- Click Apply to sync and reboot phone

3) Subscribe to Status Group

Busy Lamp Field Configuration
3) Subscribe to Status Group

- Phone Display after reboot





Logging Calls

NetVanta IP Telephony Course

Logging Calls

Station Message Detail Recoding (SMDR)

Logging Methods

Logging Methods

- Real-Time Event messages are generated by many parts of the NetVanta 7000 and can be displayed real-time in the console or telnet session or logged to the following:
  - Internal event-log
  - External syslog-server.
  - SMTP mail server.
- Can filter which events are logged based on severity

Logging – SMDR

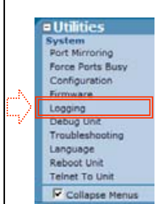
Logging - SMDR

- The NetVanta 7000 supports sending SMDR events to an external logging server
- SMDR events are sent AFTER the call is completed
- SMDR messages contain information about the call including
  - Time initiated
  - Billing code
  - Billable Duration
  - Trunk
- Requires an external host running a syslog server
- Sends messages on UDP port 514

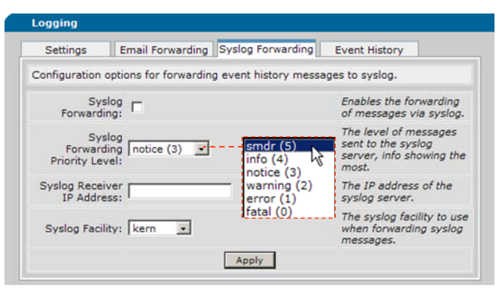
SMDR - Station Messaging Detail Record (Call Detail Records)

Utilities / System - Logging – SMDR

Utilities / System
Logging - SMDR



- Enable SMDR Logging



1. Enable Syslog Forwarding
2. Set Syslog Forwarding Priority Level to SMDR
3. Define IP address of Syslog server
4. Optional – define Syslog facility

Logging – SMDR Events

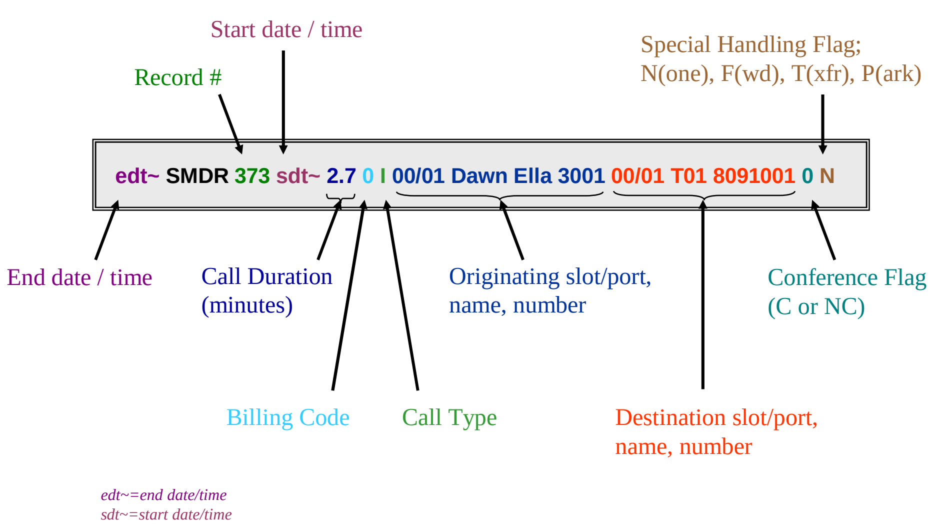
Logging - SMDR Events

- Sample SMDR Log

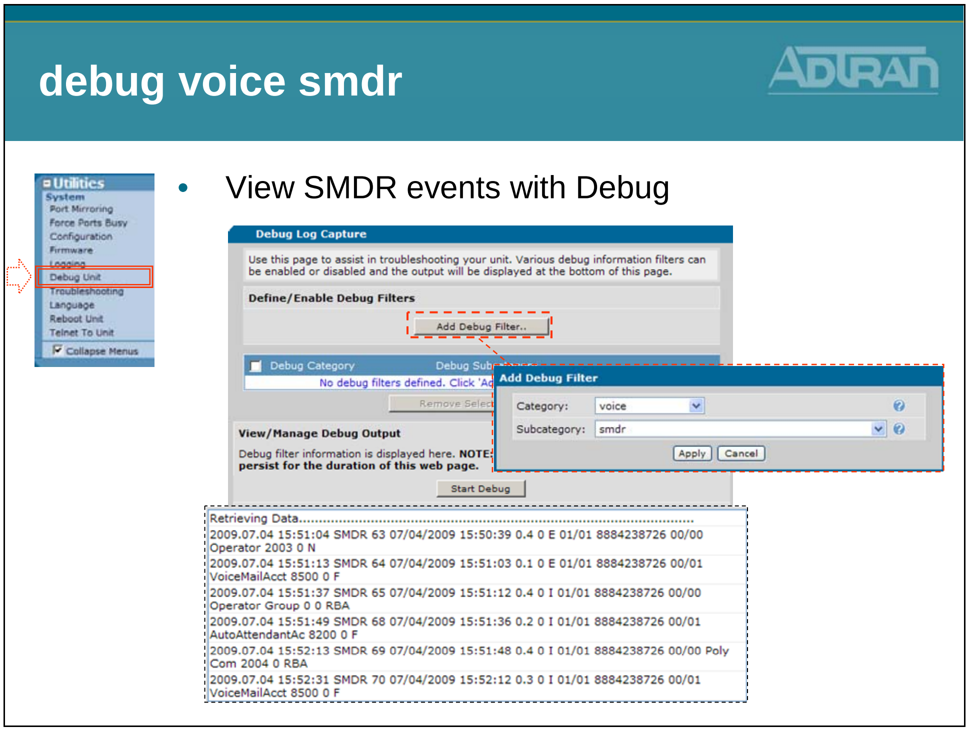
```
2008.04.30 13:52:51 SMDR 369 04/30/2008 13:52:49 0.0 0 I 00/00 Dawn Ella 3001 00/01 Rob Wade 5001 0 N
2008.04.30 13:52:53 SMDR 368 04/30/2008 13:52:51 0.0 0 I 00/00 Dawn Ella 3001 00/01 Bob Sup 2003 0 N
2008.04.30 13:57:01 SMDR 370 04/30/2008 13:53:35 3.4 0 I 00/01 Rob Wade 5001 00/01 T01 8041000 0 N
2008.04.30 13:57:41 SMDR 371 04/30/2008 13:57:27 0.2 0 E 00/01 8081000 00/01 AutoAttendantAc 8200 0 N
2008.04.30 13:59:11 SMDR 372 04/30/2008 13:57:39 1.5 0 I 00/01 8081000 00/01 Dawn Ella 3001 0 RBA
2008.04.30 14:05:28 SMDR 373 04/30/2008 14:02:46 2.7 0 I 00/01 Dawn Ella 3001 00/01 T01 8091001 0 N
2008.04.30 14:05:28 SMDR 374 04/30/2008 14:02:40 2.8 0 I 00/00 Dawn Ella 3001 00/00 Rob Wade 5001 0 N
```


Logging – SMDR Events

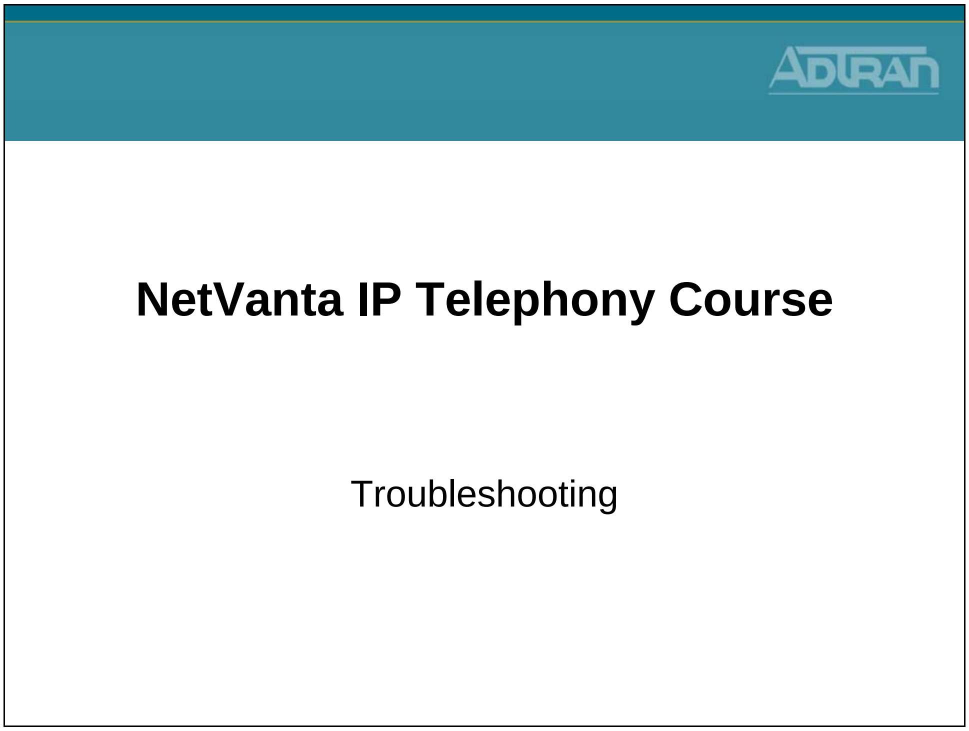
Logging - SMDR Events



debug voice smdr



Troubleshooting



Layer 1 Troubleshooting T1 Alarm Conditions

View T1 Alarms and Errors

Detailed troubleshooting can be accomplished via the Command Line Interface (CLI) via either a console or telnet connection.

The **show interface t1 1/1** command shows the up/down state of the T1 along with the following:

- Alarm state (current/history)
- Framing and coding
- Clock source
- Test mode
- Channel status
- Signal state (A/B bits)
- Performance statistics

show int t1 1/1 – No Alarms

show int t1 1/1

- Display the T1 interface – No Alarms

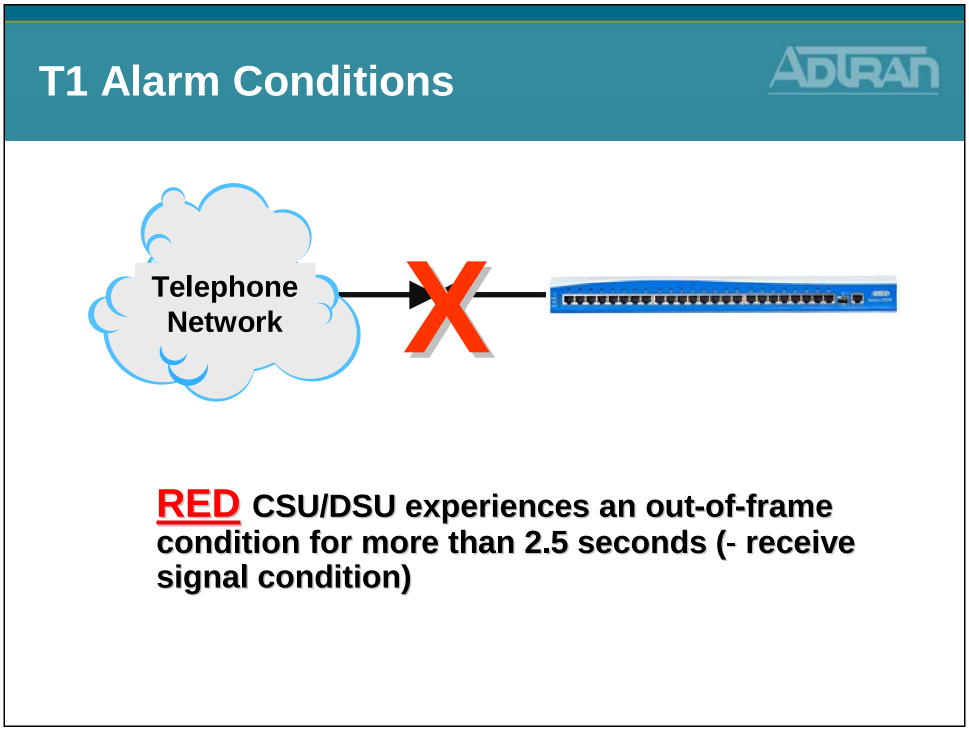
```
NV7000# show int t1 1/1
t1 1/1 is UP
  Receiver has no alarms
  T1 coding is B8ZS, framing is ESF
  Clock source is line, FDL type is ANSI
  Line build-out is 0dB
  No remote loopbacks, No network loopbacks
  Acceptance of remote loopback requests enabled
  Tx Alarm Enable: rai
  Last clearing of counters 01:05:16
    loss of frame  : 0
    loss of signal : 0
    AIS alarm      : 0
    Remote alarm   : 1, last occurred 00:21:23

  DS0 Status: 123456789012345678901234
              -------------------XXXXX
  Status Legend:  '-' = DS0 is not allocated
                  'X' = DS0 is allocated (nailed)
             :
```

* Continues on next slide

show int t1 1/1 (Continued…)

```
Continued…

  Signaling Bit Status: 123456789012345678901234
          RxA:          -------------------11111
          RxB:          -------------------11111

          TxA:          ------------------- 11111
          TxB:          -------------------01111
                        123456789012345678901234

  Line Status: -- No Alarms --

  5 minute input rate 0 bits/sec, 0 packets/sec
  5 minute output rate 0 bits/sec, 0 packets/sec
  Current Performance Statistics:
    0 Errored Seconds, 0 Bursty Errored Seconds
    0 Severely Errored Seconds, 0 Severely Errored Frame Seconds
    0 Unavailable Seconds, 0 Path Code Violations
    0 Line Code Violations, 0 Controlled Slip Seconds
    0 Line Errored Seconds, 0 Degraded Minutes

  TDM group 1, line protocol is not set
    Encapsulation is not set
```

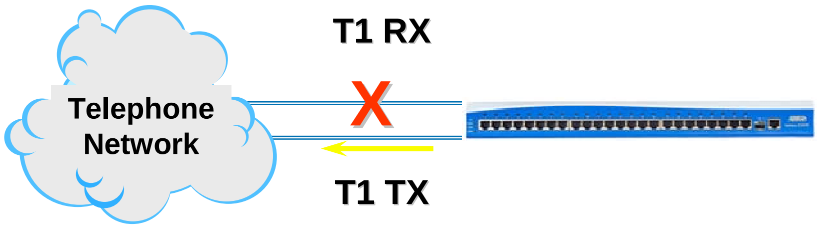
T1 - Red Alarm



Red Alarm is declared when the CSU cannot synchronize on the framing pattern on the network interface. This may be due to excessive errors on the T1 or an incorrect framing pattern. Red Alarm will be declared if an Out of Frame (OOF) condition exists for 2.5 seconds or more. A common cause of Red Alarm is a mismatch on framing configuration (D4 versus ESF) between the telco and the customer's CSU.

T1 - Yellow Alarm

T1 Alarm Conditions



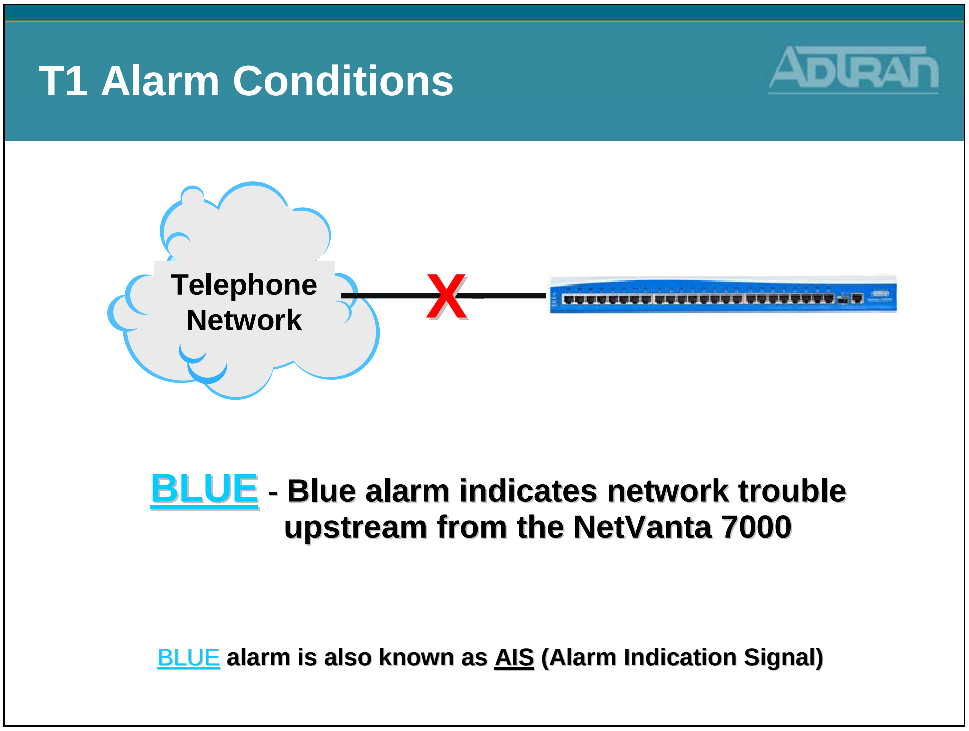
YELLOW - CSU/DSU responds to Red alarm condition by transmitting yellow alarm to inform far end of alarm condition (far end signal condition)

Remote Alarm Indication (RAI) is being received at the Network Interface to indicate that the far end is in Red Alarm. It may be inferred that the path from the far end to the near end is good since the RAI is being received successfully. (Note: "far end" refers only to the far end of the local loop, which may extend to the customers' "other" site or may only go to an intermediate Central Office.) This is inferred because framing must be adequate in order to receive a transmitted Yellow alarm.

In ESF, the Yellow Alarm is transmitted over the Facility Data Link (FDL). In SF (or D4), it is transmitted inband, by setting the second bit in every DS0 to zero; consequently, it is possible for payload data to mimic the code and cause a "false yellow alarm".

Any time a unit is in Red Alarm it will always be transmitting Yellow alarm toward the far end. There will be no indication of this on the local unit. The only indication will be at the far end unit if the transmit path is functioning properly.

T1 – Blue Alarm



Blue alarm indicates network trouble upstream from the NetVanta 7000. BLUE alarm is also known as AIS (Alarm Indication Signal) or an All 1’s pattern.

LOS (LOS of Signal)

A LOS is an alarm indication that occurs when the CSU does not receive a valid T1 signal (i.e., approximately 1.544 Mbps, nominally 3V peak). A common cause of LOS is an improperly wired cable from the demarcation point to the TSU. Additionally, if excessive zeros are received on an AMI line, LOS can be declared.

When an LOS condition is present the Red alarm will always be active because framing is absent as well.

show int t1 1/1 – In Alarm

show int t1 1/1 – In Alarm

- Display the T1 interface – In Alarm

NV7000# **show int t1 1/1**
t1 1/1 is **DOWN**
Transmitter is **sending remote alarm**
Receiver has **loss of signal**, **loss of frame**
T1 coding is B8ZS, framing is ESF
Clock source is line, FDL type is ANSI
Line build-out is 0dB
No remote loopbacks, No network loopbacks
Acceptance of remote loopback requests enabled
Tx Alarm Enable: rai
Last clearing of counters 20:26:52
loss of frame : 1, current duration 00:02:45
loss of signal : 1, current duration 00:02:44
AIS alarm : 0
Remote alarm : 0

DS0 Status: 123456789012345678901234
XXXXXXXXXXXXXXXXXXXXXXXX
Status Legend: '-' = DS0 is not allocated
'X' = DS0 is allocated (nailed)

* Continues on next slide

show int t1 1/1 (Continued…)

Continued…

Signaling Bit Status: 123456789012345678901234
RxA: ------------------------
RxB: ------------------------

TxA: ------------------------
TxB: ------------------------
123456789012345678901234

Line Status: -- **LOS** -- **Red** -- **Tx Yellow** --

5 minute input rate 0 bits/sec, 0 packets/sec
5 minute output rate 0 bits/sec, 0 packets/sec
Current Performance Statistics:
10 Errored Seconds, 0 Bursty Errored Seconds
0 Severely Errored Seconds, 167 Severely Errored Frame Seconds
167 Unavailable Seconds, 0 Path Code Violations
1 Line Code Violations, 1 Controlled Slip Seconds
0 Line Errored Seconds, 0 Degraded Minutes

TDM group 1, line protocol is not set
Encapsulation is not set

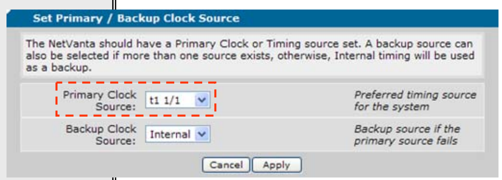
show int pri 1

show int pri 1

ADTRAN

- Display the ISDN PRI interface

```
NV7000# show int pri 1
pri 1 is DOWN
  Description: pri 1
  Switch protocol National ISDN 2
  calling-party override none
  calling-party presentation allowed
  calling-party name
  calling-party number
  area-code
  connect t1 1/1 tdm-group 1
  digits transferred all prefix
  isdn name-delivery none
  role user
  Number Conversion as dialed
  resource-selection circular
  Channel status 123456789012345678901234
                 ---------------------------. . . . .
   Legend: - = Unallocated       . = Inactive
         A = Active B channel   B = Backup D channel
         D = Active D channel   M = Maintenance
         R = Restart
              :
```



No D channel ?

Check System Clock Source from the System Summary screen

show int pri 1

ADTRAN

- Display the ISDN PRI interface

```
NV7000# show int pri 1
pri 1 is UP
  Description: pri 1
  Switch protocol:  National ISDN 2
  Signaling role:  user (TE)
  Calling-party override:  disabled
  Calling-party presentation:  allowed
  Calling-party number:  (no number configured)
  digits transferred 4
  ISDN name-delivery:  disabled
  Connected interface: t1 1/1 tdm-group 1
  Channel status 123456789012345678901234
         ------------------- . . AAD
   Legend: - = Unallocated       . = Inactive
         A = Active B channel   B = Backup D channel
         D = Active D channel   M = Maintenance
         R = Restart
   329 packets input, 1701 bytes, 0 no buffer
   0 runts, 0 giants, 0 throttles
   37 input errors, 1 CRC, 28 frame
   8 abort, 0 discards, 0 overruns
   441 packets output, 2068 bytes, 0 underruns
```

PRI Channels in use

PRI 1 D channel is UP

debug isdn l2-formatted

debug isdn l2-formatted

- Display ISDN Layer 2 formatted messages

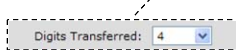
```
NV7000# debug isdn l2-formatted
14:57:09 ISDN.L2_FMT PRI  1 =====================================================
14:57:09 ISDN.L2_FMT PRI  1 Recd = Sapi:00  C/R:C Tei:00
14:57:09 ISDN.L2_FMT PRI  1       Ctl:INFO     Ns:4   Nr:4
14:57:09 ISDN.L2_FMT PRI  1       Prot:08  CRL:2  CRV:0002
14:57:09 ISDN.L2_FMT PRI  1         M - 05 SETUP
14:57:09 ISDN.L2_FMT PRI  1        IE - 04 BEARER CAPABILITY   Len=3
14:57:09 ISDN.L2_FMT PRI  1            80 Xfer Cap.:SPEECH
14:57:09 ISDN.L2_FMT PRI  1            90 Xfer Rate:64k
14:57:09 ISDN.L2_FMT PRI  1            A2 Layer 1:u-Law
14:57:09 ISDN.L2_FMT PRI  1        IE - 18 CHANNEL ID         Len=3
14:57:09 ISDN.L2_FMT PRI  1            A1 Primary Rate
14:57:09 ISDN.L2_FMT PRI  1               Intfc ID:IMPLICIT
14:57:09 ISDN.L2_FMT PRI  1               Pref/Excl:PREFERRED
14:57:09 ISDN.L2_FMT PRI  1               D-Chan Indicated:NO
14:57:09 ISDN.L2_FMT PRI  1               Chan. Sel:FOLLOWS
14:57:09 ISDN.L2_FMT PRI  1            83 Numb/Map:NUMBER
14:57:09 ISDN.L2_FMT PRI  1            97 Channel:23
14:57:09 ISDN.L2_FMT PRI  1        IE - 6C CALLING PARTY #     Len=12
14:57:09 ISDN.L2_FMT PRI  1            21 Numb. Type:NATIONAL
14:57:09 ISDN.L2_FMT PRI  1               Numb. Plan:ISDN/Telephony
14:57:09 ISDN.L2_FMT PRI  1            80 Presentation:ALLOWED
14:57:09 ISDN.L2_FMT PRI  1               Ph.# 8884238726
14:57:09 ISDN.L2_FMT PRI  1        IE - 70 CALLED PARTY #     Len=11
14:57:09 ISDN.L2_FMT PRI  1            A1 Numb. Type:NATIONAL
14:57:09 ISDN.L2_FMT PRI  1               Numb. Plan:ISDN/Telephony
14:57:09 ISDN.L2_FMT PRI  1               Ph.# 2568012003
```

debug isdn endpoint

debug isdn endpoint

- Display ISDN endpoint events

```
NV7000# debug isdn endpoint
15:17:13 ISDN.EP    PRI  1  Incoming call :'2568012003' from '8884238726'.
15:17:13 ISDN.EP    PRI  1  Call from 8884238726, wait for Name Facility msg
15:17:16 ISDN.EP    PRI  1  Call from 8884238726 - timeout waiting for Name Facility
15:17:16 ISDN.EP    PRI  1  Incoming number '2568012003' converted to '2003'
15:17:16 ISDN.EP    PRI  1  Incoming call to '2568012003' accepted
15:17:40 ISDN.EP    PRI  1  Call to '2568012003' connected.
15:17:59 ISDN.EP    PRI  1  Call to '2568012003' Process clearing. CCR 16
```



Physical Interface / PRI Config
- Digits Transferred set to 4

debug voice summary

debug voice summary

- View call routing summary real time
  - Can confirm proper trunk is being used

```
NV7000# debug voice summary
13:50:44:007 VOICE.SUMMARY voice user 2003 cos allowed the call to Long Distance
13:50:44:009 VOICE.SUMMARY 2003 is calling T03 (12568022003).
13:50:46:743 VOICE.SUMMARY RTP for Call from 0 to 12568022003:  Codec PCMU
13:50:49:931 VOICE.SUMMARY 2003 is connected to T03 (12568022003)
13:50:54:493 VOICE.SUMMARY Call from 2003 to T03 (12568022003) ended by 2003:
13:51:24:119 VOICE.SUMMARY T01 is calling AutoAttendantAcct (8200).
13:51:24:421 VOICE.SUMMARY RTP for Call from 8021000 to 8200:  Codec G729
13:51:24:421 VOICE.SUMMARY T01 is connected to AutoAttendantAcct (8200)
13:51:27:598 VOICE.SUMMARY T01 is calling 2003 (2003).
13:51:27:843 VOICE.SUMMARY Call from T01 to AutoAttendantAcct (8200) ended by Au
toAttendantAcct: normal clearing
```

Voice Trunk ID

debug voice autoattendant

debug voice autoattendant

- Display Auto Attendant events

```
NV7000# debug voice autoattendant

16:07:06 VXMLInterpreter vxml.8201 Ca:0 # Started prompt 'CFLASH:/AA/Prompts/Main.wav'
16:08:35 VXMLInterpreter vxml.8201 Ca:0 ProcessingLogic.dtmf input '0101' matched ''
16:08:35 VXMLInterpreter vxml.8201 Ca:0 # Started prompt 'CFLASH:/AA/Prompts/HOLD.wav'
16:08:37 VXMLInterpreter vxml.8201 Ca:0   Transfering call to 'tel:8301'

16:08:37 VXMLInterpreter vxml.8301 Ca:2 # Started prompt 'CFLASH:/AA/Prompts/Choice.wav'
16:08:52 VXMLInterpreter vxml.8301 Ca:2 ProcessingLogic.dtmf input '3' matched ''
16:08:57 VXMLInterpreter vxml.8301 Ca:2   Transfering call to 'tel:912568012003'
```

** Partial output displayed*

debug voice mail

debug voice mail

- Display Voicemail events

NV7000# **debug voice mail**

16:27:10:073 VOICEMAIL.Appearance Acct SC: CA 0:   8500 to **VM DigitGathering**.

16:26:23 VXMLInterpreter vxml. Ca:0 # Set 'VMTargetBox' to **'2003**'

VoiceMail.MailboxManager: Sending **New Message** Filename of
CFLASH:/VoiceMail/Messages/**G04Q26RMCI.wav** to VSCO for Mailbox **2003**

** Partial output displayed*

Module Summary

Module Summary

- At the end of this module, you should be able to:
- Recognize NetVanta 7000 IP PBX Applications
- Configure T1–RBS/ISDN PRI Voice Trunks
- Create and Configure a Multi-level Auto Attendant
- Create and Configure Dial by Name Directories
- Configure Busy Lamp Field/ Public Park Zones
- Enable SMDR Call Logging
- Conduct Voice Troubleshooting in a NetVanta 7000 IP PBX Application

Module 7: NetVanta 7000 Series Data Configuration – Part 2

Module Objectives

Module Objectives

- Discuss Switch/Router concept
- Create another VLAN interface
- Configure Firewall policies
- Create a network DMZ
- Introduce Quality of Service concepts
- Configure QoS Maps
- Basic Firewall and QoS troubleshooting

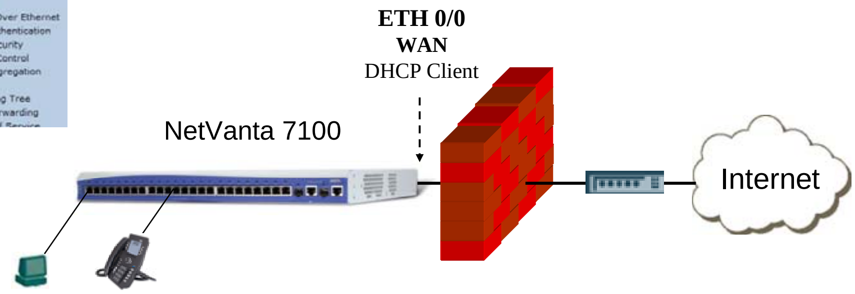
VLAN (Network) Interfaces

NetVanta IP Telephony Course

VLAN (Network) Interfaces

Data / Switch Defaults Review

Data / Switch Defaults Review



VLANs
Data - VLAN 1
IP Address: 10.10.10.1/24
Voice - VLAN 2
IP Address: 10.10.20.1/24

Ethernet 0/1-24
Switchport mode: trunk
Allowed VLANs: all
Native VLAN: 1
Spanning Tree Mode: edgeport

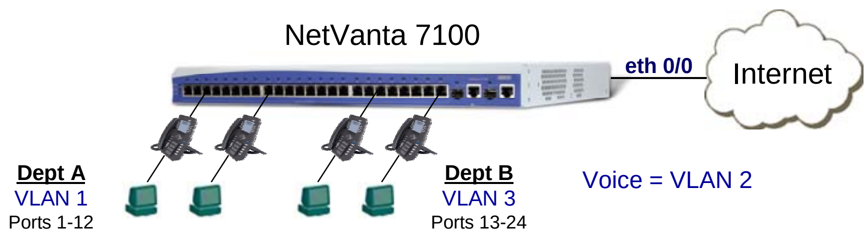
Gigabit 0/1-2
VLAN membership: trunk
Allowed VLANs: all
Native VLAN: 1

Adding Network Interfaces

Adding Network Interfaces

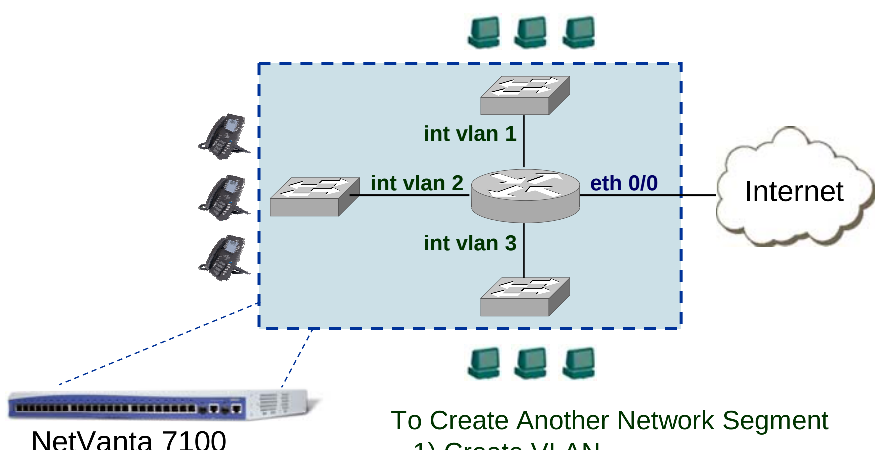
Company XYZ has two departments. Both departments have IP phones that will reside on VLAN 2. The PCs for each department need to be on isolated network segments.

Solution: Create Another Network Segment
1) Create VLAN
2) Assign IP Address to VLAN
3) Change Native VLAN on desired ports



NetVanta 7100 - Switch Router Concept

NetVanta 7100 Switch Router Concept



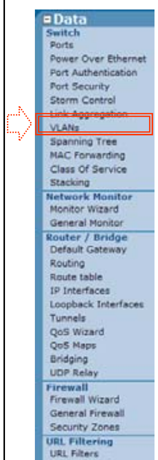
To Create Another Network Segment
1) Create VLAN
2) Assign IP Address to VLAN
3) Assign Switchport to VLAN

VLAN Interfaces

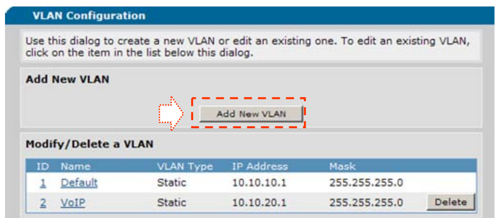
VLANs can be configured with IP information to allow the built in router to route information between them. This is known as Inter-VLAN routing. The VLAN becomes an actual router interface with it own unique network IP address. The IP address assigned to the VLAN interface will act as the default gateway to devices connected to ports that are members of this VLAN.

To Create a New VLAN

To Create a New VLAN



1) From the VLANs screen, select Add New VLAN



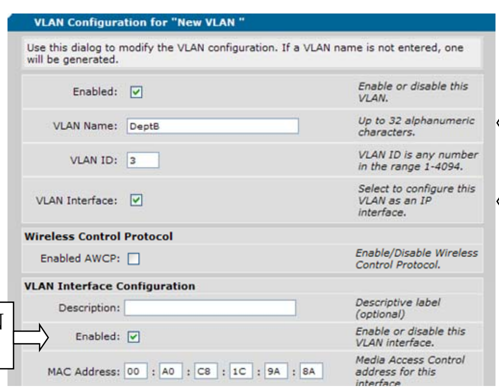
– The NetVanta 7100 supports up to 255 active VLANs

more

To Create a New VLAN

To Create a New VLAN

2) Configure new VLAN



New VLAN
Name: DeptB
ID: 3

Assign VLAN Name and ID

Enable IP on this interface

Enable VLAN interface

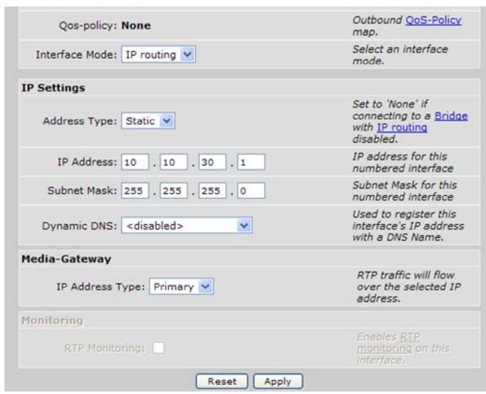
more

To Create a New VLAN

To Create a New VLAN

2) Configure new VLAN (Continued…)

<u>Static IP Address</u>
10.10.30.1
255.255.255.0



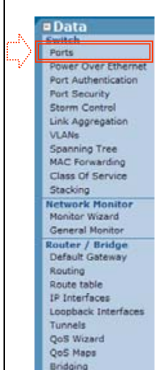
Address Type set to Static

VLAN IP address and subnet mask

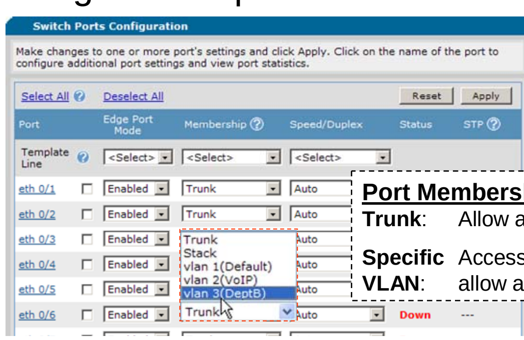
Media-Gateway set to Primary

To Create a New VLAN

To Create a New VLAN



3) Assign Switchport to VLAN



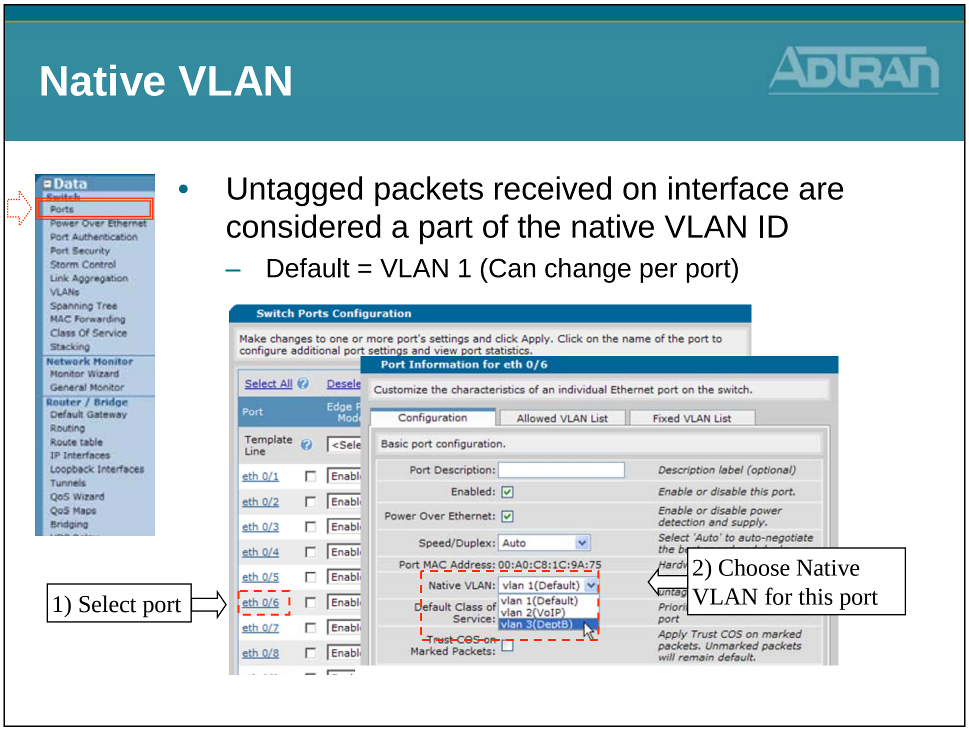
<u>Port Membership:</u>

Trunk: Allow all VLANs

Specific VLAN: Access Port - allow assigned VLAN only

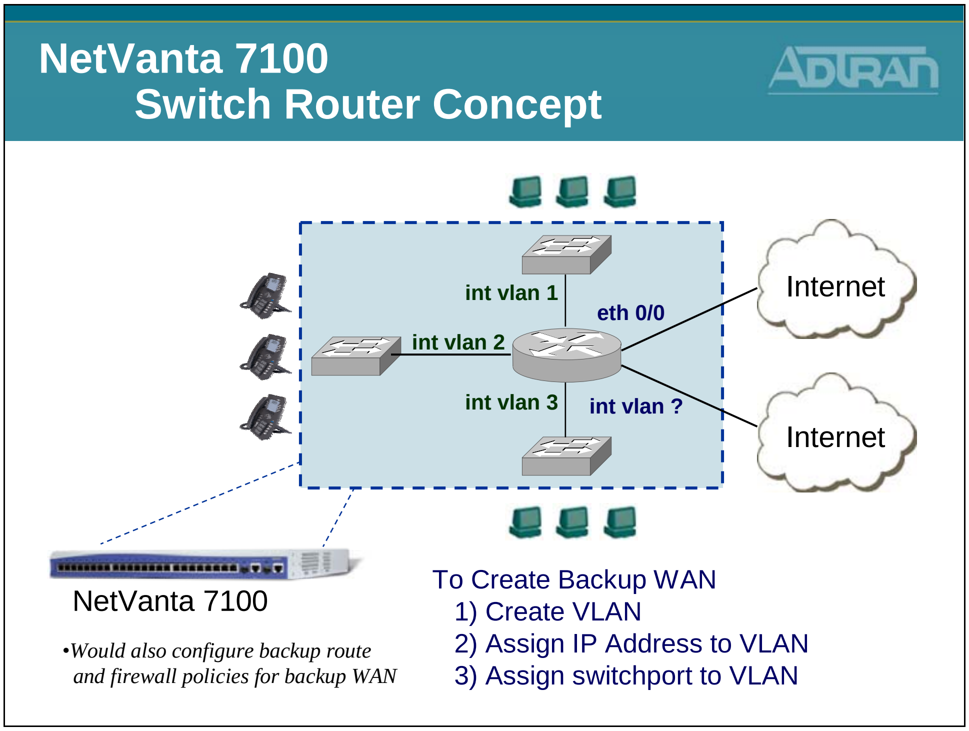
<u>InterVLAN Routing:</u>
Traffic from one VLAN destined for another VLAN must go through the router. Firewall policies can be configured to allow or disallow.

Native VLAN



A Switchport configured as a Trunk port (802.1Q) allows all VLANs by default. When traffic enters a switchport, it knows what VLAN it is assigned to based on the 802.1Q VLAN ID. The Native VLAN option is used to associate untagged (no VLAN ID) traffic to a VLAN. By default, untagged traffic is assigned to VLAN 1.

NetVanta 7100 - Switch Router Concept



As illustrated in an earlier example, a routable VLAN interface can be created by adding a new VLAN, assigning an IP address to that VLAN, and then assigning the new VLAN to a Switchport. This new routable interface can be an additional LAN network, an isolated DMZ, or a backup WAN as illustrated above.

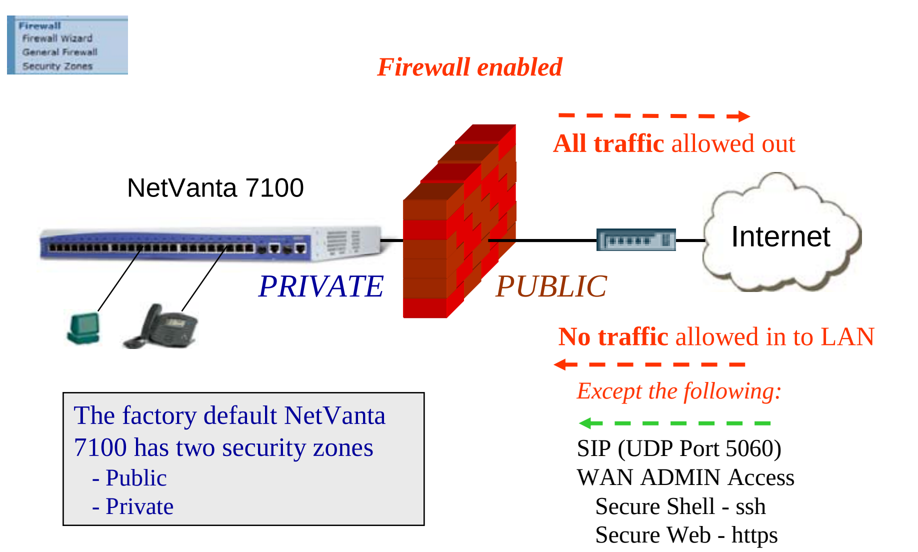
Firewall Configuration

NetVanta IP Telephony Course

Firewall Configuration

NetVanta 7100 - Data/Firewall Factory Defaults

NetVanta 7100
Data / Firewall Factory Defaults

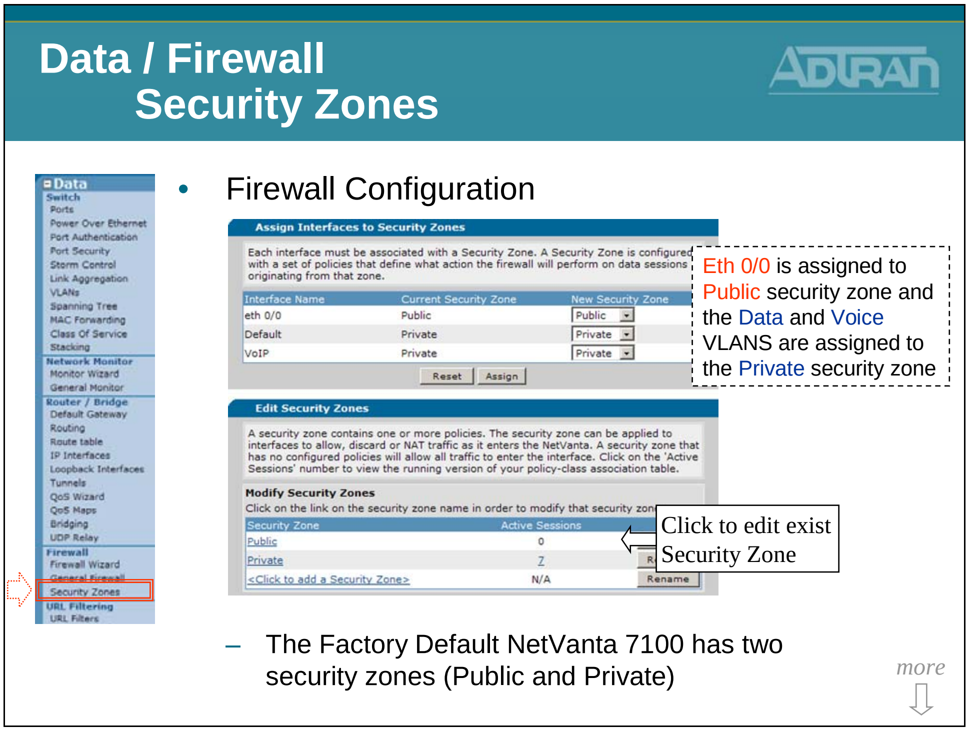


The factory default NetVanta 7100 allows (and NATs) all traffic going to the internet. UDP port 5060 SIP traffic, secure shell, and secure web traffic are the only traffic allowed in the PUBLIC interface by default. The policies allowing this traffic can be removed if you do not currently wish to allow that type of traffic.

The NetVanta 7100 is equipped with a stateful inspection firewall. A stateful inspection firewall operates by monitoring traffic passing through it. It only allows traffic it is specifically configured to allow as well as return traffic matching traffic that was specifically allowed.

For example, if a computer sends a request to a web site, through the firewall, it is only necessary to configure an allow for the outbound traffic, the traffic from the requesting computer to the web server. The response traffic from the website will be automatically allowed. All traffic that has not been initiated from within the network will be automatically blocked unless otherwise specified.

Data/Firewall - Security Zones



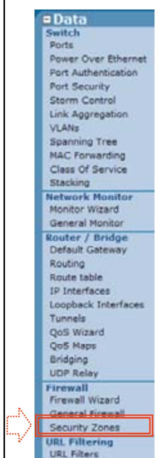
Each interface should be associated with a Security Zone. A Security Zone is configured with a set of policies that define what action the firewall will perform on data sessions originating from that zone. A security zone that has no configured policies will allow all traffic to enter the interface.

The Public and Private Security Zone listed above are present with the factory delivered NetVanta 7100. The firewall inspects traffic inbound. To control traffic coming from the Internet, modify the Public Security Zone. To control traffic coming from VLAN 1 or VLAN 2, modify the Private Security Zone.

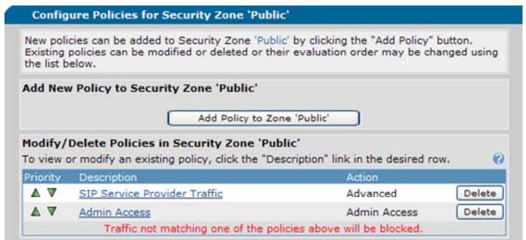
Data/Firewall - Public Security Zone

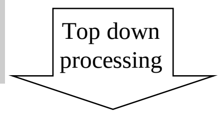
Data / Firewall
Public Security Zone

ADTRAN



- Access from Outside the NetVanta 7100



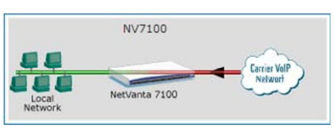


- – SIP Service Provider Traffic
  - Allow SIP (UDP 5060) traffic in
- – Admin Access
  - Allow allows https and ssh access from the Public security zone

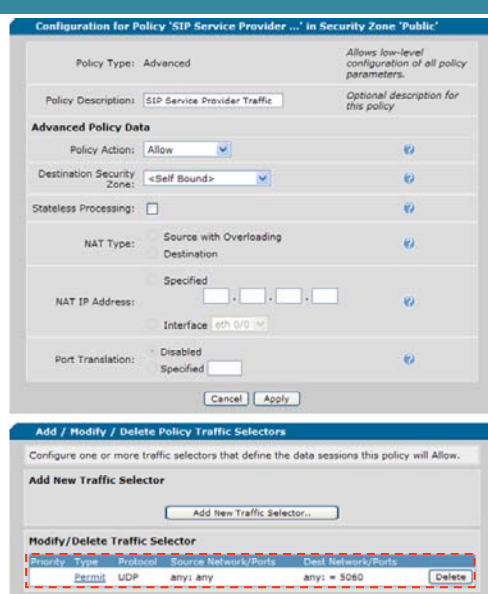
Public Security Zone - SIP Service Provider Traffic

Public Security Zone
SIP Service Provider Traffic

ADTRAN

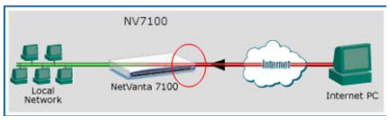


- The SIP Service Provider policy allows SIP – UDP port 5060 from anywhere in to the NetVanta 7100
  - – If this is truly from a SIP service provider, the traffic selector should be modified defining the source as the specific IP address of service provider



Public Security Zone – Admin Access

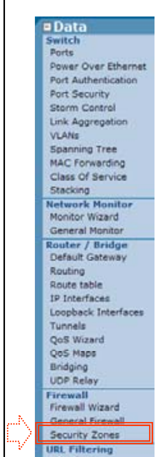
Public Security Zone Admin Access



- The Admin Access policy allows https and ssh access from the Public security zone by default
  - Can be more specific
  - Could also allow other admin traffic such as:
    - HTTP, FTP, SNMP, Telnet, and Ping

Data/Firewall - Private Security Zone

Data / Firewall Private Security Zone

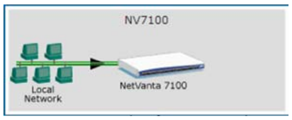


- Access from LAN

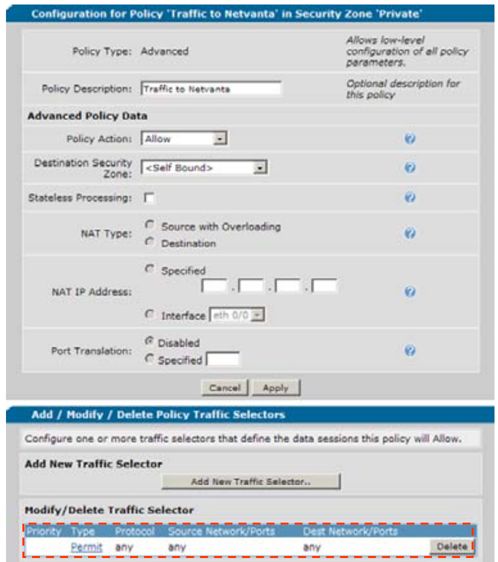
- Traffic to NetVanta
  - Inside traffic with destination of NetVanta 7100 allowed
- Voice / Data VLAN Traffic
  - Allow VLAN to VLAN traffic
- NAT list NAT
  - Allow all traffic going to Internet

Private Security Zone – Traffic to NetVanta

Private Security Zone
Traffic to NetVanta

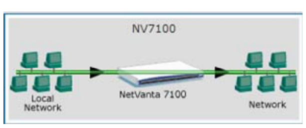


- Inside traffic with a destination of the NetVanta 7100 is allowed
  - Examples:
    - SIP
    - RTP
    - DHCP
    - TFTP
    - FTP

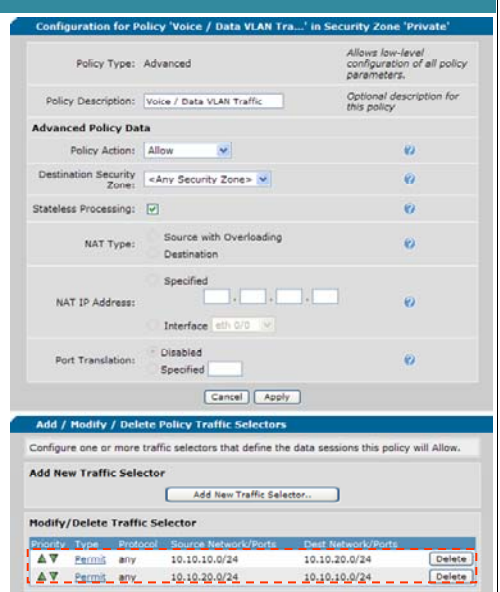


Private Security Zone – Voice / Data VLAN Traffic

Private Security Zone
Voice / Data VLAN Traffic

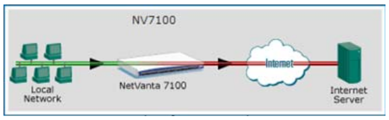


- Allow VLAN to VLAN traffic
- Required if you want to allow the following:
  - PC with Softphone to call a SIP hard phone
  - PC to access WEB GUI of an IP phone

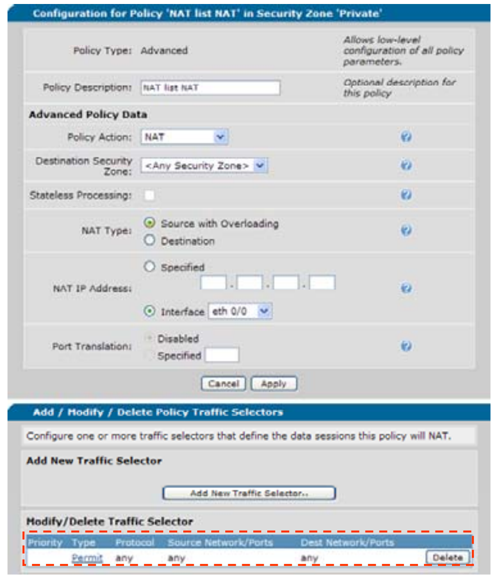


Private Security Zone – NAT list NAT

Private Security Zone NAT list NAT



- Allow all traffic going to Internet
  - Traffic selectors matches all traffic
  - Outbound traffic is translated from the private inside IP address to the public IP address assigned to the outgoing interface

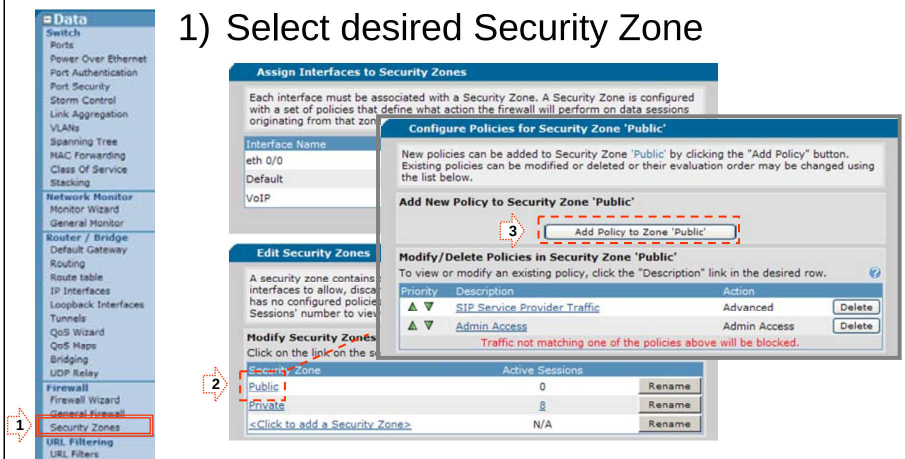


Security Zones - Adding New Policies

Data / Firewall Security Zones Adding New Policies

ADTRAN

1) Select desired Security Zone



– A Security Zone is configured with a set of policies that define what action the firewall will perform on data sessions originating from that zone (inbound)

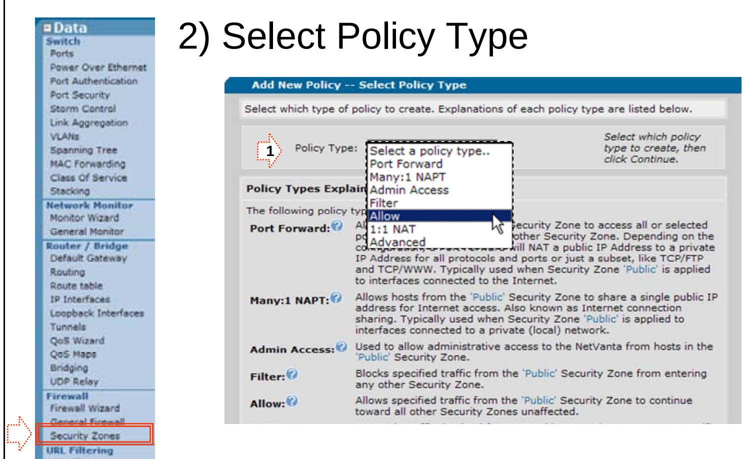
more

Security Zones - Adding New Policies

Data / Firewall / Security Zones Adding New Policies

ADTRAN

2) Select Policy Type

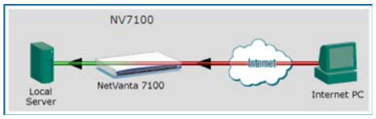


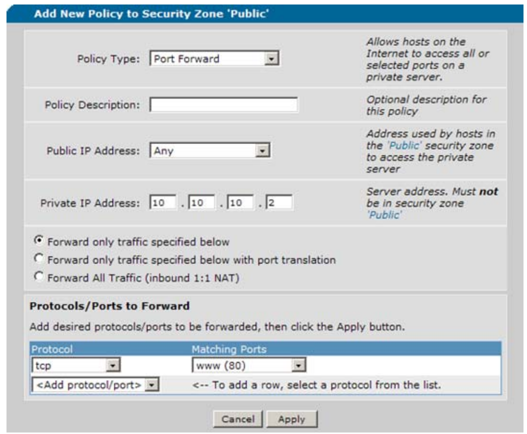
3) Configure new Policy Settings
- Based on Policy Type selected

more

Security Zones Policies – Port Forward

Security Zone Policies
Port Forward



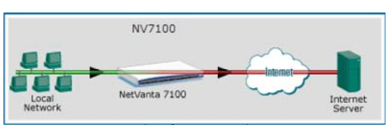


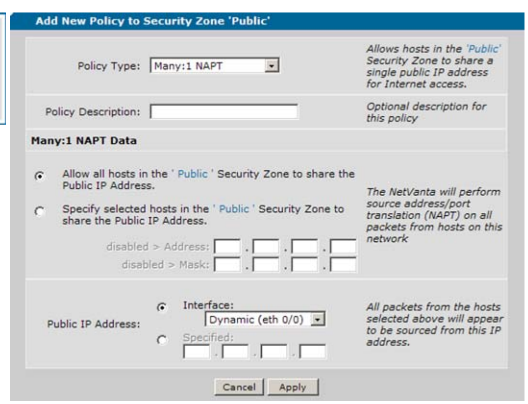
Port Forward Policy

Allows hosts from the selected Security Zone to access all or selected ports on a private server in another Security Zone

Security Zones Policies – Many:1 NAPT

Security Zone Policies
Many:1 NAPT

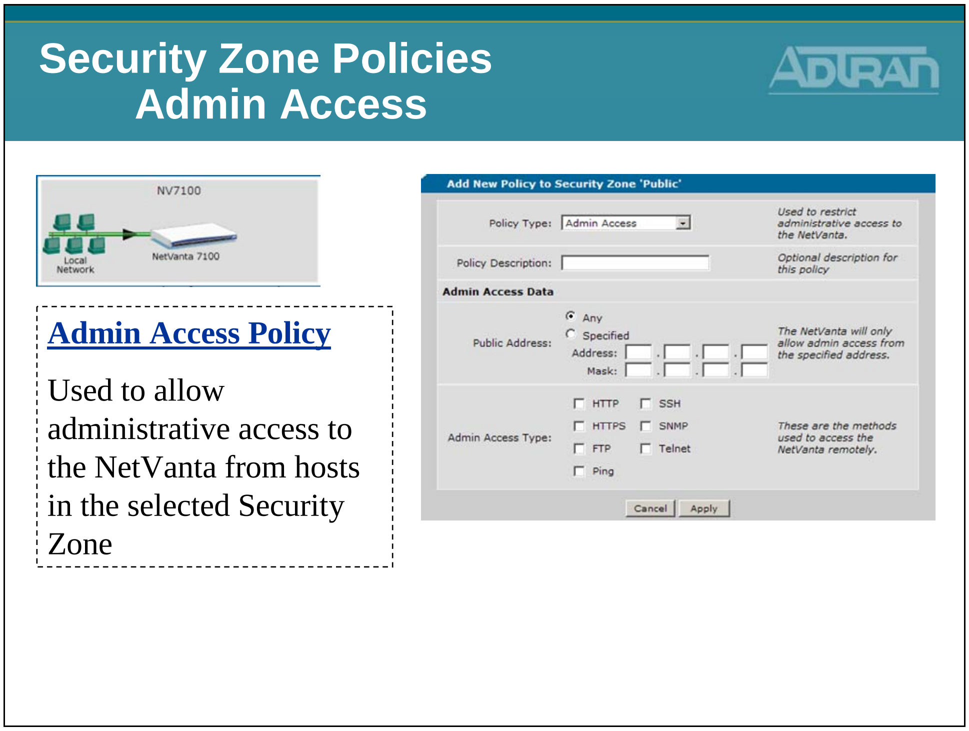




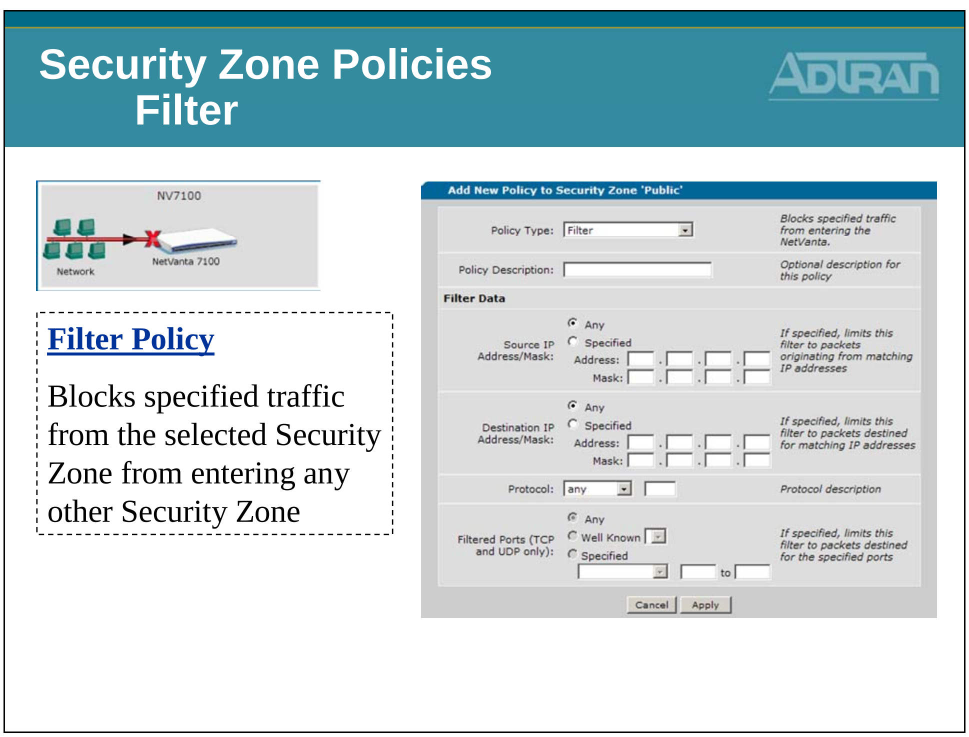
Many:1 NAPT Policy

Allows hosts from the selected Security Zone to share a single public IP address for Internet access. Also known as Internet connection sharing

Security Zones Policies – Admin Access

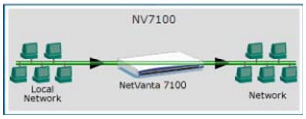


Security Zones Policies – Filter



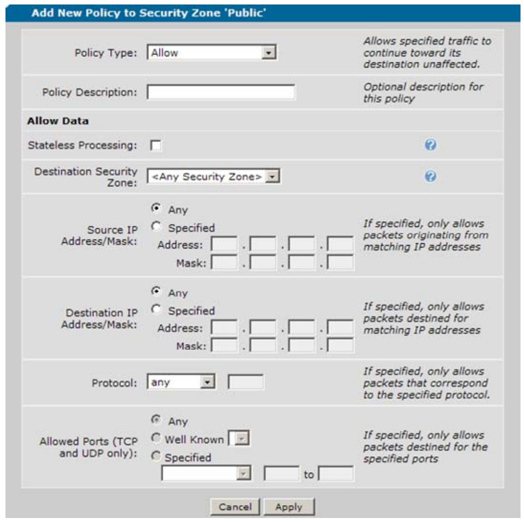
Security Zones Policies – Allow

Security Zone Policies Allow



Allow Policy

Allows specified traffic from the selected Security Zone to continue toward all other Security Zones unaffected

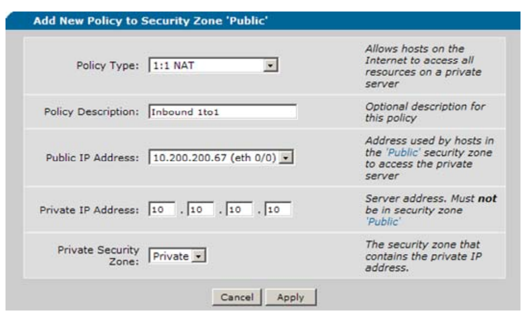


Security Zones Policies – 1:1 NAT

Security Zone Policies 1:1 NAT

1:1 NAT Policy

Forwards traffic destined for an IP address on the system to a specific IP address in another security zone by changing the destination IP address



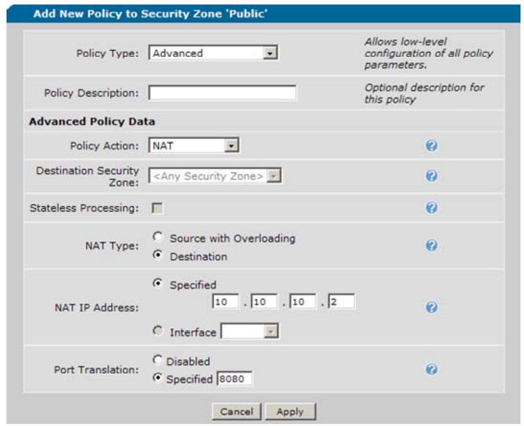
Security Zones Policies – Advanced

Security Zone Policies
Advanced

ADTRAN

Advanced Policy

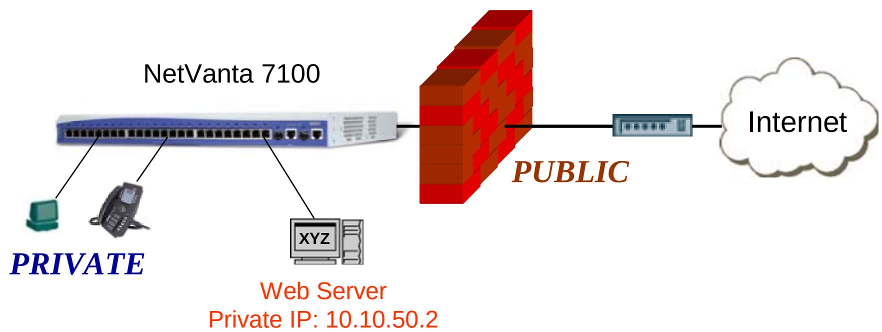
Allows low-level configuration of all policy parameters



Firewall Example - Public Web Server

Data / Firewall Example
Public Web Server

1. Modify the Public Security Zone
2. Add a Port Forwarding Rule
   – Traffic destined to public IP address - port 80, will be forward to the private IP address of the web server

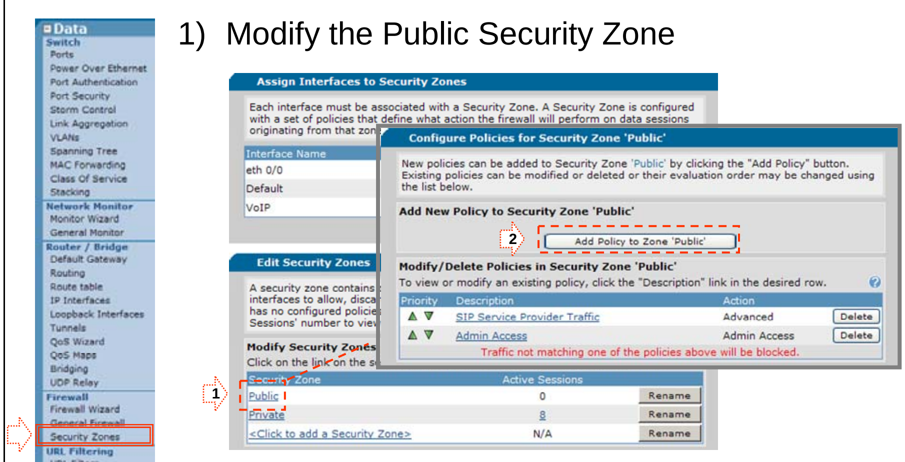


** Placing public web server in DMZ is covered later in this module*

Firewall Example – Add Port Forwarding Rule

Data / Firewall Example
Add Port Forwarding Rule

1) Modify the Public Security Zone



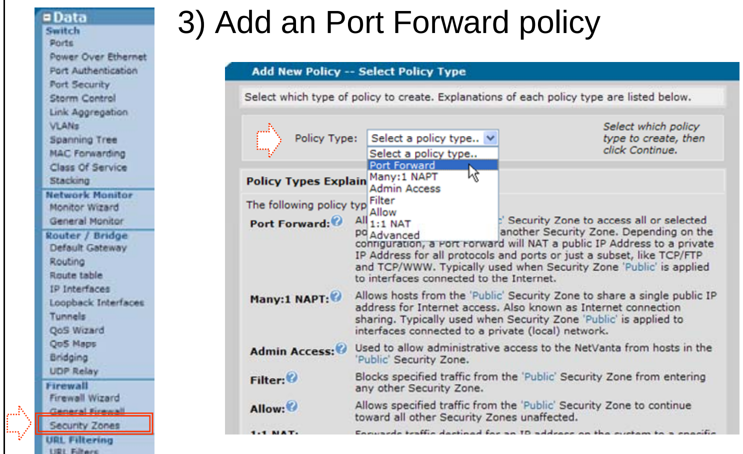
2) Add a policy to zone Public

more

Firewall Example – Add Port Forwarding Rule

Data / Firewall Example
Add Port Forwarding Rule

3) Add an Port Forward policy



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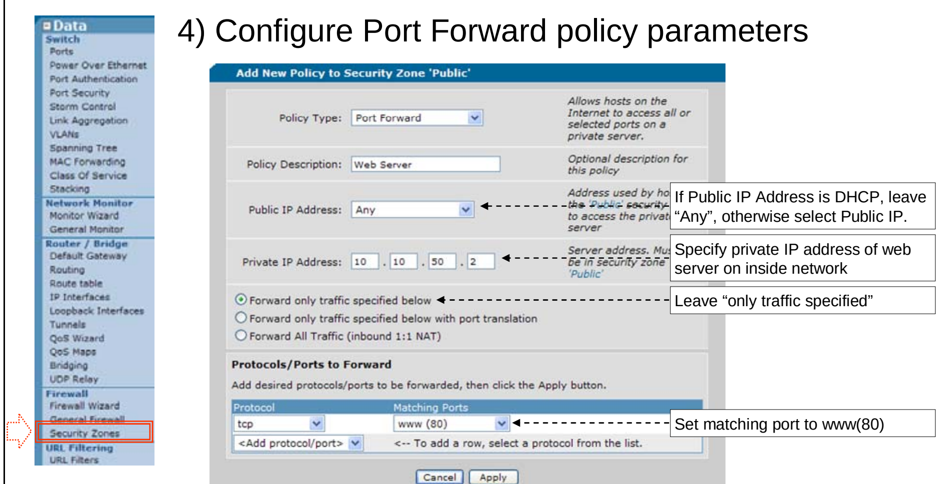
Firewall Example – Add Port Forwarding Rule

Data / Firewall Example
Add Port Forwarding Rule

4) Configure Port Forward policy parameters

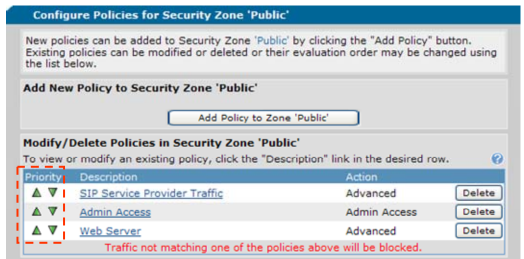


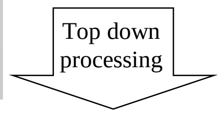
Firewall Example – Port Forwarding Rule

Data / Firewall Example Port Forwarding Rule

ADTRAN

- New “Web Server” Port Forwarding Rule was added to bottom of Policy list



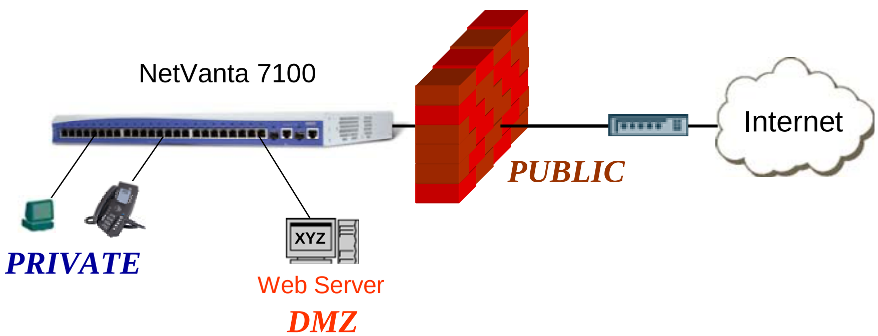


  - Arrow keys can be used to change the order of rules.

NetVanta 7100 - DMZ

NetVanta 7100
DMZ

A network DMZ (demilitarized zone) is an isolated network used to host an organization's external services. It adds an additional level of security. Only specified traffic is allowed into the DMZ. No traffic initiated from the outside is allowed into the PRIVATE LAN. If an external attacker breaks into a server in the DMZ, they only have access to that server, not the PRIVATE LAN.

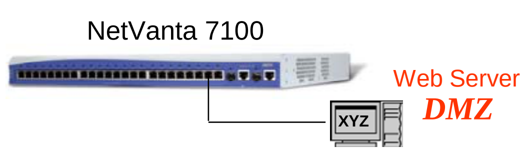


NetVanta 7100 - DMZ

NetVanta 7100
Creating a DMZ

1. Create a DMZ VLAN
   – Assign an IP Address to the new DMZ VLAN
   – Assign a switch port to the DMZ VLAN
2. Create a new DMZ Security Zone
   – Block traffic initiated in the DMZ security zone from entering the NetVanta 7100
   – Assign the new DMZ VLAN to the new DMZ security zone
3. Add a Port Forwarding rule for Web traffic
   – Forward all www traffic destined to the NetVanta 7100 public interface in to a web server located in the new DMZ VLAN

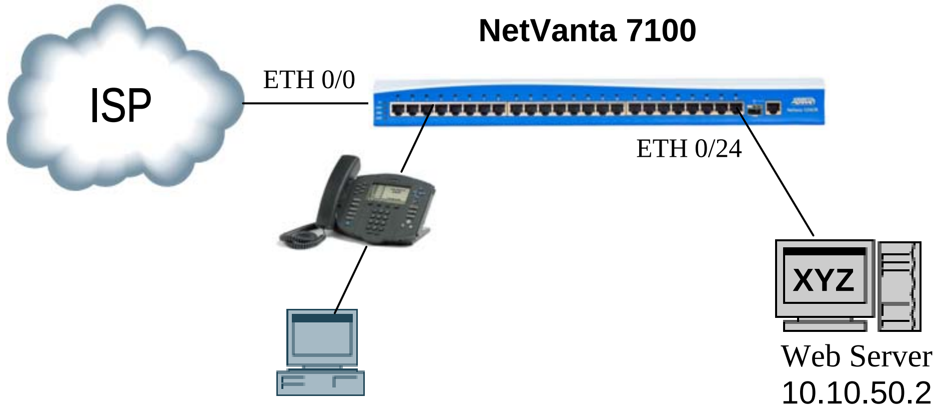
** Step by step instructions included with this document*



NetVanta 7100 - DMZ Creation

Instructor Led Exercise

DMZ VLAN
VLAN ID: 5
IP Address: 10.10.50.1
Subnet Mask: 255.255.255.0
DMZ Port: Eth 0/24



NetVanta 7100 Firewall Configuration

In this exercise you will add a DMZ VLAN to the NetVanta 7100 and then make it routable by assigning an IP address to it. You will then create a Port Forwarding policy to forward web traffic destined to the Public interface of the NetVanta 7100 in to a web server located in the DMZ VLAN. Finally, you will create a DMZ security zone to block traffic initiated within the DMZ VLAN.

DMZ Creation Overview

1. Create a DMZ VLAN
   - Assign an IP Address to the new DMZ VLAN
   - Assign a switch port to the DMZ VLAN
2. Create a new DMZ Security Zone
   - Block traffic initiated in the DMZ security zone from entering the NetVanta 7100
   - Assign the new DMZ VLAN to the new DMZ security zone
3. Add a Port Forwarding rule for Web traffic
   - Forward all www traffic destined to the NetVanta 7100 public interface in to a web server located in the new DMZ VLAN

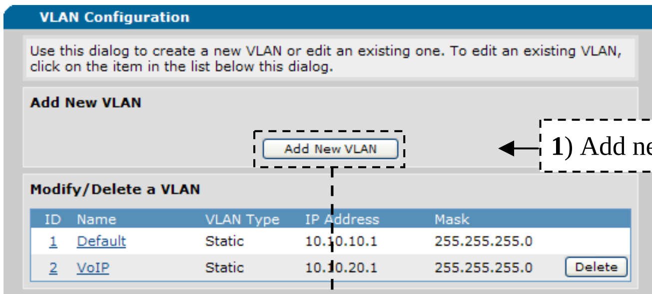
SETUP

- ❑ This exercise builds on the NetVanta 7100 factory default configuration
- ❑ Plug the NetVanta 7100 in to an AC power source.
- ❑ Connect one end of an Ethernet cable to Ethernet port of the PC (Configured as DHCP Client) and the other end to Ethernet 0/1 on the NetVanta 7100.
- ❑ Connect an Ethernet cable between Ethernet 0/0 of the NetVanta 7100. Connect the other end to the Internet connection provided by your ISP.
- ❑ From your PC, open the installed browser (if not already open) and enter **10.10.10.1/admin** in the Address field. The NetVanta login window appears. Enter **admin** as the username, **password** as the password, and then click the OK button.

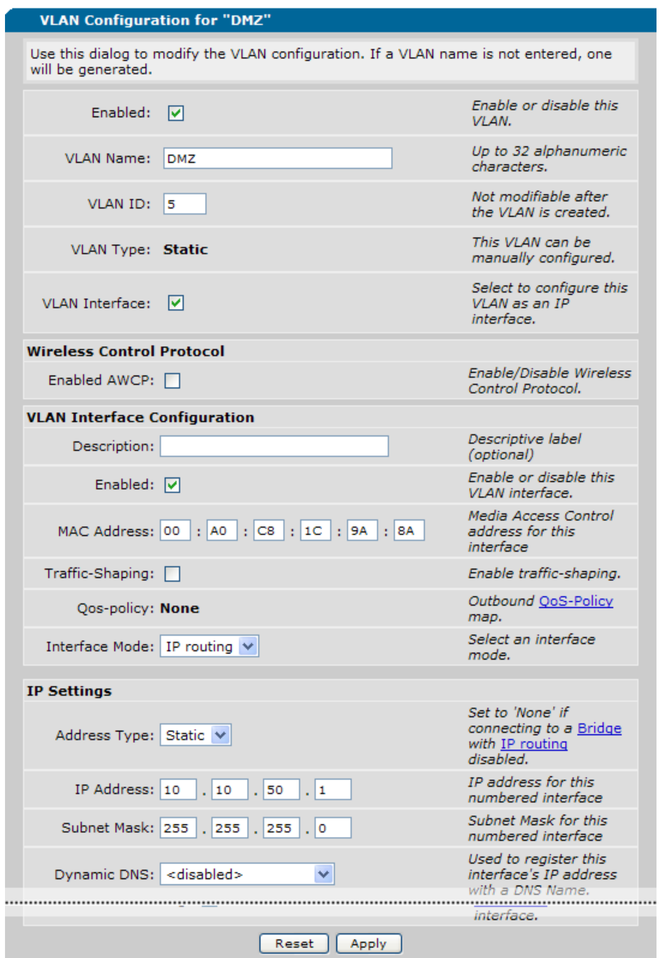
Step 1) Create a new VLAN to be used for a DMZ

From the NetVanta 7100 *Data / Switch / VLANs* screen, add a new VLAN, enable IP, and configure the IP address for the DMZ VLAN. Then add port *eth 0/24* to the DMZ VLAN.

VLAN Name: **DMZ**
VLAN ID: **5**
VLAN IP Address: **10.10.50.1**
Subnet Mask: **255.255.255.0**



1) Add new VLAN



2) Enable, name, and assign VLAN ID

3) Enable IP on this VLAN interface

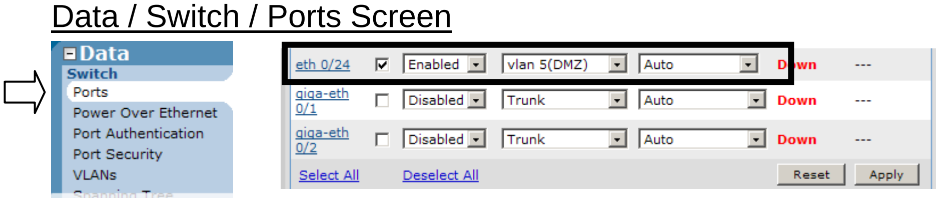
4) Enable VLAN (layer 3)

5) Select Static

6) Assign static IP address
10.10.50.1
255.255.255.0

7) Click Apply

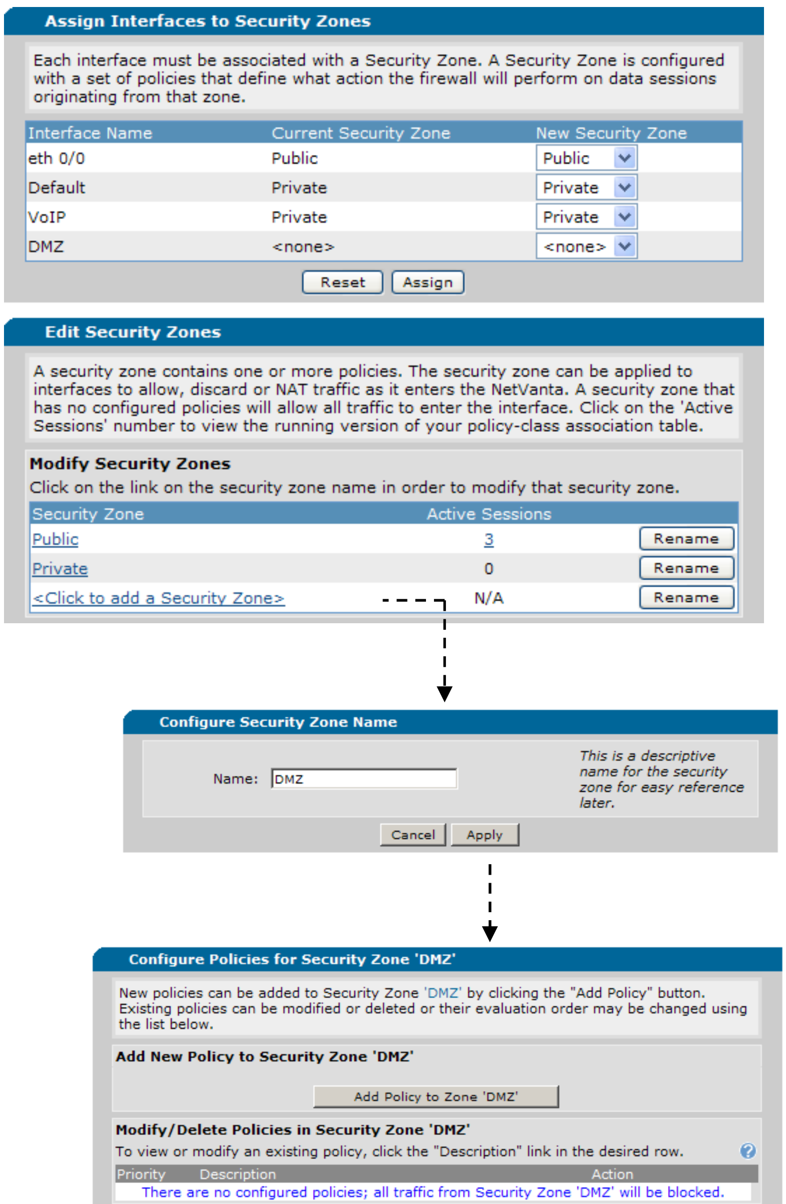
Data / Switch / Ports Screen



8) Add port eth 0/24 to the DMZ VLAN then click Apply

Step 2) Create a new Security Zone for the DMZ

From the NetVanta 7100 *Data / Firewall / Security Zones* screen configure an 'Unused Security Zone' to be used as the DMZ. This security zone will be configured to block traffic initiated from within the DMZ VLAN.



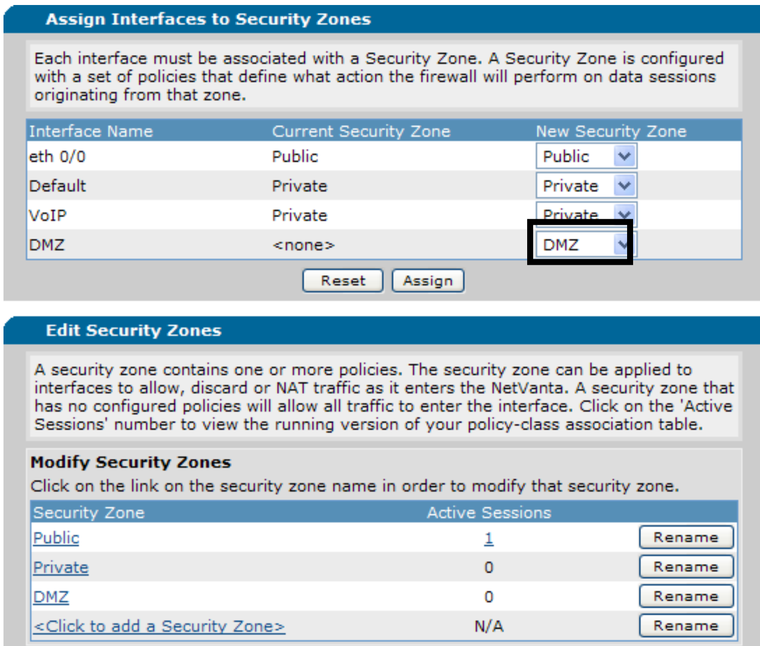
1) Click to add a new Security Zone

2) Type 'DMZ' for the Security Zone name and then click Apply

A new security zone has been created. By default, there are no configured polices in this security zone. All traffic initiated from within the DMZ Security Zone will be blocked from entering the NetVanta 7100.

Step 2 *(Continued…)* Assign VLAN #5 to the DMZ Security Zone

From the *Data / Firewall / Security Zones* screen, place interface DMZ (VLAN #5) in the new DMZ security zone. All traffic originating in the DMZ VLAN will be blocked from entering the NetVanta 7100.

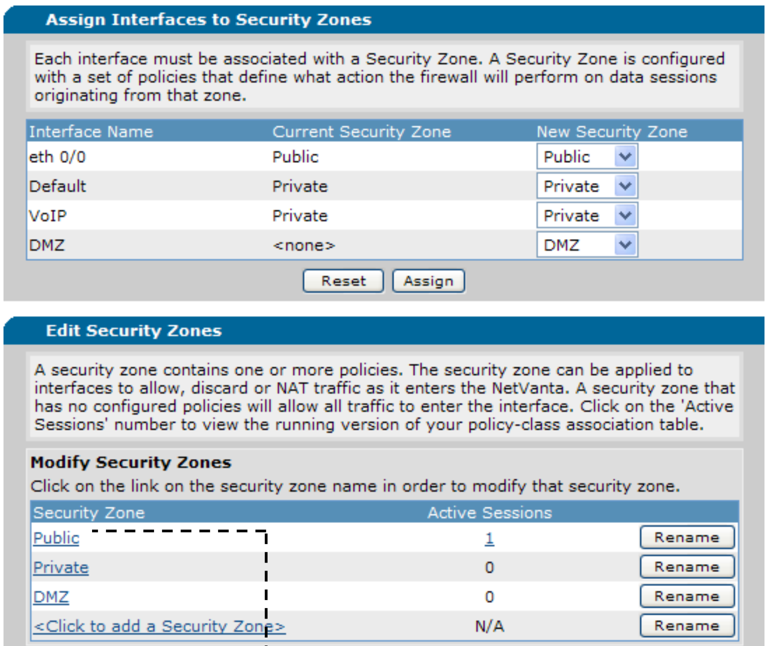


1) Select DMZ as the Security Zone

2) Click Assign

Step 3) Add a Port Forwarding Rule

From the *Data / Firewall / Security Zones* screen, add a port forwarding rule to the Public Security Zone. The new rule will be configured to forward all WEB traffic destined to the public IP address of the NetVanta 7100 in to the private IP address of the WEB server located in the DMZ security zone.



1) Select the Public Security Zone

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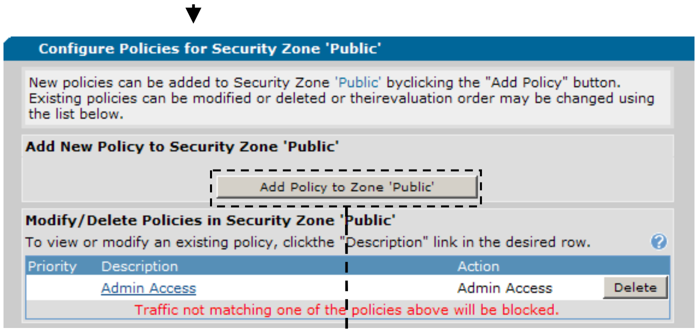
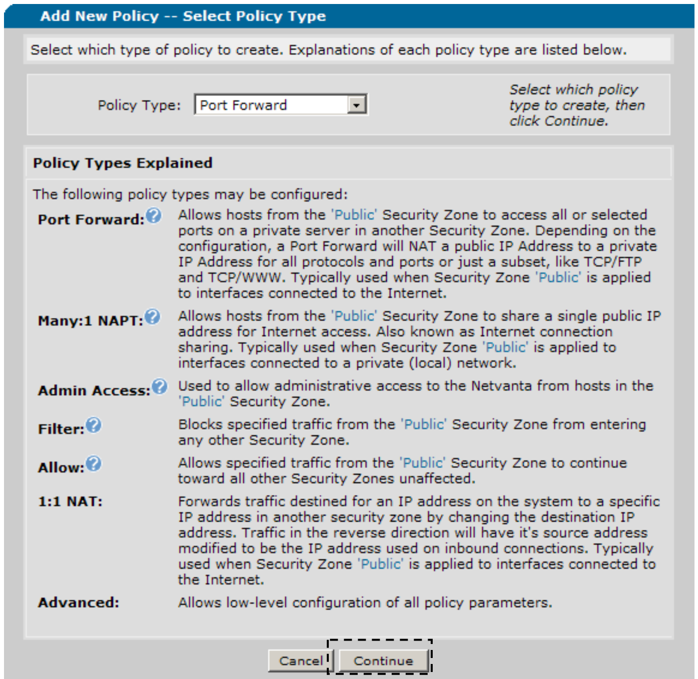


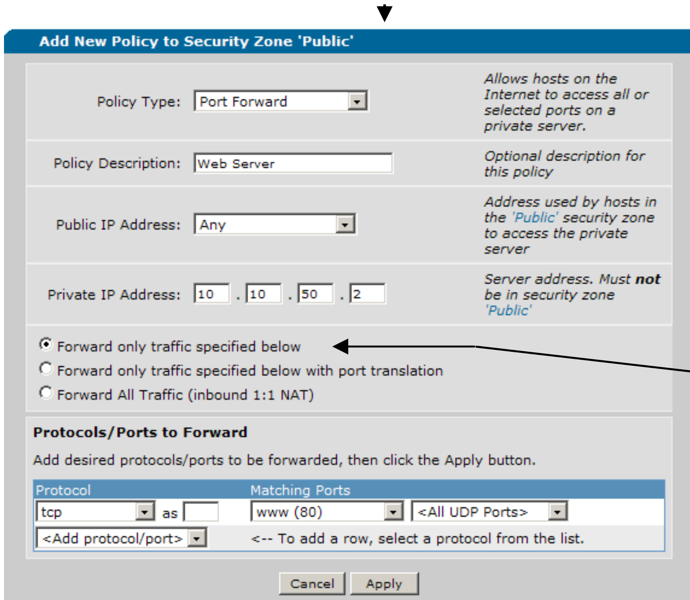
2) Click Add Policy to Zone Public

Add a Port Forwarding Rule *(Continued…)*



← **3**) Select Policy Type **Port Forward**

← **4**) Click Continue



← **5**) Enter description

← **6**) Set Public IP Address to **Any**

← **7**) Set Private IP Address to **10.10.50.2**

8) Choose the "Forward only traffic specified below" option button

← **9**) Set matching port to **www** (port **80**)

← **10**) Click Apply

The new port forwarding rule has been added to the Public security zone. All port 80 web traffic destined for the public IP address of the NetVanta 7100 will be forward in to the private IP address of the WEB server located in the DMZ security zone.

Quality of Service

NetVanta IP Telephony Course

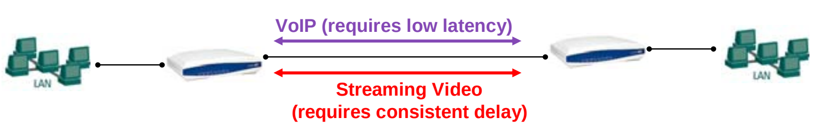
Quality of Service

Quality of Service (QoS)

Quality of Service (QoS)

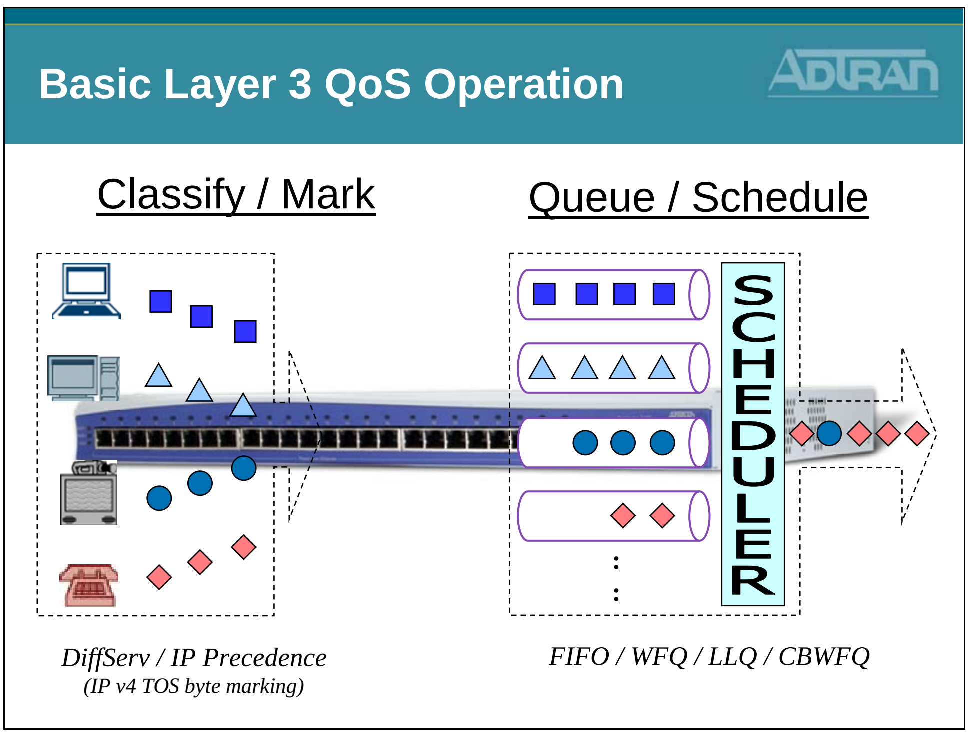
(QoS) – A technique used to differentiate between packet types and allow important traffic to receive higher priority

- A diverse mixture of protocols typically share the same data path in today's networks
- Different traffic types can impact each other across the connection
- QoS is intended to allow certain applications to achieve the level of performance considered necessary for optimal function
- The whole point is to provide a predictable level of service



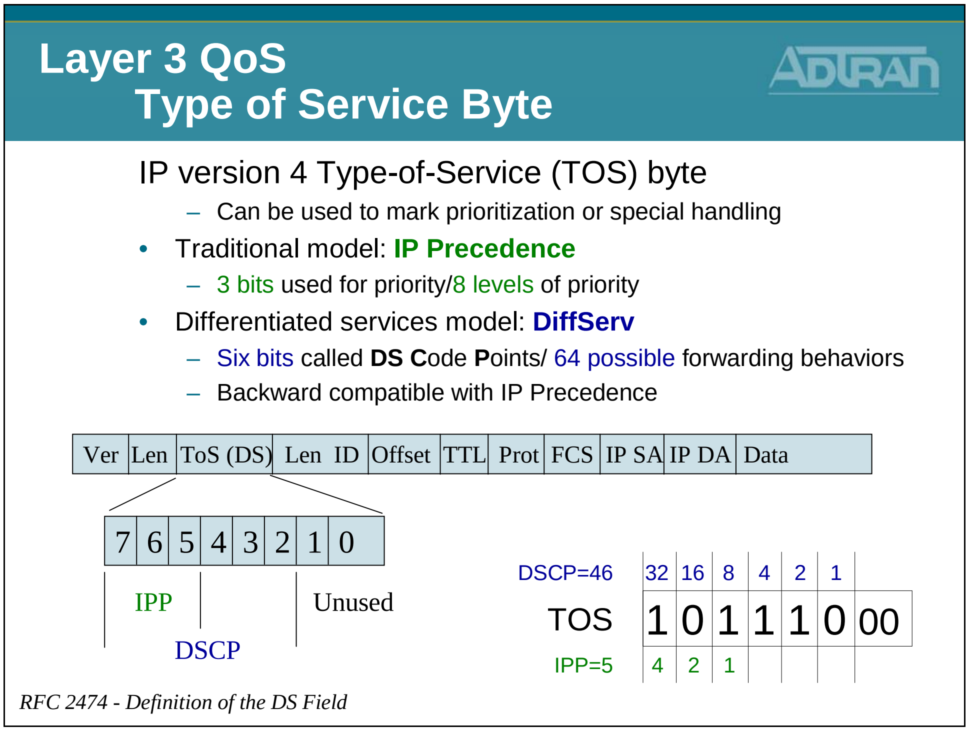
Quality of Service (QoS) is a technique used to differentiate between packet types and allow important traffic to receive higher priority. In a non-QoS-enabled IP network, all packets generally receive the same best-effort service. QoS is intended to allow applications that may require a certain type of network performance to be able to achieve that level of performance. Network applications require different types of response. Some may need very low latency, like Voice over IP. Others can handle longer latency, but need consistent delay. An example of this is streaming video. QoS helps give these types of applications a predictable level of service.

Basic Layer 3 QoS Operation



The basic operation of QoS involves classifying the different types of traffic and then marking the traffic to give a certain level of priority. Marking might be done by the originating equipment or by the router. Queuing only takes place when the transmitting interface is congested (or full). Traffic is placed in queues where it waits to get serviced out the transmitting interface. There are different scheduling methods that can be used to schedule traffic from the queues to the transmitting interface. We will look at the scheduling methods supported by the NetVanta AOS products in this module.

Layer 3 QoS - Type of Service Byte



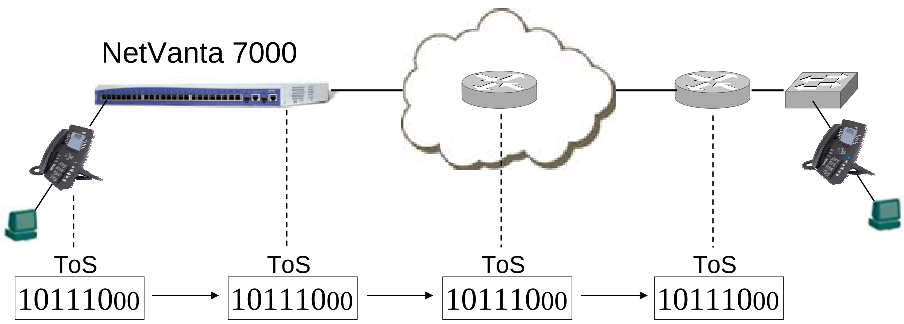
To mark, or tag traffic with different priorities Type of Service (TOS) byte in the IP packet is used. The TOS byte can be used two different ways. The traditional means of tagging the packet with a priority value was done using only three bits of the TOS byte – bits 7, 6, and 5. This method is referred to as the IP Precedence value. Using these three bits of the TOS, the IP Precedence value allows for eight levels of differentiation.

More commonly, six bits of the TOS are used to define the DiffServ, or Differentiated Services Code Point (DSCP) value. Bits 7, 6, 5, 4, 3, and 2 in the TOS field define the DSCP. The DiffServ bits allow for 64 levels of priority, but are also backward compatible with IP Precedence values.

Layer 3 QoS - Type of Service Byte

Layer 3 QoS
Type of Service Byte

- Layer 3 QoS “ToS” is End to End
  - Once ToS is written, does not change unless rewritten
- Per Hop Behavior
  - Each router must be configured to give special treatment
    - Otherwise, “Best Effort”



As shown above, layer 3 QoS is considered End to End. Once the IP Type of Service field is written, it does not change as it routes from one network to another. The only way it changes is if someone rewrites it.

Even though layer 3 QoS is considered End to End, it is still a Per Hop Behavior. In order for a packet to get special treatment, the router that the packet crosses must be configured to give this packet special treatment.

Layer 3 QoS - Type of Service Byte

Layer 3 QoS
Type of Service Byte

DSCP Values are translated to IP Precedence Values

DSCP	IP Precedence
0-7	0
8-15	1
16-23	2
24-31	3
32-39	4
40-47	5
48-55	6
56-63	7

In order for DSCP bits to be backward compatible with IP Precedence values, the DSCP ranges are mapped to corresponding IPP values. These values are known as Class-Selector per-hop Behaviors. In these per-hop behaviors, the last three bits of the DSCP value are set to zero, so only the first three bits are significant for differentiating the eight classes of service. The chart shown here indicates the values for these Class-Selector per-hop Behaviors.

DiffServ Value	DSCP	First 3 Bits (IPP)	IPP Value	Traffic Type
0	000000	000	0	Routine
8	001000	001	1	Priority
16	010000	010	2	Immediate
24	011000	011	3	Flash
32	100000	100	4	Flash Override
40	101000	101	5	Critical
48	110000	110	6	Internetwork Control
56	111000	111	7	Network Control

IP ToS Byte and IP Precedence

IP Precedence	Bits	Class Name	ToS Decimal	Byte Value
0	000	Routine	0	(0x00)
1	001	Priority	32	(0x20)
2	010	Immediate	64	(0x40)
3	011	Flash	96	(0x60)
4	100	Flash Override	128	(0x80)
5	101	Critical	160	(0xA0)
6	110	Internetwork Ctl	192	(0xC0)
7	111	Network Control	224	(0xE0)

The IP Precedence values provide network routers with information about what kind of traffic is contained in the IP packet. Based on the IP Precedence values, some networks (when supported) can offer special handling to certain packets. In addition, providing IP Precedence values to critical traffic (such as route information) ensures that critical packets will always be delivered regardless of network congestion. This traffic is often critical to network and internetwork operation. In general, the higher the IP Precedence value, the more important the traffic and the better handling it should receive in the network. It is important to remember that not all equipment in the public IP network will be configured to recognize and handle IP precedence values. While it is a good idea to set the values for critical traffic, it does not guarantee special handling. As just discussed, the IP Precedence value uses the first three high-order bits of the TOS field to define precedence values. This chart lists the IP Precedence value, the TOS bits and class name of the priority value.

DSCP Classes

DSCP Class Name	Binary Value	Decimal Value
BE (Best Effort)	000000	0
AF11 (Assured Forwarding) (RFC 2597)	001010	10
AF12	001100	12
AF13	001110	14
AF21	010010	18
AF22	010100	20
AF23	010110	22
AF31	011010	26
AF32	011100	28
AF33	011110	30
AF41	100010	34
AF42	100100	36
AF43	100110	38
EF (Expedited Forwarding) (RFC 2598)	101110	46

Assured Forwarding PHB

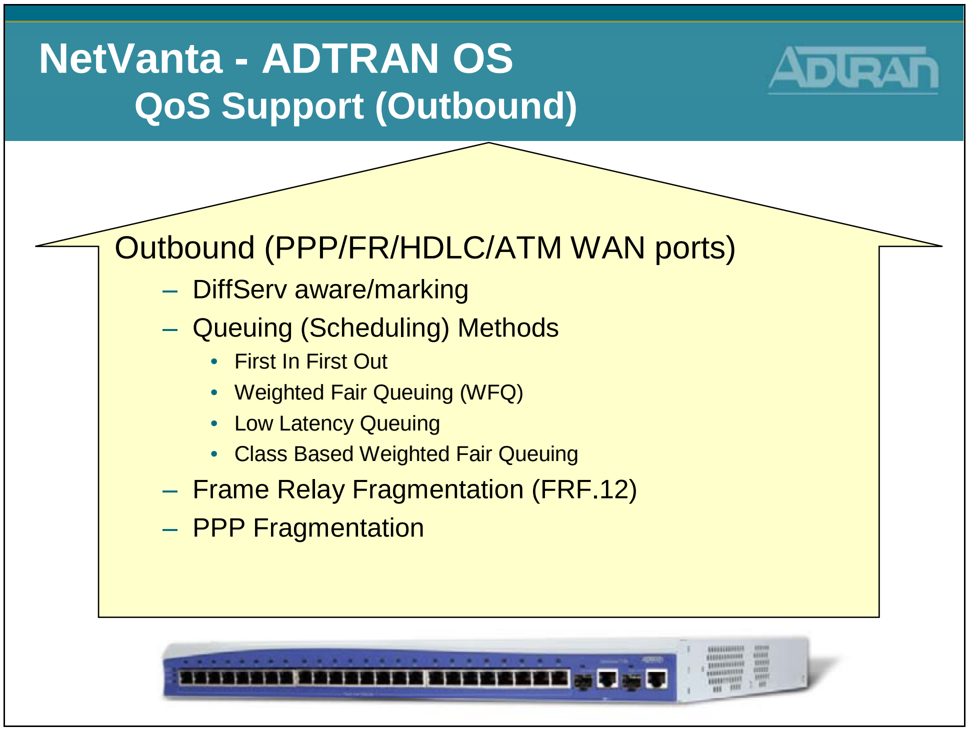
The flexibility of DiffServ allows for more developed sub-classes of service within each main class using the last three bits of the DSCP. As defined in RFC2597, the Assured Forwarding PHB creates four main classes of service: AF1, AF2, AF3, AF4

The first three bits of the DSCP specify the class and the last bit is always zero. Each class is separated into subclasses using the two remaining bits in the DSCP (bits 3 and 4). The subclasses are divided based on the likelihood that packets in the class are dropped in the event of network congestion. The higher the value for bits 3 and 4, the greater the likelihood that the packets will be dropped. The following table lists the Assured Forwarding PHB subclasses and their corresponding DSCP bits and values.

Expedited Forwarding PHB

RFC2598 created a new DiffServ PHB intended to provide the best service possible on an IP network. Packets using the Expedited Forwarding PHB markings should be provided service to reduce latency, jitter, dropped packets, and be guaranteed bandwidth during the entire end-to-end transmission journey through the network. The DSCP value for the Expedited Forwarding PHB is 46 (DSCP bits are 101110).

ADTRAN OS - QoS Support (Outbound)



Outbound QoS occurs in the AOS devices on WAN interfaces (i.e. PPP, Frame Relay) when there is congestion on the interface. The equipment recognizes IPP or DSCP values that are already marked, or the device may also tag the traffic.

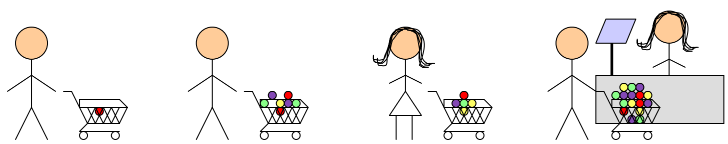
Once traffic is tagged, it is scheduled using one of several queuing methods. The AOS devices support First In First Out (FIFO), Weighted Fair Queuing (WFQ), Class-Based Weighted Fair Queuing (CBWFQ), or Low Latency Queuing (LLQ). Frame Relay Fragmentation (FRF.12) and PPP Fragmentation are also supported. We will discuss these queuing methods in more detail over the next few slides.

Layer 3 Queuing Methods - First In First Out

Layer 3 Queuing Methods First In First Out

First In First Out (FIFO)

- Most common and simplest to implement
- Packets are transmitted in the order they are placed in the queue
- Works best in situations where the ingress and egress ports are similarly matched in speed
- Not adequate for time sensitive traffic



Waiting in line on a first come, first served basis is similar to how FIFO works. When you go to the grocery store and are ready to check out, you wait in line to be processed by the cashier. The cashier will process each person in line based on the order in which they arrived. It doesn't matter how many groceries you have in your shopping cart or how much of a hurry you are in. You must wait until customers in front of you are processed first.

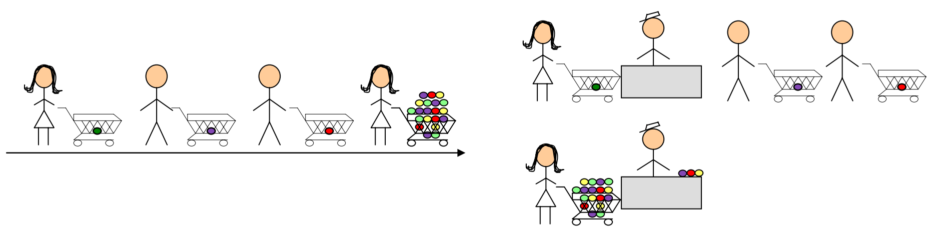
First In First Out (FIFO) queuing is familiar to almost everyone. This method is what we are used to in everyday life. Consider a single line at the grocery store. When you go to the grocery store and are ready to check out, you wait in line to be processed by the cashier. The cashier will process each person in line based on the order in which they arrived. It does not matter how many groceries you have in your shopping cart or how much of a hurry you are in. You must wait until customers in front of you are processed first. Waiting in line on a first come, first served basis is similar to how FIFO works. Packets are transmitted simply in the order they are placed in the queue. This method works best in situations where the ingress and egress ports are similarly matched in speed, but it is not adequate for time sensitive traffic, such as Voice over IP.

Layer 3 Queuing Methods - Weighted Fair Queuing

Layer 3 Queuing Methods
Weighted Fair Queuing

Weighted Fair Queuing (WFQ)

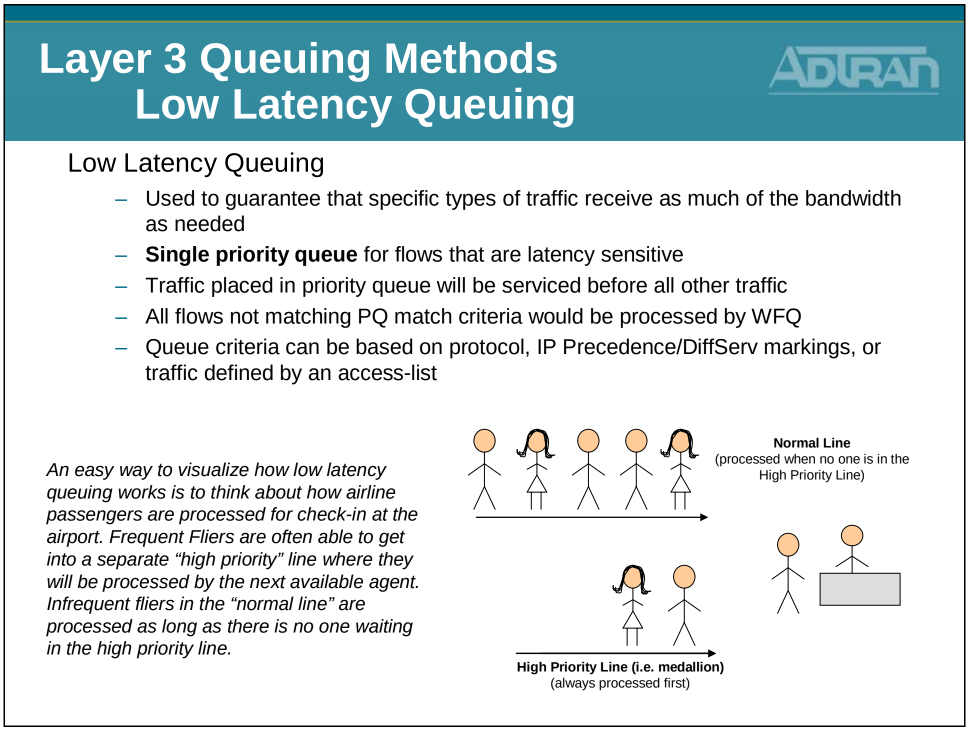
- Enabled by default on WAN interfaces with speed E1 or less
- WFQ uses a number of individual queues, one for each flow or conversation
- Up to 256 conversation queues
- Conversations determined by hash of src/dest IP address, ports, protocol type, and IP Precedence value
- Each flow or traffic class is assigned weight based on IP Precedence
- Provides priority among unequally weighted flows
- Prevents small volume, interactive traffic such as Telnet from being starved out by high volume traffic such as FTP



Another queuing type supported by the AOS devices is Weighted Fair Queuing (WFQ). WFQ is the default queuing method on WAN interfaces with a speed of E1 or less. WFQ uses queues for each conversation flow, and there can be up to 256 conversation queues on the single WAN interface. Conversations are determined by a combination of the source/destination IP address, ports, protocol type, and IP Precedence value. Each conversation flow is then assigned a weight based on IP Precedence to ensure priority. Traffic marked with a higher IPP value, or interactive traffic will be given more weight or 'priority' when waiting to get out the WAN interface. For example, interactive traffic such as Telnet would be given priority over high volume traffic such as FTP. Going back to our grocery line example, this is similar to having a 10 items or less express lane. If someone has only a few items that will be quick to process, they can go to the express lane. If they have many items that may take a little longer to process, the customer goes to the regular line.

The differentiating factor here is that both lines can be processed simultaneously, so the people with few items no longer have to wait in the same line as those with a lot of items.

Layer 3 Queuing Methods - Low Latency Queuing



While Weighted Fair Queuing processes multiple lines at the same time, Low Latency Queuing guarantees that as long as there are people in the priority line, no other lines will be processed. In other words, Low Latency Queuing reserves a single queue for priority traffic and low latency traffic is placed in that queue. This queue is then always serviced before other queues. This guarantees that specific types of traffic receive as much of the bandwidth as needed. All other traffic that does not match the priority queue criteria is processed via WFQ. Queue criteria can be configured based on protocol, IP Precedence values, DiffServ markings, or traffic defined by an access-list.

An easy way to visualize how low latency queuing works is to think about how airline passengers are processed for check-in at the airport. Frequent Fliers are often able to get into a separate 'high priority' line where the next available agent will process them. Infrequent fliers in the 'normal line' are only processed as long as there is no one waiting in the high priority line.

Layer 3 Queuing Methods - Class Based WFQ

Layer 3 Queuing Methods
Class Based WFQ

- Class Based Weighted Fair Queuing
  - Used to guarantee that specific types of traffic receive as much of the bandwidth as needed
  - Single priority queue which is serviced first for flows that are latency sensitive, as previously described with LLQ
  - Up to four bandwidth queues that reserve interface bandwidth for other types of traffic
  - Bandwidth queues are serviced after the priority queue
  - Traffic not in the priority queue or the bandwidth queues is serviced by WFQ

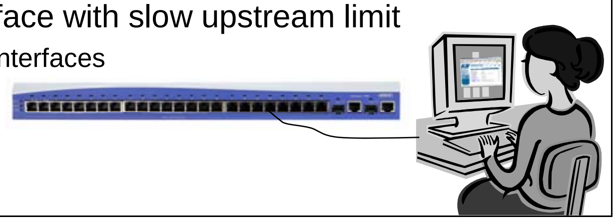
Finally, Class-Based Weighted Fair Queuing (CBWFQ) combines some of the attributes of Low Latency Queuing (LLQ) and regular Weighted Fair Queuing (WFQ) to provide priority traffic as much bandwidth as needed, assign bandwidth to other classes of traffic, and process remaining traffic using Weighted Fair Queuing. A single priority queue is used for latency sensitive traffic, which is serviced first as previously described with LLQ. Up to four bandwidth queues may also be configured that reserve interface bandwidth for other types of traffic that are grouped into 'classes' by the user. These bandwidth queues are serviced after the priority queue, and finally, traffic not in the priority queue or the bandwidth queues is serviced by WFQ. Next we will look at configuration parameters for each of these queuing methods.

NetVanta 7100 - Layer 3 QoS Configuration

NetVanta 7100 Layer 3 QoS Configuration

Basic Configuration Steps

1. Configure Bandwidth (Assumed)
2. Choose a Queue Type
   - Weighted Fair Queuing
   - Low Latency Queuing (QoS Maps)
   - Class Based Weighted Fair Queuing (QoS Maps)
3. Fragment links less than 768Kbps
   - PPP or Frame Relay
4. Rate Limit on interface with slow upstream limit
   - Ethernet or VLAN interfaces



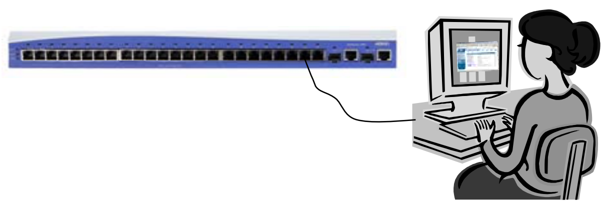
Layer 3 queuing can be configured in three general steps. The first step is to configure bandwidth values on affected interfaces. This is an informational parameter that is used in cost calculations by the queuing algorithms. Bandwidth is configured at interface configuration mode. The second step is to choose a queuing method and configure parameters associated with that type of queuing. Finally, you will want to fragment any WAN interfaces with links of 768 Kbps or less to avoid delays caused by long packets.

QoS Map Configuration - Low Latency Queuing

QoS Map Configuration
Low Latency Queuing (LLQ)

Basic Configuration Steps

1. Create the QoS map
2. Specify traffic to match
3. Configure priority queue bandwidth
4. Assign QoS Map to outgoing interface

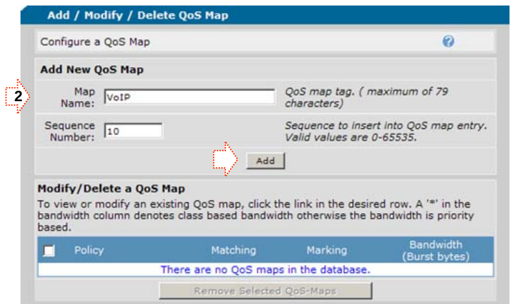


Low Latency, and Class-Based Weighted Fair Queuing require a few more configuration steps: First you will create a QoS Map. Within the QoS Map, you will define matching traffic, and then use a set command to specify an action to apply to the matching traffic. A priority command is available to configure the priority queue used in LLQ and CBWFQ, and a bandwidth statement will define bandwidth reserved for different 'classes' of traffic used in CBWFQ. Finally, you will apply the QoS Map to a WAN interface.

QoS Map Configuration

QoS Map Configuration
QoS Map Configuration

1. Select the Data / Router / QoS Maps menu

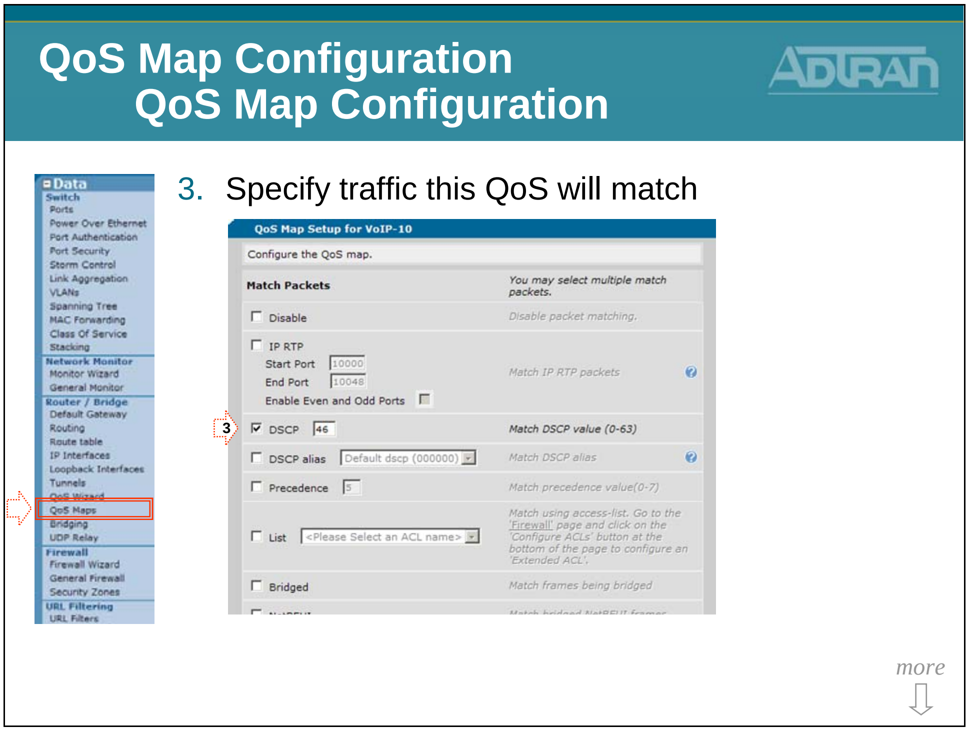


2. Type QoS map name, assign sequence number, and then click Add to create QoS map

more

The first step in Low Latency or Class-Based Weighted Fair Queuing configuration is to create a QoS map. A QoS map is a named list with sequenced entries. An entry contains a single match reference and one or more actions. The actions are then performed on traffic matching the QoS policy criteria. Multiple map entries for the QoS map are differentiated by sequence number, but the sequence number is also used to assign match order. The router searches maps with the lowest number first. Once created, a QoS map must be applied to an interface in order to actively process traffic.

QoS Map Configuration



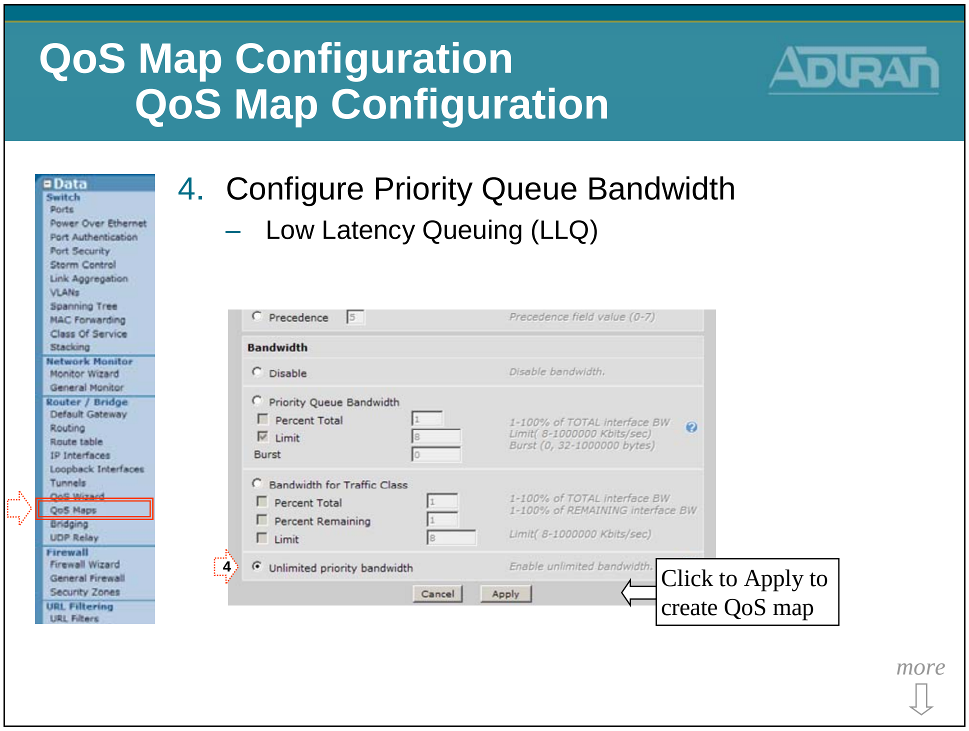
QoS policies contain at least one match reference and one or more action items (using the **priority**, **bandwidth**, or **set** commands).

The **match** section specifies the criteria used when determining whether incoming traffic is a candidate for the QoS policy action items. Multiple **match** statements can exist within the same QoS policy, allowing a single QoS policy to service various types of traffic. Use the **Match Packets** section to specify which traffic should be processed by this QoS map.

Possible Match selections:

> **dscp** <*value*>
> **ip rtp <*port #*>**
> **ip rtp <*begin port range*> <*end port range*>**
> **ip rtp <*begin port range*> <*end port range*> all**
> **Access Control List**
> **precedence <*value*>**
> **protocol bridge**
> **protocol bridge netbeui**

QoS Map Configuration



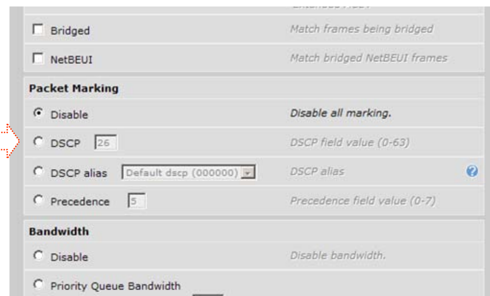
To enable Low Latency Queuing (LLQ) the **priority** option is used to provide a high-priority queue, prioritizing this traffic above all others. If no traffic is present in any other queue, priority traffic is allowed to burst up to the interface rate; otherwise, priority traffic above the specified bandwidth is dropped.

The priority queue is intended for constant bit rate traffic such as voice, due to the rate limiting. The sum of the bandwidths reserved by priority commands for all entries of a QoS map cannot exceed the **max-reserved-bandwidth** rate specified for the interfaces that the map is applied to.

QoS Map Configuration

QoS Map Configuration
QoS Map Configuration

- Optional – Configure DSCP or IP Precedence Packet Marking



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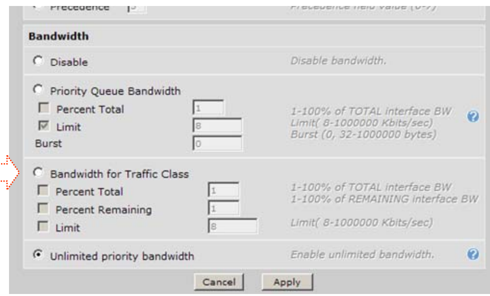
When traffic matches the configured criteria, you may specify an action to be performed on that traffic. If traffic matched is not already marked with a DSCP or IPP value, use Packet Marking to mark the packet a DSCP value (0-63) or an IP precedence value (0-7) before packet leaves the router interface.

QoS Map Configuration

QoS Map Configuration
QoS Map Configuration

- Optional – Configure Bandwidth for Traffic Class
  - Class Based Weighted Fair Queue



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When configuring CBWFQ, the **bandwidth** option is used to specify bandwidth allocation for individual traffic classes.

Options include:

Percent Total
Allocates a minimum bandwidth for a traffic class, specifying the minimum as a percentage of the total interface bandwidth.

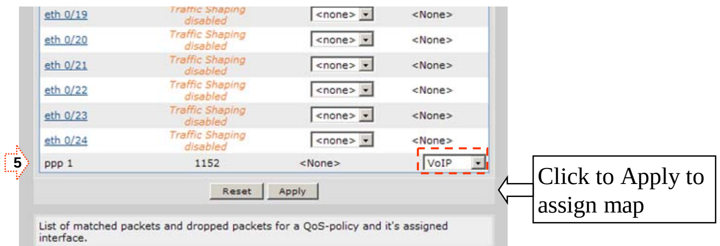
Percent remaining
Allocates a minimum bandwidth for a traffic class, specifying the minimum, as a percentage of the total interface bandwidth not allocated to priority classes in the QoS map.

Limt
Allocates the minimum bandwidth for a traffic class, specifying the minimum as an absolute bandwidth in kilobits per second. Range is 8 to 2,000,000 Kbps.

QoS Map Configuration

QoS Map Configuration
QoS Map Configuration

5. Assign Outbound QoS-policy to the interface



– The QoS Map is assigned to the outgoing transmitting router interface

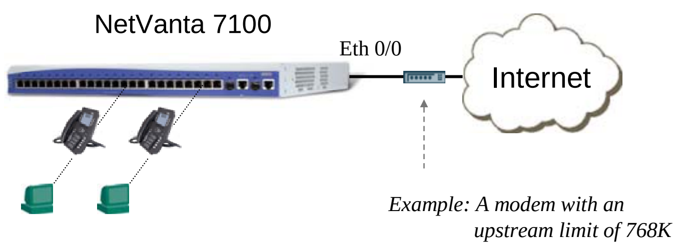
Once created, a QoS map must be applied to an interface in order to actively process traffic. Any traffic for the interface that is not sent to the priority queue is sent using the default queuing method for the interface (such as weighted fair queuing).

Note: A QoS map can not be applied to a router Ethernet or VLAN interface until Traffic Shaping is enabled on that interface.

QoS Map Configuration - Rate Limiting

QoS Map Configuration
Rate Limiting

- The WAN connection may be an Ethernet connection to a broadband modem
- It may be necessary to rate limit the Ethernet or VLAN interface to the upstream bandwidth
- This does not affect downstream bandwidth

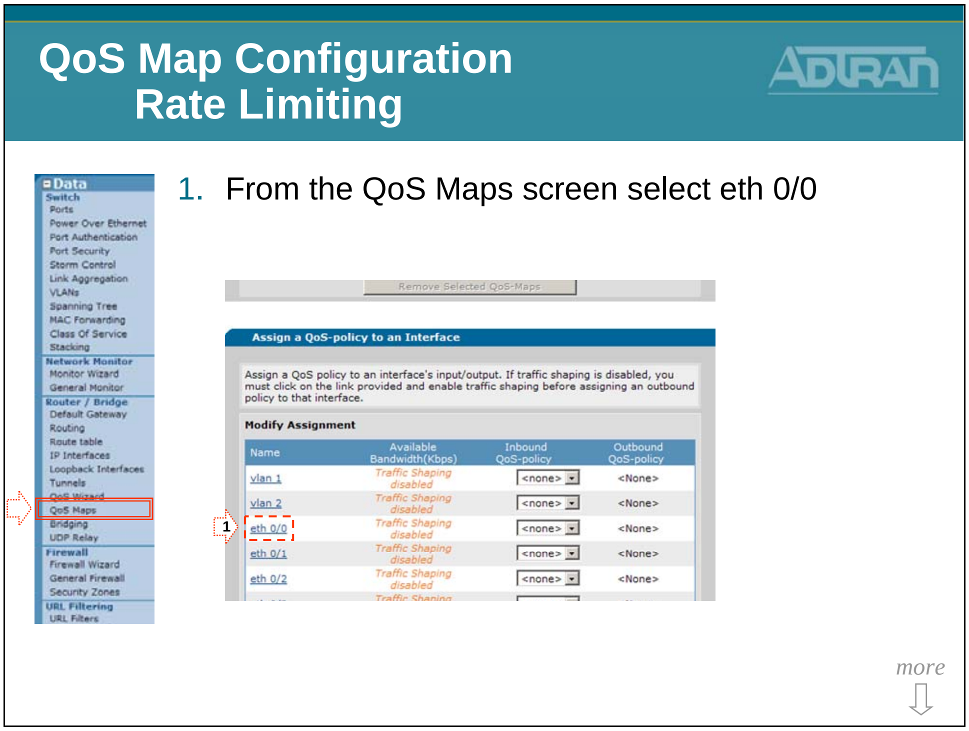


The WAN connection may be an Ethernet connection to a broadband modem. Traffic shaping can be used to limit an Ethernet segment to a particular rate or to specify use of QoS on Ethernet or VLAN interfaces. The **traffic-shape rate** command allows traffic to be limited on upstream, or outbound traffic only. This command does not affect downstream bandwidth. The value specified is the outbound rate of bits per second. By default, traffic-shaping is disabled. Variations of this command include:

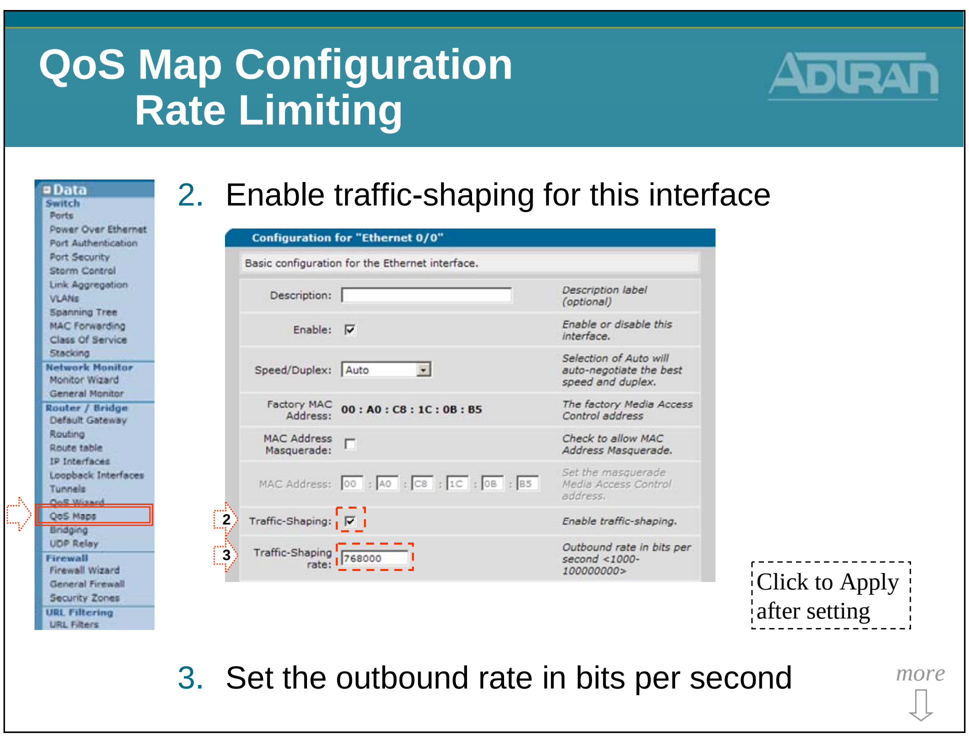
Rate Limiting - Basic Configuration Steps

1. Edit the Ethernet or VLAN interface
2. Enable traffic-shaping
3. Set the outbound rate
4. Assign QoS Map to outgoing interface

QoS Map Configuration - Rate Limiting

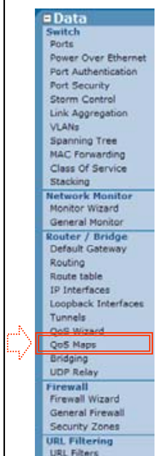


QoS Map Configuration - Rate Limiting

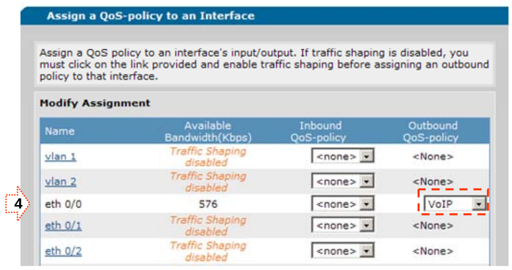


QoS Map Configuration

QoS Map Configuration
QoS Map Configuration



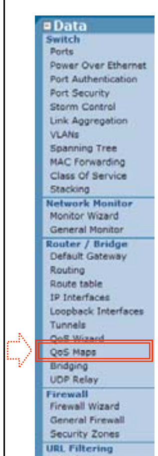
4. Assign Outbound QoS-policy to the interface



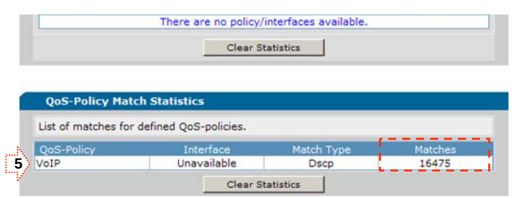
– Traffic-shaping must be enabled on an Ethernet/VLAN interface before a QoS map can be applied

QoS Map Configuration

QoS Map Configuration
QoS Map Configuration



5. Confirm specified traffic is being matched by the QoS map



– Also confirm that other traffic is not be matched by the QoS map

Basic Firewall and QoS Troubleshooting

NetVanta IP Telephony Course

Basic Firewall and QoS
Troubleshooting

show ip interfaces brief

show ip interfaces brief

- Display status of all IP interfaces

```
NV7000# show ip interfaces brief
Interface        IP Address      Status    Protocol
  eth 0/0        172.23.102.41    UP         UP
  vlan 1          10.10.10.1      UP         UP
  vlan 2          10.10.20.1      UP         UP
NV7100#
```

show ip policy-stats

show ip policy-stats

ADTRAN

- View access firewall policy statistics

NV7000# **show ip policy-stats**
Current sessions: 7
Maximum sessions: 30000
Policy-class "**Private**":
 2 current sessions (10000 max)
 Entry 1 - allow list self self
   555935 in bytes, 1813280 out bytes, 625 hits
 Entry 2 - allow list InterVLAN stateless
   0 in bytes, 0 out bytes, 0 hits
 Entry 3 - nat source list NAT interface eth 0/0 overload
   2513596 in bytes, 2481556 out bytes, 11 hits
Policy-class "**Public**":
 5 current sessions (10000 max)
 Entry 1 - allow list SIP self
   275901 in bytes, 226455 out bytes, 50 hits
 Entry 2 - allow list Admin self
   6008462 in bytes, 21224947 out bytes, 8329 hits
   :

View number of hits per policy

* Partial output displayed

show ip policy-sessions

show ip policy-sessions

ADTRAN

- View current policy-class associations

NV7000# **show ip policy-sessions**

Src IP Address	Src Port	Dest IP Address	Dst Port	NAT IP Address	NAT Port
Policy class "Private":					
udp (45) -> Public					
10.10.20.2	3000	172.23.102.42	50024	s 172.23.102.41	50020
udp (45) -> Public					
10.10.20.3	2227	172.23.102.42	50023	s 172.23.102.41	50019
Policy class "Public":					
udp (45) -> Private					
172.23.102.42	50025	172.23.102.41	50021	d 10.10.20.2	3001
udp (45) -> Private					
172.23.102.42	50022	172.23.102.41	50018	d 10.10.20.3	2226

* Partial output displayed

reload in command

reload in command

- Reload after a time interval
  - *Useful with remote configuration (may help get you out of a jam)*

```
NV7000# reload in ?
<input>              - Delay before reload (mmm or hhh:mm)

NV7100# reload in 15
Save System Configuration? [y/n] n
Reload scheduled in 15 minutes
You are about to reboot the system. Continue?[y/n] y

2009.07.05 15:52:06 OPERATING_SYSTEM System reboot scheduled in 15 minutes!

NV7100# reload cancel

******RELOAD CANCELLED******

2009.07.05 15:59:41 OPERATING_SYSTEM Scheduled system reboot cancelled.
```

show qos map

show qos map

- Display QoS Map Statistics

```
NV7000# show qos map

 qos map VoIP
   map entry 10
     match IP packets with a dscp value of 46
     priority bandwidth: 50 (% of total)  burst: default
     packets matched by map: 68372

   map entry 20
     match IP packets with a dscp value of 26
     class bandwidth:  10 (% of remaining)
     packets matched by map: 78

   Interfaces using qos map VoIP:
     eth 0/0:Output (enabled)

NV7100#
```

View packets matched per entry

show qos map interface

show qos map int eth 0/0

- Display QoS Map Statistics for specific interface

```
NV7000# show qos map interface ethernet 0/0
eth 0/0
  qos-policy out: VoIP
   map entry 10
     match IP packets with a dscp value of 46
     priority bandwidth: 50 (% of total)
     burst budget 9364/9600 bytes (current/max)
     packets matched on interface: 81158
     packets dropped: 0
   map entry 20
     match IP packets with a dscp value of 26
     class bandwidth:  10 (% of remaining)
     conversation: 233
     packets matched on interface: 0
     packets dropped: 0
```

- View packets matched per entry
- Check drop status

Module Summary

Module Summary

- At the end of this module, you should be able to:
- Create VLAN interfaces
- Configure Firewall policies
- Create a network DMZ
- Understand Quality of Service concepts
- Configure QoS Maps
- Perform basic Firewall and QoS troubleshooting

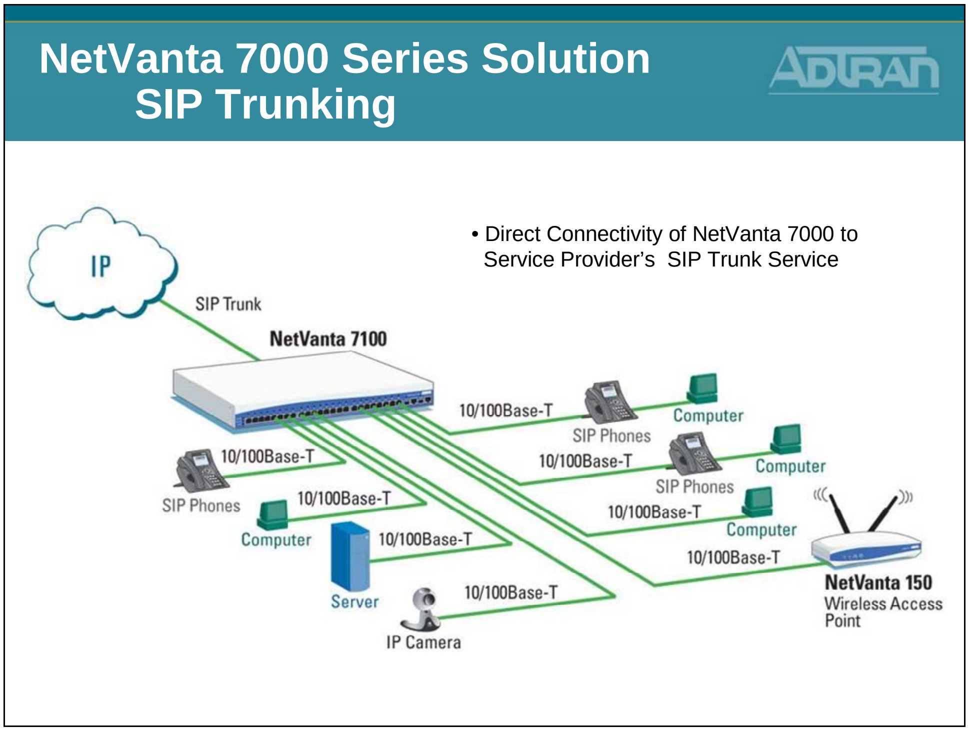
Module 8: NetVanta 7000 Remote Telephony Applications

Module Objectives

Module Objectives

- Introduce NetVanta 7000 Remote Telephony Applications
- Configure Service Provider SIP Trunk
- Configure SIP Networking between Sites
- Preview Remote User Configuration
- Introduce VoIP Quality Monitoring (VQM)
- Conduct Voice Troubleshooting in a NetVanta 7000 Remote Telephony Application

SIP Trunking



For businesses that are looking for ways to reduce costs, ADTRAN's SIP Trunking is an ideal solution. SIP trunking is a packet-based service which will dynamically consolidate all voice and data traffic over a single IP circuit and enables the SIP Service Provider to carry local, domestic and international long distance, and toll free calls, in addition to video, email, Internet, and other data. The combination of ADTRAN's NetVanta 7000 Series IP PBX and the service provider's SIP trunk offers a proven solution for not only reducing immediate costs, but also ongoing savings up to 40% each month.

- Integrates Multiple Functions into Single Solution including PBX, Switch, Router, Firewall/VPN functions
- Provides Key System Functionality across SIP Trunking such as Busy Lamp Field (BLF) and Share Line Appearances (SLA)
- Built in Quality of Service for Voice to monitor and report VoIP performance statistics.

ADTRAN's SIP Trunking alliances offer proven ways to consolidate voice and data onto a converged IP service that lowers costs and achieves high quality reliable service - all backed by industry leading service and support.

SIP Trunking Overview

SIP is the industry standard ASCII-based peer to peer signaling protocol responsible for the initiation and management of IP voice communication sessions. SIP is designed to control call setup and tear down between IP endpoint devices. The basic function of SIP is to locate endpoints, signal a desire to communicate, establish sessions, and tear down sessions between endpoints. The current version of SIP (2.0) is defined in RFC 3261.

SIP Trunks Overview

Voice over IP (VoIP) rapidly gained popularity due to the cost savings achieved by simultaneously routing voice calls and data over the same network, eliminating the need for separate voice and data circuits at customer premises. The common method of combining voice and data together on one circuit is PRI. PRI carries voice traffic over the dedicated channels with the data channels, and routes or terminates the voice traffic between two PRI-compatible private branch exchanges (PBXs) or key systems. The more advanced alternative to a PRI trunk is a SIP trunk.

SIP trunking is a packet-based voice service that routes calls over an IP network to an IP-compatible PBX or voice switch using SIP signaling to place and receive calls. The typical SIP trunk service provider offers extensive cost savings, compared to conventional trunk services. The IP connection to the provider carries all traffic, such as local, long distance, and toll free calls, video, email, Internet, data, and other media over a single circuit. Calls into public switched telephone network (PSTN) are also handled by the SIP service provider by passing the calls off to a media gateway that connects to the PSTN for users not using VoIP service.

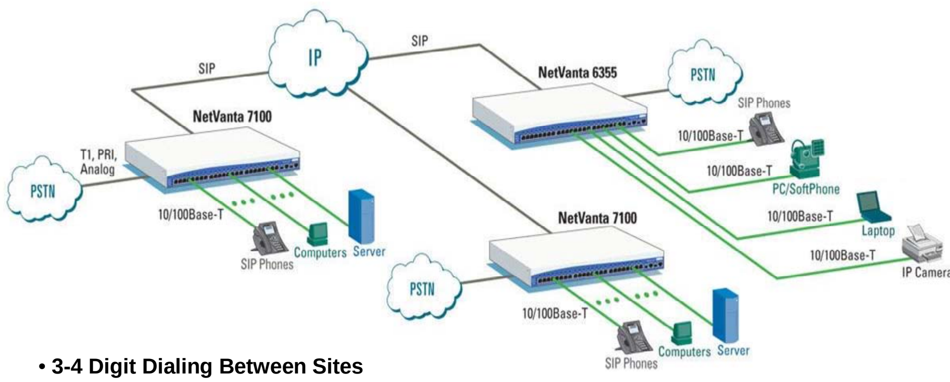
AOS SIP networking is an interconnection of NetVanta 7000 Series units or ADTRAN IP Business Gateways over an IP network. The SIP networking configuration is very similar to configuring SIP trunking between a NetVanta 7000 Series unit and a service provider's SIP trunking service. The main difference is that configuring the SIP registrar is not required.

SIP Trunking Advantages

Using SIP trunks has advantages over PRI(s) such as more significant cost savings, and control over the number of channels on the trunk (SIP trunks can be purchased in increments of simultaneous calls or DIDs). When connected to an ADTRAN IP PBX device, the SIP trunk solution offers all the traditional hosted telephony features of a PRI. Reference configuration guides on compatible AOS voice features (such as source and ANI based routing (SABR), voice quality monitoring (VQM), voicemail, etc.), are available on your AOS Documentation CD shipped with your AOS unit or visit our website at **http://kb.adtran.com**.

SIP Networking

NetVanta 7000 Series Solution SIP Networking

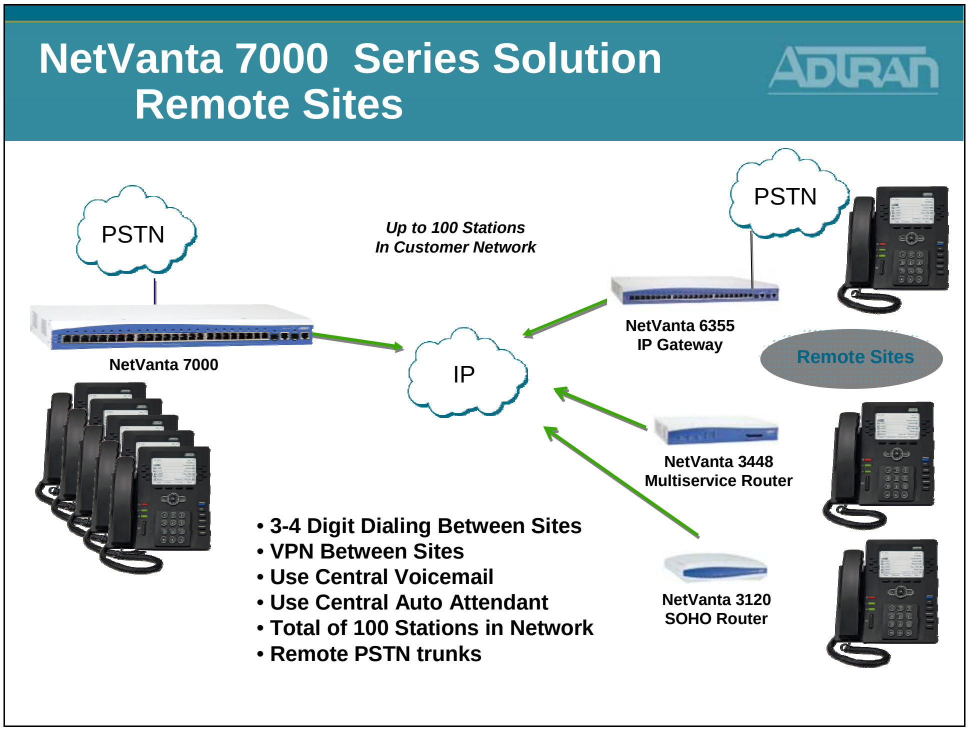


- **3-4 Digit Dialing Between Sites**
- **VPN Between Sites**
- **Toll Bypass**
- **Access Remote Site Local Trunks**

The ADTRAN NetVanta 7000 Series supports SIP Networking between multiple locations. With SIP networking, businesses can connect multiple sites and have three- to four-digit dialing, local call routing and survivability, and on-net calls for toll bypass. The NetVanta 7100 and 7060 are best for locations that need local voice mail; while ADTRAN's NetVanta 6355 IP Business Gateway and Total Access 900 Series provide the ideal solution for locations that will use a central NetVanta 7000 voice mail.

- Links multiple sites together to reduce costs
- Direct dials between offices
- Supports inter-office, three- to four-digit dialing
- Provides local PSTN access
- Allows local sites to share remote site trunks

Remote Sites



SIP trunking feature allows remote IP Business Gateways, such as the ADTRAN Total Access 900(e) Series and NetVanta 6355, to connect to a central IPT device (NetVanta 7000 Series) for the use of local trunks at each remote location. This application functions similar to a single PBX with each remote user registering back to the IPT either via transparent proxy (SIP) or directly (analog phones). The phones at the remote locations rely on the main site (IPT device) to provide voicemail and auto attendant services to incoming calls.

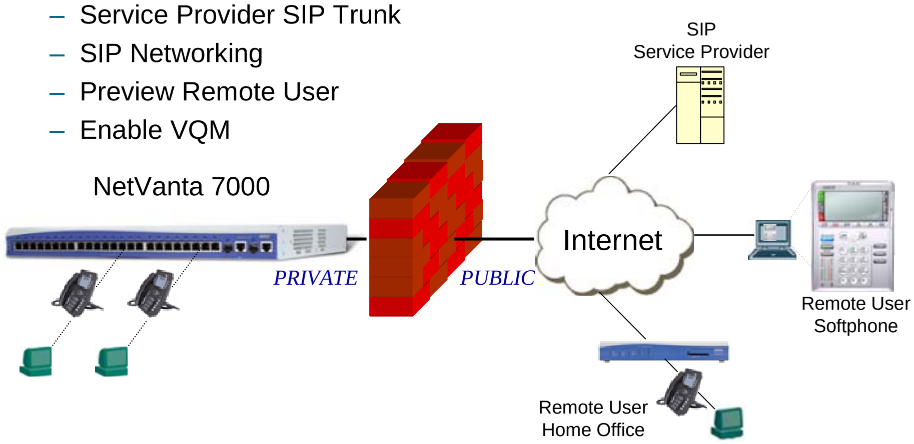
The NetVanta 7100 and NetVanta 3120 enable secure, always-on, voice, data and high-speed data access to business resources from a remote home office. Using a single cable or DSL broadband connection and secure IPSec-compliant VPN NetVanta technology, workers can have the same convenience and functionality in their home office.

- Ideal solution to extend voice/data capabilities to small, remote offices
- Enables one or more teleworkers to have same features as the main business office
- Improves teleworking productivity
- Provides phone feature transparency over IP connectivity
- Uses the same desktop phone at remote home or small offices

NetVanta 7000 - Remote Telephony Applications

NetVanta 7000
Remote Telephony Applications

- Configuration of the following Remote Telephony Applications are introduced in this section:
  - Service Provider SIP Trunk
  - SIP Networking
  - Preview Remote User
  - Enable VQM



Service Provider SIP Trunk Configuration

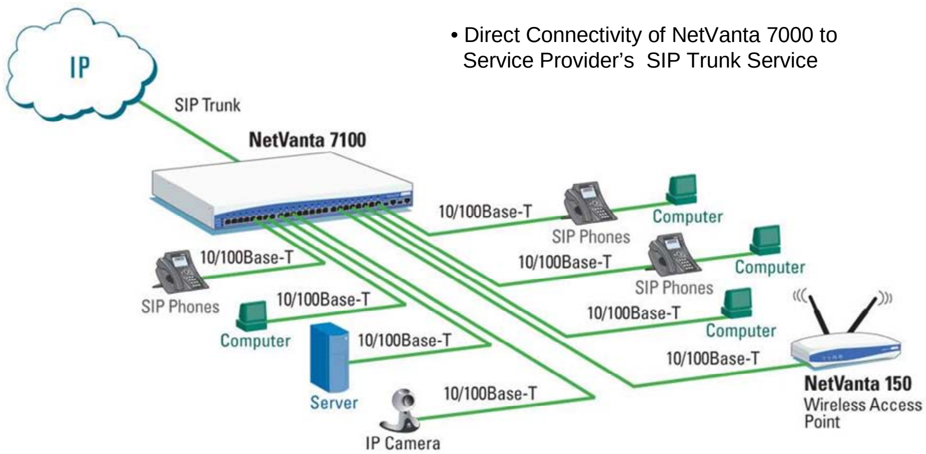
NetVanta IP Telephony Course

Service Provider SIP Trunk Configuration

Service Provider SIP Trunk

NetVanta 7000 Series Solution
Service Provider SIP Trunk

• Direct Connectivity of NetVanta 7000 to Service Provider's SIP Trunk Service

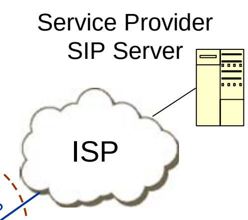


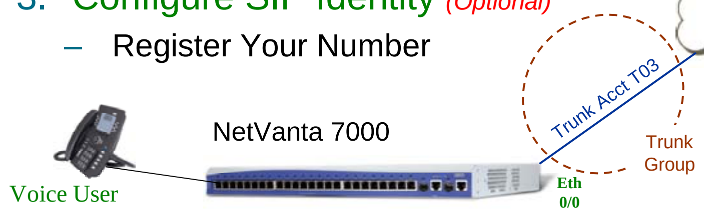
To configure an incoming SIP trunk from your service provider, verify that NetVanta 7000 Series Call Routing Mode and Transfer Mode are set to Local (feature support is provided internally by the NetVanta unit). The softswitch only has control of the call routing up to the SIP trunk interface. The ADTRAN IPT device will send and receive all basic SIP call setup messages and will accept advanced setup messages, but the REFER and INVITE with Replaces (SIP signaling methods) messages will not be sent out the trunk (T01). The incoming SIP trunk will behave similar to a PRI and all the traditionally supported call features will remain functional. The use of the SIP trunk can be controlled with other IPT features, such as SABR and least cost routing (LCR). Only one service-provider SIP trunk is allowed in this application. Precise trunk group and dial plan configuration allow users to take advantage of the LCR out of any trunk configured on the system. In the illustration, the additional trunk (T02) that is directly connected to the PSTN can be analog, T1, or PRI. T02 can be mainly used for local calls by assigning a high cost to the long distance outbound call template, or it can be used for survivability during possible failure of the main SIP trunk service.

Provider SIP Trunk - Basic Configuration Steps

Service Provider SIP Trunk Basic Configuration Steps

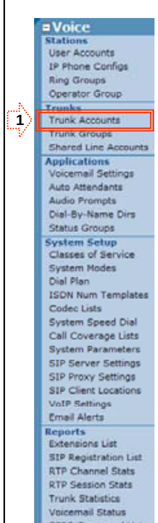
1. Create Trunk Account
   – Configure SIP and Registrar servers
   – Register Your Number *(Optional)*
2. Create Trunk Group
   – Add SIP Trunk Account
   – Define outbound call templates
3. Configure SIP Identity *(Optional)*
   – Register Your Number



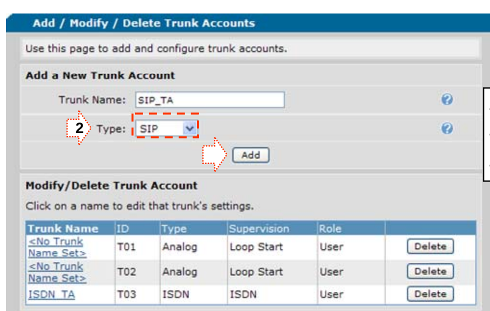


SIP Trunk Configuration - 1) Create Trunk Account

Provider SIP Trunk Configuration 1) Create Trunk Account



1. Select the Voice / Trunks / Trunk Accounts menu

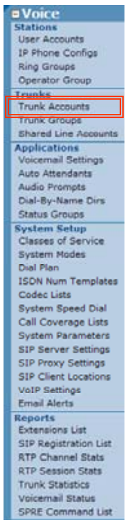


- Type Trunk Name
- Set Type to SIP
- Click Add

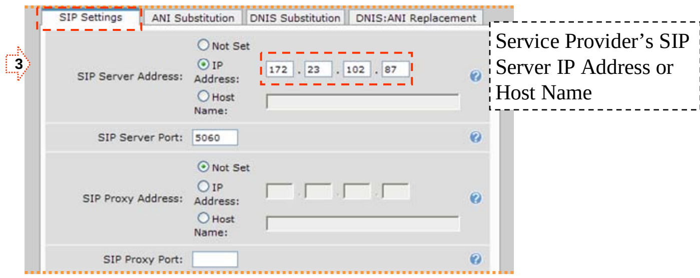
2. Create a SIP Trunk Account
   - Will be used to point to Service Provider's SIP Server

SIP Trunk Account - Define SIP Server Address

Provider SIP Trunk Configuration
1) Create Trunk Account (Cont..)



3. Define address or host name of SIP Server

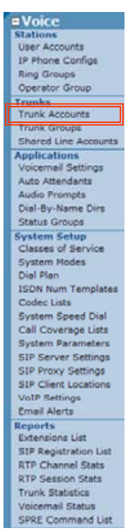


– Default SIP server Port is 5060

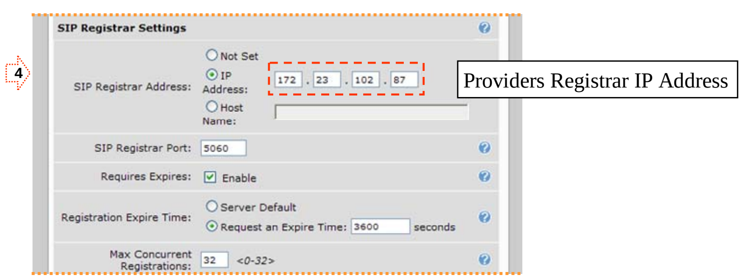
- Define SIP Proxy Server address if one is being used

SIP Trunk Account - Define SIP Registrar Address

Provider SIP Trunk Configuration
1) Create Trunk Account (Cont..)



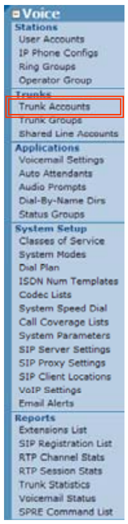
4. Define SIP Registrar address or host name



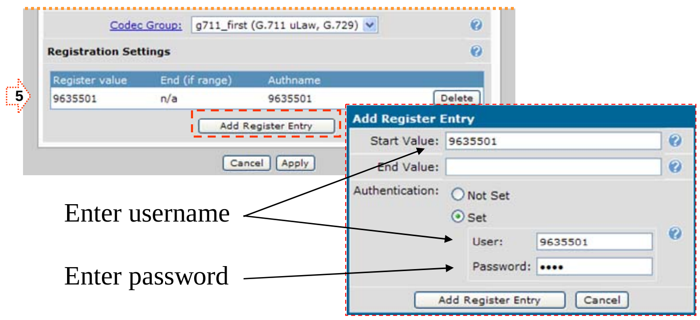
– Default SIP Registrar Port is 5060

SIP Trunk Account - Register Number

Provider SIP Trunk Configuration 1) Create Trunk Account (Cont..)



5. Register the Number provided from the SIP Service Provider
   – You should receive a username, password, and Service Provider's SIP Server address



If user's phone number does not match the SIP username entered here, an **Alias** will need to be configured matching the SIP username to the actual user extension

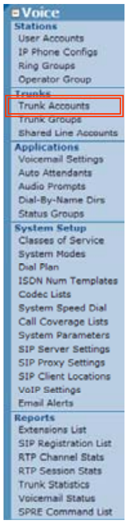
Calling Party – ANI Substitution

Calling Party - ANI Substitution

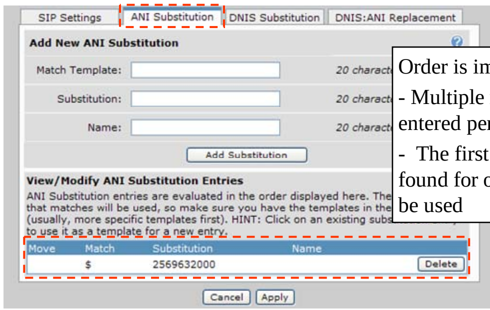
- **Problem**: Many SIP Service Providers will reject a call from an unknown number, that is, a number not registered to their switch
  – Problems can occur when a user on the NetVanta 7000 forwards their phone to an external number
  – When the user locally forwards their phone, the original calling party information will be preserved and sent in the From header of the SIP INVITE back out to the provider's softswitch
- **Solution**: Configure Calling Party (ANI) Match/Substitutions to allow forwarding of calls out a SIP Trunk
  – Create an **ANI Substitution** template to match all numbers
  – This will be used to replace the From header on forwarded calls along with all other calls routed out the SIP trunk

SIP Trunk Account – ANI Substitution

Provider SIP Trunk Configuration
1) Create Trunk Account (Cont..)



- *Optional*: Add ANI substitution

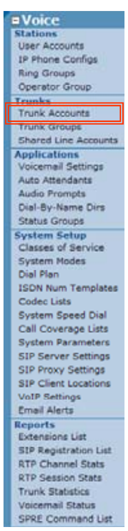


Order is important:
- Multiple match statements can be entered per trunk account
- The first valid match that is found for outbound numbers will be used

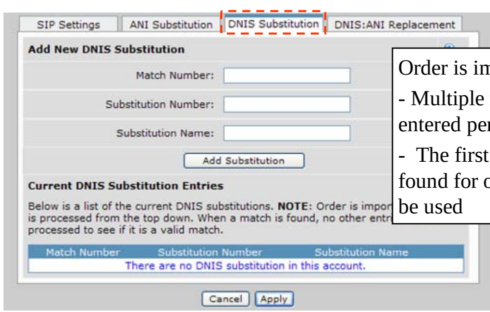
  – Examples:
    - match ani "**2XXX**" substitute "**2569632100**" name "**Shanes Cable Co**"
    - match ani "**3XXX**" substitute "**2569632200**" name "**Hunters Cable Co**"
    - match ani "**$**" substitute "**2569632000**"

SIP Trunk Account – DNIS Substitution

Provider SIP Trunk Configuration
1) Create Trunk Account (Cont..)



- *Optional*: Add DNIS substitution

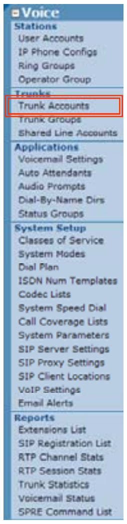


Order is important:
- Multiple match statements can be entered per trunk account
- The first valid match that is found for outbound numbers will be used

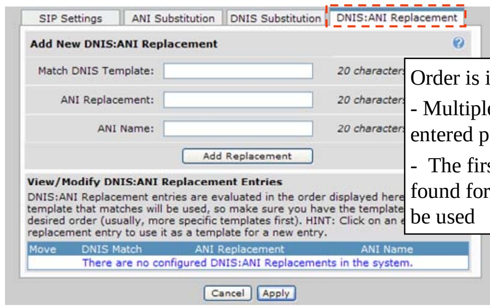
  – Examples:
    - Match: **NXX-XXXX** Subst: **256-NXX-XXXX**
    - Match: **1-256-XXX-XXXX** Subst: **NXX-XXX-XXXX**
    - Match: **1-NXX-NXX-XXXX** Subst: **10-10-220-NXX-NXX-XXXX**

SIP Trunk Account – DNIS:ANI Replacement

Provider SIP Trunk Configuration
1) Create Trunk Account (Cont..)



- *Optional*: Add DNIS:ANI Replacement



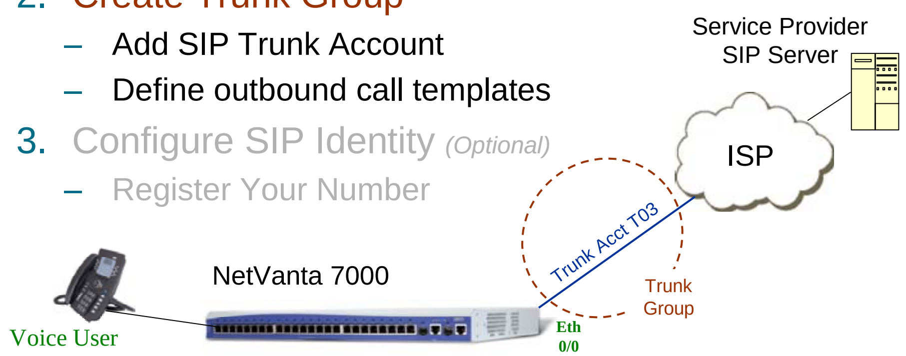
Order is important:
- Multiple match statements can be entered per trunk account
- The first valid match that is found for outbound numbers will be used

  – Examples:
    - match dnis "**1NXXNXXXXXX**" replace ani "**18884238726**" name "**National Network Co**"
    - match dnis "**NXXXXXX**" replace ani "**9638716** " name "**Huntsville Network Co**"

Provider SIP Trunk - Basic Configuration Steps

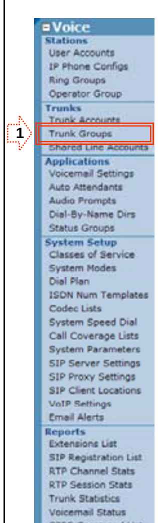
Service Provider SIP Trunk Basic Configuration Steps

1. Create Trunk Account
   - Configure SIP and registrar servers
   - Optional - Register Your Number *(Optional)*
2. Create Trunk Group
   - Add SIP Trunk Account
   - Define outbound call templates
3. Configure SIP Identity *(Optional)*
   - Register Your Number

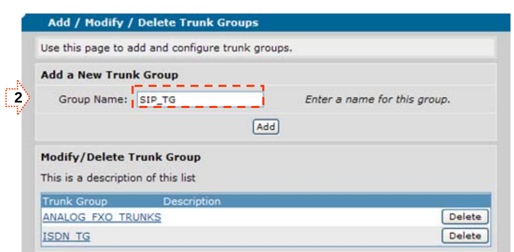


SIP Trunk Configuration - 2) Create Trunk Group

Provider SIP Trunk Configuration 2) Create Trunk Group



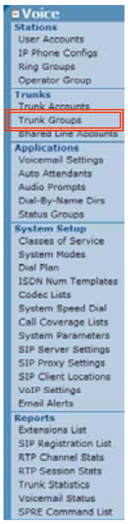
1. Select the Voice / Trunks / Trunk Groups menu



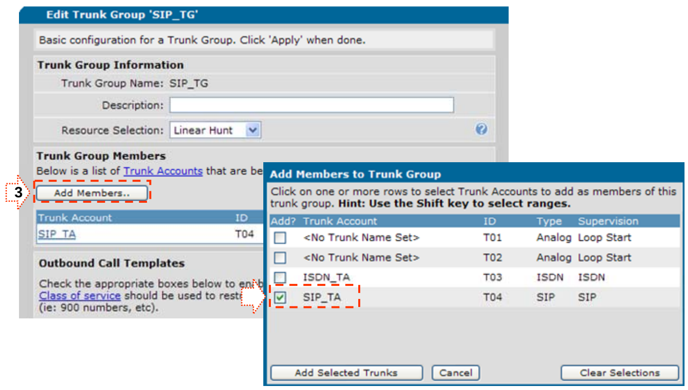
2. Create a Trunk Group
   - Point to the new SIP Trunk Account
   - Define call types allowed out this Trunk Group

Trunk Group – Add SIP Trunk Account

Provider SIP Trunk Configuration
2) Create Trunk Group

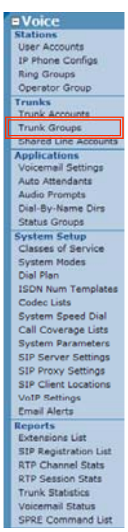


3. Click Add Members to add existing SIP Trunk Account to this Trunk Group

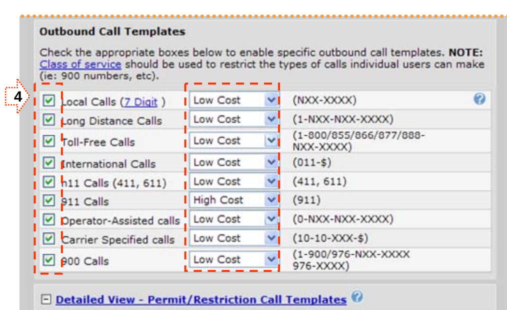


Trunk Group – Define Outbound Call Template

Provider SIP Trunk Configuration
2) Create Trunk Group

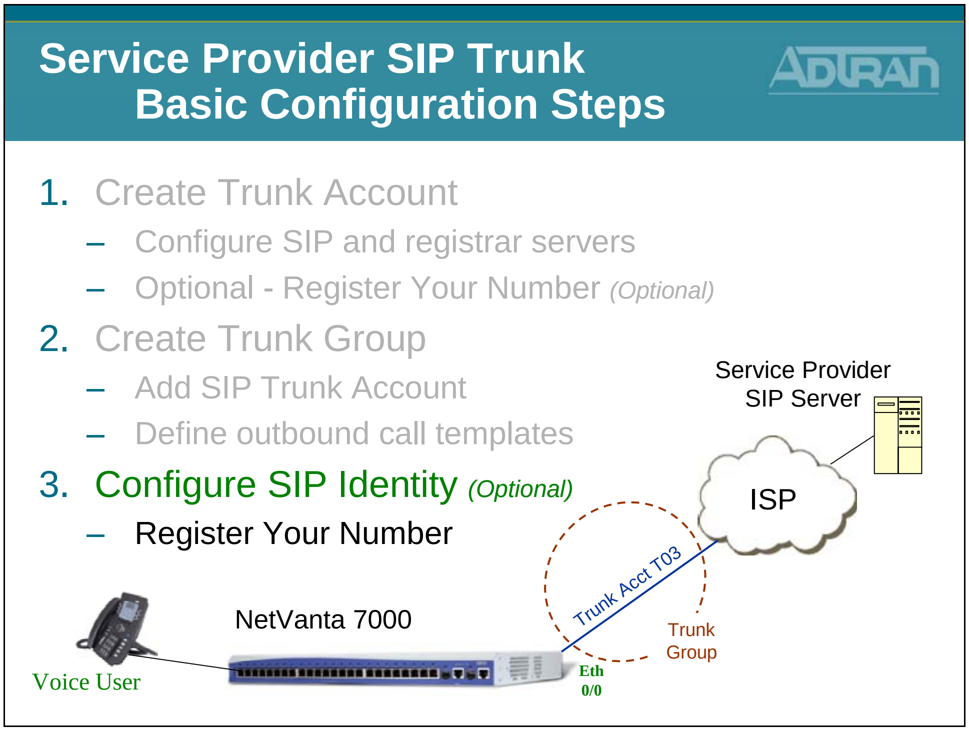


4. Outbound Call Template
   – Define call types allowed out this SIP Trunk Group

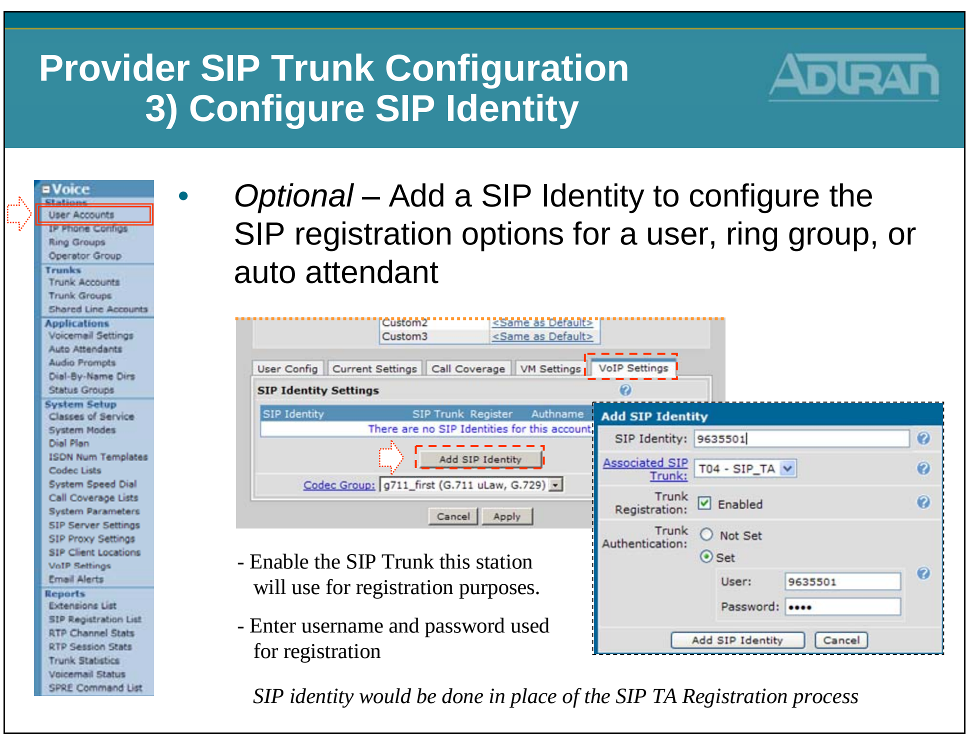


   – *Optional*: Define cost for each type of call
     - Least cost routing

Provider SIP Trunk - Basic Configuration Steps

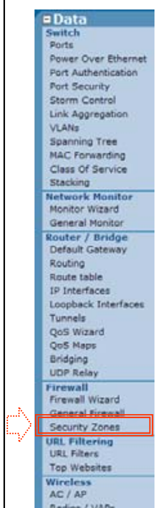


SIP Trunk Configuration - 3) Configure SIP Identity

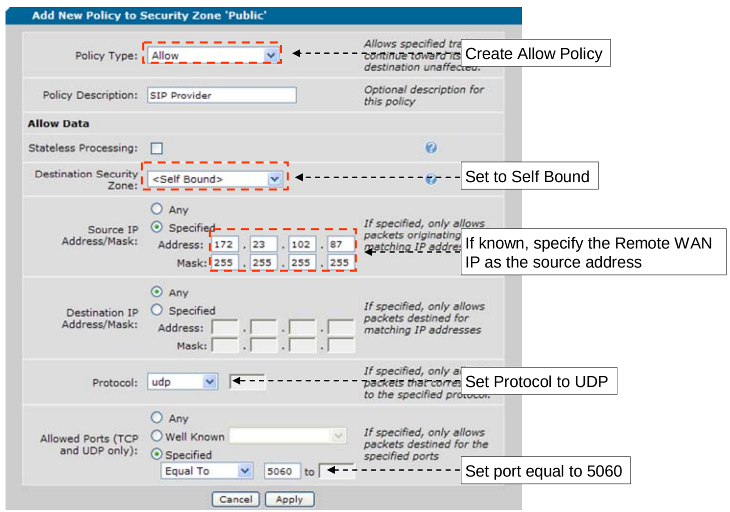


Additional VoIP Config – Allow UDP 5060

Additional VoIP Configuration Allow UDP 5060 traffic

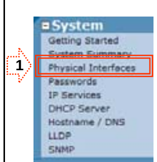


- Allow UDP traffic in **Public Security Zone** (WAN)

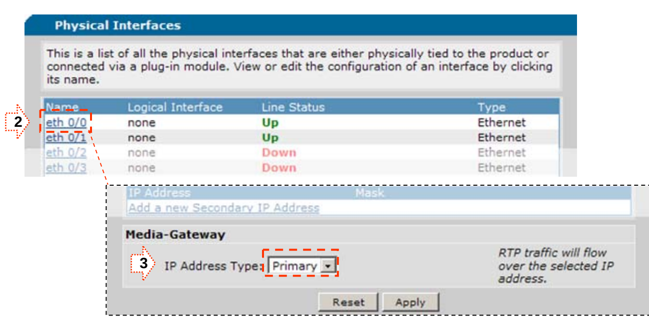


Additional VoIP Config - Eth 0/0 Media Gateway

Additional VoIP Configuration Eth 0/0 Media Gateway



- Enable WAN Eth 0/0 media-gateway as Primary

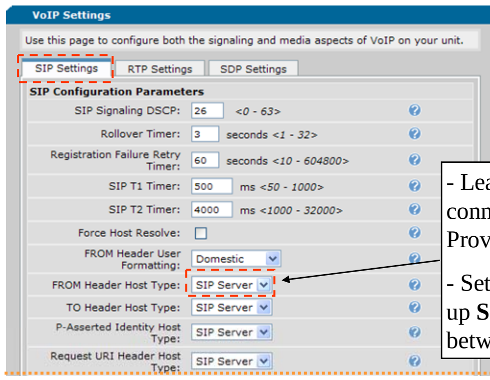


- Specifies that RTP traffic will be sourced from the Primary IP address of this interface

Additional VoIP Config - VoIP / SIP Settings

Additional VoIP Configuration VoIP Settings / SIP Settings

- Select the Voice / System Setup / VoIP Settings menu



- Leave **SIP Server** when connecting to SIP Service Provider

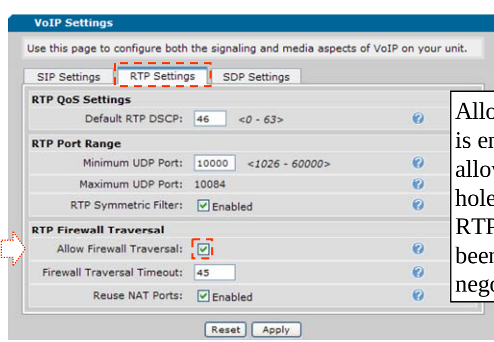
- Set to **Local** when setting up **SIP Networking** between NetVanta 7000s

  – Leave SIP From Header Host Type as **SIP Server**

Additional VoIP Config - VoIP / RTP Settings

Additional VoIP Configuration VoIP Settings / RTP Settings

- Allow RTP traffic associated with the allowed SIP traffic



Allow Firewall Traversal is enabled by default. This allows the 7000 to open holes in the firewall for RTP streams that have been created by SIP negotiation

SIP Networking Configuration

NetVanta IP Telephony Course

SIP Networking Configuration

SIP Networking Features

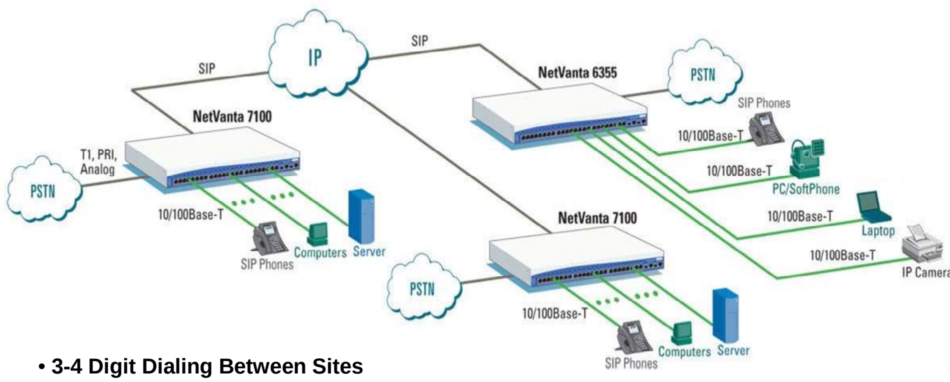
- Links multiple sites together to reduce costs
- Support for up to 10 SIP trunks
  - Remote devices or service provider
- Direct dials between offices
  - Supports inter-office, three- to four-digit dialing
  - Transfer calls between sites
- Provides local PSTN access
  - Allows local sites to share remote site trunks

- Independent Sites
  - Each Site has own Voicemail and Auto Attendant
  - Can not forward to a Mailbox (Could forward in email)

SIP Networking

NetVanta 7000 Series Solution SIP Networking

ADTRAN



- **3-4 Digit Dialing Between Sites**
- **VPN Between Sites**
- **Toll Bypass**
- **Access Remote Site Local Trunks**

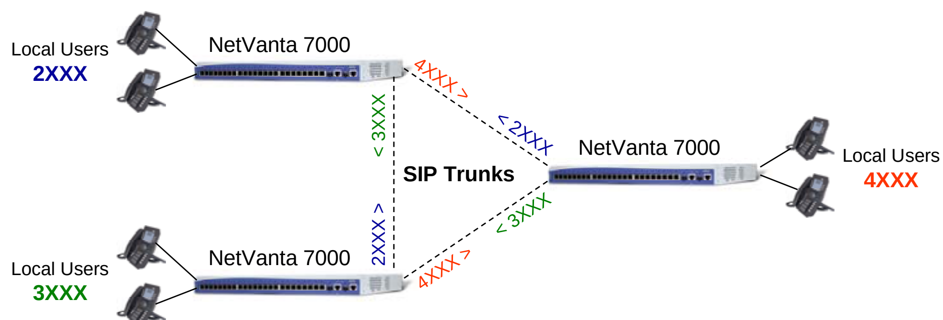
In the SIP Networking application shown above, the NetVanta 7000 Series unit at the main location is connected to remote NetVanta 7000 Series. This type of SIP networking application can support a maximum of ten remote SIP trunks at each site. Voice users connected to the NetVanta 7000 Series at Site A will be able to connect to all endpoints at all locations, including access to voicemail, auto attendant, ring groups, and other phone users. Voicemail features will not be extended across the facing SIP trunks; each IPT will have local voicemail and auto attendant services. Remote users will not be automatically entered into the system directory at remote locations. Also, remote users will not appear in the selection list boxes for Trunk Number and Ring Groups. Precise trunk group and dial plan configuration will allow users to take advantage of the LCR out of any trunk configured on the system. Each NetVanta 7000 Series Call Routing Mode, Forward Mode, and Transfer Mode must be set to Local.

Each remote site can also have a SIP trunk connection to an IP Business Gateway (Total Access 900(e) or NetVanta 6355). In the illustration, the additional trunk that is directly connected to the PSTN can be analog, T1, or PRI. This trunk can be mainly used for local calls by assigning a high cost to the long distance outbound call template, or it can be used for survivability during possible failure of the main SIP trunk service.

SIP Networking – Design Considerations

SIP Networking
Design Considerations

- **Fully Meshed** SIP Trunks
- Each site has **unique extensions**
  - 2XXX, 3XXX, 4XXX, etc…
  - 21XX, 22XX, 23XX, etc…

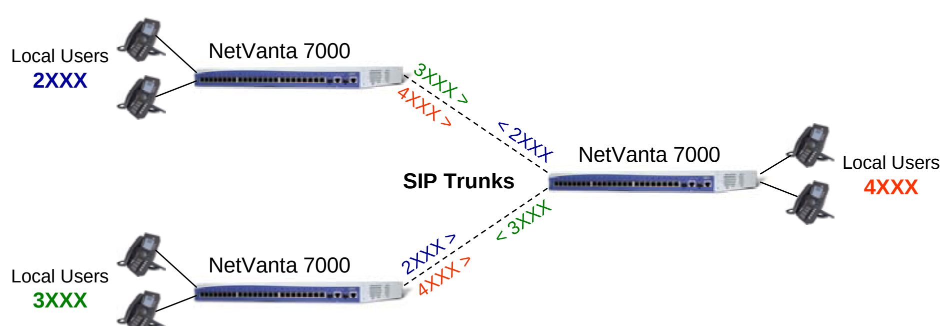


Each site has local PSTN (not shown)

SIP Networking – Design Considerations

SIP Networking
Design Considerations

- **Hub & Spoke** SIP Trunk
- Each site has **unique extensions**
  - 2XXX, 3XXX, 4XXX, etc…
  - 21XX, 22XX, 23XX, etc…

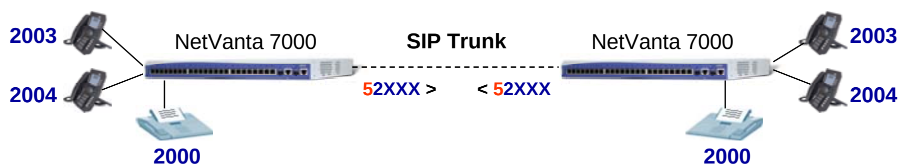


Each site has local PSTN (not shown)

SIP Networking – Design Considerations

SIP Networking
Design Considerations

- Each site has **same extensions**
  - Requires use of an **extra digit** for use as a site identifier
    - For example, **5** along with extension **2XXX** could cause the call to route out the SIP trunk to the other site (**52XXX**)
  - Dial plan at both sites needs to be modified for both the System and the IP phones
    - Extension = MXXX and 52XXX

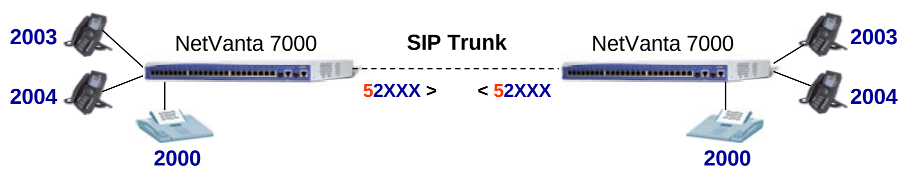


Each site has local PSTN (not shown)

SIP Networking – Basic Configuration Steps

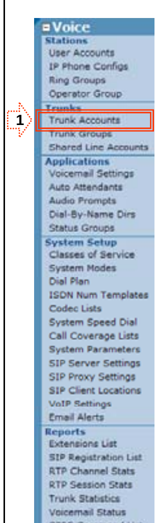
SIP Networking
Basic Configuration Steps

1. Create Trunk Account
   - Configure SIP server as other sides WAN IP
   - Configure the SIP FROM Header Host Type
   - Configure DNSI substitution (if required)
2. Create Trunk Group
   - Add SIP Trunk Account
   - Define outbound call templates for other site

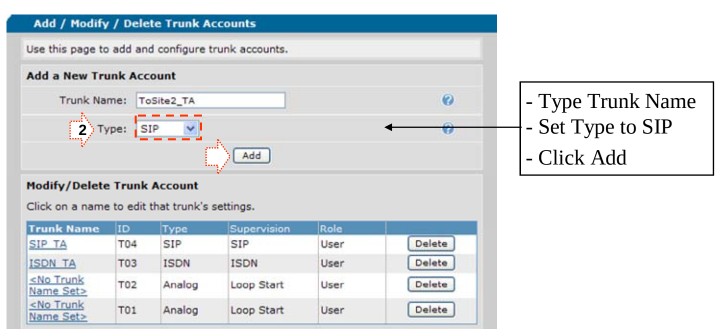


SIP Networking – 1) Create Trunk Account

SIP Networking Configuration
1) Create Trunk Account



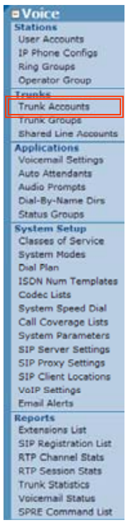
1. Select the Voice / Trunks / Trunk Accounts menu



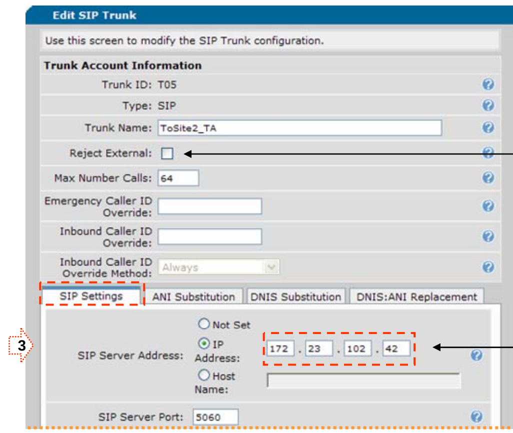
2. Create a SIP Trunk Account
   - Will be used to point to remote site WAN IP address

Trunk Account – Define Remote Site WAN IP

SIP Networking Configuration
1) Create Trunk Account (Cont..)



3. Define address Remote Site WAN IP address

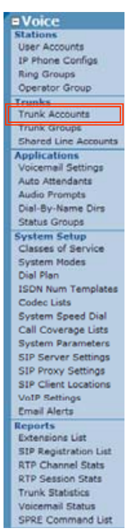


Disable **Reject External** if you want to allow Trunk to Trunk calls

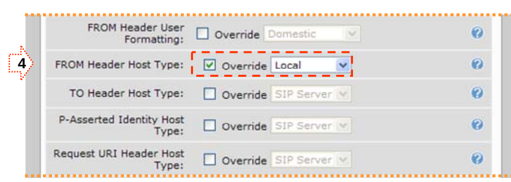
Remote WAN IP

Trunk Account – Configure FROM Header

SIP Networking Configuration
1) Create Trunk Account (Cont..)



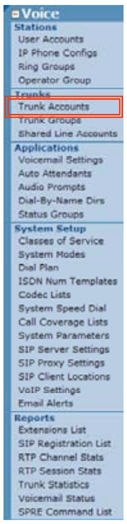
4. From the Trunk Accounts SIP settings tab, configure the FROM Header Host Type as **Local**



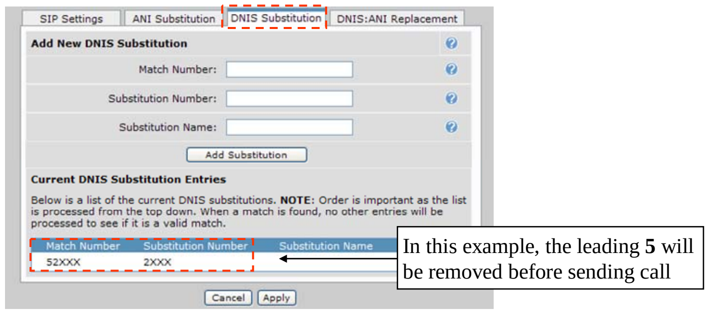
- This overrides the global default setting found on the System Setup / VoIP Settings / SIP Settings tab

Trunk Account – DNIS Substitution

SIP Networking Configuration
1) Create Trunk Account (Cont..)



- *Optional*: Add DNIS substitution

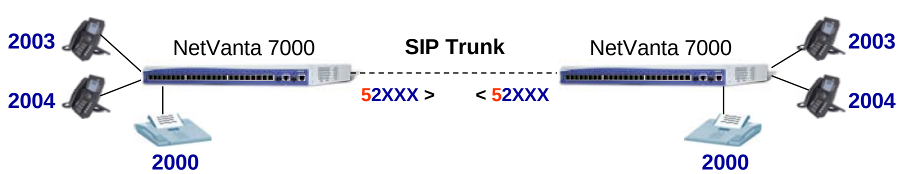


  – If both sides have the same extension, additional digits are needed to point calls out a particular trunk
    - Once the call routing decision has been made, the extra digit(s) need to be removed before sending call

SIP Networking - Basic Configuration Steps

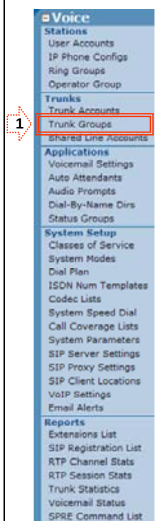
SIP Networking
Basic Configuration Steps

1. Create Trunk Account
   – Configure SIP server as other sides WAN IP
   – Configure the SIP FROM Header Host Type
   – Configure DNSI substitution (if required)
2. Create Trunk Group
   – Add SIP Trunk Account
   – Define outbound call templates for other site

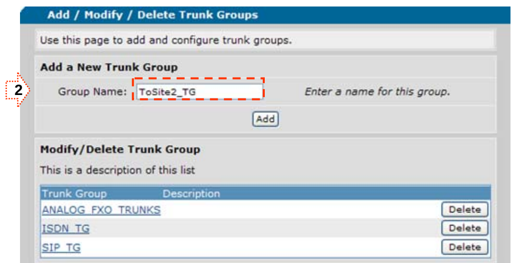


SIP Trunk Configuration - 2) Create Trunk Group

SIP Networking Configuration
2) Create Trunk Group



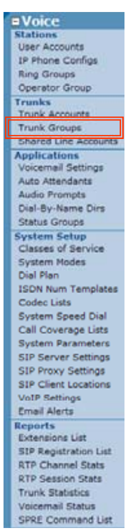
1. Select the Voice / Trunks / Trunk Groups menu



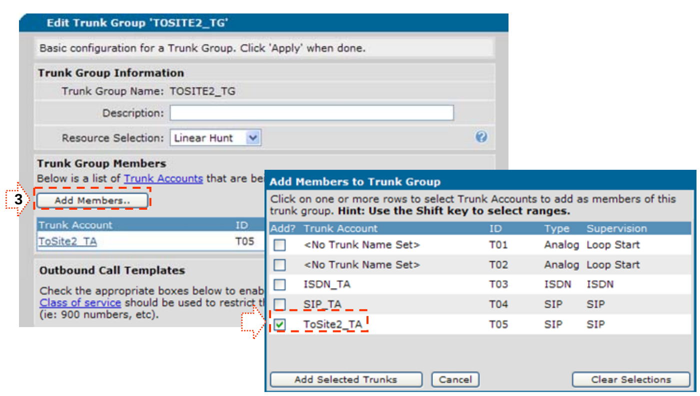
2. Create a Trunk Group
   – Points to the new SIP Trunk Account
   – Define call types allowed to other site

Trunk Group – Add SIP Trunk Account

SIP Networking Configuration
2) Create Trunk Group

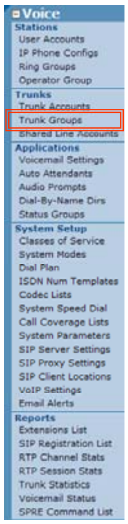


3. Click Add Members to add the Remote Site Trunk Account to this Trunk Group

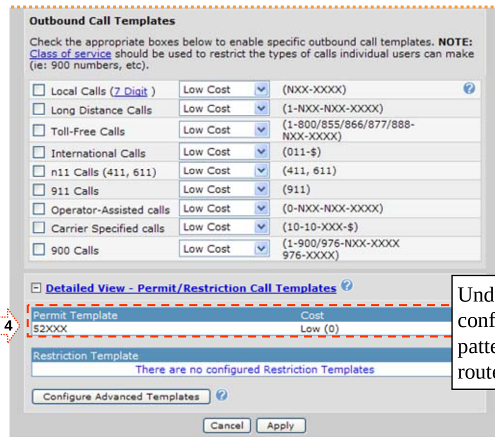


Trunk Group – Define Outbound Call Template

SIP Networking Configuration
2) Create Trunk Group



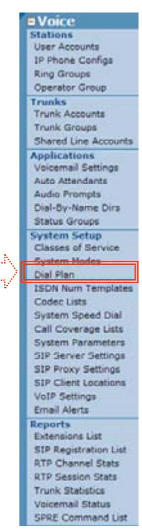
4. Add a custom Call Template
   – Define remote sites extension pattern



Under **Advanced Templates**, configure the extension pattern that will be used to route calls to the remote site

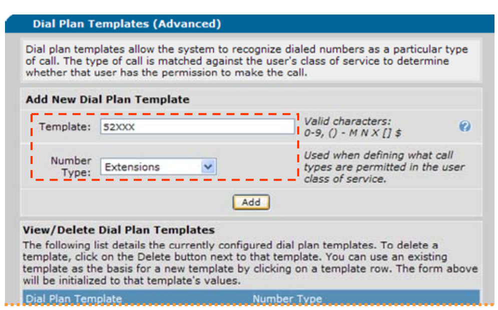
SIP Networking – System Dial Plan

SIP Networking Configuration
System Dial Plan



- Note: Configuration in the next two slides only needs to be done if both sides have the same extensions. An **extra digit** will be used as a site identifier – In this example 5 (52XXX)

1. Select Voice / System Setup / Dial Plan

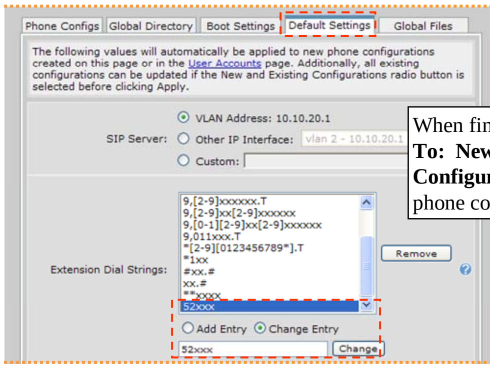


2. For our example, add 52XXX as an Extension

SIP Networking – IP Phone Configs Dial Plan

SIP Networking Configuration
IP Phone Configs Dial Plan

1. Select Voice / Stations / IP Phones Configs



When finish, click **Apply Settings To: New and Existing Configurations** to modify existing phone configuration files

2. Add 52XXX to the Extension Dial Strings
   – Note: The existing extension pattern of [12345678]xxx could also be changed to [1234678]xxx (removing 5XXX as an extension)

Remote User Preview

NetVanta IP Telephony Course

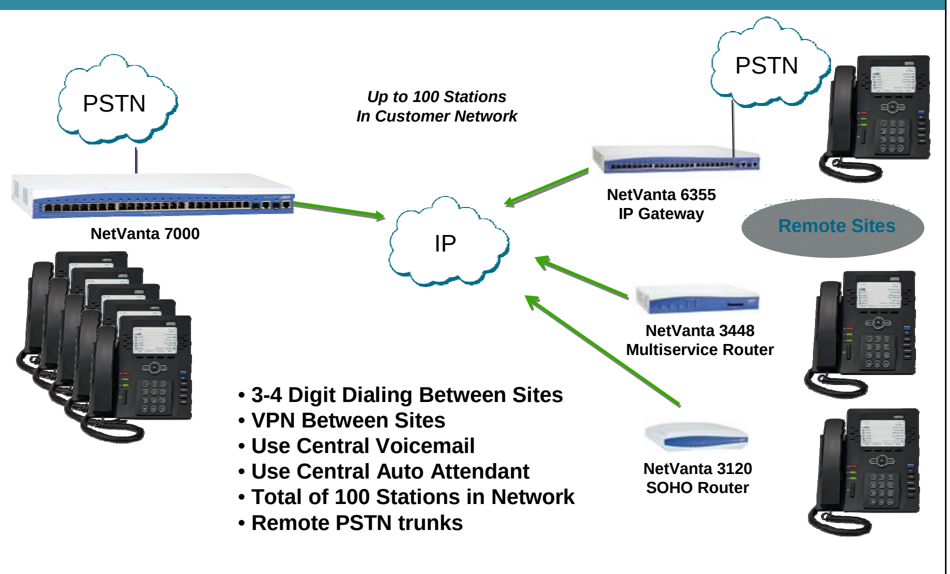
Remote User Preview

Configuration Guide can be found at

kb.adtran.com

NetVanta 7000 Solution – Remote Sites

NetVanta 7000 Series Solution Remote Sites

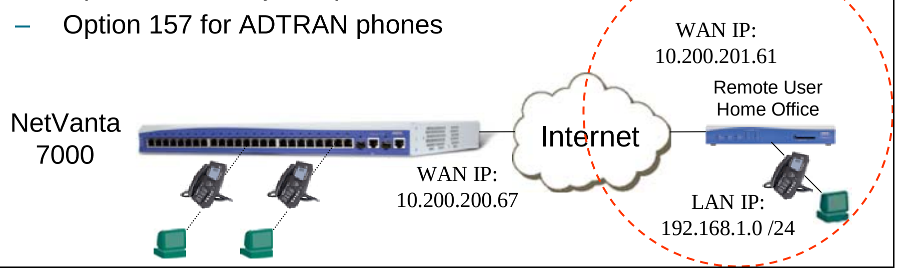


Remote User - Basic Configuration Steps

Remote User – Home Office Basic Configuration Steps

Remote NetVanta

1. Normal router configuration
   – Change LAN IP to 192.168.1.0 /24
2. Normal firewall configuration
   – Allow UDP traffic in WAN
3. Add DHCP Boot Server option for phone
   – Option 66 for Polycom phones
   – Option 157 for ADTRAN phones

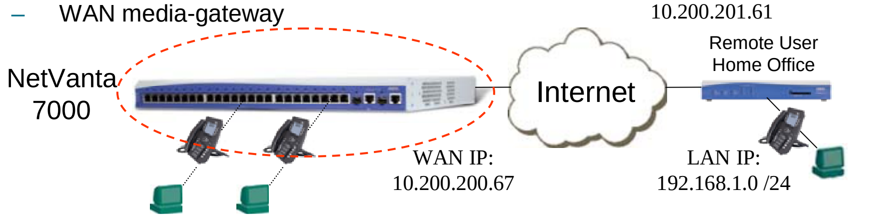


Remote User - Basic Configuration Steps

Remote User – Home Office Basic Configuration Steps

Local NetVanta 7000

1. Remote Phone Configuration
   – Create SIP User by defining MAC Address for remote phone
   – Define Boot Settings for remote phones
   – Set Boot Profile of new SIP user to Remote Phone
2. Firewall Configuration
   – Allow SIP and FTP traffic in WAN
3. VoIP Settings
   – SIP from header / RTP Firewall Traversal
   – WAN media-gateway



VPN Preview

NetVanta IP Telephony Course

VPN Preview

Configuration Guide can be found at

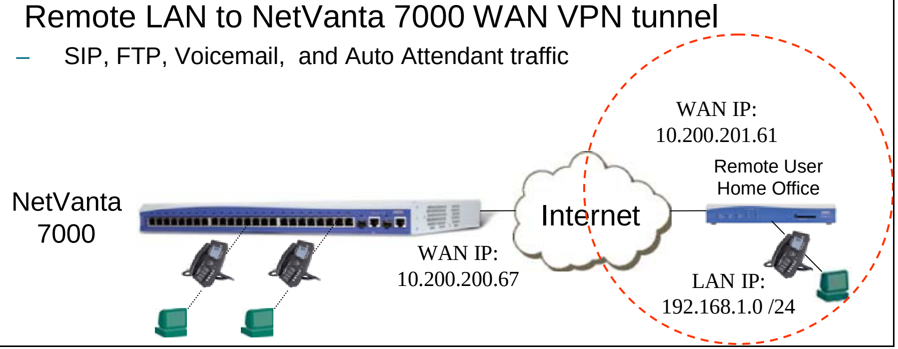
kb.adtran.com

Remote User over VPN - Basic Configuration

Remote User over VPN Basic Configuration Steps

Remote NetVanta

1. Router/DHCP/VoIP settings config
   - Same as remote user with no VPN
2. Remote LAN to NetVanta 7000 LAN VPN tunnel
   - RTP traffic
3. Remote LAN to NetVanta 7000 WAN VPN tunnel
   - SIP, FTP, Voicemail, and Auto Attendant traffic

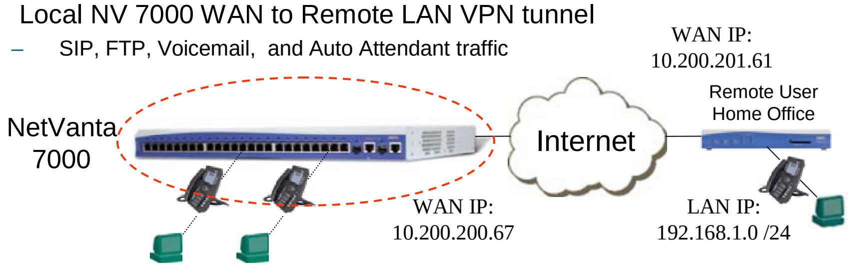


Remote User Over VPN - Basic Configuration

Remote User Over VPN
Basic Configuration Steps

Local NetVanta 7000

1. Remote Phone Configuration
   - Create SIP User by defining MAC Address for remote phone
   - Define Boot Settings for remote phones
   - Set Boot Profile of new SIP user to Remote Phone
2. WAN media-gateway enabled
3. Local NV 7000 LAN to Remote LAN VPN tunnel
   - RTP traffic
4. Local NV 7000 WAN to Remote LAN VPN tunnel
   - SIP, FTP, Voicemail, and Auto Attendant traffic



VoIP Quality Monitoring

NetVanta IP Telephony Course

VoIP Quality Monitoring (VQM)

VQM – What is it?

VQM – What is it?

- Voice Quality Monitoring
  - Provides visibility into VoIP networks
  - VoIP network troubleshooting and monitoring
  - Works in conjunction with QoS
  - Graphically intuitive web interface
- Allows network device to make real-time VoIP quality measurements on SIP-signaled RTP VoIP calls
- Measurements provide live and historical performance data on a per-call basis (inbound and outbound)
- Call quality monitoring using MOS, delay, jitter, packet loss and out of order packets

VQM – How it benefits you!

VQM – How it benefits you!

- Improves customer VoIP experience
  - VoIP performance can be proactively monitored
- Provides visibility into VoIP networks
  - Identify problem areas: Local LAN, Remote LAN, or WAN
  - Identify interface errors
  - Verify QoS configurations
  - Monitor network utilization
- Reduces Operating Expense
  - Reduce turn up time
  - Allows remote monitoring of VoIP performance
  - Reduce truck rolls
- Reduces Downtime
  - Segments network issues

VQM – Understanding Terms

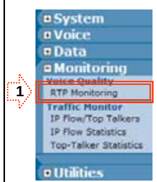
VQM – Understanding Terms

- Quality of Service (QoS) - The ability to assigning priority to specific network traffic
- Mean Opinion Score (MOS) - a numerical measure of the quality of human speech at the destination end of a circuit
  - A MOS score ranges from 1 to 5
  - Acceptable MOS scores are 4 and above
- Delay - The amount of time between the transmission and reception of packets
- Jitter - Variations in the total delay for a single packet
- Loss - Packets dropped along the way
- Out of Order - Packets received out of order causes reassembling problems

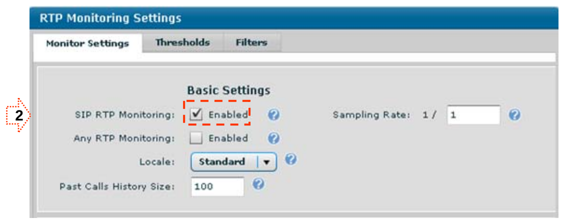
Enabling VQM

Enabling VQM

ADTRAN



1. Select the Monitoring / Voice Quality / RTP Monitoring menu



2. Enable SIP RTP Monitoring
   - This enables the ability to monitor RTP streams associated with SIP traffic that traverse the filewall

VQM – Graphically Intuitive Interface

VQM - Graphically Intuitive Interface

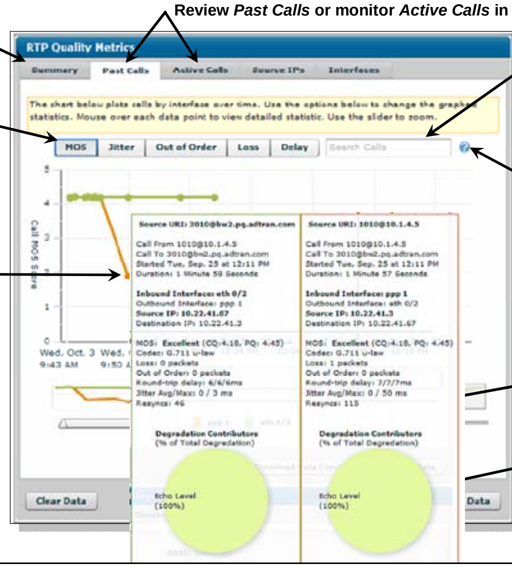
ADTRAN

Review *Past Calls* or monitor *Active Calls* in real-time

Summary tab shows overall network health

Monitor Mean Opinion Score (MOS), Jitter, and Out of Order, Lost, or Delayed packets

An example of a data point on the graph with a low MOS (1.86) indicates poor voice quality. Use your mouse to hover over any data point allows you to see detailed information pertaining to that specific call.



Use the *Search Field* to sort a large number of data points.

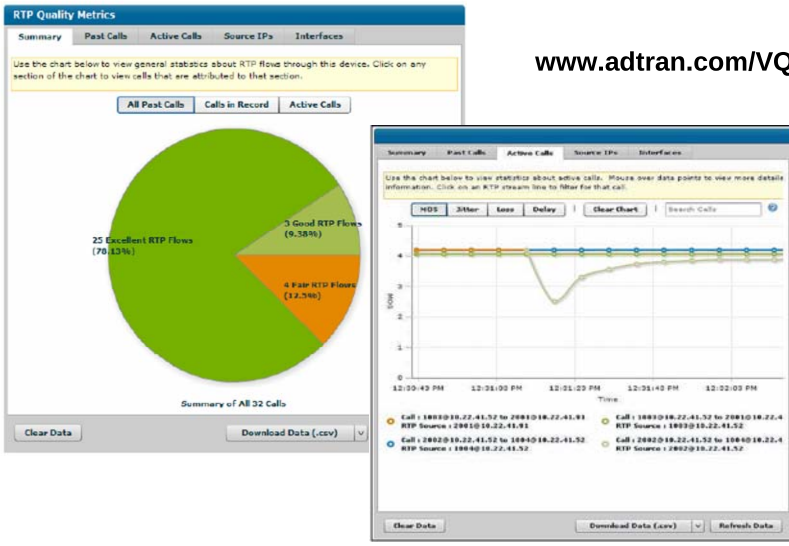
Hovering over the question mark offers examples of multiple-term search and valid filters.

Use the slider bar with color-coded interfaces to filter RTP streams.

Export data as Comma Delimited or XML

VQM - Demonstration

VQM – Demonstration



www.adtran.com/VQM

Troubleshooting

NetVanta IP Telephony Course

Troubleshooting

show sip user-registration

show sip user-registration

- Display local SIP Server registration information

NV7000# **show sip user-registration**

EXTENSION	TYPE	IP ADDRESS	PORT	PROT	EXPIRES
2003	Adtran-SIP-IP712/v1.3.7	10.10.20.2	5060	UDP	3537
2004	PolycomSoundPointIP_601..	10.10.20.3	5060	UDP	2009

Total phones registered: 2

show sip trunk-registration

show sip trunk-registration

- Display local SIP Trunk registration information

```
NV7000# show sip trunk-registration

Trk   Identity   Reg'd  Grant   Expires Success Failed Requests Challenges Rollovers
----  ---------  -----  ------- ------- ------- ------ -------- ---------- ---------
T04   9635501    Yes    3600    833     9       0      18       9          0

Total Displayed: 1
```

sip trunk-registration force-register

sip trunk-registration force-register

- Force a SIP registration

```
NV7100# sip trunk-registration force-register
```

debug sip stack message summary

debug sip stack messages

- View a summary of SIP messages (first line only)

```
NV7000# debug sip stack messages summary
15:16:35 SIP.STACK MSGSUM  Tx: REGISTER sip:172.23.102.87:5060 SIP/2.0
15:16:35 SIP.STACK MSGSUM  Rx: SIP/2.0 401 Unauthorized
15:16:35 SIP.STACK MSGSUM  Tx: REGISTER sip:172.23.102.87:5060 SIP/2.0
15:16:35 SIP.STACK MSGSUM  Rx: SIP/2.0 200 OK
15:16:35 SIP.STACK MSGSUM  Rx: NOTIFY sip:9635501@172.23.102.41 SIP/2.0
15:16:35 SIP.STACK MSGSUM  Tx: SIP/2.0 200 OK NV7100#
15:16:52 SIP.STACK MSGSUM  Tx: NOTIFY sip:2003@10.10.20.2 SIP/2.0
15:16:53 SIP.STACK MSGSUM  Rx: SIP/2.0 200 OK
```

debug voice summary

debug voice summary

- View call routing summary real time
  - Can confirm proper trunk is being used

```
NV7000# debug voice summary
15:22:47:830 VOICE.SUMMARY voice user 2001 cos allowed the call to Local
15:22:47:832 VOICE.SUMMARY 2001 is calling T04 (9635502).
15:22:51:681 VOICE.SUMMARY RTP for Call from 2001 to 9635502:  Codec PCMU
15:22:51:683 VOICE.SUMMARY 2001 is connected to T04 (9635502)
15:22:57:845 VOICE.SUMMARY Call from 2001 to T04 (9635502) ended by 2001:
15:23:23:178 VOICE.SUMMARY T03 is calling 2003 (2003).
15:23:26:316 VOICE.SUMMARY RTP for Call from 2003 to 2003:  Codec PCMU
15:23:26:317 VOICE.SUMMARY T03 is connected to 2003 (2003)
15:23:31:612 VOICE.SUMMARY Call from T03 to 2003 (2003) ended b
15:23:41:532 VOICE.SUMMARY voice user 2003 cos allowed the call to Local
15:23:41:534 VOICE.SUMMARY 2003 is calling T01 (8021000).
15:23:43:950 VOICE.SUMMARY RTP for Call from 0 to 8021000:  Codec PCMU
15:23:43:951 VOICE.SUMMARY 2003 is connected to T01 (8021000)
15:23:52:842 VOICE.SUMMARY Call from 2003 to T01 (8021000) ended by 2003:
```

Voice Trunk ID

Module Summary

Module Summary

- At the end of this module, you should be able to:
- Recognize NetVanta 7000 Remote Telephony Applications
- Configure Service Provider SIP Trunk
- Configure SIP Networking between Sites
- Enable VoIP Quality Monitoring (VQM)
- Conduct Voice Troubleshooting in a NetVanta 7000 Remote Telephony Application

Module 9: NetVanta 7000 Miscellaneous Tools and Utilities

Module Objectives

Module Objectives

- Introduce the following Tools:
  - Top Talkers
  - Top Visited Web Sites
  - Wireless Controller
  - n-Command
- Introduce System Utilities
  - Port Mirroring
  - Firmware Upgrades
  - Configuration Backup

Top Talkers

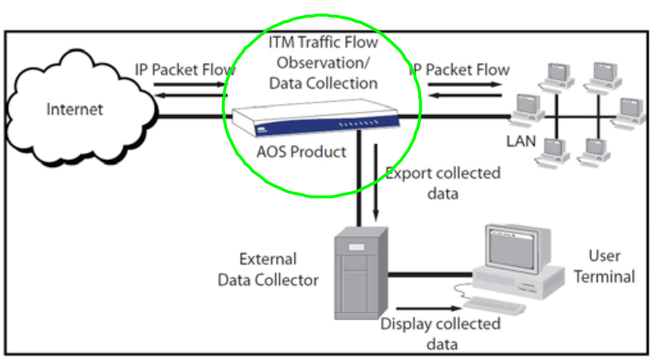
NetVanta 7100

Top Talkers

Top Talkers

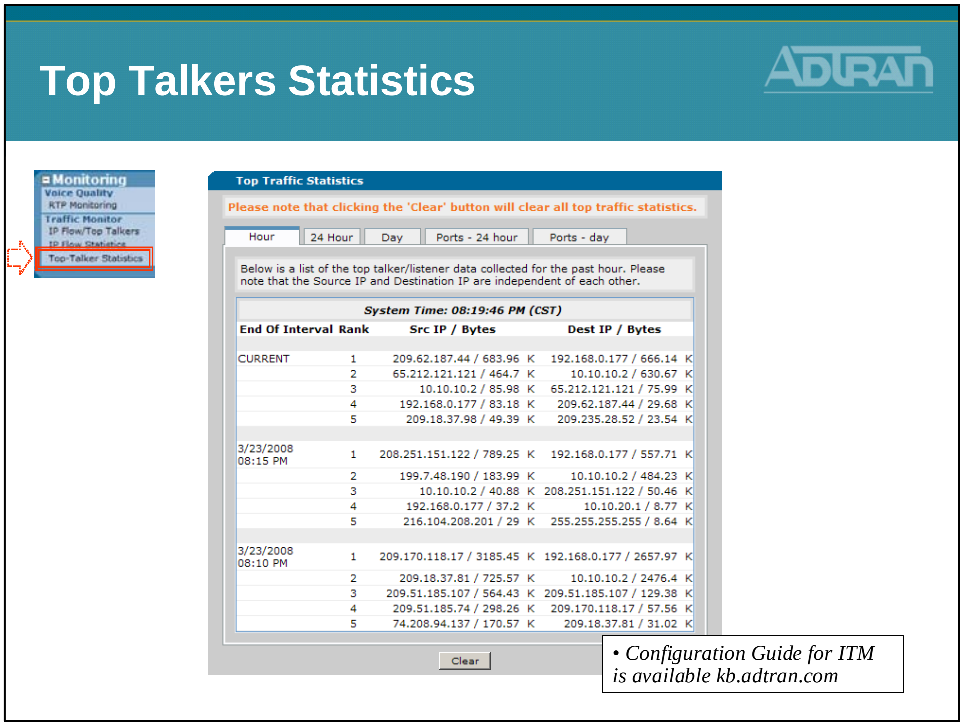
Top Talkers

- Top Talkers enhances ITM by adding a simple internal data collector to NetVanta Products



• Integrated Traffic Monitoring (ITM) is a method of tracking traffic flow patterns across interfaces on a network using Netflow v9 as an export protocol for maximum operability with external data collectors.

Top Talkers Statistics



Top Visited Web Sites

NetVanta 7100

Top Visited Web Sites

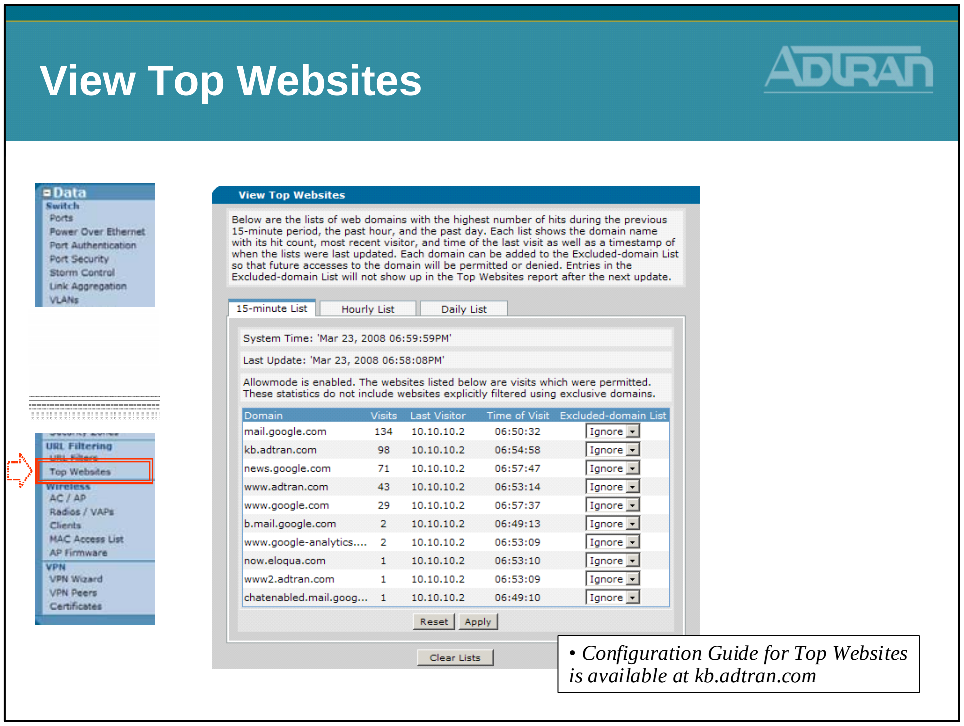
Top Visited Web Sites

Top Visited Web Sites

- Report top websites requested by users
  - Can be used without a Websense server

mail.google.com	134	10.10.10.2	06:50:32	Ignore
kb.adtran.com	98	10.10.10.2	06:54:58	Ignore
news.google.com	71	10.10.10.2	06:57:47	Ignore
www.adtran.com	43	10.10.10.2	06:53:14	Ignore
www.google.com	29	10.10.10.2	06:57:37	Ignore
b.mail.google.com	2	10.10.10.2	06:49:13	Ignore
www.google-analytics....	2	10.10.10.2	06:53:09	Ignore
now.eloqua.com	1	10.10.10.2	06:53:10	Ignore
www2.adtran.com	1	10.10.10.2	06:53:09	Ignore
chatenabled.mail.goog...	1	10.10.10.2	06:49:10	Ignore

View Top Websites



Wireless Controller

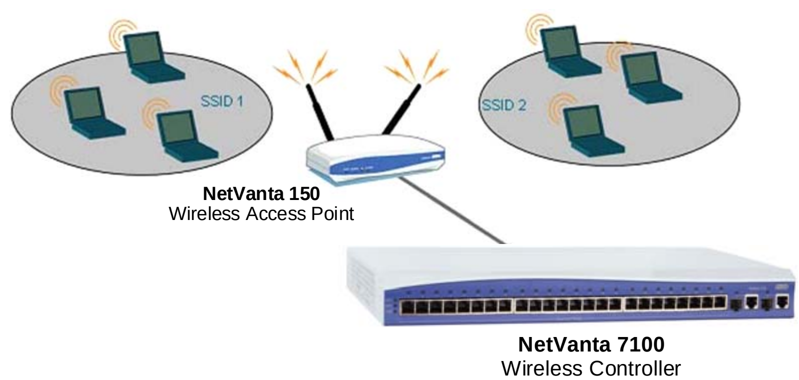
NetVanta 7100

Wireless Controller

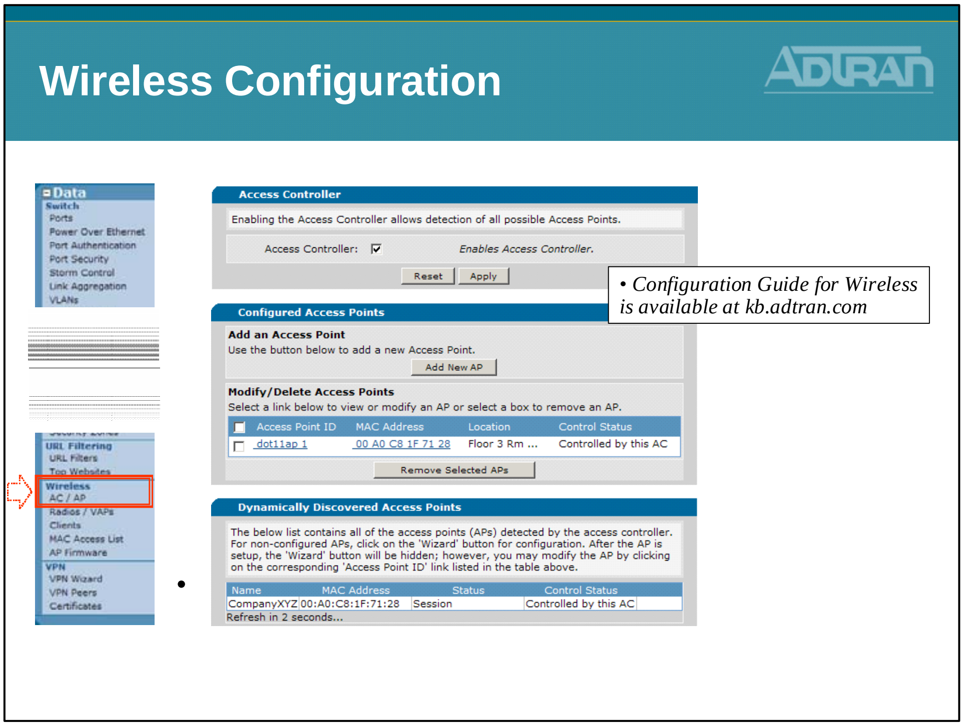
Wireless Controller

Wireless Controller

- Wi-Fi enable the NetVanta 7100



Wireless Configuration

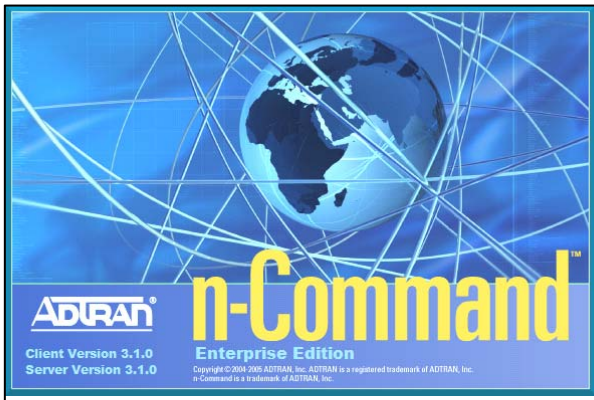


n-Command

NetVanta 7100

n-Command

Added Value To Dealers



n-Command

Enterprise Edition:
Network Productivity
Suite

ADDED VALUE TO DEALERS

ADTRAN – Management Solution

ADTRAN - Management Solution

- Individual Device Configuration
  - Familiar Command Line Interface (CLI)
  - Web Browser
    - For easy graphical device configuration
- Manage Large Deployments of Devices
  - n-Command
    - Configuration Management
    - Asset Management

n-Command – What is it?

n-Command – What is it?

- n-Command is a suite of productivity tools that help IT departments save time and money on daily network operations
  - VARs also use n-Command to offer their customers value-add services
- n-Command is designed to facilitate global device activities:
  - Firmware Upgrades
  - Configuration changes
  - Configuration Backups
  - Asset management
- n-Command is NOT an NMS that monitors performance, detects alarms or provides performance reports
  - It complements any NMS system that may already exist since it focuses on features that a NMS does not support

n-Command: Product Support

n-Command: Product Support

- * Supports all NetVanta AOS products and more …
  - NetVanta 7000 Series
  - NetVanta 300 Series
  - NetVanta 1000 Series
  - NetVanta 2000 Series
  - NetVanta 3k, 4k, & 5k Series
  - TA 900 Series
  - TA 600/850 MDL
    - (Add-on Module)

n-Command: Services Offered

n-Command: Services Offered

- Manage Multiple Customer Networks: Separate customer networks by creating different Folders or Sub-Folders
  - Flexibility to organize each customer's network devices differently
  - Modify user access per folder/customer network
- Inventory: Keep track of Customer Inventory - Device part number, serial number, firmware revision, boot code revision, etc…
  - Provide an Excel Inventory Report periodically or when requested
- Configuration Changes: Make Configuration changes to multiple Devices and keep a log of all user activity per device
  - Push changes to a single device or large numbers of devices quickly
  - Provides a detailed summary of the job upon completion
- Uses Secure Communication (SSH and Secure Copy)

n-Command: Services Offered

n-Command: Services Offered

- Backup Configuration: Schedule recurring configuration backups and monitor for any unexpected changes in configuration
  - NetVanta 7100 – Backup the full system including:
    - All Phone Files
    - All NetVanta data and voice configs
    - VoiceMail Greetings (Note: VoiceMail Messages are not backed up)
  - Send e-mail notification summary if a config. change is found or if the running config. and startup config. are different
- Configuration Restore: Quickly restore a previous configuration based on a date or event to one or many devices
- Firmware Update: Notify customer whenever a new firmware revision is available for their devices, and update their firmware without a truck roll
- Uses Secure Communication (SSH and Secure Copy)

n-Command Specifics

n-Command Specifics

- ACL Manager (Policy Pushing)
  - Quickly change Access Control lists used for packet selection on interfaces, QOS and Firewall Polices etc...
    - Find & Mark Common ACL Statements in existing ACLs
    - Create / Edit / Delete / Insert Templates
    - Syntax Checking
    - Preview of ACL after change / View “Fix Up” Code
    - Backup Configuration Before / After ACL change
    - Fail Safe Push of “Fix Up” Code
    - Device Log of All actions

Folder Management

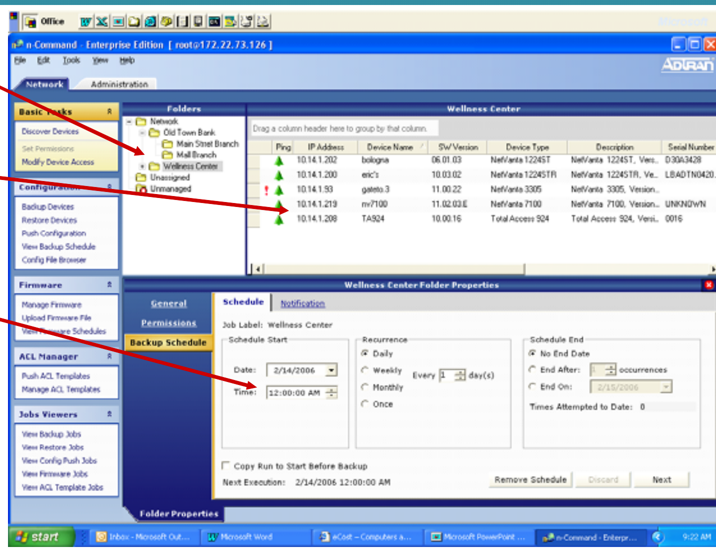
Folder Management

Create Folders and sub-folders

Drag Devices into folders

Set Automatic backup schedule on a folder basis

Set User Permissions on a per Folder basis



n-Command Part Numbers

n-Command Part Numbers

Part Number & Description:

- n-Command Enterprise Edition CD* 1950843L1
- 500-node Incremental License 1950844L1
- 250-node Incremental License 1950844L2
- 2-Client Incremental License 1950845L1
- ACL Manager Module 1950850L1
- Total Access 600/850 Module 1950852L1

- Annual Maintenance - Access to download patches, upgrades and phone support
  - n-Command Enterprise Edition CD comes with support for 100-Nodes, 5-Clients, and one year of software maintenance included

n-Command – Other Info

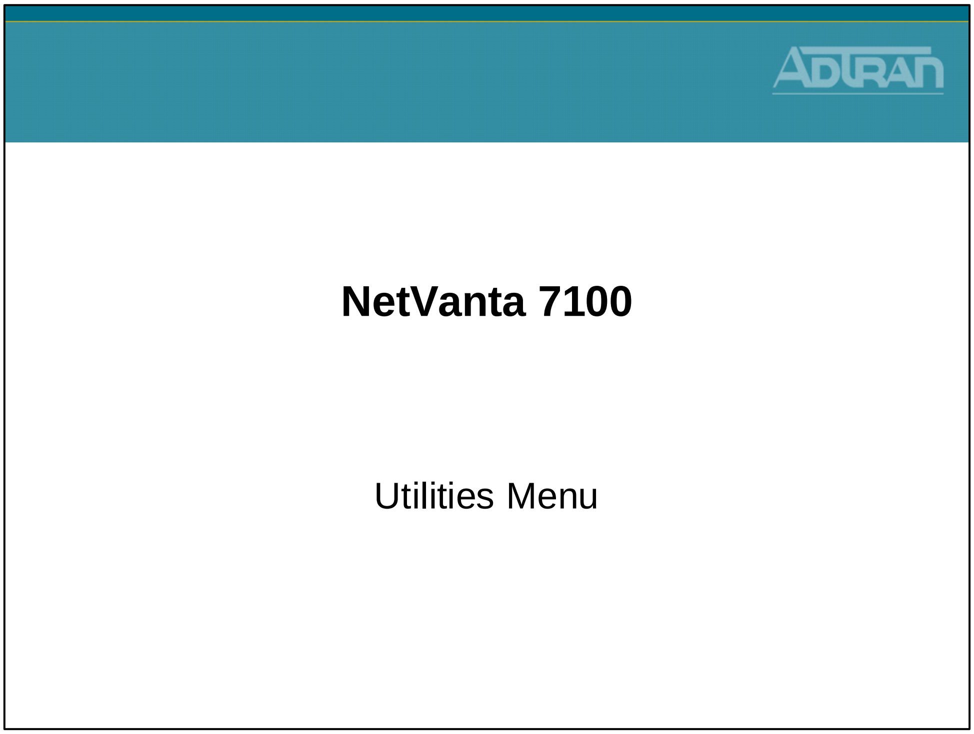
n-Command – Other Info.

- There is a worthwhile flash video on our web site which a customer can watch. It demonstrates all functionality.

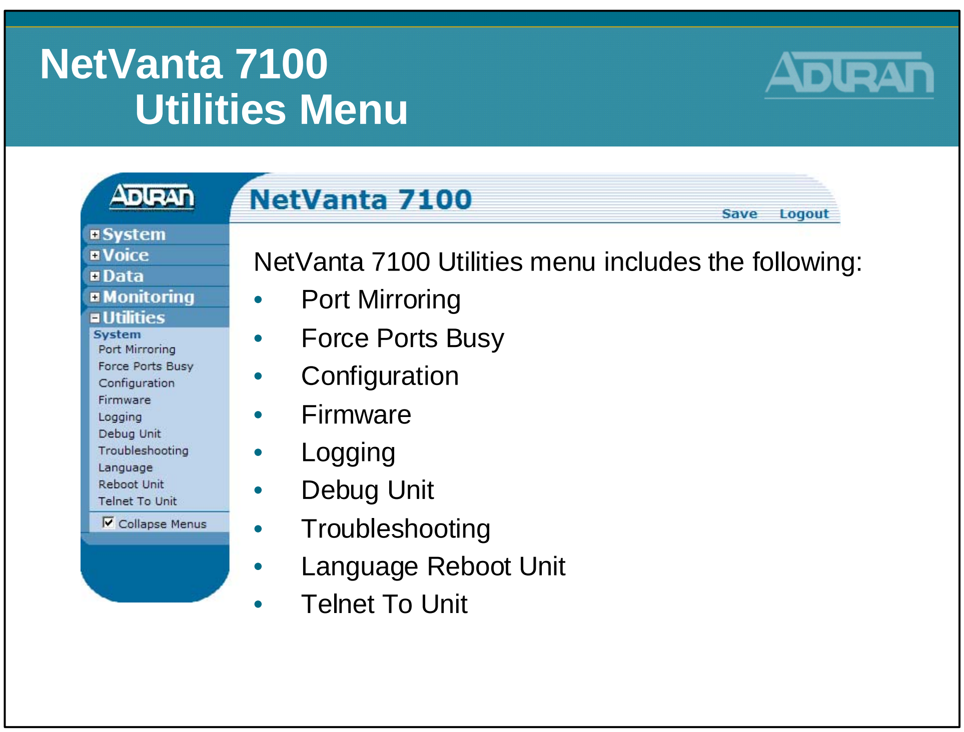
  www.adtran.com/n-command

- Demo disks are available through our publication department. It allows all features to work, but is limited to 5-nodes.

Utilities Menu

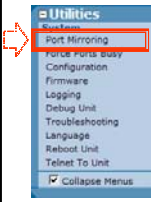


NetVanta 7000 - Utilities Menu

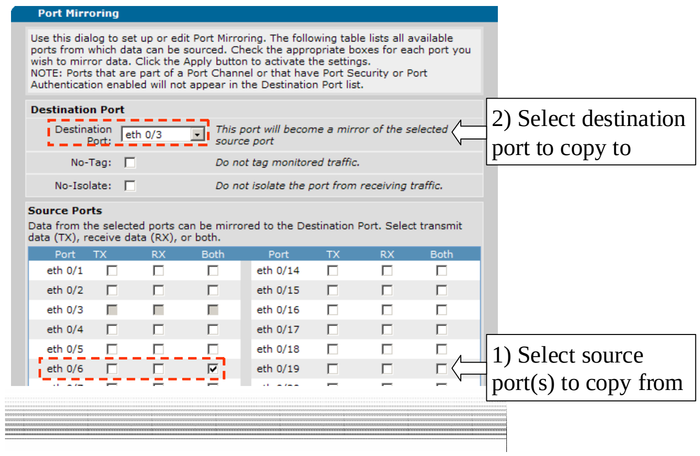


Utilities / System - Port Mirroring

Utilities / System
Port Mirroring

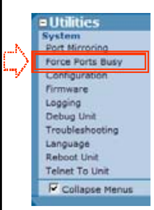


- Mirror (copy) source traffic from a source port to a destination port

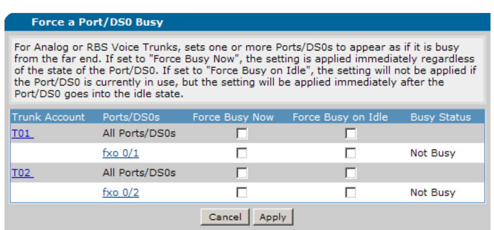


Utilities / System - Force Ports Busy

Utilities / System
Force Ports Busy



- Force ports busy for maintenance
  - Analog and RBS Voice Trunks

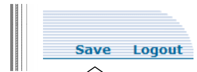


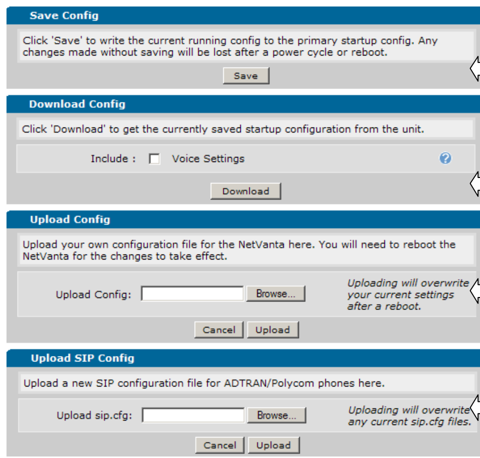
  - Force Busy Now
    - Setting immediately applied
  - Force Busy on Idle
    - Setting applied when port becomes idle

Utilities / System - Configuration

Utilities / System Configuration

- Saving / Backup Configuration





Same function as **copy run start**

Save config to any location on your PC

Upload a config to from your PC

Upload sip.cfg from your PC

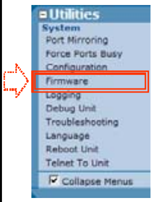
Firmware Upgrades

Firmware Upgrades

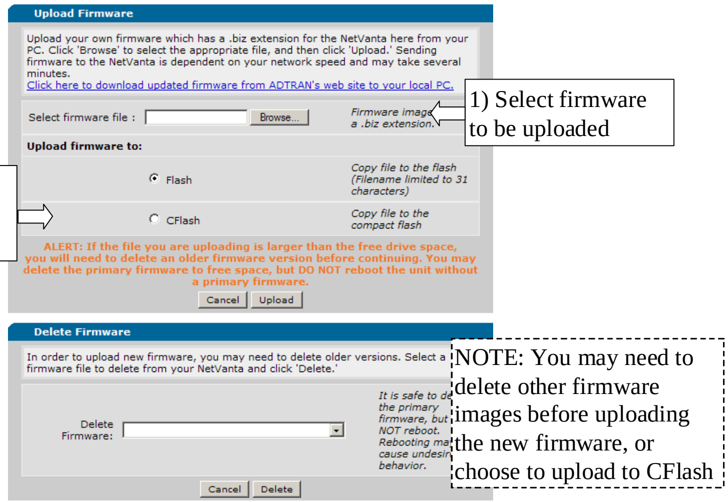
- Firmware upgrades can be handled in mass with n-Command
- Firmware upgrades can be done via the Web GUI
- Firmware upgrades can be done via the command line
  - FTP
  - TFTP
  - SCP

Utilities / System - Firmware

Utilities / System Firmware

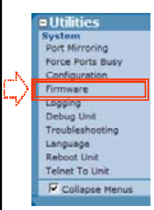


1. Upload the Firmware image to NetVanta 7100

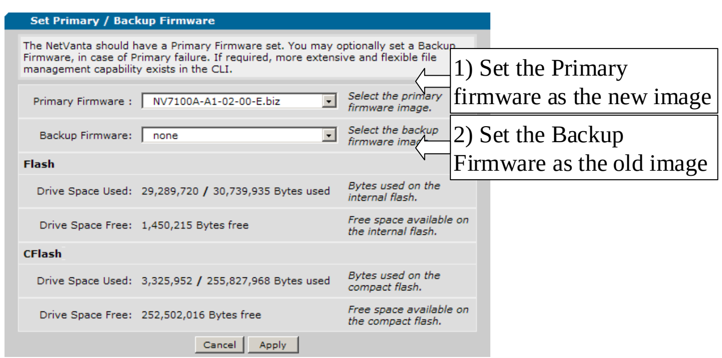


Utilities / System - Firmware

Utilities / System Firmware



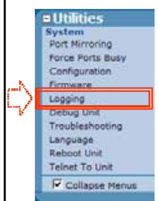
2. Set Primary and Backup Firmware Image



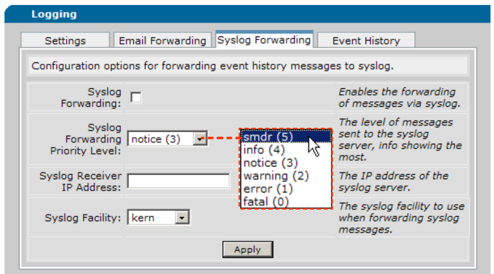
3. Click Apply and then reboot to complete the firmware up grade process

Utilities / System - Logging – SMDR

Utilities / System Logging - SMDR



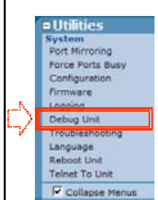
- Enable SMDR Logging



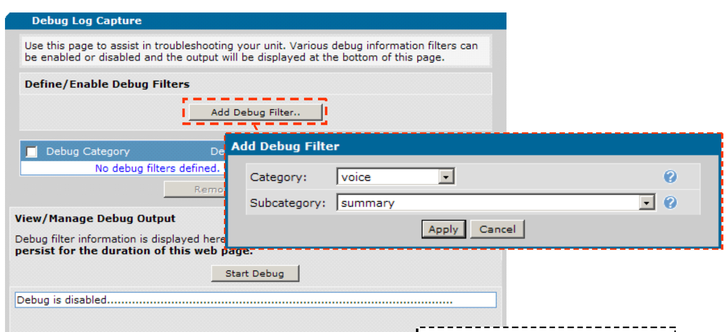
1. Enable Syslog Forwarding
2. Set Syslog Forwarding Priority Level to SMDR
3. Define IP address of Syslog server
4. Optional – define Syslog facility

Utilities / System - Debug Unit

Utilities / System Debug Unit



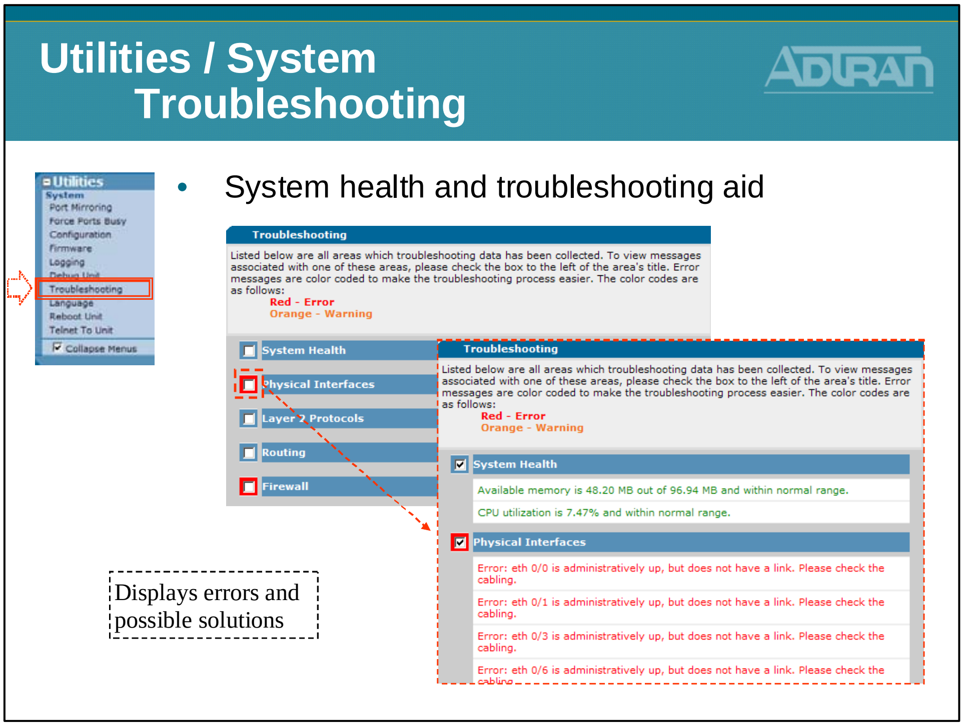
- View debug events real time



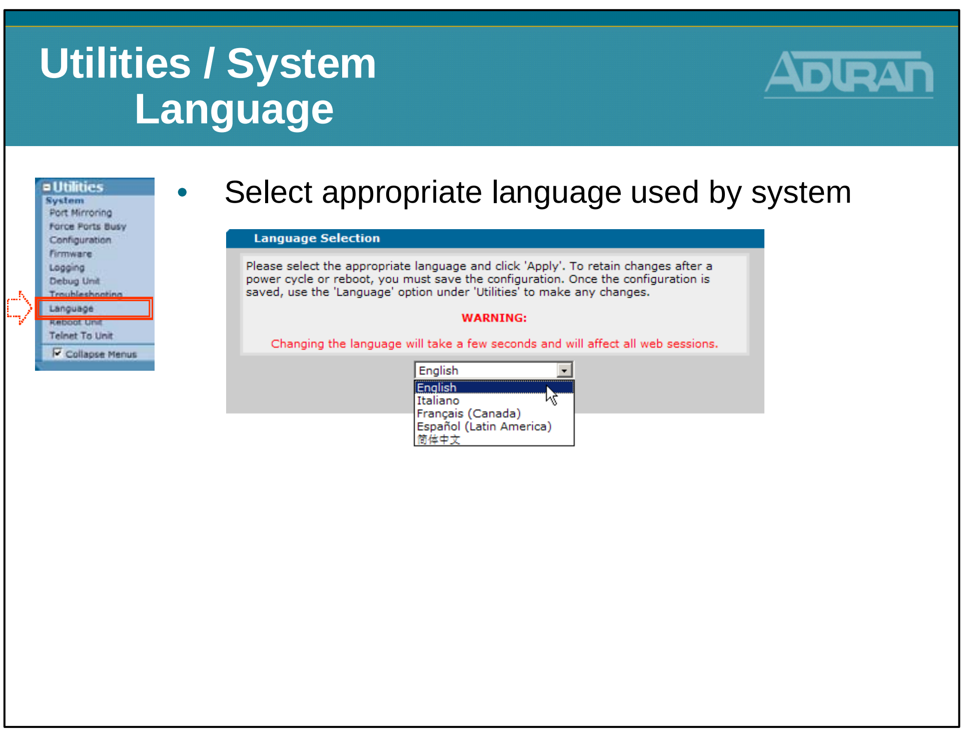
Debug information will display on screen here

1. Select Add Debug Filter
2. Choose a Category
3. Choose Sub-category
4. Click Start Debug

Utilities / System - Troubleshooting

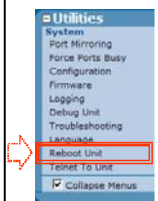


Utilities / System - Language

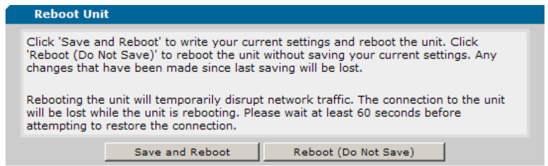


Utilities / System - Reboot Unit

Utilities / System
Reboot Unit



- Rebooting the NetVanta 7100

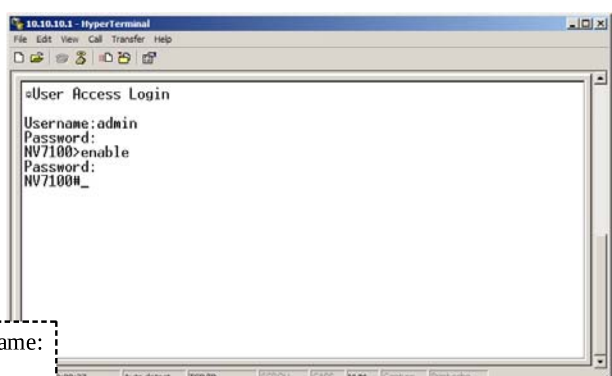


- Save and Reboot
  - Saves the running configuration to startup-config and dynvoice-config
- Reboot (Do Not Save)
  - Reboots the unit without saving

Utilities / System - Telnet To Unit

Utilities / System
Telnet To Unit

- Access the Command Line Interface through an IP connection using Telnet



Default Username:
admin
Default Password:
password

Module Summary

Module Summary

- At the end of this module, you should be:
- Familiar with the following Tools:
  - Top Talkers
  - Top Visited Web Sites
  - Wireless Controller
  - n-Command
- Troubleshoot with Port Mirroring
- Upgrade the NetVanta 7000 Firmware
- Save and backup configurations