# Instituto Tecnológico de Costa Rica

Escuela de Ingeniería en Electrónica


**Unisys de Centroamérica**

**“Diseño y ampliación de Comunicaciones”**

**Informe de Proyecto de Graduación para optar por el grado de Bachiller en Ingeniería Electrónica**

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***DEDICATORIA***

*Dedico este trabajo a mi madre. Eres tú quien me impulsó y quien me mostró el camino para llegar a dónde hoy estoy... gracias por estar allí siempre... gracias mamá.*

## AGRADECIMIENTO

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**Resumen**

El objetivo principal de este informe es determinar las características necesarias para realizar un enlace inalámbrico que permita unir al segmento principal de la red de Unisys de Centroamérica con otro segmento de red que se ubicará en un local en el que se desea establecer un Nuevo Centro Logístico. Este enlace transmitirá tanto voz (Voz sobre IP) como datos.

El informe detalla las características del enlace que se desea realizar: ancho de banda para voz, ancho de banda para datos, equipo de comunicaciones necesario para la realización del enlace, actualizaciones de software (para el equipo en existencia) necesarias para la realización del enlace y también detalla los requerimientos del Nuevo Centro Logístico.

Palabras clave: VoIP; Voz sobre IP; Enlace Inalámbrico; Ancho de Banda; Equipo Inalámbrico.



**Abstract**

The main objective of this inform is to determinate the characteristics needed to make a wireless link that will allow to unite the main segment of the Unisys of Centro America Network with another segment that is going to be located in a warehouse where a New Logistic Center is going to be build. This link will transmit voice (Voice over IP) and data.

The inform details the characteristics of the link that is going to be make: total bandwidth, voice bandwidth, data bandwidth, communications equipment needed to achieve the link, software upgrades for the existing equipment and also details the New Logistic Center requirements.

Keywords: VoIP, Voice over IP, Wireless Link, Bandwidth, Wireless Equipment.



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Nota: Todas las tablas se realizaron en Microsoft Word

# CAPÍTULO 1: INTRODUCCIÓN

## 1.1 Descripción General

Unisys de Centroamérica es una empresa que tiene como fin principal el brindar servicios en el área de la informática y de las comunicaciones digitales. Estos servicios se encuentran orientados hacia el área de los sistemas automáticos que son empleados por compañías con bases de datos dinámicas.

El servicio que brinda la compañía comprende lo que se denomina “soluciones completas”; ésto significa que con base en el proyecto que se esté realizando, Unisys subcontrata los servicios de otras compañías que puede ofrecer otro tipo de equipo que permitirá brindar la “solución completa” al cliente.

Además de brindar servicios prácticos, Unisys de Centroamérica también brinda servicios de asesorías y evaluaciones de sistemas de redes actuales a las compañías que así lo soliciten (la asesoría comprende una evaluación de la red así como también un plan de acción).

La empresa tiene más de 40 años de prestar sus servicios. En ella laboran 110 empleados directos (propios de Unisys, en planilla) y 50 empleados por contratos temporales. El Gerente General es el Ingeniero Jorge Villalobos.

En el área de ingeniería la empresa está constituida por dos grandes divisiones, la división IS, o logística, que se encarga del desarrollo de proyectos y la división GNS, que se encarga del servicio al cliente, diseño e instalación de redes.



El proyecto de graduación se realizará en la división GNS, específicamente en implantación. Este departamento está formado por 4 ingenieros en las áreas de sistemas y electrónica, y sus funciones están orientadas a lo que se llama Equipo de Implantación, que como su nombre lo indica, consiste en la formación de grupos coordinados de ingenieros para instalar, configurar, probar y monitorear los sistemas que el equipo de diseño haya seleccionado y aprobado para ofrecer una solución a un problema. El coordinador del departamento es el Ingeniero Ricardo Zamora, el cual coordina además otros proyectos.

Para realizar los proyectos, el departamento cuenta con todo el equipo necesario, desde materiales y equipo, y en casos necesarios se encarga de la subcontratación de algunos elementos a otras compañías. Los organigramas de la compañía se muestran en el Anexo 6.

## 1.2 Descripción del Problema y su importancia

El edificio principal de Unisys de Centroamérica se localiza en el Parque Empresarial Fórum. En este edificio se encuentra el centro de cómputo y el centro de comunicaciones de la empresa.

Este edificio se encuentra enlazado a un Centro Logístico, conocido como CLU, que se encuentra en La Uruca. El enlace se realiza a través de líneas telefónicas dedicadas, equipo Cisco y fibra óptica (para las conexiones internas) y funciona con un ancho de banda de 64Kbits. Del mismo modo, el edificio principal (de Fórum) se encuentra enlazado con Racsa, y desde Racsa se realiza un enlace inalámbrico con el Centro de Comunicaciones Corporativo, ubicado en Minnesota, Estados Unidos de América.



El ancho de banda actual con el que cuenta el edificio principal de Unisys es de 384 Kbits. Dicho ancho de banda se encuentra distribuido de la siguiente forma:

- 8 canales de voz de 8Kb cada uno
- 1 canal de datos de 256 Kb para el enlace Fórum - Minnesota
  - 64 Kb para el enlace con el Centro Logístico de la Uruca (este ancho de banda es independiente del ancho de banda principal).
- Se desperdician 64 Kb debido a limitaciones del equipo que realiza la multiplexación por división de tiempo.

Unisys ha planeado el alquiler de un local para la creación de un Nuevo Centro Logístico que se ubicará muy cerca al edificio principal (poco menos de 1 Km de distancia). Este Centro estará constituido por una bodega, laboratorios, oficinas de ingenieros además del Centro de Servicio. Este nuevo Centro tendrá una función similar al Centro Logístico de la Uruca.

### 1.3 Objetivo General

Realizar un estudio que permita determinar las características necesarias del enlace inalámbrico que unirá al Edificio Central de Unisys de Centroamérica con un nuevo Centro Logístico en un periodo de 15 semanas.



## 1.4 Objetivos Específicos

1. Realizar una capacitación sobre los principios básicos de ambientes de redes (networking), tecnología y equipo activo de comunicaciones durante un periodo de 7 semanas.
2. Determinar los requerimientos del nuevo Centro Logístico que se enlazará con el Edificio Central de Unisys de Centroamérica de acuerdo con el equipo que allí se utilizará en un periodo de 1 semana.

3. Escoger diferentes métodos que permitan determinar el ancho de banda necesario para la transmisión de datos del nuevo Centro Logístico que se enlazará con el Edificio Central de Unisys de Centroamérica en un periodo de 0.75 semanas.

4. Obtener el ancho de banda, a partir del método óptimo, necesario para la transmisión de datos del nuevo Centro Logístico que se enlazará con el Edificio Central de Unisys de Centroamérica en un periodo de 0.75 semanas.

5. Escoger diferentes métodos que permitan determinar el ancho de banda necesario para la transmisión de voz del nuevo Centro Logístico que se enlazará con el Edificio Central de Unisys de Centroamérica en un periodo de 0.75 semanas.

6. Obtener el ancho de banda, a partir del método óptimo, necesario para la transmisión de voz del nuevo Centro Logístico que se enlazará con el Edificio Central de Unisys de Centroamérica en un periodo de 0.75 semanas.



7. Evaluar si el ancho de banda actual del Edificio Central de Unisys de Centroamérica es suficiente para realizar un enlace que permita la transmisión de voz y datos entre el Edificio Central y el Nuevo Centro Logístico en un periodo de 1 semana.

8. Determinar si el equipo de comunicaciones en existencia en el Edificio Central de Unisys de Centroamérica permitirá realizar un enlace que permita la transmisión de voz y datos entre el Edificio Central y el nuevo Centro Logístico en un periodo de 1 semana.

9. Seleccionar el equipo activo de comunicaciones que se requerirá en el nuevo Centro Logístico para unirlo, mediante un enlace inalámbrico que transmitirá voz y datos, al Edificio Central de Unisys de Centroamérica en un periodo de 1 semana.

10. Determinar las características de la distribución física del equipo de comunicaciones que se requerirá en el nuevo Centro Logístico para unirlo, mediante un enlace inalámbrico que transmitirá voz y datos, al Edificio Central de Unisys de Centroamérica en un periodo de 2 semanas.



# CAPÍTULO 2: ANTECEDENTES

Básicamente, en proyecto en cuestión no presenta antecedentes. Como ya se ha mencionado en el apartado anterior, el Nuevo Centro no se ha creado sino que se espera que se apruebe el alquiler del local. Debido ha esto, no se ha realizado ningún estudio que haya permitido determinar las características del enlace.

## 2.1 Estudio del problema por resolver

La solución proyectada consiste en realizar un enlace inalámbrico para unir al edificio central de Unisys con el Nuevo Centro que se planea crear. Dicho enlace se realizará a partir de las características que el estudio ha establecido.

El ancho de banda del enlace que se desea realizar no dependerá del ancho de banda que utiliza el edificio principal (ya que el ancho de banda del enlace completamente independiente del ancho de banda principal), además de que se apoyará en el hardware en existencia en el edificio principal.

El estudiante fue el encargado de realizar un estudio que permitió determinar las características que se necesitarán para el nuevo enlace: el ancho de banda total que se requerirá, el ancho de banda para la transmisión de voz, el ancho de banda para la transmisión de datos, determinar la posibilidad de ampliar el ancho de banda del edificio central, determinar la posibilidad de actualizar el equipo de comunicaciones en existencia en el edificio central, determinar el equipo activo de comunicaciones que se requerirá en el nuevo Centro Logístico y la distribución interna del mismo.

Dicho estudio servirá de base para realizar el enlace una vez que se haya aprobado la construcción del nuevo Centro.



La figura 2.1 muestra un diagrama de interconexión de la red interna del Edificio Principal de Unisys de Centroamérica. En este diagrama se puede observar que la red emplea un total de 8 switches (los cuales se encuentran conectados en cascada y las interfaces entre los mismos es de 1 Giga); estos switches representan un total de 192 puertos, de los cuales se emplean 180.

El switch 7 se conecta al Router Cisco 3660 a través del puerto Ethernet. El puerto Serial 0 del router se encuentra conectado al slot Pos 5D del Timeplex; el puerto Serial 1 del router está conectado a la línea dedicada, de allí pasa al módem del CLU (Centro Logístico de la Uruca). El diagrama de la conexión de la red del CLU se puede observar en la figura 2.2.

El Timeplex es un Mainframe MicroLink / 2+ de 6 slots. Este equipo es el encargado de realizar la multiplexación por división de tiempo (TDM) de la voz (proveniente de la central telefónica que se encuentra conectada a los slots 2A, 2B, 3A y 3B del Timeplex; en la conexión de la central telefónica al Timeplex se consumen 64 Kb del enlace total) y de los datos que le llegan por el Slot 5D. El slot 6A se encuentra conectado al módem que permite comunicarse con Racsa, a través de líneas dedicadas, para realizar el enlace inalámbrico con Minnesota.

En CLU se cuenta con un Router Cisco 3640 (ver figura 4.2). Las estaciones de trabajo (son 11) se conectan a 2 Hubs. Los Hubs se conectan al router a través del puerto Ethernet del mismo; el módem con el que se realiza el enlace a Fórum se conecta con el router a través del puerto Serial 0 del mismo (del router). El CLU no cuenta con una central telefónica, sino con 6 números de teléfono independientes proporcionados por el ICE.



Como se puede observar de las figuras 2.1 y 2.2, desde CLU hacia Fórum sólo se puede transmitir datos, la configuración actual del equipo no permite la transmisión de voz.

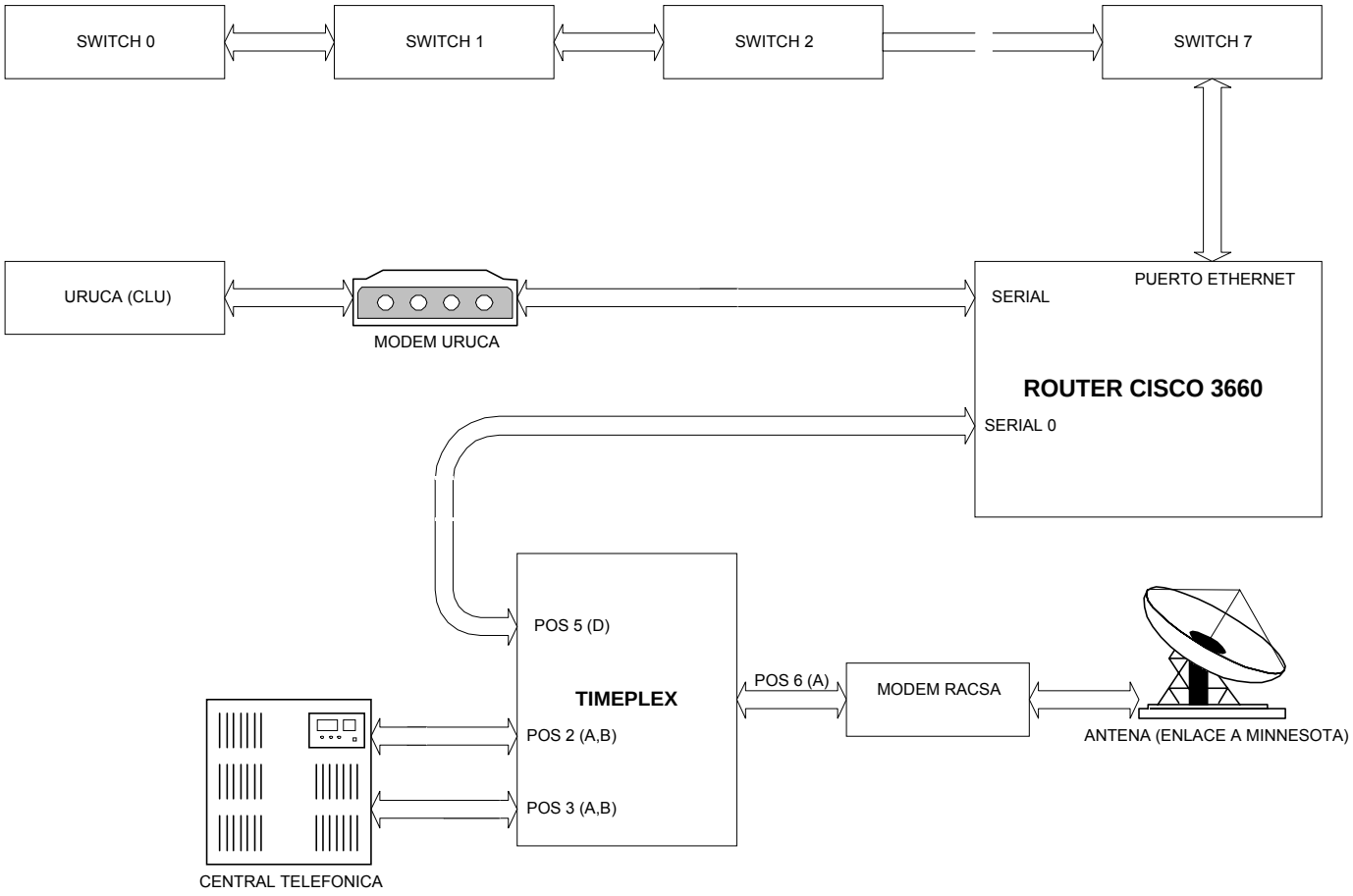
Desde el Edificio Principal de Unisys de Centroamérica se puede transmitir voz y datos, hacia Minnesota, a través del enlace inalámbrico en existencia.

El estudio que se desarrolló tuvo como finalidad determinar las especificaciones que permitirán lograr establecer un enlace inalámbrico entre el nuevo Centro Logístico y el Edificio Principal. Dicho enlace debe soportar la transmisión de datos (como se logra con el enlace actual hacia CLU), pero además también debe permitir la transmisión de voz a través de la red en existencia.

## 2.2 Requerimientos de la empresa

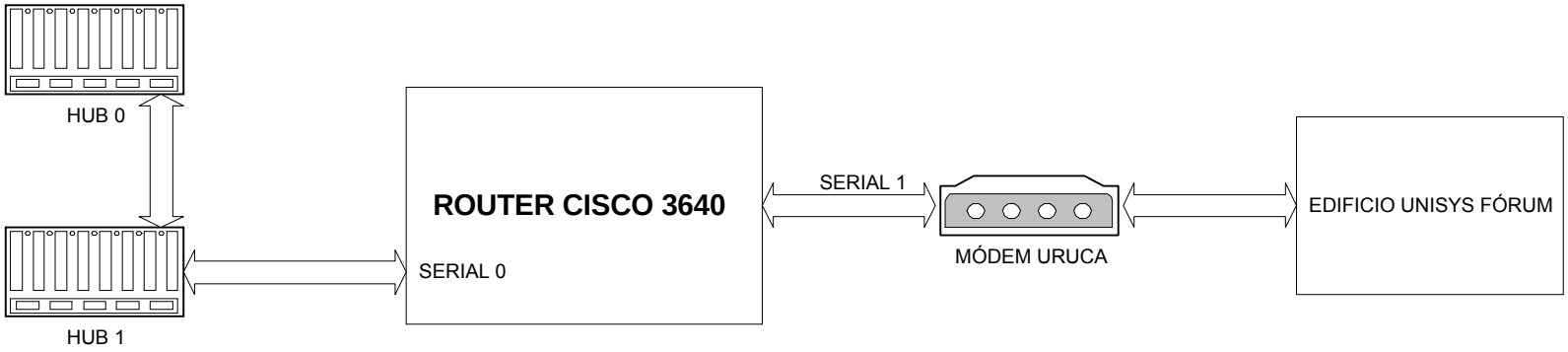
Unisys de Centroamérica requirió de un informe en el que se detallarán las características del enlace (ancho de banda de voz y datos) y los estudios realizados; en dicho informe además se recomienda la clase de equipo de comunicaciones que se empleará en el Centro Logístico para lograr el enlace y se presenta una propuesta para la distribución física del Nuevo Centro Logístico.






**Visio 4.5**

**Figura 2.1** Diagrama físico de la red interna de Unisys de Centroamérica




**Visio 4.5**

**Figura 2.2** Diagrama físico de la red interna de CLU (Realizado en Visio 4.5)

## 2.3 Solución propuesta

Se desea enlazar el nuevo Centro con el edificio principal de Unisys. Sin embargo, dicho enlace se desea que sea inalámbrico. La ventaja de realizar un enlace inalámbrico y no mediante líneas telefónicas ni líneas dedicadas es que se logra aumentar la capacidad de manejo de datos del Centro puesto que no se dependerá de líneas telefónicas además de modernizar los enlaces. El estudio realizado determina las características del ancho de banda que se deberá emplear para realizar este enlace además del equipo activo y de la distribución física del centro.

Como ya se mencionó, el Centro no se ha creado aún, sino que se encuentra como un proyecto por realizar. El estudio realizado sirve como base para que, una vez que se haya aprobado el alquiler del local, se cuente con las especificaciones necesarias para realizar el enlace.

La figura 2.3 muestra un diagrama de la solución planteada. Puede observarse que la solución propuesta utiliza el equipo en existencia en Unisys de Centroamérica, no se requiere de la adquisición de nuevo equipo (aparte de las interfaces inalámbricas). Se puede observar que los cambios principales se realizan el router.

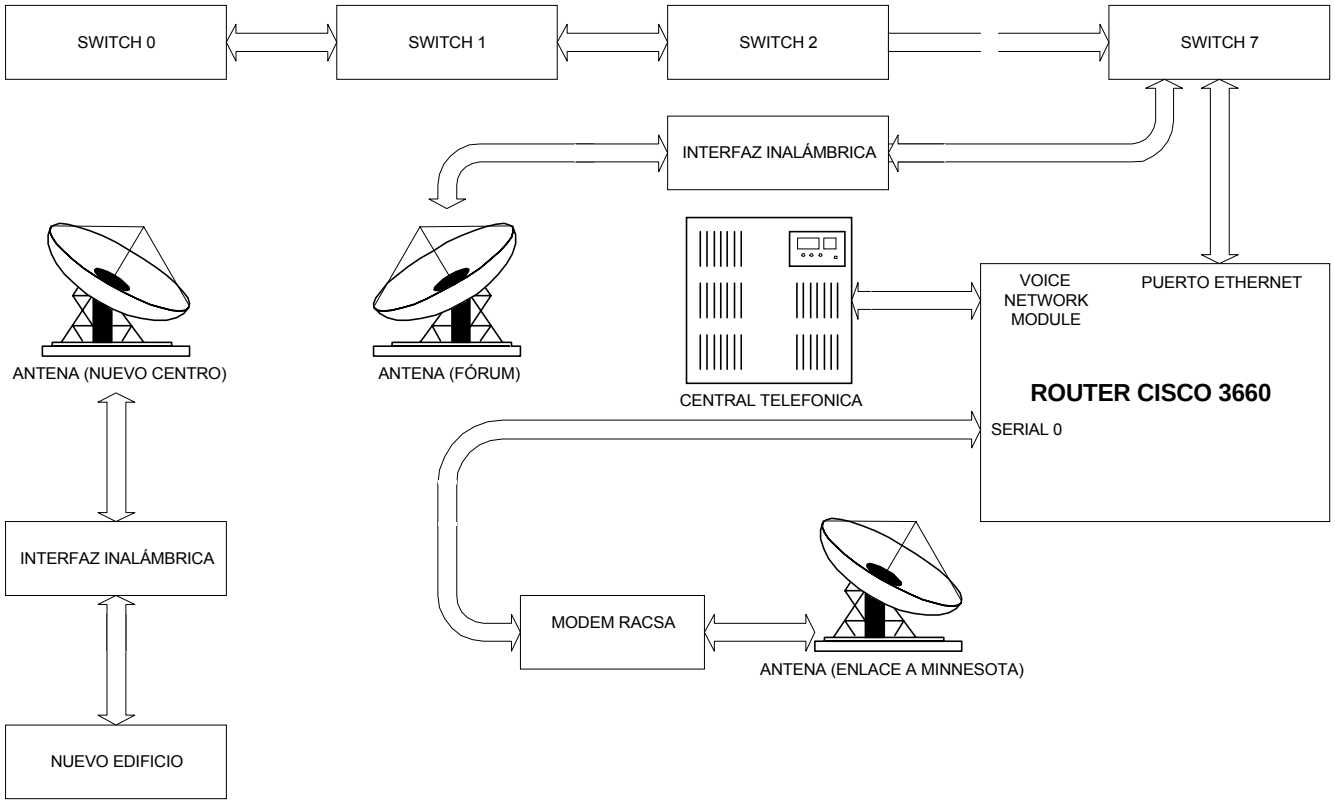
Al router se conectará la central telefónica de Unisys de Centroamérica. Para realizar esto, el router cuenta con un módulo de voz para red (Voice Network Module); este módulo permite la conexión de una central telefónica al router (la descripción detallada se realiza en el Capitulo 4); la interfaz serie del router mantiene su configuración original (enlace con Minnesota) y al switch 7 se conecta la interfaz inalámbrica para la realización del enlace (el bloque de interfaz inalámbrica contendrá un puente inalámbrico Aironet AIR - WBR342R). La principal diferencia que presenta el diagrama de la solución propuesta con el diagrama de la red actual, es la eliminación del Timeplex (multiplexor) puesto que ahora el router asume las funciones del multiplexor.

La figura 2.4 muestra un diagrama de la conexión de los equipos en el Nuevo Centro Logístico. El principal bloque de este diagrama el Router Cisco 3640. Al router se le conectará un teléfono además de una máquina de fax; ésto se puede lograr ya que el router cuenta con un Voice Network Module que permitirá que se le conecte el equipo mencionado (ver Capitulo 4).

El router mantiene su conexión con la red y se le agrega el bloque de interfaz inalámbrica para la realización del enlace (el bloque de interfaz inalámbrica ya ha sido especificado).

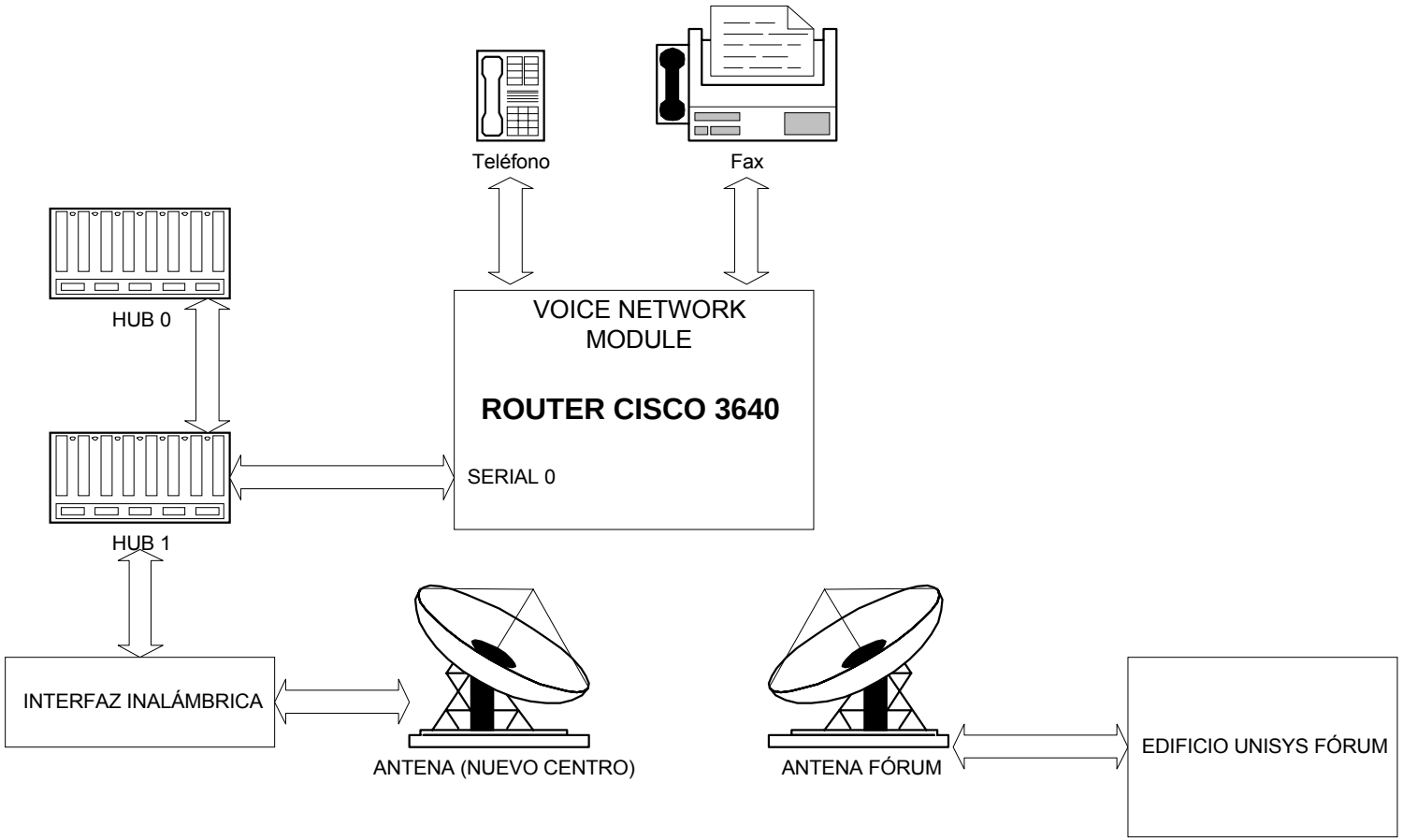
Además de las conexiones especificadas, se hace necesario reprogramar los routers para que soporten la transmisión de voz sobre IP (VoIP).






**Visio 4.5**

**Figura 2.3** Diagrama físico de la solución planteada




**Visio 4.5**

**Figura 2.4** Diagrama físico de la solución propuesta (Nuevo Centro)

# CAPÍTULO 3: PROCEDIMIENTO METODOLÓGICO

A continuación se detallan cada uno de los pasos que conforma la metodología:

### 3.1 Capacitación sobre los principios básicos de ambientes en redes y equipo activo de comunicaciones

En esta etapa se estudiaron conceptos generales sobre redes, el esquema de capas OSI, las funciones de cada capa y su interacción; tecnologías de transmisión de datos (ATM, Frame Relay), conceptos de transmisión de datos (paquetización y conmutación de circuitos); herramientas de Calidad de Servicio (QoS); conceptos de redes inalámbricas Esta etapa se realizo en Unisys, donde existe documentación al respecto.

La capacitación también estuvo orientada a familiarizar al estudiante con el equipo activo de comunicaciones en existencia en Unisys de Centroamérica, además de poner al tanto al estudiante con la nueva tecnología en existencia en el área de comunicaciones y redes. Esta actividad tuvo una duración de 6 semanas.

### 3.2 Determinación de los requerimientos del nuevo Centro Logístico

Consistió en la determinación, por parte del estudiante, en el tipo y clase de equipo que albergará el nuevo Centro Logístico. Esto se hizo necesario para poder determinar el ancho de banda requerido para realizar el enlace deseado. Esta actividad tuvo una duración de 1 semana.

### 3.3 Escoger diferentes métodos para determinar el ancho de banda para la transmisión de datos en el nuevo enlace

Esta etapa consistió en la investigación y evaluación de diferentes métodos que permitieran calcular, experimentalmente, el ancho de banda requerido para lograr la transmisión de datos en el nuevo enlace. El método evaluado fue la utilización del Packet Shaper, el cual es un administrador inteligente de ancho de banda (ver Capitulo 4). Esta actividad tuvo una duración de 1 semana.

### 3.4 Obtención del ancho de banda necesario para transmitir datos con el nuevo enlace, a partir del método óptimo

Una vez que se comprendió el funcionamiento del Packet Shaper, se procedió a conectarlo al enlace CLU con el fin de determinarla utilización del ancho de banda en existencia en este enlace. Se dejo conectado el equipo por un periodo de una semana con el fin de que recolectara datos acerca de la utilización del enlace.

### 3.5 Escoger diferentes métodos para determinar el ancho de banda para la transmisión de voz en el nuevo enlace

La explicación de este apartado coincide con la realizada para el punto 3.3

### 3.6 Obtención del ancho de banda necesario para transmitir voz con el nuevo enlace, a partir del método óptimo

Este punto coincide con la explicación realizada para el apartado 3.4

### 3.7 Evaluación del ancho de banda actual

Esta etapa consistió en un análisis del ancho de banda actual que emplea Unisys de Centroamérica. A partir de los resultados experimentales obtenidos en los pasos anteriores, se tuvieron bases para recomendar que se amplíe o que se mantenga igual el ancho de banda actual de Unisys de Centroamérica. Esta actividad tuvo una duración de 1 semana.

### 3.8 Evaluación del equipo actual

Esta etapa consistió en la evaluación del equipo actual con el que labora Unisys de Centroamérica. Basado en los resultados experimentales obtenidos en los pasos anteriores y en el ancho de banda actual con el que labora la empresa, se recomendó mantener el equipo de comunicaciones como se encuentra en la actualidad y se recomendó someterlo a actualización (software). Esta actividad tuvo una duración de 1 semana.

### 3.9 Selección del equipo activo de comunicaciones

Esta etapa consistió en la determinación del equipo activo de comunicaciones que se requerirá para realizar el enlace. Basado en los resultados experimentales obtenidos en los pasos anteriores y en el ancho de banda actual con el que labora la empresa, se recomendó el equipo de comunicaciones que se deberá emplear en el nuevo Centro Logístico para lograr el nuevo enlace. Esta actividad tuvo una duración de 1 semana.

### 3.10 Distribución física

Esta etapa consistió en la determinación de las características de la distribución física (del equipo) que se requerirá en el Nuevo Centro Logístico. Las características mencionadas comprenden los requerimientos del Nuevo Centro. Esta actividad tuvo una duración de 2 semanas.

# CAPÍTULO 4: DESCRIPCIÓN DEL HARDWARE UTILIZADO

Como ya se ha mencionado anteriormente, el estudio desarrollado ha involucrado tres herramientas principales en lo que a hardware se refiere. A continuación se dará una descripción más detallada de cada unas de estas herramientas.

**Router Cisco Serie 3600**

Los Router Cisco de la serie 3600 son una plataforma modular multifuncional que combina los servicios LAN – LAN, enrutamiento e integración de voz, vídeo y datos en el mismo dispositivo. En esta serie se incluyen los modelos 3660, 3640 y 3620.

El Router Cisco 3660 (ver figura 4.1) tiene 6 slots para módulos mientras que el 3640 cuenta con 4 (ver figura 4.2). Cada slot de módulo acepta una variedad de tarjetas de interface de módulos de red que soportan, a su vez, una gran variedad de tecnologías LAN y WAN. Este router cuenta con un módulos de 4 puertos seriales (ver figura 4.4). Cuando se emplea el cable de transmisión serial adecuado, cada puerto de este módulo puede proveer de las interfaces seriales EIA / TIA – 232, EIA / TIA – 449, V.35, X.21, DTE / DCE, EIA – 530 DTE, NRZ / NRZI, en cualquier combinación. El módulo serial permite tasa de datos sincrónicos de 8MB/seg en el puerto 0, 4 MB/seg en los puertos 0 y 2, ó 4 MB/seg en los cuatro puertos simultáneamente.

La serie 3600 ha sido provista de mayores herramientas para el manejo de voz: voz sobre Frame Relay (VoFR), voz sobre ATM (Asinchronous Transfer Mode) sobre las interfaces de voz digitales (E1 y T1), voz sobre IP (VoIP).

Para el manejo de la voz, se hace necesario que el router tenga instalado un Voice Network Module. En este módulo, se instalan las diferentes clases de tarjetas de interface de voz. Hay 3 tipos de estas tarjetas:

- FXS (Foreing Exchange Station): a este tipo de tarjeta (ver figura 4.7 y 4.8) se le puede conectar directamente un teléfono, una máquina de fax o un dispositivo similar. Esta interfaz provee el voltaje de repique, tono de marcado y también en la estación. Para conectar el teléfono se debe emplear un cable de teléfono estándar del tipo RJ –11. Esta tarjeta incluye dos puertos. El router 3660 no cuenta con este tipo de interfaz; el router 3640 tiene 2 tarjetas del tipo FXS instaladas.

- FXO (Foreing Exchange Office): esta tarjeta (ver figuras 4.9 y 4.10) interconecta las llamadas locales a una oficina central (PSTN: Public Switched Telephone Network) o a una PBX (central telefónica). Esta tarjeta tiene puertos del tipo RJ – 11.

- E & M (RecEive & transMit): esta es una técnica de señalización para interfaces de teléfono de dos o de cuatro cables. Este tipo de interfaz (ver figuras 4.11 y 4.12) típicamente conecta llamadas remotas, realizadas desde una red IP, a una central telefónica. Esta tarjeta tiene puertos de tipo RJ – 48C y no se debe conectar directamente a una línea telefónica. El router 3660 tiene instaladas 2 tarjetas de este tipo.

Del mismo modo, los router cuentan con módulos de red Ethernet (ver figura 4.5). El router 3660 tiene un módulo de 4 puertos de red Ethernet, mientras que el router 3640 tiene un módulo de un puerto (ver figura 4.6). El puerto de Ethernet 0 puede utilizar el conector AUI (Attachement Unit Interface) (DB – 15) o el conector 10BaseT (RJ – 45) que se encuentra a la par. Sólo uno de estos conectores se puede activar a la vez. Los puertos de Ethernet 1, 2 y 3 (en el módulo de 4 puertos) sólo utilizan conectores 10BaseT; éstos puertos no proveen conectores AUI. En el puerto 0, el módulo detecta automáticamente el tipo de conexión de red y no se requiere seleccionar el tipo de medio cuando se configura el software. Si se conectan cables en los conectores AUI y 10BaseT al mismo tiempo, se selecciona la conexión 10BaseT.

El router 3660 utiliza dos tipos de memoria (las cuales son reemplazables y actualizables): memoria SDRAM y memoria FLASH. La memoria FLASH (puede utilizar de 4 a 64 MB) es implementada con SIMMs, mientras que la memoria SDRAM (puede utilizar de 16 a 256 MB) utiliza DIMMs. El router 3640 emplea memoria FLASH (4 a 32 MB) y memoria DRAM (4 a 128 MB).

El router Cisco 3660 es el que se encuentra en existencia en Unisys de Centroamérica, mientras que en CLU se encuentra un router Cisco 3640.

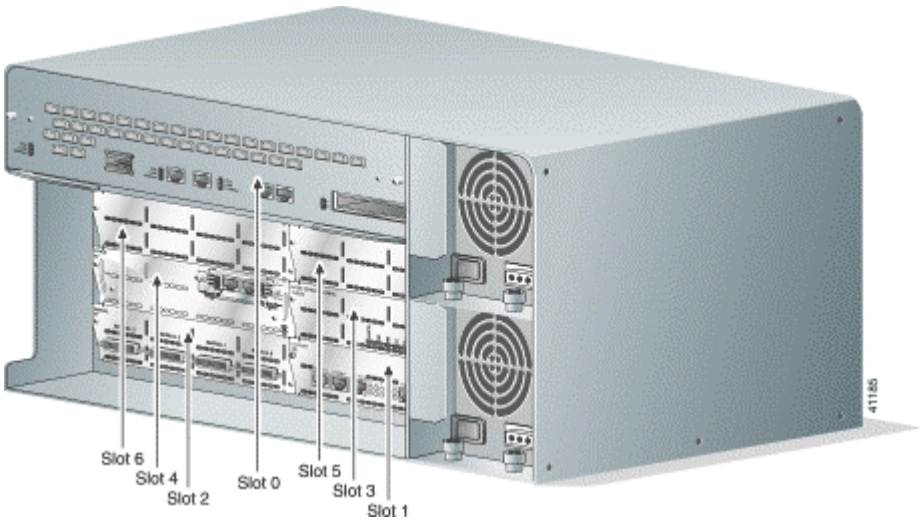



Figura 4.1 **Vista posterior del Router Cisco 3660**

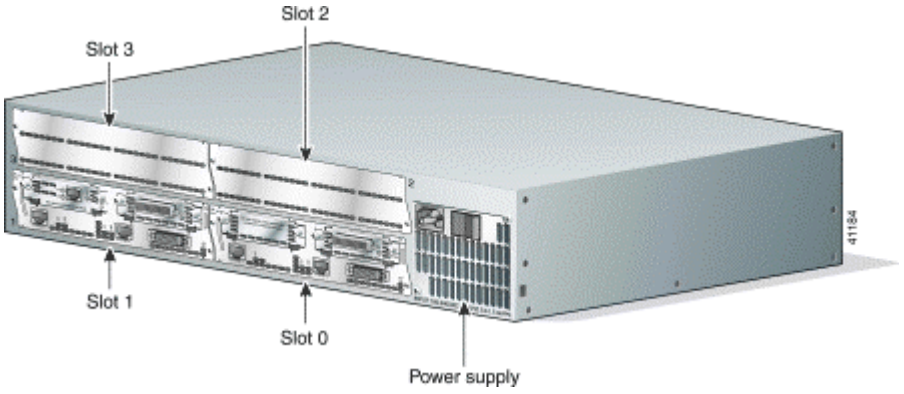



Figura 4.2 **Vista posterior del Router Cisco 3640**

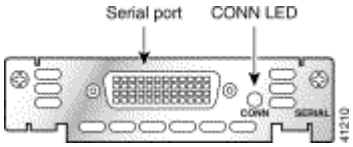



**Figura 4.3** Módulo serial de un puerto

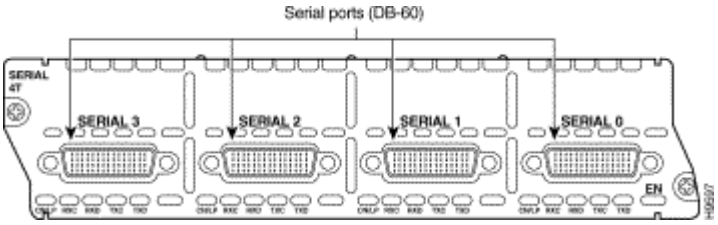



**Figura 4.4** Módulo serial de 4 puertos

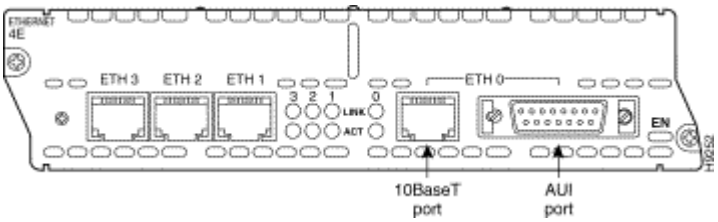



**Figura 4.5** Módulo de red Ethernet de 4 puertos

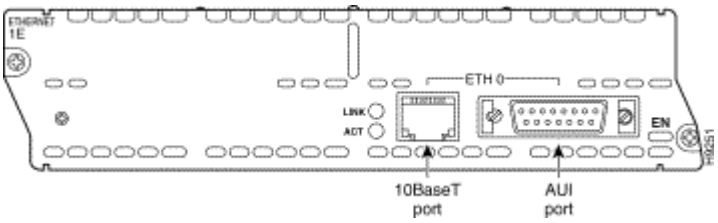



**Figura 4.6** Módulo de red Ethernet de 1 puerto

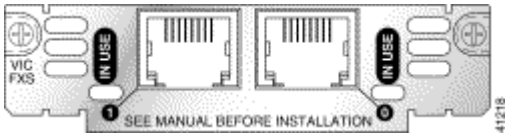



**Figura 4.7** Tarjeta de Interfaz de voz FXS

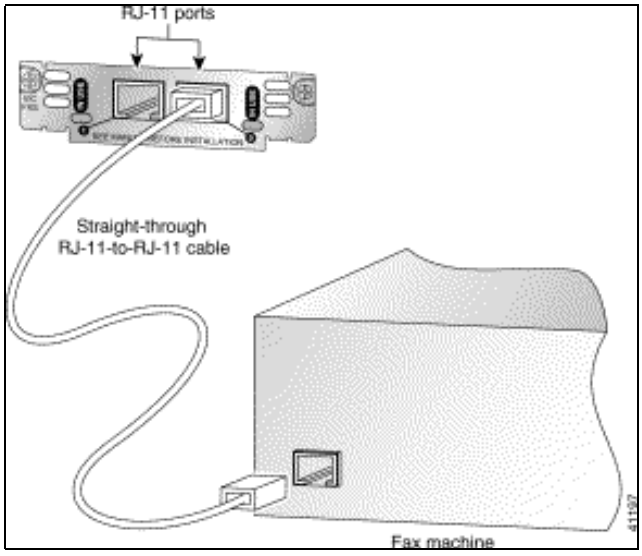



**Figura 4.8** Conexión de la tarjeta FXS

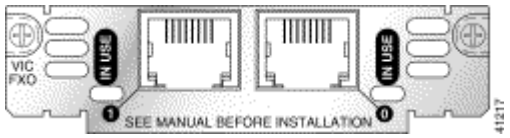



**Figura 4.9** Tarjeta de Interfaz de voz FXO

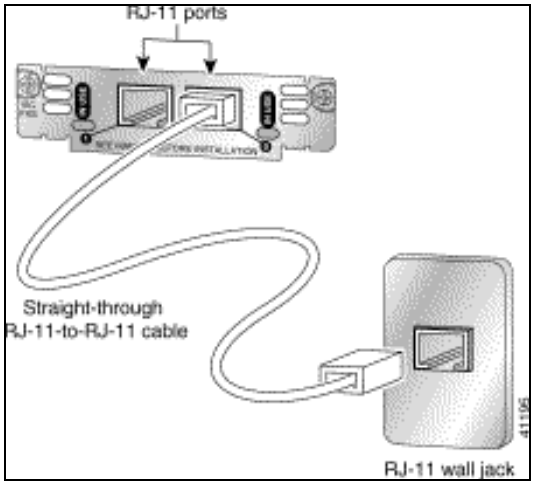



**Figura 4.10** Conexión de la tarjeta FXO

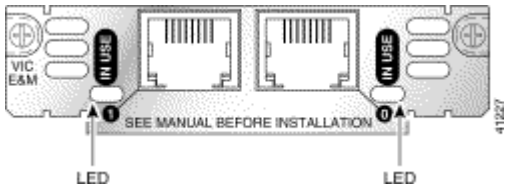



**Figura 4.11** Tarjeta de Interfaz de voz E & M

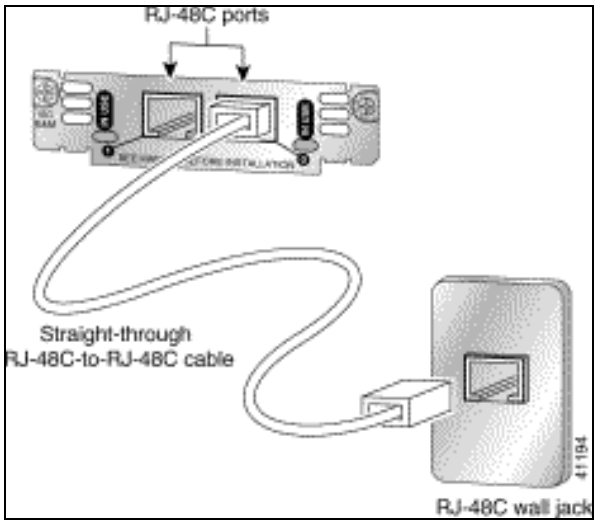



**Figura 4.12** Conexión de la Tarjeta E & M

**Packeteer (Intelligent Bandwidth Maganement)**

El Packetshaper es un dispositivo que, tal como su nombre lo indica, es un administrador inteligente de ancho de banda. Este dispositivo se conecta entre el router de acceso a la red y los servidores Web.

Una vez instalado, el Packetshaper se le asigna una dirección IP disponible de las que emplee la red a la que se ha conectado, se configura con la dirección IP del Router de la red así como del Gateway y se le asigna el ancho de banda total que emplea la red. De este modo, el Packeteer logra identificar los protocolos de los diferentes tipos de tráficos que se dan en la red (tanto entrantes – Inbound – como los salientes – Outbound - de la red). El Packeteer realiza y almacena un muestreo del ancho de banda que consume cada uno de los protocolos que emplea la red.

Del mismo modo, el Packeteer esta en la capacidad de generar gráficos de la utilización del enlace (ancho de banda total) de la red, indicando picos y valores promedios; puede generar gráficos acerca de la eficiencia de la red; gráficos del consumo de ancho de banda de cada protocolo; gráficos de tiempos de respuesta; gráficos de los protocolos más utilizados, etc. Del mismo modo, el Packeteer permite asignar anchos de banda (fijos o expansibles sí existe ancho de banda disponible) a los diferentes protocolos de la red; también se pueden asignar prioridades de aplicaciones, valores máximos y mínimos de ancho de banda, etc.

Este equipo se instaló en el CLU con el fin de realizar un estudio acerca de la utilización de los 64Kb del enlace entre el Edificio Principal de Unisys de Centroamérica y CLU. El equipo se dejó conectado por un periodo de una semana, al final de la cual se procedió a la revisión y estudio de los datos muestreados en ese periodo de tiempo. Este estudio permitió determinar las aplicaciones que más ancho de banda consumen en el enlace Unisys – CLU, también permitió determinar si el ancho de banda del enlace es suficiente o si hay que aumentarlo. A partir de estos resultados, se logró determinar un ancho de banda apropiado (tanto para la transmisión de voz como para la transmisión de datos) del nuevo enlace que se desea realizar.

Se puede obtener más información acerca de este dispositivo en la dirección de internet *www.packetshaper.com*

**Sistema Global de Posicionamiento**

Un GPS (Global Position System) es un dispositivo que permite a un usuario, desde cualquier punto de la tierra, determinar las coordenadas en las que se encuentra con un alto grado de precisión. Los GPS utilizan un sistema de satélites que les permiten determinar su ubicación en el globo terráqueo. A continuación se realiza una explicación acerca del funcionamiento de los GPS.

- **Triangulación**: la idea general detrás del GPS es utilizar los satélites en el espacio como puntos de referencia para ubicaciones aquí en la tierra. Esto se logra mediante una muy, pero muy exacta, medición de la distancia hacia al menos tres satélites, lo que permite "triangular" la posición en cualquier parte de la tierra.

- **Medición de distancias:** el GPS calcula la distancia entre el satélite con el que se está comunicando y él mismo. Esto lo hace midiendo el tiempo que tarda una señal emitida por el satélite en llegar hasta el receptor de GPS. Una vez obtenido el tiempo que tarda la señal en llegar, y sabiendo que en el vacío cualquier onda electromagnética viaja a la velocidad de la luz, es posible calcular la distancia a la que se encuentra el satélite. Para efectuar dicha medición se asume que ambos, el receptor GPS y el satélite, están generando el mismo Código Pseudo Aleatorio en exactamente el mismo momento. Comparando cuanto retardo existe entre la llegada del Código Pseudo Aleatorio proveniente del satélite y la generación del código del receptor GPS, se puede determinar cuanto tiempo le llevó a dicha señal llegar hasta el receptor. Multiplicando dicho tiempo de viaje por la velocidad de la luz y se obtiene la distancia al satélite.

- **Posicionamiento de satélites:** La altura de 20.000 km (de la tierra a los satélites) es en realidad un gran beneficio para este caso, porque algo que está a esa altura está bien despejado de la atmósfera. Eso significa que orbitará de manera regular y predecible mediante ecuaciones matemáticas sencillas. La Fuerza Aérea de los EEUU colocó cada satélite de GPS en una órbita muy precisa, de acuerdo al Plan Maestro de GPS. En tierra, todos los receptores de GPS tienen un almanaque programado en sus computadoras que les informan donde está cada satélite en el espacio, en cada momento. Las órbitas básicas son muy exactas pero con el fin de mantenerlas así, los satélites de GPS son monitoreados de manera constante por el Departamento de Defensa.

Existen muchos modelos de GPS. Para las mediciones realizadas se utilizó un GPS Magellan 300 (como el que se muestra en la figura 4.13).


**Figura 4.13** Vista frontal del GPS Magellan 300

# CAPÍTULO 5: DESARROLLO DE SOFTWARE

## 5.1 Configuración de voz sobre IP (VoIP)

Existen diferentes factores que deben tenerse en cuenta cuando se desea transmitir voz sobre IP. Entre estos factores se pueden mencionar los siguientes:

- Delay: se refiere al tiempo que toma a los paquetes de voz viajar entre dos puntos. Las redes de hoy día están diseñadas para minimizar este tiempo, sin embargo siempre se debe esperar un retardo. El oído humano normalmente acepta un delay no mayor de 150 ms, por lo que una conversación telefónica con un delay mayor a los 150 ms puede volverse ininteligible.

- Jitter: se refiere a delays de diferente longitud. Este también es un factor que puede producir que una conversación se vuelva ininteligible.

- Serialización: este es un término que describe lo que ocurre cuando un router intenta enviar tanto voz como datos (en paquetes) fuera de una interfaz. Generalmente, los paquetes de voz son muy pequeños (80 a 256 bytes) mientras que los paquetes de datos son muy grandes (1500 a 18000 bytes). En enlaces lentos (como conexiones WAN), paquetes grandes de datos pueden demorar mucho tiempo en transmitirse; cuando estos paquetes se mezclan con paquetes de voz más pequeños, el exceso de tiempo de transmisión puede producir delay y jitter. Para eliminar esto se utiliza la fragmentación para reducir el tamaño de los paquetes.

- Consumo de ancho de banda: conversaciones tradicionales consumen 64 Kb de ancho de banda (equivalente en una red). Cuando se realiza este tráfico a través de una red IP, este ancho de banda se puede comprimir y digitalizar (a través de procesadores digitales de señales) hasta 5.3 Kb. Una vez que estos paquetes son enviados a la red IP, los encabezados (IP/UDP/RTP) se agregan (alrededor de 40 bytes por paquete de voz). Otras tecnologías como RTP Header Compression pueden comprimir estos encabezados a 2 bytes (ver figura 5.1); VAD (voice activity detection) no envía paquetes de voz a menos que la conversación esté activa.

Existen una serie de pasos para configurar voz sobre IP en un router Cisco de la serie 3600. Estos pasos son los siguientes:

### 5.1.1 Configurar la red para que soporte tráfico de voz en tiempo real

Esta configuración se logra a partir de la activación y configuración de diferentes herramientas de calidad y servicio (QoS) (se detallan las herramientas que se aconseja emplear para la realización del nuevo enlace):

- RSVP: debido a que los paquetes de voz son de longitud variable y se transmiten por ráfagas, este protocolo que trocea los paquetes de datos grandes y da prioridad a los paquetes de voz en el caso de que haya una congestión en el router. Se aconseja la utilización de esta herramienta si la implementación de voz es a pequeña escala, se busca la mejor calidad de voz, hay enlaces de alta utilización o enlaces menores a 2 Mbps. Se aconseja la utilización de esta herramienta en ambos routers (para realizar el nuevo enlace inalámbrico si el ancho de banda es menor a los 2 Mbps).

Esta herramienta se puede programar mediante el comando

***ip rsvp bandwidth*** [*interfaz-kbps*] [*flujo-simple-kbps*]

El valor interfaz se reemplaza con el máximo ancho de banda que se desea reservar para el tráfico de voz; el valor flujo-simple indica el máximo ancho de banda que puede consumir una sola conversación (este valor puede tener un máximo de un 75% del valor interfaz. RSVP es la única herramienta de QoS que se aconseja emplear para la realización del enlace inalámbrico deseado (si el enlace logrado es menor a los 2 Mbps). Existen otras herramientas que sólo se mencionarán, pero no se describirán pues no es necesario emplearlas para el mencionado enlace:

- Multilink PPP: se emplea únicamente en interfaces multienlace.

- RTP Header Compression: se emplea para interfaces seriales (ver descripción del apartado "consumo de ancho de banda" en la parte inicial de este capítulo). Se aconseja emplear esta herramienta en enlaces lentos y si se necesita ahorrar ancho de banda.

- Queuning: se emplea para definir prioridades a cierto tipo de tráfico. Se aconseja emplear esta herramienta en redes con un ancho de banda congestionado.

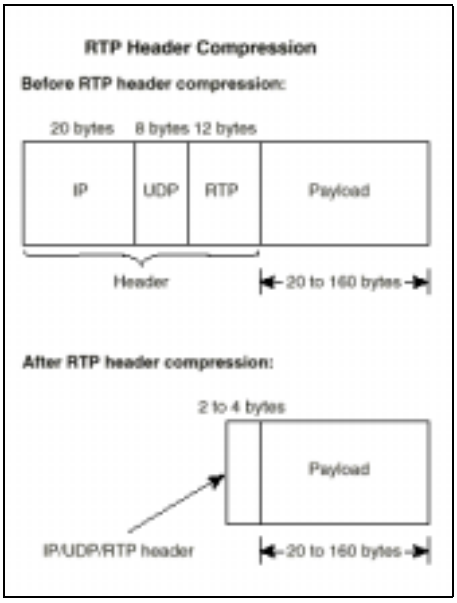



**Figura 5.1** RTP Header Compression

### 5.1.2 Dial peers

Para entender el funcionamiento de VoIP, primero que todo se debe entender lo que se denomina ***dial peers***. Cada ***dial peers*** define las características de cada "segmento de llamada". Un segmento de llamada (call leg) es un segmento discreto de una conexión en la que se transmite voz (ver figura 5.2); dicho segmento yace entre dos puntos de la conexión. Cada uno de los segmentos de llamada, para una conexión particular, tienen el mismo identificador (ID).

Como se observa en la figura 5.2, una llamada completa está compuesta de 4 segmentos de llamada (dos desde la perspectiva del emisor y dos desde la perspectiva del destino). Los ***dial peers*** se emplean para colocar aplicar atributos a los segmentos de llamada y para identificar el origen y el destino. Los atributos que se aplican a los segmentos de llamada incluyen QoS (calidad y servicio), CODEC, VAD y fax rate.

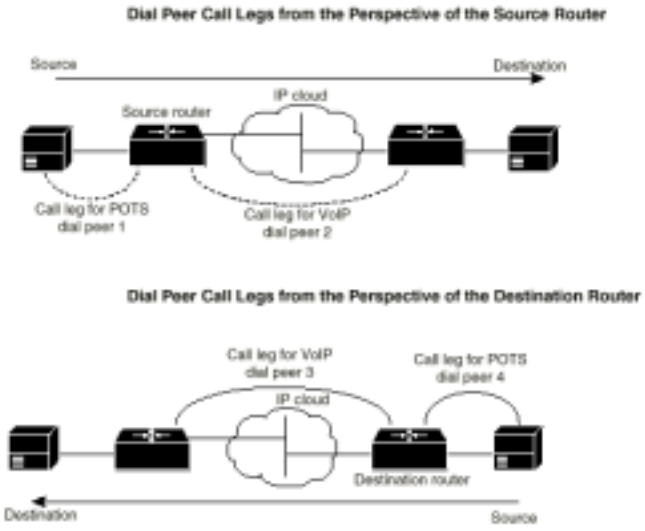



**Figura 5.2** Segmentos de llamada vistos desde el router emisor (source) y el router de destino (destination).

Existen dos clases de ***dial peers***:

- **POTS (Plain Old Telephone Service):** describe las características de una conexión de red de telefonía tradicional. POTS peers apuntan a un puerto particular de voz en un dispositivo de red para voz y permiten que las llamadas entrantes sean recibidas por un dispositivo telefónico particular. Para configurarlos, se requiere identificar al ***dial peer*** con un número único al que se les denomina *tags* (son números enteros que abarcan un rango que va desde 1 hasta $(2^{31} – 1)$. Los *tags* en un router deben ser únicos, pero se pueden reutilizar en otro router.

- **VoIP**: describe las características de una conexión de red de paquetes (en el caso de Voz sobre IP, se trata de una red IP). VoIP apunta a un dispositivo específico de voz sobre IP (puesto que asocia los números de teléfono de destino con una dirección IP específica), de modo que las llamadas entrantes puedan ser recibidas y las llamadas salientes puedan ser enviadas.

Se aconseja la creación de una tabla en la que se muestren los *tags* que se empleen para los diferentes VoIP y POTS, además de incluír los números de teléfono de destino y los atributos asignados.

La tabla 5.1 muestra una lista (y la descripción) de los comandos que se emplean para configurar VoIP.

**Tabla 5.1** Comandos empleados en la configuración de voz sobre IP

| Comando | Propósito |
|---|---|
| **dial-peer voice** *número* **pots** | Entra al modo de configuración dial-peer para configurar un POTS (*número* se reemplaza con el tag asignado) |
| **Destination-pattern** *cadena* | Define el número de teléfono asociado con el POTS (*cadena* se reemplaza con el número telefónico asociado al dial peer).<br>La *cadena* es el número telefónico especificado de acuerdo con el estándar E.164 |
| **port** *número de puerto* | Asocia el dial peer con una interfaz de voz específica. |
| **dial-peer voice** *número* **voip** | Entra al modo de configuración dial-peer para configurar un VoIP. |
| **Ip precedence** *número* | Selecciona el nivel de precedencia para el tráfico de voz asociado con el dial peer.<br>- "número" es un valor entre 1 y 7; los números de 1 a 5 identifica clases para flujos IP; los números 6 y 7 se emplean para actualizaciones de red y enrutamiento de backbone. La precedencia es ascendente.<br><br>Este comando se debe emplear si no se ha habilitado RSVP y se desea dar mayor prioridad a los paquetes de voz. |

| | |
|---|---|
| **req-qos** [ **best-effort** \| **controlled-load** \| **guaranteed-delay** ] | Especifica la calidad deseada de servicio.<br>Se aconseja emplear la opción **controlled-load** |

**Continuación de la tabla 5.1**

| | |
|---|---|
| | (asegura preferencia a un tipo de tráfico en el caso de que el ancho de banda se sature).<br>- **best-effort:** indica que no se reseva ancho de banda<br>- **guaranteed-dealy:** indica que se mantiene una tasa de datos mínima y se realiza queueining si no se excede el valor de ancho de banda reservado. |
| **codec** [ **g711alaw** \| **g711ulaw** \| **g729r8** ] | Coder – Decoder.<br>Determina cuanto ancho de banda emplea la sesion de voz.<br>g729r8 (8000 bps) es el valor predeterminado (es el más deseado); para redes con bajo ancho de banda y con calidad de voz de muy alta importancia, se debe emplear alguna de las otras opciones. Así se logrará la mejor calidad de voz pero se consumirá un mayor ancho de banda.<br>Las otras dos opciones tienen tasas de 64000 bps. |
| **Vad** | Voice Activity Detection<br>Desactiva la transmisión de paquetes de silencio.<br>El valor predeterminado de este comando es habilitado; si se requiere una mejor calidad de voz, se debe deshabilitar **vad** pero se consumirá más ancho de banda. |

## 5.2 Programación de las interfaces FXS en los Routers Cisco 3640

- El primer paso paso para realizar la programación de voz sobre IP, del Router Cisco 3640, es crear una tabla en la que se asignen los números de teléfono que se desean utilizar con el router (estos números son asignados por el programador y nunca por la compañía telefónica), el patrón de destino y de la asignación de los *dial peer pots*.

**Tabla 5.2** Router 3640 dial peers locales

| Número de teléfono | Destination Pattern | Puerto de Voz (Router) | Dial peer |
|---|---|---|---|
| 204 - 0000 | 506 – 2040000 | 1/0/0 | 001 |
| . | . | 1/0/1 | 002 |
| . | . | 1/1/0 | 003 |
| 204- 0003 | 506 – 2040003 | 1/1/1 | 004 |

Una vez realizada esta tabla, se pueden programar los números de teléfonos en el router con la siguiente rutina de programación (se supone que el router se llama 3640):

3640 **global config**

```
3640(config)# dial-peer voice 001 pots
3640(config-dial-peer)# dest-pat +5062040000
3640(config-dial-peer)# port 1/0/0
      .
      .
      .
3640(config-dial-peer)# exit
3640(config)#
```

Se deben repetir las líneas 2 – 4 para programar cada puerto de voz en el router.

Los números de teléfono se pueden asignar por extensiones (con la cantidad de de dígitos deseada) o con la secuencia completa de siete números establecida de acuerdo con el área del país (para que el número indique el área a la que se está llamando). Esta asignación de números servirá de guía para saber a dónde se está llamando; estos números no tienen nada que ver con los empleados por las líneas telefónicas convencionales. Los números empleados en este documento son a modo de ejemplo; no tienen que ser necesariamente los que se utilicen en la programación real.

- Las centrales telefónicas (PBX), generalmente, están configuradas de modo que un usuario puede realizar una llamada local (en la misma central telefónica) marcando sólo una porción del número telefónico en vez del número completo. En la telefonía sobre IP también se puede programar esta facilidad de acceso mediante el comando **num-exp** (expansión de número). A continuación se muestra un ejemplo de la utilización de este comando

Para expandir la extensión 0000 en el número completo incluída el código de país, se debe escribir la siguiente línea al programar el router:

3640(config)**# num-exp 0000 +5062040000**

Para expandir 4141 en 506204-0000, se debe escribir:

3640(config)**# num-exp 4141 +5062044141**

Se pueden emplear periodos (..) en lugar de escribir los números completos:

3640(config)**# num-exp .... +506204....**

Este comando expande cualquier secuencia de números de 4 dígitos precediéndole la secuencia 506204.

Para emplear extensiones de 5 números (que empiecen con el número 0), se debe emplear el siguiente comando:

3640(config)**# num-exp 0.... +1408555....**

Este comando permite utilizar extensiones de cinco dígitos en lugar de cuatro.

Luego de realizar esta programación, se debe poder realizar llamadas entre los teléfonos conectados al mismo router. El comando **show num-exp** permitirá verificar si los datos programados son correctos.

- Para lograr la comunicación entre las extensiones telefónicas que se encuentran conectadas al router 3640 y la central telefónica en el Edificio Principal de Unisys en Fórum, se necesita programar al router 3640 asociándole un ***dial peer remoto*** (voip) además de indicarle la dirección IP del router en Fórum (comando **session-target**):

```
3640(config)# dial-peer voice 001 voip
3640(config-dial-peer)# dest-pat +506204....
3640(config-dial-peer)# session-target ipv4:direcciónip.de.Unisys.Fórum
```

### 5.3 Programación de las interfaces E&M en los Routers Cisco 3660

- De modo similar a la programación del router 3640, al router 3660 se le debe definir un ***dial peer local***, un patrón de destino y el puerto que se está empleando:

```
3660(config)# dial-peer voice 111 pots
3660(config-dial-peer)# dest-pat +506204....
3660(config-dial-peer)# port 1/0/0
```

- El siguiente paso consiste en programar los números telefónicos que se están empleando en el router 3640. Se debe crear una tabla similar a la tabla 5.2, pero también se pueden emplear periodos para facilitar la programación:

**Tabla 5.3** Router 3660 dial peers remotos

| Número de teléfono | Destination Pattern | Dirección IP | Dial peer |
|---|---|---|---|
| 506204-XXXX | +506204…. | dir.ip.router.3640 | 111 |

Luego, las líneas de código para programar al router son:

```
3660(config)# dial-peer voice 111 voip
3660(config-dial-peer)# dest-pat +506204....
3660(config-dial-peer)# session-target ipv4:dir.ip.router.3640.
3660(config-dial-peer)#exit
3660(config)# num-exp 0…. +506204….
```

La utilización de los periodos permite que con sólo estas líneas de código, la programación del router sea completa para que se puedan realizar llamadas a cualquiera de los teléfonos conectados al router 3640.

- El último paso consiste en configurar el puerto E&M utilizando los comandos necesarios. Un ejemplo de configuración sería:

```
3660(config)# voice-port 1/0/0
3660(config-voice-port)# signal immediate
3660(config-voice-port)# operation 4-wire
3660(config-voice-port)# type 2
```

Cada uno de estos parámetros o comandos se ha definido anteriormente en el apartado 5.1.1. Los valores asignados por los comandos dependen del tipo de central telefónica que se esté conectando.

## 5.4 Configuración de los puertos de voz FXS

Generalmente, los valores de la configuración de los puertos de voz son adecuados para el funcionamiento de las interfaces de voz del router (FXO, FXS, E&M). Para cambiar la configuración de los puertos de voz, se hace necesario realizar las siguientes tareas:

1. Identificar el puerto de voz (comando **voice-port**) y entrar al modo de configuración de puerto de voz (comando **configure terminal**).
2. Configurar los siguientes parámetros:

- Tipo de señal (comando **signal**)
- Tono de llamada en progreso (comando **cptone**)
- Frecuencia de repique (comando **ring frequency**)
- Configurar uno o más parámetros opcionales:
  - Modo de conexión PLAR (comando **connection plar**)
  - Nivel de música (comando **music-threshold**)
  - Descripción (comando **description**)
  - Tono de fondo (comando **comfort-noise**)

Existen otros parámetros configurables en la interface FXS. Estos parámetros son llamados de ajuste fino. Generalmente, los valores predeterminados de estos parámetros son los adecuados para que la interface funcione correctamente luego de configurar los parámetros principales. (estos parámetros de ajuste fino no se detallan en este documento. Dichos parámetros y comandos relacionados se muestran en el Anexo 3).

La Tabla 5.4 describe la sintaxis de los comandos que permiten configurar los parámetros principales.

**Tabla 5.4** Descripción de los comandos necesarios para configurar la tarjeta de interfaz FXS

| Comando | Propósito |
|---|---|
| **voice-port** *No.slot / No.subunidad / puerto* | Identifica al puerto de voz que se desea configurar. |
| **configure terminal** | Entra al modo de configuración globlal. |
| **signal** { l**oop-start \| ground-start** } | Selecciona el tipo de señal apropiado para este tipo de interface.<br>**loop-start**: sólo un lado de la conexión puede colgar.<br>**ground-start**: cualquiera de los lados de la conexión puede empezar o terminar una conversación. |
| **Cptone** *country* | Selecciona el tono de llamada en progreso apropiado para esta interfaz (especifica una interfaz de voz analógica regional, tono, timbrado y cadencia para un puerto de voz específico). El valor predeterminado es US (se puede seleccionar otra región). |
| **Ring frequency** { **25 \| 50** } | Selecciona el valor de la frecuencia de repique en Hz (25 ó 50). |
| **connection plar** *cadena* | Especifica la conexión para Private Line Auto Ringdown (se asocia un peer directamente con una interfaz; cuando se descuelga la interfaz, el peer conecta el segundo segmento de llamada con el número de destino sin la necesidad de |

**Continuación de la tabla 5.4**

| | |
|---|---|
| | que la persona que llama tenga que marcar ningún número). La *cadena* especifica el número de teléfono de destino. |
| **music threshold** *número* | Especifica el volumen de la música de espera (en dB). Una entrada válida se encuentra entre –70 y –30. |
| **description** *cadena* | Agrega texto descriptivo acerca de esta conexión de puerto de voz (permite escribir de 1 a 255 caracteres). |
| **comfort- noise** | Especifica que se generará música de fondo. |

## 5.5 Configuración de los puertos de voz E & M

Generalmente, los valores de la configuración de los puertos de voz E & M son adecuados para el funcionamiento de la interfaz. Para cambiar la configuración de los puertos de voz, se hace necesario realizar los siguientes tareas:

1. Identificar el puerto de voz (comando **voice-port**) y entrar al modo de configuración de puerto de voz (comando **configure terminal**).
2. Configurar los siguientes parámetros:
   - Tipo de dial (comando **dial-type**)
   - Tipo de señal (comando **signal**)
   - Tono de llamada en progreso (comando **cptone**)
   - Operación (comando **operation**)
   - Tipo (comando **type**)
   - Impedancia (comando **impedance**)
   - Configurar uno o más parámetros opcionales:
     - Modo de conexión PLAR (comando **connection plar**)
     - Nivel de música (comando **music-threshold**)
     - Descripción (comando **description**)
     - Tono de fondo (comando **comfort-noise**)
4. Existen otros parámetros configurables en la interface E&M. Estos parámetros son llamados de ajuste fino. Generalmente, los valores predeterminados de estos parámetros son los adecuados para que la interfaz funcione correctamente luego de configurar los parámetros principales. (estos parámetros de ajuste fino no se detallan en este documento. Dichos parámetros y comandos relacionados se muestran en el Anexo 3).

La Tabla 5.5 describe la sintaxis de los comandos que permiten configurar los parámetros principales.

Luego de configurar el puerto de voz, se necesita activar este puerto para poder utilizarlo. De hecho, se aconseja apagar el puerto y luego volver a encenderlo (ciclarlo). Para activar el puerto de voz se emplea el comando **no shutdown** (se debe utilizar en el modo configuración).

Para ciclar el puerto se deben utilizar los siguientes comandos (en la secuencia especificada):

- **shutdown** : desactiva el puerto de voz
- **voice-port** : se emplea con la misma sintaxis especificada anteriormente.
- **no shutdown** : activa el puerto de voz.

**Tabla 5.5** Descripción de los comandos necesarios para configurar la tarjeta de interfaz E&M

| Comando | Propósito |
|---|---|
| **voice-port** No.slot / No.subunidad / puerto | Identifica al puerto de voz que se desea configurar. |
| **configure terminal** | Entra al modo de configuración globlal. |
| **signal** { **dtmf \| pulse** } | Selecciona el tipo de señal apropiado para la marcación externa (pulso o tonos). |
| **cptone** country | Ver descripción de la tabla 5.4 |
| **num-exp** número1 número2 | Define cómo expandir un set específico de números en un patrón de |

| | |
|---|---|
| | destino específico.<br>“número1” indica la extensión con la que se desea establecer una comunicación.<br>“número2” indica el número en que se va a expandir la extensión marcada para lograr la comunicación (el número se especifica según el formato E.164). |
| **operation** { **2-wire \| 4-wire** } | Selecciona el cableado apropiado para este puerto de voz (depende de la central telefónica). |
| **type** { **1 \| 2 \| 3 \| 5** } | El tipo 1 indica lo siguiente:<br>E – salida, relacionada a tierra<br>M – entrada, referenciada a tierra |

**Continuación de la tabla 5.5**

| | |
|---|---|
| | El tipo 2 indica lo siguiente:<br>E – salida, relacionada a SG<br><br>M – entrada, referenciada a tierra<br>SB – alimentación para M, conectada a –48 V.<br>SG – retorno para E, aislada galvánicamente de tierra. |
| | El tipo 3 indica lo siguiente:<br>E – salida, relacionada a tierra<br>M – entrada, referenciada a tierra<br>SB – conectada a –48 V.<br>SG – conectada a tierra.<br>El tipo 5 indica lo siguiente:<br>E – salida, relacionada a SG |

| | |
|---|---|
| | M – entrada, referenciada a –48 V |
| l**mpedance** { 600c \| 600r\| 900c \| complex1 \| complex2 } | Especifica el valor de la impedancia terminal. Este valor debe coincidir con la especificación de la central telefónica.<br>La “r” indica que es un valor real; la “c” indica que es un valor complejo; *complex1* y *complex2* indican Complex 1 y Complex 2 respectivamente. |
| **connection plar** *cadena* | Ver descripción de la tabla 5.4 |

**Continuación de la tabla 5.5**

| | |
|---|---|
| **music threshold** *número* | Especifica el volumen de la música de espera (en dB). Una entrada válida se encuentra entre –70 y –30. |
| **description** *cadena* | Agrega texto descriptivo acerca de esta<br><br>conexión de puerto de voz. |
| **comfort- noise** | Especifica que se generará música de fondo. |

## 5.6 Configuración del Puente Inalámbrico AIR – WGBR342R

El puente inalámbrico permitirá unir la red del Nuevo Centro Logístico con la red del Edificio Principal de Unisys de Centroamérica (se requerirá de dos puentes, uno en cada edificio). Este puente está en capacidad de transmitir la paqueteria TCP/IP con un ancho de banda de hasta 11 Mbps. Como regla general, cuanto mayor sea el ancho de banda del enlace, menor será el alcance de la señal del puente (idealmente, la señal tiene un alcance de hasta 15 Km); del mismo modo, el alcance de la señal del puente disminuirá en la misma proporción de la altura de la antena que se utilice y se debe mantener *"linea vista"* entre las antenas de los puentes (esto significa que se debe si se traza una línea recta imaginaria entre ambas antenas, dicha línea debe estar libre de obstáculos).

Este puente emplea una encriptación de 128 bits (Same System Identifier – **SSID**). Para que estos dispositivos se puedan comunicar entre sí, todos los equipos deben utilizar el mismo SSID, sino, no se podrán comunicar entre ellos; la potencia de la señal de salida es de 100 mW, permite la conexión de una antena externa y emplea la técnica de **Modulación DSSS** (Direct Sequence Spread Spectrum), la cual fue desarrollada inicialmente con propósitos militares y es una tecnología con muy baja probabilidad de intercepción.

Como ya se mencionó, se utilizarán dos puentes: uno en el edificio en Fórum y otro en el Nuevo Centro Logístico. El puente que se deberá instalar en el Edificio en Fórum se utilizará como ***root*** (está conectado al segmento principal de red) y con el otro puente (que estará en el Nuevo Centro Logístico) se realizará una conexión de punto a punto (se enlazarán los dos segmentos de red). Un diagrama de conexión punto a punto se muestra en la figura 5.3. Como se observa, mediante la utilización de los puentes inalámbricos se logra la comunicación entre los dos segmentos de red. Cada segmento de red accesa la información que necesita de su respectivo servidor (a través de la red alambrada); sólo en el caso de que se requiera el intercambio de información entre ambos segmentos de red, se emplea el enlace inalámbrico.

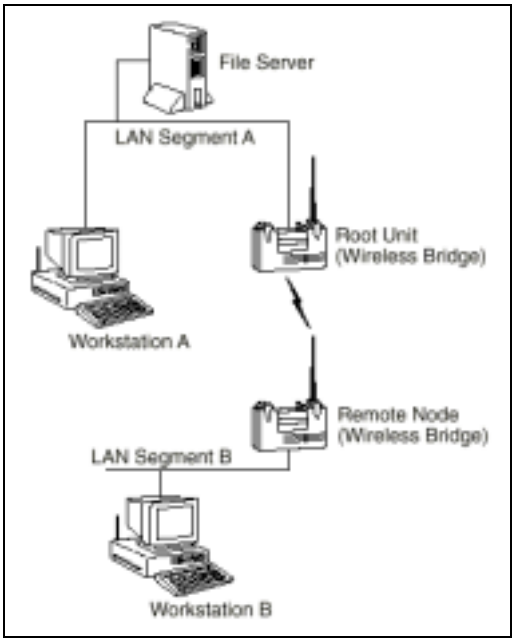



**Figura 5.3** Conexión punto a punto empleando puentes inalámbricos

La figura 5.4 muestra una vista general de los puentes de la serie Aironet. Cada puente posee tres conectores de red (estos puentes se conectan directamente al switch o al hub de la red): un conector 10BaseTm (cable twisted pair), 10Base5 (puerto AUI), 10Base2 (conector BNC -T). Además también tiene un conector para la antena externa, un puerto RS-232 para la conexión de una consola externa (para realizar la configuración), leds indicadores, botón de encendido/apagado y una unidad de alimentación AC/DC.

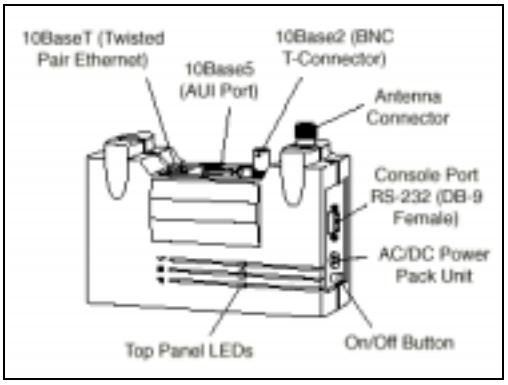



**Figura 5.4** Vista general del puente de la serie Aironet

En el Anexo 4 se muestran las instrucciones para la conexión de la antena externa y protección contra rayos.

La configuración inicial del puente se debe realizar a través del puerto de consola mediante un programa de comunicación serial (9600 bps, 1 start bit, 1 stop bit, sin paridad, 8 bits datos y compatibilidad ANSI). Una vez que ya se ha instalado el puente, se puede variar la configuración también mediante TELNET, SNMP ó HTML (con Internet Explorer o Netscape Navigator) mediante su dirección IP.

Al realizar la configuración inicial mediante el puerto de consola, una vez que se ha establecido la comunicación, se muestra la siguiente pantalla:

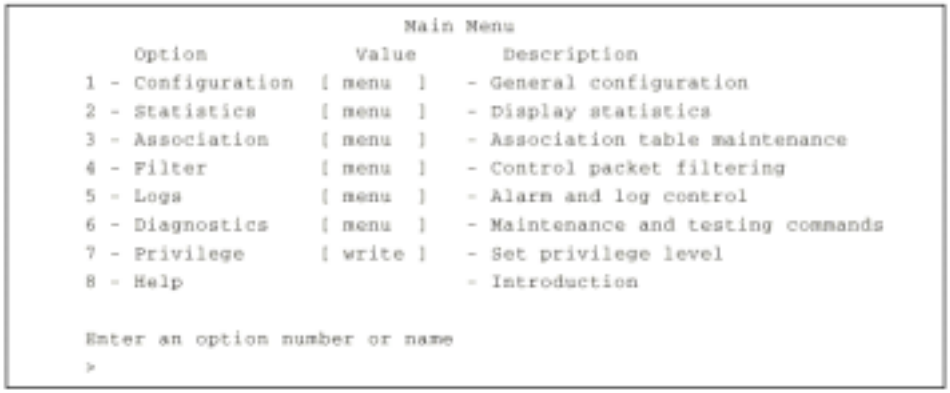



**Figura 5.5** Pantalla incial de configuración del puente Aironet AIR-WGB342R

A continuación se realiza una breve descripción acerca de cada una de las opciones del menú principal:

- **Configuration**: permite establecer los parámetros de Ethernet y de la comunicación, estableciendo identificadores de red.
- **Statistics**: permite ver una variedad de información estadística del puente (datos enviados y recibidos, errores generados, estatus normal del puente.
- **Association**: es una tabla que contiene la dirección de todos los nodos (que emplean señales de RF) que se encuentran conectados a un nivel menor que el del puente. Esta tabla permite ver dichas direcciones, borrar y quitar entradas estáticas y permitir la detección automática de adiciones a la tabla.
- **Filter**: controla el filtrado de la paquetería; permite controlar el envío de mensajes multicast bloqueando aquellos protocolos no empleados en la red RF (Radio Frequency).
- **Logs**: guarda un archivo acerca todas las alarmas que ocurren en la unidad; permite establecer niveles de alarma e imprimir el archivo antes mencionado.

- **Diagnostics**: permite realizar pruebas diagnóstico (de transmisión) entre la unidad Aironet y algún otro nodo de la infraestructura para probar la calidad del radio enlace. También permite descargar nuevas versiones del software del puente.
- **Privilege**: permite establecer niveles de privilegio y passwords para accesar la configuración del puente.
- **Help**: pantalla de ayuda de comandos, menús, etc.

La figura 5.6 muestra el menú que se despliega al seleccionar la opción **Configuration** del menú principal. A continuación se realiza una descripción de la configuración básica del puente. Sólo se describirán en detalle las configuraciones **Radio** y **Ethernet** (ya que son básicas para lograr una configuración adecuada).

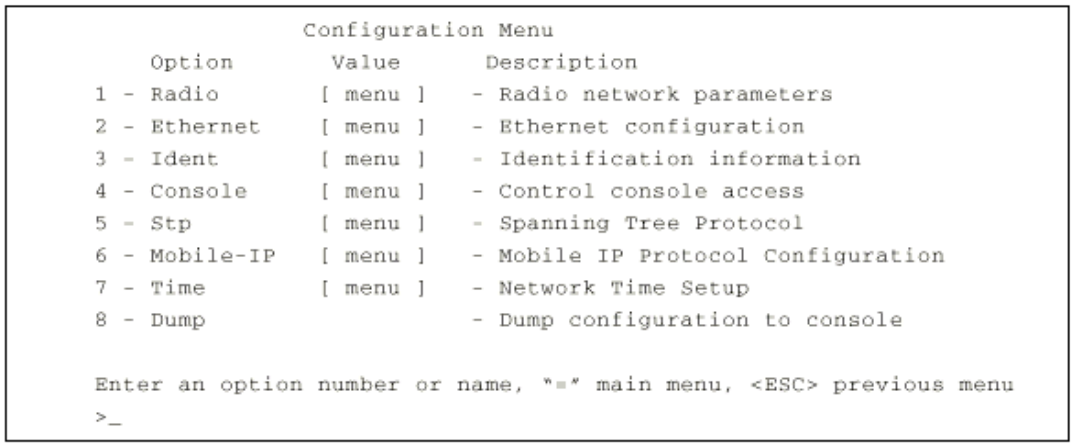



**Figura 5.6** Submenú Configuration

### 5.6.1 Configuración de la red de radio (Radio Network)

Esta configuración se realiza seleccionando la opción ***Configuration*** del menú principal y luego la opción ***Radio***. Las configuraciones que se deben realizar son las siguientes:

- **Configurar SSID**: es un valor de cadena que funciona como una palabra clave que permitirá la comunicación entre el puente y los demás dispositivos deseados (en este caso, el otro puente que se utilizará). En ambos se debe configurar la misma SSID, en caso contrario, los dispositivos no se comunicarán entre ellos.
- **Establecer la unidad Root**: establece cual es el puente que estará conectado al segmento principal de la red. El puente de Unisys – Fórum será el **Root,** al otro puente se le debe asignar *off* en esta opción.
- **Rates**: establece el valor de la tasa de datos a la que se le permitirá a la unidad enviar y recibir información (este valor se especifica en Mbps). La unidad también podrá recibir información que venga a tasas menores que la especificada.
- **Basic_rates:** se configura en el puente Root. Establece los diferentes valores de velocidad que todos los nodos de la conexión inalámbrica deben soportar o no se podrán asociar con el puente Root. El valor menor se emplea para transmitir broadcast y paquetes de control. Si se emplea el valor menor se asegura que los broadcasts y paquetes de control serán recibidos por todos los nodos, incluso los más distantes. El valor más alto determina la máxima velocidad a la que se puede transmitir un paquete de respuesta. Este parámetro no es tan crítico al momento de la configuración, pues el puente Root sólo estará enviando información a un solo nodo (en el Nuevo Centro)

- **Frequency:** establece la frecuencia que se empleará para realizar la transmisión de los datos. Si se asigna el valor manualmente, se debe asignar uno de acuerdo con las entidades reguladoras de la utilización del espectro radiofónico; si se deja la opción en ***auto***, la unidad verificará todas las frecuencias permitidas y elegirá una que no se encuentre en uso.
- **Distance:** se debe establecer la distancia que debe recorrer el radio enlace. Este parámetro sólo se debe configurar en el puente Root y permite establecer valores aproximados de delay en la transmisión.

Esta es la configuración básica del radio enlace. Para revisar los demás parámetros de configuración del radio enlace, se debe revisar el manual de usuario del puente (este manual se encuentra en el Anexo 5).

### 5.6.2 Configuración del Puerto Ethernet (Ethernet)

La configuración del puerto Ethernet se realiza seleccionando ***Configuration*** del menú principal y luego la opción ***Port***. La configuración que se realiza comprende lo siguiente:

- **Active:** establece el estado del puerto del puente. Si se deja en *on*, el puente sigue enviando y recibiendo información por el puerto. Si se establece en *off*, se le informa al software que deje de enviar paquetes al puerto y se deja de revisar la existencia de actividad de ethernet.
- **Size:** permite incrementar el tamaño máximo de los frames que se transmiten desde y hacia la estructura de ethernet. El valor típico se encuentra entre 1518 y 4096.

- **Port:** establece el puerto mediante el que el puente se conecta a la estructura de ethernet. Con la opción *Auto*, el puente busca automáticamente un cable en cualquiera de sus tres conectores.

**Asignación de Identificadores de Red (Ident)**

Esta opción permite la configuración de diferentes parámetros entre los que se pueden mencionar: dirección IP asignada al puente, dirección IP del router principal, dirección IP del Gateway, máscara, DNS, dominio, etc.

**Spanning Tree Protocol (STP)**

No se realiza una descripción de esta opción puesto que éste es un protocolo que se emplea si se están manejando múltiples puentes en un ambiente de LAN extendida.

**Configuración de Mobile IP (Mobile-IP)**

Este es un protocolo que permite moverse a través de diferenes sub-redes IP mientras que se mantiene la misma dirección IP.

# CAPÍTULO 6: ANÁLISIS Y RESULTADOS

## 6.1 Resultados del estudio de ancho de banda

A continuación se muestran las gráficas que se obtuvieron de la utilización del ancho de banda del enlace hacia CLU.

Las figuras 6.1 a 6.7 son gráficas obtenidas para los datos Inbound (entrantes), mientras que las gráficas 6.8 a 6.14 son las obtenidas para los datos Outbound (salientes).

La figura 6.2 muestra la utilización del enlace durante un periodo de una semana. Puede observarse que la utilización del enlace se da prácticamente a un 100 % (64K) durante las horas del día (a pesar de que el valor promedio es bajo, los picos de utilización son altos), mientras que durante las horas de la noche, el ancho de banda se mantiene en un valor aproximado de 40 K (este valor se mantiene así por la utilización del protocolo NETBIOS – IP, como se puede observar en la figura 6.5).

La figura 6.3 muestra un gráfico de la eficiencia de la red para los datos Inbound. Puede observarse que la eficiencia se mantiene casi en un valor de un 100%. Esto significa que para el tráfico de datos que se maneja en el enlace CLU, el ancho de banda es suficiente.

La figura 6.9 muestra la utilización del enlace durante un periodo de una semana para los datos Outbound. Puede observarse que la utilización del enlace se da prácticamente a un 100 % (64K) durante las horas del día (según los picos observados), mientras que durante las horas de la noche, el ancho de banda se mantiene en un valor aproximado de 5 K. Las siguientes figuras muestran las gráficas de utilización del ancho de banda para los protocolos más empleados durante la semana del muestreo.

La figura 6.10 muestra un gráfico de la eficiencia de la red para los datos Outbound. Puede observarse que la eficiencia se mantiene casi en un valor de un 100%. Esto significa que para el tráfico de datos que se maneja en el enlace CLU, el ancho de banda es suficiente.

Como ya se mencionó, la eficiencia de la red es muy alta, pero sólo para el transporte de datos. Si se desea realizar el transporte de voz por IP y aumentar el número de puntos de conexión, se debe ampliar este ancho de banda.

Si se realiza un enlace inalámbrico, el ancho de banda del enlace será de 11 Mbps (aproximadamente), por lo que el transporte de voz y de datos no tendrá ningún problema. Si se realiza el enlace por medio de una línea dedicada, se recomienda emplear un ancho de banda mínimo de 128 Kbps. Con este ancho de banda se asegura que se pueda realizar tanto un tráfico de datos como el tráfico de voz a través del enlace (línea dedicada).

Para ninguno de los dos casos (de enlace inalámbrico o de la utilización de una línea dedicada), se hace necesario la ampliación del ancho de banda de Unisys de Centroamérica (actualmente el ancho de banda es de 384 Kb). Esto es debido a que con los 384 Kb se realiza en enlace con Minnesotta; el ancho de banda con el que se realiza el enlace con la bodega (CLU) es completamente independiente del ancho de banda principal.

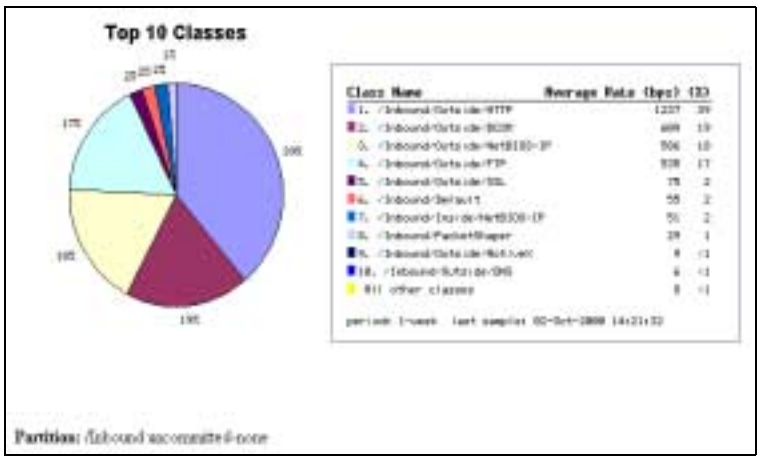



**Figura 6.1** Gráfico de los 10 protocolos más utilizados (para los datos Inbound) en el CLU durante un periodo de 1 semana (al 02 de Octubre de 2000)

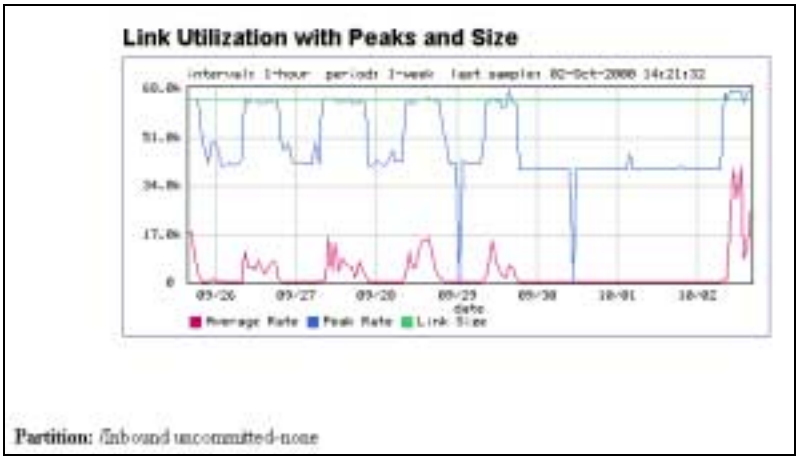



**Figura 6.2** Utilización del ancho de banda del enlace a CLU (datos Inbound) durante el periodo de una semana (al 02 deOctubre de 2000)

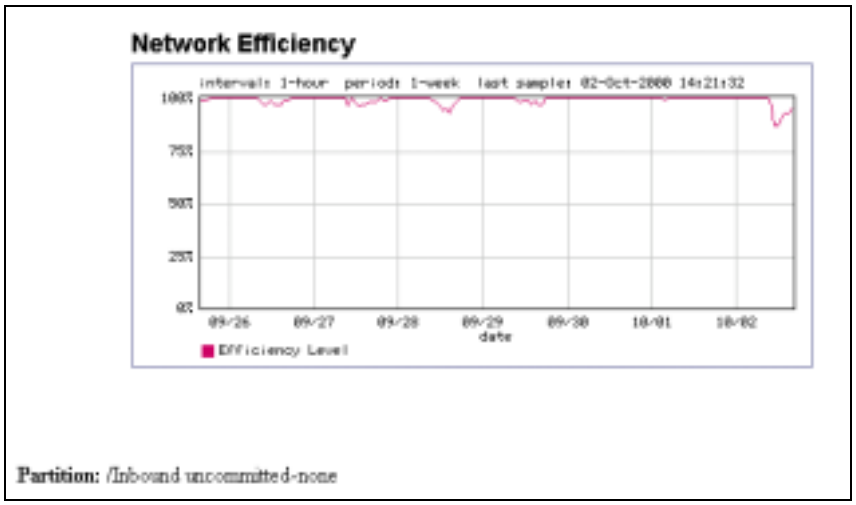



**Figura 6.3** Eficiencia de la red de CLU (datos Inbound) para un periodo de una semana (al 02 de Octubre de 2000)

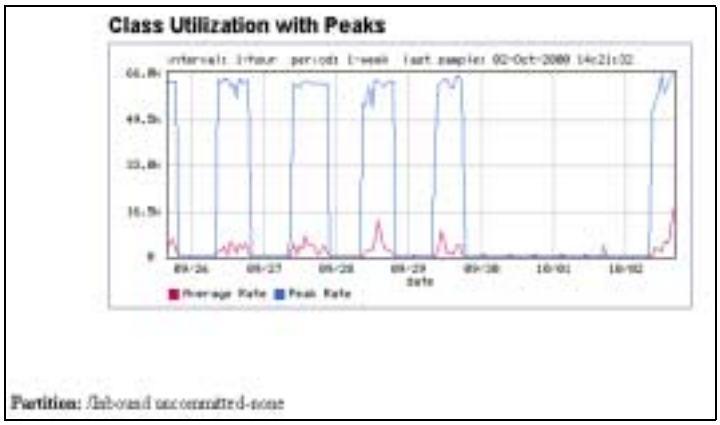



**Figura 6.4** Gráfico que muestra la utilización de ancho de banda del protocolo HTTP (Inbound) para el enlace CLU para un periodo de una semana (al 02 de Octubre de 2000)

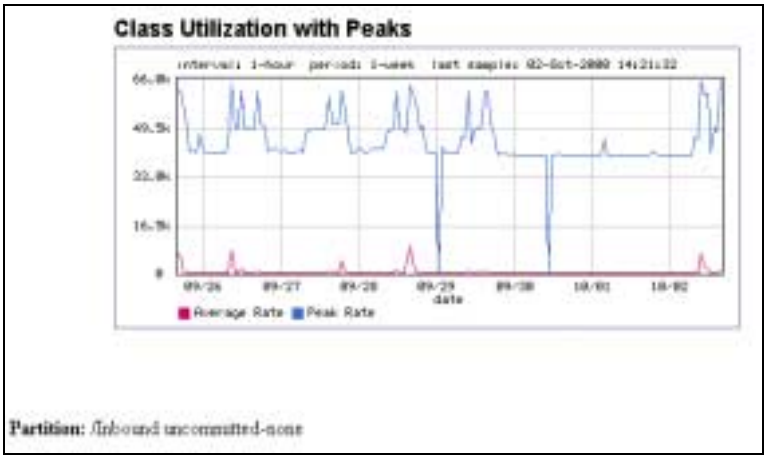



**Figura 6.5** Gráfico que muestra la utilización de ancho de banda del protocolo Netbios-IP (Inbound) para el enlace CLU para un periodo de una semana (al 02 de Octubre de 2000)

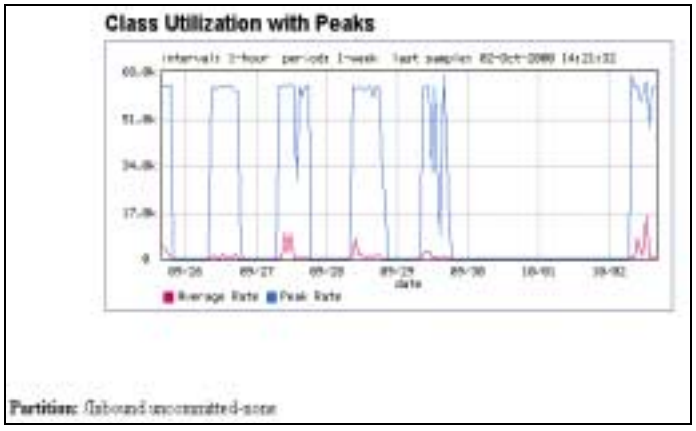



**Figura 6.6** Gráfico que muestra la utilización de ancho de banda del protocolo DCOM (Inbound) para el enlace CLU para un periodo de una semana (al 02 de Octubre de 2000)

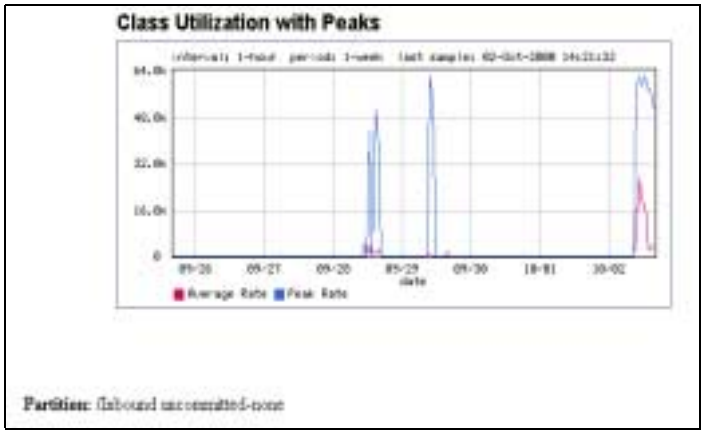



**Figura 6.7** Gráfico que muestra la utilización de ancho de banda del protocolo FTP (Inbound) para el enlace CLU para un periodo de una semana (al 02 de Octubre de 2000)

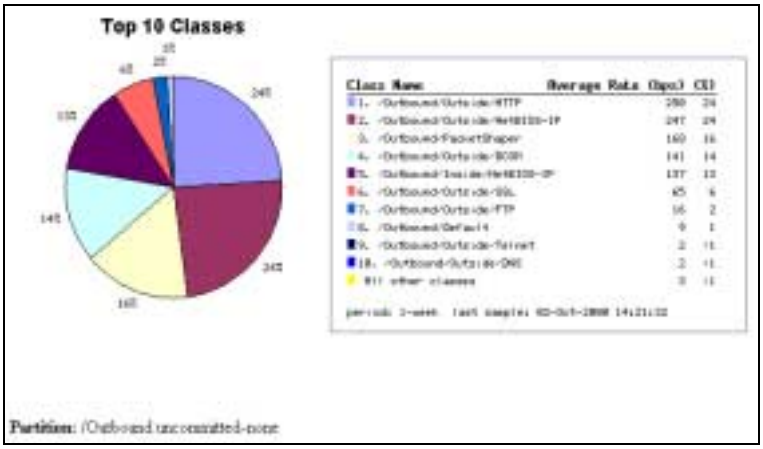



**Figura 6.8** Gráfico de los 10 protocolos más utilizados (para los datos Outbound) en el CLU durante un periodo de 1 semana (al 02 de Octubre de 2000)

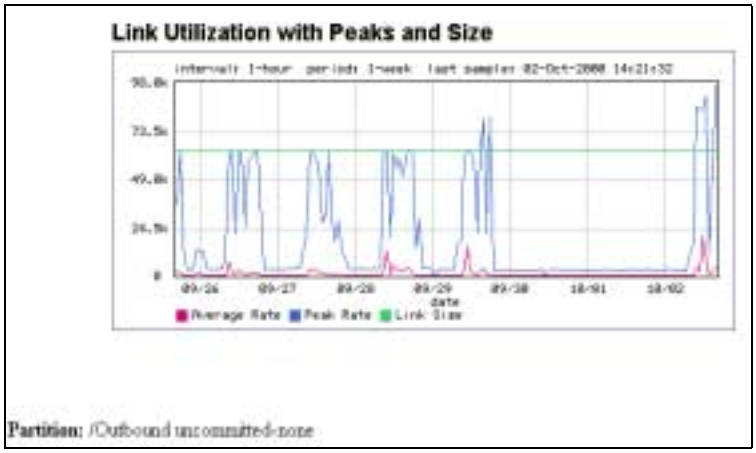



**Figura 6.9** Utilización del ancho de banda del enlace a CLU (datos Outbound) durante el periodo de una semana (al 02 deOctubre de 2000)

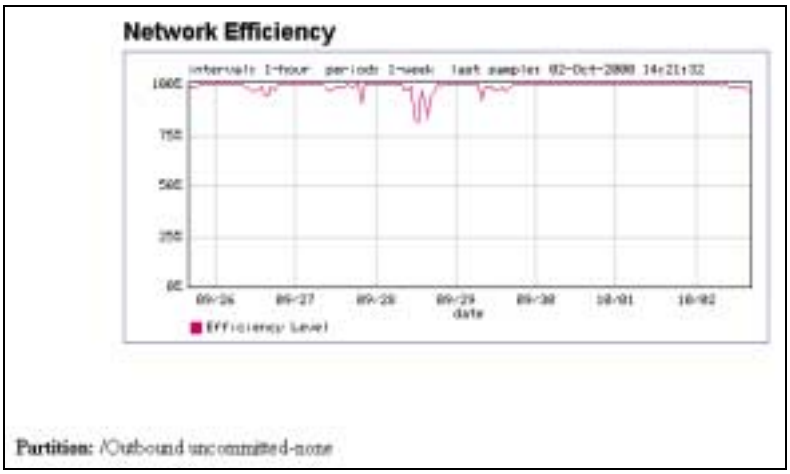



**Figura 6.10** Gráfico de la eficiencia de la red de CLU (para los datos Outbound) para un periodo de una semana (al 02 de Octubre de 2000)

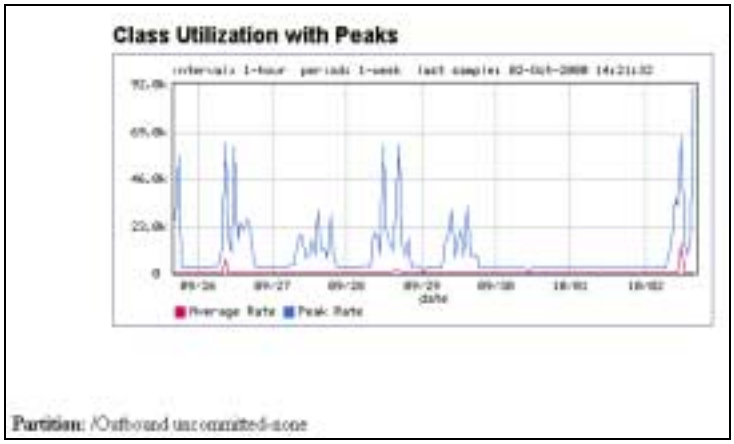



**Figura 6.11** Gráfico que muestra la utilización de ancho de banda del protocolo NETBIOS-IP/OUTSIDE (Outbound) para el enlace CLU para un periodo de una semana (al 02 de Octubre de 2000)

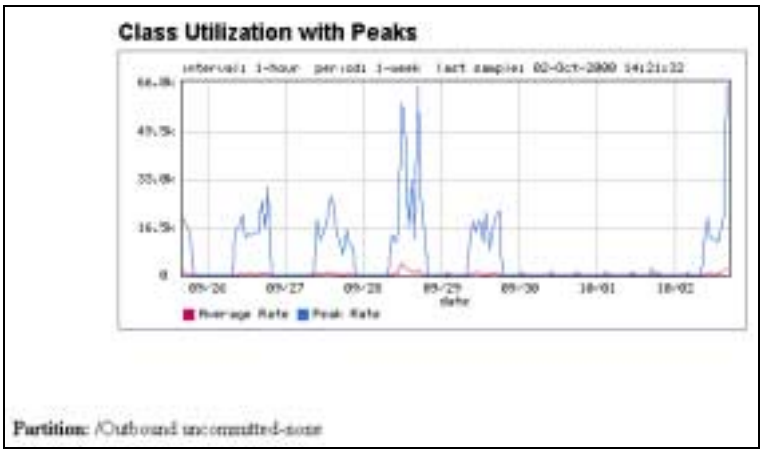



**Figura 6.12** Gráfico que muestra la utilización de ancho de banda del protocolo HTTP (Outbound) para el enlace CLU para un periodo de una semana (al 02 de Octubre de 2000)

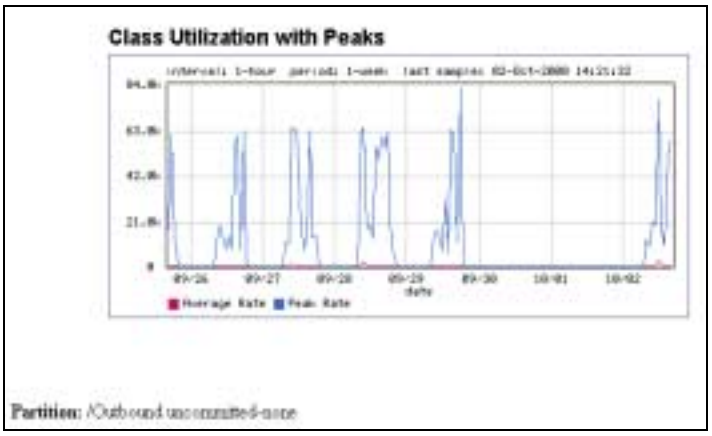



**Figura 6.13** Gráfico que muestra la utilización de ancho de banda del protocolo DCOM (Outbound) para el enlace CLU para un periodo de una semana (al 02 de Octubre de 2000)

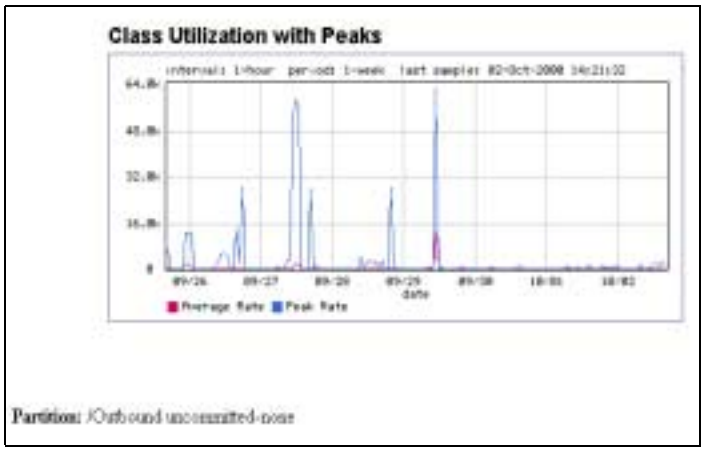



**Figura 6.14** Gráfico que muestra la utilización de ancho de banda del protocolo NETBIOS-IP/INSIDE (Outbound) para el enlace CLU para un periodo de una semana (al 02 de Octubre de 2000)

## 6.2 Requerimientos del Nuevo Centro Logístico

El local al que Unisys de Centroamérica desea trasladar el equipo que se encuentra en CLU son Las Bodegas de Matra, las cuales son administradas por Mabinsa (las bodegas se encuentran en Santa Ana).

Unisys de Centroamérica desea trasladar su equipo a la bodega No. 5. Dicha bodega cuenta con un área de 350 $m^2$ y 10 líneas telefónicas. El equipo que se desea trasladar es todo el que se encuentra en CLU: computadores personales, equipo en reparación, periféricos, material de bodega, etc.

Actualmente en CLU se emplean 11 computadores personales y existen un total de 16 conexiones (las conexiones extra corresponden a los dispositivos periféricos).

Los requerimientos básicos de la nueva bodega son los siguientes:

- Capacidad para albergar el equipo que se encuentra en existencia en CLU.
- Creación de un laboratorio de pruebas de equipo de redes.
- Aumento de los puntos de conexión.
- Capacidad de distinguir los diferentes segmentos de red.

### 6.2.1 Requerimientos Físicos

El nuevo local albergará el equipo en existencia en CLU. Este equipo comprende los dispositivos periféricos, las computadoras personales, equipo en reparación y equipo almacenado. Además de ésto, se desea aumentar la cantidad de puntos de conexión.

Como ya se mencionó, actualmente en CLU se emplean 16 puntos de conexión, 11 de los cuales corresponden a computadoras. Para el Nuevo Centro Logístico se desea doblar la cantidad de puntos de conexión (tentativamente a 22 puntos). Cada punto de conexión deberá contar con su conexión de cable de red UTP, su extensión telefónica (de una central telefónica) además de las conexiones eléctricas necesarias.

Para realizar el enlace inalámbrico se ha seleccionado puentes inalámbricos de la Serie Aironet. El modelo seleccionado corresponde al Aironet AIR – WGBR342R. Se seleccionó este tipo de puente pues es el que presenta la mayor potencia de salida (100 mW) y la mayor opción de seguridad (ID de 128 bits). En el desarrollo de software se realiza una mayor descripción de este puente.

### 6.2.2 Laboratorio de Pruebas

Se desea que el Nuevo Centro cuente con un laboratorio de equipo de comunicaciones. No se cuenta con una descripción específica acerca del equipo de comunicaciones que albergará este laboratorio, sin embargo, se pueden mencionar los siguientes: Routers, Hubs, Packeteers, equipo Nortel, equipo Cisco, etc.

Además del equipo de comunicaciones, el laboratorio deberá contar con, al menos, 10 puntos de conexión (conexión a la red, conexiones eléctricas y extensiones telefónicas) además de un mínimo de dos líneas telefónicas directas (para la realización de pruebas). Los 10 puntos de conexión del laboratorio no se deben contar entre los 22 puntos de conexión propios del Nuevo Centro.

### 6.2.3 Segmentos de Red

También se desea distinguir la localización física de las computadoras que conforman la red de Unisys de Centroamérica; en otras palabras, se desea poder diferenciar cuáles máquinas se encuentran en el Edificio de Fórum y cuales estarán en el Nuevo Centro.

Para realizar esto no se requiere de ningún tipo de herramienta. El edificio en Fórum cuenta con un Router propio, en CLU también se encuentra otro Router. Puesto que cada Router cuenta con su dirección IP, el Router de Fórum define el segmento de red de Fórum y el Router de CLU define al segmento de red de CLU. Puesto que este Router será el mismo que se instalará en el Nuevo Centro, el segmento de red del Nuevo Centro se podrá diferenciar del segmento de red de Unisys en Fórum.

**Cálculo de la altura de las torres para realizar el enlace deseado**

La figura 6.15 muestra un diagrama de la ubicación de los edificios para la realización del enlace inalámbrico.

El enlace se desea realizar entre el edificio de Unisys de Centroamérica y el Nuevo Centro. Como se puede observar, entre ambas edificaciones se encuentran los edificios de la compañía Procter & Gamble. El edificio de Unisys tiene una altura total de 15 metros, el edificio más alto de Procter & Gamble tiene una altura de 25 metros y la bodega No.5 presenta una altura de 9 metros.

El edificio de Unisys se encuentra en el terreno más elevado, los edificios de Procter & Gamble se encuentran en un desnivel aproximado de tres metros y el nuevo local de Unisys se encuentra en un desnivel mucho mayor (como se puede apreciar en el diagrama).

Con la ayuda de un GPS (Global Position System), se pudieron realizar las medidas de las distancias entre los edificios, las alturas relativas de los terrenos en los que están ubicados y la latitud y longitud en la que están ubicados. La distancia existente entre el edificio de Unisys (9º 56' 37''$^{N}$, 84º 11' 38''$^{W}$) y los edificios de Procter & Gamble ($\mathbf{d_1}$) medida fue de aproximadamente 225 metros; la distancia entre los Edificios de Procter & Gamble (9º 56' 44''$^{N}$, 84º 11' 39''$^{W}$) y la bodega ($\mathbf{d_2}$) es de aproximadamente 800 metros; la distancia entre el edificio de Unisys y la bodega ($\mathbf{d_T}$) es de aproximadamente 980 metros. Sumando las distancias parciales se observa que el valor obtenido es muy similar al valor de la distancia total, lo cual demuestra que los tres edificios se encuentran prácticamente en la misma línea de visión (existe una diferencia en la distancia sumada y la real de 45 metros, pero se puede aproximar a una línea recta). Esto significa que para enlazar el edificio de Unisys con la bodega (9º 57' 06''$^{N}$, 84º 11' 25''$^{W}$), se debe pasar por encima del los edificios de Procter & Gamble.

Esto se puede realizar colocando las antenas, de los puentes inalámbricos, a las alturas necesarias para pasar por encima de los edificios de P&G.

En el diagrama se observan una serie de abreviaturas que señalan diferentes alturas. Esta es la descripción de dichas alturas (todas las alturas y distancias se miden en metros):

***$h_{UNISYS}$ :*** es la altura completa del edificio principal de Unisys incluida la torre que se deberá construir.

***$h_{EXTRA}$*** : altura con la que se desea que la señal de RF pase por encima del edificio de P&G. Este valor se debe asignar libremente.

***$h_{TP\&G}$*** : altura del edificio de P&G más la $h_{EXTRA.}$

***$h_{D1}$*** : desnivel entre el terreno del edificio de Unisys y el terreno de los edificios de P&G. Con el GPS se obtuvo un valor aproximado de 3 metros.

***$h_{TUNISYS}$*** : altura necesaria de la torre que se deberá instalar en el edificio de Unisys.

***$h_{D2}$***: desnivel entre el terreno de los edificios de P&G y el terreno en el que se encuentra la bodega. Con el GPS se midió un valor aproximado de 10 metros.

***$h_{TOTBOD}$ :*** es la altura completa del edificio de la bodega incluida la torre que se deberá construir.

***$h_{TBOD}$*** : es la altura de la torre que se deberá construir en la bodega. Este valor se asigna según consideraciones de diseño.

***$\Delta h$*** : es un valor de altura relativa entre ***$h_{TOTBOD}$*** y ***$h_{D2}$***. Este valor podrá ser positivo o negativo.

***Ángulo $\theta$***: en un ángulo que se emplea con fines matemáticos y permite calcular alturas necesarias en el diseño.

Para poder calcular la altura de las torres que se deben construir, el primer paso consiste en asignar un valor deseado para $h_{EXTRA}$. Una vez realizado esto, se debe asignar un valor para la altura de la torre de la bodega ($h_{TBOD}$).
Con estos valores, se puede calcular $h_{TOTBOD}$:

$$\boldsymbol{h_{TOTBOD} = 9m + h_{TBOD}}$$

Trazando una línea recta imaginaria (a cero grados de inclinación) entre la punta de la torre hasta que corte al edificio de P&G o al terreno del mismo, se puede calcular el valor de Δh:

$$\boldsymbol{\Delta h = h_{D2} - h_{TOTBOD}}$$

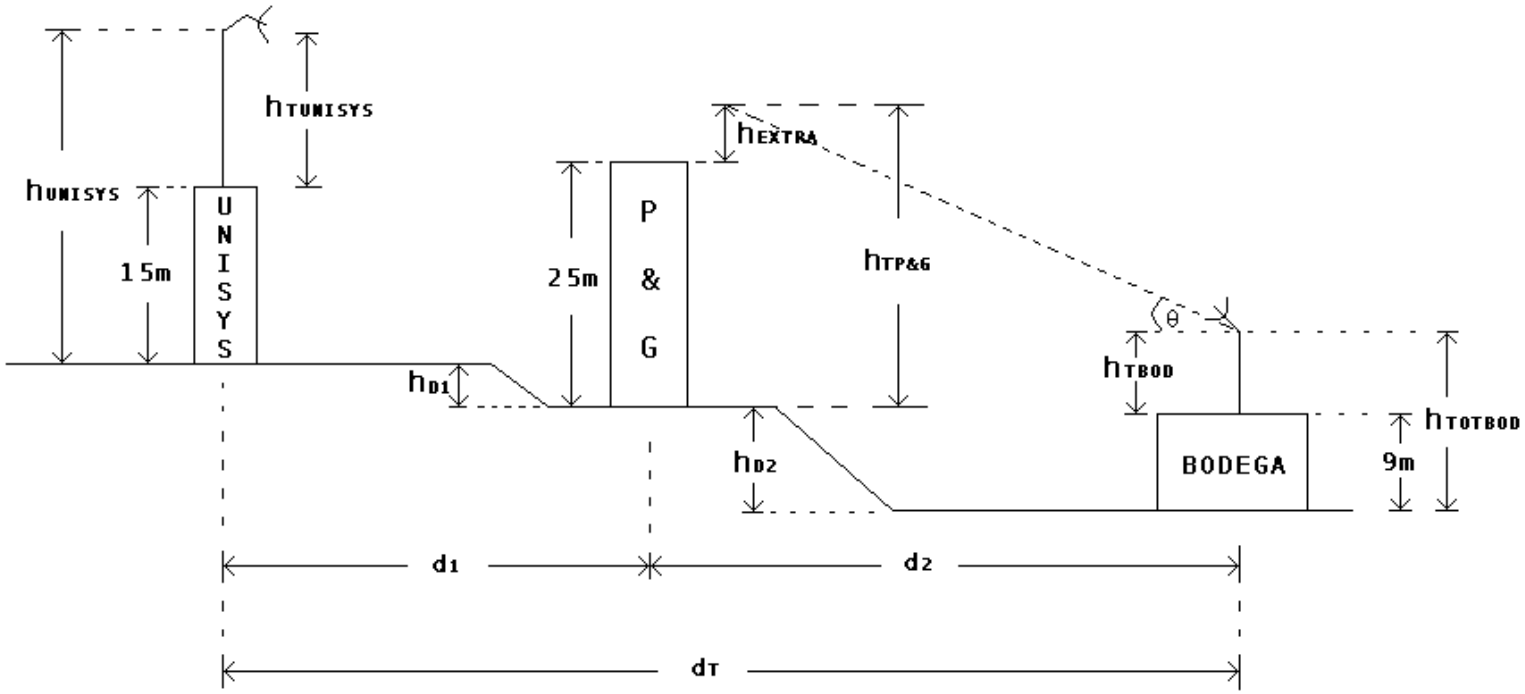



**Figura 6.15** Diagrama de la ubicación de los edificios para realizar el enlace inalámbrico (diagrama realizado en Paint)

Con el valor de deltah, los 25 metros de alto del edificio de P&G y $h_{EXTRA}$ se puede calcular el valor equivalente de $\boldsymbol{h_{TP\&G}}$:

$$\boldsymbol{h_{TP\&G} = \Delta h + 25m + h_{EXTRA}}$$

De este modo, se forma un triángulo rectángulo imaginario que está formado por la distancia $\boldsymbol{d_2}$ (eje X), la altura $h_{TP\&G}$ y el segmento que une a $h_{TP\&G}$ con $h_{TBOD}$. Con estos valores, se puede calcular el valor del *ángulo θ*:

$$\boldsymbol{\theta = TAN^{-1} (h_{TP\&G} / d_2)}$$

Con el valor $d_T$ y con el valor de θ, se puede calcular la altura equivalente $h_{UNISYS}$:

$$\boldsymbol{h_{UNISYS} = d_T (TAN\theta)}$$

Una vez obtenido este valor, se puede calcular el valor de la torre que se debe construir en Fórum para colocar la antena del puente inalámbrico (y con la que se logrará tener línea vista entre ambas antenas):

$$\boldsymbol{h_{TUNISYS} = h_{UNISYS} - 15m - \Delta h - h_{D1}}$$

Este valor de altura calculado es medido desde el suelo del edificio de Unisys.

Para tener una idea de los cálculos, suponiendo una $h_{EXTRA}$ de 2m y una altura de $h_{TBOD}$ de 10 metros (32.8 pie), se obtiene una altura de $h_{TUNISYS}$ de 13.05 metros (42.81 pie)[1].

[1] Las alturas se detallan en pie pues Cisco Systems especifica la longitud de los cables de bajas pérdidas en pie.

La altura de 13.05 metros de la torre de Unisys convertida en pie, representa 42.81 pie (por lo que se haría necesario emplear un mínimo de 100 pie de cable de bajas pérdidas para la conexión de la antena al puente).

Se aconseja utilizar la antena **AIR-ANT1949;** esta antena presenta las siguientes características:

- Ganancia de 13.5 dBi
- Alcance aproximado de 10.5 Km a 2 Mbps y de 3.3 Km a 11 Mbps
- Lóbulo de radiación ubicado a 30º H y 25º V
- Se utiliza en aplicaciones de alcance medio y direccionales.

Para el cable que unirá a la antena con el puente, se aconseja emplear un cable de bajas pérdidas del tipo **AIR-420-003346-100** el que presenta una atenuación de 6.7 dBi por cada 100 pie de cable; las pérdidas totales se calculan a partir de:

***Pérdidas = 100 pie * ( 6.7 dBi / 100 pie) = 6.7 dBi.***

Esto significa que para cuando la señal generada por el puente llegue a la antena, habrá perdido 6.7 dBi de potencia.



Para el lado de la bodega, se debe emplear una antena del mismo tipo que la que se aconseja utilizar en Unisys de Centroamérica. Se recomienda emplear es mismo tipo de antena debido a que en radioenlaces se aconseja emplear antenas direccionales; la antena direccional evita que se desperdicie potencia irradiando en direcciones en las que no existe un receptor. Asumiendo que se empleará una torre de 10 metros (32.8 pie), se aconseja emplear 75 pie de cable de bajas pérdidas del tipo **AIR-420-003346-075** (el que presenta pérdidas de 5.0d dBi por cada 75 pie de cable). Las pérdidas producidas en este caso son menores:

***Pérdidas = 75 pie * (5.0 dBi / 75pie) = 5.0 dBi.***

Esto significa que para cuando la señal generada por el puente llegue a la antena, habrá perdido 5.0 dBi de potencia.

Con estos valores de alturas, se puede lograr el enlace deseado. Se pueden realizar otros cálculos basándose en las ecuaciones planteadas.

Los datos de distancias que se mencionan en este informe se obtuvieron con un GPS. Para poder realizar cálculos más exactos, se recomienda realizar un trabajo topográfico para calcular las distancias y las alturas de los terrenos con mayor precisión.



# CAPÍTULO 7: CONCLUSIONES Y RECOMENDACIONES

## Conclusiones

La etapa de capacitación permitió adquirir conocimientos básicos sobre el ambiente de redes, técnicas de transmisión de datos y equipo activo de comunicaciones.

El Nuevo Centro Logístico albergará al mismo equipo en existencia en el Centro Logístico de la Uruca (computadores personales, equipo de taller, equipo de bodega, etc). El Nuevo Centro Logístico requerirá de la creación de un laboratorio de pruebas (para equipo de comunicaciones). El Nuevo Centro además requerirá de la duplicación de los puntos de conexión, en comparación con la cantidad que existen en la Uruca).

La utilización del Packetshaper, al conectarlo al enlace Unisys Fórum – CLU, permitió determinar la utilización de los 64 Kbits de este enlace. Una vez determinada la utilización de dicho enlace, se pudo calcular los anchos el ancho de banda necesario para la transmisión de voz y de datos en el nuevo enlace que se desea realizar.

Si se realiza un enlace inalámbrico para la transmisión de voz y de datos, el ancho de banda empleado estará entre los 2 Mbps y los 11 Mbps, por lo que no se requiere de ninguna herramienta Qos (Calidad de servicio). Si se realiza el enlace a través de una línea dedicada, el ancho de banda mínimo aconsejado es de 128 Kbits; además se sugiere la utilización de las herramientas de QoS. Este ancho de banda es el total para el enlace, tanto como para voz como para datos. Las herramientas de QoS administran dinámicamente este ancho de banda en el caso de la transmisión de voz.



No se requiere que en Unisys de Centroamérica se amplíe el ancho de banda (actualmente el mismo es de 384 Kbps). El nuevo enlace es independiente del ancho de banda principal de Unisys.

Al Router Cisco de la serie 3660 que se emplea en Unisys de Centroamérica se le deben programar herramientas QoS para que soporte el tráfico de voz. Ocurre lo mismo con el Router Cisco de la serie 3640 en existencia en CLU.

Para realizar el enlace inalámbrico entre Unisys de Centroamérica y el Nuevo Centro Logístico se recomienda emplear equipo inalámbrico de la serie Aironet 340, además de las antenas y cable de bajas pérdidas.

**Recomendaciones**

Se recomienda la realización de un estudio topográfico para determinar las distancias exactas entre los edificios de Unisys, Procter & Gamble y el Nuevo Centro, a partir de ese estudio se pueden determinar las alturas exactas de los terrenos y la dirección de cada edificio. Con estos datos más precisos y con las ecuaciones planteadas se pueden calcular las alturas de las torres para la colocación de las antenas.



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# ANEXOS

## Anexo 1: Comandos de configuración de los puertos de Voz.

El siguiente archivo presenta una descripción más detallada acerca de los comandos necesarios para la configuración de los puertos de voz de los routers Cisco.



# ANEXOS

## Anexo 1: Comandos de configuración de los puertos de Voz.

El siguiente archivo presenta una descripción más detallada acerca de los comandos necesarios para la configuración de los puertos de voz de los routers Cisco.



# Configuring Voice Ports

This chapter describes how to configure voice ports for both the Cisco 3600 series (for Voice over IP) and for the Cisco MC3810 (for Voice over Frame Relay, Voice over ATM, and Voice over HDLC). For a description of the voice port commands, refer to the "Voice Port Commands" chapter in the *Voice, Video, and Home Applications Command Reference*.

## Voice Ports on the Cisco 3600 Series

The Cisco 3600 currently provides only analog voice ports for its implementation of Voice over IP. The type of signaling associated with these analog voice ports depend on the interface module installed into the device. The Cisco 3600 series router supports either a two-port or four-port voice network module (VNM); VNMs can hold either two or four voice interface cards (VICs).

Each VIC is specific to a particular signaling type; therefore, VICs determine the type of signaling for the voice ports on that particular VNM. This means that even though VNMs can hold multiple VICs, each VIC on a VNM must conform to the same signaling type. For more information about the physical characteristics of VNMs and VICs or how to install them, refer to the installation document, *Voice Network Module and Voice Interface Card Configuration Note*, that came with your VNM.

Voice ports on the Cisco 3600 series support three basic voice signaling types:

- FXO—Foreign Exchange Office interface. The FXO interface is an RJ-11 connector that allows a connection to be directed at the PSTN's central office (or to a standard PBX interface, if the local telecommunications authority permits). This interface is of value for off-premise extension applications.
- FXS—The Foreign Exchange Station interface. This interface is an RJ-11 connector that allows connection for basic telephone equipment, keysets, PBXs, and supplies ring, voltage, and dial tone.
- E&M—The "Ear and Mouth" interface (or "RecEive and TransMit") interface. This interface is an RJ-48 connector that allows connection for PBX trunk lines (tie lines). It is a signaling technique for two-wire and four-wire telephone and trunk interfaces.

## Configuring Voice Ports on the Cisco 3600 Series

In general, voice port commands define the characteristics associated with a particular voice port signaling type. Under most circumstances, the default voice port command values are adequate to configure FXO and FXS ports to transport voice data over your existing IP network. Because of the inherent complexities involved with PBX networks, E&M ports might need specific voice port values configured, depending on the specifications of the devices in your telephony network.



# Voice Ports for the Cisco 3600 Series Configuration Task List

Perform the following tasks to configure the voice ports on the Cisco 3600 series:

- Configuring FXO or FXS Voice Ports
- Fine-Tuning FXO and FXS Voice Ports
- Configuring E&M Voice Ports
- Fine-Tuning E&M Voice Ports
- Activating the Voice Port

## Configuring FXO or FXS Voice Ports

Under most circumstances the default voice port values are adequate for both FXO and FXS voice ports. If you need to change the default configuration for these voice ports, perform the following tasks. The first two tasks are required; the third task is optional.

1. Identify the voice port and enter the voice-port configuration mode.
2. Configure the following mandatory voice-port parameters:
   (a) Dial type (FXO only)
   (b) Signal type
   (c) Call progress tone
   (d) Ring frequency (FXS only)
   (e) Ring number (FXO only)
3. Configure one or more of the following optional voice-port parameters:
   (a) PLAR connection mode
   (b) Music-threshold
   (c) Description
   (d) Comfort noise (if VAD is activated—VAD is a dial peer command)

To configure FXO and FXS voice ports, use the following commands beginning in privileged EXEC mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **configure terminal** | Enter global configuration mode. |
| 2 | **voice-port** *slot-number/subunit-number/port* | Identify the voice port you want to configure and enter voice-port configuration mode. |
| 3 | **dial-type** {**dtmf** \| **pulse**} | (For FXO ports only) Select the appropriate dial type for out-dialing. |
| 4 | **signal** {**loop-start** \| **ground-start**} | Select the appropriate signal type for this interface. |
| 5 | **cptone** *country* | Select the appropriate voice call progress tone for this interface.<br>The default for this command is **us**. For a list of supported countries, refer to the *Voice, Video, and Home Applications Command Reference*. |

| Step | Command | Purpose |
|---|---|---|
| 6 | **ring frequency** {**25** \| **50**} | (For FXS ports only) Select the appropriate ring frequency (in Hertz) specific to the equipment attached to this voice port. |
| 7 | **ring number** *number* | (For FXO ports only) Specify the maximum number of rings to be detected before answering a call. |
| 8 | **connection plar** *string* | (Optional) Specify the private line auto ringdown (PLAR) connection, if this voice port is used for a PLAR connection. The *string* value specifies the destination telephone number. |
| 9 | **music-threshold** *number* | (Optional) Specify the threshold (in decibels) for on-hold music. Valid entries are from –70 to –30. |
| 10 | **description** *string* | (Optional) Attach descriptive text about this voice port connection. |
| 11 | **comfort-noise** | (Optional) Specify that background noise will be generated. |

### Validation Tips

You can check the validity of your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for a dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.
- Use the **show voice-port** command to verify that the data configured is correct.

### Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Network Protocols Configuration Guide, Part 1*.
- Use the show voice-port command to make sure that the port is enabled. If the port is offline, use the **no shutdown** command.
- If you have configured E&M interfaces, make sure that the values pertaining to your specific PBX setup, such as timing and/or type, are correct.
- Check to see if the voice network module has been correctly installed. For more information, refer to the installation document, *Voice Network Module and Voice Interface Card Configuration Note*, that came with your voice network module.

## Fine-Tuning FXO and FXS Voice Ports

Depending on the specifics of your particular network, you may need to adjust voice parameters involving timing, input gain, and output attenuation for FXO or FXS voice ports. Collectively, these commands are referred to as voice-port tuning commands.

**Note** In most cases, the default values for voice-port tuning commands will be sufficient.

To configure voice-port tuning for FXO and FXS voice ports, perform the following tasks:

1. Identify the voice port and enter the voice-port configuration mode.
2. For each of the following parameters, select the appropriate value:
   (a) Input gain
   (b) Output attenuation
   (c) Echo cancel coverage
   (d) Non-linear processing
   (e) Initial digit timeouts
   (f) Interdigit timeouts
   (g) Timing other than timeouts

To fine-tune FXO or FXS voice ports, use the following commands beginning in privileged EXEC mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **configure terminal** | Enter global configuration mode. |
| 2 | **voice-port** *slot-number/subunit-number/port* | Identify the voice-port you want to configure and enter voice-port configuration mode. |
| 3 | **input gain** *value* | Specify (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from –6 to 14. |
| 4 | **output attenuation** *value* | Specify (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14. |
| 5 | **echo-cancel enable** | Enable echo-cancellation of voice that is sent out the interface and received back on the same interface. |
| 6 | **echo-cancel coverage** *value* | Adjust the size (in milliseconds) of the echo-cancel. Acceptable values are 16, 24, and 32. |
| 7 | **non-linear** | Enable non-linear processing, which shuts off any signal if no near-end speech is detected. (Non-linear processing is used with echo-cancellation.) |
| 8 | **timeouts initial** *seconds* | Specify the number of seconds the system will wait for the caller to input the first digit of the dialed digits. Valid entries for this command are from 0 to 120. |
| 9 | **timeouts interdigit** *seconds* | Specify the number of seconds the system will wait (after the caller has input the initial digit) for the caller to input a subsequent digit. Valid entries for this command are from 0 to 120. |

| Step | Command | Purpose |
|---|---|---|
| 10 | **timing digit** *milliseconds* | If the voice-port dial type is DTMF, configure the DTMF digit signal duration. The range of the DTMF digit signal duration is from 50 to 100. The default is 100. |
| 11 | **timing inter-digit** *milliseconds* | If the voice-port dial type is DTMF, configure the DTMF inter-digit signal duration. The range of the DTMF inter-digit signal duration is from 50 to 500. The default is 100. |
| 12 | **timing pulse-digit** *milliseconds* | (FXO ports only) If the voice-port dial type is pulse, configure the pulse digit signal duration. The range of the pulse digit signal duration is from10 to 20. The default is 20. |
| 13 | **timing pulse-inter-digit** *milliseconds* | (FXO ports only) If the voice-port dial type is pulse, configure the pulse inter-digit signal duration. The range of the pulse inter-digit signal duration is from 100 to 1000. The default is 500. |

**Note** After you change any voice-port command, it is a good idea to cycle the port by using the **shutdown** and **no shutdown** commands.

# Configuring E&M Voice Ports

Unlike FXO and FXS voice ports, the default E&M voice-port parameters most likely will not be sufficient to enable voice data transmission over your IP network. E&M voice-port values must match those specified by the particular PBX device to which it is connected.

**Note** E&M voice-port values must match those of the PBX to which it is connected. Refer to the documentation that came with your specific PBX for the appropriate E&M voice-port configuration command values.

To configure an E&M voice port, perform the following tasks. The first two tasks are required; the third task is optional.

1. Identify the voice port and enter the voice-port configuration mode.
2. For each of the following required parameters, select the appropriate parameter value:
   (a) Dial type
   (b) Signal type
   (c) Call progress tone
   (d) Operation
   (e) Type
   (f) Impedance
3. Select one or more of the following optional parameters:
   (a) Connection mode
   (b) Music-threshold

(c) Description

(d) Comfort tone (if VAD is activated)

To configure E&M voice ports, use the following commands beginning in privileged EXEC mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **configure terminal** | Enter global configuration mode. |
| 2 | **voice-port** *slot-number/subunit-number/port* | Identify the voice port you want to configure and enter voice-port configuration mode. |
| 3 | **dial-type** {**dtmf** \| **pulse**} | Select the appropriate dial type for out-dialing. |
| 4 | **signal** {**wink-start** \| **immediate** \| **delay-dial**} | Select the appropriate signal type for this interface. |
| 5 | **cptone** {**australia** \| **brazil** \| **china** \| **finland** \| **france** \| **germany** \| **japan** \| **northamerica** \| **unitedkingdom**} | Select the appropriate voice call progress tone for this interface. |
| 6 | **operation** {**2-wire** \| **4-wire**} | Select the appropriate cabling scheme for this voice port. |
| 7 | **type** {**1** \| **2** \| **3** \| **5**} | Select the appropriate E&M interface type.<br>Type 1 indicates the following lead configuration:<br>E—output, relay to ground<br>M—input, referenced to ground<br>Type 2 indicates the following lead configuration:<br>E—output, relay to SG<br>M—input, referenced to ground<br>SB—feed for M, connected to –48V<br>SG—return for E, galvanically isolated from ground<br>Type 3 indicates the following lead configuration:<br>E—output, relay to ground<br>M—input, referenced to ground<br>SB—connected to –48V<br>SG—connected to ground<br>Type 5 indicates the following lead configuration:<br>E—output, relay to ground<br>M—input, referenced to –48V |
| 8 | **impedance** {**600c** \| **600r** \| **900c** \| **complex1** \| **complex2**} | Specify a terminating impedance. This value must match the specifications from the telephony system to which this voice port is connected. |
| 9 | **connection plar** *string* | (Optional) Specify the private line auto ringdown (PLAR) connection, if this voice port is used for a PLAR connection. The *string* value specifies the destination telephone number. |
| 10 | **music-threshold** *number* | (Optional) Specify the threshold (in decibels) for on-hold music. Valid entries are from –70 to –30. |
| 11 | **description** *string* | (Optional) Attach descriptive text about this voice port connection. |
| 12 | **comfort-noise** | (Optional) Specify that background noise will be generated. |

### Validation Tips

You can check the validity of your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for a dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.
- Use the **show voice-port** command to verify that the data configured is correct.

### Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Network Protocols Configuration Guide, Part 1*.
- Use the **show voice-port command** to make sure that the port is enabled. If the port is offline, use the **no shutdown** command.
- If you have configured E&M interfaces, make sure that the values pertaining to your specific PBX setup, such as timing and/or type, are correct.
- Check to see if the voice network module has been correctly installed. For more information, refer to the installation document that came with your voice network module.

## Fine-Tuning E&M Voice Ports

Depending on the specifics of your particular network, you may need to adjust voice parameters involving timing, input gain, and output attenuation for E&M voice ports. Collectively, these commands are referred to as voice-port tuning commands.

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**Note** In most cases, the default values for voice-port tuning commands will be sufficient.

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To configure voice-port tuning for E&M voice ports, perform the following tasks:

1. Identify the voice port and enter the voice-port configuration mode.
2. For each of the following parameters, select the appropriate value:
   (a) Input gain
   (b) Output attenuation
   (c) Echo cancel coverage
   (d) Non-linear processing
   (e) Initial digit timeouts
   (f) Interdigit timeouts
   (g) Timing other than timeouts

To fine-tune E&M voice ports, use the following commands beginning in privileged EXEC mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **configure terminal** | Enter global configuration mode. |
| 2 | **voice-port** *slot-number/subunit-number/port* | Identify the voice port you want to configure and enter voice-port configuration mode. |
| 3 | **input gain** *value* | Specify (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from –6 to 14. |
| 4 | **output attenuation** *value* | Specify (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14. |
| 5 | **echo-cancel enable** | Enable echo-cancellation of voice that is sent out the interface and received back on the same interface. |
| 6 | **echo-cancel coverage** *value* | Adjust the size (in milliseconds) of the echo-cancel. Acceptable values are 16, 24, and 32. |
| 7 | **non-linear** | Enable non-linear processing, which shuts off any signal if no near-end speech is detected. (Non-linear processing is used with echo-cancellation.) |
| 8 | **timeouts initial** *seconds* | Specify the number of seconds the system will wait for the caller to input the first digit of the dialed digits. Valid entries for this command are from 0 to 120. |
| 9 | **timeouts interdigit** *seconds* | Specify the number of seconds the system will wait (after the caller has input the initial digit) for the caller to input a subsequent digit. Valid entries for this command are from 0 to 120. |
| 10 | **timing clear-wait** *milliseconds* | Specify the minimum amount of time between the inactive seizure signal and the call being cleared. Valid entries for **clear-wait** are from 200 to 2000 milliseconds. |
| 11 | **timing delay-duration** *milliseconds* | Specify the delay signal duration for delay dial signaling. Valid entries for **delay-duration** are from 100 to 5000 milliseconds. |
| 12 | **timing delay-start** *milliseconds* | Specify the minimum delay time from outgoing seizure to outdial address. Valid entries for **delay-start** are from 20 to 2000 milliseconds. |
| 13 | **timing dial-pulse min-delay** *milliseconds* | Specify the time between generation of wink-like pulses. Valid entries for **dial-pulse min-delay** are from 0 to 5000 milliseconds. |
| 14 | **timing digit** *milliseconds* | If the voice-port dial type is DTMF, configure the DTMF digit signal duration.Valid entries for **digit** are from 50 to 100 milliseconds. |
| 15 | **timing inter-digit** *milliseconds* | If the voice-port dial type is DTMF, specify the DTMF inter-digit duration. Valid entries for **inter-digit** are from 50 to 500 milliseconds. |
| 16 | **timing pulse** *pulse-per-second* | If the voice-port dial type is pulse, specify the pulse dialing rate. Valid entries for **pulse** are from 10 to 20 pulses per second. |

| Step | Command | Purpose |
|---|---|---|
| 17 | **timing pulse-inter-digit** *milliseconds* | If the voice-port dial type is pulse, specify the pulse dialing inter-digit timing. Valid entries for **pulse-inter-digit** are 100 to 1000 milliseconds. |
| 18 | **timing wink-duration** *milliseconds* | Specify the maximum wink signal duration. Valid entries for **wink-duration** are from 100 to 400 milliseconds. |
| 19 | **timing wink-wait** *milliseconds* | Specify the maximum wink-wait duration for a wink start signal. Valid entries for **wink-wait** are from 100 to 5000 milliseconds. |

**Note** After you change any voice-port command, it is a good idea to cycle the port by using the **shutdown** and **no shutdown** commands.

## Activating the Voice Port

After you have configured the voice port, you need to activate the voice port to bring it online. In fact it is a good idea to cycle the port—meaning to shut the port down and then bring it online again.

To activate a voice port, use the following commands in voice-port configuration mode:

| Command | Purpose |
|---|---|
| **no shutdown** | Activate the voice port. |

To cycle a voice port, use the following command in voice-port configuration mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **shutdown** | Deactivate the voice port. |
| 2 | **voice-port** *slot-number/subunit-number/port* | Identify the voice port you want to activate and enter the voice-port configuration mode. |
| 3 | **no shutdown** | Activate the voice port. |

**Note** If you are not going to use a voice port, shut it down.

# Voice Ports on the Cisco MC3810

The Cisco MC3810 hardware features two models, each providing different configuration options for voice ports:

- Six analog voice interfaces— The Cisco MC3810 version with the analog voice module (AVM) supports up to six analog voice personality modules (APMs) with each voice module supporting a single signaling type (see below). Each voice personality module maps to a single analog voice port. The FXO, FXS, and E&M voice modules can be installed in any combination.
- One digital voice module (DVM)— The Cisco MC3810 version with the DVM provides support for up to 24 voice channels, one for each voice port. Depending on whether the controller is T1 or E1, different DS0 voice channels are used. The DVM supports Channel Associated Signaling

(CAS) for the following types: FXO, FXS, and E&M. For E&M signaling, the DVM also supports E1 Mercury Exchange Limited Channel Associated Signaling (MELCAS), a standard used primarily in the United Kingdom.

The Cisco MC3810 voice ports provide support for three basic voice signaling formats:

- FXO—The Foreign Exchange Office interface. This interface allows a connection to be directed to the PSTN's central office. The FXO interface also allows a connection to be directed to a standard PBX interface if the local telecommunications authority permits. This interface is of value for off-premise extension applications.
- FXS—The Foreign Exchange Station interface. This interface allows connection for basic telephone equipment and keysets, and supplies ring, voltage, and dial tone.
- E&M—The "Ear and Mouth" (or "RecEive and TransMit") interface allows connection for PBX trunk lines (tie lines). E&M is a signaling technique for two-wire and four-wire telephone and trunk interfaces.

# Configuring Voice Ports on the Cisco MC3810

In general, voice-port commands define the characteristics associated with a particular voice-port signaling type. Under most circumstances, the default voice-port command values are adequate to configure FXO and FXS ports to transport voice data using the Cisco MC3810. Because of the inherent complexities involved with PBX networks, E&M ports might need specific voice-port values configured, depending on the specifications of the devices in your telephony network.

Table 10 lists the valid slot and port numbers for the different voice interfaces.

**Table 10 Voice Interface Slot and Port Number**

| Interface Type | Slot | Valid Port Numbers |
|---|---|---|
| Analog voice module (AVM) | 1 | 1–6 |
| Digital voice module (DVM) | 1 | Digital T1: 1–24<br>Digital E1: 1–15 and 17–31 |
| Multiflex Trunk (MFT) | 0 | Digital T1: 1–24<br>Digital E1: 1–15 and 17–31 |

**Note** The voice-port number designations start with 1. Unlike serial port interfaces and interfaces on other Cisco products, there is no *port* 0 for voice ports.

# Voice Ports for the Cisco MC3810 Configuration Task List

Perform the following tasks to configure the voice ports on the Cisco MC3810:

- Configuring FXO or FXS Voice Ports
- Fine-Tuning FXO and FXS Voice Ports
- Configuring E&M Voice Ports
- Fine-Tuning E&M Voice Ports

- Activating the Voice Port

# Configuring FXO or FXS Voice Ports

Under most circumstances the default voice-port values are adequate for both FXO and FXS voice ports. If you need to change the default configuration for these voice ports, perform the following tasks. The first two tasks are required; the third task is optional.

1. Identify the voice port and enter the voice-port configuration mode.
2. Configure the following mandatory voice-port parameters:
   (a) Connection
   (b) Dial type
   (c) Signal type
   (d) CODEC
   (e) Call progress tone
3. Configure one or more of the following optional voice-port parameters:
   (a) Description
   (b) VAD
   (c) Comfort noise (if VAD is activated)

To configure FXO and FXS voice ports, use the following commands beginning in privileged EXEC mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **voice-port** *slot/port* | Enter voice-port configuration mode. The slot number for analog voice ports on the Cisco MC3810 is always 1. There is no port 0 for voice ports. |
| 2 | **connection** {**plar** \| **tie-line** \| **plar-opx**} *string* | Configure the voice-port connection mode type and the destination telephone number.<br>The **plar** value is used for Private Line Auto Ringdown (PLAR) connections. The **tie-line** value is used for a tie-line connection to a PBX. The **plar-opx** value is used for PLAR Off-Premises eXtension, to allow the local voice port to provide a local response before the remote voice port receives an answer. |
| 3 | **dial-type** {**pulse** \| **dtmf**} | Configure the voice-port dial-type. The default is dtmf (FXO only). |
| 4 | **signal** {**loop-start** \| **ground-start**} | Configure the signaling type for analog FXO and FXS voice ports. The default is loop-start. |

| Step | Command | Purpose |
|---|---|---|
| 5 | **codec** {**g729r8** \| **g729ar8** \| **g726r32** \| **g711alaw** \| **g711ulaw**} | Configure the voice-port compression mode.The g729ar8 value is the default and is recommended.<br>The g729ar8 compression mode can support a maximum of 24 simultaneously active on-net voice calls while the g729r8 value can only support a maximum of 12. The g729 compression modes have a nominal data rate of 8 kbps. |
| 6 | **compand-type** {**u-law** \| **a-law**} | Configure the companding standard used to convert between analog and digital signals in PCM systems. This command applies to digital voice-ports only. |
| 7 | **cptone** *country* | Configure the appropriate call progress tone for the local region.<br>The default for this command is **northamerica**. For a list of supported countries, refer to the *Voice, Video, and Home Applications Command Reference*. |
| 8 | **description** *string* | (Optional) Enter a string description for the voice port. The string describes the voice port in displays. You can use the description command to note the voice port's location or use. |
| 9 | **vad** | (Optional) Enable voice activity detection (VAD). |
| 10 | **voice confirmation-tone** | (Optional) If the voice port is configured for **connection plar-opx** for Off-Premises eXtension, disable the two-beep confirmation tone that a caller hears when picking up the handset. |

## Validation Tips

You can check the validity of your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for a dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.
- Use the **show voice port** command to verify that the data configured is correct.
- Use the **show voice dsp** command to verify the current status of all DSP voice channels.
- Use the **show voice call** summary command to verify the call status for all voice ports.

### Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Network Protocols Configuration Guide, Part 1*.
- Use the **show voice port** command to make sure that the port is enabled. If the port is offline, use the **no shutdown** command.
- Check the dial-peer configuration.
- Check the Frame Relay, ATM, or HDLC configuration.
- Check to see if the voice network module has been correctly installed. For more information, refer to the *Cisco MC3810 Multiservice Concentrator Hardware Installation Guide*.

## Fine-Tuning FXO and FXS Voice Ports

Depending on the specifics of your particular network, you may need to adjust voice parameters involving timing, input gain and output attenuation for FXO or FXS voice ports. Collectively, these commands are referred to as voice-port tuning commands.

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**Note** In most cases, the default values for voice-port tuning commands will be sufficient.

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To configure voice-port tuning applicable for both FXO and FXS voice ports, perform the following tasks:

1. Identify the voice port and enter the voice-port configuration mode.
2. For each of the following parameters, select the appropriate value:
   (a) Input gain
   (b) Output attenuation
   (c) Echo cancel coverage
   (d) Initial digit timeouts
   (e) Interdigit timeouts
   (f) Timing other than timeouts
   (g) Impedance (FXO voice ports only)
   (h) Ring number (FXO voice ports only)
   (i) Ring frequency (FXS voice port only)
   (j) Ring cadence (FXS voice port only)

To fine-tune FXO or FXS voice ports, perform the following steps beginning in privileged EXEC mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **configure terminal** | Enter global configuration mode. |
| 2 | **voice-port** *slot/port* | Identify the voice port you want to configure and enter voice-port configuration mode. |
| 3 | **input gain** *value* | Specify (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from –6 to 14. |
| 4 | **output attenuation** *value* | Specify (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14. |
| 5 | **echo-cancel enable** | Enable echo-cancellation of voice that is sent out the interface and received back on the same interface. |
| 6 | **echo-cancel coverage** *value* | Adjust the size (in milliseconds) of the echo-cancel. Acceptable values are 16, 24, and 32. |
| 7 | **timeouts initial** *seconds* | Configure the initial timeout value. The initial timeout value specifies the number of seconds the system waits for the caller to input the first digit of the dialed digits. The default is 10 seconds. |
| 8 | **timeouts interdigit** *seconds* | Configure the interdigit timeout value. The timeouts interdigit value specifies the number of seconds the system waits (after the caller has input the initial digit) for the caller to input a subsequent digit of the dialed digits. The default is 10 seconds. |
| 9 | **timing digit** *milliseconds* | If the voice-port dial type is DTMF, configure the DTMF digit signal duration. The range of the DTMF digit signal duration is from 50 to 100 milliseconds. The default is 100. |
| 10 | **timing inter-digit** *milliseconds* | If the voice-port dial type is DTMF, configure the DTMF inter-digit signal duration. The range of the DTMF inter-digit signal duration is from 50 to 500 milliseconds. The default is 100. |
| 11 | **timing pulse-digit** *milliseconds* | If the voice-port dial type is pulse, configure the pulse digit signal duration. The range of the pulse digit signal duration is from10 to 20 milliseconds. The default is 20. |
| 12 | **timing pulse-inter-digit** *milliseconds* | If the voice-port dial type is pulse, configure the pulse inter-digit signal duration. The range of the pulse inter-digit signal duration is from 100 to 1000 milliseconds. The default is 500. |
| 13 | **impedance** {**600r** \| **600c** \| **900r** \| **900c**} | (For FXO ports only) Configure the impedance. The default is 600 ohms real. |
| 14 | **ring number** *number* | (For FXO ports only) Configure the number of rings detected before a connection is closed on the FXO port. |

| Step | Command | Purpose |
|---|---|---|
| 15 | **ring frequency** *number* | (For FXS ports only) Specify the local ring frequency for the FXS voice port. The *number* value should be set to either 20 or 30. |
| 16 | **ring cadence** [**on1** \| **off1**] [**on2** \| **off2**] [**on3** \| **off3**] [**on4** \| **off4**] [**on5** \| **off5**] [**on6** \| **off6**] | (For FXS only) Specify the local ring cadence for the FX voice port. Using this command, specify the on and off pulses for the ring. The ring cadence differs depending on the local region. The units are in 100-millisecond units. |

**Note** After you change any voice-port command, it is a good idea to cycle the port by using the **shutdown** and **no shutdown** commands.

## Configuring E&M Voice Ports

Unlike FXO and FXS voice ports, the default E&M voice-port parameters most likely will not be sufficient to enable voice data transmission over your network.

**Note** E&M voice-port values must match those of the PBX to which it is connected. Refer to the documentation that came with your specific PBX for the appropriate E&M voice-port configuration command values.

To configure an E&M voice port on the Cisco MC3810, perform the following tasks. The first two tasks are required; the third task is optional.

1. Identify the voice port and enter the voice-port configuration mode.
2. Configure the appropriate value for each of the following required parameters:
   (a) Connection
   (b) Dial type
   (c) Cabling scheme
   (d) Interface type
   (e) Signal type
   (f) CODEC
   (g) Call progress tone
3. Configure one or more of the following optional parameters:
   (a) Description
   (b) VAD
   (c) Comfort noise (if VAD is activated)

To configure E&M voice ports, use the following commands beginning in privileged EXEC mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **configure terminal** | Enter global configuration mode. |
| 2 | **voice-port** *slot/port* | Identify the voice port you want to configure and enter voice-port configuration mode. |
| 3 | **connection** {**plar** \| **tie-line** \| **plar-opx**} *string* | Configure the voice-port connection mode type and the destination telephone number.<br>The **plar** value is used for Private Line Auto Ringdown (PLAR) connections. The **tie-line** value is used for a tie-line connection to a PBX. The **plar-opx** value is used for PLAR Off-Premises eXtension, to allow the local voice port to provide a local response before the remote voice port receives an answer. |
| 4 | **dial-type dtmf** | Select the appropriate dial type for out-dialing. For E&M voice ports, the only available choice is DTMF. |
| 5 | **operation** {**2-wire** \| **4-wire**} | Select the appropriate cabling scheme for this voice port. |
| 6 | **type** {**1** \| **2** \| **3** \| **5**} | Select the appropriate E&M interface type.<br>Type 1 indicates the following lead configuration:<br>E—output, relay to ground<br>M—input, referenced to ground<br>Type 2 indicates the following lead configuration:<br>E—output, relay to SG<br>M—input, referenced to ground<br>SB—feed for M, connected to –48V<br>SG—return for E, galvanically isolated from ground<br>Type 3 indicates the following lead configuration:<br>E—output, relay to ground<br>M—input, referenced to ground<br>SB—connected to –48V<br>SG—connected to ground<br>Type 5 indicates the following lead configuration:<br>E—output, relay to ground<br>M—input, referenced to –48V. |
| 7 | **signal** {**wink-start** \| **immediate** \| **delay-dial**} | Configure the signaling type for E&M voice ports. The default is wink-start. |
| 8 | **codec** {**g729r8** \| **g729ar8**} | Configure the voice-port compression mode.The g729ar8 value is the default and is recommended.<br>The g729ar8 compression mode can support a maximum of 24 simultaneously active on-net voice calls while the g729r8 value can only support a maximum of 12. Both compression modes have a nominal data rate of 8 kbps. |
| 9 | **compand-type** {**u-law** \| **a-law**} | Configure the companding standard used to convert between analog and digital signals in PCM systems. This command applies to digital voice-posts only. |

| Step | Command | Purpose |
|---|---|---|
| 10 | **cptone** *country* | Configure the appropriate call progress tone for the local region.<br>The default for this command is **northamerica**. For a list of supported countries, refer to the *Voice, Video, and Home Applications Command Reference*. |
| 11 | **description** *string* | (Optional) Attach descriptive text about this voice port connection. |
| 12 | **vad** | (Optional) Enable voice activity detection (VAD). |
| 13 | **voice confirmation-tone** | (Optional) If the voice port is configured for **connection plar-opx** for Off-Premises eXtension, disable the two-beep confirmation tone that a caller hears when picking up the handset. |

### Validation Tips

You can check the validity of your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for a dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.
- Use the **show voice port** command to verify that the data configured is correct.
- Use the **show voice dsp** command to verify the current status of all DSP voice channels.
- Use the **show voice call** summary command to verify the call status for all voice ports.

### Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Network Protocols Configuration Guide, Part 1*.
- Use the **show voice port** command to make sure that the port is enabled. If the port is offline, use the **no shutdown** command.
- Make sure that the values pertaining to your specific PBX setup, such as timing and type, are correct.
- Check to see if the voice network module has been correctly installed. For more information, refer to the Cisco MC3810 *Multiservice Concentrator Hardware Installation Guide*.

## Fine-Tuning E&M Voice Ports

Depending on the specifics of your particular network, you may need to adjust voice parameters involving timing, input gain and output attenuation for E&M voice ports. Collectively, these commands are referred to as voice-port tuning commands.

**Note** In most cases, the default values for voice-port tuning commands will be sufficient.

To configure voice-port tuning for E&M voice ports, perform the following tasks:

1. Identify the voice port and enter the voice-port configuration mode.
2. Select the appropriate value for each of the following parameters:
   (a) Input gain
   (b) Output attenuation
   (c) Echo cancel coverage
   (d) Non-linear processing
   (e) Initial digit timeouts
   (f) Interdigit timeouts
   (g) Timing other than timeouts

To fine-tune E&M voice ports, use the following commands beginning in privileged EXEC mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **configure terminal** | Enter global configuration mode. |
| 2 | **voice-port** *slot/port* | Identify the voice port you want to configure and enter voice-port configuration mode. |
| 3 | **input gain** *value* | Specify (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from –6 to 14. |
| 4 | **output attenuation** *value* | Specify (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14. |
| 5 | **echo-cancel enable** | Enable echo-cancellation of voice that is sent out the interface and received back on the same interface. |
| 6 | **echo-cancel coverage** *value* | Adjust the size (in milliseconds) of the echo-cancel. Acceptable values are 16, 24, and 32. |
| 7 | **non-linear** | Enable non-linear processing, which shuts off any signal if no near-end speech is detected. (Non-linear processing is used with echo-cancellation.) |
| 8 | **timeouts initial** *seconds* | Configure the initial timeout value. The initial timeout value specifies the number of seconds the system waits for the caller to input the first digit of the dialed digits. The default is 10 seconds. |
| 9 | **timeouts interdigit** *seconds* | Configure the interdigit timeout value. The timeouts interdigit value specifies the number of seconds the system waits (after the caller has input the initial digit) for the caller to input a subsequent digit of the dialed digits. The default is 10 seconds. |
| 10 | **timeouts wait-release** {*value* \| **infinity**} | Configure the timeout value for releasing voice ports. This command limits the duration that a voice port stays in the call failure state while the Cisco MC3810 sends a busy tone, reorder tone or out-of-service tone to the port. |

| Step | Command | Purpose |
|---|---|---|
| 11 | **timing digit** *milliseconds* | If the voice-port dial type is DTMF, configure the DTMF digit signal duration. The range of the DTMF digit signal duration is from 50 to 100 milliseconds. The default is 100. |
| 12 | **timing inter-digit** *milliseconds* | If the voice-port dial type is DTMF, configure the DTMF inter-digit signal duration. The range of the DTMF inter-digit signal duration is from 50 to 500 milliseconds. The default is 100. |
| 13 | **timing pulse-digit** *milliseconds* | If the voice-port dial type is pulse, configure the pulse digit signal duration. The range of the pulse digit signal duration is from10 to 20 milliseconds. The default is 20. |
| 14 | **timing pulse-inter-digit** *milliseconds* | If the voice-port dial type is pulse, configure the pulse inter-digit signal duration. The range of the pulse inter-digit signal duration is from 100 to 1000 milliseconds. The default is 500. |
| 15 | **timing wink-duration** *milliseconds* | Configure the timing wink-duration value. This value sets the wink signal duration for a wink-start signal. This value applies only if the **signal** command is set to "wink-start." The range is from 100 to 400 milliseconds and the default is 200. |
| 16 | **timing wink-wait** *milliseconds* | Configure the timing wink-wait value. This value sets the wink wait duration for a wink-start signal. This value applies only if the **signal** command is set to "wink-start." The range is from 100 to 5000 milliseconds and the default is 200. |
| 17 | **timing clear-wait** *milliseconds* | Configure the timing clear-wait value. This value sets the amount of time between the inactive seizure signal and the call being cleared. The range is from 100 to 2000 milliseconds and the default is 400. |
| 18 | **timing delay-duration** *milliseconds* | Configure the timing delay-duration value. This value sets the delay signal duration for delay dial signaling. This value applies only if the **signal** command is set to "delay-dial." The range is from 100 to 5000 milliseonds and the default is 140. |
| 19 | **timing delay-start** *milliseconds* | Configure the timing delay-start value. This value sets the delay interval between the generation of the delay-start signal from incoming detection seizure. This value applies only if the **signal** command is set to "delay-dial." The range is from 100 to 290 milliseconds and the default is 150. |
| 20 | **timing percentbreak** *percent* | Configure the timing percent-break value. This value sets the percentage of the break period for a dialing pulse. The default is 50 percent. |

**Note** After you change any voice-port command, it is a good idea to cycle the port by using the **shutdown** and **no shutdown** commands.

## Activating the Voice Port

After you have configured the voice port, you need to activate the voice port to bring it online. In fact it is a good idea to cycle the port—meaning to shut the port down and then bring it online again.

To activate a voice port, use the following command in voice-port configuration mode:

| Command | Purpose |
|---|---|
| **no shutdown** | Activate the voice port. |

To cycle a voice port, use the following commands in voice-port configuration mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **shutdown** | Deactivate the voice port. |
| 2 | **voice-port** *slot-number/subunit-number/port*<br>or<br>**voice-port** *slot/port* | Identify the voice port you want to activate and enter the voice-port configuration mode. |
| 3 | **no shutdown** | Activate the voice port. |

**Note** If you are not going to use a voice port, shut it down.

## Anexo 2: Configuración de VoIP en los puertos de voz

El siguiente archivo presenta una descripción más detallada acerca de los pasos necesarios para programar los puertos de voz de los routers Cisco.



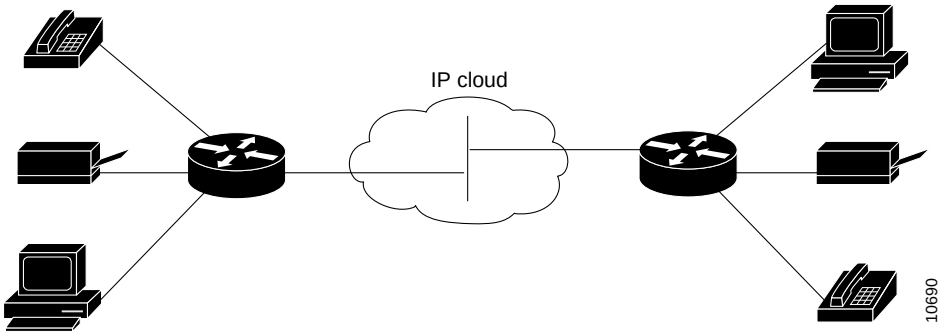
Customer Order Number: DOC-VOICEOIP-QSG=
Text Part Number: 78-4936-01

# Voice over IP Quick Start Guide

**Product Numbers: NM-1V=, NM-2V=**
**VIC-2E/M=, VIC-2FXO=, VIC-2FXS=**

The voice over IP feature enables Cisco 3600 series modular routers to carry voice traffic, such as telephone calls and faxes, over an IP network, simultaneously with data traffic. (See Figure 1.)

**Figure 1 Voice and Data Traffic on an IP Network**




This guide explains briefly how to install voice hardware and how to set up basic configurations of Cisco IOS software for a voice over IP network. It contains the following sections:

- **What You Should Know**, page 3—Describes what you should know before starting, what you should do before configuring your voice network, conventions used in this guide, and how to find more information.
- **Installing Voice Network Modules and Voice Interface Cards**, page 4—Describes voice hardware and explains how to install and connect it.
- **Voice Port Numbering**, page 8—Explains the interface numbering convention for voice ports.
- **Entering Configuration Mode**, page 10—Offers hints for using Cisco IOS software to configure a voice over IP network.

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# What You Should Know

This guide is intended for data communications managers or telecommunications managers who are installing, configuring, or maintaining voice network modules and interface cards. It assumes that you already have a working IP LAN or WAN, or that you know how to set one up, and that you want to add voice capability to it. It does not tell you how to design or install a LAN or WAN, assign IP addresses, install routers and configure them for IP traffic, or perform other basic tasks. It also assumes that you are familiar with Cisco IOS software, and that you know how to configure the routers on your LAN or WAN for IP service.

For E&M installations, you should also be familiar with telephony concepts and PBX operation.

The "List of Terms" near the end of this guide provides a glossary of many terms and concepts used by voice over IP. For further information on voice over IP features, refer to the *Voice Network Module and Voice Interface Card Configuration Note*, the *Voice over IP Software Configuration Guide*, your router installation and configuration guide, the *Regulatory Compliance and Safety Information* document for your router, and the Cisco IOS configuration guides and command references.

## Document Conventions

In this guide:

- Commands in the text are shown in **boldface**.
- Variables for which you supply values are in *italics*.
- *Italics* are also used for the titles of publications and for new words or concepts.
- Information the router displays on the console screen is in `screen font`.
- Information that you enter at the console keyboard is in **`boldface screen font`**.

## To Learn More

Cisco documentation and additional literature are available in a CD-ROM package, which ships with your product. The Documentation CD-ROM, a member of the Cisco Connection Family, is updated monthly. Therefore, it might be more up to date than printed documentation. To order additional copies of the Documentation CD-ROM, contact your local sales representative or call customer service. The CD-ROM package is available as a single package or as an annual subscription.

You can also read Cisco documentation on the World Wide Web at http://www.cisco.com, http://www-china.cisco.com, or http://www-europe.cisco.com. From here, you can submit comments electronically. Click **Feedback** on the title bar, and then select **Documentation**. After you complete the form, click **Submit** to send it to Cisco. We appreciate your comments.

If you have questions or need help, refer to the section "Cisco Connection Online" at the end of this guide for further information.

# Installing Voice Network Modules and Voice Interface Cards

The voice over IP feature for Cisco 3600 series routers uses two types of hardware:

- Voice network modules
- Voice interface cards (VICs)

---

**Note** To transmit voice calls over an IP LAN or WAN, you need (in addition to the voice hardware) at least one other network module or WAN interface card in the router to provide the connection to the LAN or WAN. In most cases, this network module, or another network module or WAN interface card in the router, also serves to carry data traffic.

---

You should install and cable voice modules and VICs before performing the software configuration tasks explained later. If you need more detailed installation instructions, refer to the *Voice Network Module and Voice Interface Card Configuration Note*.

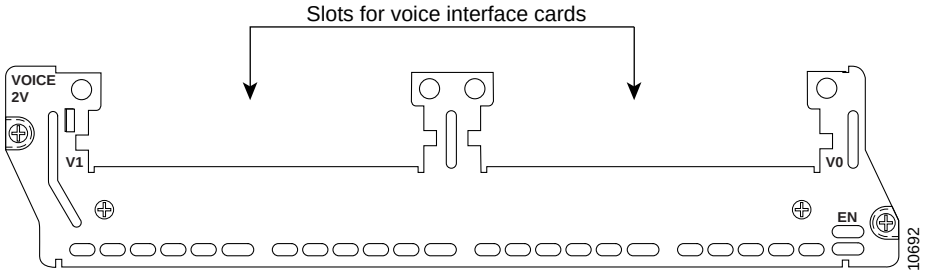
**Warning** Be sure to observe all warnings and safety precautions in the configuration note.

**Caution** Network modules and voice interface cards do not support online insertion and removal (hot swap). To avoid equipment damage, before you insert a network module into a chassis slot, or insert a voice interface card into a voice network module in the router chassis, you must turn OFF electrical power and disconnect network cables.

## Voice Network Modules

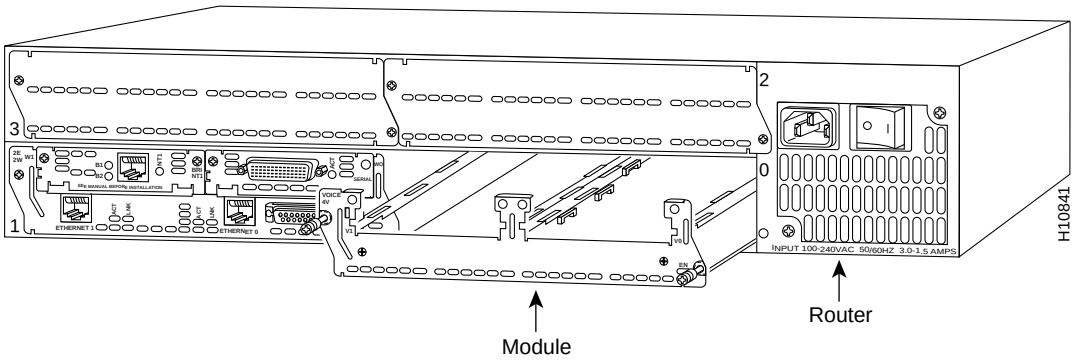
Voice network modules install in a slot in a Cisco 3600 series router, and convert telephone voice signals into a form that can be transmitted over an IP network. The one-slot voice network module provides one slot for a voice interface card. The two-slot voice network module provides two slots for voice interface cards. Figure 2 shows a two-slot voice network module.

**Figure 2 Two-Slot Voice Network Module**



To install a voice network module, slide it into a slot in the router chassis (Figure 3). Use a number 1 Phillips or small flat-blade screwdriver to fasten the module's mounting screws.

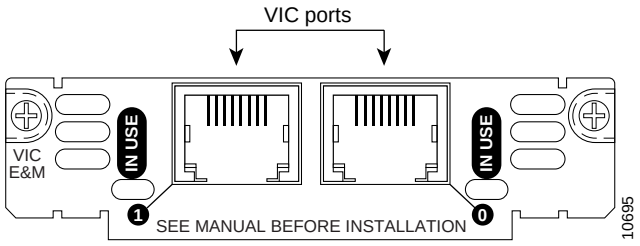
**Figure 3 Installing a Voice Network Module in a Router**




# Voice Interface Cards (VICs)

Voice interface cards install in slots in the voice network module, and provide connections to the telephone equipment or network. Figure 4 shows a typical VIC.

**Figure 4 Voice Interface Card**




There are three types of VIC interface:

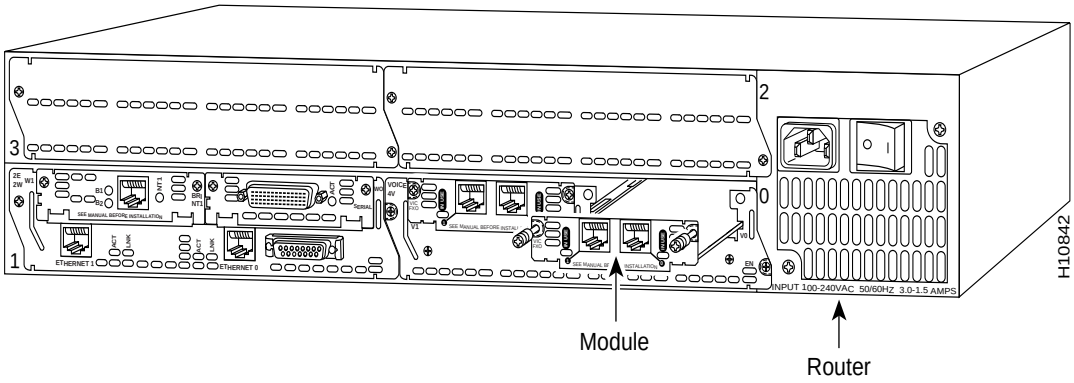
- An FXS (foreign exchange station) interface connects directly to a standard telephone, fax machine, or similar device. The FXS VIC interface supplies ringing voltage, dial tone, and similar signals to the station. Ports on this VIC are color-coded gray.
- An FXO (foreign exchange office) interface connects local calls to a public switched telephone network central office, or to a PBX that does not support E&M signaling. This is the interface a standard telephone provides. Ports on this VIC are color-coded pink.
- E&M (ear and mouth) is a signaling technique for two-wire and four-wire telephone and trunk interfaces. The E&M VIC connects remote calls from an IP network to a PBX for local distribution. Ports on this VIC are color-coded brown.

Each VIC provides two ports. You need one VIC port for each voice connection.

You can install one VIC (two voice ports) in a one-slot voice network module, and two VICs (four voice ports) in a two-slot voice network module.

To install a VIC, slide it into a slot in the voice network module, as shown in Figure 5. Use a number 1 Phillips or small flat-blade screwdriver to fasten the VIC's mounting screws.

**Figure 5 Installing a Voice Interface Card in a Voice Network Module**

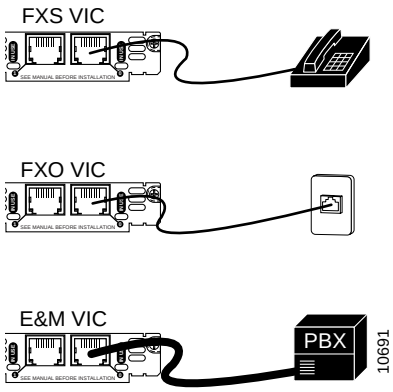


# Connecting VICs to the Network

Figure 6 shows how to connect VICs to the network:

- Use a standard RJ-11 modular telephone cable to connect FXS VIC ports (color-coded gray) to a telephone or fax machine.
- Use a standard RJ-11 modular telephone cable to connect FXO VIC ports (color-coded pink) to the public switched telephone network, or to a PBX that does not support E&M signaling, through a telephone wall outlet.
- The E&M VIC uses an RJ-48S connector and cable. The cable wiring depends on the type of connection. See the *Voice Network Module and Voice Interface Card Configuration Note* for details.

**Figure 6 Connecting VICs to the Network**




When you are finished, reinstall any network interface cables you removed and turn ON power to the router.

# Checking the Installation

If you installed an FXS VIC, connect a handset to the VIC port. When router power is on, you should be able to hear dial tone when you lift the handset. Dial tone should stop after you dial a digit. If you have trouble, use the **show voice port** command to make sure that the VIC is installed correctly. If it is, try connecting a different handset to the VIC.

# Voice Port Numbering

Cisco IOS configuration commands identify voice ports in the form

*router-slot*/*voice-slot*/*VIC-port*

**Timesaver** You can use the Cisco IOS **show voice port** command to identify the port numbers of voice interfaces installed in your router.

Figure 7 shows Cisco 3620 router slot numbering.

**Figure 7 Cisco 3620 Slot Numbers**

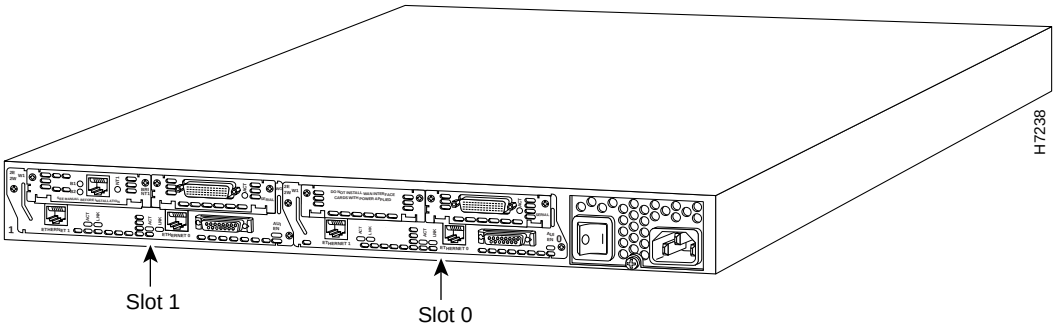



Figure 8 shows Cisco 3640 router slot numbering.

**Figure 8 Cisco 3640 Slot Numbers**

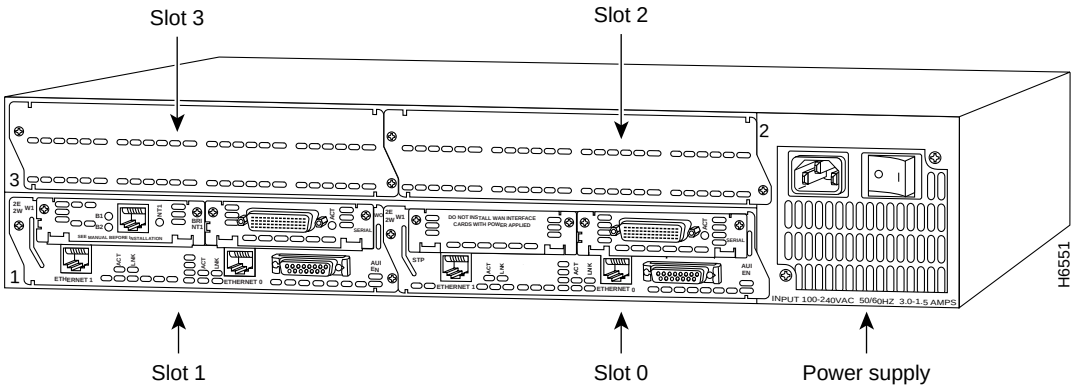


Figure 9 shows voice slot numbering.

**Figure 9 Voice Slot Numbering**

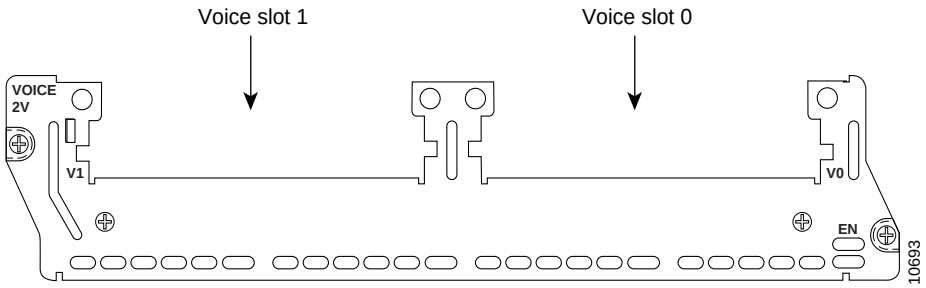
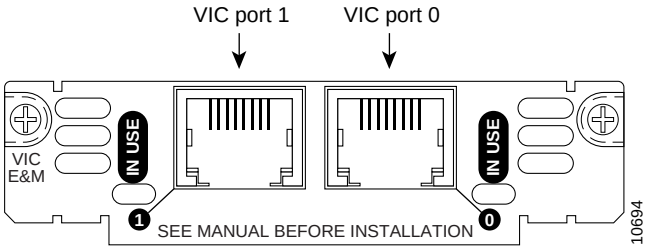



Figure 10 shows VIC port numbering.

**Figure 10 VIC Port Numbering**




# Example

Suppose you install a two-slot voice network module in the upper right slot of a Cisco 3640 router, and install two VICs in the module. Each VIC has two ports. From right to left, these ports would be numbered 2/0/0, 2/0/1, 2/1/0, and 2/1/1.

# Entering Configuration Mode

You configure Cisco IOS software for voice over IP by typing commands on the command line. This method of entering commands is called configuration mode.

**Note** Voice over IP commands require the IP Plus, Desktop Plus, or Enterprise Plus image, Cisco IOS Release 11.3(1)T or later.

To enter configuration mode, follow this procedure:

**Step 1** Connect a console to the router. Configure the console for 9600 baud, 8 data bits, 1 stop bit, and no parity. Power up the router.

**Step 2** If the current configuration is no longer valid (for instance, because you added an interface), after about one minute you see the following prompt:

```
Would you like to enter the initial configuration dialog? [yes]:
```

Answer **no**. You now enter the normal operating mode of the router.

**Note** If the current configuration is valid, you enter the normal operating mode automatically.

**Step 3** After a few seconds, you see the user EXEC prompt (`Router>`). Type **enable** and the password to enter enable mode:

```
Router> enable
Password:
```

Configuration changes can be made only in enable mode. The prompt changes to the privileged EXEC (enable) prompt (`Router#`):

```
Router#
```

**Step 4** Enter the **config terminal** command to enter configuration mode:

```
Router# config terminal
Router(config)#
```

The router enters global configuration mode, indicated by the `Router(config)#` prompt.

**Step 5** If you have not configured the router before, or want to change the configuration, configure global parameters, passwords, network management, and routing protocols. In this example, IP routing, AppleTalk routing, and IPX routing are all enabled:

```
Router(config)# ip routing
Router(config)# appletalk routing
Router(config)# ipx routing
```

For complete information about global configuration commands, and about configuring LAN and WAN interfaces on your router, refer to the Cisco IOS configuration guides and command references.

The rest of this guide explains how to configure Cisco IOS software for voice over IP traffic.

**Timesaver** Voice configuration uses a number of new Cisco IOS commands. For complete information about these commands, see the *Voice over IP Software Configuration Guide*. You can also enter a question mark after a command or partial command at the `Router(config)#` prompt to get help with syntax and arguments.

Remember that when you finish configuring the software, you must save the new configuration to NVRAM. This procedure is described in the section “Saving the Configuration” later in this guide. You may also want to save periodically during the configuration process.

**Timesaver** At any point, you can see the current operating configuration, including changes you just made, by entering the **show running-config** command.

# Configuring the IP Network for Real-Time Voice Traffic

Voice traffic is much more sensitive to timing variations than data traffic. For good voice performance, you might need to configure your data network so voice packets are not lost or delayed. This section describes three important methods of improving quality of service, or QoS (the level of network performance needed for voice over IP connections):

- RSVP
- Multilink PPP interleaving
- RTP header compression

Cisco IOS software provides many other tools for ensuring QoS, such as custom queuing, priority queuing, weighted fair queuing, and IP precedence. For further information and more detailed examples of QoS configuration, see the *Voice over IP Software Configuration Guide*.

---

**Note** QoS measures the level of network performance. It does not directly measure the quality of the voice signal.

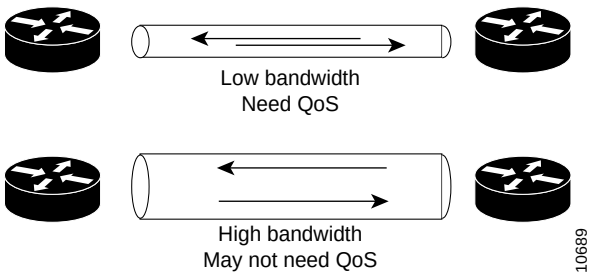
---

Configuring voice over IP on a Frame Relay link involves special considerations. These are discussed in the section “Configuring Frame Relay for Voice over IP.”

## Do You Need QoS?

On a relatively low-bandwidth connection, such as a PPP or HDLC serial link, you should consider using methods to ensure QoS. If you have a high-bandwidth network, such as Ethernet or Fast Ethernet, and voice and data traffic together occupy only a small fraction of the bandwidth available, you may not need to provide QoS. (See Figure 11.) If you are not sure, you may want to try skipping the following sections and continuing with “Configuring FXS Interfaces” later in this guide. Return to this section if you need to improve the reliability of voice connections.

**Figure 11 Bandwidth v. Quality of Service**



# Do You Need RSVP?

Resource Reservation Protocol (RSVP) enables routers to reserve enough bandwidth for reliable performance. RSVP works well on PPP, HDLC, and similar serial line interfaces. It does not work well on multiaccess LANs.

You should configure RSVP if you have a serial interface and any of the following:

- Links slower than 2 Mbps
- Links with high utilization
- Need for the best possible voice quality

---

**Note** If you configure multilink PPP interleaving, you can use the **ip rtp reserve** command instead of configuring RSVP. See the next section on "Do You Need Multilink PPP Interleaving?"

---

## Configuring RSVP

By default, RSVP is disabled for compatibility with routers that do not implement it. To enable RSVP on an IP network, enter the **ip rsvp bandwidth** command from interface configuration mode. The following example shows how to configure RSVP on serial interface 0/0:

```
Router> enable
Password:
Router# configure terminal
Router(config)# interface serial 0/0
Router(config-if)# ip rsvp bandwidth
```

The default maximum bandwidth is 75 percent of the bandwidth available on each interface.

RSVP must be enabled at each LAN or WAN interface that voice packets will travel across. You must also configure each remote dial peer to request an RSVP session, using the **req-qos** command. See the section "Calling Between Routers" later in this guide.

# Do You Need Multilink PPP Interleaving?

On a dialer, ISDN PRI or BRI interface, or a virtual template, you can configure multilink PPP interleaving, which helps to make sure that voice packets are transmitted without delay.

You should configure multilink PPP interleaving if you have one of these interfaces, and either of the following:

- Point-to-point connections using PPP encapsulation
- Links slower than 2 Mbps

---

**Note** Do not use multilink PPP on links faster than 2 Mbps.

---

### Configuring Multilink PPP Interleaving

To configure multilink PPP and interleaving on a dialer, ISDN PRI, or ISDN BRI interface, or a virtual template, you must first configure multilink PPP and interleaving on the interface or template by entering the following commands in interface configuration mode:

```
Router(config-if)# encapsulation ppp
Router(config-if)# ppp multilink
Router(config-if)# ppp multilink interleave
```

Optionally, configure a maximum fragment delay:

```
Router(config-if)# ppp multilink fragment-delay milliseconds
```

You can also reserve a special queue for real-time packet flows to specified destination UDP ports, allowing real-time traffic to have higher priority than other flows. This command is needed only if you have not configured RSVP:

```
Router(config-if)# ip rtp reserve lowest-UDP-port range-of-ports
```

For virtual templates only, apply the virtual template to the multilink bundle:

```
Router(config-if)# multilink virtual-template 1
```

## Do You Need RTP Header Compression?

RTP header compression on a PPP, HDLC, or similar serial interface compresses the packet header to reduce network overhead.

You should configure RTP header compression on a serial interface if you have either of the following:

- Links slower than 2 Mbps
- Need to save bandwidth

**Note** Do not use RTP header compression on links faster than 2 Mbps.

### Configuring RTP Header Compression

Enable RTP header compression at both ends of the serial link by entering the following command in interface configuration mode:

```
Router(config-if)# ip rtp header-compression
```

# Configuring Frame Relay for Voice over IP

Configuring voice over IP on a Frame Relay link involves certain special considerations to ensure acceptable voice quality. For Frame Relay links with slow output rates (64 kbps or less), where data and voice are being transmitted over the same PVC, you should configure the following:

- Lower MTU size—Voice packets are generally small. If you decrease the MTU size to 300 bytes, large data packets can be broken up into smaller data packets that are more easily interleaved with voice packets.

  The following example configures an MTU size of 300 bytes over serial interface 0/0:

  ```
  Router# interface serial 0/0
  Router(config-if)# mtu 300
  ```

- RSVP—Configure RSVP on the subinterfaces (which correspond to PVCs) to reserve bandwidth for voice channels. See the section on "Do You Need RSVP?" earlier in this guide.

  The following example configures RSVP over serial subinterface 0/0.1:

  ```
  Router(config-if)# interface serial 0/0.1 point-to-point
  Router(config-if)# ip address 192.168.19.0 255.0.0.0
  Router(config-if)# ip rsvp bandwidth
  ```

- RTP header compression—Configure RTP header compression on the subinterfaces to minimize the size of the voice packet. See the previous section on "Do You Need RTP Header Compression?"

  The following command configures RTP header compression over the currently selected subinterface:

  ```
  Router(config-if)# frame-relay ip rtp header-compression
  ```

- Traffic shaping—Use traffic shaping to control the outbound traffic rate; otherwise, voice packets might be discarded. In Cisco IOS Release 11.3, Frame Relay traffic shaping is not compatible with RSVP. Use generic traffic shaping instead, setting the CIR equal to the port speed. Doing this prevents the router from exceeding the CIR rate, which could lead to frames being discarded.

  The following command configures generic traffic shaping with a CIR of 32000 bps:

  ```
  Router(config-if)# traffic-shape rate 32000
  ```

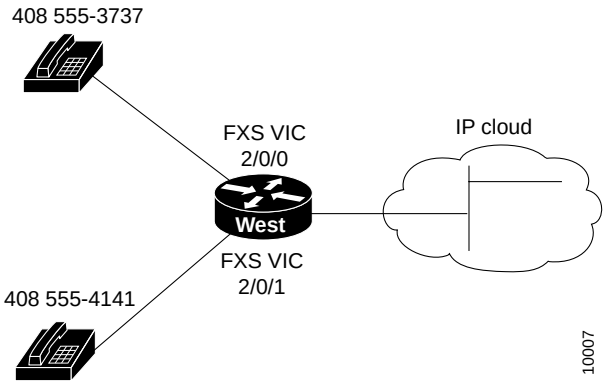
For further information and more detailed examples of Frame Relay configuration, see the *Voice over IP Software Configuration Guide*.

# Configuring FXS Interfaces

This section explains how to configure FXS VICs. Ports on this VIC connect directly to a standard telephone, fax machine, or similar device.

Figure 12 shows a basic voice network. A small business uses a Cisco 3600 series router (named West) to provide telephone and fax connections among employees in its office. Figure 12 shows two of these telephones, each connected to an FXS VIC port in the West router.

**Figure 12 Basic Voice Network (West Router)**




**Note** You can name your router by using the **hostname** command in global configuration mode.

Table 1 lists telephone numbers and voice ports for the West router. (For information about port numbering, see the section on “Voice Port Numbering” earlier in this guide.)

**Table 1 West Router Telephone Numbers and Voice Ports**

| Telephone Number | Voice Port |
|---|---|
| 408 555-3737 | 2/0/0 |
| 408 555-4141 | 2/0/1 |

**Note** Additional telephones and fax machines could be connected in this example, up to a total of 12 if the router is a Cisco 3640 configured with three two-slot voice network modules. The module in the remaining slot would provide interfaces for IP connectivity to the LAN or WAN and for data traffic. To accommodate more than 12 voice devices, you would need to add more routers, or to use an E&M VIC and a local PBX, rather than connecting every telephone to its own FXS VIC.

# Local Dial Peers

To route a received voice call to the right destination, the router needs to know which telephone number belongs to each voice port. For instance, if a call comes in for 408 555-3737, the router needs to know that this telephone is connected to voice port 2/0/0 (as shown in Figure 12). In other words, the router needs to know the information in Table 1.

To hold this information, Cisco IOS software uses objects called *dial peer*s. A telephone number, a voice port, and other call parameters are tied together by associating them all with the same dial peer. Configuring dial peers is similar to configuring static IP routes—you are telling the router what path to follow to route the call.

Dial peers are identified by numbers, but to avoid confusing these numbers with telephone numbers, they are usually referred to as *tags*. Dial peer tags are integers that can range from 1 to $2^{31}$ -1 (2147483647), which should be enough for most purposes. Dial peers on the same router must have unique tags, but you can reuse the tags on other routers.

Table 2 assigns each phone number-voice port pair on the West router a dial peer tag. Within the allowed range, you can choose any dial peer tag or any system that is convenient or makes sense to you. This type of dial peer is called a *POTS dial peer* or a *local dial peer.* The term “POTS” (plain old telephone service) means that the dial peer associates a physical voice port with a local telephone device. (Another type of dial peer is explained in the section “Calling Between Routers.”)

**Table 2 West Router Local Dial Peers**

| Telephone Number | Voice Port | Dial Peer Tag |
|---|---|---|
| 408 555-3737 | 2/0/0 | 401 |
| 408 555-4141 | 2/0/1 | 402 |

You should construct a table similar to Table 2 for your own routers, assigning your own telephone numbers and dial peer tags.

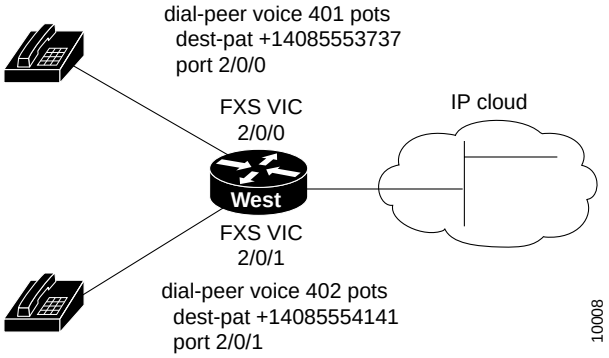
**Note** The telephone numbers used in this guide are only examples, and are generally invalid for public use in the United States. When you configure your network, be sure to substitute your own telephone numbers.

To configure the router with the information in the table, enter the following commands in global configuration mode:

```
West(config)# dial-peer voice 401 pots
West(config-dial-peer)# dest-pat +14085553737
West(config-dial-peer)# port 2/0/0
West(config)# dial-peer voice 402 pots
West(config-dial-peer)# dest-pat +14085554141
West(config-dial-peer)# port 2/0/1
West(config-dial-peer)# exit
West(config)#
```

These commands are summarized in Figure 13.

**Figure 13 West Router Configured for Local Dial Peers**




The **dial-peer** command always takes the argument **voice**. The number following it is the dial peer tag, and **pots** is the type of dial peer.

Cisco IOS software refers to a telephone number as a *destination pattern*, because it is the destination for an incoming or outgoing call. Enter these numbers with the **destination-pattern** (abbreviated **dest-pat**) command. A destination pattern always begins with a plus sign (+). It can also include asterisks (*) and pound signs (#) from the telephone keypad, and commas (,) and periods (.), which have special meaning. Parentheses ( () ), hyphens (-), slashes (/), and spaces ( ), which are often used to make telephone numbers easier for humans to read, are not allowed.

Notice that the commands in the examples append the prefix 1 (used in the United States to indicate a long-distance number) and an area code to the destination pattern. You may need to include similar codes for your country if the voice over IP equipment needs to establish a connection to the PSTN. In other situations, you might be able to simplify configuration by omitting this information.

**Note** The Cisco IOS software does not check the validity of the telephone number. It accepts any string of permitted characters as a valid number.

The business that owns the West router also has a branch office in the East. Figure 14 shows the East network, and Table 3 lists phone numbers, voice ports, and dial peer tags for this office.

**Figure 14 Basic Voice Network (East Router)**

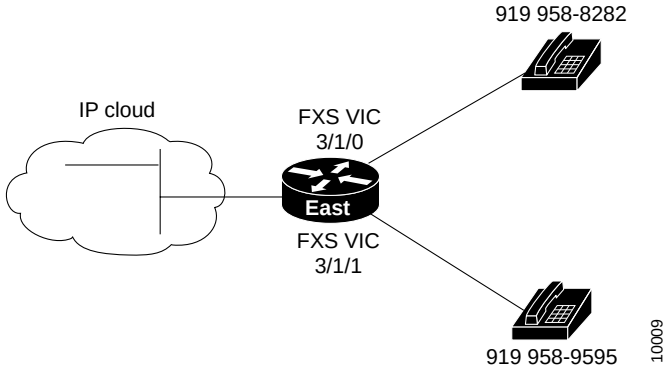



**Table 3 East Router Local Dial Peers**

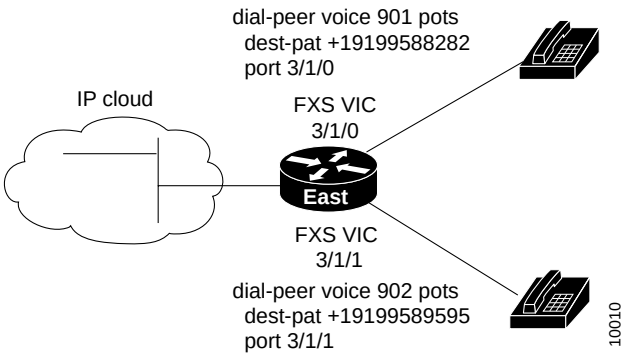
| Telephone Number | Destination Pattern | Voice Port | Dial Peer Tag |
|---|---|---|---|
| 919 958-8282 | +19199588282 | 3/1/0 | 901 |
| 919 958-9595 | +19199589595 | 3/1/1 | 902 |

The following commands configure local ports on the East Router:

```
East(config)# dial-peer voice 901 pots
East(config-dial-peer)# dest-pat +19199588282
East(config-dial-peer)# port 3/1/0
East(config)# dial-peer voice 902 pots
East(config-dial-peer)# dest-pat +19199589595
East(config-dial-peer)# port 3/1/1
East(config-dial-peer)# exit
East(config)#
```

These commands are summarized in Figure 15.

**Figure 15 East Router Configured for Local Dial Peers**




## Checking the Configuration

If you configured your router following these examples, you should now be able to place calls between telephones connected to the same router. You can also use the **show dial-peer voice** command to verify that the data you configured is correct.

**Timesaver** If the voice port is offline, use the **no shutdown** command from interface configuration mode to enable it.

**Note** Although placing calls directly between ports on the same router helps to verify your configuration, it is not recommended for general telecommunications use.

# Wild Cards and Number Expansion

Office PBXs are usually configured so a user can dial a local call (within the same PBX) by dialing the extension only—for instance, the four-digit extension 3737 or the five-digit extension 53737 instead of the full telephone number, 1 408 555-3737.

You can provide the same shortcut on a voice over IP network by using the **number-expansion** (**num-exp**) command. This command tells the router to expand a particular sequence of dialed numbers into a complete telephone number (destination pattern). For instance, to expand 3737 into 408 555-3737, enter the following command:

```
West(config)# num-exp 3737 +14085553737
```

To expand 4141 into 1 408 555-4141, enter the following command:

```
West(config)# num-exp 4141 +14085554141
```

More generally, you can use the period (.) as a wild-card character representing a single digit. For instance, the command

```
West(config)# num-exp .... +1408555....
```

expands any dialed sequence of four digits by prefixing +1408555 to it.

---

**Note** You must still configure each telephone number in full as a local dial peer so the router can find the voice port it belongs to.

---

To use five-digit extensions beginning with the numeral 5 rather than four-digit extensions, you would enter the following command:

```
West(config)# num-exp 5.... +1408555....
```

The corresponding commands for the East router would be (for four-digit extensions):

```
East(config)# num-exp .... +1919958....
```

or for five-digit extensions:

```
East(config)# num-exp 8.... +1919958....
```

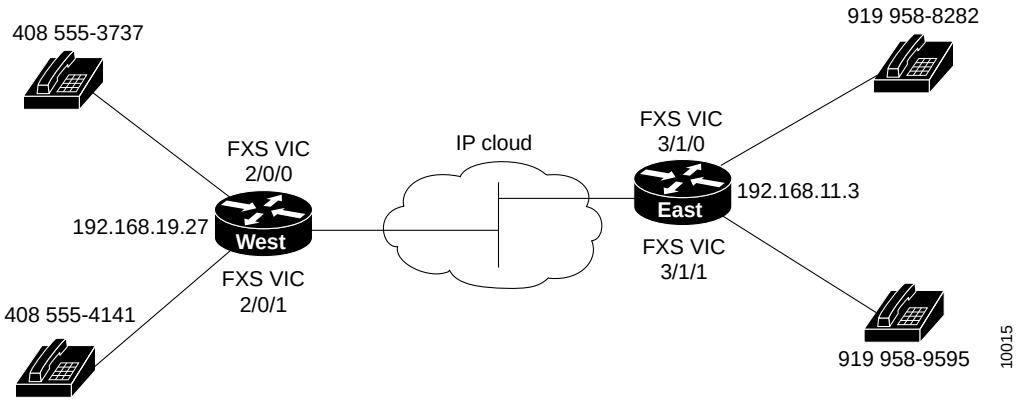
## Checking the Configuration

If you followed the examples, you should now be able to place calls between telephones connected to the same router using the extension instead of the full telephone number. You can use the **show num-exp** command to verify that the data you configured is correct.

# Calling Between Routers

Naturally, the West and East offices would like to send voice traffic to each other over the same IP network they use for data traffic. A WAN port of some type on each router connects to the IP WAN, as shown in Figure 16.

**Figure 16 IP Connection Between Routers**




Look at the connection between the West router and the IP network. This connection does not include a voice port or an attached telephone—it leads from a WAN interface to a remote destination somewhere on the IP network. IP routers know how to locate IP addresses on the network, but they do not know how to locate telephone numbers. To route an outgoing voice call over this connection, the West router has to associate a telephone number in the East office with the IP address of the East router.

This is done by associating both pieces of information with a *remote dial peer* or *VoIP dial peer* on the West router, as shown in Table 4. (Remember, the dial peer tags are arbitrary.) The term "VoIP" (voice over IP) means that the dial peer associates a telephone number with an IP address.

**Table 4 West Router Remote Dial Peers**

| Remote Location | Telephone Number | Destination Pattern | IP Address | Dial Peer Tag |
|---|---|---|---|---|
| East | 919 958-8282 | +19199588282 | 192.168.11.3 | 501 |
| East | 919 958-9595 | +19199589595 | 192.168.11.3 | 502 |

To do this, you could create a remote dial peer on the West router for every telephone on the East router, all associated with the same IP address. But it is much easier to use periods as wild cards, as shown in Table 5.

**Table 5 West Router Remote Dial Peers with Wild Cards**

| Remote Location | Telephone Number | Destination Pattern | IP Address | Dial Peer Tag |
|---|---|---|---|---|
| East | 919 958-xxxx | +1919958.... | 192.168.11.3 | 501 |

You should construct a table similar to Table 5 for your own routers, assigning your own telephone numbers, IP addresses, and dial peer tags.

**Note** The IP addresses shown in this guide are reserved, and are meant only as examples. When you configure your network, be sure to substitute your own IP addresses.

Now you need to enter only the following commands on the West router from global configuration mode:

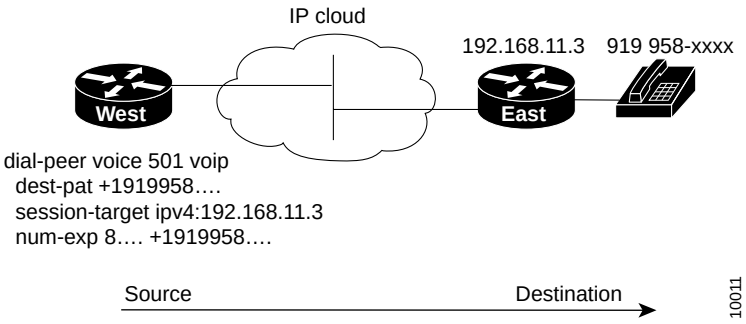
```
West(config)# dial-peer voice 501 voip
West(config-dial-peer)# dest-pat +1919958....
West(config-dial-peer)# session-target ipv4:192.168.11.3
```

Cisco IOS software calls the remote network the *session target*. This command is followed by the IP address of the remote router. The prefix **ipv4** means IP version 4. Alternatively, you can use the prefix **dns** followed by the DNS name—for instance:

```
West(config-dial-peer)# session-target dns:voice.eastrouter.com
```

These commands are summarized in Figure 17.

**Figure 17 West Router Configured for Remote Dial Peers**




In general, you have to configure a dial peer on each router for every telephone number (or group of wild cards) on every other router.

You can make things even easier by configuring number expansion for East router telephone numbers on the West router:

```
West(config)# num-exp 8.... +1919958....
```

Now users can dial a five-digit extension beginning with 8 from a telephone on the West router to reach a telephone on the East router.

Now the West router is configured to send calls to the East router. Table 6 shows how to configure the East router to send calls to the West router.

**Table 6 East Router Remote Dial Peers with Wild Cards**

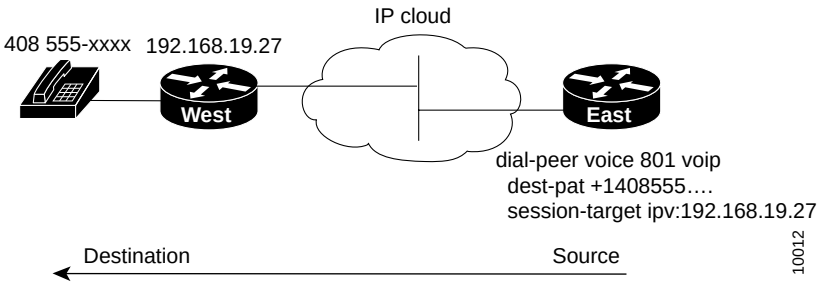
| Remote Location | Telephone Number | IP Address | Dial Peer Tag |
|---|---|---|---|
| West | 408 555-xxxx | 192.168.19.27 | 801 |

To create this configuration, you would enter the following commands on the East router:

```
East(config)# num-exp 5.... +1408555....
East(config)# dial-peer voice 801 voip
East(config-dial-peer)# dest-pat +1408555....
East(config-dial-peer)# session-target ipv4:192.168.19.27
```

These commands are summarized in Figure 18.

**Figure 18 East Router Configured for Remote Dial Peers**




## Requesting RSVP

If you configured RSVP on the WAN interface (see the section “Do You Need RSVP?” earlier in this guide), then you must also configure each VoIP dial peer to request an RSVP session, using the **req-qos** command:

```
West(config-dial-peer)# req-qos controlled-load
```

Otherwise no bandwidth is reserved for voice traffic. For further information about this command, see the *Voice over IP Software Configuration Guide*.

## Other Routers on the Network

If the path between endpoints of a voice call travels through intermediate routers, you should configure those routers for VoIP traffic, as described in the section “Configuring the IP Network for Real-Time Voice Traffic” earlier in this guide.

You need to configure local or remote dial peers on an intermediate router only if that router also has voice devices attached.

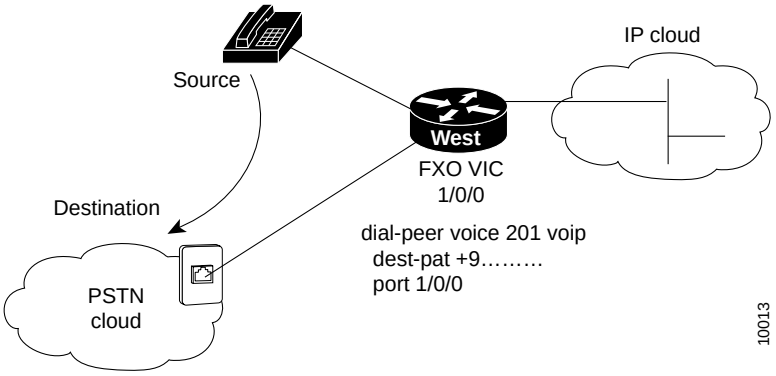
## Checking the Configuration

After you configure remote dial peers (VoIP dial peers) on a router, you should be able to place calls from that router to telephones on the remote routers (using just the extension if you configured number expansion). If you have trouble, use the **show dial-peer voice** command to verify that the data you configured is correct, and ping the remote router to make sure you have connectivity.

# Configuring FXO Interfaces

FXO interfaces provide a gateway from the VoIP network to the analog PSTN, or to a PBX that does not support E&M signaling, so users can reach telephones and fax machines outside the VoIP network. Figure 19 shows a typical PSTN gateway attached to the West router.

**Figure 19 FXO Gateway to PSTN**




To create a local (POTS) dial peer for an FXS interface, as explained earlier, you enter the complete telephone number of the attached telephone as the destination pattern for incoming calls. When you create a local dial peer for an FXO interface, however, the destination pattern refers to outgoing calls, and you can include wild cards in it, because the PSTN performs the switching.

The VoIP feature can also remove digits that you don't want to send to the PSTN. For instance, suppose you want to dial 9 to reach an outside line (that is, the analog PSTN). You would enter commands similar to these:

```
West(config)# dial-peer voice 201 pots
West(config-dial-peer)# dest-pat +9..........
West(config-dial-peer)# port 1/0/0
```

Now, when you dial 9 followed by an 11-digit telephone number from a telephone attached to the West router, your call is connected through voice port 1/0/0 to the PSTN. The router software automatically removes the fixed part of the destination pattern—in this case, the digit 9—and sends the remaining 11 digits to the PSTN.

---

**Note** The fixed part of the destination pattern is removed only on calls sent to analog (FXO) interfaces. The entire number is always sent to digital (FXS and E&M) interfaces.

---

To enable East router users to make calls over the West router's local PSTN, you could enter commands similar to these:

```
East(config)# dial-peer voice 701 voip
East(config-dial-peer)# dest-pat +7..........
East(config-dial-peer)# session-target ipv4:192.168.19.27

West(config)# dial-peer voice 601 pots
West(config-dial-peer)# dest-pat +7..........
West(config-dial-peer)# port 1/0/0
```

The East router now sends all calls whose numbers begin with the special prefix 7 over the IP network to the West router. (The entire number is sent, including the 7.) The West router removes the 7 and passes the calls through its analog FXO gateway to the local PSTN.

**Note** In this example, West voice port 1/0/0 has two separate POTS dials peers associated with it. Dial peer 201 matches calls beginning with the digit 9, and handles PSTN calls originating from the West router. Dial peer 601 matches calls beginning with the digit 7, and handles calls to the West PSTN originating from the East router.

## Checking the Configuration

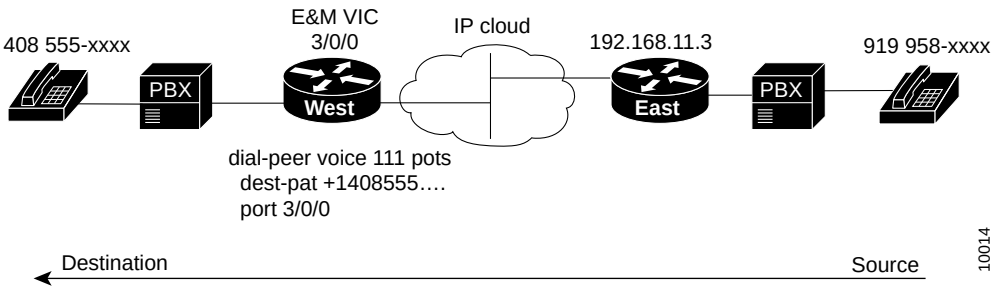
If you configured your FXO interface according to this example, you should now be able to place outgoing calls over the PSTN. If you have trouble, use the **show voice port** command to make sure that the VIC is installed correctly. Use the **show dial-peer voice** command to make sure that the data you configured is correct, and test the PSTN by connecting a handset directly to the PSTN outlet and placing a call.

# Configuring E&M Interfaces

If you have more than a few voice users at each location, then the cost of voice ports and routers, and the effort needed to configure dial peers for all the combinations of origins and destinations, increases rapidly. In this situation, it may be more efficient to use a PBX at each location to switch local traffic and to direct incoming calls, and to connect the PBXs over an IP network using E&M voice interface cards.

The following example again shows a company with two offices, West and East. Now each office has a PBX to operate its internal telephone network, while the IP network carries voice traffic between the offices. Figure 20 illustrates the topology of this connection. Each PBX connects to the IP router over an E&M interface connection.

**Figure 20 Linking PBXs over the IP Network (Local Dial Peers)**




Both PBXs in this example use E&M interface Type 2, with four-wire operation and immediate-start signaling. The values you should use for your configuration depend on your PBX, and should be available from your telecommunications department or the PBX manufacturer. For more information about E&M interface configuration commands, see the "Voice over IP Commands" chapter of the *Voice over IP Software Configuration Guide*.

In the following configuration, West users can dial 8 and a four-digit extension to reach telephones in the East Office. East users can dial 5 and a four-digit extension to reach telephones in the West office.

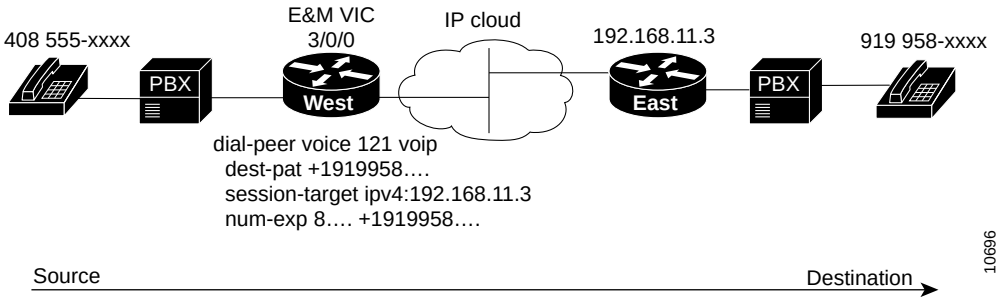
The West router is connected to the PBX over E&M port 3/0/0. That means that this port is associated with local (POTS) dial peers for incoming calls. But you no longer need to associate every telephone number with its own port. Instead, you can configure a local dial peer as if all the West telephones (represented by a wild-card destination pattern) are connected directly to this port, as shown in the following commands:

```
West(config)# dial-peer voice 111 pots
West(config-dial-peer)# dest-pat +1408555....
West(config-dial-peer)# port 3/0/0
```

Remote (VoIP) dial peers for outgoing calls associate destination phone numbers on the East router with that router's IP address, just as before (see Figure 21):

```
West(config)# dial-peer voice 121 voip
West(config-dial-peer)# dest-pat +1919958....
West(config-dial-peer)# session-target ipv4:192.168.11.3
West(config-dial-peer)# exit
West(config)#
```

**Figure 21 Linking PBXs over the IP Network (Remote Dial Peers)**




Now configure number expansion, so that numbers beginning with 8 (belonging to the East office) and sent by the West PBX to the West router are expanded into the full destination pattern:

```
West(config)# num-exp 8.... +1919958....
```

**Note** You do not need to configure number expansion for calls from one West telephone to another West telephone, because the PBX switches those calls.

Finally, configure the E&M port similarly to any other network interface, using the **voice-port** command from global configuration mode:

```
West(config)# voice-port 3/0/0
West(config-voice-port)# signal immediate
West(config-voice-port)# operation 4-wire
West(config-voice-port)# type 2
```

**Note** Configure the PBX to pass all DTMF signals to the router.

Configuration for the East router is similar. The PBX for this router is connected to E&M port 2/0/1. The following commands configure a local dial peer for all East telephones:

```
East(config)# dial-peer voice 211 pots
East(config-dial-peer)# dest-pat +1919958....
East(config-dial-peer)# port 2/0/1
```

These commands configure a remote dial peer for telephones on the West router:

```
East(config)# dial-peer voice 221 voip
East(config-dial-peer)# dest-pat +1408555....
East(config-dial-peer)# session-target ipv4:192.168.19.27
East(config-dial-peer)# exit
East(config)#
```

Configure number expansion, to make it easy for East users to dial numbers on the West router:

```
West(config)# num-exp 5.... +1408555....
```

Finally, configure the E&M port:

```
East(config)# voice-port 2/0/1
East(config-voice-port)# signal immediate
East(config-voice-port)# operation 4-wire
East(config-voice-port)# type 2
```

## Checking the Configuration

If you configured the E&M interfaces correctly, you should now be able to place calls from a telephone served by one PBX to a telephone served by the other PBX (using just the extension, if you configured number expansion). If you have trouble, ping the remote router to make sure you have connectivity.

# Saving the Configuration

When you finish router configuration, follow this procedure for each router to write the new configuration to NVRAM:

**Step 1** Exit configuration mode and return to the enable prompt by pressing **Ctrl-Z**. To see the current operating configuration, including any changes you just made, enter the **show running-config** command:

```
Router# show running-config
```

To see the configuration currently stored in NVRAM, enter the **show startup-config** command at the enable prompt:

```
Router# show startup-config
```

**Step 2** The results of the **show running-config** and **show startup-config** commands differ from each other if you have made changes to the configuration, but have not yet written them to NVRAM. To write your changes to NVRAM, making them permanent, enter the **copy running-config startup-config** command at the enable prompt:

```
Router# copy running-config startup-config
Building configuration. . .
[OK]
Router#
```

The router is now configured to boot in the new configuration.

If you have questions or need help, see the last section, “Cisco Connection Online.”

# List of Terms

This section defines some of the terms and concepts used by voice over IP.

**BRI**—Basic Rate Interface.

**call leg**—a segment of a call path; for instance, between a telephone and a router, a router and a network, a router and a PBX, or a router and the PSTN. Each call leg corresponds to a dial peer.

**CIR**—committed information rate. The average rate of information transfer a subscriber (for example, the network administrator) has stipulated for a Frame Relay PVC.

**destination pattern**—the pattern of numbers that identifies the destination of an incoming call; in other words, a phone number.

**dial peer**—a software object that ties together a voice port and a local telephone number (*local dial peer* or *POTS dial peer*) or an IP address and a remote telephone number (*remote dial peer* or *VoIP dial peer*). Each dial peer corresponds to a call leg.

**E&M**—ear and mouth (or recEive and transMit), a signaling technique for two-wire and four-wire interfaces between PSTN central offices or PBXs.

**FXO**—foreign exchange office, a type of VIC interface. The FXO VIC connects local calls to a PSTN central office or PBX over a standard RJ-11 modular telephone cable. This is the interface a standard telephone provides.

**FXS**—foreign exchange station, a type of VIC interface. The FXS VIC connects directly to a standard telephone, fax machine, or similar device over a standard RJ-11 modular telephone cable, and supplies ringing voltage, dial tone, and similar signals to it.

**IPX**—Internetwork Packet Exchange.

**ISDN**—Integrated Services Digital Network.

**local dial peer**—a software object that ties together a voice port and the telephone number of a device attached to the port (also called *POTS dial peer*).

**MTU**—maximum transmission unit.

**Multilink PPP**—Multilink Point-to-Point Protocol, a method of splitting, recombining, and sequencing datagrams across multiple logical data links.

**PBX**—private branch exchange, a private telephone switching system.

**POTS**—plain old telephone service.

**POTS dial peer**—a software object that ties together a voice port and the telephone number of a device attached to the port (also called *local dial peer*).

**PRI**—Primary Rate Interface.

**PSTN**—public switched telephone network.

**PVC**—permanent virtual circuit.

**QoS**—quality of service, a measure of the level of performance needed for a particular application, such as a voice over IP connection.

**remote dial peer**—a software object that ties together an IP address and a telephone number at a remote site reached over the IP network (also called *VoIP dial peer*).

**RSVP**—resource reservation protocol, a network protocol that enables routers to reserve the bandwidth necessary for reliable performance.

**RTP**—real-time transport protocol (RTP), a network protocol used to carry packetized audio and video traffic over an IP network.

**session target**—a remote IP or DNS address specified as one end of a voice connection.

**tag**—a positive integer in the range 1 to $2^{31}$ -1 (2147483647) used to identify a dial peer.

**UDP**—User Datagram Protocol.

**VIC**—voice interface card. VICs install in a slot in a voice network module, and provide the connection to the telephone equipment or network. There are three types: FXS, FXO, and E&M interface. Each VIC provides two ports of the same type.

**voice network module**—a network module that installs in a slot in a Cisco 3600 series router, converts telephone voice signals into a form that can be transmitted over an IP network, and provides one or two slots for voice interface cards.

**VoIP**—voice over IP, a feature that carries voice traffic, such as telephone calls and faxes, over an IP network, simultaneously with data traffic.

**VoIP dial peer**—a software object that ties together an IP address and a telephone number at a remote site reached over the IP network (also called *remote dial peer*).

# Cisco Connection Online

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- WWW: http://www.cisco.com
- WWW: http://www-europe.cisco.com
- WWW: http://www-china.cisco.com
- Telnet: cco.cisco.com
- Modem: From North America, 408 526-8070; from Europe, 33 1 64 46 40 82. Use the following terminal settings: VT100 emulation; databits: 8; parity: none; stop bits: 1; and connection rates up to 28.8 kbps.

For a copy of CCO's Frequently Asked Questions (FAQ), contact cco-help@cisco.com. For additional information, contact cco-team@cisco.com.

**Note** If you are a network administrator and need personal technical assistance with a Cisco product that is under warranty or covered by a maintenance contract, contact Cisco's Technical Assistance Center (TAC) at 800 553-2447, 408 526-7209, or tac@cisco.com. To obtain general information about Cisco Systems, Cisco products, or upgrades, contact 800 553-6387, 408 526-7208, or cs-rep@cisco.com.

Use this document in conjunction with your router installation and configuration guide, the *Regulatory Compliance and Safety Information* document for your router, the *Voice Network Module and Voice Interface Card Configuration Note*, the *Voice over IP Software Configuration Guide*, and the Cisco IOS configuration guides and command references.



## Anexo 3: Configuración de los routers Cisco para que soporten VoIP

Los siguientes tres archivos detallan los comandos necesarios para que los routers Cisco soporten la transmisión de voz sobre IP. Los dos primeros archivos detallan las herramientas QoS disponibles en los routers Cisco y el ajuste fino de la tarjetas de módulo de voz. El último archivo detalla todos los comandos que se pueden utilizar para la configuración de VoIP, (tanto en el router como en los puertos de voz).



# Configuring Voice over IP for the Cisco 3600 Series

This chapter explains how to configure Voice over IP (VoIP) on Cisco 3600 series routers and contains the following sections:

- Introduction
- List of Terms
- Prerequisites Tasks
- How Voice over IP Handles a Typical Telephone Call
- Configuration Tasks
- Configure IP Networks for Real-Time Voice Traffic
  - Configure RSVP for Voice
  - Configure Multilink PPP with Interleaving
  - Configure RTP Header Compression
  - Configure Custom Queuing
  - Configure Weighted Fair Queuing
- Configure Frame Relay for Voice over IP
- Configure Number Expansion
  - Create a Number Expansion Table
  - Configure Number Expansion
- Configure Dial Peers
  - Create a Peer Configuration Table
  - Configure POTS Peers
  - Configure VoIP Peers
- Configure Voice Ports
  - Configuring FXO or FXS Voice Ports
  - Fine-Tune FXO and FXS Voice Ports
  - Configure E&M Voice Ports
  - Fine-Tune E&M Voice Ports

- Optimize Dial Peer and Network Interface Configurations
  - Configure IP Precedence for Dial Peers
  - Configure RSVP for Dial Peers
  - Configure CODEC and VAD for Dial Peers
- Configure Voice over IP for Microsoft NetMeeting

All of these tasks are described in the following sections.

# Introduction

Voice over IP (VoIP) enables a Cisco 3600 series router to carry voice traffic (for example, telephone calls and faxes) over an IP network. Voice over IP is primarily a software feature; however, to use this feature on a Cisco 3600 series router, you must install a Voice Network Module (VNM). The VNM can hold either 2 or 4 Voice Interface Cards (VICs). Each VIC is specific to a particular signaling type associated with a voice port; therefore, VICs determine the type of signaling for the voice ports on that particular VNM. For more information about the physical characteristics of the VNM, as well as installing or configuring a VNM in your Cisco 3600 series router, refer to the *Voice Network Module and Voice Interface Card Configuration Note* that came with your VNM.

Voice over IP offers the following benefits:

- Toll bypass
- Remote PBX presence over WANs
- Unified voice/data trunking
- POTS-Internet telephony gateways

# List of Terms

**ACOM**—Term used in G.165, "General Characteristics of International Telephone Connections and International Telephone Circuits: Echo Cancellers." ACOM is the combined loss achieved by the echo canceller, which is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.

**Call leg**—A logical connection between the router and either a telephony endpoint over a bearer channel or another endpoint using a session protocol.

**Channel Associated Signaling (CAS)**—A form of signaling used on a T1 line. With CAS, a signaling element is dedicated to each channel in the T1 frame. This type of signaling is sometimes called Robbed Bit Signaling (RBS) because a bit is taken out (or robbed) from the user's data stream to provide signaling information to and from the switch.

**CIR**—Committed Information Rate. The average rate of information transfer a subscriber (for example, the network administrator) has stipulated for a Frame Relay PVC.

**CODEC**—Coder-decoder compression scheme or technique. In Voice over IP, it specifies the voice coder rate of speech for a dial peer.

**Dial peer**—An addressable call endpoint. In Voice over IP, there are two kinds of dial peers: POTS and VoIP.

**DS0**—A 64K channel on an E1 or T1 WAN interface.

**DTMF**—Dual tone multifrequency. Use of two simultaneous voice-band tones for dial (such as touch tone).

**E&M**—Stands for recEive and transMit (or Ear and Mouth). E&M is a trunking arrangement generally used for two-way switch-to-switch or switch-to-network connections. Cisco's E&M interface is an RJ-48 connector that allows connections to PBX trunk lines (tie lines).

**FIFO**—First-in, first-out. In data communication, FIFO refers to a buffering scheme where the first byte of data entering the buffer is the first byte retrieved by the CPU. In telephony, FIFO refers to a queuing scheme where the first calls received are the first calls processed.

**FXO**—Foreign Exchange Office. An FXO interface connects to the PSTN's central office and is the interface offered on a standard telephone. Cisco's FXO interface is an RJ-11 connector that allows an analog connection to be directed at the PSTN's central office. This interface is of value for off-premise extension applications.

**FXS**—Foreign Exchange Station. An FXS interface connects directly to a standard telephone and supplies ring, voltage, and dial tone. Cisco's FXS interface is an RJ-11 connector that allows connections to basic telephone service equipment, keysets, and PBXs.

**Multilink PPP**—Multilink Point-to-Point Protocol. This protocol is a method of splitting, recombining, and sequencing datagrams across multiple logical data links.

**PBX**—Private Branch Exchange. Privately-owned central switching office.

**PLAR**—Private Line Auto Ringdown. This type of service results in a call attempt to some particular remote endpoint when the local extension is taken off-key.

**POTS**—Plain Old Telephone Service. Basic telephone service supplying standard single line telephones, telephone lines, and access to the public switched telephone network.

**POTS dial peer**—Dial peer connected via a traditional telephony network. POTS peers point to a particular voice port on a voice network device.

**PSTN**—Public Switched Telephone Network. PSTN refers to the local telephone company.

**PVC**—Permanent Virtual Circuit.

**QoS**—Quality of Service, which refers to the measure of service quality provided to the user.

**RSVP**—Resource Reservation Protocol. This protocol supports the reservation of resources across an IP network.

**Trunk**—Service that allows quasi-transparent connections between two PBXes, a PBX and a local extension, or some other combination of telephony interfaces to be permanently conferenced together by the session application and signaling passed transparently through the IP network.

**VoIP dial peer**—Dial peer connected via a packet network; in the case of Voice over IP, this is an IP network. VoIP peers point to specific VoIP devices.

# Prerequisites Tasks

Before you can configure your Cisco 3600 series router to use Voice over IP, you must first:

- Establish a working IP network. For more information about configuring IP, refer to the "IP Overview," "Configuring IP Addressing," and "Configuring IP Services" chapters in the *Network Protocols Configuration Guide, Part 1*.
- Install the one-slot or two-slot (NM-1V/NM-2V) voice network module into the appropriate bay of your Cisco router. For more information about the physical characteristics of the voice network module, or how to install it, refer to the installation documentation, *Voice Network Module and Voice Interface Card Configuration Note*, that came with your voice network module.
- Complete your company's dial plan.

- Establish a working telephony network based on your company's dial plan.
- Integrate your dial plan and telephony network into your existing IP network topology. Merging your IP and telephony networks depends on your particular IP and telephony network topology. In general, we recommend the following suggestions:
  - Use canonical numbers wherever possible. It is important to avoid situations where numbering systems are significantly different on different routers or access servers in your network.
  - Make routing and/or dialing transparent to the user—for example, avoid secondary dial tones from secondary switches, where possible.
  - Contact your PBX vendor for instructions about how to reconfigure the appropriate PBX interfaces.

After you have analyzed your dial plan and decided how to integrate it into your existing IP network, you are ready to configure your network devices to support Voice over IP.

# How Voice over IP Handles a Typical Telephone Call

Before configuring Voice over IP on your Cisco 3600 series router, it helps to understand what happens at an application level when you place a call using Voice over IP. The general flow of a two-party voice call using Voice over IP is as follows:

1. The user picks up the handset; this signals an off-hook condition to the signaling application part of Voice over IP in the Cisco 3600 series router.
2. The session application part of Voice over IP issues a dial tone and waits for the user to dial a telephone number.
3. The user dials the telephone number; those numbers are accumulated and stored by the session application.
4. After enough digits are accumulated to match a configured destination pattern, the telephone number is mapped to an IP host via the dial plan mapper. The IP host has a direct connection to either the destination telephone number or a PBX that is responsible for completing the call to the configured destination pattern.
5. The session application then runs the H.323 session protocol to establish a transmission and a reception channel for each direction over the IP network. If the call is being handled by a PBX, the PBX forwards the call to the destination telephone. If RSVP has been configured, the RSVP reservations are put into effect to achieve the desired quality of service over the IP network.
6. The CODECs are enabled for both ends of the connection and the conversation proceeds using RTP/UDP/IP as the protocol stack.
7. Any call-progress indications (or other signals that can be carried in-band) are cut through the voice path as soon as end-to-end audio channel is established. Signaling that can be detected by the voice ports (for example, in-band DTMF digits after the call setup is complete) is also trapped by the session application at either end of the connection and carried over the IP network encapsulated in RTCP using the RTCP APP extension mechanism.
8. When either end of the call hangs up, the RSVP reservations are torn down (if RSVP is used) and the session ends. Each end becomes idle, waiting for the next off-hook condition to trigger another call setup.

# Configuration Tasks

To configure Voice over IP on the Cisco 3600 series, you need to perform the following steps:

**Step 1** Configure your IP network to support real-time voice traffic. Fine-tuning your network to adequately support VoIP involves a series of protocols and features geared toward Quality of Service (QoS). To configure your IP network for real-time voice traffic, you need to take into consideration the entire scope of your network, then select and configure the appropriate QoS tool or tools:

- RSVP
- Multilink PPP with Interleaving
- RTP Header Compression
- Custom Queuing
- Weighted Fair Queuing

Refer to "Configure IP Networks for Real-Time Voice Traffic" section for information about how to select and configure the appropriate QoS tools to optimize voice traffic on your network.

**Step 2** (Optional) If you plan to run Voice over IP over Frame Relay, you need to take certain factors into consideration when configuring Voice over IP for it to run smoothly over Frame Relay. For example, a public Frame Relay cloud provides no guarantees for QoS. Refer to the "Configure Frame Relay for Voice over IP" section for information about deploying Voice over IP over Frame Relay.

**Step 3** Use the **num-exp** command to configure number expansion if your telephone network is configured so that you can reach a destination by dialing only a portion (an extension number) of the full E.164 telephone number. Refer to the "Configure Number Expansion" section for information about number expansion.

**Step 4** Use the **dial-peer voice** command to define dial peers and switch to the dial-peer configuration mode. Each dial peer defines the characteristics associated with a call leg. A call leg is a discrete segment of a call connection that lies between two points in the connection. An end-to-end call is comprised of four call legs, two from the perspective of the source access server, and two from the perspective of the destination access server. Dial peers are used to apply attributes to call legs and to identify call origin and destination. There are two different kinds of dial peers:

- POTS—dial peer describing the characteristics of a traditional telephony network connection. POTS peers point to a particular voice port on a voice network device. To minimally configure a POTS dial peer, you need to configure the following two characteristics: associated telephone number and logical interface. Use the **destination-pattern** command to associate a telephone number with a POTS peer. Use the **port** command to associate a specific logical interface with a POTS peers. In addition, you can specify direct inward dialing for a POTS peer by using the **direct-inward-dial** command.
- VoIP—dial peer describing the characteristics of a packet network connection; in the case of Voice over IP, this is an IP network. VoIP peers point to specific VoIP devices. To minimally configure a VoIP peer, you need to configure the following two characteristics: associated destination telephone number and a destination IP address. Use the **destination-pattern** command to define the destination telephone number associated with a VoIP peer. Use the **session-target** command to specify a destination IP address for a VoIP peer.

In addition, you can use VoIP peers to define characteristics such as IP precedence, additional QoS parameters (when RSVP is configured), CODEC, and VAD. Use the **ip precedence** command to define IP precedence. If you have configured RSVP, use either the **req-qos** or **acc-qos** command to configure QoS parameters. Use the **codec** command to configure specific voice coder rates. Use the **vad** command to disable voice activation detection and the transmission of silence packets.

Refer to the "Configure Dial Peers" section and the "Optimize Dial Peer and Network Interface Configurations" section for additional information about configuring dial peers and dial-peer characteristics.

**Step 5** You need to configure your Cisco 3600 series router to support voice ports. In general, voice-port commands define the characteristics associated with a particular voice-port signaling type. voice ports on the Cisco 3600 series support three basic voice signaling types:

- FXO—Foreign Exchange Office interface
- FXS—Foreign Exchange Station interface
- E&M—"RecEive and TransMit" interface or the "Ear and Mouth" interface

Under most circumstances, the default voice-port command values are adequate to configure FXO and FXS ports to transport voice data over your existing IP network. Because of the inherent complexities involved with PBX networks, E&M ports might need specific voice-port values configured, depending on the specifications of the devices in your telephony network. For information about configuring voice ports, refer to the "Configuring Voice Ports" section.

# Configure IP Networks for Real-Time Voice Traffic

You need to have a well-engineered network end-to-end when running delay-sensitive applications such as VoIP. Fine-tuning your network to adequately support VoIP involves a series of protocols and features geared toward Quality of Service (QoS). It is beyond the scope of this document to explain the specific details relating to wide-scale QoS deployment. Cisco IOS software provides many tools for enabling QoS on your backbone, such as Random Early Detection (RED), Weighted Random Early Detection (WRED), Fancy Queuing (meaning custom, priority, or weighted fair queuing), and IP Precedence. To configure your IP network for real-time voice traffic, you need to take into consideration the entire scope of your network, then select the appropriate QoS tool or tools.

The important thing to remember is that QoS must be configured throughout your network—not just on the Cisco 3600 series devices running VoIP—to improve voice network performance. Not all QoS techniques are appropriate for all network routers. Edge routers and backbone routers in your network do not necessarily perform the same operations; the QoS tasks they perform might differ as well. To configure your IP network for real-time voice traffic, you need to take into consideration the functions of both edge and backbone routers in your network, then select the appropriate QoS tool or tools.

In general, edge routers perform the following QoS functions:

- Packet classification
- Admission control
- Bandwidth management
- Queuing

In general, backbone routers perform the following QoS functions:

- High-speed switching and transport
- Congestion management
- Queue management

Scalable QoS solutions require cooperative edge and backbone functions.

Although not mandatory, some QoS tools have been identified as being valuable in fine-tuning your network to support real-time voice traffic. To configure your IP network for QoS using these tools, perform one or more of the following tasks:

- Configure RSVP for Voice
- Configure Multilink PPP with Interleaving
- Configure RTP Header Compression
- Configure Custom Queuing
- Configure Weighted Fair Queuing

Each of these tasks is discussed in the following sections.

# Configure RSVP for Voice

RSVP allows end systems to request a particular Quality of Service (QoS) from the network. Real-time voice traffic requires network consistency. Without consistent QoS, real-time traffic can experience jitter, insufficient bandwidth, delay variations, or information loss. RSVP works in conjunction with current queuing mechanisms. It is up to the interface queuing mechanism (such as weighted fair queuing or weighted random early detection) to implement the reservation.

RSVP can be equated to a dynamic access list for packet flows.

You should configure RSVP to ensure QoS if the following conditions exist in your network:

- Small scale voice network implementation
- Slow links
- Links with high utilization
- Links less than 2 Mbps
- Need for the best possible voice quality

## Enable RSVP

To minimally configure RSVP for voice traffic, you must enable RSVP on each interface where priority needs to be set.

By default, RSVP is disabled so that it is backward compatible with systems that do not implement RSVP. To enable RSVP on an interface, use the following command in interface configuration mode:

| Command | Purpose |
|---|---|
| **ip rsvp bandwidth** [*interface-kbps*] [*single-flow-kbps*] | Enable RSVP for IP on an interface. |

This command starts RSVP and sets the bandwidth and single-flow limits. The default maximum bandwidth is up to 75 percent of the bandwidth available on the interface. By default, the amount reservable by a flow can be up to the entire reservable bandwidth.

On subinterfaces, this applies the more restrictive of the available bandwidths of the physical interface and the subinterface.

Reservations on individual circuits that do not exceed the single flow limit normally succeed. If, however, reservations have been made on other circuits adding up to the line speed, and a reservation is made on a subinterface which itself has enough remaining bandwidth, it will still be refused because the physical interface lacks supporting bandwidth.

Cisco AS5300s running VoIP and configured for RSVP request allocations per the following formula:

```
bps=packet_size+ip/udp/rtp header size * 50 per second
```

For G.729, the allocation works out to be 24,000 bps. For G.711, the allocation is 80,000 bps.

For more information about configuring RSVP, refer to the "Configuring RSVP" chapter of the Cisco IOS Release 11.3 *Network Protocols Configuration Guide, Part 1*.

### RSVP Configuration Example

The following example enables RSVP and sets the maximum bandwidth to 100 kbps and the maximum bandwidth per single request to 32 kbps (the example presumes that both VoIP dial peers have been configured):

```
interface serial 1/0/0
 ip rsvp bandwidth 100 32
 fair-queue
 end
!
dial-peer voice 1211 voip
 req-qos controlled-load
!
dial-peer voice 1212 voip
 req-qos controlled-load
```

## Configure Multilink PPP with Interleaving

Multi-class Multilink PPP Interleaving allows large packets to be multilink-encapsulated and fragmented into smaller packets to satisfy the delay requirements of real-time voice traffic; small real-time packets, which are not multilink-encapsulated, are transmitted between fragments of the large packets. The interleaving feature also provides a special transmit queue for the smaller, delay-sensitive packets, enabling them to be transmitted earlier than other flows. Interleaving provides the delay bounds for delay-sensitive voice packets on a slow link that is used for other best-effort traffic.

---

**Note** Interleaving applies only to interfaces that can configure a multilink bundle interface. These include virtual templates, dialer interfaces, and Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) or Primary Rate Interface (PRI) interfaces.

---

In general, Multilink PPP with interleaving is used in conjunction with weighted fair queuing and RSVP or IP Precedence to ensure voice packet delivery. Use Multilink PPP with interleaving and weighted fair queuing to define how data will be managed; use RSVP or IP Precedence to give priority to voice packets.

You should configure Multilink PPP if the following conditions exist in your network:

- Point-to-point connection using PPP Encapsulation
- Slow links

**Note** Multilink PPP should not be used on links greater than 2 Mbps.

Multilink PPP support for interleaving can be configured on virtual templates, dialer interfaces, and ISDN BRI or PRI interfaces. To configure interleaving, you need to complete the following tasks:

- Configure the dialer interface or virtual template, as defined in the relevant chapters of the Cisco IOS Release 11.3 *Dial Solutions Configuration Guide*.
- Configure Multilink PPP and interleaving on the interface or template.

To configure Multilink PPP and interleaving on a configured and operational interface or virtual interface template, use the following commands in interface mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **ppp multilink** | Enable Multilink PPP. |
| 2 | **ppp multilink interleave** | Enable real-time packet interleaving. |
| 3 | **ppp multilink fragment-delay** *milliseconds* | Optionally, configure a maximum fragment delay. |
| 4 | **ip rtp reserve** *lowest-UDP-port range-of-ports* [*maximum-bandwidth*] | Reserve a special queue for real-time packet flows to specified destination User Datagram Protocol (UDP) ports, allowing real-time traffic to have higher priority than other flows. This is only applicable if you have not configured RSVP. |

**Note** The **ip rtp reserve** command can be used instead of configuring RSVP. If you configure RSVP, this command is not required.

For more information about Multilink PPP, refer to the "Configuring Media-Independent PPP and Multilink PPP" chapter in the Cisco IOS Release 11.3 *Dial Solutions Configuration Guide*.

## Multilink PPP Configuration Example

The following example defines a virtual interface template that enables Multilink PPP with interleaving and a maximum real-time traffic delay of 20 milliseconds, and then applies that virtual template to the Multilink PPP bundle:

```
interface virtual-template 1
 ppp multilink
 encapsulated ppp
 ppp multilink interleave
 ppp multilink fragment-delay 20
 ip rtp reserve 16384 100 64

multilink virtual-template 1
```

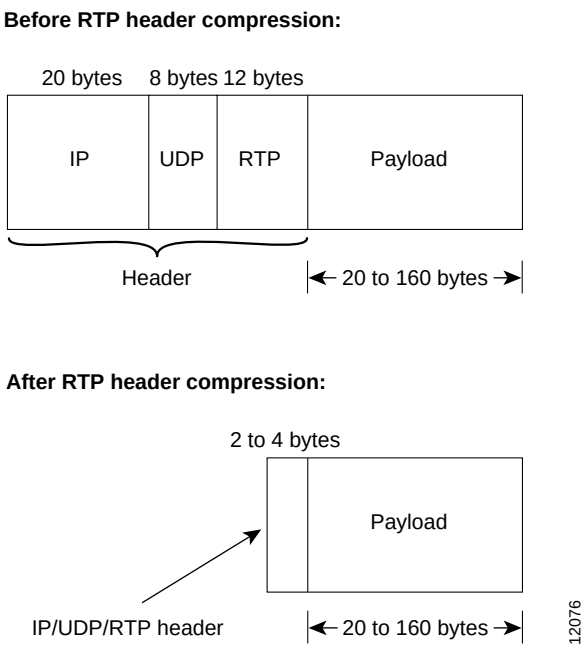
# Configure RTP Header Compression

Real-Time Transport Protocol (RTP) is used for carrying packetized audio traffic over an IP network. RTP header compression compresses the IP/UDP/RTP header in an RTP data packet from 40 bytes to approximately 2 to 4 bytes (most of the time), as shown in Figure 2-1.

This compression feature is beneficial if you are running Voice over IP over slow links. Enabling compression on both ends of a low-bandwidth serial link can greatly reduce the network overhead if there is a lot of RTP traffic on that slow link.

Typically, an RTP packet has a payload of approximately 20 to 160 bytes for audio applications that use compressed payloads. RTP header compression is especially beneficial when the RTP payload size is small (for example, compressed audio payloads between 20 and 50 bytes).

**Figure 2-1 RTP Header Compression**




You should configure RTP header compression if the following conditions exist in your network:

- Slow links
- Need to save bandwidth

**Note** RTP header compression should not be used on links greater than 2 Mbps.

Perform the following tasks to configure RTP header compression for Voice over IP. The first task is required; the second task is optional.

- Enable RTP Header Compression on a Serial Interface
- Change the Number of Header Compression Connections

### Enable RTP Header Compression on a Serial Interface

To use RTP header compression, you need to enable compression on both ends of a serial connection. To enable RTP header compression, use the following command in interface configuration mode:

| Command | Purpose |
|---|---|
| **ip rtp header-compression** [**passive**] | Enable RTP header compression. |

If you include the **passive** keyword, the software compresses outgoing RTP packets only if incoming RTP packets on the same interface are compressed. If you use the command without the **passive** keyword, the software compresses all RTP traffic.

### Change the Number of Header Compression Connections

By default, the software supports a total of 16 RTP header compression connections on an interface. To specify a different number of RTP header compression connections, use the following command in interface configuration mode:

| Command | Purpose |
|---|---|
| **ip rtp compression connections** *number* | Specify the total number of RTP header compression connections supported on an interface. |

### RTP Header Compression Configuration Example

The following example enables RTP header compression for a serial interface:

```
interface 0
 ip rtp header-compression
 encapsulation ppp
 ip rtp compression-connections 25
```

For more information about RTP header compression, see the "Configuring IP Multicast Routing" chapter of the Cisco IOS Release 11.3 *Network Protocols Configuration Guide, Part 1*.

## Configure Custom Queuing

Some QoS features, such as IP RTP reserve and custom queuing, are based on the transport protocol and the associated port number. Real-time voice traffic is carried on UDP ports ranging from 16384 to 16624. This number is derived from the following formula:

```
16384 = 4(number of voice ports in the Cisco 3600 series router)
```

Custom Queuing and other methods for identifying high priority streams should be configured for these port ranges. For more information about custom queuing, refer to the "Managing System Performance" chapter in the Cisco IOS Release 11.3 *Configuration Fundamentals Configuration Guide*.

## Configure Weighted Fair Queuing

Weighted fair queuing ensures that queues do not starve for bandwidth and that traffic gets predictable service. Low-volume traffic streams receive preferential service; high-volume traffic streams share the remaining capacity, obtaining equal or proportional bandwidth.

In general, weighted fair queuing is used in conjunction with Multilink PPP with interleaving and RSVP or IP Precedence to ensure that voice packet delivery. Use weighted fair queuing with Multilink PPP to define how data will be managed; use RSVP or IP Precedence to give priority to voice packets. For more information about weighted fair queuing, refer to the "Managing System Performance" chapter in the Cisco IOS Release 11.3 *Configuration Fundamentals Configuration Guide*.

# Configure Frame Relay for Voice over IP

You need to take certain factors into consideration when configuring Voice over IP for it to run smoothly over Frame Relay. A public Frame Relay cloud provides no guarantees for QoS. For real-time traffic to be transmitted in a timely manner, the data rate must not exceed the committed information rate (CIR) or there is the possibility that packets will be dropped. In addition, Frame Relay traffic shaping and RSVP are mutually exclusive. This is particularly important to remember if multiple DLCIs are carried on a single interface.

For Frame Relay links with slow output rates (less than or equal to 64 kbps), where data and voice are being transmitted over the same PVC, we recommend the following solutions:

- Separate DLCIs for voice and data—By providing a separate subinterface for voice and data, you can use the appropriate QoS tool per line. For example, each DLCI would use 32 kbps of a 64 kbps line.
  - Apply adaptive traffic shaping to both DLCIs.
  - Use RSVP or IP Precedence to prioritize voice traffic.
  - Use compressed RTP to minimize voice packet size.
  - Use weighted fair queuing to manage voice traffic.
- Lower MTU size—Voice packets are generally small. By lowering the MTU size (for example, to 300 bytes), large data packets can be broken up into smaller data packets that can more easily be interwoven with voice packets.

**Note** Lowering the MTU size affects data throughput speed.

- CIR equal to line rate—Make sure that the data rate does not exceed the CIR. This is accomplished through generic traffic shaping.
  - Use RSVP or IP Precedence to prioritize voice traffic.
  - Use compressed RTP to minimize voice packet header size.
- Traffic shaping—Use adaptive traffic shaping to throttle back the output rate based on the BECN. If the feedback from the switch is ignored, packets (both data and voice) might be discarded. Because the Frame Relay switch does not distinguish between voice and data packets, voice packets could be discarded, which would result in a deterioration of voice quality.
  - Use RSVP, compressed RTP, reduced MTU size, and adaptive traffic shaping based on BECN to hold data rate to CIR.
  - Use generic traffic shaping to obtain a low interpacket wait time. For example, set Bc to 4000 to obtain an interpacket wait of 125 ms.

In Cisco IOS Release 11.3, Frame Relay Traffic Shaping is not compatible with RSVP. We suggest one of the following workarounds:

- Provision the Frame Relay PVC to have the CIR equal to the port speed.
- Use Generic Traffic Shaping with RSVP.

## Frame Relay for Voice over IP Configuration Example

For Frame Relay, it is customary to configure a main interface and several subinterfaces, one subinterface per PVC. The following example configures a Frame Relay main interface and a subinterface so that voice and data traffic can be successfully transported:

```
interface Serial0/0
 mtu 300
 no ip address
 encapsulation frame-relay
 no ip route-cache
 no ip mroute-cache
 fair-queue 64 256 1000
 frame-relay ip rtp header-compression

interface Serial0/0.1 point-to-point
 mtu 300
 ip address 40.0.0.7 255.0.0.0
 ip rsvp bandwidth 48 48
 no ip route-cache
 no ip mroute-cache
 bandwidth 64
 traffic-shape rate 32000 4000 4000
 frame-relay interface-dlci 16
 frame-relay ip rtp header-compression
```

In this configuration example, the main interface has been configured as follows:

- MTU size is 300 bytes.
- No IP address is associated with this serial interface. The IP address must be assigned for the subinterface.
- Encapsulation method is Frame Relay.
- Fair-queuing is enabled.
- IP RTP header compression is enabled.

The subinterface has been configured as follows:

- MTU size is inherited from the main interface.
- IP address for the subinterface is specified.
- Bandwidth is set to 64 kbps.
- RSVP is enabled to use the default value, which is 75 percent of the configured bandwidth.
- Generic traffic shaping is enabled with 32 kbps CIR where Bc=4000 bits and Be=4000 bits.
- Frame Relay DLCI number is specified.
- IP RTP header compression is enabled.

**Note** When traffic bursts over the CIR, output rate is held at the speed configured for the CIR (for example, traffic will not go beyond 32 kbps if CIR is set to 32 kbps).

For more information about Frame Relay, refer to the "Configuring Frame Relay" chapter in the Cisco IOS Release 11.3 *Wide-Area Networking Configuration Guide*.

# Configure Number Expansion

In most corporate environments, the telephone network is configured so that you can reach a destination by dialing only a portion (an extension number) of the full E.164 telephone number. Voice over IP can be configured to recognize extension numbers and expand them into their full E.164 dialed number by using two commands in tandem: **destination-pattern** and **num-exp**. Before you configure these two commands, it is helpful to map individual telephone extensions with their full E.164 dialed numbers. This can be done easily by creating a number expansion table.

## Create a Number Expansion Table

In Figure 2-2, a small company wants to use Voice over IP to integrate its telephony network with its existing IP network. The destination pattern (or expanded telephone number) associated with Access Server 1 (located to the left of the IP cloud) is (408) 526-xxxx, where xxxx identifies the individual dial peers by extension. The destination pattern (or expanded telephone number) associated with Access Server 2 (located to the right of the IP cloud) is (729) 422-xxxx.

**Figure 2-2 Sample Voice over IP Network**

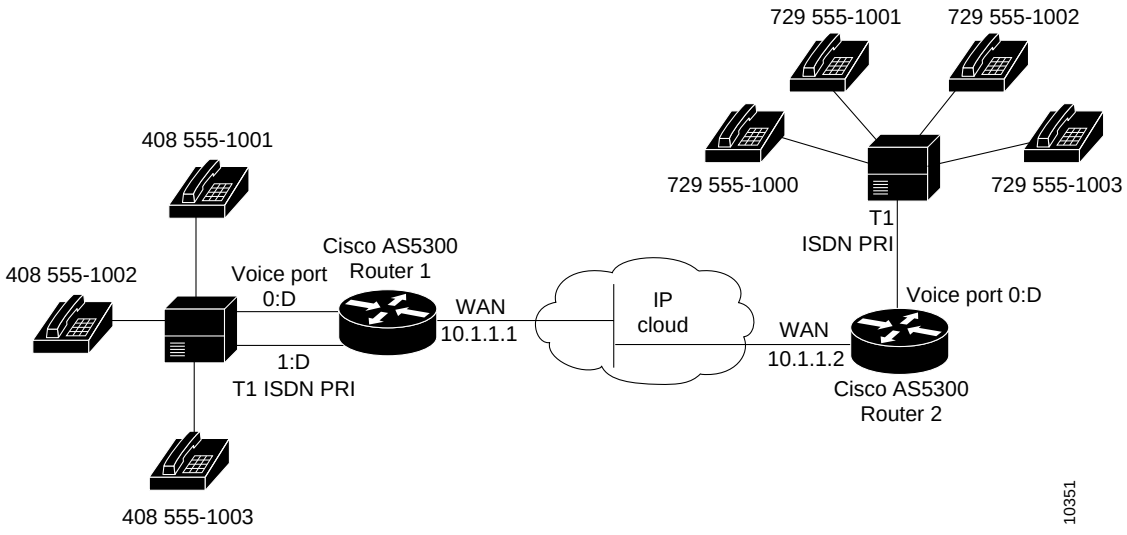



Table 2-1 shows the number expansion table for this scenario.

**Table 2-1 Sample Number Expansion Table**

| Extension | Destination Pattern | Num-Exp Command Entry |
|---|---|---|
| 5.... | 40852..... | num-exp 5.... 408525.... |
| 6.... | 40852..... | num-exp 6.... 408526.... |
| 7.... | 40852..... | num-exp 7.... 408527.... |
| 1... | 729422.... | num-exp 2.... 729422.... |

**Note** You can use the period symbol (.) to represent variables (such as extension numbers) in a telephone number.

The information included in this example needs to be configured on both Access Server 1 and Access Server 2.

## Configure Number Expansion

To define how to expand an extension number into a particular destination pattern, use the following command in global configuration mode:

| Command | Purpose |
|---|---|
| **num-exp** *extension-number extension-string* | Configure number expansion. |

You can verify the number expansion information by using the **show num-exp** command to verify that you have mapped the telephone numbers correctly.

After you have configured dial peers and assigned destination patterns to them, you can verify number expansion information by using the **show dialplan number** command to see how a telephone number maps to a dial peer.

# Configure Dial Peers

The key point to understanding how Voice over IP functions is to understand dial peers. Each dial peer defines the characteristics associated with a call leg, as shown in Figure 2-3 and Figure 2-4. A call leg is a discrete segment of a call connection that lies between two points in the connection. All of the call legs for a particular connection have the same connection ID.

There are two different kinds of dial peers:

- POTS—Dial peer describing the characteristics of a traditional telephony network connection. POTS peers point to a particular voice port on a voice network device.
- VoIP—Dial peer describing the characteristics of a packet network connection; in the case of Voice over IP, this is an IP network. VoIP peers point to specific VoIP devices.

An end-to-end call is comprised of four call legs, two from the perspective of the source access server as shown in Figure 2-3, and two from the perspective of the destination access server as shown in Figure 2-4. A dial peer is associated with each one of these call legs. Dial peers are used to apply attributes to call legs and to identify call origin and destination. Attributes applied to a call leg include QoS, CODEC, VAD, and fax rate.

**Figure 2-3 Dial Peer Call Legs from the Perspective of the Source Router**

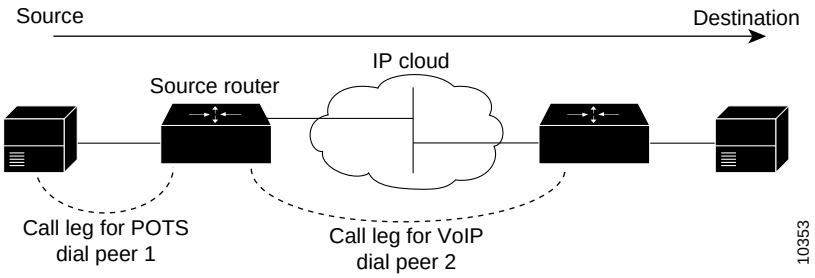
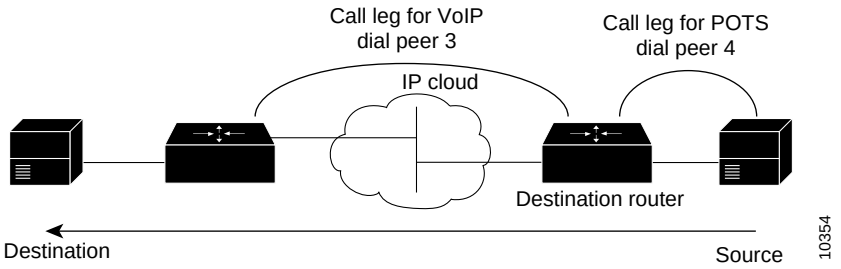



**Figure 2-4 Dial Peer Call Legs from the Perspective of the Destination Router**




## Inbound versus Outbound Dial Peers

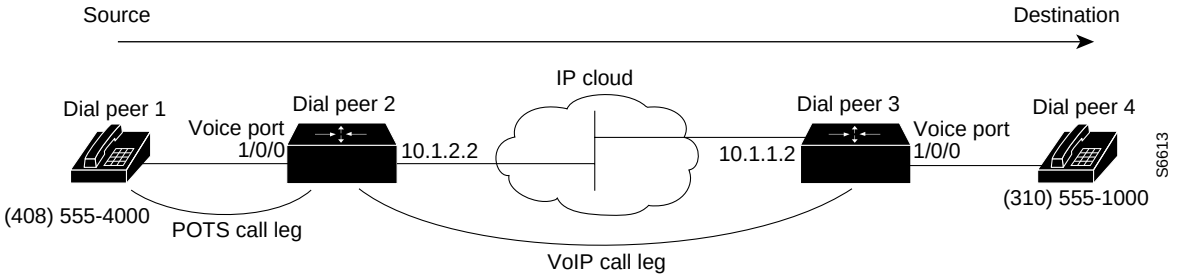
Dial peers are used for both inbound and outbound call legs. It is important to remember that these terms are defined from the *router's* perspective. An inbound call leg originates *outside* the router. An outbound call leg originates *from* the router.

For inbound call legs, a dial peer might be associated to the calling number or the port designation. Outbound call legs always have a dial peer associated with them. The destination pattern is used to identify the outbound dial peer. The call is associated with the outbound dial peer at setup time.

POTS peers associate a telephone number with a particular voice port so that incoming calls for that telephone number can be received and outgoing calls can be placed. VoIP peers point to specific devices (by associating destination telephone numbers with a specific IP address) so that incoming calls can be received and outgoing calls can be placed. Both POTS and VoIP peers are needed to establish Voice over IP connections.

Establishing communication using Voice over IP is similar to configuring an IP static route: you are establishing a specific voice connection between two defined endpoints. As shown in Figure 2-5, for outgoing calls (from the perspective of the POTS dial peer 1), the POTS dial peer establishes the source (via the originating telephone number or voice port) of the call. The VoIP dial peer establishes the destination by associating the destination telephone number with a specific IP address.

**Figure 2-5 Outgoing Calls from the Perspective of POTS Dial Peer 1**




To configure call connectivity between the source and destination as illustrated in Figure 2-5, enter the following commands on router 10.1.2.2:

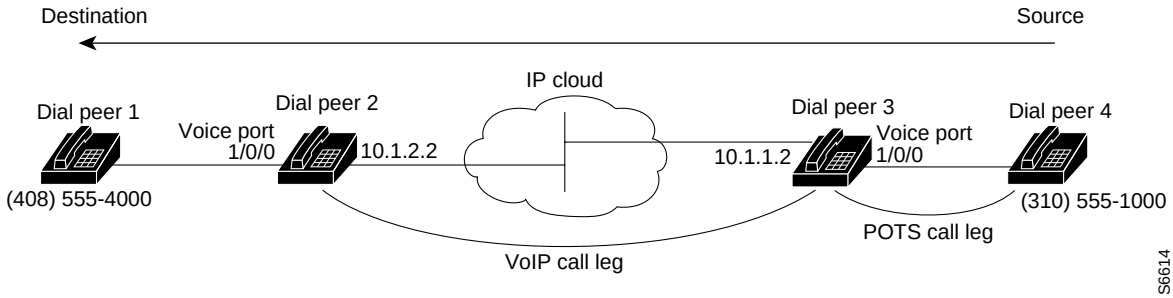
```
dial-peer voice 1 pots
 destination-pattern 1408526....
 port 1/0/0

dial-peer voice 2 voip
 destination-pattern 1310520....
 session target ipv4:10.1.1.2
```

In the previous configuration example, the last four digits in the VoIP dial peer's destination pattern were replaced with wildcards. This means that from access server 10.1.2.2, calling any number string that begins with the digits "1310520" will result in a connection to access server 10.1.1.2. This implies that access server 10.1.1.2 services all numbers beginning with those digits. From access server 10.1.1.2, calling any number string that begins with the digits "1408526" will result in a connection to access server 10.1.2.2. This implies that access server 10.1.2.2 services all numbers beginning with those digits. For more information about stripping and adding digits, see the "Outbound Dialing on POTS Peers" section.

Figure 2-6 shows how to complete the end-to-end call between dial peer 1 and dial peer 4.

**Figure 2-6 Outgoing Calls from the Perspective of POTS Dial Peer 2**




To complete the end-to-end call between dial peer 1 and dial peer 4 as illustrated in Figure 2-6, enter the following commands on router 10.1.1.2:

```
dial-peer voice 4 pots
 destination-pattern 1310520....
 port 1/0/0

dial-peer voice 3 voip
 destination-pattern 1408526....
 session target ipv4:10.1.2.2
```

## Create a Peer Configuration Table

There is specific data relative to each dial peer that needs to be identified before you can configure dial peers in Voice over IP. One way to do this is to create a peer configuration table.

Using the example in Figure 2-2, Router 1, with an IP address of 10.1.1.1, connects a small sales branch office to the main office through Router 2. There are three telephones in the sales branch office that need to be established as dial peers. Access Server 2, with an IP address of 10.1.1.2, is the primary gateway to the main office; as such, it needs to be connected to the company's PBX. There are four devices that need to be established as dial peers in the main office, all of which are basic telephones connected to the PBX. Figure 2-2 shows a diagram of this small voice network.

Table 2-2 shows the peer configuration table for the example illustrated in Figure 2-2.

**Table 2-2 Peer Configuration Table for Sample Voice Over IP Network**

| | | Commands | | | | | |
|---|---|---|---|---|---|---|---|
| **Dial Peer Tag** | **Ext** | **Dest-Pattern** | **Type** | **Voice Port** | **Session-Target** | **CODEC** | **QoS** |
| *Access Server 1* | | | | | | | |
| 1 | 6.... | +1408526.... | POTS | | | | |
| 10 | | +1729422.... | VoIP | | IPV4 10.1.1.2 | G.729 | Best Effort |
| *Access Server 2* | | | | | | | |
| 11 | | +1408526.... | VoIP | | IPV4 10.1.1.1 | G.729 | Best Effort |
| 4 | 2.... | +1729422.... | POTS | | | | |

## Configure POTS Peers

Once again, POTS peers enable incoming calls to be received by a particular telephony device. To configure a POTS peer, you need to uniquely identify the peer (by assigning it a unique tag number), define its telephone number(s), and associate it with a voice port through which calls will be established. Under most circumstances, the default values for the remaining dial-peer configuration commands will be sufficient to establish connections.

To enter the dial-peer configuration mode (and select POTS as the method of voice-related encapsulation), use the following command in global configuration mode:

| Command | Purpose |
|---|---|
| **dial-peer voice** *number* **pots** | Enter the dial-peer configuration mode to configure a POTS peer. |

The *number* value of the **dial-peer voice pots** command is a tag that uniquely identifies the dial peer. (This number has local significance only.)

To configure the identified POTS peer, use the following commands in dial-peer configuration mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **destination-pattern** *string* | Define the telephone number associated with this POTS dial peer. |
| 2 | **port** *controller number*:**D** | Associate this POTS dial peer with a specific logical dial interface. |

## Outbound Dialing on POTS Peers

When a router receives a voice call, it selects an outbound dial peer by comparing the called number (the full E.164 telephone number) in the call information with the number configured as the destination pattern for the POTS peer. The router then strips out the left-justified numbers corresponding to the destination pattern matching the called number. If you have configured a prefix, the prefix will be put in front of the remaining numbers, creating a dial string, which the router will then dial. If all numbers in the destination pattern are stripped-out, the user will receive (depending on the attached equipment) a dial tone.

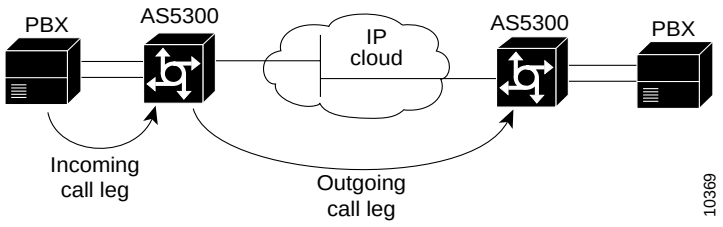
For example, suppose there is a voice call whose E.164 called number is 1(310) 767-2222. If you configure a destination-pattern of “1310767” and a prefix of “9,” the router will strip out “1310767” from the E.164 telephone number, leaving the extension number of “2222.” It will then append the prefix, “9,” to the front of the remaining numbers, so that the actual numbers dialed is “9, 2222.” The comma in this example means that the router will pause for one second between dialing the “9” and the “2” to allow for a secondary dial tone.

For additional POTS dial-peer configuration options, refer to the “Voice over IP Commands for the Cisco 3600 Series” section.

## Direct Inward Dial for POTS Peers

Direct inward dial (DID) is used to determine how the called number is treated for incoming POTS call legs. As shown in Figure 2-7, incoming means from the perspective of the router. In this case, it is the call leg coming into the access server to be forwarded through to the appropriate destination pattern.

**Figure 2-7 Incoming and Outgoing POTS Call Legs**




Unless otherwise configured, when a call arrives on the access server, the server presents a dial tone to the caller and collects digits until it can identify the destination dial peer. After the dial peer has been identified, the call is forwarded through the next call leg to the destination.

There are cases where it might be necessary for the server to use the called-number (DNIS) to find a dial peer for the outgoing call leg—for example, if the switch connecting the call to the server has already collected the digits. DID enables the server to match the called-number with a dial peer and then directly place the outbound call. With DID, the server does not present a dial tone to the caller and does not collect digits; it forwards the call directly to the configured destination.

To use DID and incoming called-number, a dial peer must be associated with the incoming call leg. Before doing this, it helps if you understand the logic behind the algorithm used to associate the incoming call leg with the dial peer.

The algorithm used to associate incoming call legs with dial peers uses three inputs (which are derived from signaling and interface information associated with the call) and four defined dial peer elements. The three signaling inputs are:

- Called-number (DNIS)—Set of numbers representing the destination, which is derived from the ISDN setup message or CAS DNIS.
- Calling-number (ANI)—Set of numbers representing the origin, which is derived from the ISDN setup message or CAS DNIS.
- Voice port—The voice port carrying the call.

The four defined dial peer elements are:

- Destination pattern—A pattern representing the phone numbers to which the peer can connect.
- Answer address—A pattern representing the phone numbers from which the peer can connect.
- Incoming called-number—A pattern representing the phone numbers that associate an incoming call leg to a peer based on the called-number or DNIS.
- Port—The port through which calls to this peer are placed.

Using the elements, the algorithm is as follows:

```
For all peers where call type (VoIP versus POTS) match dial peer type:
if the type is matched, associate the called number with the incoming called-number
 else if the type is matched, associate calling-number with answer-address
 else if the type is matched, associate calling-number with destination-pattern
 else if the type is matched, associate voice port to port
```

This algorithm shows that if a value is not configured for answer-address, the origin address is used because, in most cases, the origin address and answer-address are the same.

To configure DID for a particular POTS dial peer, use the following commands, initially in global configuration mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **dial-peer voice** *number* **pots** | Enter the dial-peer configuration mode to configure a POTS peer. |
| 2 | **direct-inward-dial** | Specify direct inward dial for this POTS peer. |

**Note** Direct inward dial is configured for the calling POTS dial peer.

For additional POTS dial-peer configuration options, refer to the “Voice over IP Commands for the Cisco 3600 Series” chapter.

# Configure VoIP Peers

Once again, VoIP peers enable outgoing calls to be made from a particular telephony device. To configure a VoIP peer, you need to uniquely identify the peer (by assigning it a unique tag number), define its destination telephone number and destination IP address. As with POTS peers, under most circumstances, the default values for the remaining dial-peer configuration commands will be adequate to establish connections.

To enter the dial-peer configuration mode (and select VoIP as the method of voice-related encapsulation), use the following command in global configuration mode:

| Command | Purpose |
|---|---|
| **dial-peer voice** *number* **voip** | Enter the dial-peer configuration mode to configure a VoIP peer. |

The *number* value of the **dial-peer voice voip** command is a tag that uniquely identifies the dial peer.

To configure the identified VoIP peer, use the following commands in dial-peer configuration mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **destination-pattern** *string* | Define the destination telephone number associated with this VoIP dial peer. |
| 2 | **session-target** {**ipv4:***destination-address* \| **dns:***host-name*} | Specify a destination IP address for this dial peer. |

For additional VoIP dial-peer configuration options, refer to the "Voice over IP Commands for the Cisco 3600 Series" chapter. For examples of how to configure dial peers, refer to the chapter, "Voice over IP Configuration Examples for the Cisco 3600 Series."

## Verify

You can check the validity of your dial-peer configuration by performing the following tasks:

- If you have relatively few dial peers configured, you can use the **show dial-peer voice** command to verify that the data configured is correct. Use this command to display a specific dial peer or to display all configured dial peers.
- Use the **show dialplan number** command to show the dial peer to which a particular number (destination pattern) resolves.

## Tips

If you are having trouble connecting a call and you suspect the problem is associated with dial-peer configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the chapter "Configuring IP" in the Cisco IOS 11.3 *Network Protocols Configuration Guide, Part 1*.
- Use the **show dial-peer voice** command to verify that the operational status of the dial peer is up.
- Use the **show dialplan number** command on the local and remote routers to verify that the data is configured correctly on both.
- If you have configured number expansion, use the **show num-exp** command to check that the partial number on the local router maps to the correct full E.164 telephone number on the remote router.

- If you have configured a CODEC value, there can be a problem if both VoIP dial peers on either side of the connection have incompatible CODEC values. Make sure that both VoIP peers have been configured with the same CODEC value.
- Use the **debug vpm spi** command to verify the output string the router dials is correct.
- Use the **debug cch323 rtp** command to check RTP packet transport.
- Use the **debug cch323 h225** command to check the call setup.

# Configure Voice Ports

The Cisco 3600 currently provides only analog voice ports for its implementation of Voice over IP. The type of signaling associated with these analog voice ports depends on the interface module installed into the device. The Cisco 3600 series router supports either a two-port or four-port voice network module (VNM); VNMs can hold either 2 or 4 voice interface cards (VICs).

Each VIC is specific to a particular signaling type; therefore, VICs determine the type of signaling for the voice ports on that particular VNM. This means that even though VNMs can hold multiple VICs, each VIC on a VNM must conform to the same signaling type. For more information about the physical characteristics of VNMs and VICs or how to install them, refer to the installation document, *Voice Network Module and Voice Interface Card Configuration Note*, that came with your VNM.

Voice ports on the Cisco 3600 series support three basic voice signaling types:

- FXO—Foreign Exchange Office interface. The FXO interface is an RJ-11 connector that allows a connection to be directed at the PSTN's central office (or to a standard PBX interface, if the local telecommunications authority permits). This interface is of value for off-premise extension applications.
- FXS—The Foreign Exchange Station interface is an RJ-11 connector that allows connection for basic telephone equipment, keysets, PBXs, and supplies ring, voltage, and dial tone.
- E&M—The "Ear and Mouth" interface (or "RecEive and TransMit" interface) is an RJ-48 connector that allows connection for PBX trunk lines (tie lines). It is a signaling technique for 2-wire and 4-wire telephone and trunk interfaces.

In general, voice port commands define the characteristics associated with a particular voice port signaling type. Under most circumstances, the default voice port command values are adequate to configure FXO and FXS ports to transport voice data over your existing IP network. Because of the inherent complexities involved with PBX networks, E&M ports might need specific voice port values configured, depending on the specifications of the devices in your telephony network.

## Configuring FXO or FXS Voice Ports

Under most circumstances the default voice port values are adequate for both FXO and FXS voice ports. If you need to change the default configuration for these voice ports, perform the following tasks. Items included in Step 1 and Step 2 are required; items included in Step 3 are optional.

**Step 1** Identify the voice port and enter the voice-port configuration mode by using the **voice-port** command.

**Step 2** Configure the following mandatory voice-port parameters by using the indicated commands:

- Dial type (FXO only) using the **dial-type** command
- Signal type using the **signal** command

- Call progress tone using the **cptone** command
- Ring frequency (FXS only) using the **ring frequency** command
- Ring number (FXO only) using the **ring number** command

**Step 3** Configure one or more of the following optional voice-port parameters by using the indicated commands:

- PLAR connection mode using the **connection plar** command
- Music-threshold using the **music-threshold** command
- Description using the **description** command
- Comfort noise (if VAD is activated—VAD is a dial-peer command) using the **comfort-noise** command

To configure FXO and FXS voice ports, use the following commands beginning in privileged EXEC mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **configure terminal** | Enter the global configuration mode. |
| 2 | **voice-port** *slot-number/subunit-number/port* | Identify the voice port you want to configure and enter the voice-port configuration mode. |
| 3 | **dial-type** {**dtmf** \| **pulse**} | (For FXO ports only) Select the appropriate dial type for out-dialing. |
| 4 | **signal** {**loop-start** \| **ground-start**} | Select the appropriate signal type for this interface. |
| 5 | **cptone** *country* | Select the appropriate voice call progress tone for this interface.<br>The default for this command is **us**. For a list of supported countries, refer to the *Voice, Video, and Home Applications Command Reference*. |
| 6 | **ring frequency** {**25** \| **50**} | (For FXS ports only) Select the appropriate ring frequency (in Hertz) specific to the equipment attached to this voice port. |
| 7 | **ring number** *number* | (For FXO ports only) Specify the maximum number of rings to be detected before answering a call. |
| 8 | **connection plar** *string* | (Optional) Specify the private line auto ringdown (PLAR) connection, if this voice port is used for a PLAR connection. The *string* value specifies the destination telephone number. |
| 9 | **music-threshold** *number* | (Optional) Specify the threshold (in decibels) for on-hold music. Valid entries are from -70 to -30. |
| 10 | **description** *string* | (Optional) Attach descriptive text about this voice port connection. |
| 11 | **comfort-noise** | (Optional) Specify that background noise will be generated. |

### Validation Tips

You can check the validity of your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for a dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.
- Use the **show voice-port** command to verify that the data configured is correct.

### Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Network Protocols Configuration Guide, Part 1*.
- Use the **show voice port** command to make sure that the port is enabled. If the port is offline, use the **no shutdown** command.
- If you have configured E&M interfaces, make sure that the values pertaining to your specific PBX setup, such as timing and/or type, are correct.
- Check to see if the voice network module has been correctly installed. For more information, refer to the installation document, *Voice Network Module and Voice Interface Card Configuration Note*, that came with your voice network module.

## Fine-Tune FXO and FXS Voice Ports

Depending on the specifics of your particular network, you might need to adjust voice parameters involving timing, input gain, and output attenuation for FXO or FXS voice ports. Collectively, these commands are referred to as voice-port tuning commands.

---

**Note** In most cases, the default values for voice-port tuning commands will be sufficient.

---

To configure voice-port tuning for FXO and FXS voice ports, perform the following steps:

**Step 1** Identify the voice port and enter the voice-port configuration mode using the **voice-port** command.

**Step 2** For each of the following parameters, select the appropriate value using the commands indicated:

- Input gain using the **input gain** command
- Output attenuation using the **output attenuation** command
- Echo cancel coverage using the **echo-cancel enable** and **echo-cancel coverage** commands
- Non-linear processing using the **non-linear** command
- Initial digit timeouts using the **timeouts initial** command
- Interdigit timeouts using the **timeouts interdigits** command
- Timing other than timeouts, using the **timing digit**, **timing inter-digit**, **timing pulse-digit** and **timing pulse-inter-digit** commands.

To fine-tune FXO or FXS voice ports, use the following commands beginning in privileged EXEC mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **configure terminal** | Enter the global configuration mode. |
| 2 | **voice-port** *slot-number/subunit-number/port* | Identify the voice-port you want to configure and enter the voice-port configuration mode. |
| 3 | **input gain** *value* | Specify (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from -6 to 14. |
| 4 | **output attenuation** *value* | Specify (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14. |
| 5 | **echo-cancel enable** | Enable echo-cancellation of voice that is sent out the interface and received back on the same interface. |
| 6 | **echo-cancel coverage** *value* | Adjust the size (in milliseconds) of the echo-cancel. Acceptable values are 16, 24, and 32. |
| 7 | **non-linear** | Enable non-linear processing, which shuts off any signal if no near-end speech is detected. (Non-linear processing is used with echo-cancellation.) |
| 8 | **timeouts initial** *seconds* | Specify the number of seconds the system will wait for the caller to input the first digit of the dialed digits. Valid entries for this command are from 0 to 120. |
| 9 | **timeouts interdigit** *seconds* | Specify the number of seconds the system will wait (after the caller has input the initial digit) for the caller to input a subsequent digit. Valid entries for this command are from 0 to 120. |
| 10 | **timing digit** *milliseconds* | If the voice-port dial type is DTMF, configure the DTMF digit signal duration. The range of the DTMF digit signal duration is from 50 to 100. The default is 100. |
| 11 | **timing inter-digit** *milliseconds* | If the voice-port dial type is DTMF, configure the DTMF inter-digit signal duration. The range of the DTMF inter-digit signal duration is from 50 to 500. The default is 100. |
| 12 | **timing pulse-digit** *milliseconds* | (FXO ports only) If the voice-port dial type is pulse, configure the pulse digit signal duration. The range of the pulse digit signal duration is from10 to 20. The default is 20. |
| 13 | **timing pulse-inter-digit** *milliseconds* | (FXO ports only) If the voice-port dial type is pulse, configure the pulse inter-digit signal duration. The range of the pulse inter-digit signal duration is from 100 to 1000. The default is 500. |

**Note** After you change any voice-port command, it is a good idea to cycle the port by using the **shutdown** and **no shutdown** commands.

# Configure E&M Voice Ports

Unlike FXO and FXS voice ports, the default E&M voice-port parameters most likely will not be sufficient to enable voice data transmission over your IP network. E&M voice-port values must match those specified by the particular PBX device to which it is connected.

To configure an E&M voice port, perform the following steps. Items included in Step 1 and Step 2 are required; items included in Step 3 are optional.

**Step 1** Identify the voice port and enter the voice-port configuration mode using the **voice-port** command.

**Step 2** For each of the following required parameters, select the appropriate parameter value using the commands indicated:

- Dial type using the **dial-type** command
- Signal type using the **signal** command
- Call progress tone using the **cptone** command
- Operation using the **operation** command
- Type using the **type** command
- Impedance using the **impedance** command

**Step 3** Select one or more of the following optional parameters, using the indicated commands:

- Connection mode using the **connection plar** command
- Music-threshold using the **music-threshold** command
- Description using the **description** command
- Comfort tone (if VAD is activated) using the **comfort-noise** command

To configure E&M voice ports, use the following commands beginning in privileged EXEC mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **configure terminal** | Enter the global configuration mode. |
| 2 | **voice-port** *slot-number/subunit-number/port* | Identify the voice port you want to configure and enter the voice-port configuration mode. |
| 3 | **dial-type** {**dtmf** \| **pulse**} | Select the appropriate dial type for out-dialing. |
| 4 | **signal** {**wink-start** \| **immediate** \| **delay-dial**} | Select the appropriate signal type for this interface. |
| 5 | **cptone** {**australia** \| **brazil** \| **china** \| **finland** \| **france** \| **germany** \| **japan** \| **northamerica** \| **unitedkingdom**} | Select the appropriate voice call progress tone for this interface. |
| 6 | **operation** {**2-wire** \| **4-wire**} | Select the appropriate cabling scheme for this voice port. |

| Step | Command | Purpose |
|---|---|---|
| 7 | **type** {**1** \| **2** \| **3** \| **5**} | Select the appropriate E&M interface type.<br>Type 1 indicates the following lead configuration:<br>— E—output, relay to ground<br>M—input, referenced to ground<br>Type 2 indicates the following lead configuration:<br>— E—output, relay to SG<br>M—input, referenced to ground<br>SB—feed for M, connected to -48V<br>SG—return for E, galvanically isolated from ground<br>Type 3 indicates the following lead configuration:<br>— E—output, relay to ground<br>M—input, referenced to ground<br>SB—connected to -48V<br>SG—connected to ground<br>Type 5 indicates the following lead configuration:<br>— E—output, relay to ground<br>M—input, referenced to -48V. |
| 8 | **impedance** {**600c** \| **600r** \| **900c** \| **complex1** \| **complex2**} | Specify a terminating impedance. This value must match the specifications from the telephony system to which this voice port is connected. |
| 9 | **connection plar** *string* | (Optional) Specify the private line auto ringdown (PLAR) connection, if this voice port is used for a PLAR connection. The *string* value specifies the destination telephone number. |
| 10 | **music-threshold** *number* | (Optional) Specify the threshold (in decibels) for on-hold music. Valid entries are from -70 to -30. |
| 11 | **description** *string* | (Optional) Attach descriptive text about this voice port connection. |
| 12 | **comfort-noise** | (Optional) Specify that background noise will be generated. |

## Validation Tips

You can check the validity of your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for a dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.
- Use the **show voice port** command to verify that the data configured is correct.

### Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Network Protocols Configuration Guide, Part 1*.
- Use the **show voice-port command** to make sure that the port is enabled. If the port is offline, use the **no shutdown** command.
- If you have configured E&M interfaces, make sure that the values pertaining to your specific PBX setup, such as timing and/or type, are correct.
- Check to see if the voice network module has been correctly installed. For more information, refer to the installation document that came with your voice network module.

## Fine-Tune E&M Voice Ports

Depending on the specifics of your particular network, you may need to adjust voice parameters involving timing, input gain, and output attenuation for E&M voice ports. Collectively, these commands are referred to as voice-port tuning commands.

**Note** In most cases, the default values for voice-port tuning commands will be sufficient.

To configure voice-port tuning for E&M voice ports, perform the following steps:

**Step 1** Identify the voice port and enter the voice-port configuration mode by using the **voice-port** command.

**Step 2** For each of the following parameters, select the appropriate value, using the commands indicated:

- Input gain using the **input gain** command
- Output attenuation using the **output attenuation** command
- Echo cancel coverage using **echo-cancel enable** and **echo-cancel coverage** commands
- Non-linear processing using the **non-linear** command
- Initial digit timeouts using the **timeouts initial** command
- Interdigit timeouts using the **timeouts interdigit** command
- Timing other than timeouts using the **timing clear-wait**, **timing delay-duration**, **timing delay-start**, **timing dial-pulse min-delay**, **timing digit**, **timing pulse**, **timing pulse-inter-digit**, **timing wink-duration**, and **timing wink-wait** commands.

To fine-tune E&M voice ports, use the following commands beginning in privileged EXEC mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **configure terminal** | Enter the global configuration mode. |
| 2 | **voice-port** *slot-number/subunit-number/port* | Identify the voice port you want to configure and enter the voice-port configuration mode. |

| Step | Command | Purpose |
|---|---|---|
| 3 | **input gain** *value* | Specify (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from -6 to 14. |
| 4 | **output attenuation** *value* | Specify (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14. |
| 5 | **echo-cancel enable** | Enable echo-cancellation of voice that is sent out the interface and received back on the same interface. |
| 6 | **echo-cancel coverage** *value* | Adjust the size (in milliseconds) of the echo-cancel. Acceptable values are 16, 24, and 32. |
| 7 | **non-linear** | Enable non-linear processing, which shuts off any signal if no near-end speech is detected. (Non-linear processing is used with echo-cancellation.) |
| 8 | **timeouts initial** *seconds* | Specify the number of seconds the system will wait for the caller to input the first digit of the dialed digits. Valid entries for this command are from 0 to 120. |
| 9 | **timeouts interdigit** *seconds* | Specify the number of seconds the system will wait (after the caller has input the initial digit) for the caller to input a subsequent digit. Valid entries for this command are from 0 to 120. |
| 10 | **timing clear-wait** *milliseconds*<br>**timing delay-duration** *milliseconds*<br>**timing delay-start** *milliseconds*<br>**timing dial-pulse min-delay** *milliseconds*<br>**timing digit** *milliseconds*<br>**timing inter-digit** *milliseconds*<br>**timing pulse** *pulse-per-second*<br>**timing pulse-inter-digit** *milliseconds*<br>**timing wink-duration** *milliseconds*<br>**timing wink-wait** *milliseconds* | Specify timing parameters. Valid entries for **clear-wait** are from 200 to 2000 milliseconds. Valid entries for **delay-duration** are from 100 to 5000 milliseconds. Valid entries for **delay-start** are from 20 to 2000 milliseconds. Valid entries for **dial-pulse min-delay** are from 0 to 5000 milliseconds. Valid entries for **digit** are from 50 to 100 milliseconds. Valid entries for **inter-digit** are from 50 to 500 milliseconds. Valid entries for **pulse** are from 10 to 20. Valid entries for **pulse-inter-digit** are 100 to 1000 milliseconds. Valid entries for **wink-duration** are from 100 to 400 milliseconds. Valid entries for **wink-wait** are from 100 to 5000 milliseconds. |

**Note** After you change any voice-port command, it is a good idea to cycle the port by using the **shutdown** and **no shutdown** commands.

# Optimize Dial Peer and Network Interface Configurations

Depending on how you have configured your network interfaces, you might need to configure additional VoIP dial-peer parameters. This section describes the following topics:

- Configure IP Precedence for Dial Peers
- Configure RSVP for Dial Peers
- Configure CODEC and VAD for Dial Peers

## Configure IP Precedence for Dial Peers

If you want to give real-time voice traffic a higher priority than other network traffic, you can weight the voice data traffic associated with a particular VoIP dial peer by using IP Precedence. IP Precedence scales better than RSVP but provides no admission control.

To give real-time voice traffic precedence over other IP network traffic, use the following commands, beginning in global configuration mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **dial-peer voice** *number* **voip** | Enter the dial-peer configuration mode to configure a VoIP peer. |
| 2 | **ip precedence** *number* | Select a precedence level for the voice traffic associated with that dial peer. |

In IP Precedence, the numbers 1 through 5 identify classes for IP flows; the numbers 6 through 7 are used for network and backbone routing and updates.

For example, to ensure that voice traffic associated with VoIP dial peer 103 is given a higher priority than other IP network traffic, enter the following:

```
dial-peer voice 103 voip
 ip precedence 5
```

In this example, when an IP call leg is associated with VoIP dial peer 103, all packets transmitted to the IP network via this dial peer will have their precedence bits set to 5. If the networks receiving these packets have been configured to recognize precedence bits, the packets will be given priority over packets with a lower configured precedence value.

## Configure RSVP for Dial Peers

If you have configured your WAN or LAN interfaces for RSVP, you must configure the QoS for any associated VoIP peers. To configure quality of service for a selected VoIP peer, use the following commands, beginning in global configuration mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **dial-peer voice** *number* **voip** | Enter the dial-peer configuration mode to configure a VoIP peer. |
| 2 | **req-qos** [**best-effort** \| **controlled-load** \| **guaranteed-delay**] | Specify the desired quality of service to be used. |

**Note** We suggest that you select **controlled-load** for the requested quality of service.

For example, to specify guaranteed delay QoS for VoIP dial peer 108, enter the following:

```
Dial-peer voice 108 voip
 destination-pattern +1408528
 req-qos controlled-load
 session target ipv4:10.0.0.8
```

In this example, every time a connection is made through VoIP dial peer 108, an RSVP reservation request is made between the local router, all intermediate routers in the path, and the final destination router.

To generate an SNMP trap message if the reserved QoS is less than the configured value for a selected VoIP peer, use the following commands, beginning from the global configuration mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **dial-peer voice** *number* **voip** | Enter the dial-peer configuration mode to configure a VoIP peer. |
| 2 | **acc-qos** [**best-effort** \| **controlled-load** \| **guaranteed-delay**] | Specify the QoS value below which an SNMP trap will be generated. |

**Note** RSVP reservations are only one-way. If you configure RSVP, the VoIP dial peers on both ends of the connection must be configured for RSVP.

# Configure CODEC and VAD for Dial Peers

Coder-decoder (CODEC) and voice activity detection (VAD) for a dial peer determine how much bandwidth the voice session uses. CODEC typically is used to transform analog signals into a digital bit stream and digital signals back into analog signals—in this case, it specifies the voice coder rate of speech for a dial peer. VAD is used to disable the transmission of silence packets.

## Configure CODEC for a VoIP Dial Peer

To specify a voice coder rate for a selected VoIP peer, use the following commands, initially beginning in global configuration mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **dial-peer voice** *number* **voip** | Enter the dial-peer configuration mode to configure a VoIP peer. |
| 2 | **codec** [**g711alaw** \| **g711ulaw** \| **g729r8**] | Specify the desired voice coder rate of speech. |

The default for the **codec** command is **g729r8**; normally the default configuration for this command is the most desirable. If, however, you are operating on a high bandwidth network and voice quality is of the highest importance, you should configure the **codec** command for **g711alaw** or **ulaw**. Using this value will result in better voice quality, but it will also require higher bandwidth requirements for voice.

For example, to specify a CODEC rate of G.711a-law for VoIP dial peer 108, enter the following:

```
Dial-peer voice 108 voip
 destination-pattern +1408528
 codec g711alaw
 session target ipv4:10.0.0.8
```

### Configure VAD for a VoIP Dial Peer

To disable the transmission of silence packets for a selected VoIP peer, use the following commands, beginning in global configuration mode:

| Step | Command | Purpose |
|---|---|---|
| 1 | **dial-peer voice** *number* **voip** | Enter the dial-peer configuration mode to configure a VoIP peer. |
| 2 | **vad** | Disable the transmission of silence packets (enabling VAD). |

The default for the **vad** command is enabled; normally the default configuration for this command is the most desirable. If you are operating on a high bandwidth network and voice quality is of the highest importance, you should disable **vad**. Using this value will result in better voice quality, but it will also require higher bandwidth requirements for voice.

For example, to enable VAD for VoIP dial peer 108, enter the following:

```
Dial-peer voice 108 voip
 destination-pattern +1408528
 vad
 session target ipv4:10.0.0.8
```

# Configure Voice over IP for Microsoft NetMeeting

Voice over IP can be used with Microsoft NetMeeting (Version 2.x) when the Cisco 3600 series router is used as the voice gateway. Use the latest version of DirectX drivers from Microsoft on your PC to improve the voice quality of NetMeeting.

## Configure Voice over IP to Support Microsoft NetMeeting

To configure Voice over IP to support NetMeeting, create a VoIP peer that contains the following information:

- Session Target—IP address or DNS name of the PC running NetMeeting
- CODEC—g711ulaw or g711alaw

## Configure Microsoft NetMeeting for Voice over IP

To configure NetMeeting to work with Voice over IP, complete the following steps:

**Step 1** From the Tools menu in the NetMeeting application, select **Options**. NetMeeting will display the Options dialog box.

**Step 2** Click the **Audio** tab.

**Step 3** Click the “Calling a telephone using NetMeeting” check box.

**Step 4** Enter the IP address of the Cisco 3600 series router in the **IP address** field.

**Step 5** Under **General**, click **Advanced**.

**Step 6** Click the “Manually configured compression settings” check box.

**Step 7** Select the CODEC value **CCITT ulaw 8000Hz**.

**Step 8** Click the **Up** button until this CODEC value is at the top of the list.

**Step 9** Click **OK** to exit.

## Initiate a Call Using Microsoft NetMeeting

To initiate a call using Microsoft NetMeeting, perform the following steps:

**Step 1** Click the Call icon from the NetMeeting application. Microsoft NetMeeting will open the call dialog box.

**Step 2** From the Call dialog box, select **call using H.323 gateway**.

**Step 3** Enter the telephone number in the **Address** field.

**Step 4** Click **Call** to initiate a call to the Cisco 3600 series router from Microsoft NetMeeting.

# Managing System Performance

This chapter describes the basic tasks that you can perform to manage the general system performance.

For a complete description of the performance management commands in this chapter, refer to the "Performance Management Commands" chapter of the *Configuration Fundamentals Command Reference*. To locate documentation of other commands that appear in this chapter, use the command reference master index or search online.

## System Performance Management Task List

Perform any of the tasks in the following sections to manage system performance:

- Set the Interval for Load Data
- Limit TCP Transactions
- Configure Switching and Scheduling Priorities
- Establish Queueing and Congestion Strategies
- Configure Generic Traffic Shaping
- Modify the System Buffer Size

In addition, most chapters in this guide include performance tasks specific to the chapter content, and the *Internetworking Design Guide* includes detailed information on performance issues that arise when designing a network.

See the "Performance Management Examples" section at the end of this chapter for examples.

## Set the Interval for Load Data

You can change the period of time over which a set of data is used for computing load statistics. Decisions, such as dial backup decisions, are dependent on these statistics. If you decrease the load interval, the average statistics are computed over a shorter period of time and are more responsive to bursts of traffic.

To change the length of time for which a set of data is used to compute load statistics, perform the following task in interface configuration mode:

| Task | Command |
|---|---|
| Set the length of time for which data is used for load calculations. | **load-interval** *seconds* |

# Limit TCP Transactions

When using a standard TCP implementation to send keystrokes between machines, TCP tends to send one packet for each keystroke typed, which can use up bandwidth and contribute to congestion on larger networks.

John Nagle's algorithm (RFC 896) helps alleviate the small-packet problem in TCP. The first character typed after connection establishment is sent in a single packet, but TCP holds any additional characters typed until the receiver acknowledges the previous packet. Then the second, larger packet is sent, and additional typed characters are saved until the acknowledgment comes back. The effect is to accumulate characters into larger chunks, and pace them out to the network at a rate matching the round-trip time of the given connection. This method is usually good for all TCP-based traffic. However, do not enable the Nagle slow packet avoidance algorithm if you have XRemote users on X Window sessions.

By default, the Nagle algorithm is not enabled. To enable the Nagle algorithm and thereby reduce TCP transactions, perform the following task in global configuration mode:

| Task | Command |
|---|---|
| Enable the Nagle slow packet avoidance algorithm. | **service nagle** |

# Configure Switching and Scheduling Priorities

The normal operation of the network server allows the switching operations to use as much of the central processor as is required. If the network is running unusually heavy loads that do not allow the processor the time to handle the routing protocols, you might need to give priority to the system process scheduler. To do so, perform the following task in global configuration mode:

| Task | Command |
|---|---|
| Define the maximum amount of time that can elapse without running the lowest-priority system processes. | **scheduler interval** *milliseconds* |

To change the amount of time that the CPU spends on fast switching and process level operations on the Cisco 7200 series and Cisco 7500 series, perform the following task in global configuration mode:

| Task | Command |
|---|---|
| For the Cisco 7200 series and Cisco 7500 series, change the default time the CPU spends on process tasks and fast switching. | **scheduler allocate** *network-microseconds process-microseconds* |

**Caution** Cisco recommends that you do not change the default values of the **scheduler allocate** command.

# Establish Queueing and Congestion Strategies

There are four possible queueing algorithms used: first-come-first-serve (FCFS), weighted fair queueing, priority queueing, and custom queueing. For serial interfaces at E1 (2.048 Mbps) and below, weighted fair queueing is used by default. When no other queueing strategies are configured, all other interfaces use FCFS by default. There is also one congestion avoidance algorithm available: random early detection.

You can configure the Cisco IOS software to support the following types of queueing and congestion strategies for prioritizing network traffic:

- Weighted Fair Queueing
- Priority Queueing
- Custom Queueing
- Random Early Detection

You can configure weighted fair queueing, priority queueing, custom queueing, or random early detection, but you can assign only one type to an interface.

**Note** Weighted fair queueing, priority queueing, and custom queueing are not supported on tunnels.

## Weighted Fair Queueing

When enabled for an interface, weighted fair queueing provides traffic priority management that automatically sorts among individual traffic streams without requiring that you first define access lists.

Weighted fair queueing can manage duplex data streams, such as those between pairs of applications, and simplex data streams such as voice or video. From the perspective of weighted fair queueing, there are two categories of data streams: high-bandwidth sessions and low-bandwidth sessions. Low-bandwidth traffic has effective priority over high-bandwidth traffic, and high-bandwidth traffic shares the transmission service proportionally according to assigned weights.

When you enable weighted fair queueing for an interface, new messages for high-bandwidth conversations are discarded after the congestive-messages threshold you set or the default one has been met. However, low-bandwidth conversations, which include control-message conversations, continue to enqueue data. As a result, the fair queue may occasionally contain more messages than are specified by the threshold number.

## Priority Queueing

Priority output queueing is a mechanism that allows the administrator to set priorities on the type of traffic passing through the network. Packets are classified according to various criteria, including protocol and subprotocol type, and then queued on one of four output queues (high, medium, normal, and low).

When the server is ready to transmit a packet, it scans the priority queues in order, from highest to lowest, to find the highest-priority packet. After that packet is completely transmitted, the server scans the priority queues again. If a priority output queue fills up, packets are dropped and, for IP, quench indications are sent to the original transmitter.

Although you can enable priority output queueing for any interface, the intended application was for low-bandwidth, congested serial interfaces. Cisco's priority output queueing mechanism allows traffic control based on protocol or interface type. You can also set the size of the queue and defaults for what happens to packets that are not defined by priority output queue rules.

The priority output queueing mechanism can be used to manage traffic from all networking protocols. Additional fine-tuning is available for IP and for setting boundaries on the packet size.

**Note** Priority queueing introduces extra overhead that is acceptable for slow interfaces, but may not be acceptable for higher-speed interfaces such as Ethernet.

The four priority queues—high, medium, normal, and low—are listed in order from highest to lowest priority. Keepalives sourced by the network server are always assigned to the high-priority queue; all other management traffic (such as IGRP updates) must be configured. Packets that are not classified by the priority list mechanism are assigned to the normal queue.

A priority list is a set of rules that describes how packets should be assigned to priority queues. A priority list might also describe a default priority or the queue size limits of the various priority queues.

## Custom Queueing

Priority queueing introduces a fairness problem in that packets classified to lower-priority queues might not get serviced in a timely manner or at all, depending upon the bandwidth used by packets sent from the higher-priority output queues.

With custom output queueing, a "weighted fair" queueing strategy is implemented for the processing of interface output queues. You can control the percentage of an interface's available bandwidth that is used by a particular kind of traffic. When custom queueing is enabled on an interface, the system maintains 17 output queues for that interface that can be used to modify queueing behavior. You can specify queues 1 through 16.

For queue numbers 1 through 16, the system cycles through the queues sequentially, delivering packets in the current queue before moving on to the next. Associated with each output queue is a configurable byte count, which specifies how many bytes of data the system should deliver from the current queue before it moves on to the next queue. When a particular queue is being processed, packets are sent until the number of bytes sent exceed the queue byte count or the queue is empty. Bandwidth used by a particular queue can only be indirectly specified in terms of byte count and queue length.

Queue number 0 is a system queue; it is emptied before any of the queues numbered 1 through 16 are processed. The system queues high-priority packets, such as keepalive packets, to this queue. Other traffic cannot be configured to use this queue.

On most platforms, all protocols are classified in the fast switching path.

**Note** With custom or priority queueing enabled, the system takes longer to switch packets because the packets are classified by the processor card.

## Random Early Detection

Random early detection is useful in high-speed networks to provide a congestion avoidance mechanism (as opposed to a congestion management mechanism such as queueing). When enabled on an interface, random early detection begins dropping packets at a rate you select during configuration when congestion occurs.

Random early detection is recommended only for TCP/IP networks. You can use random early detection as a way to cause TCP to back off traffic. TCP not only pauses, but it also restarts quickly and adapts its transmission rate to the rate that the network can support.

Random early detection is not recommended for protocols, such as AppleTalk or Novell Netware, that respond to dropped packets by retransmitting the packets at the same rate. Random early detection should only be configured on an interface where most of the traffic is TCP/IP traffic.

For interfaces configured to use RSVP, random early detection chooses packets from other flows to drop rather than the RSVP flows. Also, IP precedence governs which packets are dropped—traffic that is at a lower precedence has a higher drop rate and therefore is more likely to be throttled back.

## Queueing Task List

You can set up weighted fair queueing, priority queueing, custom queueing, or random early detection on your network, but you can assign only one of the four to an interface.

The following sections describe the tasks that you can choose from, depending on the needs of your network:

- Set Weighted Fair Queueing for an Interface
- Enable Priority Queueing
- Enable Custom Queueing
- Enable Random Early Detection on an Interface

---

**Note** To configure priority queueing or custom queueing over X.25 or LAPB, refer to the "Configuring X.25 and LAPB" chapter in the *Wide-Area Networking Configuration Guide*.

---

## Set Weighted Fair Queueing for an Interface

To enable weighted fair queueing for an interface, set the congestion threshold after which messages for high-bandwidth conversations are dropped, and specify the number of dynamic and reservable queues, perform the following task in interface configuration mode after specifying the interface:

| Task | Command |
|---|---|
| Configure an interface to use weighted fair queueing. | f**air-queue** [*congestive-discard-threshold* [*dynamic-queues* [*reservable-queues*]]] |

To disable weighted fair queueing for an interface, use the **no fair-queue** command.

Fair queueing is enabled by default for physical interfaces whose bandwidth is less than or equal to 2.048 megabits per second (Mbps) and that do not use Link Access Procedure, Balanced (LAPB), X.25, or Synchronous Data Link Control (SDLC) encapsulations. (Fair queueing is not an option for these protocols.) However, if custom queueing or priority queueing is enabled for a qualifying link,

it overrides fair queueing, effectively disabling it. Additionally, fair queueing is automatically disabled if you enable autonomous or SSE switching. Fair queueing is now enabled by default on interfaces configured for Multilink PPP.

## Enable Priority Queueing

To enable priority queueing, perform the tasks in the following sections. The first and third tasks are required.

- Assign Packets to Priority Queues
- Specify the Maximum Packets in the Priority Queues
- Assign a Priority Group to an Interface
- Monitor the Priority Queueing Lists

### Assign Packets to Priority Queues

You can assign packets to priority lists based on the protocol type or the interface where the packets enter the router. In addition, you can set the default queue for packets which do not match other assignment rules. To define the priority lists, perform the following tasks in global configuration mode:

| Task | Command |
|---|---|
| Establish queueing priorities based upon the protocol type. | **priority-list** *list-number* **protocol** *protocol-name* {**high** \| **medium** \| **normal** \| **low**} *queue-keyword keyword-value* |
| Establish queueing priorities for packets entering from a given interface. | **priority-list** *list-number* **interface** *interface-type interface-number* {**high** \| **medium** \| **normal** \| **low**} |
| Assign a priority queue for those packets that do not match any other rule in the priority list. | **priority-list** *list-number* **default** {**high** \| **medium** \| **normal** \| **low**} |

All protocols supported by Cisco are allowed. The *queue-keyword* variable provides additional options including byte-count, TCP service and port number assignments, and AppleTalk, IP, IPX, VINES, or XNS access list assignments. See the **priority-list** command syntax description in the "Performance Management Commands" chapter in the *Configuration Fundamentals Command Reference*.

When using multiple rules, remember that the system reads the **priority-list** commands in order of appearance. When classifying a packet, the system searches the list of rules specified by **priority-list** commands for a matching protocol or interface type. When a match is found, the packet is assigned to the appropriate queue. The list is searched in the order it is specified, and the first matching rule terminates the search.

### Specify the Maximum Packets in the Priority Queues

You can specify the maximum number of packets allowed in each of the priority queues. To do so, perform the following task in global configuration mode:

| Task | Command |
|---|---|
| Specify the maximum number of packets allowed in each of the priority queues. | **priority-list** *list-number* **queue-limit** *high-limit medium-limit normal-limit low-limit* |

### Assign a Priority Group to an Interface

You can assign a priority list number to an interface. Only one list can be assigned per interface. To assign an priority group to an interface, perform the following task in interface configuration mode:

| Task | Command |
|---|---|
| Assign a priority list number to the interface. | **priority-group** *list-number* |

### Monitor the Priority Queueing Lists

You can display information about the input and output queues when priority queueing is enabled on an interface. To do so, perform the following task in EXEC mode:

| Task | Command |
|---|---|
| Show the status of the priority queueing lists. | **show queueing priority** |

## Enable Custom Queueing

To enable custom queueing, perform the tasks in the following sections. The first and third tasks are required.

- Assign Packets to Custom Queues
- Specify the Maximum Packets and Bytes in the Custom Queues
- Assign a Custom Queue to an Interface
- Monitor the Custom Queueing Lists

### Assign Packets to Custom Queues

You can assign packets to custom queues based on the protocol type or the interface where the packets enter the router. In addition, you can set the default queue for packets which do not match other assignment rules. To define the custom queueing lists, perform the following tasks in global configuration mode:

| Task | Command |
|---|---|
| Establish queueing priorities based upon the protocol type. | **queue-list** *list-number* **protocol** *protocol-name queue-number queue-keyword keyword-value* |
| Establish custom queueing based on packets entering from a given interface. | **queue-list** *list-number* **interface** *interface-type interface-number queue-number* |
| Assign a queue number for those packets that do not match any other rule in the custom queue list. | **queue-list** *list-number* **default** *queue-number* |

All protocols supported by Cisco are allowed. The *queue-keyword* variable provides additional options, including byte-count, TCP service and port number assignments, and AppleTalk, IP, IPX, VINES, or XNS access list assignments. See the **queue-list** command syntax description in the "Performance Management Commands" chapter in the *Configuration Fundamentals Command Reference*.

When using multiple rules, remember that the system reads the **queue-list** commands in order of appearance. When classifying a packet, the system searches the list of rules specified by **queue-list** commands for a matching protocol or interface type. When a match is found, the packet is assigned to the appropriate queue. The list is searched in the order it is specified, and the first matching rule terminates the search.

### Specify the Maximum Packets and Bytes in the Custom Queues

You can specify the maximum number of packets allowed in each of the custom queues or the maximum queue size in bytes. To do so, perform one of the following tasks in global configuration mode:

| Task | Command |
|---|---|
| Specify the maximum number of packets allowed in each of the custom queues. | **queue-list** *list-number* **queue** *queue-number* **limit** *limit-number* |
| Designate the byte size allowed per queue. | **queue-list** *list-number* **queue** *queue-number* **byte-count** *byte-count-number* |

### Assign a Custom Queue to an Interface

You can assign a custom queue list number to an interface. Only one list can be assigned per interface. To assign an custom queue to an interface, perform the following task in interface configuration mode:

| Task | Command |
|---|---|
| Assign a custom queue list number to the interface. | **custom-queue-list** *list-number* |

### Monitor the Custom Queueing Lists

You can display information about the input and output queues when custom queueing is enabled on an interface. To do so, perform one of the following tasks in EXEC mode:

| Task | Command |
|---|---|
| Show the status of the custom queueing lists. | **show queueing custom** |
| Show the current status of the custom output queues when custom queueing is enabled. | s**how interface** *type number* |

## Enable Random Early Detection on an Interface

To enable random early detection on the interface, perform the following task in interface configuration mode:

| Task | Command |
|---|---|
| Enable random early detection on an interface. | **random-detect** [*weighting*] |

To monitor the various drop statistics for early random detection, perform the following tasks in EXEC mode:

| Task | Command |
|---|---|
| Show the drop statistics for the interface. | **show interface** [*type number*] |

# Configure Generic Traffic Shaping

Traffic shaping allows you to control how fast packets are sent out on the interface to avoid congestion and meet the needs of remote interfaces. You may want to configure traffic shaping on the interface if you have a network with differing access rates or if you are offering a subrate service. For example, if one end of the link in a Frame Relay network is 256 kbps and the other end of the link is only 128 kbps, sending packets at 256 kbps could cause failure of the applications using the link.

Traffic shaping is supported on all media and encapsulation types on the router. Traffic shaping can also be applied to a specific access list on an interface. To perform traffic shaping on Frame Relay virtual circuits, you can also use the **frame-relay traffic-shaping** command. For more information on Frame Relay traffic shaping, refer to the "Configuring Frame Relay" chapter in the *Wide-Area Network Configuration Guide*.

To enable traffic shaping for outbound traffic on an interface, perform one of the following tasks in interface configuration mode:

| Task | Command |
|---|---|
| Enable traffic shaping for outbound traffic on an interface. | **traffic-shape rate** *bit-rate* [*burst-size* [*excess-burst-size*]] |
| Enable traffic shaping for outbound traffic on an interface for a specified access list. | **traffic-shape group** *access-list bit-rate* [*burst-size* [*excess-burst-size*]] |

**Note** Traffic shaping is not supported with optimum, distributed, or flow switching. If you enable traffic shaping, all interfaces will revert to fast switching.

If traffic shaping is performed on a Frame Relay network with the **traffic-shape rate** command, you can also use the **traffic-shape adaptive** command to specify the minimum bit rate the traffic is shaped to.

To configure a Frame Relay subinterface to estimate the available bandwidth when backward explicit congestion notifications (BECNs) are received, perform the following task in interface configuration mode:

| Task | Command |
|---|---|
| Configure minimum bit rate that traffic is shaped to when BECNs are received on an interface. | **traffic-shape adaptive** [*bit-rate*] |

The **traffic-shape adaptive** command uses the configured bit rate as a lower bound of the range and the bit rate specified by the **traffic-shape rate** command as the upper bound. The rate that the traffic is actually shaped to will be between those two rates. Configure the **traffic-shape adaptive** command at both ends of the link because it also configures the device at the flow end to reflect forward explicit congestion notification (FECN) signals as BECNs, enabling the router at the high-speed end to detect and adapt to congestion even when traffic is flowing primarily in one direction.

To display the current traffic-shaping configuration and statistics, perform the following tasks in EXEC mode:

| Task | Command |
|---|---|
| Display the current traffic-shaping configuration. | **show traffic-shape** [*interface*] |
| Display the current traffic-shaping statistics. | **show traffic-shape statistics** [*interface*] |

For an example of configuring traffic shaping, see the section "Generic Traffic Shaping Example" at the end of this chapter.

# Modify the System Buffer Size

You can adjust initial buffer pool settings and the limits at which temporary buffers are created and destroyed. To do so, perform the following tasks in global configuration mode:

| Task | Command |
|---|---|
| Adjust the system buffer sizes. | **buffers** {**small** \| **middle** \| **big** \| **verybig** \| **large** \| **huge** \| *type number*} {**permanent** \| **max-free** \| **min-free** \| **initial**} *number* |
| Dynamically resize all huge buffers to the value that you supply. | **buffers huge size** *number* |

**Caution** Normally you need not adjust these parameters; do so only after consulting with technical support personnel. Improper settings can adversely impact system performance.

During normal system operation, there are two sets of buffer pools: public and interface.

- The buffers in the public pools grow and shrink based upon demand. Some public pools are temporary and are created and destroyed as needed. Other public pools are permanently allocated and cannot be destroyed. The public buffer pools are small, middle, big, large, very big, and huge.
- Interface pools are static—that is, they are all permanent. One interface pool exists for each interface. For example, a Cisco 4000 1E 4T configuration has one Ethernet buffer pool and four serial buffer pools. In the **buffers** command, the *type* and *number* arguments allow the user to tune the interface pools.

See the section "Buffer Modification Examples" at the end of this chapter.

The server has one pool of queueing elements and six public pools of packet buffers of different sizes. For each pool, the server keeps count of the number of buffers outstanding, the number of buffers in the free list, and the maximum number of buffers allowed in the free list. To display statistics about the buffer pool on the system, perform the following tasks in EXEC mode:

| Task | Command |
|---|---|
| Display all public pool information. | **show buffers** |
| Display all public and interface pool information. | **show buffers all** |
| Display a brief listing of all allocated buffers. | **show buffers alloc** |
| Display interface pool information. | **show buffers** [*type number*] |
| Dump all allocated buffers. | **show buffers alloc dump** |
| Display all interface pool information. | **show buffers interface** |

| Task | Command |
|---|---|
| If the specified interface has its own buffer pool, display information for that pool. | **show buffers interface** *type number* |
| Display a brief listing of buffers allocated for this interface. | **show buffers interface** *type number* **alloc** |
| Dump the buffers allocated to this interface. | **show buffers interface** *type number* **alloc dump** |

# Performance Management Examples

The following sections provide performance management examples:

- Generic Traffic Shaping Example
- Buffer Modification Examples

## Generic Traffic Shaping Example

This example shows the configuration of two traffic-shaped interfaces on a router. Ethernet 0 is configured to limit User Datagram Protocol (UDP) traffic to 1 Mbps. Ethernet 1 is configured to limit all output to 5 Mbps.

```
access-list 101 permit udp any any
interface Ethernet0
 traffic-shape group 101 1000000 125000 125000
!
interface Ethernet1
 traffic-shape rate 5000000 625000 625000
```

The following is a sample display for the **show traffic-shape** command for the example shown:

```
Router# show traffic-shape

          access Target    Byte   Sustain   Excess    Interval  Increment Adapt
I/F       list   Rate      Limit  bits/int  bits/int  (ms)       (bytes)  Active
Et0       101    1000000   23437  125000    125000    63        7813      -
Et1              5000000   87889  625000    625000    16        9766      -
```

The following is a sample display for the **show traffic-shape statistics** command for the example shown:

```
Router# show traffic-shape statistics

          Access Queue     Packets   Bytes     Packets   Bytes     Shaping
I/F       List   Depth                        Delayed   Delayed   Active
Et0       101    0         2         180       0         0         no
Et1              0         0         0         0         0         no
```

## Buffer Modification Examples

The following example instructs the system to keep at least 50 small buffers free:

```
buffers small min-free 50
```

The following example instructs the system to keep no more than 200 medium buffers free:

```
buffers middle max-free 200
```

The following example instructs the system to create one large temporary extra buffer, just after a reload:

```
buffers large initial 1
```

The following example instructs the system to create one permanent huge buffer:

```
buffers huge permanent 1
```

# Voice Over IP for the Cisco 3600 Series Commands

This chapter documents new commands. All other commands used with this feature are documented in the Cisco IOS Release 11.3 command reference documents.

The following new commands are used to configure and monitor Voice over IP:

- **acc-qos**
- **answer-address**
- **codec**
- **comfort-noise**
- **connection**
- **cptone**
- **description**
- **destination-pattern**
- **dial-control-mib**
- **dial-peer voice**
- **dial-type**
- **echo-cancel coverage**
- **echo-cancel enable**
- **expect-factor**
- **fax-rate**
- **icpif**
- **impedance**
- **input gain**
- **ip precedence**
- **ip udp checksum**
- **music-threshold**
- **non-linear**
- **num-exp**
- **operation**

- **output attenuation**
- **port**
- **prefix**
- **req-qos**
- **ring frequency**
- **ring number**
- **session-protocol**
- **session-target**
- **show call active voice**
- **show call history voice**
- **show dial-peer voice**
- **show dialplan incall number**
- **show dialplan number**
- **show num-exp**
- **show voice port**
- **shutdown** (dial-peer configuration)
- **shutdown** (voice-port configuration)
- **signal**
- **snmp enable peer-trap poor-qov**
- **snmp-server enable traps**
- **snmp trap link-status**
- **timeouts initial**
- **timeouts interdigit**
- **timing clear-wait**
- **timing delay-duration**
- **timing delay-start**
- **timing dial-pulse min-delay**
- **timing digit**
- **timing interdigit**
- **timing pulse**
- **timing pulse-interdigit**
- **timing wink-duration**
- **timing wink-wait**
- **type**
- **vad**
- **voice-port**

A subset of the commands listed are voice-port commands. Voice-port commands are supported on different voice signaling types, which vary depending on the platform. Table 4-1 lists the Cisco 3600 series voice port commands and the signaling types supported.

**Table 4-1 Cisco 3600 Series Commands and Signaling Types Supported**

| Voice Port Command | FXO | FXS | E&M |
|---|---|---|---|
| **comfort-noise** | | | |
| **connection** | | | |
| **cptone** | X | X | X |
| **description** | X | X | X |
| **dial-type** | X | | X |
| **echo-cancel coverage** | | | |
| **echo-cancel enable** | | | |
| **impedance** | X | X | X |
| **input gain** | X | X | X |
| **music-threshold** | | | |
| **non-linear** | | | |
| **operation** | | | X |
| **output attenuation** | X | X | X |
| **ring frequency** | | X | |
| **ring number** | X | | |
| **shutdown** | X | X | X |
| **signal** | X | X | X |
| **snmp trap link-status** | | | |
| **timeouts initial** | | | |
| **timeouts interdigit** | | | |
| **timing** | | | |
| **timing clear-wait** | | | X |
| **timing delay-duration** | | | X |
| **timing delay-start** | | | X |
| **timing dial-pulse min-delay** | | | X |
| **timing digit** | X | X | X |
| **timing inter-digit** | X | X | X |
| **timing pulse** | X | | X |
| **timing pulse-inter-digit** | X | | X |
| **timing wink-duration** | | | X |
| **timing wink-wait** | | | X |
| **type** | | | X |

# Command Syntax Conventions

Table 4-2 describes the syntax used with the commands in this chapter.

**Table 4-2 Command Syntax Guide**

| Convention | Description |
|---|---|
| **boldface font** | Commands and keywords. |
| *italic font* | Command input that is supplied by you. |
| [ ] | Keywords or arguments that appear within square brackets are optional. |
| { x \| x \| x } | A choice of keywords (represented by x) appears in braces separated by vertical bars. You must select one. |
| ^ or Ctrl | Represent the key labeled *Control*. For example, when you read *^D* or *Ctrl-D*, you should hold down the Control key while you press the D key. |
| `screen font` | Examples of information displayed on the screen. |
| **`boldface screen font`** | Examples of information that you must enter. |
| < > | Nonprinting characters, such as passwords, appear in angled brackets. |
| [ ] | Default responses to system prompts appear in square brackets. |
| **Note** | Means *reader take note*. Notes contain helpful suggestions or references to additional information and material. |
|  **Caution** | Means *reader be careful*. In this situation, you might do amoething that could result in equipemnt damage or loss of data. |
|  **Timesaver** | Means *the described action saves time*. You can save time by performing the action described in the paragraph. |



# acc-qos

To generate an SNMP event if the quality of service for a dial peer drops below a specified level, use the **acc-qos** dial-peer configuration command. Use the **no** form of this command to use the default value for this feature.

**acc-qos** {**best-effort** | **controlled-load** | **guaranteed-delay**}
**no acc-qos**

## Syntax Description

| | |
|---|---|
| **best-effort** | Indicates that Resource Reservation Protocol (RSVP) makes no bandwidth reservation. |
| **controlled-load** | Indicates that RSVP guarantees a single level of preferential service, presumed to correlate to a delay boundary. The controlled load service uses admission (or capacity) control to assure that preferential service is received even when the bandwidth is overloaded. |
| **guaranteed-delay** | Indicates that RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queuing if the bandwidth reserved is not exceeded. |

## Default

The default value is best-effort. Using the **no** form of this command is the same as the default.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **acc-qos** dial-peer command to generate an SNMP event if the quality of service for specified dial peer drops below the specified level. When a dial peer is used, the Cisco IOS software reserves a certain amount of bandwidth so that the selected quality of service can be provided. Cisco IOS software uses Resource Reservation Protocol (RSVP) to request quality of service guarantees from the network.

To select the most appropriate value for this command, you need to be familiar with the amount of traffic this connection supports and what kind of impact you are willing to have on it. The Cisco IOS software generates a trap message when the bandwidth required to provide the selected quality of service is not available.

This command is only applicable to VoIP peers.

## Example

The following example selects guaranteed-delay as the specified level below which an SNMP trap message will be generated:

```
dial-peer voice 10 voip
 acc-qos guaranteed-delay
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**req-qos**



# answer-address

To specify the full E.164 telephone number to be used to identify the dial peer of an incoming call, use the **answer-address** dial-peer configuration command. Use the **no** form of this command to disable this feature.

**answer-address** [+]*string*
**no answer-address**

## Syntax Description

| | |
|---|---|
| *string* | Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are:<br>• Digits 0 through 9, letters A through D, pound sign (#), and asterisk (*), which represent specific digits that can be entered.<br>• Plus sign (+), which is optionally used as the first digit to indicate an E.164 standard number.<br>• Comma (,), which inserts a pause between digits.<br>• Period (.), which matches any entered digit. |

## Default

The default value is enabled with a null string.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **answer-address** command to identify the origin (or dial peer) of incoming calls from the IP network. Cisco IOS software identifies the dial peers of a call in one of two ways: either by identifying the interface through which the call is received or through the telephone number configured with the **answer-address** command. In the absence of a configured telephone number, the peer associated with the interface will be associated with the incoming call.

For calls coming in from a POTS interface, the **answer-address** command is not used to select an incoming dial peer. The incoming POTS dial peer is selected on the basis of the port configured for that dial peer.

This command is applicable to both VoIP and POTS dial peers.

---

**Note** The Cisco IOS software does not check the validity of the E.164 telephone number; it will accept any series of digits as a valid number.

---

## Example

The following example configures the E.164 telephone number, “555-9626” as the dial peer of an incoming call:

```
dial-peer voice 10 pots
 answer-address +5559626
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**destination-pattern**
**port**
**prefix**

# codec

To specify the voice coder rate of speech for a dial peer, use the **codec** dial-peer configuration command. Use the **no** form of this command to reset the default value for this command.

**codec** {**g711alaw** | **g711ulaw** | **g729r8**}
**no codec**

## Syntax Description

| | |
|---|---|
| **g711alaw** | G.711 A-Law 64,000 bits per second (bps). |
| **g711ulaw** | G.711 u-Law 64,000 bps. |
| **g729r8** | G.729 8000 bps. |

## Default

The default value for this command is **g729r8**.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **codec** command to define a specific voice coder rate of speech for a dial peer.

For toll quality, use **g711alaw** or **g711ulaw**. These values provide high-quality voice transmission but use a significant amount of bandwidth. For almost toll quality (and a significant savings in bandwidth), use the **g729r8** value.

If **codec** values for the VoIP peers of a connection do not match, the call will fail.

This command is only applicable to VoIP peers.

## Example

The following example configures a voice coder rate that provides toll quality and uses a relatively high amount of bandwidth:

```
dial-peer voice 10 voip
 codec g711alaw
```

# comfort-noise

To specify whether or not background noise should be generated, use the **comfort-noise** voice-port configuration command. Use the **no** form of this command to disable this feature.

**comfort-noise**
**no comfort-noise**

## Syntax Description

This command has no arguments or keywords.

## Default

The default value for this command is enabled.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **comfort-noise** command to generate background noise to fill silent gaps during calls if VAD is activated. If **comfort noise** is not enabled, and VAD is enabled at the remote end of the connection, the user will hear dead silence when the remote party is not speaking.

The configuration of **comfort noise** only affects the silence generated at the local interface; it does not affect the use of VAD on either end of the connection, or the silence generated at the remote end of the connection.

## Example

The following example enables background noise:

```
voice-port 1/0/0
 comfort-noise
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**vad**



# connection

To specify a connection mode for a specified voice port, use the **connection** voice-port configuration command. Use the **no** form of this command to disable the selected connection mode.

**connection plar** *string*
**no connection plar** *string*

## Syntax Description

| | |
|---|---|
| **plar** | Specifies a private line auto ringdown (PLAR) connection. PLAR is handled by associating a peer directly with an interface; when an interface goes off-hook, the peer is used to set up the second call leg and conference them together without the caller having to dial any digits. |
| *string* | Specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number. |

## Default

The default value for this command is no connection.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **connection** command to specify a connection mode for a specific interface. Use the **connection plar** command to specify a PLAR interface. The string you configure for this command is used as the called number for all calls coming in over this voice port. The destination dial peer is determined on the basis of this called number.

If the **connection** command is not configured, the standard session application outputs a dial tone when the interface goes off-hook until enough digits are collected to match a dial-peer and complete the call.

## Example

The following example selects PLAR as the connection mode, with a destination telephone number of 555-9262:

```
voice-port 1/0/0
 connection plar 5559262
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**session-protocol**

# cptone

To configure a voice call progress tone locale, use the **cptone** voice-port configuration command. Use the **no** form of this command to disable this feature.

**cptone** {**australia** | **brazil** | **china** | **france** | **germany** | **japan** | **northamerica** | **unitedkingdom**}
**no cptone**

## Syntax Description

| | |
|---|---|
| **australia** | Specifies an analog voice interface-related default tone, ring, and cadence setting for Australia. |
| **brazil** | Specifies an analog voice interface-related default tone, ring, and cadence setting for Brazil. |
| **china** | Specifies an analog voice interface-related default tone, ring, and cadence setting for China. |
| **finland** | Specifies an analog voice interface-related default tone, ring, and cadence setting for Finland. |
| **france** | Specifies an analog voice interface-related default tone, ring, and cadence setting for France. |
| **germany** | Specifies an analog voice interface-related default tone, ring, and cadence setting for Germany. |
| **japan** | Specifies an analog voice interface-related default tone, ring, and cadence setting for Japan. |
| **northamerica** | Specifies an analog voice interface-related default tone, ring, and cadence setting for North America. |
| **unitedkingdom** | Specifies an analog voice interface-related default tone, ring, and cadence setting for the United Kingdom. |

## Default

The default value for this command is **northamerica**.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **cptone** command to specify a regional analog voice interface-related tone, ring, and cadence setting for a specified voice port. This command only affects the tones generated at the local interface. It does not affect any information passed to the remote end of a connection, or any tones generated at the remote end of a connection.

## Example

The following example configures North America as the call progress tone locale:

```
voice-port 1/0/0
 cptone northamerica
```

# description

To include a description of what this voice port is connected to, use the **description** voice-port configuration command. Use the **no** form of this command to disable this feature.

**description** *string*
**no description**

## Syntax Description

| | |
|---|---|
| *string* | Character string from 1 to 255 characters. |

## Default

The default for this command is enabled with a null string.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **description** command to include descriptive text about this voice-port connection. This information is displayed when you enter a **show** command and does not affect the operation of the interface in any way.

## Example

The following example identifies this voice port as being connected to the Purchasing department:

```
voice-port 1/0/0
 description purchasing_dept
```

# destination-pattern

To specify either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer, use the **destination-pattern** dial-peer configuration command. Use the **no** form of this command to disable this feature.

**destination-pattern** [+]*string*
**no destination-pattern**

## Syntax Description

*string* Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are:

- Digits 0 through 9, letters A through D, pound sign (#), and asterisk (*), which represent specific digits that can be entered.
- Plus sign (+), which is optionally used as the first digit to indicate an E.164 standard number.
- Comma (,), which inserts a pause between digits.
- Period (.), which matches any entered digit.

## Default

The default value for this command is enabled with a null string.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **destination-pattern** command to define the E.164 telephone number for this dial peer. This pattern is used to match dialed digits to a dial peer. The dial peer is then used to complete the call.

This command is applicable to both VoIP and POTS dial peers.

---

**Note** The Cisco IOS software does not check the validity of the E.164 telephone number; it will accept any series of digits as a valid number.

---

## Example

The following example configures the E.164 telephone number, “555-7922,” for a dial peer:

```
dial-peer voice 10 pots
 destination-pattern +5557922
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**answer-address**
**prefix**

# dial-control-mib

To specify attributes for the call history table, use the **dial-control-mib** global configuration command.

**dial-control-mib** {**max-size** *number* | **retain-timer** *number*}

## Syntax Description

| | |
|---|---|
| **max-size** *number* | Specifies the maximum size of the call history table. Valid entries are from **0** to **500** table entries. A value of 0 will prevent any history from being retained. |
| **retain-timer** *number* | Specifies the length of time, in minutes, for entries in the call history table. Valid entries are from **0** to **2147483647** minutes. A value of 0 will prevent any history from being retained. |

## Default

The default call history table length is 50 table entries. The default retain timer is 15 minutes.

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

## Example

The following example configures the call history table to hold 400 entries, with each entry remaining in the table for 10 minutes:

```
configure terminal
 dial-control-mib max-size 400
 dial-control-mib retain-timer 10
```

# dial-peer voice

To enter the dial-peer configuration mode (and specify the method of voice-related encapsulation), use the **dial-peer voice** global configuration command.

**dial-peer voice** *number* {**voip** | **pots**}

## Syntax Description

| | |
|---|---|
| *number* | Digit(s) defining a particular dial peer. Valid entries are from 1 to 2147483647. |
| **voip** | Indicates that this is a VoIP peer using voice encapsulation on the POTS network. |
| **pots** | Indicates that this is a POTS peer using Voice over IP encapsulation on the IP backbone. |

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **dial-peer voice** global configuration command to switch to the dial-peer configuration mode from the global configuration mode. Use the **exit** command to exit the dial-peer configuration mode and return to the global configuration mode.

## Example

The following example accesses the dial-peer configuration mode and configures a POTS peer identified as dial peer 10:

```
configure terminal
 dial-peer voice 10 pots
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**voice-port**

# dial-type

To specify the type of out-dialing for voice port interfaces, use the **dial-type** voice-port configuration command. Use the **no** form of this command to disable this feature.

**dial-type** {**dtmf** | **pulse**}
**no dial-type**

## Syntax Description

| | |
|---|---|
| **dtmf** | Specifies a touch-tone dialer. |
| **pulse** | Specifies a pulse dialer. |

## Default

The default value for this command is **dtmf**.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **dial-type** command to specify an out-dialing type for an FXO or E&M voice port interface; this command is not applicable to FXS voice ports because they do not generate out-dialing. Voice ports can always detect dtmf and pulse signals. This command does not affect voice port dialing detection.

The **dial-type** command affects out-dialing as configured for the dial peer.

## Example

The following example configures a voice port to support a touch-tone dialer:

```
voice-port 1/0/0
 dial-type dtmf
```

# echo-cancel coverage

To adjust the size of the echo cancel, use the **echo-cancel coverage** voice-port configuration command. Use the **no** form of this command to reset this command to the default value.

**echo-cancel coverage** *value*
**no echo-cancel coverage** *value*

## Syntax Description

| | |
|---|---|
| *value* | Number of milliseconds the echo-canceller will cover on a given signal. Valid values are 16, 24, and 32. |

## Default

The default value for this command is 16.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **echo-cancel coverage** command to adjust the coverage size of the echo canceller. This command enables cancellation of voice that is sent out the interface and received back on the same interface within the configured amount of time. If the local loop (the distance from the analog interface to the connected equipment producing the echo) is longer, the configured value of this command should be extended.

If you configure a longer value for this command, it will take the echo canceller longer to converge; in this case, the user might hear slight echo when the connection is initially set up. If the configured value for this command is too short, the user might hear some echo for the duration of the call because the echo canceller is not cancelling the longer delay echoes.

There is no echo or echo cancellation on the IP side of the connection.

---

**Note** This command is valid only if the echo cancel feature has been enabled. For more information, refer to the **echo-cancel enable** command.

---

## Example

The following example adjusts the size of the echo canceller to 16 milliseconds:

```
voice-port 1/0/0
 echo-cancel enable
 echo-cancel coverage 16
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**echo-cancel enable**

# echo-cancel enable

To enable the echo cancel feature, use the **echo-cancel enable** voice-port configuration command. Use the **no** form of this command to disable this feature.

**echo-cancel enable**
**no echo-cancel enable**

## Syntax Description

This command has no arguments or keywords.

## Default

The default value for this command is enabled for all interface types.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The **echo-cancel** command enables cancellation of voice that is sent out the interface and is received back on the same interface. Disabling echo cancellation might cause the remote side of a connection to hear an echo. Because echo cancellation is an invasive process that can minimally degrade voice quality, this command should be disabled if it is not needed.

The **echo-cancel** command does not affect the echo heard by the user on the analog side of the connection.

There is no echo path for a 4-wire E&M interface. The echo canceller should be disabled for that interface type.

---

**Note** This command is valid only if the **echo-cancel coverage** command has been configured. For more information, refer to the **echo-cancel coverage** command.

---

## Example

The following example enables the echo cancel feature for 16-millisecond echo coverage:

```
voice-port 1/0/0
 echo-cancel enable
 echo-cancel coverage 16
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**echo-cancel coverage**
**non-linear**

# expect-factor

To specify when the router will generate an alarm to the network manager, indicating that the expected quality of voice has dropped, use the **expect-factor** dial-peer configuration command. Use the **no** form of this command to reset the default value for this command.

**expect-factor** *value*
**no expect-factor** *value*

## Syntax Description

| | |
|---|---|
| *value* | Integers that represent the ITU specification for quality of voice as described in G.113. Valid entries are from 0 to 20, with 0 representing toll quality. |

## Default

The default value for this command is 10.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Voice over IP monitors the quality of voice received over the network. Use the **expect-factor** command to specify when the router will generate an SNMP trap to the network manager.

This command is only applicable to VoIP peers.

## Example

The following example configures toll quality of voice when connecting to a dial peer:

```
dial-peer voice 10 voip
 expect-factor 0
```

# fax-rate

To establish the rate at which a fax will be sent to the specified dial peer, use the **fax-rate** dial-peer configuration command. Use the **no** form of this command to reset the default value for this command.

**fax-rate**{**2400** | **4800** | **7200** | **9600** | **14400** | **disable** | **voice**}
**no fax-rate**

## Syntax Description

| | |
|---|---|
| **2400** | Specifies a fax transmission speed of 2400 bits per second (bps). |
| **4800** | Specifies a fax transmission speed of 4800 bps. |
| **7200** | Specifies a fax transmission speed of 7200 bps. |
| **9600** | Specifies a fax transmission speed of 9600 bps. |
| **14400** | Specifies a fax transmission speed of 14,400 bps. |
| **disable** | Disables fax relay transmission capability. |
| **voice** | Specifies the highest possible transmission speed allowed by voice rate. |

## Default

The default value for this command is **voice**.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **fax-rate** command to specify the fax transmission rate to the specified dial peer.

The values for this command apply only to the fax transmission speed and do not affect the quality of the fax itself. The higher values provide a faster transmission speed but monopolize a significantly larger portion of the available bandwidth. Slower transmission speeds use less bandwidth.

If the **fax-rate** command is set above the CODEC value in the same dial peer, the data sent over the network for fax transmission will be above the bandwidth reserved for RVSP. Because more network bandwidth will be monopolized by the fax transmission, we do not recommend setting the **fax-rate** value higher than the **codec** value. If the **fax-rate** value is set lower than the **codec** value, faxes will take longer to transmit but will use less bandwidth.

This command is only applicable to VoIP peers.

## Example

The following example configures a facsimile rate of 9600 bps for faxes sent to a dial peer:

```
dial-peer voice 10 voip
 fax-rate 9600
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**codec**

# icpif

To specify the Calculated Planning Impairment Factor (ICPIF) for calls sent by a dial peer, use the **icpif** dial-peer configuration command. Use the **no** form of this command to restore the default value for this command.

**icpif** *number*
**no icpif** *number*

## Syntax Description

| | |
|---|---|
| *number* | Integer, expressed in equipment impairment factor units, specifying the ICPIF value. Valid entries are 0 to 55. |

## Default

The default value for this command is 30.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **icpif** command to specify the maximum acceptable impairment factor for the voice calls sent by the selected dial peer.

This command is applicable only to VoIP peers.

## Example

The following example disables the **icpif** command:

```
dial-peer voice 10 voip
 icpif 0
```

# impedance

To specify the terminating impedance of a voice port interface, use the **impedance** voice-port configuration command. Use the **no** form of this command to restore the default value.

**impedance** {**600c** | **600r** | **900c** | **complex1** | **complex2**}
**no impedance**

## Syntax Description

| | |
|---|---|
| **600c** | Specifies 600 Ohms complex. |
| **600r** | Specifies 600 Ohms real. |
| **900c** | Specifies 900 Ohms complex. |
| **complex1** | Specifies Complex 1. |
| **complex2** | Specifies Complex 2. |

## Default

The default value for this command is 600.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **impedance** command to specify the terminating impedance of an FXO voice-port interface. The impedance value selected needs to match the specifications from the specific telephony system to which it is connected. Different countries often have different standards for impedance. CO switches in the United States are predominantly 600r. PBXes in the United States are normally either 600r or 900c.

If the impedance is set incorrectly (if there is an impedance mismatch), there will be a significant amount of echo generated (which could be masked if the **echo-cancel** command has been enabled). In addition, gains might not work correctly if there is an impedance mismatch.

Configuring the impedance on a voice port will change the impedance on both voice ports of a VNM card. This voice port must be shut down and then opened for the new value to take effect.

This command is applicable to FXS, FXO, and E&M voice ports.

## Example

The following example configures an FXO voice port for a terminating impedance of 600 Ohms:

```
impedance 600r
```

# input gain

To configure a specific input gain value, use the **input gain** voice-port configuration command. Use the **no** form of this command to disable this feature.

**input gain** *value*
**no input gain** *value*

## Syntax Description

| | |
|---|---|
| *value* | Specifies, in decibels, the amount of gain to be inserted at the receiver side of the interface. Acceptable value is any integer from -6 to 14. |

## Default

The default value for FXO, FXS, and E&M ports is 0.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

A system-wide loss plan must be implemented using both **input gain** and **output attenuation** commands. Other equipment (including PBXes) in the system must be taken into account when creating a loss plan. This default value for this command assumes that a standard transmission loss plan is in effect, meaning that normally, there must be -6 dB attenuation between phones. Connections are implemented to provide -6 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 0.

Please note that you can't increase the gain of a signal going out into the PSTN, but you can decrease it. Therefore, if the voice level is too high, you can decrease the volume by either decreasing the input gain value or by increasing the output attenuation.

You can increase the gain of a signal coming in to the router. If the voice level is too low, you can increase the input gain.

## Example

The following example configures a 3-decibel gain to be inserted at the receiver side of the interface:

```
input gain 3
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**output attenuation**

# ip precedence

To set IP precedence (priority) for packets sent by the dial peer, use the **ip precedence** dial-peer configuration command. Use the **no** form of this command to restore the default value for this command.

**ip precedence** *number*
**no ip precedence**

## Syntax Description

| | |
|---|---|
| *number* | Integer specifying the IP precedence value. Valid entries are 0 to 7. A value of 0 means that no precedence (priority) has been set. |

## Default

The default value for this command is 0.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **ip precedence** command to configure the value set in the IP precedence field when voice data packets are sent over the IP network. This command should be used if the IP link utilization is high and the QoS for voice packets need to have a higher priority than other IP packets. The **ip precedence** command should also be used if RSVP is not enabled, and the user would like to give voice packets a higher priority over other IP data traffic.

This command is applicable to VoIP peers.

## Example

The following example sets the IP precedence at 5:

```
dial-peer voice 10 voip
 ip precedence 5
```

# ip udp checksum

To calculate the UDP checksum for voice packets transmitted by the dial peer, use the **ip udp checksum** dial-peer configuration command. Use the **no** form of this command to disable this feature.

**ip udp checksum**
**no ip udp checksum**

## Syntax Description

This command has no arguments or keywords.

## Default

The default value for this command is disabled.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **ip udp checksum** command to enable UDP checksum calculation for each of the outbound voice packets. This command is disabled by default to speed up the transmission of the voice packets. If you suspect that the connection has a high error rate, you should enable **ip udp checksum** to prevent bad voice packets forwarded to the DSP.

This command is applicable to VoIP peers.

## Example

The following example calculates the UDP checksum for voice packets transmitted by this dial peer:

```
dial-peer voice 10 voip
 ip udp checksum
```

# music-threshold

To specify the threshold for on-hold music for a specified voice port, use the **music-threshold** voice-port configuration command. Use the **no** form of this command to disable this feature.

**music-threshold** *number*
**no music-threshold** *number*

## Syntax Description

| | |
|---|---|
| *number* | Specifies the on-hold music threshold in decibels. Valid entries are any integer from -70 to -30. |

## Default

The default value for this command is -38.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **music-threshold** command to specify the decibel level of music played when calls are put on hold. This command tells the firmware to pass steady data above the specified level. It only affects the operation of VAD when receiving voice.

If the value for this command is set too high, VAD will interpret music-on-hold as silence, and the remote end will not hear the music. If the value for this command is set too low, VAD will compress and pass silence when the background is noisy, creating unnecessary voice traffic.

## Example

The following sets the decibel threshold to -35 for the music played when calls are put on hold:

```
voice port 1/0/0
 music-threshold -35
```

# non-linear

To enable non-linear processing in the echo canceller, use the **non-linear** voice-port configuration command. Use the **no** form of this command to disable this feature.

**non-linear**
**no non-linear**

## Syntax Description

This command has no arguments or keywords.

## Default

The default for this command is enabled for all voice-port types.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command is associated with the echo canceller operation. The **echo-cancel enable** command must be enabled for the **non-linear** command to take effect. Use the **non-linear** command to shut off any signal if no near-end speech is detected.

Enabling the **non-linear** command normally improves performance, although some users might perceive truncation of consonants at the end of sentences when this command is enabled.

This feature is also generally known as residual echo suppression.

## Example

The following example enables non-linear call processing:

```
voice-port 1/0/0
 non-linear
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**echo-cancel enable**



# num-exp

To define how to expand an extension number into a particular destination pattern, use the **num-exp** global configuration command.

**num-exp** *extension-number expanded-number*

## Syntax Description

| | |
|---|---|
| *extension-number* | Digit(s) defining an extension number for a particular dial peer. |
| *expanded-number* | Digit(s) defining the expanded telephone number or destination pattern for the extension number listed. |

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **num-exp** global configuration command to define how to expand a particular set of numbers (for example, an extension number) into a particular destination pattern. With this command, you can map specific extensions and expanded numbers together by explicitly defining each number, or you can define extensions and expanded numbers using variables. You can also use this command to convert 7-digit numbers to numbers containing less than 7 digits.

Use a period (.) as a variable or wild card, representing a single number. Use a separate period for each number you want to represent with a wildcard—meaning that if you want to replace 4 numbers in an extension with wildcards, type in 4 periods.

## Examples

The following example expands the extension number 55541 to be expanded to 14085555541:

```
num-exp 65541 14085555541
```

The following example shows how to expand all 5-digit extensions beginning with 5 to append the following numbers at the beginning of the extension number 1408555:

```
num-exp 5.... 1408555....
```

# operation

To select a specific cabling scheme for E&M ports, use the **operation** voice-port configuration command. Use the **no** form of this command as an alternative method of configuring 2-wire operation.

**operation** {**2-wire** | **4-wire**}
**no operation** {**2-wire** | **4-wire**}

## Syntax Description

| | |
|---|---|
| **2-wire** | Specifies a 2-wire E&M cabling scheme. |
| **4-wire** | Specifies a 4-wire E&M cabling scheme. |

## Default

2-wire operation

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The **operation** command will only affect voice traffic. Signaling is independent of 2-wire versus 4-wire settings. If the wrong cable scheme is specified, the user might get voice traffic in only one direction.

Configuring the **operation** command on a voice port changes the operation of both voice ports on a VPM card. The voice port must be shut down and then opened again for the new value to take effect.

This command is not applicable to FXS or FXO interfaces because those are, by definition, two-wire interfaces.

## Example

The following example specifies that an E&M port uses a 4-wire cabling scheme:

```
voice-port 1/0/0
 operation 4-wire
```

# output attenuation

To configure a specific output attenuation value, use the **output attenuation** voice-port configuration command. Use the **no** form of this command to disable this feature.

**output attenuation** *value*
**no output attenuation**

## Syntax Description

| | |
|---|---|
| *value* | Specifies, in decibels, the amount of attenuation at the transmit side of the interface. Acceptable value is any integer from 0 to 14. The default value for FXO, FXS, and E&M ports is 0. |

## Default

The default value for FXO, FXS, and E&M ports is 0.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

A system-wide loss plan must be implemented using both **input gain** and **output attenuation** commands. Other equipment (including PBXes) in the system must be taken into account when creating a loss plan. This default value for this command assumes that a standard transmission loss plan is in effect, meaning that normally, there must be -6 dB attenuation between phones. Connections are implemented to provide -6 dB of attenuation when the **input gain** and **output attenuation** commands are configured with the default value of 0.

Please note that you can't increase the gain of a signal going out into the PSTN, but you can decrease it. Therefore, if the voice level is too high, you can decrease the volume by either decreasing the input gain value or by increasing the output attenuation.

## Example

The following example configures a 3-decibel gain to be inserted at the transmit side of the interface:

```
voice-port 1/0/0
 output attenuation 3
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**input gain**

# port

To associate a dial peer with a specific voice-port, use the **port** dial-peer configuration command. Use the **no** form of this command to cancel this association.

**port** *slot-number*/*subunit-number*/*port*
**no port**

## Syntax Description

| | |
|---|---|
| *slot-number/* | Specifies the slot number in the Cisco router where the voice interface card is installed. Valid entries are from 0 to 3, depending on the slot where it has been installed. |
| *subunit-number/* | Specifies the subunit on the voice interface card where the voice port is located. Valid entries are 0 or 1. |
| *port* | Specifies the voice port. Valid entries are 0 or 1. |

## Default

No port is configured.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **port** configuration command to associate the designated voice port with the selected dial peer.

This command is used for calls incoming from a telephony interface to select an incoming dial peer and for calls coming from the VoIP network to match a port with the selected outgoing dial peer.

This command is applicable only to POTS peers.

## Example

The following example associates a dial peer with voice port 1, which is located on subunit 0, and accessed through port 0:

```
dial-peer voice 10 pots
 port 1/0/0
```

# prefix

To specify the prefix of the dialed digits for this dial peer, use the **prefix** dial-peer configuration command. Use the **no** form of this command to disable this feature.

**prefix** *string*
**no prefix**

## Syntax Description

| | |
|---|---|
| *string* | Integers representing the prefix of the telephone number associated with the specified dial peer. Valid numbers are **0** through **9**, and a comma (**,**). Use a comma to include a pause in the prefix. |

## Default

The default for this command is a null string.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **prefix** command to specify a prefix for a specific dial peer. When an outgoing call is initiated to this dial peer, the **prefix** *string* value is sent to the telephony interface first, before the telephone number associated with the dial peer.

If you want to configure different prefixes for dialed numbers on the same interface, you need to configure different dial peers.

This command is applicable only to POTS peers.

## Example

The following example specifies a prefix of “9” and then a pause:

```
dial-peer voice 10 pots
 prefix 9,
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**answer-address**
**destination-pattern**

# req-qos

To specify the desired quality of service to be used in reaching a specified dial peer, use the **req-qos** dial-peer configuration command. Use the **no** form of this command to restore the default value for this command.

**req-qos** {**best-effort** | **controlled-load** | **guaranteed-delay**}
**no req-qos**

## Syntax Description

| | |
|---|---|
| **best-effort** | Indicates that Resource Reservation Protocol (RSVP) makes no bandwidth reservation. |
| **controlled-load** | Indicates that RSVP guarantees a single level of preferential service, presumed to correlate to a delay boundary. The controlled load service uses admission (or capacity) control to assure that preferential service is received even when the bandwidth is overloaded. |
| **guaranteed-delay** | Indicates that RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queuing if the bandwidth reserved is not exceeded. |

## Default

The default value for this command is best-effort. The **no** form of this command restores the default value.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **req-qos** command to request a specific quality of service to be used in reaching a dial peer. Like **acc-qos**, when you enter this command, the Cisco IOS software reserves a certain amount of bandwidth so that the selected quality of service can be provided. Cisco IOS software uses Resource Reservation Protocol (RSVP) to request quality of service guarantees from the network.

This command is applicable only to VoIP peers.

## Example

The following example configures guaranteed-delay as the desired (requested) quality of service to a dial peer:

```
dial-peer voice 10 voip
 req-qos guaranteed-delay
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**acc-qos**

# ring frequency

To specify the ring frequency for a specified FXS voice port, use the **ring frequency** voice-port configuration command. Use the **no** form of this command to reset the default value for this command.

**ring frequency** *number*
**no ring frequency**

## Syntax Description

| | |
|---|---|
| *number* | Specifies the ring frequency (Hertz) used in the FXS interface. Valid entries are 25 and 50. |

## Default

The default value for this command is 25 Hertz.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **ring frequency** command to select a specific ring frequency for an FXS voice port. Use the **no** form of this command to reset the default value for this command, which is **25** Hertz. The ring frequency you select must match the connected equipment. If set incorrectly, the attached phone might not ring or might buzz. In addition, the ring frequency is usually country-dependent and you should take into account the appropriate ring frequency for your area before configuring this command.

This command does not affect ringback, which is the ringing a user hears when placing a remote call.

## Example

The following example configures the ring frequency for 50 Hertz:

```
voice-port 1/0/0
 ring frequency 50
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**ring number**

# ring number

To specify the number of rings for a specified FXO voice port, use the **ring number** voice-port configuration command. Use the **no** form of this command to reset the default value for this command.

**ring number** *number*
**no ring number** *number*

## Syntax Description

| | |
|---|---|
| *number* | Specifies the number of rings detected before answering the call. Valid entries are numbers from 1 to 10. |

## Default

The default value is 1 ring.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **ring number** command to set the maximum number of rings to be detected before answering a call over an FXO voice port. Use the **no** form of this command to reset the default value, which is **1** ring.

Normally, this command should be set to the default so that incoming calls are answered quickly. If you have other equipment available on the line to answer incoming calls, you might want to set the value higher to give the equipment sufficient time to respond. In that case, the FXO interface would answer if the equipment online did not answer the incoming call in the configured number of rings.

This command is not applicable to FXS or E&M interfaces because they do not receive ringing to receive a call.

## Example

The following example sets 5 rings as the maximum number of rings to be detected before closing a connection over this voice port:

```
voice port 1/0/0
 ring number 5
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**ring frequency**

# session protocol

To establish a session protocol for calls between the local and remote routers via the packet network, use the **session protocol** dial-peer configuration command. Use the **no** form of this command to reset the default value for this command.

**session protocol cisco**
**no session protocol**

## Syntax Description

| | |
|---|---|
| **cisco** | Specifies Cisco Session Protocol. |

## Default

The default value for this command is **cisco**.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

For this release, Cisco Session Protocol (**cisco**) is the only applicable session protocol. This command is applicable only to VoIP peers.

## Example

The following example selects Cisco Session Protocol as the session protocol:

```
dial-peer voice 10 voip
 session protocol cisco
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**session target**

# session target

To specify a network-specific address for a specified dial peer, use the **session target** dial-peer configuration command. Use the **no** form of this command to disable this feature.

**session target** {**ipv4:***destination-address* | **dns:**[**$s$. | $d$. | $u$.**] *host-name* | **loopback:rtp** | **loopback:compressed** | **loopback:uncompressed**}
**no session target**

## Syntax Description

| | |
|---|---|
| **ipv4:***destination-address* | IP address of the dial peer. |
| **dns:***host-name* | Indicates that the domain name server will be used to resolve the name of the IP address. Valid entries for this parameter are characters representing the name of the host device.<br><br>(Optional) You can use 1 of the following 3 wildcards with this keyword when defining the session target for VoIP peers:<br><br>• **$s$.**—Indicates that the source destination pattern will be used as part of the domain name.<br>• **$d$.**—Indicates that the destination number will be used as part of the domain name.<br>• **$u$.**—Indicates that the unmatched portion of the destination pattern (such as a defined extension number) will be used as part of the domain name. |
| **loopback:rtp** | Indicates that all voice data will be looped-back to the originating source. This is applicable for VoIP peers. |
| **loopback:compressed** | Indicates that all voice data will be looped-back in compressed mode to the originating source. This is applicable for POTS peers. |
| **loopback:uncompressed** | Indicates that all voice data will be looped-back in uncompressed mode to the originating source. This is applicable for POTS peers. |

## Default

The default for this command is enabled with no IP address or domain name defined.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **session target** command to specify a network-specific address or domain name for a dial peer. Whether you select a network-specific address or a domain name depends on the session protocol you select.

The **session target loopback** command is used for testing the voice transmission path of a call. The loopback point will depend on the call origination and the loopback type selected.

The **session target dns** command can be used with or without the specified wildcards. Using the optional wildcards can reduce the number of VoIP dial-peer session targets you need to configure if you have groups of numbers associated with a particular router.

## Example

The following example configures a session target using dns for a host, “voice_router,” in the domain “cisco.com”:

```
dial-peer voice 10 voip
 session target dns:voice_router.cisco.com
```

The following example configures a session target using dns, with the optional $u$. wildcard. In this example, the destination pattern has been configured to allow for any four-digit extension, beginning with the numbers 1310555. The optional wildcard **$u$.** indicates that the router will use the unmatched portion of the dialed number—in this case, the four-digit extension, to identify the dial peer. As in the previous example, the domain is “cisco.com.”

```
dial-peer voice 10 voip
 destination-pattern 1310555....
 session target dns:$u$.cisco.com
```

The following example configures a session target using dns, with the optional **$d$.** wildcard. In this example, the destination pattern has been configured for 13105551111. The optional wildcard $d$. indicates that the router will use the destination pattern to identify the dial peer in the “cisco.com” domain.

```
dial-peer voice 10 voip
 destination-pattern 13105551111
 session target dns:$d$.cisco.com
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**destination-pattern**
**session protocol**

# show call active voice

To show the active call table, use the **show call active voice** privileged EXEC command.

**show call active voice**

## Syntax Description

This command contains no arguments or keywords.

## Command Mode

Privileged EXEC

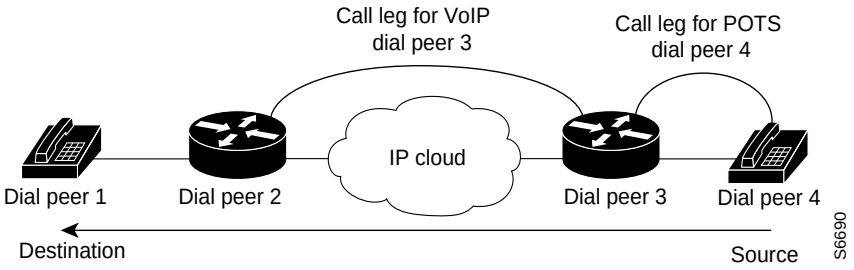
## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **show call active voice** privileged EXEC command to display the contents of the active call table, which shows all of the calls currently connected through the router.

For each call, there are 2 call legs, usually a POTS call leg and a VoIP call leg. A call leg is a discrete segment of a call connection that lies between 2 points in the connection. Each dial peer creates a call leg, as shown in Figure 4-1.

**Figure 4-1 Call Legs Example**




These 2 call legs are associated by the connection ID. The connection ID is global across the voice network, so that you can associate 2 call legs on one router with 2 call legs on another router, thereby providing an end-to-end view of a call.

## Sample Display

The following is sample output from the **show call active voice** command:

```
sloth# show call active voice
GENERIC: SetupTime=21072 Index=0 PeerAddress= PeerSubAddress= PeerId=0
PeerIfIndex=0 LogicalIfIndex=0 ConnectTime=0 CallState=3 CallOrigin=2 ChargedUnits=0
InfoType=0 TransmitPackets=375413 TransmitBytes=7508260 ReceivePackets=377734
ReceiveBytes=7554680

VOIP: ConnectionId[0x19BDF910 0xAF500007 0x0 0x58ED0] RemoteIPAddress=17635075
RemoteUDPPort=16394 RoundTripDelay=0 SelectedQoS=0 SessionProtocol=1
SessionTarget= OnTimeRvPlayout=0 GapFillWithSilence=0 GapFillWithPrediction=600
GapFillWithInterpolation=0 GapFillWithRedundancy=0 HiWaterPlayoutDelay=110
LoWaterPlayoutDelay=64 ReceiveDelay=94 VADEnable=0 CoderTypeRate=0

GENERIC: SetupTime=21072 Index=1 PeerAddress=+14085271001 PeerSubAddress=
PeerId=0 PeerIfIndex=0 LogicalIfIndex=5 ConnectTime=21115 CallState=4 CallOrigin=1
ChargedUnits=0 InfoType=1 TransmitPackets=377915 TransmitBytes=7558300
ReceivePackets=375594 ReceiveBytes=7511880

TELE: ConnectionId=[0x19BDF910 0xAF500007 0x0 0x58ED0] TxDuration=16640
VoiceTxDuration=16640 FaxTxDuration=0 CoderTypeRate=0 NoiseLevel=0 ACOMLevel=4
OutSignalLevel=-440 InSignalLevel=-440 InfoActivity=2 ERLLevel=227
SessionTarget=
```

Table 4-3 provides an alphabetical listing of the fields in this output and a description of each field.

**Table 4-3 Show Call Active Voice Field Descriptions**

| Field | Description |
|---|---|
| ACOM Level | Current ACOM level for the call. This value is sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call. |
| CallOrigin | Call origin; answer versus originate. |
| CallState | Current state of the call. |
| CoderTypeRate | Negotiated coder transmit rate of voice/fax compression during the call. |
| ConnectionId | Global call identifier of a gateway call. |
| ConnectTime | Time at which the call was connected. |
| Dial-Peer | Tag of the dial peer transmitting this call. |
| ERLLevel | Current Echo Return Loss (ERL) level for this call. |
| FaxTxDuration | Duration of fax transmission from this peer to voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value. |
| GapFillWith Silence | Duration of voice signal replaced with silence because voice data was lost or not received on time for this call. |
| GapFillWithPrediction | Duration of voice signal played out with signal synthesized from parameters or samples of data preceding in time because voice data was lost or not received in time from the voice gateway for this call. An example of such pullout is frame-eraser or frame-concealment strategies in G.729 and G.723.1 compression algorithms. |
| GapFillWithInterpolation | Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because voice data was lost or not received on time from voice gateway for this call. |

**Table 4-3 Show Call Active Voice Field Descriptions (continued)**

| Field | Description |
|---|---|
| GapFillWith Redundancy | Duration of voice signal played out with signal synthesized from redundancy parameters available because voice data was lost or not received on time from voice gateway for this call. |
| HiWaterPlayoutDelay | High water mark Voice Playout FIFO Delay during this call. |
| Index | Dial-peer identification number. |
| InfoActivity | Active information transfer activity state for this call. |
| InfoType | Information type for this call. |
| InSignalLevel | Active input signal level from the telephony interface used by this call. |
| LogicalIfIndex | Index number of the logical interface for this call. |
| LoWaterPlayoutDelay | Low water mark Voice Playout FIFO Delay during the call. |
| NoiseLevel | Active noise level for the call. |
| OnTimeRvPlayout | Duration of voice playout from data received on time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values. |
| OutSignalLevel | Active output signal level to telephony interface used by this call. |
| PeerAddress | Destination pattern associated with this peer. |
| PeerId | ID value of the peer table entry to which this call was made. |
| PeerIfIndex | Voice-port index number for this peer. |
| PeerSubaddress | Subaddress to which this call is connected. |
| ReceiveBytes | Number of bytes received by the peer during this call. |
| ReceiveDelay | Average Playout FIFO Delay plus the decoder delay during the voice call. |
| ReceivePackets | Number of packets received by this peer during this call. |
| RemoteIPAddress | Remote system IP address for the VoIP call. |
| RemoteUDPPort | Remote system UDP listener port to which voice packets are transmitted. |
| RoundTripDelay | Voice packet round trip delay between the local and remote system on the IP backbone during the call. |
| SelectedQoS | Selected RSVP quality of service (QoS) for the call. |
| SessionProtocol | Session protocol used for an Internet call between the local and remote router via the IP backbone. |
| SessionTarget | Session target of the peer used for the call. |
| SetupTime | Value of the System UpTime when the call associated with this entry was started. |
| TransmitBytes | Number of bytes transmitted from this peer during the call. |
| TransmitPackets | Number of packets transmitted from this peer during the call. |
| TxDuration | Duration of transmit path open from this peer to the voice gateway for the call. |
| VADEnable | Whether or not voice activation detection (VAD) was enabled for this call. |
| VoiceTxDuration | Duration of voice transmission from this peer to voice gateway for this call. You can derive the Voice Utilization Rate by dividing the VoiceTxDuration value by the TxDuration value. |

## Related Commands

You can use the master index or search online to find documentation of related commands.

**show call history voice**
**show dial-peer voice**
**show num-exp**
**show voice port**

# show call history voice

To display the call history table, use the **show call history voice** privileged EXEC command.

**show call history voice last** *number*

## Syntax Description

| | |
|---|---|
| **last** *number* | Displays the last calls connected, where the number of calls displayed is defined by the argument *number.* A valid entry for the argument *number* is any number from 1 to 2147483647. |

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **show call history voice** privileged EXEC command to display the call history table. The call history table contains a listing of all calls connected through this router in descending time order since Voice over IP was enabled. You can display subsets of the call history table by using specific keywords. To display the last calls connected through this router, use the keyword **last**, and define the number of calls to be displayed with the argument *number.*

## Sample Display

The following is sample output from the **show call history voice** command:

```
sloth# show call history voice
GENERIC: SetupTime=20405 Index=0 PeerAddress= PeerSubAddress= PeerId=0
PeerIfIndex=0 LogicalIfIndex=0 DisconnectCause=NORMAL DisconnectText= ConnectTime=0
DisconectTime=20595 CallOrigin=2 ChargedUnits=0 InfoType=0 TransmitPackets=0
TransmitBytes=0 ReceivePackets=0 ReceiveBytes=0

VOIP: ConnectionId[0x19BDF910 0xAF500006 0x0 0x56590] RemoteIPAddress=17635075
RemoteUDPPort=16392 RoundTripDelay=0 SelectedQoS=0 SessionProtocol=1
SessionTarget= OnTimeRvPlayout=0 GapFillWithSilence=0 GapFillWithPrediction=0
GapFillWithInterpolation=0 GapFillWithRedundancy=0 HiWaterPlayoutDelay=0
LoWaterPlayoutDelay=0 ReceiveDelay=0 VADEnable=0 CoderTypeRate=0

TELE: ConnectionId=[0x19BDF910 0xAF500006 0x0 0x56590] TxDuration=3030
VoiceTxDuration=2700 FaxTxDuration=0 CoderTypeRate=0 NoiseLevel=0 ACOMLevel=0
SessionTarget=
```

Table 4-4 provides an alphabetical listing of the fields in this output and a description of each field.

**Table 4-4 Show Call History Voice Field Descriptions**

| Field | Description |
|---|---|
| ACOMLevel | Average ACOM level for this call. This value is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call. |
| CallOrigin | Call origin; answer versus originate. |
| CoderTypeRate | Negotiated coder rate. This value specifies the transmit rate of voice/fax compression to its associated call leg for the call. |
| ConnectionID | Global call identifier for the gateway call. |
| ConnectTime | Time the call was connected. |
| DisconnectCause | Description explaining why the call was disconnected. |
| DisconnectText | Descriptive text explaining the disconnect reason. |
| DisconnectTime | Time the call was disconnected. |
| FaxDuration | Duration of fax transmitted from this peer to the voice gateway for this call. You can derive the Fax Utilization Rate by dividing this value by the TxDuration value. |
| GapFillWithSilence | Duration of voice signal replaced with silence because the voice data was lost or not received on time for this call. |
| GapFillWithPrediction | Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because the voice data was lost or not received on time from the voice gateway for this call. |
| GapFillWithInterpolation | Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because the voice data was lost or not received on time from the voice gateway for this call. |
| GapFillWithRedundancy | Duration of voice signal played out with signal synthesized from redundancy parameters available because the voice data was lost or not received on time from the voice gateway for this call. |
| HiWaterPlayoutDelay | High water mark Voice Playout FIFO Delay during the voice call. |
| Index | Index number identifying the voice-peer for this call. |
| InfoType | Information type for this call. |
| LogicalIfIndex | Index of the logical voice port for this call. |
| LoWaterPlayoutDelay | Low water mark Voice Playout FIFO Delay during the voice call. |
| NoiseLevel | Average noise level for this call. |
| OnTimeRvPlayout | Duration of voice playout from data received on time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values. |
| PeerAddress | Destination pattern or number to which this call is connected. |
| PeerId | ID value of the peer entry table to which this call was made. |
| PeerIfIndex | Index number of the logical interface through which this call was made. For ISDN media, this would be the index number of the B channel used for the call. |
| PeerSubAddress | Subaddress to which this call is connected. |
| ReceiveBytes | Number of bytes received by the peer during this call. |
| ReceiveDelay | Average Playout FIFO Delay plus the decoder delay during the voice call. |
| ReceivePackets | Number of packets received by this peer during the call. |

**Table 4-4 Show Call History Voice Field Descriptions (continued)**

| Field | Description |
|---|---|
| RemoteIPAddress | Remote system IP address for the call. |
| RemoteUDPPort | Remote system UDP listener port to which voice packets for this call are transmitted. |
| RoundTripDelay | Voice packet round trip delay between the local and remote system on the IP backbone for this call. |
| SelectedQoS | Selected RSVP quality of service for the call. |
| Session Protocol | Session protocol to be used for an Internet call between the local and remote router via the IP backbone. |
| Session Target | Session target of the peer used for the call. |
| SetUpTime | Value of the System UpTime when the call associated with this entry was started. |
| TransmitBytes | Number of bytes transmitted by this peer during the call. |
| TransmitPackets | Number of packets transmitted by this peer during the call. |
| TxDuration | Duration of the transmit path open from this peer to the voice gateway for the call. |
| VADEnable | Whether or not voice activation detection (VAD) was enabled for this call. |
| VoiceTxDuration | Duration of voice transmitted from this peer to voice gateway for this call. You can derive the Voice Utilization Rate by dividing the VoiceTxDuration by the TxDuration value. |

## Related Commands

You can use the master index or search online to find documentation of related commands.

**show call active voice**
**show dial-peer voice**
**show num-exp**
**show voice port**

# show dial-peer voice

To display configuration information for dial peers, use the **show dial-peer voice** privileged EXEC command.

**show dial-peer voice** [*number*]

## Syntax Description

| | |
|---|---|
| *number* | Displays configuration for the dial peer identified by the argument *number*. Valid entries are any integers that identify a specific dial peer, from 1 to 32767. |

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **show dial-peer voice** privileged EXEC command to display the configuration for all VoIP and POTS dial peers configured for the router. To show configuration information for only one specific dial peer, use the argument *number* to identify the dial peer.

## Sample Display

The following is sample output from the **show dial-peer voice** command for a POTS dial peer:

```
sloth# show dial-peer voice 1
VoiceEncapPeer1
        tag = 1, dest-pat = `+14085551000',
        answer-address = `',
        group = 0, Admin state is up, Operation state is down
        Permission is Both,
        type = pots, prefix = `',
        session-target = `', voice-port =
        Connect Time = 0, Charged Units = 0
        Successful Calls = 0, Failed Calls = 0
        Accepted Calls = 0, Refused Calls = 0
        Last Disconnect Cause is “”
        Last Disconnect Text is “”
        Last Setup Time = 0
```

The following is sample output from the **show dial-peer voice** command for a VoIP dial peer:

```
sloth# show dial-peer voice 10
VoiceOverIpPeer10
        tag = 10, dest-pat = `',
        incall-number = `+14087',
        group = 0, Admin state is up, Operation state is down
        Permission is Answer,
        type = voip, session-target = `',
        sess-proto = cisco, req-qos = bestEffort,
        acc-qos = bestEffort,
        fax-rate = voice, codec = g729r8,
        Expect factor = 10,Icpif = 30, VAD = disabled, Poor QOV Trap = disabled,
        Connect Time = 0, Charged Units = 0
        Successful Calls = 0, Failed Calls = 0
        Accepted Calls = 0, Refused Calls = 0
        Last Disconnect Cause is “”
        Last Disconnect Text is “”
        Last Setup Time = 0
```

Table 4-5 explains the fields contained in both of these examples.

**Table 4-5 Show Dial-Peer Voice Field Descriptions**

| Field | Description |
|---|---|
| AcceptedCalls | Number of calls from this peer accepted since system startup. |
| acc-qos | Lowest acceptable quality of service configured for calls for this peer. |
| Admin state | Administrative state of this peer. |
| Charged Units | Total number of charging units applying to this peer since system startup.The unit of measure is in hundredths of seconds. |
| codec | Default voice coder rate of speech for this peer. |
| Connect Time | Accumulated connect time to the peer since system startup for both incoming and outgoing calls. The unit of value is in hundredths of seconds. |
| dest-pat | Destination pattern (telephone number) for this peer. |
| Expect factor | User-requested Expectation Factor of voice quality for calls via this peer. |
| fax-rate | Fax transmission rate configured for this peer. |
| Failed Calls | Number of failed call attempts to this peer since system startup. |
| group | Group number associated with this peer. |
| ICPIF | Configured Calculated Planning Impairment Factor (ICPIF) value for calls sent by a dial peer. |
| incall-number | Full E.164 telephone number to be used to identify the dial peer. |
| Last Disconnect Cause | Encoded network cause associated with the last call. This value will be updated whenever a call is started or cleared and depends on the interface type and session protocol being used on this interface. |
| Last Disconnect Text | ASCII text describing the reason for the last call termination. |
| Last Setup Time | Value of the System Up Time when the last call to this peer was started. |
| Operation state | Operational state of this peer. |
| Permission | Configured permission level for this peer. |
| Poor QOV Trap | Whether Poor Quality of Voice trap messages have been enabled or disabled. |
| Refused Calls | Number of calls from this peer refused since system startup. |

**Table 4-5 Show Dial-Peer Voice Field Descriptions (continued)**

| Field | Description |
|---|---|
| req-qos | Configured requested quality of service for calls for this dial peer. |
| session-target | Session target of this peer. |
| sess-proto | Session protocol to be used for Internet calls between local and remote router via the IP backbone. |
| Successful Calls | Number of completed calls to this peer. |
| tag | Unique dial-peer ID number. |
| VAD | Whether or not voice activation detection (VAD) is enabled for this dial peer. |

## Related Commands

You can use the master index or search online to find documentation of related commands.

**show call active voice**
**show call-history voice**
**show num-exp**
**show voice port**

# show dialplan incall number

To pair different voice ports and telephone numbers together for troubleshooting, use the **show dialplan incall number** privileged EXEC command.

**show dialplan incall** *slot-number/subunit-number/port* **number** *dial string*

## Syntax Description

| | |
|---|---|
| *slot-number/* | Specifies the slot number in the Cisco router where the voice network module is installed. Valid entries are from 0 to 3, depending on the voice interface card you have installed. |
| *subunit-number/* | Specifies the subunit on the voice network module where the voice port is located. Valid entries are 0 or 1. |
| *port* | Specifies the voice port. Valid entries are 0 or 1. |
| *dial string* | Specifies a particular destination pattern (telephone number). |

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Occasionally, an incoming call cannot be matched to a dial peer in the dial-peer database. One reason this might occur is that the specified destination cannot be reached via the voice interface through which the incoming call came. Use the **show dialplan incall number** command as a troubleshooting method to resolve the call destination by pairing voice ports and telephone numbers together until there is a match.

## Example

The following example tests whether the telephone extension 57681 can be reached through voice port 1/0/1:

```
show dialplan incall 1/0/1 number 57681
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**show dialplan number**

# show dialplan number

To show which dial peer is reached when a particular telephone number is dialed, use the **show dial plan number** privileged EXEC command.

**show dial plan number** *dial string*

## Syntax Description

| | |
|---|---|
| *dial string* | Specifies a particular destination pattern (telephone number). |

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

## Example

The following example displays the dial peer associated with the destination pattern of 54567:

```
show dialplan number 54567
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**show dialplan incall number**

# show num-exp

To show the number expansions configured, use the **show num-exp** privileged EXEC command.

**show num-exp** [*dialed- number*]

## Syntax Description

*dialed-number* Displays number expansion for the specified dialed number.

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **show num-exp** privileged EXEC command to display all of the number expansions configured for this router. To display number expansion for only one number, specify that number by using the *dialed-number* argument.

## Sample Display

The following is sample output from the **show num-exp** command:

```
sloth# show num-exp
Dest Digit Pattern = '0...'     Translation = '+14085550...'
Dest Digit Pattern = '1...'     Translation = '+14085551...'
Dest Digit Pattern = '3..'      Translation = '+140855503..'
Dest Digit Pattern = '4..'      Translation = '+140855504..'
Dest Digit Pattern = '5..'      Translation = '+140855505..'
Dest Digit Pattern = '6....'    Translation = '+1408555....'
Dest Digit Pattern = '7....'    Translation = '+1408555....'
Dest Digit Pattern = '8...'     Translation = '+14085558...'
```

Table 4-6 explains the fields in the sample output.

**Table 4-6 Show Dial-Peer Voice Field Descriptions**

| Field | Description |
|---|---|
| Dest Digit Pattern | Index number identifying the destination telephone number digit pattern. |
| Translation | Expanded destination telephone number digit pattern. |

## Related Commands

You can use the master index or search online to find documentation of related commands.

**show call active voice**
**show call history voice**
**show dial-peer voice**
**show voice port**

# show voice port

To display configuration information about a specific voice port, use the **show voice port** privileged EXEC command.

**show voice port** *slot-number/subunit-number/port*

## Syntax Description

| | |
|---|---|
| *slot-number/* | Specifies the slot number in the Cisco router where the voice interface card is installed. Valid entries are from 0 to 3, depending on the slot where it has been installed. |
| *subunit-number/* | Specifies the subunit on the voice interface card where the voice port is located. Valid entries are 0 or 1. |
| *port* | Specifies the voice port. Valid entries are 0 or 1. |

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **show voice port** privileged EXEC command to display configuration and voice interface card-specific information about a specific port.

## Sample Display

The following is sample output from the **show voice port** command for an E&M voice port:

```
sloth# show voice port 1/0/0
E&M Slot is 1, Sub-unit is 0, Port is 0
 Type of VoicePort is E&M
 Operation State is unknown
 Administrative State is unknown
 The Interface Down Failure Cause is 0
 Alias is NULL
 Noise Regeneration is disabled
 Non Linear Processing is disabled
 Music On Hold Threshold is Set to 0 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is disabled
 Echo Cancel Coverage is set to 16ms
 Connection Mode is Normal
 Connection Number is
 Initial Time Out is set to 0 s
 Interdigit Time Out is set to 0 s
 Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
```

```
 Voice card specific Info Follows:
 Signal Type is wink-start
 Operation Type is 2-wire
 Impedance is set to 600r Ohm
 E&M Type is unknown
 Dial Type is dtmf
 In Seizure is inactive
 Out Seizure is inactive
 Digit Duration Timing is set to 0 ms
 InterDigit Duration Timing is set to 0 ms
 Pulse Rate Timing is set to 0 pulses/second
 InterDigit Pulse Duration Timing is set to 0 ms
 Clear Wait Duration Timing is set to 0 ms
 Wink Wait Duration Timing is set to 0 ms
 Wink Duration Timing is set to 0 ms
 Delay Start Timing is set to 0 ms
 Delay Duration Timing is set to 0 ms
```

The following is sample output from the **show voice port** command for an FXS voice port:

```
sloth# show voice port 1/0/0
Foreign Exchange Station 1/0/0 Slot is 1, Sub-unit is 0, Port is 0
 Type of VoicePort is FXS
 Operation State is DORMANT
 Administrative State is UP
 The Interface Down Failure Cause is 0
 Alias is NULL
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to 0 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 16ms
 Connection Mode is Normal
 Connection Number is
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
 Voice card specific Info Follows:
 Signal Type is loopStart
 Ring Frequency is 25 Hz
 Hook Status is On Hook
 Ring Active Status is inactive
 Ring Ground Status is inactive
 Tip Ground Status is inactive
 Digit Duration Timing is set to 100 ms
 InterDigit Duration Timing is set to 100 ms
 Hook Flash Duration Timing is set to 600 ms
```

Table 4-7 explains the fields in the sample output.

**Table 4-7 Show Voice Port Field Descriptions**

| Field | Description |
|---|---|
| Administrative State | Administrative state of the voice port. |
| Alias | User-supplied alias for this voice port. |
| Clear Wait Duration Timing | Time of inactive seizure signal to declare call cleared. |
| Connection Mode | Connection mode of the interface. |
| Connection Number | Full E.164 telephone number used to establish a connection with the trunk or PLAR mode. |
| Currently Processing | Type of call currently being processed: none, voice, or fax. |
| Delay Duration Timing | Maximum delay signal duration for delay dial signaling. |
| Delay Start Timing | Timing of generation of delayed start signal from detection of incoming seizure. |
| Dial Type | Out-dialing type of the voice port. |
| Digit Duration Timing | DTMF Digit duration in milliseconds. |
| E&M Type | Type of E&M interface. |
| Echo Cancel Coverage | Echo Cancel Coverage for this port. |
| Echo Cancellation | Whether or not echo cancellation is enabled for this port. |
| Hook Flash Duration Timing | Maximum length of hook flash signal. |
| Hook Status | Hook status of the FXO/FXS interface. |
| Impedance | Configured terminating impedance for the E&M interface. |
| In Gain | Amount of gain inserted at the receiver side of the interface. |
| In Seizure | Incoming seizure state of the E&M interface. |
| Initial Time Out | Amount of time the system waits for an initial input digit from the caller. |
| InterDigit Duration Timing | DTMF interdigit duration in milliseconds. |
| InterDigit Pulse Duration Timing | Pulse dialing interdigit timing in milliseconds. |
| Interdigit Time Out | Amount of time the system waits for a subsequent input digit from the caller. |
| Maintenance Mode | Maintenance mode of the voice-port. |
| Music On Hold Threshold | Configured Music-On-Hold Threshold value for this interface. |
| Noise Regeneration | Whether or not background noise should be played to fill silent gaps if VAD is activated. |
| Number of signaling protocol errors | Number of signaling protocol errors. |
| Non-Linear Processing | Whether or not Non-Linear Processing is enabled for this port. |
| Operations State | Operation state of the port. |
| Operation Type | Operation of the E&M signal: 2-wire or 4-wire. |
| Out Attenuation | Amount of attenuation inserted at the transmit side of the interface. |
| Out Seizure | Outgoing seizure state of the E&M interface. |
| Port | Port number for this interface associated with the voice interface card. |
| Pulse Rate Timing | Pulse dialing rate in pulses per second (pps). |
| Regional Tone | Configured regional tone for this interface. |

**Table 4-7 Show Voice Port Field Descriptions (continued)**

| Field | Description |
|---|---|
| Ring Active Status | Ring active indication. |
| Ring Frequency | Configured ring frequency for this interface. |
| Ring Ground Status | Ring ground indication. |
| Signal Type | Type of signaling for a voice port: loop-start, ground-start, wink-start, immediate, and delay-dial. |
| Slot | Slot used in the voice interface card for this port. |
| Sub-unit | Subunit used in the voice interface card for this port. |
| Tip Ground Status | Tip ground indication. |
| Type of VoicePort | Type of voice port: FXO, FXS, and E&M. |
| The Interface Down Failure Cause | Text string describing why the interface is down, |
| Wink Duration Timing | Maximum wink duration for wink start signaling. |
| Wink Wait Duration Timing | Maximum wink wait duration for wink start signaling. |

## Related Commands

You can use the master index or search online to find documentation of related commands.

**show call active voice**
**show call history voice**
**show dial-peer voice**
**show num-exp**

# shutdown (dial-peer configuration)

To change the administrative state of the selected dial peer from up to down, use the **shutdown** dial-peer configuration command. Use the **no** form of this command to change the administrative state of this dial peer from down to up.

**shutdown**
**no shutdown**

## Syntax Description

This command has no arguments or keywords.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

When a dial peer is shut down, you cannot initiate calls to that peer. This command is applicable to both VoIP and POTS peers.

## Example

The following example changes the administrative state of voice telephony dial peer 10 to down:

```
configure terminal
 dial-peer voice 10 pots
 shutdown
```

# shutdown (voice-port configuration)

To take the voice ports for a specific voice interface card offline, use the **shutdown** voice-port configuration command. Use the **no** form of this command to put the ports back in service.

**shutdown**
**no shutdown**

## Syntax Description

This command has no arguments or keywords.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

When you enter the **shutdown** command, all ports on the voice interface card are disabled. When you enter the **no shutdown** command, all ports on the voice interface card are enabled. A telephone connected to an interface will hear dead silence when a port is shut down.

## Example

The following example takes voice port 1/1/0 offline:

```
configure terminal
 voice port 1/1/0
 shutdown
```

**Note** The preceding configuration example will shut down both voice ports 1/1/0 and 1/1/1.

# signal

To specify the type of signaling for a voice port, use the **signal** voice-port configuration command. Use the **no** form of this command to restore the default value for this command.

**signal** {**loop-start** | **ground-start** | **wink-start** | **immediate** | **delay-dial**}
**no signal**

## Syntax Description

| | |
|---|---|
| **loop-start** | Specifies Loop Start signaling. Used for FXO and FXS interfaces. With Loop Start signaling only one side of a connection can hang up. This is the default setting for FXO and FXS voice ports. |
| **ground-start** | Specifies Ground Start signaling. Used for FXO and FXS interfaces. Ground Start allows both sides of a connection to place a call and to hang up. |
| **wink-start** | Indicates that the calling side seizes the line by going off-hook on its E lead then waits for a short off-hook “wink” indication on its M lead from the called side before sending address information as DTMF digits. Used for E&M tie trunk interfaces. This is the default setting for E&M voice ports. |
| **immediate** | Indicates that the calling side seizes the line by going off-hook on its E lead and sends address information as DTMF digits. Used for E&M tie trunk interfaces. |
| **delay-dial** | Indicates that the calling side seizes the line by going off-hook on its E lead. After a timing interval, the calling side looks at the supervision from the called side. If the supervision is on-hook, the calling side starts sending information as DTMF digits; otherwise, the calling side waits until the called side goes on-hook and then starts sending address information. Used for E&M tie trunk interfaces. |

## Default

The default value is **loop-start** for FXO and FXS interfaces. The default value is **wink-start** for E&M interfaces.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Configuring the **signal** command for an FXS or FXO voice port will change the signal value for both voice ports on a VPM card.

**Note** If you change the signal type for an FXO voice port, you need to move the appropriate jumper in the voice interface card of the voice network module. For more information about the physical characteristics of the voice network module, refer to the *Voice Network Module and Voice Interface Card Configuration Note* that came with your voice network module

Configuring this command for an E&M voice port will change only the signal value for the selected voice port. In either case, the voice port must be shut down and then activated before the configured values will take effect.

Some PBXes will miss initial digits if the E&M voice port is configured for Immediate signaling. If this occurs, use Delay-Dial signaling instead. Some non-Cisco devices have a limited number of DTMF receivers. This type of equipment must delay the calling side until a DTMF receiver is available.

## Example

The following example configures Ground Start signaling, which means that both sides of a connection can place a call and hang up, as the signaling type for a voice port:

```
configure terminal
 voice-port 1/1/1
 signal ground-start
```

# snmp enable peer-trap poor-qov

To generate poor quality of voice notification for applicable calls associated with VoIP dial peers, use the **snmp enable peer-trap poor-qov** dial-peer configuration command. Use the **no** form of this command to disable this feature.

**snmp enable peer-trap poor-qov**
**no snmp enable peer-trap poor-qov**

## Syntax Description

This command has no arguments or keywords.

## Default

Disabled

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **snmp enable peer-trap poor qov** command to generate poor quality of voice notifications for applicable calls associated with this dial peer. If you have an SNMP manager that will use SNMP messages when voice quality drops, you might want to enable this command. Otherwise, you should disable this command to reduce unnecessary network traffic.

This command is applicable only to VoIP peers.

## Example

The following example enables poor quality of voice notifications for calls associated with VoIP dial peer 10:

```
dial-peer voice 10 voip
 snmp enable peer-trap poor-qov
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**snmp-server enable trap voice poor-qov**
**snmp trap link-status**

# snmp-server enable traps

To enable the router to send SNMP traps, use the **snmp-server enable traps** global configuration command. Use the **no** form of this command to disable SNMP traps.

**snmp-server enable traps** [*trap-type*] [*trap-option*]
**no snmp-server enable traps** [*trap-type*] [*trap-option*]

## Syntax Description

*trap-type* (Optional) Type of trap to enable. If no type is specified, all traps are sent (including the **envmon** and **repeater** traps). The trap type can be one of the following keywords:

- **bgp**—Sends Border Gateway Protocol (BGP) state change traps.
- **config**—Sends configuration traps.
- **entity**—Sends Entity MIB modification traps.
- **envmon**—Sends Cisco enterprise-specific environmental monitor traps when an environmental threshold is exceeded. When the **envmon** keyword is used, you can specify a *trap-option* value.
- **frame-relay**—Sends Frame Relay traps.
- **isdn**—Sends Integrated Services Digital Network (ISDN) traps. When the **isdn** keyword is used on Cisco 1600 series routers, you can specify a *trap-option* value.
- **repeater**—Sends Ethernet hub repeater traps. When the **repeater** keyword is selected, you can specify a *trap-option* value.
- **rtr**—Sends response time reporter (RTR) traps.
- **snmp**—Sends Simple Network Management Protocol (SNMP) traps. When the **snmp** keyword is used, you can specify a *trap-option* value.
- **syslog**—Sends error message traps (Cisco Syslog MIB). Specify the level of messages to be sent with the **logging history level** command.
- **voice**—Sends SNMP poor quality of voice traps, when used with the **qov** *trap-option*.

*trap-option* (Optional) When the **envmon** keyword is used, you can enable a specific environmental trap type, or accept all trap types from the environmental monitor system. If no option is specified, all environmental types are enabled. The option can be one or more of the following keywords: **voltage**, **shutdown**, **supply**, **fan**, and **temperature**.

When the **isdn** keyword is used on Cisco 1600 series routers, you can specify the **call-information** keyword to enable an SNMP ISDN call information trap for the ISDN MIB subsystem, or you can specify the **isdnu-interface** keyword to enable an SNMP ISDN U interface trap for the ISDN U interface MIB subsystem.

When the **repeater** keyword is used, you can specify the repeater option. If no option is specified, all repeater types are enabled. The option can be one or more of the following keywords:

- **health**—Enables IETF Repeater Hub MIB (RFC 1516) health trap.
- **reset**—Enables IETF Repeater Hub MIB (RFC 1516) reset trap.

When the **snmp** keyword is used, you can specify the **authentication** option to enable SNMP Authentication Failure traps. (The **snmp-server enable traps snmp authentication** command replaces the **snmp-server trap-authentication** command.) If no option is specified, all SNMP traps are enabled.

When the **voice** keyword is used, you can enable SNMP poor quality of voice traps by using the **qov** option.

## Defaults

This command is disabled by default. No traps are enabled.

Some trap types cannot be controlled with this command. These traps are either always enabled or enabled by some other means. For example, the linkUpDown messages are disabled by the **no snmp trap link-status** command.

If you enter this command with no keywords, the default is to enable all trap types.

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.1.

This command is useful for disabling traps that are generating a large amount of uninteresting or useless noise.

If you do not enter an **snmp-server enable traps** command, no traps controlled by this command are sent. In order to configure the router to send these SNMP traps, you must enter at least one **snmp-server enable traps** command. If you enter the command with no keywords, all trap types are enabled. If you enter the command with a keyword, only the trap type related to that keyword is enabled. In order to enable multiple types of traps, you must enter a separate **snmp-server enable traps** command for each trap type and option.

The **snmp-server enable traps** command is used in conjunction with the **snmp-server host** command. Use the **snmp-server host** command to specify which host or hosts receive SNMP traps. In order to send traps, you must configure at least one **snmp-server host** command.

For a host to receive a trap controlled by this command, both the **snmp-server enable traps** command and the **snmp-server host** command for that host must be enabled. If the trap type is not controlled by this command, just the appropriate **snmp-server host** command must be enabled.

The trap types used in this command all have an associated MIB object that allows them to be globally enabled or disabled. Not all of the trap types available in the **snmp-server host** command have notificationEnable MIB objects, so some of these cannot be controlled using the **snmp-server enable traps** command.

## Examples

The following example enables the router to send SNMP poor quality of voice traps:

```
configure terminal
 snmp-server enable trap voice poor-qov
```

The following example enables the router to send all traps to the host myhost.cisco.com using the community string *public*:

```
snmp-server enable traps
snmp-server host myhost.cisco.com public
```

The following example enables the router to send Frame Relay and environmental monitor traps to the host myhost.cisco.com using the community string *public*:

```
snmp-server enable traps frame-relay
snmp-server enable traps envmon temperature
snmp-server host myhost.cisco.com public
```

The following example will not send traps to any host. The BGP traps are enabled for all hosts, but the only traps enabled to be sent to a host are ISDN traps.

```
snmp-server enable traps bgp
snmp-server host bob public isdn
```

## Related Commands

You can use the master indexes or search online to find documentation of related commands.

**snmp enable peer-trap peer-qov**
**snmp-server host**
**snmp-server trap-source**
**snmp trap illegal-address**
**snmp trap link-status**

# snmp trap link-status

To enable Simple Network Management Protocol (SNMP) trap messages to be generated when this voice port is brought up or down, use the **snmp trap link-status** voice-port configuration command. Use the **no** form of this command to disable this feature.

**snmp trap link-status**
**no snmp trap link-status**

## Syntax Description

This command contains no arguments or keywords.

## Default

The default for this command is enabled.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **snmp trap link-status** command to enable SNMP trap messages (linkup and linkdown) to be generated whenever this voice port is brought online or offline.

If you are managing the equipment with an SNMP manager, this command should be enabled. Enabling link-status messages will allow the SNMP manager to learn of a status change without polling the equipment. If you are not using an SNMP manager, this command should be disabled to avoid unnecessary network traffic.

## Example

The following example enables SNMP trap messages for voice-port 2/1/0:

```
voice-port 2/1/0
 snmp trap link-stat
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**smnp enable peer-trap poor-qov**
**snmp-server enable trap poor-qov**

# timeouts initial

To configure the initial digit timeout value for a specified voice port, use the **timeouts initial** voice-port configuration command. Use the **no** form of this command to restore the default value for this command.

**timeouts initial** *seconds*
**no timeouts initial** *seconds*

## Syntax Description

| | |
|---|---|
| **initial** *seconds* | Specifies the initial timeout duration in seconds. Valid entries are any integer from 0 to 120. |

## Default

The default value is 10 seconds.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **timeouts initial** command to specify the number of seconds the system will wait for the caller to input the first digit of the dialed digits. The timeouts initial timer is activated when the call is accepted and is deactivated when the caller inputs the first digit. If the configured timeout value is exceeded, the caller is notified through the appropriate tone and the call is terminated.

To disable the timeouts initial timer, set the *seconds* value to **0**.

## Example

The following example sets the initial digit timeout value to 15 seconds:

```
voice-port 1/0/0
 timeouts initial 15
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**timeouts interdigit**
**timing**

# timeouts interdigit

To configure the interdigit timeout value for a specified voice port, use the **timeouts interdigit** voice-port configuration command. Use the **no** form of this command to restore the default value for this command.

**timeouts interdigit** *seconds*
**no timeouts interdigit** *seconds*

## Syntax Description

| | |
|---|---|
| *seconds* | Specifies the interdigit timeout duration in seconds. Valid entries are any integer from 0 to 120. |

## Default

The default value is 10 seconds.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **timeouts interdigit** command to specify the number of seconds the system will wait (after the caller has input the initial digit) for the caller to input a subsequent digit of the dialed digits. The timeouts interdigit timer is activated when the caller inputs a digit and restarted each time the caller inputs another digit until the destination address is identified. If the configured timeout value is exceeded before the destination address is identified, the caller is notified through the appropriate tone and the call is terminated.

To disable the timeouts interdigit timer, set the *seconds* value to **0**.

## Example

The following example sets the interdigit timeout value for 15 seconds:

```
voice-port 1/0/0
 timeouts interdigit 15
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**timeouts initial**
**timing**

# timing clear-wait

To indicate the minimum amount of time between the inactive seizure signal and the call being cleared for a specified voice port, use the **timing clear-wait** voice-port configuration command. Use the **no** form of this command to reset the default value.

**timing clear-wait** *milliseconds*
**no timing clear-wait** *milliseconds*

## Syntax Description

| | |
|---|---|
| *milliseconds* | Minimum amount of time, in milliseconds, between the inactive seizure signal and the call being cleared. Valid entries on the Cisco 3600 series are numbers from 200 to 2000. Supported on E&M ports only. |

## Default

400 milliseconds

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

## Examples

The following example configures the clear-wait duration on a Cisco 3600 series voice port to 300 milliseconds:

```
voice-port 1/0/0
 timing clear-wait 300
```

## Related Commands

You can use the master indexes or search online to find documentation of related commands.

**timeouts initial**
**timeouts interdigit**
**timing clear-wait**
**timing delay-duration**
**timing delay-start**
**timing dial-pulse min-delay**
**timing digit**
**timing interdigit**
**timing pulse**
**timing pulse-interdigit**
**timing wink-duration**
**timing wink-wait**

# timing delay-duration

To specify the delay signal duration for a specified voice port, use the **timing delay-duration** voice-port configuration command. Use the **no** form of this command to reset the default value.

**timing delay-duration** *milliseconds*
**no timing delay-duration** *milliseconds*

## Syntax Description

| | |
|---|---|
| *milliseconds* | Delay signal duration for delay dial signaling, in milliseconds. Valid entries are numbers from 100 to 5000. Supported on E&M ports only. |

## Default

2000 milliseconds

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call direction for the **timing delay-duration** command is out.

## Examples

The following example configures the delay signal duration on a Cisco 3600 series voice port to 3000 milliseconds:

```
voice-port 1/0/0
 timing delay-duration 3000
```

## Related Commands

You can use the master indexes or search online to find documentation of related commands.

**timeouts initial**
**timeouts interdigit**
**timing clear-wait**
**timing delay-start**
**timing dial-pulse min-delay**
**timing digit**
**timing interdigit**
**timing pulse**
**timing pulse-interdigit**
**timing wink-duration**
**timing wink-wait**

# timing delay-start

To specify the minimum delay time from outgoing seizure to out-dial address for a specified voice port, use the **timing delay-start** voice-port configuration command. Use the **no** form of this command to reset the default value.

**timing delay-start** *milliseconds*
**no timing delay-start** *milliseconds*

## Syntax Description

| | |
|---|---|
| *milliseconds* | Minimum delay time, in milliseconds, from outgoing seizure to out-dial address. Valid entries are numbers from 20 to 2000. Supported on E&M ports only. |

## Default

300 milliseconds

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call direction for the **timing delay-start** command is out.

## Examples

The following example configures the delay-start duration on a Cisco 3600 series voice port to 250 milliseconds:

```
voice-port 1/0/0
 timing delay-start 250
```

## Related Commands

You can use the master indexes or search online to find documentation of related commands.

**timeouts initial**
**timeouts interdigit**
**timing clear-wait**
**timing delay-duration**
**timing dial-pulse min-delay**
**timing digit**
**timing interdigit**
**timing pulse**
**timing pulse-interdigit**
**timing wink-duration**
**timing wink-wait**

# timing dial-pulse min-delay

To specify the time between wink-like pulses for a specified voice port on the Cisco 3600 series, use the **timing dial-pulse min-delay** voice-port configuration command. Use the **no** form of this command to reset the default value.

**timing dial-pulse min-delay** *milliseconds*
**no timing dial-pulse min-delay** *milliseconds*

## Syntax Description

| | |
|---|---|
| *milliseconds* | Time, in milliseconds, between the generation of wink-like pulses. Valid entries are numbers from 0 to 5000. |

## Default

300 milliseconds

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **timing** command with the **dial-pulse min-delay** keyword with PBXes requiring a wink-like pulse, even though they have been configured for delay-dial signaling. If the value for this keyword is set to 0, the router will not generate this wink-like pulse. The call signal direction for this command is in.

## Example

The following example configures the time between the generation of wink-like pulses on a Cisco 3600 series voice port to 350 milliseconds:

```
voice-port 1/0/0
 timing dial-pulse min-delay 350
```

## Related Commands

You can use the master indexes or search online to find documentation of related commands.

**timeouts initial**
**timeouts interdigit**
**timing clear-wait**
**timing delay-duration**
**timing dialout-delay**
**timing digit**
**timing interdigit**
**timing pulse**
**timing pulse-interdigit**
**timing wink-duration**
**timing wink-wait**

# timing digit

To specify the DTMF digit signal duration for a specified voice port, use the **timing digit** voice-port configuration command. Use the **no** form of this command to reset the default value.

**timing digit** *milliseconds*
**no timing digit** *milliseconds*

## Syntax Description

| | |
|---|---|
| *milliseconds* | The DTMF digit signal duration, in milliseconds. Valid entries are numbers from 50 to 100. Supported on FXO, FXS and E&M ports. |

## Default

100 milliseconds

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call signal direction for the **timing digit** command is out.

## Examples

The following example configures the DTMF digit signal duration on a Cisco 3600 series voice port to 50 milliseconds:

```
voice-port 1/0/0
 timing digit 50
```

## Related Commands

You can use the master indexes or search online to find documentation of related commands.

**timeouts initial**
**timeouts interdigit**
**timing clear-wait**
**timing delay-duration**
**timing delay-start**
**timing dial-pulse min-delay**
**timing interdigit**
**timing pulse**
**timing pulse-interdigit**
**timing wink-duration**
**timing wink-wait**

# timing interdigit

To specify the DTMF interdigit duration for a specified voice port, use the **timing interdigit** voice-port configuration command. Use the **no** form of this command to reset the default value.

**timing interdigit** *milliseconds*
**no timing interdigit** *milliseconds*

## Syntax Description

| | |
|---|---|
| *milliseconds* | DTMF interdigit duration, in milliseconds. Valid entries are numbers from 50 to 500. Supported on FXO, FXS and E&M ports. |

## Default

100 milliseconds

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call signal direction for the **timing interdigit** command is out.

## Examples

The following example configures the DTMF interdigit duration on a Cisco 3600 series voice port to 150 milliseconds:

```
voice-port 1/0/0
 timing interdigit 150
```

## Related Commands

You can use the master indexes or search online to find documentation of related commands.

**timeouts initial**
**timeouts interdigit**
**timing clear-wait**
**timing delay-duration**
**timing delay-start**
**timing dial-pulse min-delay**
**timing digit**
**timing pulse**
**timing pulse-interdigit**
**timing wink-duration**
**timing wink-wait**

# timing pulse

To specify the pulse dialing rate for a specified voice port, use the **timing pulse** voice-port configuration command. Use the **no** form of this command to reset the default value.

**timing pulse** *pulses-per-second*
**no timing pulse** *pulses-per-second*

## Syntax Description

| | |
|---|---|
| *pulses-per-second* | Pulse dialing rate, in pulses per second. Valid entries are numbers from 10 to 20. Supported on FXO and E&M ports only. |

## Default

20

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call signal direction for the **timing pulse** command is out.

## Examples

The following example configures the pulse dialing rate on a Cisco 3600 series voice port to 15 pulses per second:

```
voice-port 1/0/0
 timing pulse 15
```

## Related Commands

You can use the master indexes or search online to find documentation of related commands.

**timeouts initial**
**timeouts interdigit**
**timing clear-wait**
**timing delay-duration**
**timing delay-start**
**timing dial-pulse min-delay**
**timing digit**
**timing interdigit**
**timing pulse-interdigit**
**timing wink-duration**
**timing wink-wait**

# timing pulse-interdigit

To specify the pulse interdigit timing for a specified voice port, use the **timing pulse-interdigit** voice-port configuration command. Use the **no** form of this command to reset the default value.

**timing pulse-interdigit** *milliseconds*
**no timing pulse-interdigit** *milliseconds*

## Syntax Description

| | |
|---|---|
| *milliseconds* | Pulse dialing interdigit timing, in milliseconds. Valid entries are numbers from 100 to 1000. Supported on FXO and E&M ports only. |

## Default

500 milliseconds

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call signal direction for the **timing pulse-interdigit** command is out.

## Examples

The following example configures the pulse-dialing interdigit timing on a Cisco 3600 series voice port to 300 milliseconds:

```
voice-port 1/0/0
 timing pulse-interdigit 300
```

## Related Commands

You can use the master indexes or search online to find documentation of related commands.

**timeouts initial**
**timeouts interdigit**
**timing clear-wait**
**timing delay-duration**
**timing delay-start**
**timing dial-pulse min-delay**
**timing digit**
**timing interdigit**
**timing pulse**
**timing wink-duration**
**timing wink-wait**

# timing wink-duration

To specify the maximum wink signal duration for a specified voice port, use the **timing wink-duration** voice-port configuration command. Use the **no** form of this command to restore the default value.

**timing wink-duration** *milliseconds*
**no timing wink-duration** *milliseconds*

## Syntax Description

| | |
|---|---|
| *milliseconds* | Maximum wink signal duration, in milliseconds, for a wink-start signal. Valid entries are numbers from 100 to 400. Supported on E&M ports only. |

## Default

200 milliseconds

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call signal direction for the **timing wink-duration** command is out.

## Examples

The following example configures the wink signal duration on a Cisco 3600 series voice port to 300 milliseconds:

```
voice-port 1/0/0
 timing wink-duration 300
```

## Related Commands

You can use the master indexes or search online to find documentation of related commands.

**timeouts initial**
**timeouts interdigit**
**timing clear-wait**
**timing delay-duration**
**timing delay-start**
**timing dial-pulse min-delay**
**timing digit**
**timing interdigit**
**timing pulse**
**timing pulse-interdigit**
**timing wink-wait**

# timing wink-wait

To specify the maximum wink-wait duration for a specified voice port, use the **timing wink-wait** voice-port configuration command. Use the **no** form of this command to reset the default value.

**timing wink-wait** *milliseconds*
**no timing wink-wait** *milliseconds*

## Syntax Description

| | |
|---|---|
| *milliseconds* | Maximum wink-wait duration, in milliseconds, for a wink start signal. Valid entries are numbers from 100 to 5000. Supported on E&M ports only |

## Default

200 milliseconds

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

The call signal direction for the **timing wink-wait** command is out.

## Examples

The following example configures the wink-wait duration on a Cisco 3600 series voice port to 300 milliseconds:

```
voice-port 1/0/0
 timing wink-wait 300
```

## Related Commands

You can use the master indexes or search online to find documentation of related commands.

**timeouts initial**
**timeouts interdigit**
**timing clear-wait**
**timing delay-duration**
**timing delay-start**
**timing dial-pulse min-delay**
**timing digit**
**timing interdigit**
**timing pulse**
**timing pulse-interdigit**
**timing wink-duration**

# type

To specify the E&M interface type, use the **type** voice-port configuration command. Use the **no** form of this command to reset the default value for this command.

**type** {**1** | **2** | **3** | **5**}
**no type**

## Syntax Description

| | |
|---|---|
| **1** | Indicates the following lead configuration:<br>E—output, relay to ground.<br>M—input, referenced to ground. |
| **2** | Indicates the following lead configuration:<br>E—output, relay to SG.<br>M—input, referenced to ground.<br>SB—feed for M, connected to -48V.<br>SG—return for E, galvanically isolated from ground. |
| **3** | Indicates the following lead configuration:<br>E—output, relay to ground.<br>M—input, referenced to ground.<br>SB—connected to -48V.<br>SG—connected to ground. |
| **5** | Indicates the following lead configuration:<br>E—output, relay to ground.<br>M—input, referenced to -48V. |

## Default

The default value is 1.

## Command Mode

Voice-port configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **type** command to specify the E&M interface for a particular voice port. With **1**, the tie-line equipment generates the E-signal to the PBX type grounding the E-lead. The tie line equipment detects the M-signal by detecting current flow to ground. If you select **1**, a common ground must exist between the line equipment and the PBX.

With **2**, the interface requires no common ground between the equipment, thereby avoiding ground loop noise problems. The E-signal is generated toward the PBX by connecting it to SG. M-signal is indicated by the PBX connecting it to SB. While Type 2 interfaces do not require a common ground, they do have the tendency to inject noise into the audio paths because they are asymmetrical with respect to the current flow between devices.

With **3**, the interface operates the same as Type 1 interfaces with respect to the E-signal. The M-signal, however, is indicated by the PBX connecting it to SB on assertion and alternately connecting it to SG during inactivity. If you select **3**, a common ground must be shared between equipment.

With **5**, the Type 5 line equipment indicates E-signal to the PBX by grounding the E-lead. The PBX indicates M-signal by grounding the M-lead. A Type 5 interface is quasi-symmetrical in that while the line is up, current flow is more or less equal between the PBX and the line equipment but noise injection is a problem.

## Example

The following example selects Type 3 as the interface type for your voice port:

```
voice-port 1/0/0
 type 3
```

# vad

To enable voice activity detection (VAD) for the calls using this dial peer, use the **vad** dial-peer configuration command. Use the **no** form of this command to disable this feature.

**vad**
**no vad**

## Syntax Description

This command has no arguments or keywords.

## Default

Enabled

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **vad** command to enable voice activity detection. With VAD, silence is not transmitted over the network, only audible speech. If you enable VAD, the sound quality will be slightly degraded but the connection will monopolize much less bandwidth. If you use the **no** form of this command, VAD is disabled and voice data is continuously transmitted to the IP backbone.

This command is applicable only to VoIP peers.

## Example

The following example enables VAD:

```
dial-peer voice 10 voip
 vad
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**comfort-noise**



# voice-port

To enter the voice-port configuration mode, use the **voice-port** global configuration command.

**voice-port** *slot-number*/*subunit-number*/*port*

## Syntax Description

| | |
|---|---|
| *slot-number/* | Specifies the slot number in the Cisco router where the voice network module is installed. Valid entries are from 0 to 3, depending on the voice interface card you have installed. |
| *subunit-number/* | Specifies the subunit on the voice network module where the voice port is located. Valid entries are 0 or 1. |
| *port* | Specifies the voice port. Valid entries are 0 or 1. |

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

Use the **voice-port** global configuration command to switch to the voice-port configuration mode from the global configuration mode. Use the **exit** command to exit the voice-port configuration mode and return to the global configuration mode.

For more information about the physical characteristics of the voice network module, or how to install it, refer to installation documentation that came with your voice network module.

## Example

The following example accesses the voice-port configuration mode for port 0, located on subunit 0 on a voice interface card installed in slot 1:

```
configure terminal
 voice-port 1/0/0
```

## Related Commands

You can use the master index or search online to find documentation of related commands.

**dial-peer**

**Anexo 4: Guía de antenas y accesorios necesarios en los puentes inalámbricos de la Serie Aironet**

El primer archivo detalla los tipos de antenas y cables de bajas pérdidas que se emplean con los puentes inalámbricos de la Serie Aironet 340. El segundo archivo detalla cómo se debe colocar la protección contra rayos del puente.



# Cisco Aironet Antennas and Accessories

## Product Overview

Every wireless Local Area Network (LAN) deployment is different. When engineering an in-building solution, varying facility sizes, construction materials, and interior divisions raise a host of transmission and multipath considerations. When implementing a building-to-building solution, distance, physical obstructions between facilities, and number of transmission points involved must be accounted for.

Cisco is committed to providing not only the best access points, client adapters, and bridges in the industry — it is also committed to providing a complete solution for any wireless LAN deployment. That's why Cisco has the widest range of antennas, cable, and accessories available from any wireless manufacturer.

With the Cisco FCC-approved directional1 and omnidirectional2 antennas, low-loss cable, mounting hardware, and other accessories, installers can customize a wireless solution that meets the requirements of even the most challenging applications.

## Key Features and Benefits

See Specifications below for a complete list of Antennas and Accessory features.

## Specifications

### Hardware

### Client Adapter Antennas

Cisco Aironet wireless client adapters come complete with standard antennas that provide sufficient range for most applications at 11 Mbps. To extend the transmission range for more specialized applications, a variety of optional, higher-gain antennas are provided that are compatible with selected client adapters. Client Adapter Antennas on page 1.

**Table 26-14: Technical Specifications for Cisco Aironet Client Adapter Antenna**

| Feature | AIR-ANT3351 | AIR-ANT3342 |
|---|---|---|
| Description | POS diversity dipole[1] | Diversity dipole |
| Application | Indoor diversity antenna[2] to extend the range of Aironet LMC client adapters | Indoor diversity antenna to extend the range of Aironet LMC client adapters |
| Gain | 2.2 dBi[3] | 2.2 dBi |
| Approximate Indoor Range at 1 Mbps[4] | 350 ft (107 m) | 350 ft (107 m) |
| Approximate Indoor Range at 11 Mbps[4] | 100 ft (51 m) | 100 ft (5 1m) |
| Beam Width | 360˚ H 75˚ V | 360˚ H 75˚ V |
| Cable Length | 5 ft (1.5 m) | 1 ft (0.3 m) |

1. A type of low-gain (2.2 dBi) antenna consisting of two (often internal) elements.
2. An intelligent system of two antennas that continually senses incoming radio signals and automatically selects the antenna best positioned to receive it.
3. A ratio of decibels to an isotropic antenna that is commonly used to measure antenna gain. The greater the dBi value, the higher the gain and, as such, the more acute the angle of coverage
4. All range estimations are based on an integrated client adapter antenna associating with an access point under ideal indoor conditions. The distances referenced here are approximations and should be used for estimation purposes only.

**Table 26-15: Physical and Environmental Specifications for Cisco Aironet Client Adapter Antenna**

| Feature | AIR-ANT3351 | AIR-ANT3342 |
|---|---|---|
| Dimensions | Base: 7 x 2 in. (18 x 5 cm)<br>Height: 8 in. (20 cm) | 4 x 3 in. (8.6 x 6.5 cm) |
| Weight | 9.2 oz. (261g) | 5 oz. (142g) |

## Access Point Antennas

Cisco Aironet access point antennas are compatible with all Cisco RP-TNC-equipped access points. The antennas are available with different gain and range capabilities, beam widths, and form factors. Coupling the right antenna with the right access point allows for efficient coverage in any facility, as well as better reliability at higher data rates.

**Table 26-16: Technical Specifications for Cisco Aironet Access Point Antenna**

| Feature | AIR-ANT3213 | AIR-ANT3194 | AIR-ANT1728 | AIR-ANT2561 | AIR-ANT3549 | AIR-ANT1729 |
|---|---|---|---|---|---|---|
| Description | Pillar mount diversity omni | Omnidirectional ceiling mount | High gain omnidirectional ceiling mount | Omnidirectional ground plane | Patch wall mount | Patch wall mount |
| Application | Indoor, unobtrusive medium-range antenna | Indoor short-range antenna, typically hung from crossbars of drop ceilings | Indoor medium-range antenna, typically hung from crossbars of drop ceilings | Flat, circular, medium-range indoor antenna | Indoor, unobtrusive, long-range antenna (may also be used as a medium-range bridge antenna) | Indoor, unobtrusive, medium-range antenna (may also be used as a medium-range bridge antenna) |
| Gain | 5.2 dBi | 2.2 dBi | 5.2 dBi | 5.2 dBi | 8.5 dBi | 6 dBi |
| Approximate Indoor Range at 1 Mbps[1] | 497 ft (151 m) | 350 ft (107 m) | 497 ft (151 m) | 497 ft (151 m) | Access Point: 700 ft (213 m)<br>Bridge: 2.0 miles (3.2 km) | Access Point: 542 ft (165 m)<br>Bridge: 1.1 miles (1.8 km) |
| Approximate Indoor Range at 11 Mbps[1] | 142 ft (44 m) | 100 ft (31 m) | 142 ft (44 m) | 142 ft (44 m) | Access Point: 200 ft (61 m)<br>Bridge: 3390 ft (103 2m) | Access Point: 155 ft (47 m)<br>Bridge: 1900 ft (580 m) |
| Beam Width[2] | 360˚ H 75˚ V | 360˚ H 75˚ V | 360˚ H 75˚ V | 360˚ H 80˚ V | 60˚ H 55˚ V | 75˚ H 65˚ V |
| Cable Length | 3 ft (0.91 m) | 9 ft (2.74 m) | 3 ft (0.91 m) | 3 ft (0.91 m) | 3 ft (0.91 m) | 3 ft (0.91 m) |

1. All range estimations are based on an external antenna associating with an integrated client adapter antenna under ideal indoor conditions. The distances referenced here are approximations and should be used for estimation purposes only.
2. The angle of signal coverage provided by a radio; it may be decreased by a directional antenna to increase gain.

**Table 26-17: Physical and Environmental Specifications for Cisco Aironet Access Point Antenna**

| Feature | AIR-ANT3213 | AIR-ANT3194 | AIR-ANT1728 | AIR-ANT2561 | AIR-ANT3549 | AIR-ANT1729 |
|---|---|---|---|---|---|---|
| Dimensions | 10 x 1 in. (25.4 x 2.5 cm) | Length: 9 in. (22.86 cm)<br>Diameter: 1 in. (2.5 cm) | Length: 9 in. (22.86 cm)<br>Diameter: 1 in. (2.5 cm) | Diameter: 12 in. (30.5 cm) | 5 x 5 in.<br>(12.4 x 12.4 cm) | 4 x 5 in.<br>(9.7 x 13 cm) |
| Weight | 1 lb. (460g) | 4.6 oz. (131g) | 4.6 oz. (131g) | 9 oz. (255g) | 5.3 oz. (150g) | 4.9 oz. (139g) |

## Bridge Antennas

Cisco Aironet bridge antennas allow for extraordinary transmission distances between two or more buildings. Available in directional configurations for point-to-point transmission and omnidirectional configuration for point-to-multipoint implementations, Cisco has a bridge antenna for every application.

**Table 26-18: Technical Specifications fro Cisco Aironet Bridge Antenna**

| Feature | AIR-ANT2506 | AIR-ANT4121 | AIR-ANT1949 | AIR-ANT3338 |
|---|---|---|---|---|
| Description | Omnidirectional mast mount | High-gain omnidirectional mast mount | Yagi mast mount | Solid dish |
| Application | Outdoor short-range point-to-multipoint applications | Outdoor medium-range point-to-multipoint applications | Outdoor medium-range directional connections | Outdoor long-range directional connections |
| Gain | 5.2 dBi | 12 dBi | 13.5 dBi | 21 dBi |
| Approximate Range at 2 Mbps[1] | 5000 ft (1525m) | 4.6 miles (7.4 km) | 6.5 miles (10.5 km) | 25 miles (40 km) |
| Approximate Range at 11 Mbps[1] | 1580 ft (480m) | 1.4 miles (2.3 km) | 2.0 miles (3.3 km) | 11.5 miles (18.5 km) |
| Beam Width | 360˚ H 75˚ V | 360˚ H 7˚ V | 30˚ H 25˚ V | 12.4˚ H 12.4˚ V |
| Cable Length | 3 ft (0.91 m) | 1 ft (0.30 m) | 1.5 ft (0.46 m) | 2 ft (0.61 m) |

1. All range estimations are based on use of 50 foot (15m) low-loss cable and the same type of antenna at each end of the connection under ideal outdoor conditions. The distances referenced here are approximations and should be used for estimation purposes only.

**Table 26-19: Physical and Environmental Specifications fro Cisco Aironet Bridge Antenna**

| Feature | AIR-ANT2506 | AIR-ANT4121 | AIR-ANT1949 | AIR-ANT3338 |
|---|---|---|---|---|
| Dimensions | Length: 13 in. (33 cm)<br>Diameter: 1 in. (2.5 cm) | Length: 40 in. (101 cm)<br>Diameter: 1.3 in. (3 cm) | Length: 18 in. (46 cm)<br>Diameter: 3 in. (7.6 cm) | Diameter 24 in. (61 cm) |
| Weight | 6 oz. (17g) | 1.5 lb. (0.68 kg) | 1.5 lb. (0.68 kg) | 11 lb. (5 kg) |

## Low-Loss Antenna Cable

Low-loss cable extends the length between any Cisco Aironet bridge and the antenna. With a loss of 6.7 dB per 100 feet (30m), low-loss cable provides installation flexibility without a significant sacrifice in range.

**Table 26-20: Specifications for Cisco Aironet Low-Loss Antenna Cable**

| Feature | AIR-420-003346-020 | AIR-420-003346-050 | AIR-420-003346-075 | AIR-420-003346-100 |
|---|---|---|---|---|
| Cable Length | 20 ft (6 m) | 50 ft (15 m) | 75 ft (23 m) | 100 ft (30 m) |
| Transmission Loss | 1.3 dBi | 3.4 dBi | 5.0 dBi | 6.7 dBi |

## Accessories

To complete out an installation, Cisco provides a variety of accessories that offer increased functionality, safety, and convenience.

**Table 26-21: Features for Cisco Aironet Accessories**

| Feature | 420-002537-018 | 420-002537-060 | 420-003354 | 420-003745 | 430-002662 |
|---|---|---|---|---|---|
| Description | 18 in. (46 cm) bulkhead extender | 60 in. (152 cm) bulkhead extender | Lightening arrestor | Optional antenna adapter cable | Yagi articulating mount |
| Application | Flexible antenna cable that extends access point cabling typically within an enclosure | Flexible antenna cable that extends access point cabling typically within an enclosure | Helps prevent damage due to lightning-induced surges or static electricity | Used to add higher-gain antennas to universal and multistation clients for longer-range applications | Adds swiveling capability to mast-mounted yagi antennas |

### Software

Software applications for Cisco products can be found at http://www.cisco.com/univercd/cc/td/doc/pcat/

## Ordering Information

### Part Numbers

For part numbers for all Cisco Aironet products, see Ordering Information in

# Mounting Instructions for the Cisco Aironet 340 Series Lightning Arrestor

**March 27, 2000**

This document describes the lightning arrestor kit and provides instructions for mounting the arrestor.

## Introduction

Overvoltage transients can be created through lightning static discharges, switch processes, direct contact with power lines or through earth currents.

Cisco Aironet lightning EMP protectors limit the amplitude and duration of disturbing interference voltages. Therefore, these protectors improve the overvoltage resistance of in-line equipment, systems, and components. The following mounting procedure balances the voltage potential, thus preventing inductive interference to parallel signal lines within the protected system.

## Warnings

Please note the following warnings to prevent accidents during the installation of the arrestor.

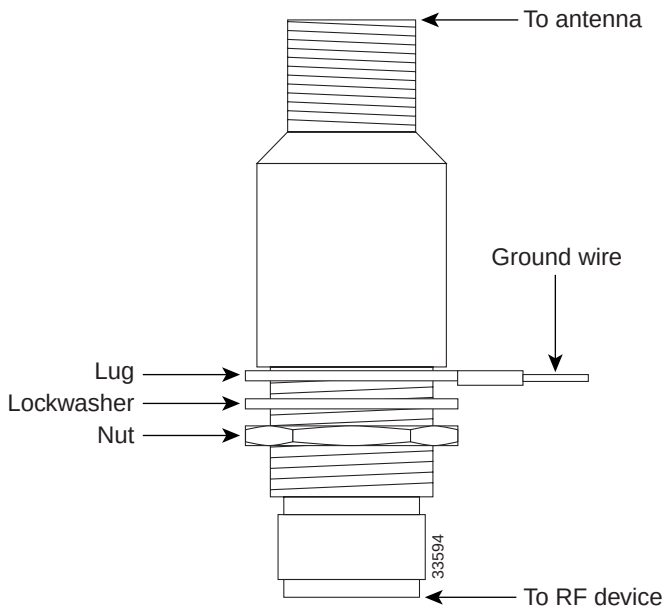
**Warning** **Disconnect or switch off in-line equipment when installing or inspecting lightning protectors during an electrical storm.**

**Warning** **When connecting lightning protectors, make sure that the succeeding equipment and components are disconnected or turned off.**

# Installation Notes

This arrestor is designed to be installed between your outdoor antenna cable and the Cisco Aironet wireless radio device. Installation should be completed indoors or inside a protected area. A good ground must be attached to the arrestor by using a ground lug attached to the arrestor and a heavy wire (#6 solid copper) connecting the lug to a good earth ground. See Figure 1.

***Figure 1 Lightning Arrestor Installation***




## Lightning Arrestor Kit

Aironet part number: 420-003354

Contains: lightning arrestor, EMP grounding ring and instruction sheet.

The kit can be ordered through your Cisco Aironet distributor.

## Anexo 5: Manual de usuario de los puentes inalámbricos de la Serie Aironet 340

Este archivo comprende la guía del usuario provista por los fabricantes de los puentes inalámbricos de la Serie Aironet.



# Using the Cisco Aironet 340 Series Wireless Bridges

March 27, 2000

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• Move the equipment farther away from the television or radio.

• Plug the equipment into an outlet that is on a different circuit from the television or radio. (That is, make certain the equipment and the television or radio are on circuits controlled by different circuit breakers or fuses.)

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*Using the Cisco Aironet*
*340 Series Wireless Bridges*

# Contents

## Chapter 3 - Before You Begin



## Chapter 4 - Configuring the Radio Network

## Chapter 5 - Configuring the Ethernet Port



## Chapter 6 - Setting Network Identifiers

## Chapter 9 - Viewing Statistics



## Chapter 10 - Setting Up the Association Table



## Chapter 11 - Using Filters

## Chapter 12 - Setting Up Event Logs



## Chapter 13 - Performing Diagnostics

## Appendix A -Aironet 340 Series Bridge Specifications



## Appendix B -Console Menu Tree

## Appendix C -SNMP Variables

## Appendix D - Cisco Technical Support

## Appendix E -Regulatory Information

## About the User's Guide

This manual covers the installation, configuration, control, and maintenance of your Aironet 340 Series Bridge.

Please read **Chapter 1** – Installing the Aironet 340 Series Bridge before attempting to install or use the hardware and software described in this manual.

The user's guide is arranged as follows:

*Chapter 1 – Installing the Aironet 340 Series Bridge* – Describes the physical installation of the Aironet 340 Series Bridge.

*Chapter 2 – Accessing the Console System* – Introduces you to the Console Port and shows you how to set up and configure the Console Port parameters.

*Chapter 3 – Before You Begin* – Provides you with an overview of the Configuration Menu and how to save and restore your configurations.

*Chapter 4 – Configuring the Radio Network* – Contains detailed procedures for configuring your Radio Network.

*Chapter 5 – Configuring the Ethernet Port* – Contains detailed procedures for configuring the Ethernet port.

*Chapter 6 – Setting Network Identifiers* – Outlines the procedures for setting the Aironet 340 Series Bridge's Network Identifiers.

*Chapter 7 – Configuring Mobile IP* – Describes how to configure the Aironet 340 Series Bridge for use with the Mobile IP Protocol.

*Chapter 8 – Using the Spanning-Tree Protocol* – Describes how to configure the Aironet 340 Series Bridge for use with the Spanning Tree Protocol.

*Chapter 9 – Viewing Statistics* – Describes how to use the Statistics Menu to monitor the performance of the Aironet 340 Series Bridge.

*Chapter 10 – Setting Up the Association Table* – Provides you with an introduction to the association process and detailed procedures for setting up the Aironet 340 Series Bridge's Association Table.

*Chapter 11 – Using Filters* – Describes how to control the forwarding of multicast messages.

*Chapter 12 – Setting Up Event Logs* – Outlines the procedures for setting up Event Logs and lists the common error log messages received on the Aironet 340 Series Bridge.

*Chapter 13 – Performing Diagnostics* – Provides you with detailed procedures for restarting your unit, returning to your default configuration, and loading new firmware versions.

*Appendix A – Aironet 340 Series Bridge Specifications* – Details the Aironet 340 Series Bridge radio and physical specifications.

*Appendix B – Console Menu Tree* – Provides you with a listing of all menus, sub-menus, and options contained in the Console Port.

*Appendix C – SNMP Variables* – Lists the SNMP variables supported by the Aironet 340 Series Bridge.

*Appendix D – Cisco Technical Support* – Describes how to contact Cisco for technical support.

*Appendix E – Regulatory Information* – Provides the FCC, DOC, and ETSI regulatory statements for the Aironet 340 Series Bridge.

## *Typographical Conventions*

When reading the user's guide, it's important to understand the symbol and formatting conventions used in the documentation. The following symbols and formatting are used in the manual.

| Convention | Type of Information |
|---|---|
|  | Indicates a note which contains important information set off from the normal text. |
|  | A caution message that appears before procedures which, if not observed, could result in loss of data or damage to the equipment. |
| **Bold** type | An action you must perform such as type or select. |
| `Monospaced font` | Information and menus that are visible on the Console Port screens. |

# Welcome to the Aironet 340 Series Bridge

The Aironet 340 Series Bridge allows the connections of two or more remote Ethernet LAN's into a single virtual LAN. Workstations on each of the remote LAN's may communicate with each other as though they were on the same physical LAN. The Aironet 340 Series Bridge can also function as a Radio Access Point and provide transparent, wireless data communications between the wired LAN (and/or within the Radio Network) and fixed, portable or mobile devices equipped with a wireless adapter employing the same modulation.

## *Data Transparency and Protocols*

The Aironet 340 Series Bridge transports data packets transparently as they move through the Wireless Infrastructure.

The bridge is also protocol independent for all packets, except those either addressed specifically to the bridge or sent as multicast address packets.

Depending on the address, packets will be processed as follows:

- All packets, except those either addressed specifically to the bridge or sent as multicast address packets, will be processed without examining the contents of the packet and without regard to the protocol used.
- Packets addressed specifically to the bridge will be examined by looking at the protocol header. If the protocol is recognized, the packet will be processed.
- Multicast address packets will also be examined by looking at the protocol header, but will be processed whether the protocol is recognized or not.

- If protocol filtering is enabled then the appropriate parts of the packet will be examined.

## *Ethernet Compatibility*

The Aironet 340 Series Bridge can attach directly to 10Base2 (Thinnet), 10Base5 (Thicknet) or 10BaseT (Twisted Pair) Ethernet LAN segments. These segments must conform to IEEE 802.3 or Ethernet Blue Book specifications.

If the existing infrastructure to which the bridge is to be attached is not Ethernet-based, an Ethernet segment can be added by installing an Ethernet Network Interface Card (NIC) in the File Server or by adding a third-party bridge.

The bridge appears as an Ethernet node and performs a routing function by moving packets from the wired LAN to remote workstations (personal computers, laptops and hand held computing devices) on the Wireless Infrastructure.

## *Protocols Supported*

Protocols supported:

- TCP/IP based protocol products
- SNMP Protocol – The resident agent is compliant with the MIB-I and MIB-II standards, TCP/IP based internets, as well as a custom MIB for specialized control of the system.

## *Radio Characteristics*

The Aironet 340 Series Bridge uses a radio modulation technique known as Direct Sequence Spread Spectrum transmission (DSSS). It combines high data throughput with excellent immunity to interference. The bridge operates in the 2.4 GHz license-free Industrial Scientific and Medical (ISM) band. Data is transmitted over a half-duplex radio channel operating at up to 11 Megabits per second (Mbps).

## Radio Ranges

The following section provides general guidelines on factors that influence infrastructure performance.

### Site Survey

Because of differences in component configuration, placement, and physical environment, every infrastructure application is a unique installation. Before installing the system, users should perform a site survey in order to determine the optimum utilization of networking components and to maximize range, coverage and infrastructure performance.

Here are some operating and environmental conditions that need to be considered:

- **Data Rates.** Sensitivity and range are inversely proportional to data bit rates. The maximum radio range is achieved at the lowest workable data rate. There will be a decrease in receiver threshold as the radio data rate increases.
- **Antenna Type and Placement.** Proper antenna configuration is a critical factor in maximizing radio range. As a general guide, range increases in proportion to antenna height.

For a detailed explanation of antenna types and configurations along with guidelines on selecting antennas for specific environments, see the Aironet Antenna Guide, document number 710-003725.

- **Physical Environments.** Clear or open areas provide better radio range than closed or filled areas. Also, the less cluttered the work environment, the greater the range.
- **Obstructions.** A physical obstruction such as shelving or a pillar can hinder the performance of the bridge. Avoid locating the computing device and antenna in a location where there is a barrier between the sending and receiving antennas.
- **Building Materials.** Radio penetration is greatly influenced by the building material used in construction. For example, drywall construction allows greater range than concrete blocks.

### *Line of Site*

A clear line of sight must be maintained between wireless bridge antennas. Any obstructions may impede the performance or prohibit the ability of the wireless bridge to transmit and receive data. Directional antennas should be placed at both ends at appropriate elevation with maximum path clearance.

## *Security Features*

The Aironet 340 Series Bridge employs Spread Spectrum Technology, previously developed for military “anti-jamming” and “low probability of intercept” radio systems.

The Aironet 340 Series Bridge must be set to the same System Identifier (SSID) as all other Aironet devices on the wireless infrastructure. Units with a different SSID will not be able to directly communicate with each other.

## *Terminology*

When configuring your system, and when reading this manual, keep in mind the following terminology:

**Infrastructure** – The wireless infrastructure is the communications system that combines Aironet bridges, mobile nodes and fixed nodes. Aironet bridges within the infrastructure can be either root units, which are physically wired to the LAN backbone, or can act as wireless repeaters. Other RF enabled devices serve as fixed nodes or mobile nodes.

**Root Unit** – The root unit is an Aironet bridge that is located at the top, or starting point, of a wireless infrastructure. The root bridge is usually connected to main wired backbone LAN. Since the radio traffic from the other bridges LANs will pass through this unit, the root unit is usually connected to the LAN which originates or receives the most traffic

**Repeater** – A repeater is an Aironet bridge that establishes a connection to the root bridge or another repeater bridge to make the wired LAN to which it is connected part of the bridged LAN.

**End Node** – A radio node that is located at the end of the network tree.

**Parent/Child Node** – Refers to the relationships between nodes in the wireless infrastructure. The complete set of relationships is sometimes described as a network tree. For example, the Aironet bridge (at the top of the tree) would be the parent of the end nodes. Conversely, the end nodes would be the children of the Aironet bridge.

**Association** – Each root unit or repeater in the infrastructure contains an association table that controls the routing of packets between the bridge and the wireless infrastructure. The association table maintains entries for all the nodes situated below the Aironet bridge on the infrastructure including repeaters and radio nodes.

**Power Saving Protocol (PSP) and Non-Power Saving Protocol** – The Power Saving Protocol allows computers (usually portable computers) to power up only part of the time to conserve energy. If a radio node is using the Power Saving Protocol to communicate with the infrastructure, the Aironet bridge must be aware of this mode and implement additional features such as message store and forward.

## *Bridge System Configurations*

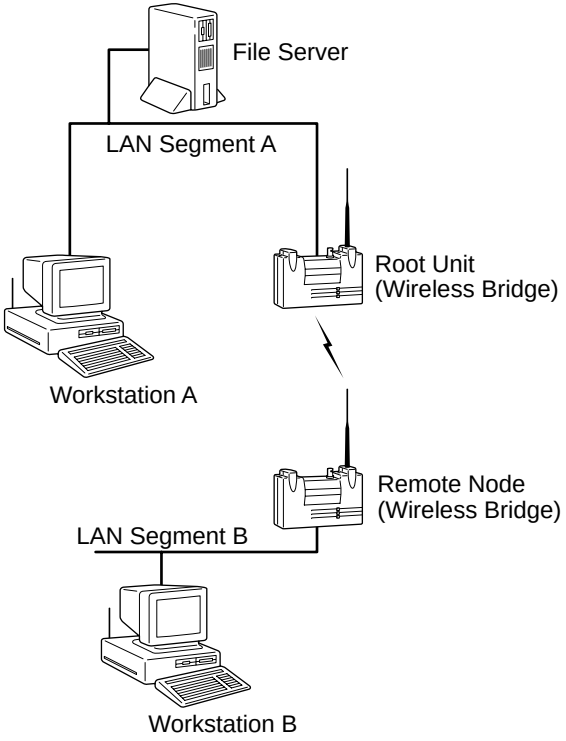
The Aironet 340 Series Bridge can be used in a variety of infrastructure configurations. How you configure your infrastructure will determine the size of the microcell, which is the area a single bridge will provide with RF coverage. You can extend the RF coverage area by creating multiple microcells on a LAN.

Examples of some common system configurations are shown on the pages that follow, along with a brief description of each.

### *Point-to-Point Wireless Bridge*

The Point-to-Point Wireless Bridge Configuration uses two units to bridge two individual LANs. Packets are sent between the file server and Workstation B through the wireless bridge units (root unit and remote node) over the radio link. Data packets sent from the file server to Workstation A go through the wired LAN segment and do not go across the wireless radio link.

**Figure 0.1 - Point-to-Point Wireless Bridge**



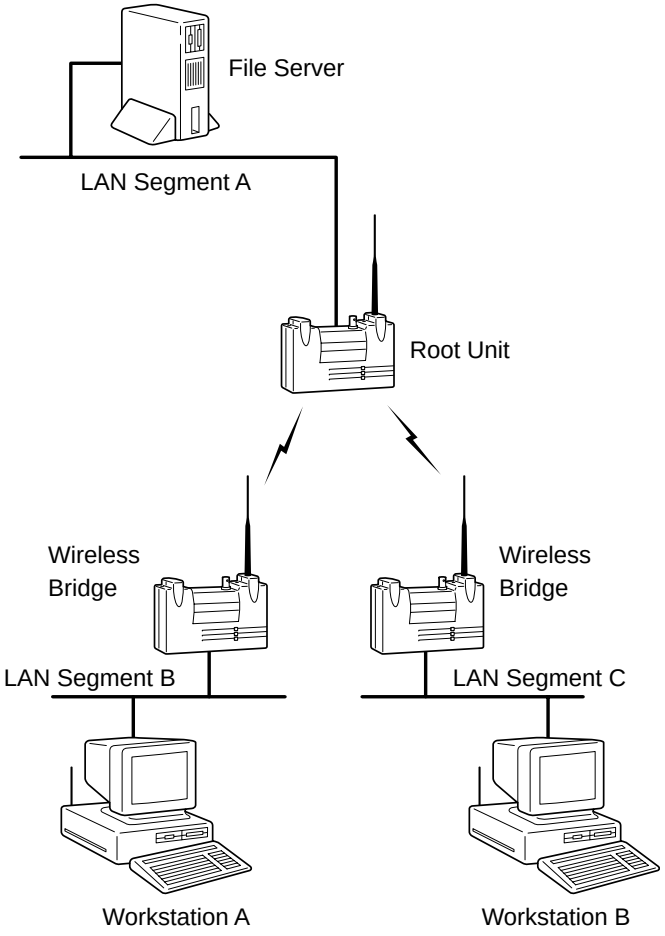

## *Point-to-Multipoint Wireless Bridge*

When connecting three or more LANs (usually in different buildings), each building requires an Aironet wireless bridge and antenna. This is called a Multipoint Wireless Bridge Configuration. One wireless bridge is designated as the central site. Its antenna is configured to transmit and receive signals from the wireless bridges at the other sites. Generally, the central site is equipped with an omni-directional antenna that provides radio signal coverage in all directions. The other wireless bridges are typically served by directional antennas that direct radio signals toward the central site.

Under a Multipoint Wireless Bridge Configuration, workstations on any of the LANs can communicate with other workstations or with any workstations on the remote LANs.

The following example shows an example of a Point-to-Multipoint Configuration. Packets sent between Workstation A and Workstation B are forwarded by their respective wireless bridges to the root unit. Then the root unit forwards these packets to the appropriate wireless bridge for routing to the workstations. Packets sent between the file server and the remote workstations are routed through the root unit and the appropriate wireless bridge.

**Figure 0.2 - Point-to-Multipoint Wireless Bridge**

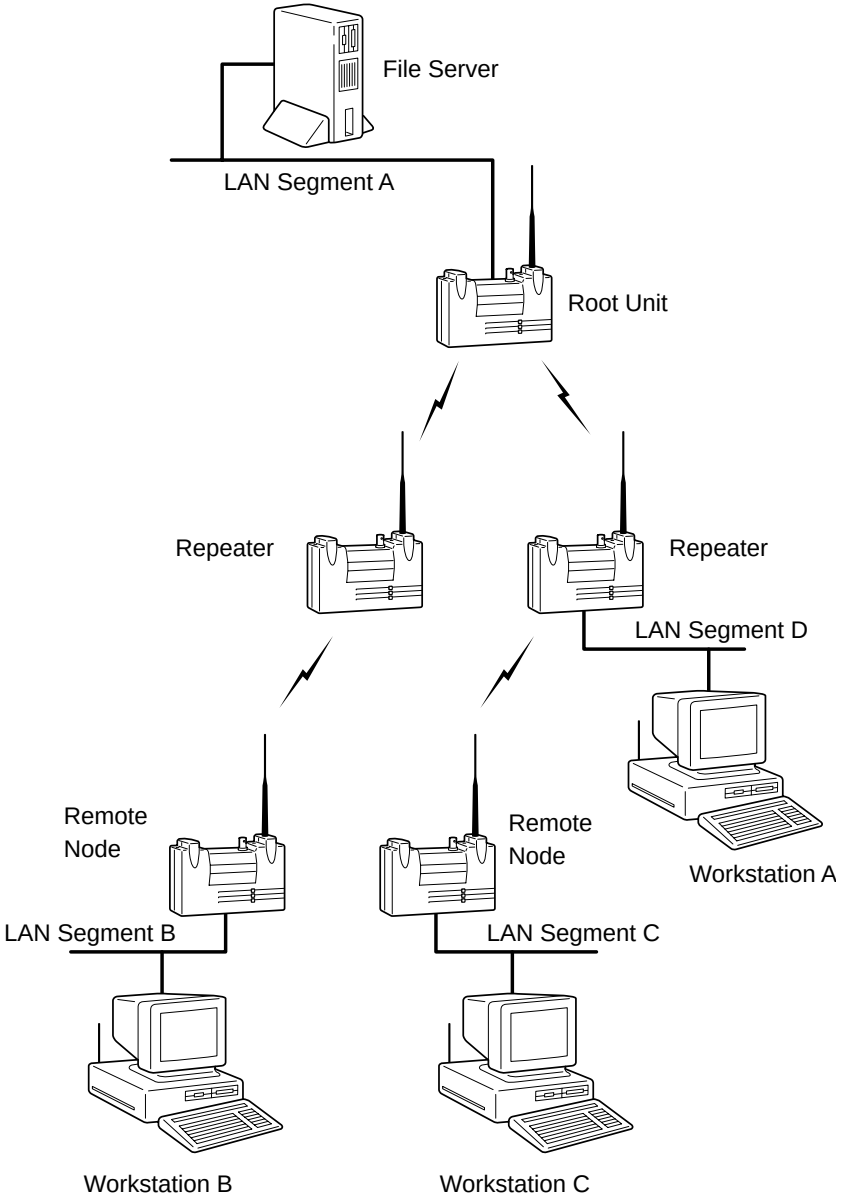



## *Infrastructure Extension with Repeaters*

Wireless bridges can be configured as repeaters to extend the range of a wireless network beyond that of a single radio hop. Repeaters can

operate as either stand-alone units or have LAN connections.

**Figure 0.3 - Infrastructure Extension with Repeaters**



# Part 1 - Getting Started

1

CHAPTER 1

# Installing the Aironet 340 Series Bridge

This chapter describes the procedures for installing the Aironet 340 Series Bridge.

Here's what you'll find in this chapter:

- Before You Start
- Installation
- Installing the Antennas
- Installing the Console Port Cable
- Installing the Ethernet Connection
- Attaching the AC/DC Power Pack and Powering On the Aironet 340 Series Bridge
- Viewing the Indicator Displays

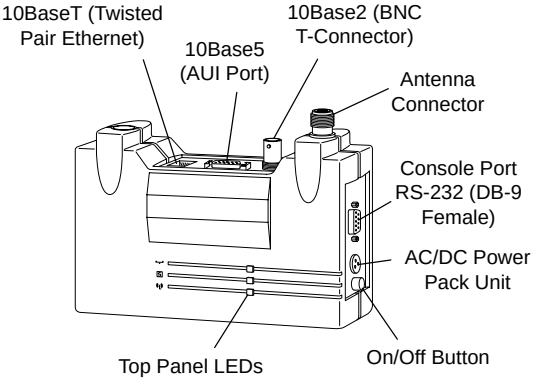
# Before You Start

After unpacking the system, make sure the following items are present and in good condition:

- Aironet 340 Series Bridge
- Power Pack. The power pack will be either 120VAC/60 Hz or 90-264VAC/47-63Hz to 12-18VDC, whichever is appropriate for country of use.
- Lightning Arrestor (Bridge Package option)
- Mounting Kit (Bridge Package option)
- Low loss antenna cable (Bridge Package option)
- Appropriate directional antenna (Bridge Package option)

If any item is damaged or missing, contact your Aironet supplier. Save all shipping and packing material in order to repack the unit should service be required.

**Figure 1.1 - Overview of the Aironet 340 Series Bridge**



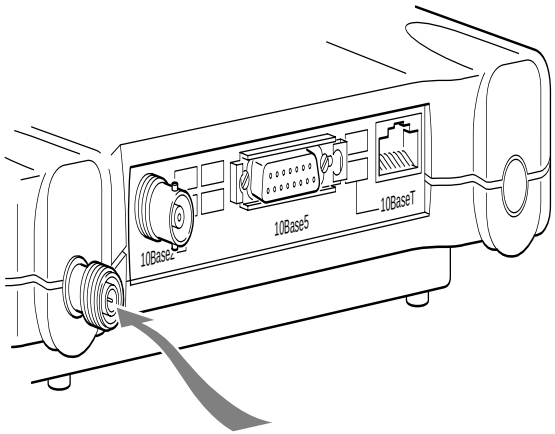
# Installation

## *Installing the Antennas*

Before installing your bridge system, we recommend that you test the bridge using the 2.2 dBi dipole antenna included in your package. Once testing is completed, install your wireless bridge for use with the appropriate antenna for your application using the following the instructions.

1. With the unit powered off, attach the lightening arrestor to the antenna connector.

**Figure 1.2 - Attaching the Antenna**




**NOTE:** Do not over-tighten; finger tight is sufficient. Position the antenna vertically for best omni-directional signal reception.

2. Connect the lightning arrestor to one end of the low loss antenna cable.

**NOTE:** The lightning arrestor should be connected to the antenna connector on the wireless bridge. The lightning arrestor is added to provide surge protection to the bridge in the event of voltage surges as a result of a lightning strike.

3. Connect the antenna to the other end of the low loss antenna cable. Mount the bridge antenna at an appropriate elevation to ensure maximum path clearance and line of sight considerations.

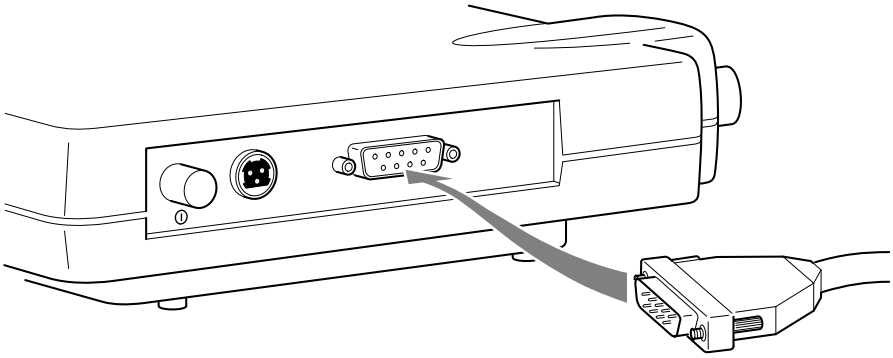
**NOTE:** Due to FCC and DOC Regulations, the antenna connectors on the Aironet 340 Series Bridge are of reverse polarity to the standard TNC connectors.

## *Installing the Console Port Cable*

1. Attach the Console Port cable to the Serial Port. Attach the other cable end to the Serial Port on a terminal or a PC running a terminal emulation program. Use a 9-pin male to 9-pin female straight through cable (**Figure 1.3**).

**NOTE:** This connection is required for setting up initial configuration information. After configuration is completed, this cable may be removed until additional configuration is required via the Serial Port.

**Figure 1.3 - Console Port Connection**



2. Set the terminal to **9600** Baud, **No**-Parity, **8** data bits, **1** Stop bit, and ANSI compatible.

## *Installing the Ethernet Connection*

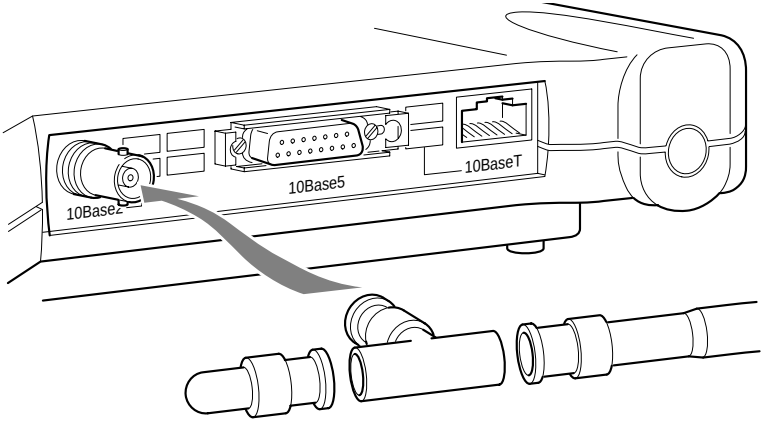
The Aironet 340 Series Bridge supports three connection types:

- 10Base2 (Thinnet)
- 10Base5 (Thicknet) AUI connector
- 10BaseT (Twisted Pair)

➔ **To Attach 10Base2 (Thinnet) Cabling:**

1. Make sure the unit is powered off.
2. Attach the Thinnet cabling to each end of a BNC T-connector, if applicable.
3. Attach the T-connector to the 10Base2 BNC (**Figure 1.4**). If the unit is at the end of the Ethernet cable, a 50-Ohm terminator must be installed on the open end of the T-connector.

**Figure 1.4 - Attaching 10Base2 (Thinnet) Cabling**

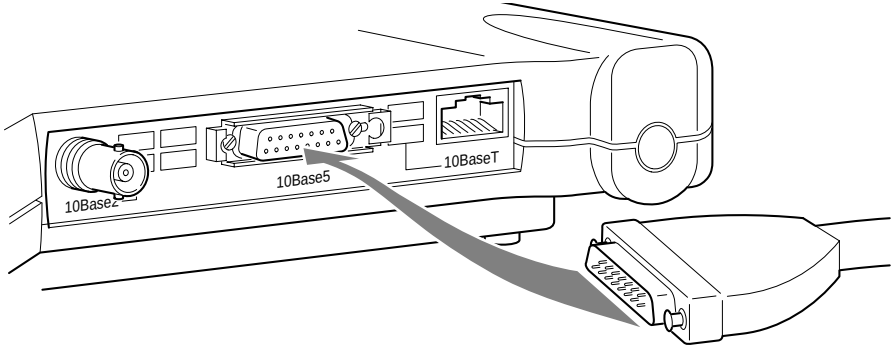



***CAUTION:*** Removing a terminator to install extra cable, or breaking an existing cable to install a T-connector, will cause a disruption in Ethernet traffic. Consult with your LAN administrator before you change any Ethernet cabling connections.

➔ **To Attach the 10Base5 (Thicknet) Cabling:**

1. Make sure the unit is powered off.
2. Attach the transceiver connector to the 10Base5 AUI port as shown in **Figure 1.5**.
3. Slide the locking mechanism in place.
4. Attach the other end of the transceiver drop cabling to an external transceiver.

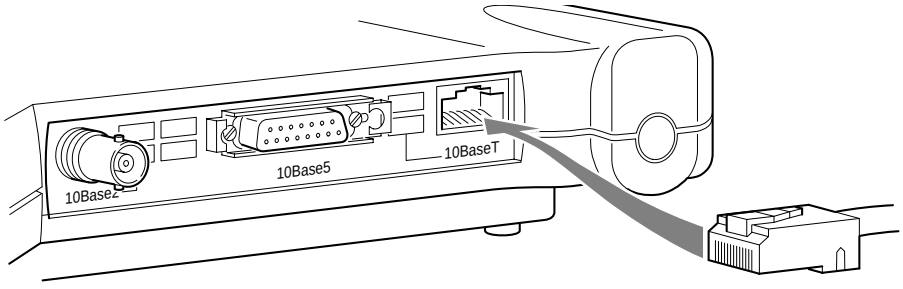
**Figure 1.5 - Attaching 10Base5 (Thicknet) Cabling**




➔ **To Attach the 10BaseT (Twisted Pair) cabling:**

1. Make sure the unit is powered off.
2. Plug the RJ-45 connector into the 10BaseT (Twisted Pair) port as shown in **Figure 1.6**.
3. Connect the other end of the Twisted Pair cabling to the LAN connection (such as a hub or concentrator).

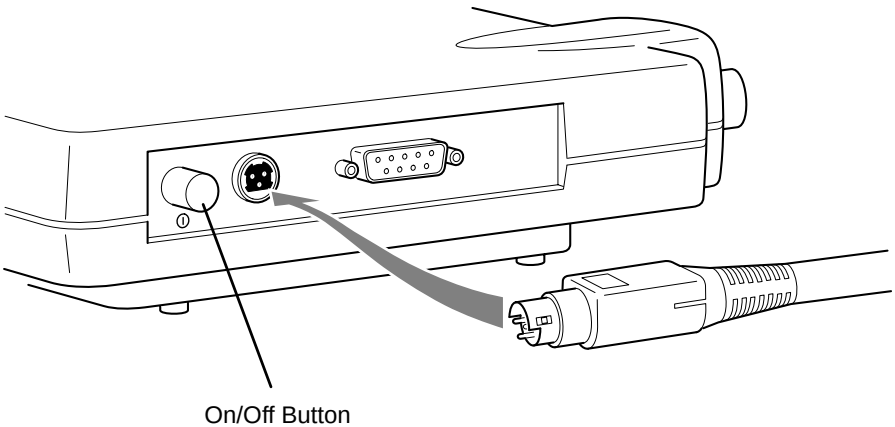
**Figure 1.6 - Attaching 10BaseT (Twisted Pair) Cabling**



# Attaching the AC/DC Power Pack and Powering On the Aironet 340 Series Bridge

1. Insert the small plug on the end of the AC/DC power pack cord into the power port.
2. Plug the AC/DC power pack into an electrical outlet. (120VAC/60 Hz or 90-264VAC as appropriate)
3. Power on the Aironet 340 Series Bridge by pushing the On/Off button.

**Figure 1.7 - AC to DC Power Pack Connections and On/Off Button**




When power is initially applied to the bridge, all three indicators will flash in sequence to test the functionality of the indicators.

# Viewing the Indicator Displays

## *Top Panel Indicators*

The indicators are a set of displays located on the top panel of the Aironet 340 Series Bridge.

- **Ethernet Indicator** – Used to indicate infrastructure traffic activity. The light is normally off, but will flash green whenever a packet is received or transmitted over the Ethernet interface.
- **Status Indicator** – Shows solid green when the bridge has accepted a radio association.
- **Radio Indicator** – Used to indicate radio traffic activity. The light is normally off, but will flash green whenever a packet is received or transmitted over the radio.

When the Aironet 340 Series Bridge is initially powered up, all three displays will flash amber, red and then green, in sequence. If a power-on test fails, the status indicator will go solid red and the unit will stop functioning. See **Table 1.1** for a detailed explanation of the Top Panel indicators.

**Figure 1.8 - Top Panel Indicators**

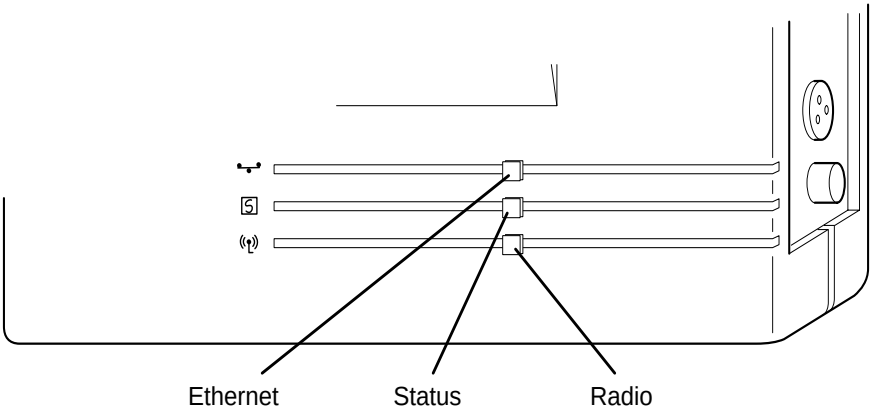


**Table 1.1 - Top Panel Indicator Description**

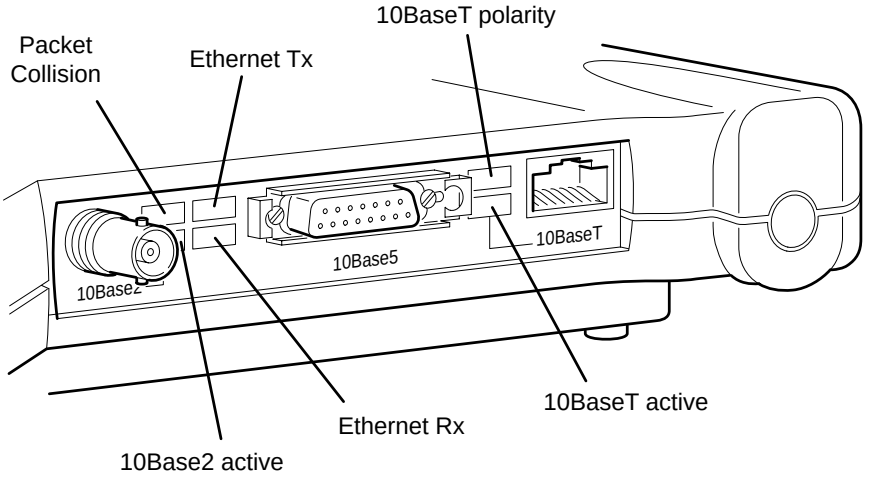
| Type | Indicator Display | | | Description |
|---|---|---|---|---|
| | Ethernet | Status | Radio | |
| Nonassociated Node | | Blinking Green | | No nodes associated |
| Operational | | Green | | One or more nodes associated |
| | | Green | Blinking Green | Transmitting/Receiving Radio packets |
| | Blinking Green | Green | | Transmitting/Receiving packets |
| Error/Warning | | Green | Blinking Amber | Maximum retries/buffer full occurred on radio |
| | Blinking Amber | Green | | Transmit/Receive errors |
| | | Blinking Amber | | General warning, check the logs |
| Failure | Red | Red | Red | Software failure |
| Firmware Upgrade | | Red | | Flashing the firmware |

## *Back Panel Indicators*

The back panel indicators shown in **Figure 1.9** are:

- **10BaseT polarity**: Solid amber to indicate the 10BaseT polarity is reversed. Check cable connections.
- **10BaseT active**: Solid green to indicate the 10BaseT has been configured as the active port.
- **Ethernet Rx**: Flashes green when an Ethernet packet has been received.
- **Ethernet Tx**: Flashes green when an Ethernet packet has been transmitted.
- **10Base2 active**: Solid green to indicate the 10Base2 has been configured as the active port.
- **Packet Collision**: Flashes amber to indicate a packet collision has occurred.

**Figure 1.9 - Back Panel Indicators**



2

CHAPTER 2

# Accessing the Console System

This chapter describes the methods used to access the Console system of the Aironet 340 Series Bridge. This system contains all commands necessary to configure and monitor the operation of the unit.

Here's what you'll find in this chapter:

- Access Methods
- Using the Console
- Telnet Access
- Web Access
- About the Menus
- Using the Configuration Console Menu
- Monitoring of DTR Signal

# Access Methods

There are many ways in which you may configure and monitor the Aironet 340 Series Bridge. When the unit is powered up, basic configuration must initially be performed by accessing the Console Serial Port. To gain access through the Serial Port, the bridge must be connected to a terminal or a PC running a terminal emulation program. See **Chapter 1** "Installing the Aironet 340 Series Bridge". Set the terminal to **9600** Baud, **No**-Parity, **8** data bits, **1** stop bit, and ANSI compatible.

Once the bridge has been assigned an IP address, you may then access the Console remotely using:

- Telnet protocol from a remote host or PC
- HTML browser, such as Netscape Navigator from a remote host
- Simple Network Management Protocol (SNMP) from a remote network management station

# Using the Console

The Console system is organized as a set of menus. Each selection in a menu list may either take you to a sub-menu or display a command that will configure or display information controlling the unit.

When the bridge is powered up, the main menu will be displayed.

```
                          Main Menu
      Option             Value        Description
1 - Configuration   [ menu  ]    - General configuration
2 - Statistics      [ menu  ]    - Display statistics
3 - Association     [ menu  ]    - Association table maintenance
4 - Filter          [ menu  ]    - Control packet filtering
5 - Logs            [ menu  ]    - Alarm and log control
6 - Diagnostics     [ menu  ]    - Maintenance and testing commands
7 - Privilege       [ write ]    - Set privilege level
8 - Help                         - Introduction

Enter an option number or name
>
```

Each menu contains the following elements:

- **Title Line**: Contains the product name, firmware version and menu name. It also contains the unique name assigned to the unit. See **Chapter 6** "Setting Network Identifiers".
- **Option Column**: Displays the menu options and option number.
- **Value Column**: Displays either the value as [menu] or displays the current settings for the option. If the value is [menu], there are additional sub-menus available.
- **Description Column**: Provides a brief description of each option on the menu.
- **Enter an Option Number or Name >**: The cursor prompt used to enter option numbers, names, or commands.

To select an item from the menu you may either enter the number displayed beside the selection, in which case you are immediately taken to the selection, or you may type the name listed in the option column followed by a carriage return. If you use the name method, you only need to enter enough characters to make the name unique from the other selection names in the menu.

When you are entering names or command information you may edit the selection by using the **BACKSPACE** character to delete a single character or the **DELETE** character to delete the entire line.

## *Sub-Menus*

If the selection you chose is a sub-menu, the new menu will be displayed. You may now either choose a selection from this menu or return to the previous menu by pressing the **ESCAPE** key**.** If you want to return to the Main Menu, type the **equal key (=)** at the menu prompt.

## *Commands and Information*

If your selection is a command, you may be prompted for information before it executes. Information may be one of the following types:

- **Token**: A list of one or more fixed strings. To select a particular token, you need only enter enough of the starting characters of the token to allow it to be uniquely identified from the characters of the other tokens in the list.

  ```
  Enter one of [off, readonly, write] : w
  ```

  You would need only enter: “o”, “r”, or “w” followed by a carriage return.

- **String**: An arbitrary amount of characters. The prompt will indicate the allowable size range of the string.

  ```
  Enter a name of from 1 to 10 characters: “abc def”
  ```

  If the string contains a space, enclose the string in quotation marks. If you wish to enter an empty string, use two quotation marks with nothing between them.

- **Integers**: A decimal integer. The prompt will indicate the range of allowed values.

  ```
  Enter a size between 1 and 100 : 99
  ```

  hexadecimal integer – a number specified in hexadecimal using the characters 0-9 and a-f or A-F.

  ```
  Enter a hex number between 1h and ffh : 1a
  ```

- **Network address**: An infrastructure or MAC level address of 12 characters or less. Omit leading zeros when entering an address.

  ```
  Enter the remote network address : 4096123456
  ```

- **IP address**: An internet address in the form of 4 numbers from 0-255 separated by dots (.). Leading zeros in any of the numbers may be omitted.

  ```
  Enter an IP address : 192.200.1.50
  ```

Once all information has been entered the command will execute. If the information entered changed a configuration item, the new value will be displayed in the menus.

Some configuration commands only allow the choice between two fixed values. When the menu item is selected, the opposite value to the current value is chosen. For example, if the configuration item is only a selection between on and off, and the current value is on, then selecting the menu option will select the off value.

Some commands which have a severe effect on the operation of the unit (such as the restart command) and will prompt to be sure you want to execute the command.

```
Are you sure [y/n] :
```

If you enter anything other than a “y” or a “Y” the command will not be executed.

If you are being prompted for information, you may cancel the command and return to the menu by typing **ESCAPE**.

## *Commands That Display Information*

There are several types of commands that display information to the operator. All displays end with a prompt before returning back to the menus. If nothing is entered at the prompt for 10 seconds, the display will automatically refresh.

- Single page non-statistical displays end with the following prompt.

  ```
  Enter space to re-display, q[uit] :
  ```

  Any character other than **space** will cause the display to exit.

- Single page statistical displays end with the following prompt.

  ```
  Enter space to re-display, C[lear stats], q[uit] :
  ```

  Entering a “C” (capital) will reset all statistics to zero.

- Multiple page table displays end with the following prompt.

  ```
  Enter space to redisplay, f[irst], n[ext], p[revi-
  ous], q[uit] :
  ```

  Parts of the prompt may or may not be present depending on the display. If you are not at the first page of the display, you may enter “f” to return to the first page or “p” to return to the previous page. If you are not at the last page you may enter “n” to go to the next page.

### *Command Line Mode*

Another way to move within the Console is to enter commands directly from the Main Menu. Commands allow you to bypass the menu system and go directly to any level sub-menu or option. Enter the list of sub-menus, command names, and information separated by space characters.

**Example 1**: To access the Radio Configuration Menu (located two sub-menus down):

1. At the Main Menu prompt type:

   ```
   configuration radio
   ```

2. Press **ENTER**. The Radio Configuration Menu appears.

**Example 2**: To access the packet size option from the Radio Link Test Menu (located three sub-menus down):

1. At the Main Menu prompt type:

   ```
   configuration radio linktest size 25
   ```

2. Press **ENTER** and the Main Menu is re-displayed.

# Telnet Access

Once the Aironet 340 Series Bridge has been assigned an IP address and connected to the infrastructure, you may connect to the Console system from a remote PC or host by executing the telnet command.

Once the connection has been made, the Main Menu will appear. The menus function in the same way for both telnet access and Serial Port connections.

While a telnet session is in progress, you may not use the Console Port to gain access to the menus. If any characters are entered, the following message is printed identifying the location of the connection.

```
Console taken over by remote operator at 192.200.1.1
<use BREAK to end>
```

If you enter a break sequence, the remote operator will be disconnected and control of the Console is returned to the Console Port.

You may disable telnet access to the bridge with a menu configuration command.

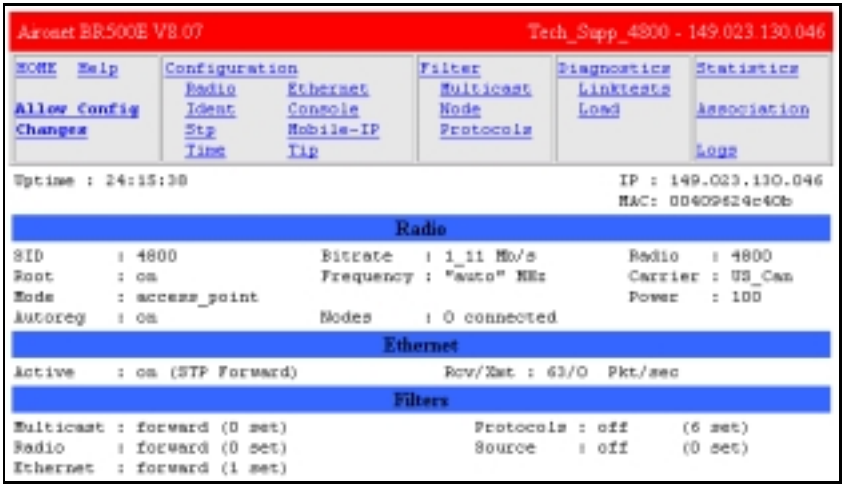
**NOTE:** If you are leaving telnet enabled, make sure you set passwords to secure the Console. See “Enabling Linemode (Linemode)”.

# Web Access

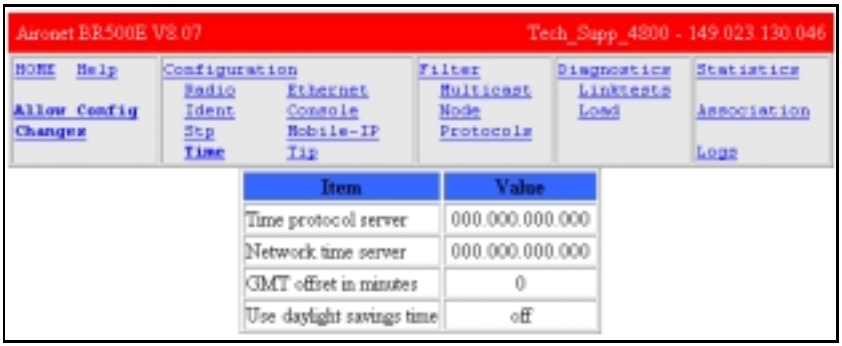
The Aironet 340 Series Bridge supports access to the Console system through the use of an HTML browser. To start a connection use:

```
http://ip address of Aironet 340 Series Bridge/
```

The page displayed will show the general status of the unit:

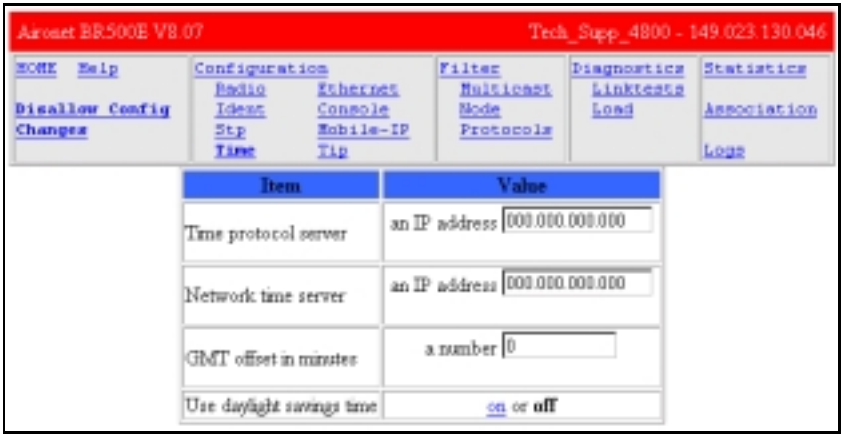


The top section of the each page contains a set of links to the various sub-pages that allow you to configure and display the status of the unit. The following is a sample configuration page




At the top left there is a “HOME” link which always returned to the main page.

By default, the web pages to display so as to not allow any changes to the configuration of the unit. This is done to try and prevent any inadvertent mouse clicks from changing the configuration. To change a configuration item you must first click on the “Allow Config Changes” link in the top left corner. The page will be re-displayed in a form that allows the changes. Once the changes have been completed you should click on the “Disallow Config Changes” link to re-protect the configuartion.



Some configuration items are displayed as a list of fixed choices. The currently active choice is displayed in bold and cannot be selected. The other choices are displayed as links that may be activated by clicking on them.

Other configuration items require the entry of some text. Enter the new value in the text box and then hit “Enter” to send the change to the AP for processing.

For those commands that display pages of information, the prompts function the same as those on the Console Port, except instead of having to type characters to select the different options, the option is a hyper-link.

You may disable web access to the bridge with a menu configuration command.

**NOTE:** If you are leaving web access enabled, make sure that you set passwords to secure the Console. See “Enabling Linemode (Linemode)”.

# About the Menus

Perform the following general functions using menus:

- **Configuration**: Allows you to configure Ethernet and Radio Parameters and establish Network Identifications. See **Chapters 3-6**.
- **Statistics**: View a variety of statistical information such as transmit and receive data throughput, Ethernet and radio errors, and the general status of the Aironet 340 Series Bridge. See **Chapter 9** "Viewing Statistics".
- **Association Table**: A table that contains the addresses of all radio nodes associated below the Aironet 340 Series Bridge on the infrastructure. You may use the association table to display, add and remove static entries, and allow automatic additions to the table. See **Chapter 10** "Setting Up the Association Table".
- **Filter**: Controls packet filtering. The filter menu allows you to control forwarding of multicast messages by blocking those multicast addresses and protocols that are not used on the radio network. See **Chapter 11** "Using Filters".
- **Logs**: Keeps a record of all events and alarms that occur on the unit. With the Logs Menu, you can view and/or print a history of all log entries, set alarm levels, and determine the type of logs you want to save. See **Chapter 12** "Setting Up Event Logs".
- **Diagnostics**: Allows you to run link tests between the Aironet 340 Series Bridge and other infrastructure nodes to test the quality of the radio link. Use the Diagnostics function to load new code versions of the bridge's firmware. See **Chapter 13** "Performing Diagnostics".
- **Privilege**: Allows you to set privilege levels and passwords to restrict access to the Console Port's menus and functions.
- **Help**: A brief help screen outlining the procedures for accessing menus and entering commands.

# Using the Configuration Console Menu

The Console system is configured using the Configuration Console Menu shown below. To access this menu, select **Configuration** from the Main Menu then select **Console** from the Configuration Menu.

```
                Configuration Console Menu
     Option           Value         Description
1  - Rpassword                   - Set readonly privilege password
2  - Wpassword                   - Set write privilege password
3  - Remote        [   on   ]    - Allow remote operators
4  - Telnet        [   on   ]    - Allow telnet connections
5  - Http          [   on   ]    - Allow http connections
6  - Display                     - Display the remote operator list
7  - Add                         - Add an operator host
8  - Delete                      - Remove an operator host
9  - Communities   [  menu  ]    - SNMP community properties
01 - Type          [  ansi  ]    - Terminal type
02 - Port          [  menu  ]    - Serial port set-up
03 - Linemode      [  off   ]    - Console expects complete lines

Enter an option number or name, “=” main menu, <ESC> previous menu
>_
```

### *Setting Privilege Levels and Passwords (Rpassword, Wpassword)*

You can restrict access to the menus by setting privilege levels and passwords. Privilege levels are set from the Main Menu. Passwords are set from the Configuration Console Menu.

There are three privilege levels:

- **Logged Out Level (Off)**: Access denied to all sub-menus. Users are only allowed access to the *privilege* and *help* options of the Main Menu.
- **Read-Only Level (Readonly)**: Read-only privileges for all sub-menus. Only those commands that do not modify the configuration may be used.
- **Read-Write Level (Write)**: Allows users complete read and write access to all sub-menus and options.

Keep in mind the following when setting Privilege Levels and Passwords:

- Only Read-Only and Read-Write privilege levels can be password protected.
- You can always go from a higher privilege level to a lower privilege level without a password. If you try to go to a higher privilege level, you will be required to enter the password.
- Passwords are upper/lower case sensitive.

When Entering the passwords you will be prompted twice to ensure they were entered correctly. The prompting will be done with echoing off.

To change the current privilege level go to the main menu and use the "privilege" function. You will be prompted for the privilege level and its associated password.

**NOTE:** After a privilege level has been assigned, anyone attempting to access that level will be prompted for the password. This allows you to set various privilege levels for individuals, providing them with access to some options, while denying them access to others. Remember passwords are case sensitive.

***CAUTION***: Make sure you write down the passwords you have established and keep them in a safe place. If you forget your password, the unit will have to be returned for factory servicing. Please contact Cisco Technical Support for further instructions.

## *Controlling Telnet and Web Access to the Console*

You may disallow telnet and/or web access to the unit with the "**telnet**" and **"http"** menu items. Setting the value to "off" completely disables this type of access.

## *Controlling SNMP access to the configuration*

All SNMP management stations must include a community name string in all of their requests for information of the unit. This string functions as a password for the snmp access. Community names have either read-only or read/write access associated with them.

The read-only and read/write console passwords automatically are allowed as SNMP community names with the appropriate privilege.

The “Configuration Console Communities” menu may be used to add more community names for use by the network management stations.

```
        Configuration Console Communities Menu
     Option     Value     Description
1 - Display             - Display SNMP communities
2 - Add                 - Add a community
3 - Remove              - Remove a community
4 - Access              - Set community access mode
5 - Remote    [ off ]   - Allow remote NMS to change community info
Enter an option number or name, “=” main menu, <ESC> previous menu
>_
```

You may use the “**Add**”, “**Remove**”, “**Display**” items to update and display the table of allowed community names. A newly added name will by default only be allowed read-only access. To change the privilege level of a community use the “**Access**” item.

The “**Remote**” item is used to control whether a management station with write access is allowed to change the community names.

By default the standard SNMP community names of “public”, “proxy”, “private”, “regional” and “core” are allowed read-only access.

## *Controlling Who Can Access the Console*

You may also control access through the use of a table of remote users. If a user is not in the table any remote access attempt will be terminated. This table controls all remote access to the unit via telnet, http, ftp, snmp, tftp, etc.

A user is identified by either IP address or the MAC address of the host he is using to attempt access. You may use the "**Add**", "**Remove**", "**Display**" items to update and display the table.

If the "**Remote**" item is set to "off" then all remote access is denied regardless of entries in the table. If it is set to "on" and there are no entries in the table then there are no restrictions on who may access the console. If there are entries in the table then only those users whose IP or MAC address match will be allowed access.words are case sensitive.

***CAUTION***: Remember that if you set remote off or make a mistake in the table, the only access to the console will be through the serial port.

## *Setting the Terminal Type (Type)*

The terminal **type** item tells the unit whether the terminal or emulation program you are using supports the ANSI escape sequences. Most modern ones do so you should select the "ansi" option. In this case colors will be added to the displays and the screen cleared to start each new page.

If the terminal does support the ANSI sequences but you do not want the page to be cleared at the start of each display, choose the "color" option.

If the terminal or program does not support the ANSI sequences, you should select "teletype" and no special formatting is done.

## *Setting the Communication Port Parameters (Port)*

Use the *port* option to set the following Aironet 340 Series Bridge port communication parameters: Baud Rate, Data Bits, Parity and Flow. When the *port* option is selected, the Configuration Console Port Menu appears. Any changes are effective immediately.

```
                Configuration Console Port Menu
     Option        Value         Description
1 - Rate      [   9600   ]    - Console baud rate
2 - Bits      [    8     ]    - Bits per character
3 - Parity    [   none   ]    - Console parity
4 - Flow      [ xon/xoff ]    - Flow control type

Enter an option number or name, "=" main menu, <ESC> previous menu
>_
```

- Baud rate selections include 300, 1200, 2400, 9600, 19200, 38400, 56800, or 115200 bits per second.
- Character size selection may be: 7 or 8 bits per character.
- Parity may be: even, odd, or none.
- Flow control selections include:

  **Off**: No flow control. Input or output may be lost if the bridge cannot handle inputs or outputs from your terminal quickly enough.

  **Xon/Xoff**: The bridge will use ASCII Xon/Xoff characters to control the input received from your terminal to prevent input buffer overflow. The unit will also control its output of characters to the terminal.

  **Hardware**: The bridge will use the RTS and CTS lines to control the flow of characters. The bridge sends characters while RTS is high and will assert CTS when the terminal is allowed to send. This mode is used for flow control by passing the Xon/Xoff characters. Make sure the DTR signal is also present on the cable. See "Monitoring of the DTR Signal".

  **Both**: Uses both hardware and Xon/Xoff flow control.

## *Enabling Linemode (Linemode)*

Enable *linemode* when working with telnet and terminal emulators that do not send characters when typed, but rather save them until the operator presses the carriage return at the end of a line.

The Console will not automatically complete any typed commands or information when a space or carriage return is inserted.

To enable linemode:

1. Select **Configuration** on the Main Menu.
2. Select **Linemode** on the Configuration Console Menu.
3. Enter “On” to enable line mode.

**NOTE:** Some telnet programs will automatically invoke linemode by sending the appropriate telnet commands when they connect to the Aironet 340 Series Bridge.

# Monitoring of the DTR Signal

The Aironet 340 Series Bridge monitors the state of the Data Terminal Ready (DTR) signal. This signal is used to indicate the presence or absence of a DTE device connected to the Console Port.

If the state of the signal changes (up or down) the following actions will occur (unless a telnet session is in progress):

- Any currently executing command or display will be terminated
- Current menu will be returned to the Main Menu
- Console Privilege Menu will be set back to the highest level not requiring a password.

If the Console is configured for hardware flow control and the DTR signal is currently down, all output will be discarded. The bridge would assume flow is off and the Console would eventually lock up.

If the cable used does not have the DTR signal connected it will not change state and no action will be taken.

# Part 2 - Configuration

3

CHAPTER 3

# Before You Begin

This chapter provides a general introduction to the Configuration Menu and describes the procedures for saving and restoring your configurations. See **Chapters 4 - 11** for more information on configurations.

Here's what you'll find in this chapter:

- Viewing the Configuration Menu
- Menu Descriptions
- Saving Configuration Parameters
- Backing up your Configuration
- Restoring your Configuration

# Viewing the Configuration Menu

Once you have completed the installation, the next step is to use the Configuration Menu commands to configure the Aironet 340 Series Bridge.

To access the Configuration Menu, select **Configuration** from the Main Menu.

```
                Configuration Menu
     Option       Value      Description
1 - Radio       [ menu ]   - Radio network parameters
2 - Ethernet    [ menu ]   - Ethernet configuration
3 - Ident       [ menu ]   - Identification information
4 - Console     [ menu ]   - Control console access
5 - Stp         [ menu ]   - Spanning Tree Protocol
6 - Mobile-IP   [ menu ]   - Mobile IP Protocol Configuration
7 - Time        [ menu ]   - Network Time Setup
8 - Dump                   - Dump configuration to console

Enter an option number or name, "=" main menu, <ESC> previous menu
>_
```

### *Menu Descriptions*

**Radio**: Used to set radio network parameters, such as system ID, frequency, and bitrate. See **Chapter 4** "Configuring the Radio Network".

**Ethernet**: Used to set the Ethernet Parameters. See **Chapter 5** "Configuring the Ethernet Port".

**Ident**: Used to set various infrastructure identifiers such as Node Names, Network ID, and Internet Address. See **Chapter 6** "Setting the Network Identifiers".

**Console**: Used to set up the Console Port. See **Chapter 2** "Accessing the Console System".

**STP**: Used to configure the spanning tree protocol. See **Chapter 8** "Using the Spanning Tree Protocol".

**Mobile-IP**: Used to configure the unit as either a home or foreign mobile P agent. See **Chapter 7** "Configuring Mobile IP".

**Time**: Used to configure time server address to set a standard time base for displaying logs and alarms. See **Chapter 6** "Setting the Network Identifiers".

**Dump**: Used to dump the configuration commands to the Console Port. See "Backing up your Configuration (Dump)".

## *Saving Configuration Parameters*

Although there is no explicit save command, your configuration parameters are automatically saved to non-volatile flash memory each time a parameter is set or modified. This will ensure the configuration is maintained during power failures or intentional power downs.

Most configuration settings become effective as soon as the command is executed. Those that do not immediately become effective will be noted in the command information.

## *Backing up your Configuration (Dump)*

Once you have set the configuration parameters for the Aironet 340 Series Bridge, use the *dump* option to dump the configuration commands to the Console Port and save them as an ASCII file on a diskette, using a PC terminal emulation program.

If the non-volatile flash memory should ever become corrupted (and you lose your saved configuration), you can use a communications program to send the configuration commands to the Console Port. The system will automatically restore your configuration based on these commands.

➔ **To Back Up Configurations:**

**NOTE:** Commands may vary depending on the communications program used.

1. In the terminal emulation program, set Save to File to **On**.
2. Select **Configuration** from the Main Menu then select **Dump**. The following message appears:

   ```
   Enter one of [all, non-default, distributable]:
   ```

- **All**: The entire configuration will be displayed.
- **Non-default**: Only the configuration options that are different from the original default settings will be displayed.
- **Distributable**: Only the configuration options that are not considered unique to this unit are displayed. You may use the "diagnostics load distribute" command to send this configuration to other units in the infrastructure.

3. Enter one of [standard, encoded]:
   - **Standard**: The configuration is displayed in normal readable text form.
   - **Encoded**: The configuration is displayed with each configuration command replaced by a unique number. This type of configuration is the best to save since the number will never change over the life of the product. Text may change or move as more items are added to the menus.
4. Enter your configuration command choice.
5. Save the file after the commands have been dumped.
6. Turn Save to File to **Off**.
7. Press any key to clear the screen.

## *Restoring your Configuration*

If your configuration is ever lost or corrupted, you can use restore your configuration using the program's ASCII upload commands.

4

CHAPTER 4

# Configuring the Radio Network

This chapter describes the procedures for configuring the Aironet 340 Series Bridge Radio Network. It describes all of the functions in the “Configuration Radio Menu” and its sub-menus.

Here’s what you’ll find in this chapter:

- Setting up the basic radio configuration
- Setting up encryption
- Setting up the advanced radio configuration
- How to test the radio links

# Overview

When configuring the radio network, all units should be configured while in close proximity to each other. This will allow your units to communicate with other radio nodes on your infrastructure as the units' parameters are set.

Once configuration is complete, the units can then be moved to their permanent location. Tests can be run to check the reliability of the radio links. See "Running a Signal Strength Test (Strength)".

The radio network parameters should be set in the order shown below:

1. Establish a system identifier.
2. Select a rate.
3. Select a frequency.
4. Enable root or repeater mode.
5. Set any extended parameters (optional).

***CAUTION:*** Changing any of the radio parameters after you have completed your configurations will cause the unit to drop all radio connections and restart with the changes you have made. Consequently, there will be a disruption in radio traffic through the unit.

# Using the Configuration Radio Menu

The radio network is configured using the Configuration Radio Menu. To access this menu, select **Configuration** from the Main Menu then select **Radio** from the Configuration Menu.

```
                  Configuration Radio Menu
   Option            Value      Description
1 - Ssid          [ "test" ]   - Service set identification
2 - Root          [   on   ]   - Enable root mode
3 - Rates         [  1_11  ]   - Allowed bit rates in megabits/second
4 - Basic_rates   [   1    ]   - Basic bit rates in megabits/second
5 - Frequency     [ "auto" ]   - Center frequency in MHz
6 - Distance      [   0    ]   - Maximum separation in kilometers
7 - I80211        [  menu  ]   - 802.11 parameters
8 - Linktests     [  menu  ]   - Test the radio link
9 - Extended      [  menu  ]   - Extended parameters

Enter an option number or name, "=" main menu, <ESC> previous menu
>_
```

### *Establishing an SSID (SSID)*

This string functions as a password to join the radio network. Nodes associating to the bridge must supply a matching value, determined by their configurations, or their association requests will be ignored.

### *Enabling Root Mode (Root)*

Use the *root* option to enable or disable root mode.

There may only be one unit serving as the root unit and it is usually connected to the primary backbone infrastructure. Those acting as remote bridges, attached to a secondary backbone and communicating via radio to the root unit, should have their Root Mode set to "Off". The default setting is "On".

### *Selecting the Allowed Data Rates (Rates)*

Use the *rates* option to define the data rate at which the unit is allowed to receive and transmit information. Other units in the radio cell are allowed to transmit data to us at any of these rates at their discretion.

When a repeater associates to a root unit data is usually transmitted between the units at the highest rate that they both support. The units may also downshift to use lower common rates if conditions warrant it.

### *Basic Rates (Basic_rates)*

The basic rates option is set on the root bridge. It is the set of rates that all nodes in the radio cell must support or they will not be allowed to associate.

The lowest basic rate is use to transmit all broadcast and multicast traffic as well as any association control packets. Using the lowest rate helps ensure they will be received by all nodes even at the farthest distances.

The highest basic rate determines the maximum rate at which an acknowledge packet may be transmitted.

### *Selecting Frequency (Frequency)*

The actual frequency allowed depends on the regulatory body that controls the radio spectrum in the location in which the unit is used. If the setting is left as “auto”, the unit will sample all the allowed frequencies when it is first started and try to pick one that is not in use.

This setting is only allowed on the root unit as it is in charge of setting up the radio cell

### *Setting the Distance (Distance)*

Since the radio link between bridges can be quite long, the time it takes for the radio signal to travel between the radios can become significant. This parameter is used to adjust the various timers used in the radio protocol to account for the extra delay.

The parameter is only entered on the root bridge, which will tell all the repeaters. It should be entered as the distance in kilometers of the longest radio link in the set of bridges.

# Using the Configuration Radio IEEE 802.11 Menu

```
              Configuration Radio I80211 Menu
     Option           Value      Description
1 - Beacon          [ 100  ]   - Beacon period in Kusec
2 - Dtim            [  2   ]   - DTIM interval
3 - Extend          [  on  ]   - Allow proprietary extensions
4 - Bcst_ssid       [  on  ]   - Allow broadcast SSID
5 - Rts             [ 2048 ]   - RTS/CTS packet size threshold
6 - Privacy         [ menu ]   - Privacy configuration
7 - Encapsulation   [ menu ]   - Configure packet encapsulation
 Enter an option number or name, "=" main menu, <ESC> previous menu
>_
```

### *Setting the Beacon Period (Beacon)*

The beacon interval is the time (in kilo-microseconds) between transmissions of the IEEE 802.11 beacon packet. The beacon packets are primarily used for radio network synchronization.

A small beacon period means faster response for roaming nodes. The default value is typically adequate.

### *Setting the Forwarding Time Interval (DTIM)*

The DTIM count determines the count of normal beacons between the special DTIM beacons. If there no power saving client nodes in a cell, as is usually the case with bridges, it is not used.

If there are power saving nodes present, the 802.11 protocol defines that all power saving nodes must, at the minimum, wake up to receive the DTIM beacons. If power save nodes are present, the AP will also buffer any multicast packets it receives from the LAN and only transmit them after the DTIM beacon.

Setting the DTIM count low causes the multicasts to be transmitted more frequently, but sets a lower upper limit as to how long a power save node may remain asleep.

### *Adding IEEE 802.11 Management Packet Extensions (Extend)*

If this parameter is enabled, the Aironet 340 Series Bridge will add extensions to some of the IEEE 802.11 management packets. This passes more information to other radio nodes allowing them to associate to the best bridge.

Even with the extensions enabled, other manufacturer's nodes should ignore the extra information. However, if they become confused, this parameter may be disabled.

### *Allowing the Broadcast SSID (Bcst_ssid)*

This option controls whether client nodes will be allowed to associate if they specify the empty or broadcast SSID. Clients that do not know the SSID of the bridge can transmit packets with the broadcast SSID. Any bridges present will respond with a packet showing their SSID. The client will then adopt the SSID and associate.

If you wish to ensure that clients know the SSID beforehand then disable this function.

### *Setting the RF RTS/CTS Parameter (RTS)*

This parameter determines the minimum size transmitted packet that will use the RTS/CTS protocol. The value entered must be in the range of 100 to 2048 bytes.

This protocol is most useful in networks where the mobile nodes may roam far enough so the nodes on one side of the cell cannot hear the transmission of the nodes on the other side of the cell.

When the transmitted packet is large enough, a small packet is sent out (an RTS). The destination node must respond with another small packet (a CTS) before the originator may send the real data packet. A node at the far end of a cell will see the RTS to/from the bridge or the CTS to/from the bridge. The node will know how long to block its transmitter to allow the real packet to be received by the bridge. The RTS and CTS are small and, if lost in a collision, they can be retried more quickly and with less overhead than if the whole packet must be retried.

The downside of using RTS/CTS is that for each data packet you transmit, you must transmit and receive another packet, which will affect throughput.

# Packet Encapsulation (Encapsulation Menu)

```
      Configuration Radio I80211 Encapsulation Menu
    Option      Value      Description
1 - Encap    [ 802.1H ]   - Default encapsulation method
2 - Show                  - Show encapsulation table
3 - Add                   - Add a protocol encapsulation method
4 - Remove                - Remove a protocol encapsulation method
 Enter an option number or name, "=" main menu, <ESC> previous menu
>_
```

The *Encap* option and the related encapsulation table commands of *Show*, *Add* and *Remove* are of concern only when both of the following conditions exist:

- You are assembling a wireless LAN that incorporates non-Aironet equipment.
- The non-Aironet equipment uses a proprietary method of packet encapsulation that is different from the method used by Aironet.

If your wireless LAN consists only of Aironet components, use the default Encap value of 802.1H and disregard the information in following discussion "Packet Encapsulation in Mixed Networks."

## *Packet Encapsulation in Mixed Networks*

Aironet LAN software allows you to assemble a wireless infrastructure using components from different suppliers. When combining equipment from different sources into a wireless LAN, you might need to accommodate different methods of packet addressing and conversion. The complete subject of packet addressing is beyond the scope of this manual, and our purpose here is to provide only basic guidelines and considerations.

To combine a mix of equipment from alternate suppliers into a wireless LAN, you need to know the packet encapsulation methods used by the different suppliers. If you determine that your infrastructure will be mixing packet encapsulation methods, you will first need to determine

your primary method, or standard, and choose that as the default setting with the Encap option. All methods other than the primary, or default, method need to be entered in the Encapsulation Table.

For all Aironet equipment, the defined packet encapsulation standard is 802.1H. The Show, Add and Remove options allow you to manage a table of alternate, non-I802.1H encapsulation methods that might be required to read data packets sent from the other, non-Aironet equipment. The primary alternate to the I802.1H standard is RFC 1042.

On an Ethernet LAN, the data portion of a frame may be in one of two formats: DIX or DSAP/SSAP. The two formats differ both in packet size specifications and in the manner of heading, or starting, the data portion. An 802 wireless LAN requires packets to start with the DSAP/SSAP format and therefore must provide a method of conversion. DSAP/SSAP packet types are easily converted since the header is already in the required style. DIX packet types present more of a problem since there are many different formats and no standard conversion method.

Aironet's 802.1H conversion protocol accommodates both DIX and DSAP/SSAP packet types. In an 802.1H conversion, DIX type packets are prepended with a header that mimics the DSAP/SSAP header. In an Aironet infrastructure, this header style is not used by any wired Ethernet nodes so the remote radio node is always able to accurately reconvert the packet.

# Packet Encryption (Privacy Menu)

```
      Configuration Radio I80211 Privacy Menu
     Option          Value       Description
1 - Encryption    [ off  ]    - Encrypt radio packets
2 - Auth          [ open ]    - Authentication mode
3 - Client        [ open ]    - Client authentication modes allowed
4 - Key                       - Set the keys
5 - Transmit                  - Key number for transmit
 Enter an option number or name, "=" main menu, <ESC> previous menu
>_
```

This menu controls the use of encryption on the data packet transmitted over the air by the radios. The packets are encrypted using the RSA RC4 algorithm using one of up to 4 known keys. Each node in the radio cell must know all the keys in use, but they may select any one to use for their transmitted data.

The "**Key**" option is used to program the encryption keys into the radio. You will be prompted as to which of the four keys you wish to set and then you are prompted twice to enter the key with echoing disabled. Depending on whether the radio is authorized to use 40 bit or 128 bit keys, you must enter either 10 or 26 hexadecimal digits to define the key.

**NOTE:** The keys must match in all nodes in the radio cell and must be entered in the same order.

You do not need to define all four keys as long as the number of keys matches in each radio in the cell.

Use the "**Transmit**" option to tell the radio which of the keys it should use to transmit its packets. Each radio is capable of de-crypting received packets sent with any of the four keys.

If the "**Encryption**" option is set to "off" then no encryption is done and the data is transmitted in the clear. If the value is set to "on" then all transmitted data packets will be encrypted and any un-encrypted received packet will be discards.

The “Encryption” value may also be set to “mixed’. In this mode a root or repeater bridge will accept association from clients that have encryption turned on or off. In this case only data packets between nodes that both support it will be encrypted. Multicast packets will be sent in the clear so that all nodes may see them.

***CAUTION***: We do not recommend the use of “mixed” mode. If a client with encryption enabled sends a multicast packet to its parent, the packet will be encrypted. The parent will then decrypt the packet and re-transmit it in the clear to the cell for the other nodes to see. Seeing a packet in both encrypted and un-encrypted form can greatly aid in breaking a key. This mode is only included for compatibility with other vendors.

The 802.11 protocol specifies a procedure in which a client must authenticate with a parent before it can associate. The “open” method of authentication is essentially a null operation. All clients will be allowed to authenticate. With the “shared key” the parent send the client a challenge text which the client encrypts and sends back to the parent. If the parent can de-crypt it correctly the client is authenticated.

***CAUTION***: With the “shared-key” mode, since a clear text and encrypted version of the same data is transmitted on the air, we again do not recommend its use. It does not really gain you anything, since if the user’s key is wrong the unit will not be able to de-crypt any of his packets and they cannot gain access to the network.

The “**Client**” option determines the authentication mode that the client nodes are allowed to use to associate to the unit. The values allowed are “open”, “shared-key”, or “both”.

The “**Auth**” is used on repeater bridges to determine which authentication mode the unit will use to connect with its parent. The allowed values are “open” or “shared-key”.

# Using the Configuration Radio LinkTests Menu

The options in this menu can be used to determine system performance on individual nodes as well as individual node radio performance.

```
          Configuration Radio Linktests Menu
     Option          Value        Description
1  - Strength                   - Run a signal strength test
2  - Carrier                    - Carrier busy statistics
3  - Multicast                  - Run a multicast echo test
4  - Unicast                    - Run a unicast echo test
5  - Remote                     - Run a remote echo test
6  - Destination   [ any  ]     - Target address
7  - Size          [ 512  ]     - Packet size
8  - Count         [ 100  ]     - Number of packets to send
9  - Rate          [ auto ]     - Data rate
01 - Errors                     - Radio error statistics
02 - Autotest      [ once ]     - Auto echo test
03 - Continuous    [  0   ]     - Repeat echo test once started
 Enter an option number or name, “=” main menu, <ESC> previous menu
>_
```

## *Running a Signal Strength Test (Strength)*

The *strength* option sends a packet once per second to our parent access point and each node in the association table. This packet is echoed back to the Aironet 340 Series Bridge which records and displays the RF signal strength associated with that particular node.

It can be used to quickly check the link to each radio partner or could be monitored while aligning directional antennas between two nodes. As the antennas are moved, the signal strength could be monitored until the maximum value is achieved.

```
                        SIGNAL LEVELS
BRxxxx  00409611d1e5 Strength   In ********************************
                                Out **********************************
(^C to exit)                        |--------------------------------|
```

## *Running a Carrier Busy Test*

The *Carrier* option can be used to determine the amount of activity on each of the available frequencies. Its main use is to pick an unused frequency or to check for the presence of a jammer.

When started, the radio is put in a special mode, in which it will scan through all the allowed frequencies, pause on each one and measure the percentage of the time that the carrier detect line for the radio is busy.

***CAUTION***: Since this test uses a special operating mode all current associations to the unit will be lost during the test.

```
                          CARRIER BUSY / FREQUENCY

     *
     *
     *
     *           *
     *           *
     *           *
     *           *
     * *         *
     * *         *
     * * * * * * * * * * *
     1 1 2 2 3 3 4 4 5 5 6
     2 7 2 7 2 7 2 7 2 7 2

     Highest point = 9% utilization
     Enter space to redisplay, q[uit] :
```

The display is a scaled bar graph with the frequencies along the bottom. The percentage utilization represented by the highest bar is given in the bottom line.

## *Running the Echo Tests (Multicast, Unicast, Remote)*

An echo test consists of sending a number of packet between units. The packets are sent with a proprietary protocol, which the target nodes recognize and will echo back to the test source along with information about how well it received the packets.

The ***multicast*** option is used to test transmission conditions within local radio cells. Packets are sent between the source and destination nodes without any acknowledgments or retries (as multicasts). This test provides a good indication of the raw state of the path to the node since no attempt is made to recover from any radio errors.

```
Testing link to 00409611d1e5 with 100 multicast packets of size 512
Please wait:
GOOD (  9% Lost)        Time   Strength %
                        msec     In   Out
                        ----  ----- -----
       Sent: 100,  Avg:  19     78    85
Lost to Tgt:   8,  Max:  29     85    92
Lost to Src:   1,  Min:  17     71    85
```

The time is displayed in milliseconds. Each packet contains the time it was sent. When a packet is received by the source, the time difference indicates the round trip time. Longer times indicate that the processor's or the radio's bandwidth is full.

The signal strength numbers indicate the strength of the radio signal at the time the packets were received at each end. Signal strength is expressed as a percentage of full power.

The ***unicast*** option can be used to test the path between the Aironet 340 Series Bridge and any other Aironet node in the wired or radio network. The packets are sent with the same error recovery as normal user data so round trip times indicate the infrastructure throughput and congestion.

```
Testing link to 00409611d1e5 with 100 unicast packets of size 512
GOOD (8% Retries)       Time   Strength %        Retries
                        msec     In   Out        In  Out
                        ----  ----- -----      ---- ----
       Sent: 100,  Avg:  25     78    85  Tot:   3   14
Lost to Tgt:   0,  Max:  91     85    92         1    2
Lost to Src:   0,  Min:  21     78    85         0    0
```

If the path to the target node was over the radio, a total number of radio retries necessary to complete the test is also displayed. If the total number of retries is large, there may be problems with the link. Look for

sources of interference.

Use the ***remote*** option to run a multicast link test between a client node associated somewhere in the infrastructure and its parent bridge. You will be prompted for the infrastructure address of the client node. A broadcast request will be made. The bridge with this associated node will run the link test and return the results which will be displayed to the operator locally.

```
   Remote linktest from 00409610d258 to 0040961064de

Sent 100 of 100 512 byte packets, Destination received
90, Source received 90
```

### *The Echo Test Parameters (Destination,Size,Count,Rate)*

The ***destination*** option is used to indicate the target node address for the link test. You may enter an infrastructure address or the string “any”. If you select “any,” the Aironet 340 Series Bridge will direct the test to the first legal address found in the association table, the access point to which the unit is registered. If you enter a network address, it may only be used for the remote or unicast linktests.

The ***size*** and ***count*** options are used to indicate the size and number of packets to be sent. The default values are 100 packets of 512 bytes each. Both the size and the count can be changed. The packet size may be set from 24 to 1500 bytes and the count of the number of packets to transmit may be set from 1 to 999 packets.

Using the ***rate*** option to control the data rate at which the packets are sent. Normally you would leave the setting at auto to allow the radio to perform it’s normal rate shifting algorithm. You might use the actual rate settings to test for the range limits at each of the data rates.

When running the link test, use the highest data bit rate possible to test the reliability of your data bit rate and frequency combination. The more packets you send and the larger the packet size, the more accurate the test.

**NOTE:** Multiple large packets will increase test time.

### *Viewing Errors (Errors)*

The *errors* option is used to view the Radio Error statistics that may have occurred during the link test. See **Chapter 9** “Viewing Statistics”.

### *Continuously Running a Link Test (Continuous)*

The ***continuous*** option is used to continuously repeat the link tests. If the value for the parameter is zero the tests are not repeated; otherwise, the value determines the delay (in seconds) between tests.

### *Setting the Automatic Link Test Mode (Autotest)*

The *autotest* option is used to control the automatic running of a link test whenever a repeater associates to its parent. The test will use the currently configured test parameters which, by default, runs a test to the parent node.

- **Off**: An automatic test is never run.
- **Once**: Only one test is run the first time the unit associates to a parent after powering on.
- **Always**: The test is run each time the unit associates to a parent.

During an automatic link test the three indicators on the unit will turn green in a cyclic pattern to indicate a test is in progress. At the end of the test, the indicators will be set to a solid pattern for 4 seconds to indicate the test results. The particular pattern that will be displayed depends on the percentage of packets lost during the test as shown in Table 4.1

**Table 4.1 - Auto Link Test Display Patterns**

| Radio | Status | Ethernet | % of Packets Lost | Quality |
|---|---|---|---|---|
| Green | Green | Green | 0-2 | Excellent |
| Green | Green | Amber | 3-5 | Very Good |
| Green | Green | Off | 6-25 | Good |
| Green | Amber | Off | 26-50 | Satisfactory |
| Amber | Off | Off | 51-75 | Fair |
| Red | Off | Off | 76-100 | Poor |

The Autotest procedure can be used to help determine the placement of repeater units. For example, at each prospective location, an installer could cycle the power on the unit and watch the indicator displays for the results of the link test. As the test begins to fail, the installer could determine the radio range to the infrastructure and adjust the location accordingly.

# Using the Configuration Radio Extended Menu

The extended radio parameters are not normally modified, but some may have to be changed when certain situations arise.

```
                Configuration Radio Extended Menu
      Option            Value          Description
1   - Bridge_mode    [ bridge_only] - Bridging mode
2   - Parentid       [     any    ] - Parent node Id
3   - Parent_timeout[     off    ] - Time to look for specified parent
4   - Time_retry     [     8     ] - Number of seconds to retry transmit
5   - Count_retry    [      0     ] - Maximum number transmit retries
6   - Refresh        [     100    ] - Refresh rate in 1/10 of seconds
7   - Roaming        [  directed  ] - Type of roaming control packets
8   - Balance        [     off    ] - Load balancing
9   - Diversity      [     off    ] - Enable the diversity antennas
01 - Power           [     20     ] - Transmit power level
02 - Fragment        [    2048    ] - Maximum fragment size
03 - Options                        - Enable radio options
Enter an option number or name, “=” main menu, <ESC> previous menu
>_
```

The Menu will display different options, depending on whether your unit is serving as an infrastructure or a repeater.

### *Setting the Operating Mode (Bridge_mode)*

This setting determines the type of client nodes that are allowed to associate to this unit. If it is set to “bridge_only” then only other bridges are allowed to associate and not any normal client nodes. Setting it to “access_point” allowed any kind of client to associate. Setting the value to “client” on a repeater bridge will not allow any other nodes to associate to this node. Setting a repeater to client mode also reduces some of the radio protocol overhead as this unit does not have to constantly advertise its presence.

### *Selecting a specific parent (Parent_id, Parent_timeout)*

The setting is only available on repeater bridges. Normally a radio node will choose its parent by polling the air waves and choosing the best available unit. If you wish to manually force a particular structure to the

radio cell, usually because of knowledge of traffic patterns, you can use the ***parent_id*** option to select the MAC address of the parent the unit would always try and associate to.

If you set the ***parent_timeout*** option to off the unit will only associate with the specified parent. If you set a value, the unit will poll for the specified parent for the given number of seconds each time it needs to associate. If the parent is not found it will choose the best available parent. If the unit ever sees that the specified parent is present it will switch its association.

## *Setting Retry Transmission Time (Time_Retries, Count_Retries)*

These settings allow the user to establish a particular level of radio performance by controlling the RF packet retry level. The lesser of the two values will be used. If the retry count is reached before the retry time is met, then retry process on this particular packet is stopped. If the destination was a child node, it will be disassociated. If the destination was a parent bridge, the unit will begin scanning for a new parent.

The retry time may be set in the range of 1 to 30 seconds. The Aironet 340 Series Bridge will continually retry the packet in this time period while contending for the air waves with other transmitting nodes.

The retry count may be set in the range of 0 to 64 times. If the count is set to zero, only the retry time applies.

Use the retry count field if the Aironet 340 Series Bridge is mobile and you want to move from bridge to bridge very quickly after moving out of range. In non-mobile applications, since you can't move out of range, it is most likely there is some temporary interference. Retry at a later time.

## *Setting the Association Refresh Interval (Refresh)*

This setting is only present on repeater bridges. If there has been no directed traffic between the unit and its parent for the specified time (in tenths of a second) the unit will send an empty packet to the parent to verify that the connection is still alive.

### *Roaming Notification Mode (Roaming)*

When a node roams from one parent to another the new parent bridge sends packets to the wired network to inform any other bridge, switches and the old parent of the change in location.

If this option is set to *directed* and if the client node knows its old parent's address, this packet is sent as a directed packet to the old parent. In all cases we have encountered this should update the other network devices correctly. If you are having any problems you may wish to set the option to *broadcast* to cause the packet to be sent as a broadcast and be guaranteed to be sent everywhere in the network.

### *Setting the Loading Balance (Balance)*

On a root bridge you may use the *balance* option to control how often the repeater bridge will execute the load balancing algorithm (i80211 Extend must be enabled). The repeater bridges will search for better parents based on the data traffic load and number of association even if they are having no trouble with their current parent. The options may be set to Off, Slow (every 30 seconds), or Fast (every 4 seconds).

### *Setting Diversity (Diversity)*

This parameter tells the unit whether you have two antennas installed. Set the parameter to "Off" if one antenna is installed. The single antenna must be installed on the right connector when facing the back of the unit with the LED display facing up.

### *Setting the Power Level (Power)*

This parameter may be used to reduce the power level of the radio transmitter down from the maximum allowed by the regulatory commission. Depending on where you are located, you may be allowed to set the power to 50 milliwatts, 100 milliwatts or to full power.

### *Setting Fragment Size (Fragment)*

This parameter determines the largest packet size that may be transmitted. Packets that are larger than this size will be broken into pieces that are transmitted separately and rebuilt on the receiving side.

If there is a lot of radio interference or collisions with other nodes, the smaller lost packets can be retried faster and with less impact on the airwaves. The disadvantage is if there is limited interference, long packets will take more time to transmit due to the extra packet overhead and acknowledgments for the fragments.

Set the fragment size between 256 and 2048 bytes.

## *Setting Purchasable Radio Options (Options)*

This selection is used to enable special features in the radio that may be purchased separately. One example is stronger encryption. To enable an option select this menu item. You will be given the MAC address of this unit and prompted for a password. Call customer support give them this address and once payment has been verified you will be given the password for this unit.

5

CHAPTER 5

# Configuring the Ethernet Port

This chapter describes the procedures for configuring the Ethernet Port of the Aironet 340 Series Bridge.

Here's what you'll find in this chapter:

- Using the Configuration Ethernet Menu
- Activating/Disabling the Ethernet Port
- Setting the Maximum Frame Size and Port Interface Type

# Using the Configuration Ethernet Menu

The Ethernet Port is configured using the Configuration Ethernet Menu. To access this menu, select **Configuration** from the Main Menu then select **Ethernet** from the Configuration Menu.

```
            Configuration Ethernet Menu
     Option       Value      Description
1 - Active     [ on   ]   - Connection active
2 - Size       [ 1518 ]   - Maximum frame size
3 - Port       [ auto ]   - Port selection

Enter an option number or name, "=" main menu, <ESC> previous menu
>_
```

## *Activating/Disabling the Ethernet Port (Active)*

**NOTE:** Do not activate the Ethernet Port until all other parameters have been set correctly.

The *active* option is used to enable or disable the Ethernet Port connection. The default setting for active is "On".

The *active* option should be disabled if the port on the Aironet 340 Series Bridge is not going to be used. This informs the software not to route packets to the port and stops the use of processing power for scanning for Ethernet activity.

## *Setting the Maximum Frame Size (Size)*

The *size* option allows you to increase the maximum size of the frames transmitted to and from the Ethernet infrastructure. Do not set the maximum frame size greater than 1518 unless you are running proprietary software that allows you to exceed this maximum. You may set the value between 1518 to 4096.

**NOTE:** After the parameter is changed, the unit must be restarted either by powering it “Off” and then “On,” or by using the “Diagnostics Restart” command for the change to occur.

## *Setting the Port Interface Type (Port)*

If this parameter is set to “Auto”, the Aironet 340 Series Bridge will scan for a cable at all three connections. When the bridge is wired to an Ethernet card that also scans, this parameter should be set to the port that is being configured. You may select AUI for 10base5 for thicknet, 10baseT for twisted pair, or 10base2 for coax and thinnet.

6

CHAPTER 6

# Setting Network Identifiers

This chapter describes the procedures for setting the Aironet 340 Series Bridge network identifiers.

Here's what you'll find in this chapter:

- Setting the IP address, subnet mask, and routing tables
- Domain name server settings
- DHCP settings
- Setting the names assigned to the unit
- Setting up a time server

# Using the Configuration Ident Menu

Network identifiers are entered using the Configuration Ident Menu shown below. To access this menu, select **Configuration** from the Main Menu then select **Ident** from the Configuration Menu.

```
                    Configuration Ident Menu
        Option              Value            Description
1  - Inaddr     [ 149.023.165.131 ] - Internet address
2  - Inmask     [ 255.255.255.000 ] - Internet subnet mask
3  - Gateway    [ 149.023.165.050 ] - Internet default gateway
4  - Routing    [      menu       ] - IP routing table configuration
5  - Dns1       [ 149.023.130.254 ] - DNS server 1
6  - Dns2       [ 000.000.000.000 ] - DNS server 2
7  - Domain     [  "aironet.com"  ] - Domain name
8  - Name       [ "BR500T_24c1e2" ] - Node name
9  - Location   [       ""        ] - System location
01 - Contact    [       ""        ] - System contact name
02 - Bootp_DHCP [       on        ] - Use BOOTP/DHCP on startup
03 - Class      [    "BR500T"     ] - DHCP class id
Enter an option number or name, “=” main menu, <ESC> previous menu
>_
```

## *Using DHCP or BOOTP*

By default the unit is configured to attempt to get a DHCP or BOOTP server to assign it an IP address and optionally set other parts of the configuration. For a complete description of this operation see “Downloading Using the Internet Boot Protocol (Bootp/DHCP)” in **Chapter 13**.

## *Assigning an IP Address (Inaddr)*

Use the *inaddr* option to establish an IP (Internet Protocol) address for the Aironet 340 Series Bridge. An IP address must be assigned to the unit before it can be accessed by either telnet, HTTP, or SNMP.

### *Specifying the IP Subnet Mask (Inmask)*

Use the *inmask* option to assign an IP Subnetwork mask to the Aironet 340 Series Bridge. The subnetwork mask determines the portion of the IP address that represents the subnet ID. A digit in a "bit" of the mask indicates that the corresponding "bit" in the IP address is part of the subnet ID.

### *Setting Up the Domain Name Servers (Dns1,Dns1,Domain)*

A domain name server allows the operator to specify the name of a known host rather than its raw IP address when accessing another node in the network You should obtain the address of the primary and backup servers and well as the domain name for your network administrator.

### *Establishing a Node Name (Name)*

The *name* option is used to establish a unique node name for the Aironet 340 Series Bridge. The *name* is a text string of up to 20 characters that appears on all Console Port Menus. It is passed in association messages to other nodes on the radio network. See **Chapter 10** "Setting Up the Association Table".

### *Setting SNMP Location and Contact Identifiers (Location,Contact)*

Use the *location* and *contact* options to specify the location of the SNMP workstation and the contact name of the individual responsible for managing it in the event of problems.

You may enter an arbitrary string of up to 20 characters for each item.

### *Configuring the IP Routing Table (Gateway, Routing)*

The IP routing table controls how IP packets originating from the bridge will be forwarded. Once the destination IP address is determined the following checks are made:

1. If the destination IP address exactly matches a host entry in the table, the packet will be forwarded to the MAC address corresponding to the next hop IP address from the table entry.

2. If the destination is in the local subnet, ARP is used to determine the nodes MAC address.

3. If the destination address is on another subnet and matches the infrastructure portion of a net entry in the table (using the associated subnet mask), the packet will be forwarded to the MAC address corresponding to the next hop IP address from the table entry.

4. If the destination address is on another subnet and does not match any entry in the table, the packet will be forwarded to the MAC address corresponding to the default gateway's IP address.

Most subnets only have one router which is always used to access the rest of the network. Use the ***Gateway*** option to set the IP address of the default router. This address is also used if the destination address does not match any entries in the routing table described below.

The ***Routing*** options allows your to set specific entries in the routing table.

```
                Configuration Ident Routing Menu
        Option              Value            Description
    1 - Display                          - Display route table entries
    2 - Host                             - Add a static host route
    3 - Net                              - Add a static network route
    4 - Delete                           - Delete a static route
  Enter an option number or name, "=" main menu, <ESC> previous menu
    >_
```

Use the ***Host*** option to add an entry for a single host. You will be prompted for the IP address of the host. Use the ***Net*** option to add an entry for an external subnet. You will be prompted for a network address and a subnet mask to identify the remote network. Use the ***Delete*** option to delete a table entry.

The ***Display*** options show the current table.

```
                      Routing Table
   Destination          Next Hop              Mask          Flags   Use
----------------     ---------------      ---------------  ------  ---
149.023.166.000      149.023.165.071      255.255.255.000   S  N     0
        default      149.023.165.050      000.000.000.000   S  N     0
149.023.130.020      149.023.165.060      255.255.255.000   S  H     0
```

The Flags column displays letters identifying the type of entry:

- **S**: Entry is static (entered by operator)
- **N**: Entry is an remote network route
- **H**: Entry is a host route

The Use column indicates the number of packets that have been forwarded using this table entry.

## *Setting up the Time Base (Configuration Time)*

This menu lets you configure the bridge to query a network time server such that any logs its printed can reference the current date and time.

```
                Configuration Time Menu
    Option              Value             Description
1 - Time_server   [ 149.023.165.080 ]   - Time protocol server
2 - Sntp_server   [ 000.000.000.000 ]   - Network time server
3 - Offset        [      -300       ]   - GMT offset in minutes
4 - Dst           [       on        ]   - Use daylight savings time
>_
Enter an option number or name, “=” main menu, <ESC> previous menu
```

The ***time_server*** option sets the IP address or name of a unix time protocol server to be queried. The ***sntp_server*** option sets the IP address or name of an internet simple network time protocol server to be queried. You should configure only one type of server.

Since the time returned by the servers is based on Greenwich mean time you must use the ***Offset*** option to give the time difference in minutes (plus or minus) from GMT.

The ***Dst*** option selects whether your time zone uses daylight savings time.

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CHAPTER 7

# Configuring Mobile IP

This chapter describes how to set up the bridge to serve as a mobile IP home or foreign agent. It assumes you understand the concepts and configuration necessary to use Mobile IP.

Mobile IP is a protocol that allows roaming across different IP subnets while maintaining their original IP address. It requires a Mobile IP stack to be set up on the client device as well. This IP stack is available from FTP corporation and other IP stack vendors.

Each client is assigned an IP address and a home agent IP address by the network administrator. The Home agent resides on the subnet for which the client's IP address is local.

When the client roams to a foreign subnet, it contacts a foreign agent on that subnet, supplying its home agent address. The foreign agent contacts the home agent with the client's information. The home agent begins relaying any packet found on its local LAN destined to the client's IP address first back to the foreign agent and from there back to the client.

# Using the Configuration Mobile IP Menu

```
                  Configuration Mobile-IP Menu
    Option        Value      Description
1 - AgentType   [ off  ]   - Home / Foreign Agent
2 - Mobile                 - Home Agent Active Mobile Nodes
3 - Visitors               - Foreign Agent Visitor List
4 - Add                    - Add Mobile Nodes
5 - Remove                 - Remove Mobile Nodes
6 - Display                - Display Home Agent Authorized Addresses
7 - Setup       [ menu ]   - Agent Configuration
8 - Advert      [ menu ]   - Advertisement Setup
Enter an option number or name, "=" main menu, <ESC> previous menu
>
```

### *Setting the Agent Type (AgentType)*

Determine the type of agent the unit is configured for, Home or Foreign. Setting this to OFF disables the Mobile IP processing.

### *Displaying the Active Clients (Mobile, Visitors)*

On a home agent the ***Mobile*** option displays information about mobile nodes that are currently away from their home network.

```
  Mobile Node      Care of Addr    Flags  Lifetime
---------------  ---------------  ------  --------
149.23.165.1     149.23.130.20    SBDMGV   120/200
```

The first column displays the node's IP address; the next shows the foreign agent it is connected with. The lifetime column displays the count in seconds since this entry was refreshed and the count at which it will be removed. To understand the meaning of the flags, you should read the internet RFCs for Mobile IP. The flags have the following meanings:

- S - allow simultaneous care of addresses
- B - forward broadcasts to the node
- D - send directly to the mobile node
- M - use minimum encapsulation method
- G - use GRE encapsulation method
- V - use Van Jacobsen compression

On a foreign agent the ***Visitors*** option display information about the mobile nodes currently visiting this agent.

```
  Mobile Node     Mac Address     Home Agent    State  Lifetime
---------------  ------------  ---------------  -----  --------
149.023.165.001  004096123456  149.023.165.050  Est     180/200
```

The first column is the node's IP address followed by its MAC address and then its home agent IP address. The state is either *established* or *waiting* for home agent confirmation. The lifetime column displays the count in seconds since this entry was refreshed and the count at which it will be removed.

## *Authorizing Mobile Nodes to Roam (Add/Remove/Display)*

Before a node is allowed to roam, the home agent must be told information about the node to validate its identity.

The ***Add*** option allows you to validate a range of IP addresses. You are prompted for the low and high IP addresses in the range. They can be the same address to validate a single user. You may require that the setup packets sent between the home agent, foreign agent, and the node itself must be encrypted. You can either enter *noauthentication* or a security parameter index and an MD5 key.

The ***Remove*** option removes entries from the table. You may either specify a single low IP address or *all*.

The ***Display*** option displays the currently authorized nodes.

```
    Low Address      High Address       SPI                   Md5 Key
---------------  ---------------  ----------  ----------------------------
149.023.165.001  149.023.165.002
149.023.165.010  149.023.165.011           1 900000000000000000000000
```

## *Set up the Agent Parameters (Setup)*

This menu lets you configure the parameters that control the operation of the agents.

```
                Configuration Mobile-IP Setup Menu

    Option               Value          Description
1 - Lifetime        [    600     ]   - Max Registration Lifetime
2 - ReplayProt      [ timestamps ]   - Replay Protection Method
3 - Broadcasts      [    off     ]   - Broadcast Forwarding
4 - RegRequired     [     on     ]   - Registration Required
5 - HostRedirects   [    off     ]   - Enable ICMP Host Redirects to MN
Enter an option number or name, "=" main menu, <ESC> previous menu
>
```

The ***Lifetime*** parameter has two functions. It is the maximum amount of time the Home Agent will grant a mobile node to be registered on a foreign network before renewing its registration. Note that the lifetime a mobile node asks for during the registration process may be more or less than this value. However, the Home Agent will only grant a lifetime up to this value.

The lifetime value is also placed in the agent advertisement packets. mobile odes typically use this field from the advertisements to generate the Lifetime value for the Registration Request.

The ***ReplayProt*** option determines the scheme used to prevent attacks based on capturing packets and playing them back at a later time. Two replay protection methods are allowed in Mobile IP: *timestamps* (mandatory) and *nonces* (optional). Due to a patent that may apply to nonce-based replay protection, we do not support nonces at this time. This value must be set to timestamps.

The ***Broadcasts*** option determines whether mobile nodes are allowed to request that broadcasts from their home network are forwarded via tunneling to the mobile node. Some protocols require broadcast packets from the home network to maintain proper operation (i.e., NetBIOS). Unless needed, this option should be left at the default value of off to avoid unnecessary traffic.

If the ***RegRequired*** is off, mobile nodes are allowed the option of registering to a Home Agent without the use of a Foreign Agent via a co-located care-of-address dynamically acquired while on the foreign net-

work. This is useful in cases where Foreign Agents have not yet been deployed on the foreign network; however, this scheme consumes IP addresses on that network. Setting this value to on will force mobile nodes on this network to always register using a Foreign Agent.

The ***HostRedirects*** option indicates whether or not the Foreign Agent will send an ICMP message to mobile nodes registered through it specifying the Address of an IP Router for the mobile node to use. If set to "off" (default), the mobile node will always use the Foreign Agent as its default gateway (router). Setting this value to "on" may improve performance while visiting a foreign network; however, there may be connectivity problems which result due to ARP broadcasts from the mobile node.

## *Control Agent Advertisements (Advert)*

Agents advertise themselves on the LAN so that the mobile nodes can find them and determine whether they are home or away.

```
                Configuration Mobile-IP Setup Menu

    Option               Value          Description
1 - AdvertType      [ multicast ]   - Advertisement type
2 - AdvertInterval  [     5     ]   - Advertisement interval
3 - PrefixLen       [    off    ]   - Advertise prefix length extension
4 - AdvertRtrs      [    on     ]   - Advertise routers
Enter an option number or name, "=" main menu, <ESC> previous menu
>
```

The ***AdvertType*** value specifies the type of datagram the Mobile Agent will use when sending out ICMP Agent Advertisements. The RFC 1256 recommendation and the default for the Access Point is to use the All Hosts Multicast address (224.0.0.1). In testing, it was discovered that some mobile nodes were not automatically joining this multicast group and thus were ignoring the agent advertisements. For these mobile nodes this value should be changed to 'broadcast,' which will use the limited broadcast address (255.255.255.255) for all unsolicited agent advertisements.

The ***AdvertInterval*** value specifies how frequently (in seconds) the Mobile Agent will send out an ICMP Router Advertisement multicast. These advertisements are used by the mobile nodes to locate the Mobile

Agents and to determine to which network they are currently attached. The more frequent the advertisement, the sooner the mobile node will be aware that it has attached to a new network and start the registration/de-registration process (if necessary). Since these are either multicast or broadcast datagrams (see below), the Access Point must be configured to forward these types of frames onto the RF network. We are currently working on a scheme to allow link layer notification of re-attachment resulting in a Router Solicitation from the mobile node. This will prompt a unicasted Router Advertisement from the Mobile Agent to the mobile node and allow multicast/broadcast forwarding on the Access Point to be turned off.

The ***PrefixLen*** option allows the Prefix Length extension to the Mobility Agent (router) advertisement to be enabled or disabled. This extension is used to indicate the number of bits in the subnet mask for the Mobility Agent generating the advertisement. The presence of the Prefix Length extension may be helpful to some mobile nodes in determining if they have attached to a foreign network. The default value is off. (**Note:** This option should be “on” for FTP TSR stacks and “off” for VxD stacks.)

RFC 2002 (Mobile IP) states that IP Routers MAY be included in the Router Advertisement (RFC 1256) portion of the Agent Advertisement. However, since the IP Address of the Agent itself is included in the router list, doing so may cause some hosts to select the Mobility Agent as its default router. In an attempt to minimize this situation, the Mobile Agent also includes the IP Address of its default router in the list of advertised routers with a higher “preference” value. If a host continues to select a Mobility Agent as its default router, the Agent can be configured to advertise zero routes by setting ***AdvertRtrs*** to “off”. The default value is “on”.

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CHAPTER 8

# Using the Spanning-Tree Protocol

This chapter describes how to configure the Aironet 340 Series Bridge for use with the Spanning Tree Protocol (STP) Protocol.

Here's what you'll find in this chapter:

- Overview
- Understanding Loops
- How STP Protocol Works
- Receiving Configuration Messages
- Determining the Root Bridge, Root Cost, and Spanning Tree
- Understanding Bridge Failures
- Avoiding Temporary Loops
- Establishing Timeouts
- Node Aging Addressing
- Implementing the STP Protocol

# Overview

STP is used to remove loops from a bridged LAN environment.

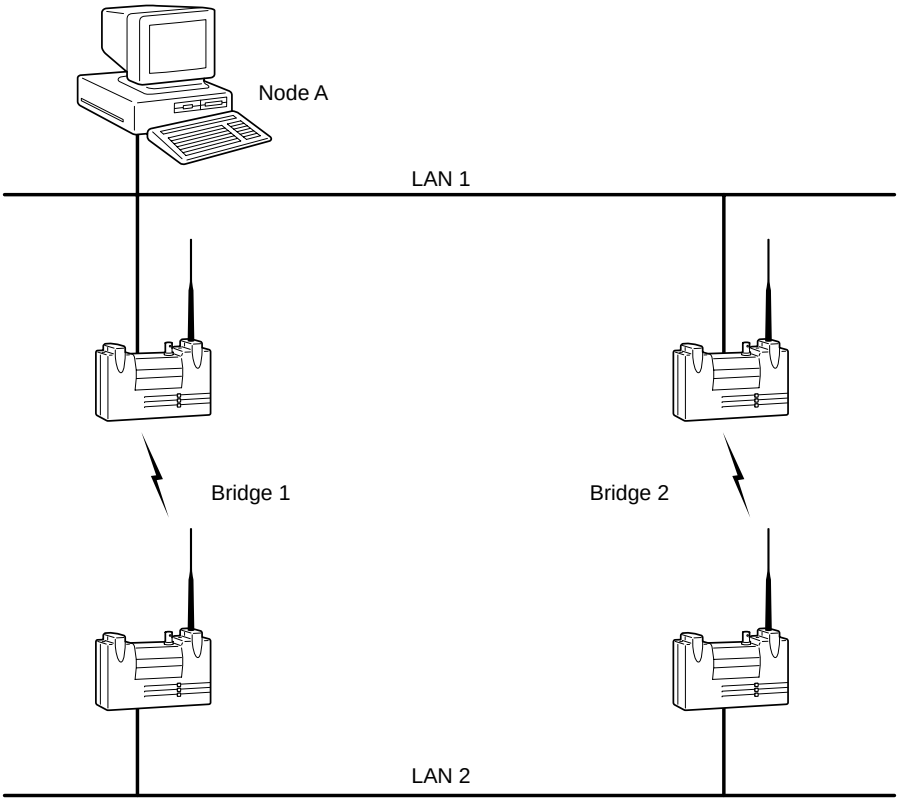
The Aironet 340 Series Bridge implements the IEEE 802.1d Spanning Tree Protocol (STP) specification to manage multiple bridges in an extended LAN environment. This allows the Aironet 340 Series Bridge to be used in bridged infrastructures with other 802.1d compliant bridges. The protocol also allows the bridges in an arbitrarily connected infrastructure to discover a topology that is loop free (a tree) and make sure there is a path between every pair of LANs (a spanning tree).

If you are administering a multiple-bridge infrastructure, this Chapter explains how the protocol works. However, if your infrastructure consists of a single bridge you can operate with the default values, although it might not be the optimal configuration required.

## *Understanding Loops*

If there is more than one path from one LAN to another, the infrastructure contains a loop.

**Figure 8.1 - Example Loop on a Bridge Infrastructure**




If Node A transmits a multicast packet, both Bridge 1 and Bridge 2 will try and forward the packet to LAN 2. Each bridge, on seeing the other's transmission on LAN 2, will forward the packet back to LAN 1. The cycle will continue and the packet will loop forever taking up all of the bandwidth of the bridges.

Topologies containing loops may be more complicated. For example, if Bridge 2 was replaced by two bridges with a LAN between them, the effect would still be the same.

# How STP Protocol Works

The STP protocol works by having the bridges transmit special configuration messages to each other. The messages contain enough information to allow the bridges to:

- Elect a single bridge. A single bridge is selected, from all the bridges on all the LAN, to be the root bridge. Each bridge then calculates the distance of the shortest path to the root bridge.
- Elect a designated bridge (for each LAN). A bridge from all the bridges residing on the LAN will be selected. This bridge will be closest to the root bridge.
- Select its own port to be root port.This bridge has the best path to the root bridge.
- Select ports are to be included in the spanning tree. Ports are included if they are a root port or the bridge itself has been selected as the designated bridge for the port's LAN.

Any ports not included in the spanning tree will be blocked and any data received from its LAN will be discarded. The bridge will not forward any traffic to this port.

### *Receiving Configuration Messages*

Configuration messages contain four main fields.

- The Bridge ID of the root bridge. This is called the root ID. A bridge's ID consists of a 16 bit priority value appended with the infrastructure address of the bridge. The infrastructure address of the bridge is usually the address of one of the bridge's ports.The priority value is assigned by the operator with a default value of 8000 hex.
- The Bridge ID of the transmitting bridge.
- The cost of the path from the transmitting bridge to the root bridge.
- The port ID of the port on which the message was transmitted.The ID is made up of an 8 bit priority value appended with an 8 bit port number assigned to the port by the bridge. The priority value is assigned by the operator with a default value of 80 hex.

Each bridge starts by assuming it is the root and its root cost is 0. When a bridge receives a configuration message, it records the values only if the message received is better than the message it would transmit out the port.

For example, message C1 is better than C2:

- If the root ID in C1 has a lower numeric value than the value from C2.
- If the root ID's are equal and C1's root cost is lower.
- If the root ID's and costs are equal and C1's transmitting Bridge ID has a lower numeric value.
- If the root ID, cost, and Bridge ID are equal and C1 was transmitted on a port with a lower port ID. This should only occur if two ports from the same bridge are connected to the same LAN.

If a port receives a better message than the one it would transmit, the bridge stops transmitting configuration messages on that port. Only one port on each LAN will be transmitting the messages. The bridge that contains this port is called the designated bridge for that LAN and the port is called the designated port.

## *Determining the Root Bridge and Root Cost*

Each bridge determines the root bridge's ID by comparing its own ID with those from the best messages received on all of its ports. The root ID is then used in all transmitted configuration messages.

If a bridge is the root, its root cost is 0. If a bridge is not a root, its cost is the minimum of the costs received in the messages from all its ports as well as the cost from the port on which the minimum cost message was received. This cost is then used in all transmitted configuration messages.

The port on which the minimum cost message was received is called the root port.

### *Determining the Spanning Tree*

All ports on a bridge, either the root port or the designated port for their LAN, are allowed to forward packets. All others are blocked and do not transmit or receive any data packets.

### *Understanding Bridge Failures*

All root and blocked ports monitor the LANs to which they are connected and watch for configuration messages transmitted by the designated bridge for the LAN.

The STP protocol specifies a timeout period in which these ports must see at least one message. Each time a message is received, the timer is restarted. If the timeout period expires, the bridge assumes the designated bridge has failed.

The bridge will discard the saved value for the port, make the port the designated port for that LAN, and restart sending configuration messages. The bridge will also recalculate its values for the root bridge and root cost based on the active ports.

Other blocked ports on the same LAN will timeout and start to transmit messages. Eventually a new designated bridge, port, and root bridge will be determined.

### *Avoiding Temporary Loops*

It will take a non-zero amount of time for the protocol to determine a stable loop free topology due to the time for messages to pass from one end of the infrastructure to the other. If the ports were allowed to forward while the protocol was stabilizing, then temporary loops could form.

To avoid temporary loops, ports are not allowed to go immediately from the blocked state to the forwarding state. They must first go through a state called listening. In this state, they may receive and transmit configuration messages as needed but must block all data traffic. The time spent in the listening state must be at least twice the end-to-end transmit time of the infrastructure.

If the port is still part of the spanning tree at the end of the listening period it is put in the learning state. In this state it can still receive and transmit configuration messages, but is also allowed to learn the source addresses from the packets received from its LAN. It is still not allowed to forward any packets. The learning state is used to lessen the amount of flooding of unknown destination addresses that would occur if the port started forwarding before there were any entries in its learning table.

Once the learning period is over, the port is allowed to forward data normally.

### *Establishing Timeouts*

The configured timeout values on the root bridge are passed to each bridge in a configuration message to ensure that all bridges on the infrastructure are using the same timeout periods.

The root bridge puts its own values in its messages. All other bridges copy the values contained in the configuration message sent to them from their root port. The value in this message is used in all of the bridge's transmitted messages. Using this method, the values are propagated throughout the infrastructure.

### *Node Address Aging*

Occasionally stations may be moved from one LAN to another. The bridges will remove learned addresses from their tables if no packets have been received from a node for a period of time.

If node addresses do not timeout, the bridge may continue to send packets for a node to the wrong LAN. If a node sends packets from its new LAN location, the tables might be corrected, however, this is not guaranteed. The default timeout period is 5 minutes.

If a new bridge or port is added to an infrastructure, the ports included in the spanning tree could change dramatically. It may appear that a node has changed location very quickly.

To allow for these quick changes of location, the spanning tree protocol specifies that every time a port enters the blocked or forwarding states, its bridge must send a topology changed message to the root bridge.

The root bridge in turn will include a flag in all the configuration messages it sends.This flag will be propagated through the infrastructure by all the other bridges. After a time period the root bridge will clear the flag. This instructs all bridges to return to the normal aging timeout.

# Implementing STP Protocol

The STP protocol is implemented on the Aironet 340 Series Bridge as follows.

- Each root bridge, with all of its repeaters, looks to other bridges in the infrastructure as a single multi-port bridge with a bridge address equal to the infrastructure address of the root bridge.
- The STP protocol runs only on the root bridge, not on repeaters. Repeaters only transmit packets or change state on commands from the root bridge.
- To reduce radio traffic, the repeaters will continue to transmit configuration messages at the timeout period without having to be told to transmit each one by the root bridge. They will also only send received configuration messages back to the root bridge if they are different from the previously received message.
- When a repeater is not associated to a parent bridge, it will put its LAN port in the blocked state and will not forward any data to or from the port. Once associated, the root bridge will take control.
- The protocol parameters are all configured from the root bridge. The local port parameters are configured on each repeater bridge.

# Using the Configuration STP Menu (Root Bridge Only)

The STP Protocol for a root bridge is configured using the Configuration STP Menu. This menu will only appear if the Root Mode is “On” as described in **Chapter 4** “Configuring the Radio Network”. To access this menu, select **Configuration** from the Main Menu, then select **STP** from the Configuration Menu**.**

```
                  Configuration Stp Menu
      Option         Value          Description
  1 - Active     [    off     ]    - Protocol enabled
  2 - Bridge     [   menu     ]    - Bridge parameters
  3 - Port       [   menu     ]    - Port parameters
  4 - Display                     - Protocol status
  5 - State      [ "Forward" ]    - Local ethernet port state
```

### *Enabling STP Protocol (Active)*

The *active* option acts as an On/Off switch for the STP protocol. The default setting is “Off”, which means all root and repeater LAN ports are placed in the forwarding state. If the option is turned “On”, the root and repeater LAN ports are placed in the listening state.

If you are running a small infrastructure, and there will never be any loops, leave the STP protocol “Off”. If you are unsure, change the setting to “On” as the overhead involved for bridges is small.

### *Setting Bridge Parameters (Bridge)*

The *bridge* option allows you to set the overall parameters and timeout values for a root bridge. When the *bridge* option is selected, the Configure STP Bridge Menu appears.

```
                Configuration Stp Bridge Menu
      Option              Value       Description
  1 - Priority          [ 8000 ]    - Bridge priority
  2 - Hello_time        [  2   ]    - Hello message interval
  3 - Forward_delay     [  15  ]    - Forwarding delay
  4 - Msg_age_timeout   [  20  ]    - Receive hello message timeout
```

### *Setting the Bridge Priority (Priority)*

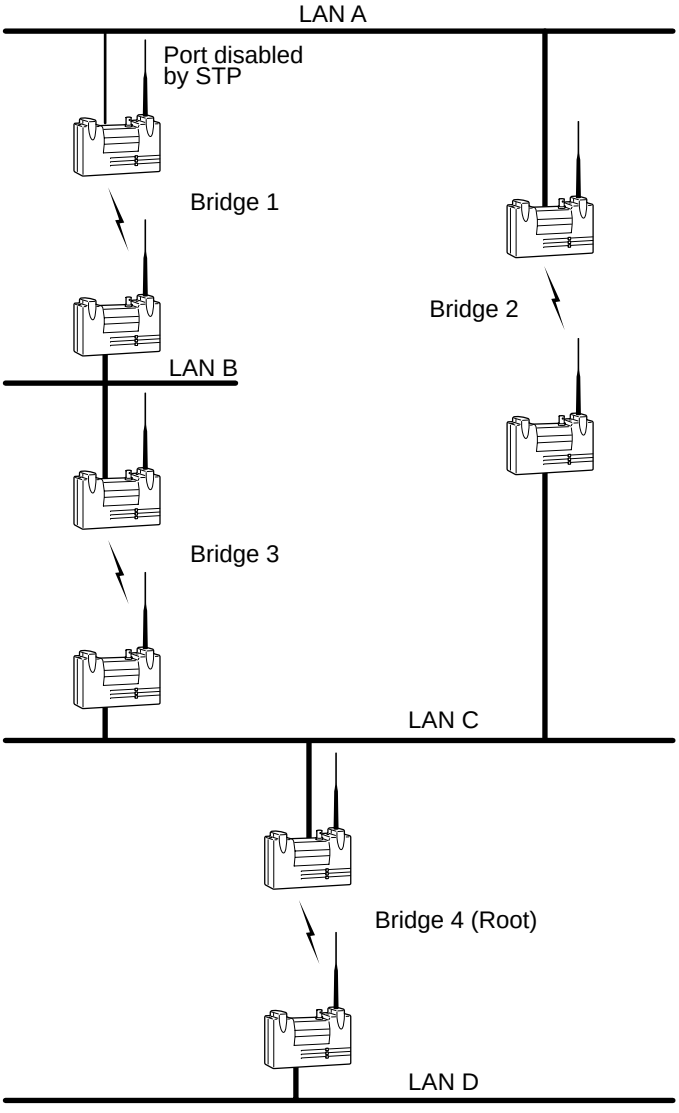
The *priority* option is used to set the priority value appended to the infrastructure address of the Bridge ID.

By changing the priority value, you can influence which bridge in the infrastructure will become the root bridge. The lower the priority value, the more likely the bridge will be the root. If all other bridges are set to the default value (8000 hex), a bridge set with a lower value will become the root.

**Figure 8.2** provides a sample configuration in which it would be useful to change the root bridge.

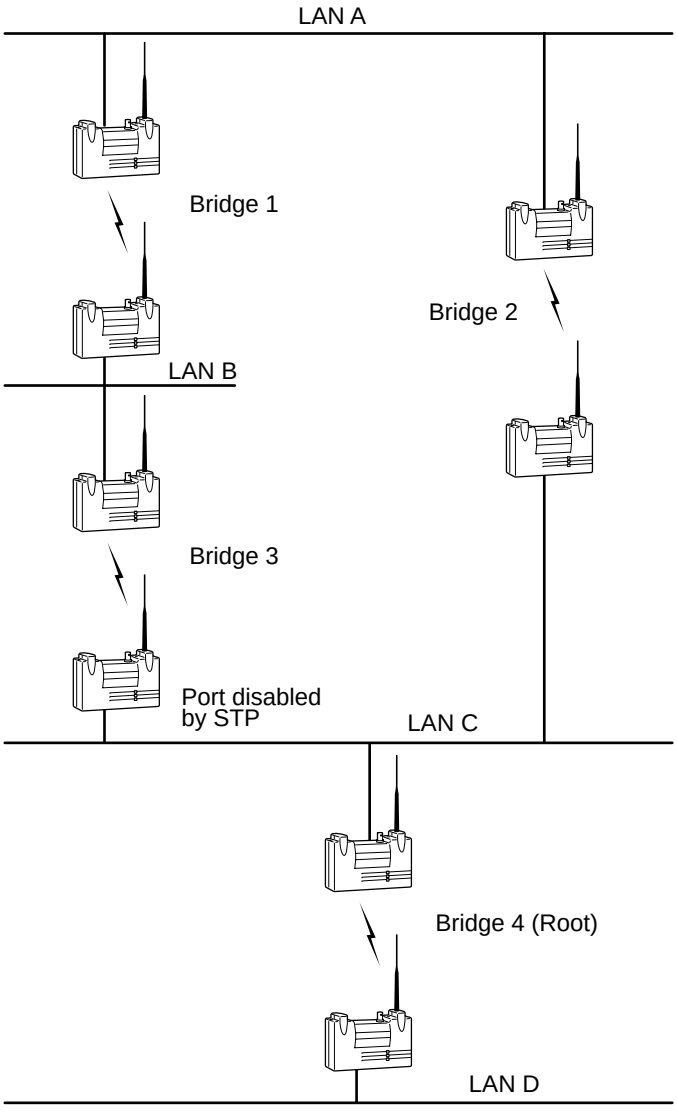
Bridge 4 is the root with the spanning tree shown by the thick line. STP has been disabled to the port on Bridge 1 to prevent a loop.

**Figure 8.2 - Non-Optimal Choice of Root Bridge**



If there is heavy traffic between LAN A and LAN B, it would be more efficient to have Bridge 1 become the root with the port on Bridge 3 being disabled.

**Figure 8.3 - Alternate Root Bridge Arrangement**



### Setting the Hello Message Interval Time (Hello_Time)

The *hello_time* option is used to set the interval time, in seconds, between the transmission of configuration messages. This value is only used if the local bridge becomes the spanning tree root bridge. If not, the value in received configuration messages transmitted from the root bridge is used.

If the interval time is set too high, the infrastructure will respond slowly in resolving any conflict problems. However, if set too low, the infrastructure will be crowded with hello message traffic. The interval time values range between 1 and 10 with a default value of 2.

### Setting the Forwarding Delay Time (Forward_Delay)

The *forward_delay* option is used to set the delay time, in seconds, that the ports will spend in the listening and learning states. This value is only used if the local bridge becomes the spanning tree root bridge. If not, the value in received configuration messages transmitted from the root bridge is used.

This option is also the timeout period used to age learned addresses whenever the spanning tree topology is changed. The value should be at least twice the transit time of a packet sent from one end of the infrastructure to the other. This allows for news of a topology change to reach all nodes and allows all ports to be blocked before new ports enter the forwarding state.

If the interval time is set too low, then temporary loops could be formed. However, if set too high, it will take longer for the infrastructure to become active after a spanning tree topology change has been made. The delay time values range between 4 and 30 with a default value of 15.

### Setting the Receive Hello Message Timeout (Msg_age_timeout)

The *msg_age_timeout* option is used to set the timeout period, in seconds, a blocked or root port watches for configuration messages from the infrastructure's designated port. This value is only used if the local bridge becomes the spanning tree bridge. If not, the value received in configuration messages transmitted from the root bridge is used.

Each time a configuration message is received, the timer is started. If the timer expires, the root bridge is assumed to have failed and the spanning tree infrastructure will be reconfigured.

If the timeout period is set too low, the spanning tree infrastructure may reconfigure itself unnecessarily and messages can be lost due to heavy traffic on the infrastructure. However, if set too high, the infrastructure will take longer than necessary to recover from failed ports or bridges.

The upper limit on the allowed range is determined by the setting of the forwarding delay. The timeout period must be less than twice the forwarding delay, minus 1 second. The timeout values range between 6 and 29 with a default value of 20.

## *Setting Port Parameters (Port)*

The *port* option allows you to set the port parameters for a root bridge's local LAN port and for the ports of any active connected repeaters. When the *port* option is selected, the Configuration STP Port Menu appears.

```
                Configuration Stp Port Menu
     Option       Value      Description
1 - Port        [ on  ]   - Protocol enabled for ethernet port
2 - Priority    [ 80  ]   - Local ethernet port priority
3 - Cost        [ 100 ]   - Local ethernet port cost
4 - Rport                 - Protocol enabled for remote port
5 - Rpriority             - Remote port priority
6 - Rcost                 - Remote port cost
```

### *Enabling the STP Protocol (Port)*

The *port* option is used to enable the STP protocol on the local port. The default setting is "On", which allows all root bridge LAN ports to be initially placed in the listening state. If the option is turned "Off", the LAN ports are placed in the forwarding state.

If the port's LAN will always be connected to the bridge and loops will never occur, turn the protocol "Off" to prevent the port from transmitting configuration messages on every timeout period.

### *Setting the Local Port Priority (Priority)*

The *priority* option is only used when two or more repeaters are connected to the same LAN for redundancy and you want to select which one will forward the packets. The port assigned the lowest priority value will be the one to forward. The priority range is from 0 to ff hex with a default setting of 80.

### *Setting the Local Port Cost (Cost)*

The value for the *cost* option is added to the root cost field from any received configuration messages to determine if the port has the least cost path to the root. Cost values can be set for 65535 or less with a default value is 100.

The larger the cost value, the more likely the port will be a backup for another active port on its LAN. If there is no active port, it is likely the LAN will be a leaf of the infrastructure tree or a less used LAN in the tree.

### *Configuring Ports of Active Connected Repeaters (Rport, Rcost, Rpriority)*

The *rport*, *rpriority* and *rcost* options are used to configure the ports of active and connected repeaters in the root's radio tree.

These options are identical to the *port*, *priority* and *cost* options described except when the values are entered, you will be prompted for the applicable port number. The port number can be obtained from the port ID field on the Protocol Status Display screen.

## *Displaying the Protocol Status (Display)*

The *display* option shows the overall status of the STP protocol and the state of each port on the local bridge. When you select **Display**, the STP Status screen appears.

```
                                STP STATUS
Bridge Id : 8000-00409611cd0e             Network Hello interval  :  2 sec
Root Id   : 8000-0000f3108678, Cost 100   Network Forward Delay   :  5 sec
Topology change : off                     Network Msg age Timeout : 20 sec
                                                ------------Designated----------
Port Address LAN  Id   Cost    State       Type      Bridge       Port Root Cost
------------ --- ---- ----- -------- ---------- ----------------- ---- ---------
00409611cd0e Eth 8001   100  Forward       Root 8000-0000f3108678 8002         0
```

- **Bridge Id:** The ID of the local bridge.
- **Root Id:** The ID of the spanning tree root. If the local bridge is not the root, then the cost to the root is also displayed.
- **Topology change:** Indicates whether the short aging timeout is currently in use because of a port state change somewhere on the infrastructure.
- **Network Hello Interval, Network Forward Delay and Network Msg age Timeout:** Shows the timeout values received from the root bridge which are in use by all bridges on the infrastructure. These values override any locally configured values.
- **Port Address:** The infrastructure address of the bridge on which the port resides.
- **Id:** The port ID, which consists of the port priority (high byte) and the port number (low byte). As each repeater connects to the root its port is assigned the next available port number.
- **Cost:** The operator configured cost for the port.
- **State:** Current state of the port. Shows one of forward, learn, listen, or blocked. The state may also be disabled if the port has been shut off by the operator.
- **Type:** Current port type. Shows one of root, designated, or blocked. The type will be disabled if the protocol is not running on the port.
- **Designated (Bridge, Port, Root Cost):** Displays the designated bridge and port for the specific LAN as well as the cost to the root from the designated port.

### *Viewing the Port State (State)*

The *state* option is a read-only value which displays the current STP state of the local LAN port. The states displayed are forward, learn, listen, or blocked.

# Part 3 - Control & Maintenance

9

CHAPTER 9

# Viewing Statistics

This chapter describes how to use the Statistics Menu to monitor the performance of the Aironet 340 Series Bridge.

Here's what you'll find in this chapter:

- Viewing the Statistics Menu
- Throughput Statistics
- Radio Error Statistics
- Ethernet Error Statistics
- Displaying Source Routes
- Displaying Overall Status
- Recording a Statistic History
- Displaying a Statistic History
- Displaying Node Information
- Setting Screen Display Time

# Viewing the Statistics Menu

The Statistics Menu provides easy access to a variety of statistical information regarding the Aironet 340 Series Bridge's performance. You can use the data to monitor the unit and detect problems when they occur. To access this menu, select **Statistics** from the Main Menu.

```
                    Statistics Menu
       Option          Value     Description
1  - Throughput                - Throughput statistics
2  - Radio                     - Radio error statistics
3  - Ethernet                  - Ethernet error statistics
4  - Status                    - Display general status
5  - Map                       - Show network map
6  - Watch                     - Record history of a statistic
7  - History                   - Display statistic history
8  - Nodes                     - Node statistics
9  - ARP                       - ARP table
01 - Display_time   [ 10  ]    - Time to re-display screens
02 - IpAdr          [ off ]    - Determine client IP addresses
Enter an option number or name, "=" main menu, <ESC> previous menu
>_
```

## *Throughput Statistics (Throughput)*

The Throughput Statistics Display provides a detailed summary of the radio data packets passing through your unit. To access this display, select **Statistics** from the Main Menu then select **Throughput** from the Statistics Menu.

```
                        THROUGHPUT STATISTICS
Cleared 19:11:52 ago
                                Recent                Average    Highest
Statistic                       Rate/s      Total     Rate/s     Rate/s
--------------------------- ---------- ---------- ---------- ----------
Radio Receive      Packets           2     110798          1        174
                   Bytes           167    7143295        103       9086
                   Filter            0          0          0          0
                   Error             0          0          0          0
Radio Transmit     Packets           4     131085          1        175
                   Bytes           377   18500991        267      37749
                   Errors            0       9036          0         27
Bridge Receive     Packets           3     151112          2        321
                   Bytes           260   30547969        442      32549
                   Filtered          5     350282          5        928
                   Errors            0          2          0          0
                   Misses            0          0          0          0
Bridge Transmit    Packets           2      54398          0        320
                   Bytes           193    1051355         93     170822
                   Errors            0          0          0          0
Enter space to redisplay, C[lear stats], q[uit] :
```

- **Recent Rate/s**: Displays the event rates, per second, averaged over the last 10 seconds.
- **Total**: Displays the number of events that have occurred since the statistics were last cleared.
- **Average Rate**: Displays the average event rates, per second, since the statistics were last cleared.
- **Highest Rate**: Displays the highest rate recorded since the statistics were last cleared.
- **Packets**: Displays the number of packets transmitted or received.
- **Bytes**: Displays the total number of data bytes in all the packets transmitted or received.
- **Filtered**: Displays the number of packets that were discarded as a result of an address filter being setup.

- **Errors**: Displays the number of errors that may have occurred.
- **Enter space to redisplay, C[lear stats], q[uit]**: To redisplay statistics, enter a space by pressing the space bar. To clear the statistics, press "C" (case sensitive). To exit the Statistics Menu, press "q".

## *Radio Error Statistics (Radio)*

The Radio Error Statistics Display provides a detailed summary of the radio receiver and transmitter errors that have occurred on the unit.

To access this display, select **Statistics** from the Main Menu then select **Radio** from the Statistics Menu.

```
                          RADIO ERROR STATISTICS
Cleared 19:23:22 ago
Receive                                 Transmit
-------------------------------------   -------------------------------------
Buffer full frames lost         0       Retries                          45
Duplicate frames                0       Max retries / frame               7 +7
CRC errors                      0       Excessive retries                 0
                                        Queue full discards               0
Enter space to redisplay, C[lear stats], q[uit]:
```

- **Buffer Full Frames Lost**: Number of frames lost due to a lack of buffer space in the unit.
- **Duplicate Frames**: Number of frames that were received more than once. This is usually due to a frame acknowledgment being lost.
- **CRC Errors**: Number of frames received with an invalid CRC. Usually caused by interference from nearby radio traffic. Occasional CRC errors can also occur due to random noise when the receiver is idle.
- **Retries**: A cumulative count of the number of times a frame had to be retransmitted due to an acknowledgment not being received.
- **Max Retries / Frame**: The maximum number of times any one frame had to be retransmitted. Excessive retries may indicate a poor quality radio link.
- **Queue Full Discards**: Number of times a packet was not transmitted due to too many retries occurring to the same destination. Discards will only occur if packets destined to this address are taking up more than their share of transmit buffers.

## *Error Statistics*

The Ethernet Error Statistics Display provides a detailed summary of the receiver and transmitter errors that have occurred on the unit. To access this display, select **Statistics** from the Main Menu then select **Ethernet** from the Statistics Menu.

### *Ethernet Error Statistics*

```
                        ETHERNET ERROR STATISTICS
Cleared 19:36:31 ago
Receive                                  Transmit
-------------------------------------  -------------------------------
Buffer full frames lost         0      Excessive collisions           0
CRC errors                      0      Deferrals                    273
Collisions                    2 +2     Excessive deferrals            0
Frame alignment errors          0      No carrier sense present       0
Over-length frames              0      Carrier sense lost             0
Short frames                    0      Out of window collisions       0
Overruns                        0      Underruns                      0
Misses                          0      Bad length                     0
Enter space to redisplay, C[lear stats], q[uit] :
```

- **Buffer Full Frames Lost**: Number of frames lost due to a lack of receiver buffer space in the unit.
- **CRC Errors**: Number of frames received with an invalid CRC.
- **Collisions**: Number of times a collision occurred while the frame was being received. This would indicate a hardware problem with an Ethernet node on the infrastructure.
- **Frame Alignment Errors**: Number of frames received whose size in bits was not a multiple of 8. Occasionally, extra bits of data are inadvertently attached to a transmitted packet causing a frame alignment error.
- **Over-length Frames**: Number of frames received that are longer than the configured maximum packet size.
- **Short Frames**: Number of frames received that are shorter than the allowed minimum packet size of 64 bytes.
- **Overruns**: Number of times the hardware receive FIFO overflow. This should be a rare occurrence.

- **Misses**: The number of Ethernet packets that were lost due to a lack of buffer space on the unit.
- **Excessive Collisions**: Number of times transmissions failed due to excessive collisions. Usually indicates the frame had to be continuously retried due to heavy traffic on the Ethernet infrastructure.
- **Deferrals**: Number of times frames had to wait before transmitting due to activity on the cable.
- **Excessive Deferrals**: Number of times the frame failed to transmit due to excessive deferrals. Usually indicates the frame had to be continuously retried due to heavy traffic on the Ethernet infrastructure.
- **No Carrier Sense Present**: Number of times the carrier was not present when a transmission was started. Usually indicates a problem with a cable on the Ethernet infrastructure.
- **Carrier Sense Lost**: Number of times the carrier was lost during a transmission. Usually indicates a problem with a cable on the Ethernet infrastructure.
- **Out of Window Collisions**: Number of times a collision occurred after the 64th byte of a frame was transmitted. Usually indicates a problem with a cable on the Ethernet infrastructure.
- **Underruns**: Number of times the hardware transmit FIFO became empty during a transmit. This should be a rare occurrence.
- **Bad Length**: Number of times an attempt was made to transmit a packet larger than the specified maximum allowed.

## *Displaying Overall Status (Status)*

This display shows the settings of the most important configuration parameters of the Ethernet unit as well as important run-time statistics. Use the display to see if anything significant is configured incorrectly. The display is broken into sections describing:

- The radio
- Any LAN connections
- Any filtering being done

All items in the display are self-explanatory or are explained in other sections of this manual.

```
                              Status
Uptime: 130:48:02
------------------------------ Radio -------------------------------
SID       :  105            Bitrate  :  1_2 Mb/s      Radio  :  LM35
Root      :  on             Pattern  :  21            Carrier:  0
                                                      Power  :  full
Autoassoc :  on             Nodes    :  1 associated
------------------------ Ethernet ---------------------------------
Active    :  on                        Pkt/sec   Rcv : 3
                                                 Xmt : 0
---------------------------- Filters -------------------------------
Multicast :  forward (0 set)              Protocols : off    (0 set)
Source    :  off     (0 set)

Enter space to redisplay, q[uit] :
```

## *Display a Network Map (Map)*

This command causes the bridge to poll all of the other bridges in the local infrastructure for information about the radio nodes associated to them. Nodes that are associated to parents are displayed indented one level from their parents on the display.

```
                    NETWORK MAP
Device          Node Id        IP Address      Ver   Name
----------- ------------  ---------------  -----  -----------------
BRE105E     00409611cd0e  149.023.165.163  4.1G   BRE105E_22ff0a
  AP4500T   00409611d1e5  149.023.165.169  4.1G   hello there
    UC4500E 004096207206  149.023.165.176  4.1G   UC4500E_207206
  LM4500    00409620222a  149.023.165.238
AP4500E     00409611855b  149.023.165.160  4.1B   AP4500E_11855b
  LM4500    00409620222d
Enter space to redisplay, q[uit]:
```

The version column displays the firmware release level currently running on the unit.

## *Recording a Statistic History (Watch)*

Use the *watch* option to record the values of a chosen Ethernet statistic over time. Once you select a statistic and a time interval, the unit will start a timer. At each timer expiration, the unit will record the current value of the statistic. The last 20 samples are saved.

➔ **To Record a Statistic History:**

1. Select the *watch* option.

```
1. ra Radio
2. re Radio Error
3. et Ethernet
4. ee Ethernet
Enter category, one of [a number from 1 to 4, a short
form]:
```

2. Type the applicable category number and press **ENTER**. For example, if you choose “Radio” the following information would appear:

```
                    Radio
  Receive                           Transmit
1 rpa Packets                     5 tpa Packets
2 rby Bytes                       6 tby Bytes
3 rfi Filtered                    7 ter Errors
4 rer Errors
Enter one of [a index from 1 to 7, a short form]:
```

3. Type the applicable statistic index number and press **ENTER**.

   ```
   Enter a sample time in seconds from 1 to 3600 :
   ```

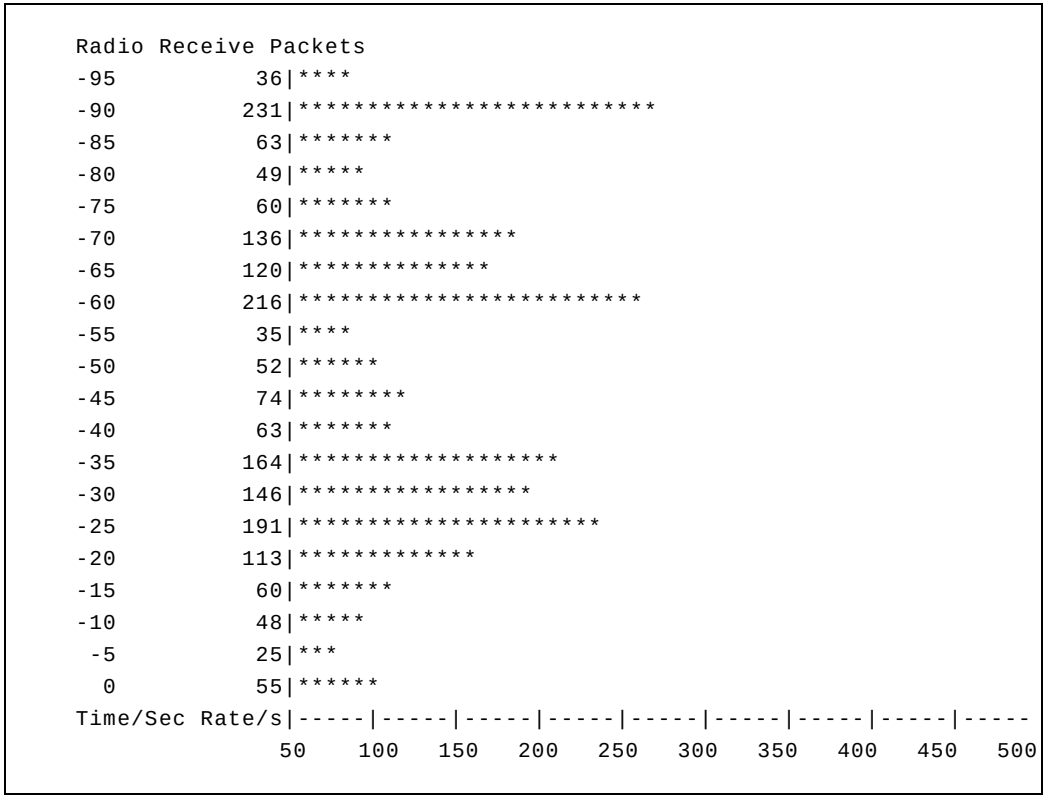
4. Type a time interval between samples and press **ENTER**. The longer the time you specify, the further back in time the samples will be saved (up to 20 samples).

## *Displaying a Statistic History (History)*

Use the *history* option to display the Ethernet history of the statistic that is currently being recorded.

➔ **To Display a Statistic History:**

1. Select the *history* option. Depending on your *watch* option selections, a display screen similar to the one below will appear.




- **Time (sec)**: Displays the number of seconds elapsed from the time the statistic sample was recorded.
- **Rate/s**: Displays the actual value of the statistic. The chart will change scale based on the largest value displayed.

## *Displaying Node Information (Node)*

The *node* command displays current Ethernet information about the client.

```
                        Radio Node Statistics
 ID   Address  Signal  Tx Pkt  Tx Byte  Tx Retry   Rx Pkt   Rx Byte   Rate
---   -------  ------  -----   -------  --------   ------   -------   -----
    004096128e76 45    1012     204322    39        1673    112386
 Enter space to redisplay, q[uit]:
```

- **Address**: Displays the address of the client.
- **Signal**: Displays the signal strength of the client.
- **Tx Pkt**: Displays the number of packets transmitted from the client.
- **Tx Byte**: Displays the actual number of bytes transmitted from the client.
- **Tx Retry**: Displays the number of transmitted packets that were resent by the client.
- **Rx Pkt**: Displays the number of packets the client has received.
- **Rx Byte**: Displays the actual number of bytes received by the client.

## *Displaying ARP Information (ARP)*

The *ARP* command displays the ARP table of IP address to MAC address. It also displays whether the node supports Ethernet Type II or IEEE 802.2 framing. The last column displays the time until the entry times out.

```
                      INTERNET ADDRESS TABLE
   Internet Address  Network Address  ETHII   802.2     Time
   ----------------  ---------------  -----  ------  ----------
   149.023.165.175      0000c0d9657f   Yes            0:14:57
   149.023.165.040      0800099e0b1a   Yes            0:14:57
   Enter space to redisplay, q[uit] :
```

### *Setting Screen Display Time (Display_Time)*

Use the *display time* option to set the Ethernet time interval for the automatic redisplay of any repeating display. The default value is 10 seconds.

### *Determine Client IP Addresses (Ipadr)*

This function can be enabled to get the unit to determine the IP address of the client nodes associated to it. The address are then display in the Map function described above.

If enabled the unit will watch traffic from the client associated to it and record the source IP addresses found.

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CHAPTER 10

# Setting Up the Association Table

This chapter describes the procedures for setting up the Association Table for the Aironet 340 Series Bridge.

Here's what you'll find in this chapter:

- Overview
- Using the Association Menu
- Displaying the Association Table
- Displaying the Association Table Summary
- Setting the Allowed Number of Child Nodes
- Controlling Associations with Static Entries
- Backbone LAN Node Stale Out Time
- Specifying How Node Addresses are Displayed

# Overview

Client nodes and repeater bridges request to be associated with a parent bridge so the parent will forward data frames. This exchange of radio packets passes back and forth information such as a node's address, device, association type, and ASCII name. This information is entered into the bridge's association table along with the address of the parent bridge. Each bridge maintains entries in its table for all nodes associated to it and all nodes associated to any repeater serving it. There may be up to 2048 entries in the table.

A bridge will accept an association from any node that requests it. The operator may set up entries in the association table to control which nodes are allowed to associate.

Using the information in the association table, the bridge can perform a variety of traffic-control functions in moving packets to their proper destination on the infrastructure. When packets are received from the Ethernet or radio network, the bridge will look in its table for the packet's destination address and do one of the following:

- If the entry shows the radio node is associated to this unit, the packet can be forwarded directly.
- If the entry indicates that the entry is associated to a repeater serving this unit, the packet is forwarded to the repeater.
- If the address is not found, a root unit will forward the packet to the wired LAN, while a repeater will forward the packet to its own parent bridge.

# Using the Association Menu

The Association Menu contains options that allow you to view the table entries, add entries, and control the routing of packets on your radio network. To access this menu, select **Association** from the Main Menu.

```
                    Association Menu
     Option       Value        Description
1 - Display                  - Display the table
2 - Summary                  - Display the table summary
3 - Maximum   [  1024   ]    - Maximum allowable child nodes
4 - Autoassoc [   on    ]    - Allow automatic table additions
5 - Add                      - Control node association
6 - Remove                   - Remove association control
7 - Staletime [   350   ]    - Backbone LAN node stale out time
8 - Niddisp   [ numeric ]    - Node Ids display mode
Enter an option number or name, "=" main menu, <ESC> previous menu
```

## *Displaying the Association Table (Display)*

Use the *display* option to view the association table entries. Select “display” to enter the type of entries to be displayed.

- **All**: Displays all entries in the table.
- **Connected**: Displays only nodes that are actively connected to the Aironet 340 Series Bridge.
- **Heirachy**: A special shortened display which shows the association tree with children indented from their parents.
- **Static**: Displays only nodes for which a static entry has been made to control the nodes’ association.
- **Multicast-filters**: Displays only those entries for multicast addresses for which filters have been added. See **Chapter 11** “Using Filters”.
- **Node-filters**: Displays only those entries for node address for which filters have been added. See **Chapter 11** “Using Filters”.

The typical hierarchy display will resemble:

```
                        RADIO HIERARCHY
Device                   Address   Name
-------------------- ------------ --------------------
BRE105E              00409611cd0e BRE105E_22ff0a
  BRE105T             00409611d1e5 hello there
    UC4500E           004096207206 UC4500E_207206
  BRE105E             00409611d602 BRE105E_22ff0a
    UC4500E           0040962068b0 UC4500E_2068b0
  LM4500              00409620222a
```

The rest of the displays will be similar to the one below.

```
                        RADIO NODES
   Address      Device  Type    Parent    Name               Src
------------- --------- ---- ------------ ------------------ ---
 00409611cd0e   BRE105E   Me               BRE105E_22ff0a     Fwd
 00409611d1e5   AP4500T  Rep         Local hello there
N00409611d602   AP4500E  Rep         Local AP4500E_11d602     Fwd
 00409620222a    LM4500              Local                    Fwd
 0040962068b0   UC4500E       00409611d602 UC4500E_2068b0     Fwd
 004096207206   UC4500E       00409611d1e5 UC4500E_207206     Fwd
Enter space to redisplay, q[uit] :
```

- **Address Column**: Displays the address (in ascending numerical order) for each node on the infrastructure. An “N” before the address indicates that the node is a static entry and not associated. An “R” before the address indicates that the node is static and associated. The letters “N” and “R” only appear beside static entries.
- **Type Column**: Displays the node association type. The following types may appear in the table:

  **Me**: Represents this Aironet 340 Series Bridge.

  **Psp**: Indicates the node that is using the Power Saving Protocol (PSP) to communicate with the system. Some radio nodes, usually wireless client devices, only power up part of the time to conserve energy. Therefore the bridge must communicate to these nodes using PSP.

  **Prnt**: Indicates a repeater’s parent node.

  **Rep**: Indicates a repeater bridge.

- **Parent Column**: Displays the node ID of the parent to which the node is associated. In place of a node ID, the column may display the following:

  **A blank entry**: The node is not associated.

  **Local**: The node is associated to this unit.

  **Local block**: The node has been blocked and will not be allowed to associate with the local system directly.

  **Name Column**: Displays the node name.

  **Rdst, Src**: Displays the type of multicast filter action that has been set for Radio (RDst) and Source (Src) packets. A blank means that the action is forward. See **Chapter 11** "Using Filters".

## *Displaying the Association Table Summary (Summary)*

Use the *summary* option to view a summary of the number of nodes associated to your unit. When you select the *summary* option, the Association Table Summary Display appears:

```
                    ASSOCIATION TABLE SUMMARY
                           Non-Psp     Psp  Repeaters
                           ------- ------- ----------
   Direct associations   :       1       0          2
   Indirect associations :       2       0          0
```

- **Direct Associations**: Number of Non-PSP, PSP, or repeater nodes associated to this bridge.
- **Indirect Associations:** Number of Non-PSP, PSP, or repeater nodes associated to the Aironet 340 Series Bridge below the current bridge, on the radio network tree.

## *Setting the Allowed Number of Child Nodes (Maximum)*

This command determines the maximum number of allowed child nodes that can be associated to the Aironet 340 Series Bridge.

## *Controlling Associations With Static Entries (Autoassoc/Add/Remove)*

Use the *auto-association* parameter and the static association table entries to control associations.

In its default configuration, the bridge will allow any radio node in range to associate to it. For a more secure installation you must add static entries to the association table for these nodes. This allows control over which radio nodes are allowed to associate with which bridge.

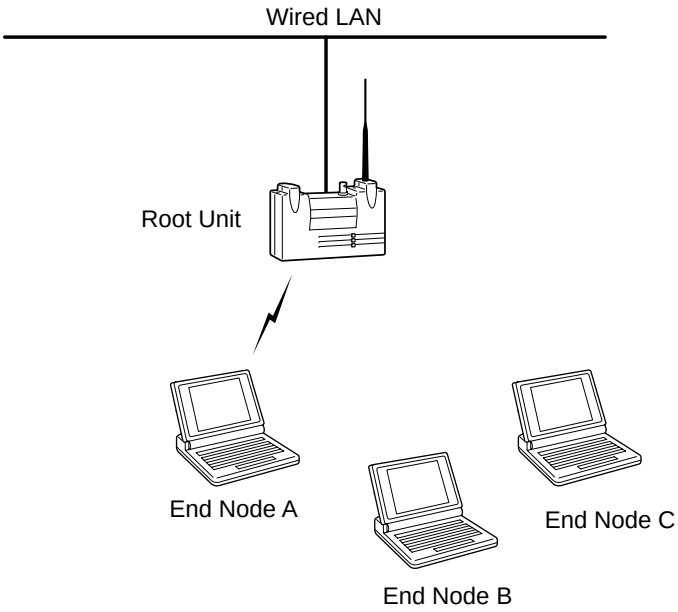
If *auto-association* is “On”, any radio node will be allowed to associate. If the parameter is “Off”, only nodes whose address matches a static table entry will be allowed to associate.

Static table entries are association table entries added manually by the operator and saved in the configuration memory. To add an entry, use “Add” on the Association Menu. “Add” supplies the address of the node that is to be controlled.

For example, suppose there is a bridge on your accounting LAN and three end nodes (A, B, and C) within radio range of the bridge. Only End Node A should be allowed access to the LAN.

1. Disable auto-association.
2. Add End Node A as a static entry. End Node A is allowed to associate to the root unit.
3. End Nodes B and C are not allowed to associate.

**Figure 10.1 - Example of Using Static Entry to Restrict Association**




As another example, suppose you only wanted to block End Node C and did not care about any other nodes. In this case you would leave auto-association “On” and add a static entry for End Node C to block it.

If you are going to use static entries to control associations, then the "association add all" command is a quick way to set up the table.

1. Leave auto-association "On" and let the nodes automatically associate to the bridge.
2. Once they have associated, select Add from the Association Menu and type "All". All entries currently in the table are now made static.
3. Turn off auto-association. You can now remove extra entries or add missing entries manually.

## *Backbone LAN Node Stale Out Time (Staletime)*

When an entry is added to the association table for a wired LAN node, a timer is started for the number of seconds specified by the value of this option. Each time a packet is received containing the same source address, the timer is restarted.

## *Specifying How Node Addresses are Displayed (NIDdisp)*

Use the *NIDdisp* option to specify how the node addresses are displayed on the Association Display Screen. The Aironet 340 Series Bridge has the ability to display node addresses as follows:

- If you specify "numeric", the addresses are displayed entirely in numeric form (default).
- If you specify "name", the Organizational Unique Identifier (OUI) portion of the address (the first three bytes) is examined to see if it is one of the known types. If it is in the list, the first three bytes will be replaced by the name of the company that owns the OUI. Otherwise the numeric value is displayed. For example, the address of a SUN workstation could be displayed as either **080020ladecc** or **Sun-ladecc**.

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CHAPTER 11

# Using Filters

This chapter describes how to control the forwarding of multicast messages.

Here's what you'll find in this chapter:

- Overview
- Using the Filter Menu
- Filtering Multicast Addresses
- Filtering Node Addresses
- Filtering Protocols

# Overview

If your Aironet 340 Series Bridge is connected to an infrastructure with a large amount of multi-protocol traffic, you may be able to reduce the amount of radio traffic by blocking out (filtering) those addresses or protocols that are not needed.

This filtering is especially important for battery operated radio nodes which might otherwise have to waste considerable battery power receiving multicast messages which are not relevant and will only be discarded.

# Using the Filter Menu

The Filter Menu is used to control the forwarding of data packets. To access this menu, select **Filter** from the Main Menu.

```
                    Filter Menu
        Option        Value      Description
    1 - Multicast   [ menu ]   - Multicast address filtering
    2 - Node        [ menu ]   - Node address filtering
    3 - Protocols   [ menu ]   - Protocol filters
    4 - Direction   [ both ]   - Packet direction affected by filters
Enter an option number or name, "=" main menu, <ESC> previous menu
>_
```

**NOTE:** In order to achieve consistent performance on your infrastructure, any configurations that you set in the Filter Menu should be duplicated on all Aironet 340 Series Bridges. This maintains consistency as nodes roam.

### *Packet Direction (Direction)*

Use the *direction* options to control the direction a packet is traveling before affected by the filters.

- **To_radio**: Only packets from the LAN will have filters applied. Packets from the radio will not be filtered. This options reduces the amount of LAN traffic to the radio network.

- **Both**: Packets in both directions will be filtered. This option allows control of the type of traffic the radio nodes may use.

## *Filtering Multicast Addresses (Multicast)*

The multicast menu allows you to control the filtering of multicasts based on the actual multicast address. When you select the *Multicast* option the Filter Multicast Menu appears.

```
                    Filter Multicast Menu
    Option             Value          Description
1 - Default     [   forward    ]- Default multicast action
2 - Show                        - Display the multicast filters
3 - Add                         - Add a multicast address filter
4 - Remove                      - Remove a multicast address filter
5 - Radio_mcst[ everywhere ]- Where to forward multicasts from radio
Enter an option number or name, “=” main menu, <ESC> previous menu
>_
```

### *Setting the Default Action (Default)*

The *default* option controls the filtering of multicasts whose address is not in the table. You may pick one of the following actions:

- **Discard**: Multicasts with no table entries will not be forwarded out the radio network.
- **Forward**: Multicasts with no table entries will be forwarded out the radio network.
- **Accesspt**: Multicasts with no table entry will only be forwarded to other access points and bridges, not to the client nodes.
- **Nonpsp**: Multicasts with no table entries will be forwarded out the radio network to non-power saving end nodes, not to any nodes using the PSP.

### *Displaying The Filters (Show)*

Use the *show* option to display the multicast filters. When you select the *show* option the Multicast Filters screen appears.

The filters are stored in the association table. The display of the multicast filters follows the format of the normal association display. At the end of each line the filter action for each address will be displayed.

This same display may also be produced with the “association display” command with either the “all” or “multicast-filters” information. See **Chapter 10** “Setting Up the Association Table”.

```
                          MULTICAST FILTERS
   Address      Device  Type    Parent    Name
------------- --------- ---- ------------ ------------------
N010203040506      Mcst                                    forward
```

### *Adding A Multicast Filter (Add)*

Use the *add* option to add a multicast filter if there are special multicast addresses you want to filter differently than the default. You will first be prompted for the address and then for an action to be applied to this address only.

### *Removing a Filter (Remove)*

Use the *remove* option to remove one or all of the non-default filters. The action for the removed entries will revert to the default action.

### *Filtering Radio Multicasts (Radio_Mcast)*

If you know that the radio nodes are not going to communicate with each other, but will only communicate with nodes on the wired LAN, set this parameter to “lan_only”. With this setting multicasts received from the radio nodes are not re-broadcast to the radio cell but are forwarded to the wired LAN.

For example, if you have a system with a large number of radio clients which only talk to the network server, enabling multicast filtering will result in much less radio traffic congestion.

If the parameter is left at the default setting of “everywhere”, then radio nodes may broadcast to each other.

## *Filtering Node Addresses (Node)*

The *node* option allows you to control the forwarding of packets based on the source node addresses. As with multicast filtering, there is a default action for those addresses not in the table. You may enter actions for specific addresses to override the default action.

Specific node filters may be entered by specifying either the 6 byte infrastructure address of the node or by specifying its IP address. If the IP address is used, the Aironet 340 Series Bridge will determine the infrastructure address associated with the IP address and use this for the actual filtering.

You may filter packets based on the source address in the received packet. For example, if you wanted to prevent all but a limited number of hosts to communicate with nodes on the radio network, you would set the default action to discard. Then add entries for the specific hosts whose action is “forward”.

```
                        Filter Node Menu
    Option       Value            Description
1 - Ethdst       [ forward ]   - Destination address from ethernet
2 - Raddst       [ forward ]   - Destination address from radio
3 - Source       [  off    ]   - Source addresses
4 - Display                    - Display the node address filters
5 - Ipdisplay                  - Display the IP address filters
6 - Add                        - Add a node address filter
7 - Remove                     - Remove a node address filter
Enter an option number or name, “=” main menu, <ESC> previous menu
>_
```

### *Setting the Destination Address (Ethdst and Raddst)*

The unit is always performing filtering based on the destination MAC address of the packets it receives. The Bridge will learn where a node is based on the source address of received packets and then can make a decision as to whether to forward a packet based on its knowledge of the location of the node.

These options set the default actions when doing destination address filtering. The ***Ethdst*** value specifies the default action for packets received on the ethernet. The ***Raddst*** action specifies the default action for packet received on the radio interface. The value allowed are *discard* or *forward*.

### *Setting the Default (Source)*

Source address filtering is “Off” by default. This saves processing power since the unit has to look up the source address of each incoming packet to see if a filter is to be applied. Before any individual source filters can be made active, one of the other values for the default must be chosen. You may set the action to *off*, *forward* or *discard*.

### *Setting Specific Node Address Filters (Add/Remove)*

Use the ***add*** option to add filters for specific addresses to the filter table.

You will be prompted for the infrastructure address or IP address of the node to which the filter applies. You are then asked whether this is a source address, radio destination address or ethernet destination address filter. Finally you are be asked for the filter action to be applied to this address which may be *off* (for remove the filter), *forward* or *discard*.

To remove one or all specific node filters use the ***remove*** option. You may enter either the keyword “all”, a single nodes infrastructure address, or a single node’s IP address. Once removed, the filter action for the removed addresses will revert to the default value.

### *Displaying The Node Address Filters (Display)*

Use the *display* option to view the table of controlled addresses. The filters are stored in the association table so that they may be accessed quickly. The display of the filters follows the format of the normal association display. At the end of each line the filter action for each address will be displayed.

This same display may also be produced using the “association display” command with either the “all” or “multicast-filters” information. See **Chapter 10** “Setting Up the Association Table”.

```
                          NODE FILTERS
   Address      Device  Type    Parent    Name               Src
------------- --------- ---- ------------ ------------------ ---
N000102030405     Unkwn                                      Fwd
Enter space to redisplay, q[uit]:
```

### *Displaying the IP to Network Address Table (IPdisplay)*

When a node address filter is entered by IP address, the Aironet 340 Series Bridge first determines the infrastructure address associated with this IP address. The actual filtering is done based on the infrastructure address.

```
                    IP ADDRESS FILTERS
   IP Address    MAC Address Src
--------------- ------------ ---
149.023.165.186  004096206892 Fwd
Enter space to redisplay, q[uit]:
```

## *Filtering Protocols (Protocols)*

Protocol filtering bases the filtering decision on the type of protocol used to encapsulate the data in the packet. This type of filtering can have the most value in almost all situations and is the preferred method of filtering. With this type of filtering you may set the Aironet 340 Series Bridge to only forward those protocols, over the radio, that are being used by the remote radio nodes. Selecting protocols is easier than setting up filters based on addresses.

The Aironet 340 Series Bridge may be set up to monitor and record the list of protocols currently being forwarded over the radio. It will record the protocols found, how may packets were encountered and whether the packet came from the LAN or the radio.

To set up the protocol filters, start the monitor and let it run for a while under normal use. Add filters by selecting the protocols from the monitor list.

There is a default action for those protocols not in the list of explicitly filtered protocols. If you know exactly which protocols are going to be used by the radio nodes, set the default action to discard and add filters to forward only those protocols that will be used. If you are not sure of all the protocols that will be used but you know that there are certain protocols you will not use, you would set the default to forward and add filters to discard only those protocols you will not use.

For filtering purposes the bridge assumes that the data portion of the packets is in one of two forms:

- The first 16 bits of the data portion contains a value that is greater than the maximum data size (1518). The value is assumed to be a protocol identifier that may be used to determine which protocol is being used within the packet.
- The first 16 bits of the data portion contains a value that is less than the maximum data size. The value is interpreted as a frame length and it is assumed that a IEEE 802.2 Logical Link Control (LLC) header follows the length.

The format of the LLC header is as follows:

DSAP, 8 bits, Destination Service Access Point (DSAP)

SSAP, 8 bits, Source Service Access Point (SSAP)

CTL, 8 bits, Control field

If the control field has a value 3 (for an un-numbered information frame), then this header may be followed by:

OUI, 24 bits, Organization Unique Identifier (OUI)

SAP-PROT, 16 bits, Protocol Identifier

You may set up filters based on either a protocol identifier or a DSAP/SSAP combination. If the filter is based on SAPs and the control field has a value of 3, the packet may also be optionally filtered based on the OUI and LLC protocol fields.

Both types of filters may also use a variable length bit mask of the packet contents to further specify which packets should be filtered.

```
                  Filter Protocols Menu
      Option       Value        Description
   1 - Default   [   off   ]   - Default action
   2 - Display                 - Display the protocol filters
   3 - Add                     - Add a protocol filter
   4 - Remove                  - Remove a protocol filter
   5 - Length    [   22    ]   - Length of packet data to log
   6 - Monitor   [   off   ]   - Protocol monitoring enabled
   7 - Show                    - Show forwarded protocol list
   8 - Clear                   - Clear forwarded protocol list
Enter an option number or name, “=” main menu, <ESC> previous menu
   >_
```

### *Setting the Default Action (Default)*

The *default* action is used for a packet whose protocol does not match any entry found in the table. It may be set to:

- **Off**: Protocol filtering is not done. It is a waste of processing power for the unit to examine each packet for its protocol only to discover no protocols need monitoring.
- **Discard**: The packet will not be forwarded out the radio network.
- **Forward**: The packet will be forwarded out the radio network.
- **Accesspt**: The packet will only be forwarded to other bridges and not to the client nodes.
- **Nonpsp**: The packet will be forwarded out the radio network to non-power saving end nodes and not to any nodes using PSP.

### *Displaying the Filters (Display)*

Use the *display* option to view the list of protocol filters you have added.

```
                        PROTOCOL FILTERS
                                      ----------LLC--------- Masks
     Name       Action   Protocol  SAPs   OUI    Protocol
-------------- --------  --------  ---- ------  ---------
 1. novell      discard    8137
 2. novell      discard              aaaa  000000    8137
 3. Ethertalk   discard              aaaa  080007    809b
 4. IPX-RIP     discard              ffff                   18- 0453, 0
Enter space to redisplay, q[uit]:
```

**Name**: The name assigned to the protocol.

**Action**: The action that has been assigned for each protocol.

**Protocol and LLC**: The protocol header.

**Masks**: A bit mask applied to the packet that must match the packet contents before the protocol is identified. The mask is displayed in the following form: 18- (start position), 0453 (value), 0 (don't care mask)

### *Adding A Filter (Add)*

Use the *add* option to add a protocol filter and specify the type of action required. There are several ways to add a filter:

- Predefined filter
- Manually add all the data
- Use an entry from the monitor table built by the unit

**➔To Add a Predefined Filter**

1. Select the *add* option.
2. Select one of the predefined strings: inet, novell, or netbios. The inet filter adds filters for both the IP and ARP protocols. The novell filter adds filters for all the types of allowed novell protocol headers.
3. You will then be prompted for the action to take when the protocol is encountered. Enter one of the actions described under the default setting above, with the exception of “Off”.

The following display shows the results if all predefined filters were added.

```
                                    ----------LLC---------
     Name       Action   Protocol  SAPs   OUI    Protocol
-------------- --------  --------  ---- ------  ---------
 1. novell      discard    8137
 2. novell      discard              aaaa  000000    8137
 3. novell      discard              ffff
 4. novell      discard              e0e0
 5. inet        forward    0800
 6. inet        forward              aaaa  000000    0800
 7. inet        forward    0806
 8. inet        forward              aaaa  000000    0806
 9. netbios     forward              f0f0
```

### ➔ To Add a Filter Using the Monitor

If protocol monitoring has been enabled, once you select the *add* command, the current monitor table will be displayed. To select a monitored protocol:

1. Enter the number displayed at the start of each line of the monitor display.
2. If the monitored protocol was un-recognized and was not given a name, you will be prompted to assign a name.
3. You will be prompted for the action to take when the protocol is encountered. Enter one of the actions described under the default setting above, with the exception of “Off”.

### ➔To Add a Filter Manually:

To start adding a filter manually:

1. Enter the *add* command and give the filter a name that does not start with a number and does not match one of the pre-defined names.
2. You will be prompted for the action to take when the protocol is encountered. Enter one of the actions described under the default setting. You may also enter the value *log*, which if chosen, the packet is not filtered, and the contents of the data portion of the packet are displayed in an information log. See “Length of Data Displayed in Log Action (Length)”. If you choose the action *high_priority* then packet*s* that match the filter will be placed on a special queue and will be transmitted before lower priority packets.
3. Choose whether the protocol is defined by an Ethernet protocol identifier or by an LLC header.

If you type “protocol”:

a. The following prompt appears:

```
Enter a value in hex from 200h to ffffh :
```

b. Type the value for the protocol identifier to be filtered and press **ENTER**.

```
Enter one of [a mask start position, none] :
```

This allows you to specify a bit mask and corresponding hexadecimal value to be applied to the packet. These two values must match the packet contents before the protocol is identified.

You must first specify a mask start position in the packet and match the mask value. The mask start position value should be a 0-based byte offset from the start of the data portion of the frame (after the MAC layer header). If you set the position to "none", no mask is tested.

c. Type a mask start position value (or "none", if applicable) and press **ENTER**.

```
Enter a hex value of 1 to 30 characters :
```

d. Type the value to be matched as a string of up to 30 hexadecimal digits and press **ENTER**. If the numbered digits is odd, the mask value will be adjusted to ignore the low 4 bits of the corresponding byte.

```
Enter a hex don't care mask of 1 to 6 characters :
```

This allows you to enter a string of hexadecimal digits to indicate which bits of the packet data are meaningful.

A bit set in this value causes the corresponding bit in the packet to be ignored. Therefore, a 0 mask means that the packet contents must exactly match the previous value entered. If the mask entered is shorter than the value entered it is automatically extended to the correct length with zeros.

e. Type the applicable hexadecimal digits and press **ENTER**.

For example, to enter a mask that matches the value 4128H in the 16th byte data portion of the packet and have the high bit of each byte ignored, complete as follows:

```
Enter one of [a mask start position, none] : 15

Enter a hex value of 1 to 30 characters : 4128
Enter a hex don't care mask of 1 to 4 characters :
8080
```

If you type **llc**:

a. When you select **llc**, the following prompt appears:

```
Enter a value in hex of ffffh or less :
```

b. Type a 16 bit value for the DSAP/SSAP combination (with the DSAP being in the high 8 bits) and press **ENTER**.

```
Enter one of [a OUI value in hex of ffffffh or
less, any] :
```

This is used to specify an OUI value to further refine the protocol identification.

If you enter “a OUI value in hex of ffffffh or less”, it must match the protocol field in addition to the SAP value.

If you enter “any”, the protocol values are not checked and the protocol is defined only by the SAP values.

c. Type the applicable OUI value or “any” and press **ENTER**. If you typed an OUI value, the following appears:

```
Enter one of [a LLC protocol value in hex of ffffh
or less, any] :
```

This is used to specify a LLC protocol identifier.

If you enter “a LLC protocol value in hex of ffffh or less”, it must match the protocol field in addition to the SAP and OUI values.

If you enter “any”, the protocol values are not checked and the protocol is defined only by the SAP and OUI values.

d. Type the applicable LLC protocol value or “any” and press **ENTER**.

e. You will be prompted for a mask description as described in the protocol section above.

### *Adding IP protocol Sub-Filters*

Once you have added a filter for the IP protocol you may also filter packets based on their UDP or TCP port number, their IP sub-protocol (i.e., UDP/TCP/ICMP) or based on an IP address range.

To filter based on an IP sub- protocol use *add ip_subprotocol*. You are then prompted as to whether you want to filter tcp or udp or another protocol based on its ID number. Finally you are prompted for a filter action as in the other manual filters.

To filter based on port number use *add ip_port*. You are prompted as to whether you want to filter a TCP or UDP source or destination port. You are then prompted for the port number and the action associated with it.

To filter based on an IP address range use *add ip_address.* You are prompted as to whether you want to filter a source or destination address and then for an IP address and address mask and a filter action. The filter matches if the value of both the packet address and the filter address ANDED with the filter mask are equal.

### *Removing an Entry (Remove)*

Use the *remove* option to remove a protocol filter entry. You may either remove all filters by entering the keyword “all” or a single entry by entering the number assigned to the filter and shown at the start of the line in the filter display.

### *Length of Data Displayed in Log Action (Length)*

Use the *length* option to display the contents of packets being forwarded to the radio.

Use this option to setup the filter mask values to properly narrow down which packets are filtered.

If you add a protocol filter whose action is “log,” each time the filter matches, the contents of the data portion of the packet (after the MAC header) will be displayed on the console (in hexadecimal) for a length in bytes determined by the value of this option.

The contents of the data portion displayed in the information log will consist of:

- "p"
- Id number of the filter shown on the Protocol Filters screen
- Bytes of the packet displayed in hexadecimal

More than one protocol at a time can be set with a filter action of "Log".

The following is an example of a protocol filter log entry:

```
p2: 01 e0 ff ff 01 eo 00 04 00 00 01 65 ff ff ff ff ff
ff 04 52 00 00
```

### *Protocol Monitoring (Monitor/ Show/ Clear)*

The Aironet 340 Series Bridge allows you to create and display a list of the protocols currently being forwarded by the unit. This allows you to test if packets that contain data for unused protocols are being forwarded to the radio nodes.

Once enabled by the *monitor* option, the Aironet 340 Series Bridge will then begin to examine the protocol used in each packet forwarded. If the protocol is not already in the list, an entry is created. Otherwise, the packet count for the given protocol is incremented.

The *show* option will display the list of currently forwarded protocols.

```
                        PROTOCOLS FOUND
                                                  --------LLC----------
      Name       Source       Count Protocol  SAPs   OUI   Protocol
---------------- ------- --------- --------  ---- ------- --------
  1. IP          RadLan        7207   0800
  2. ARP         RadLan         782   0806
  3. NetBIOS     Lan             39           f0f0
  4. ARP         RadLan          63           aaaa  000000   0806
  5. DEC MOP     Lan              3   6002
Enter space to redisplay, C[lear stats], q[uit] :
```

- **Name**: If the protocol is recognized, it will be given a name. Otherwise, the name field is left blank.

- **Source**: This will contain the string “Rad” if a packet was received from the radio and “Lan” if a packet was received from the wired LAN.
- **Count**: Displays the number of times a packet with the given protocol was encountered.
- **Protocol and LLC**: The protocol header found.

You may clear the list of found protocols either with the “clear” command or by entering a “C” (case sensitive) at the re-display prompt of the “show” command.

12

CHAPTER 12

# Setting Up Event Logs

This chapter describes how to use the Logs Menu to set up and view event logs on the Aironet 340 Series Bridge.

Here's what you'll find in this chapter:

- Overview
- Log Descriptions
- Using the Logs Menu
- Viewing History Logs
- Clearing the History Buffer
- Specifying the Type of Logs to Print/Save/Light Status LED
- Setting Statistic Parameters
- Forwarding Logs to a Unix System

# Overview

The Aironet 340 Series Bridge produces logs that record the occurrence of significant events occurring within your unit and on the infrastructure. The type of events that are recorded as logs are:

- **Information Logs**: Records status changes that occur in the normal operation of the system. For example, when an end node associates to an Aironet 340 Series Bridge.
- **Error Logs**: Records errors that occur occasionally, but are easily recovered from by the unit. For example, errors that occur during the reception and transmission of packets to and from the unit.
- **Severe Error Logs**: Records errors which drastically affect the operation of the system. The system will continue to run, but action is required to return the unit to normal operating standards.

## *Information Logs*

**BOOTP/DHCP set new IP address**

The BOOTP/DHCP server answered the request and assigned the unit an IP address different than the configured value.

**Node “node address” “device name” added**

A non-volatile entry was added to the association table.

**Node “node address” “device name” added locally “ASCII name”**

A new node associated with the local unit.

**Node “node address” “device name” restarted “ASCII name”**

A node that is currently associated to the local unit was reset.

**Node “node address” “device name” “ASCII name” removed, max radio retries**

A node was removed from the table because a response was not received from the node after attempts were made to transmit a packet to it. The node may have failed or moved to another cell.

**Node “node address” “device name” “ASCII name” removed, staled out**

A node was removed from the table because data was not received from the node within the stale-out period. Different devices have different stale-out times. PSP nodes have very short stale-out times (around 10 seconds). Non-PSP nodes have longer times (usually several minutes).

**Node “node address” “device name” “ASCII name” removed, NV removed**

A node was removed from the association table because the operator used the “association remove” command.

**Node “node address” “device name” “ASCII name” removed, deassoc notice from “address”**

The node was removed from the association table because another bridge reported that it now has the node associated locally. This log is produced whenever a node handoff occurs.

**RARP set new IP address**

A RARP server answered a request for an IP address with an address different from the one currently saved. The currently saved value is overwritten.

**Associated to router “node address”**

This log is produced when the unit, configured as a repeater, associates to its parent node.

**SNMP: “command text”**

A SNMP management node sent the unit a “set” variable request which was successfully executed. The “command text” is a similar menu command that has the same effect as the SNMP request.

**SNMP access failure from “community name” “IP address” (node address)**

A SNMP management node attempted to access the SNMP agent with an invalid community name or a name that it was not allowed to use.

**STP: Listening for other Bridges**

The spanning tree protocol is listening on the backbone port to look for other bridges in the infrastructure.

**STP: Learning Addresses**

The spanning tree protocol is listening on the backbone port. It adds any addresses it sees into the Association Table before it starts forwarding packets in order to avoid flooding packets unnecessarily.

**STP: Forwarding Data**

The spanning tree protocol has allowed the backbone port to forward data packets to the radio network.

**STP: Port Blocked**

The spanning tree protocol has determined that the backbone port must be automatically disabled to prevent a loop in the infrastructure.

**STP Port “node address” receives hello timeout**

The unit whose address is given in the log, has lost contact with the designated bridge on its LAN. It will begin to arbitrate with other bridges on the LAN to see who will take over.

**STP: Topology Changed**

Somewhere on the infrastructure a new port has been enabled or disabled. Because of these possible changes to the spanning tree, the bridge will begin using a short staleout time for backbone nodes in case the location of nodes changes.

**TFTP is loading “file name” from “ip address”**

This log is produced when the BOOTP server gives the Aironet 340 Series Bridge the name of a configuration file and then the name of a firmware file to load.

## *Error Logs*

**“Category” Error: nnn “type” errors**

This log is produced when any error occurs that is marked by an asterisk “*” after its count in the statistics displays. These errors are serious enough to affect the operation of the unit. See the sections on each display for an explanation of each error.

**Node “node address” “device name” “ascii name” removed**

These logs are similar to the information logs except that the node removed is a bridge. Since these nodes do not normally roam, it may be an indication that contact with a child repeater is lost.

**Assoctable is full**

The association table is completely full. To troubleshoot, try to force some radio nodes to associate to other bridges on the LAN using the specified router field in their association table.

**Unable to locate IP address “ip address”**

The unit was trying to send a packet to an IP address without knowing the hardware node ID. When this occurs, the unit will use the ARP protocol to try to determine the proper address. This log is produced if there was no answer to the ARP request. Usually the unit is trying to find the destination for the SNMP traps.

## *Severe Error Logs*

**Ethernet cabling problem**

If no traffic has been sent or received on the Ethernet cable in the last 10 seconds, the unit will send a packet to itself to test the connection. If the transmission succeeds, the timer is reset. If it fails, this log is produced and traffic for the connection will be discarded until the test succeeds.

**Configuration is too large to save**

The number of commands in the configuration is too large for the available non-volatile memory. This may be caused by too many non-volatile entries in the association table.

**Could not program the flash memory**

An error occurred when trying to program a new version of the firmware into flash memory. The unit must be serviced.

**EEPROM on radio is invalid**

The radio installed in each unit contains an EEPROM (Electrically Erasable Programmable Read-only Memory) chip, identifying the type of radio installed. The contents of the EEPROM were found to be invalid. Have the unit serviced. (Bad EEPROM)

**Lost our association, max radio retries**

The unit, configured as a repeater, lost communications with its parent node after trying to send a packet the maximum number of times. The unit will try to re-associate. The problem may be a parent bridge failure. All local associations will be dropped.

**Lost our association, max radio naks**

The unit, configured as a repeater, lost communications with its parent node after trying to send a packet the maximum number of times. Each time the unit sent a packet, it received a response indicating that the parent's receive buffers were full. The unit will try to re-associate. The likely cause is that the parent is handling too much traffic. All local associations will be dropped.

**Lost our association, radio restarted**

A radio configuration parameter has been changed. All associations will be dropped and the radio will be restarted.

**Lost our association, changed repeater mode**

A unit has changed from a root to a repeater or vice versa. If the unit is now a root unit, it will wait for nodes to associate to it. If the unit is now a repeater, it will attempt to associate to a parent.

**Lost our association, new specified router**

The specified router parameter of this repeater has been changed. The unit will drop its current association and try to re-associate.

**Lost our association, NAK from router**

The unit responds as though it was associated to its parent, however, the parent does not have the association. The unit will attempt to re-associate. The parent may have been rebooted.

**No response to radio loopback test**

The "config radio extended test" command was set on and no bridge in range responded to the loopback test. If you know there are units in range, then either the local radio has failed, or if there is only one remote in range, then the remote unit's radio may have failed.

**Radio Configuration Error nn**

The Aironet 340 Series Bridge could not program the radio hardware to operate at the correct frequency and bit rate. Have the unit serviced.

**Radio loopback test succeeded**

After having failed, the radio loopback test heard a response from a remote.

**The address PROM is invalid**

Each unit contains a Programmable Read-Only Memory (PROM) chip that contains the unit's hardware address. During power up, the unit was not able to read a valid address from the PROM. The unit must be serviced.

# Using the Logs Menu

The event logs are viewed using the Logs Menu. To access this menu, select **Logs** from the Main Menu.

```
                        Logs Menu
        Option           Value            Description
1  - History                 - Log and alarm history
2  - Clear                   - Clear the history buffer
3  - Printlevel [    all   ] - Type of logs to print
4  - Loglevel[       all   ] - Type of logs to save
5  -  Ledlevel   [  error/severe  ] - Type of logs to light status led
6  - Statistics              - Set alarms on statistics
7  - Network [       off   ] - Log network roaming
8  - Bnodelog[    off      ] - Log backbone node changes
9  - Snmp    [      menu   ] - Set-up SNMP traps
01 - Syslog  [149.023.165.131] - Unix syslogd address
02 - Syslevel   [  error/severe  ] - Type of logs to send to syslog
03 - Facility   [       16       ] - Syslog facility number to send
04 - Rcvsyslog  [       on       ] - Enable reception of syslog messages
 Enter an option number or name, "=" main menu, <ESC> previous menu
>_
```

## *Viewing History Logs (History)*

Use the *history* option to view history logs of events that have occurred on the unit and the infrastructure. All logs are stored within the unit in a 10KB memory buffer. The actual number of event logs the unit saves will depend on the size of each log stored in the buffer.

Log entries are always displayed in a least recent to most recent order. If the memory buffer becomes full, the oldest log in the buffer will be replaced by the most recent.

Only logs that have occurred since the unit was last powered up or since the memory buffer was cleared will be saved. See "Clearing the History Buffer (Clear)".

**NOTE:** If a power failure occurs, the logs contained in the memory will not be saved.

The display will be similar to the following:

```
OLDEST
0:00:00 I Node 004096109e30 BRE105E Floor_2_109e30 added locally
0:00:03 I Node 0040961064de AP3500-T F3_1064de added for 004096109e30
30:35:09   NEWEST, cleared at 0:00:00
b[ackward], f[orward], n[ewest], o[ldest], a[ll], C[lear], q[uit] :
```

- **First Line**: “OLDEST” indicates the end of the buffer display. This will appear at the end of the history log.
- **Display Lines**: Displays the time since power-up that the log occurred, the severity level (I-information, E-error, or S-severe) and the actual log text.
- **Last Line**: Indicates the current time and the time the buffer was last cleared by the operator. “NEWEST” indicates the start of the history log.
- **Option Line**: Indicates the movement keys to use when viewing the history logs. Since displaying the entire history will take more than a screen page, use the following keys to navigate through the history log:

  **b**: Back one page in the log

  **f**: Forward one page in the log

  **n**: Moves to the newest log entry

  **o**: Moves to the oldest log entry

  **q**: Exit the History Log screen

  **a**: Dump entire log (usually captured to a file on a PC)

## *Clearing the History Buffer (Clear)*

Use the *clear* option to delete all logs from the history buffer.

### *Specifying the Type of Logs to Print (Printlevel)*

Use the *printlevel* option to specify the type of event logs to appear on the Console screen. You will know immediately when an error or information event has occurred and then take the necessary action required.

There are four levels of logging:

- **Error/Severe**: Displays all error and severe logs.
- **Severe**: Displays severe error logs only.
- **All**: Displays all error, severe and information logs.
- **Off:** No event logs will be displayed.

### *Specifying the Type of Logs to Save (Loglevel)*

Use the *loglevel* option to specify the type of logs you want to save to memory and view on the History Log screen.

There are four levels of logging:

- **Error/Severe**: Displays all error and severe logs.
- **Severe**: Displays severe error logs only.
- **All**: Displays all error, severe and information logs.
- **Off:** No event logs will be displayed.

See “Specifying the Type of Logs to Print (Printlevel)”.

### *Specifying the Type of Logs to Light Status Indicator (Ledlevel)*

Use the *ledlevel* option to have the indicator status light turn amber when a specific type of error log occurs.

There are four levels of logging:

- **Error/Severe**: Displays all error and severe logs.
- **Severe**: Displays severe error logs only.
- **All**: Displays all error, severe and information logs.
- **Off:** No event logs will be displayed.

See “Specifying the Type of Logs to Print (Printlevel)”.

## *Setting Statistic Parameters (Statistics)*

This command allows you to control how alarms are generated based on any of the available statistics kept by the bridge. Logs may be:

- Disabled for statistics
- Generated if the statistic changes at all
- Generated if the statistic changes at a greater than specified rate

➔ **To Set Statistic Parameters:**

1. Select **statistics**. Type a number or the short form.

```
1. ra Radio
2. re Radio error
Enter one of [a number from 1 to 2, a short form]:
```

2. You will be prompted for the statistics category. Enter the number or the short form. The short form is used to store the command in the configuration.

```
                  Radio
Receive                     Transmit
1 rpa Packets           5 tpa Packets
2 rby Bytes                6 tby Bytes
3 rfi Filtered          7 ter Errors
4 rer Errors
Enter one of [a number from 1 to 7, a short form]:
```

3. Type a category number or the short form and press **ENTER**.

4. Choose the particular statistics that you wish to change. If any of the statistics already have an alarm associated, the current setting is displayed after the name.

```
Enter an action, one of [off, any, rate]:
```

5. Enter an action.

- **Off**: Turns off any alarms based on the statistics value.
- **Any**: An alarm will be generated if the statistics change value.
- **Rate**: Prompts for a rate per second change. If the statistic value changes faster than this rate, an alarm is produced.

### *Log Network Roaming (Network)*

Normally the bridges only log changes in location for radio client that move to or from this unit. If the network option is enabled, log will be produced when the unit learns of any changes to radio nodes in any of the bridges any access points in the network.

### *Logging Backbone Node changes (BnodeLog)*

Normally the bridge only log changes the state or location of its own radio nodes. If you enable this item it will produce a log whenever a backbone node is discovered or its table entry stales out.

### *Setting up SNMP traps (Snmp)*

The bridge may be set up to generate SNMP traps and send them to a network management station.

```
                        Logs Snmp Menu
        Option           Value            Description
1 - Trapdest [   none   ] - IP destination for SNMP traps
2 - Trapcomm [ "public" ] - Community for SNMP traps
3 - Loglevel [  severe  ] - Type of logs to cause a trap
4  - Authtrap   [   off       ]   - Enable authentication failure trap
 Enter an option number or name, "=" main menu, <ESC> previous menu
>_
```

Use the ***trapdest*** option to generate SNMP trap messages to a particular NMS whenever a significant event occurs.

If SNMP is enabled and the *trapdest* option is configured with a valid IP address, then the system will generate SNMP trap messages. If the *trapdest* option is set to "none," then traps will not be sent. Setting the "trapdest" parameter to address 0.0.0.0 is the same as disabling trap generation by using "none."

The following trap messages will be sent as they occur:

- A cold start trap will be sent when the unit first powers up.
- A link up trap is sent when the configuration is changed or restored for a severe error condition.
- A link down trap is sent when the configuration is changed or encounters a severe error condition.
- A link up trap is sent for an Aironet 340 Series Bridge as soon as the radio is configured.
- An authentication failure trap will be sent if an SNMP request is received with an unknown community name. This trap may be disabled by setting the “authtrap” parameter to “Off”. See “The generated trap will contain the text of the log message along with the severity of the log. See the MIB definition files for the exact layout Enabling Authentication Failure Trap (Authtrap)”.
- Any normal alarms and logs you have configured to be sent by setting the “loglevel” parameter.

**NOTE:** Since the path to the trap destination may be through a failed or not yet established radio link, it is possible that cold start and link down traps could be lost.

Use the ***trapcomm*** option to specify the community name that will be used in the trap message.

The ***LogLevel*** option configures The Aironet 340 Series Bridge to generate enterprise specific traps whenever a log of a given severity or higher is produced. The trapdest parameter must be “On”.

The generated trap will contain the text of the log message along with the severity of the log. See the MIB definition files for the exact layout Enabling Authentication Failure Trap (Authtrap)

Use the ***authtrap*** option to control the generation of SNMP authentication failure traps.

The failure traps may be sent if an NMS sends a request with an unknown community name or a community name that it is not allowed for use. You can enable or disable this option. The default setting is "Off".

## *Forwarding Logs to a Unix System (Syslog,SysLevel,Facility,Rcvsyslog)*

Use the ***syslog*** option to forward logs to a Unix host running the Syslogd deamon process. Enter the IP address of the Unix host. If the address remains at the default of 0.0.0.0., logs will not be sent. You may control the type of logs sent to the deamon with the ***Syslevel*** option which has the same arguments as the printlevel function described above.

Packets received by the Syslogd daemon process are recorded in the system log file on the Unix host. The logs are displayed on the Console in addition to being forwarded to the Unix host. If the Aironet 340 Series Bridge should fail for any reason, the logs may still be viewed on the Unix host.

The logs are sent using the syslog facility code "LOG_LOCAL0" which has a value of 16. You may change this value with the option ***Facility***. The syslog priority depends on the priority of the log locally.

On the Unix host, the Syslogd deamon process will usually add the current time and IP address of the unit that sent the log. The Aironet 340 Series Bridge will pre-pend its own name to the log before it is sent.

A message similar to the following will appear on the host:

```
Jan 11 10:46:30 192.009.200.206 A630_10172c:

Node 0000c0d1587e 630 added for 004096104546
```

By default the bridges are set up to be able to receive and display syslog messages from other bridges in the network. The ***Rcvsyslog*** option enables or disables this function. You could choose one bridge to monitor and have all other units setup with this one as their syslog host.

13

CHAPTER 13

# Performing Diagnostics

This chapter describes how to use the Diagnostics Menu to maintain the Aironet 340 Series Bridge.

Here's what you'll find in this chapter:

- Using the Diagnostics Menu
- Starting a Telnet Session
- Changing the Escape Sequence
- Running a Linktest
- Restarting the Unit
- Preparing the Unit for Shutdown
- Returning the Unit to the Default Configuration
- Physically Locating a Unit
- Sending a Ping Packet
- Loading New Code Versions

# Using the Diagnostics Menu

Diagnostics are performed using the Diagnostics Menu. To access this menu, select **Diagnostics** from the Main Menu.

```
                    Diagnostics Menu
     Option    Value        Description
1 - Network  [ menu  ]   - Network connection commands
2 - Linktest [ menu  ]   - Run a link test
3 - Restart              - Equivalent to power-up
4 - Defaults             - Return to default configuration
5 - Reset                  - Default parts of the configuration
6 - Load     [ menu  ]   - Load new version of firmware

Enter an option number or name, "=" main menu, <ESC> previous menu
     >_
```

### *Testing the Radio Link (Linktest)*

Use the *linktest* option to test the quality of the radio transmission between the Aironet 340 Series Bridge and other nodes on the radio network. See "Running a Linktest" in **Chapter 4**.

### *Restarting the Unit (Restart)*

Use the *restart* option to reboot the Aironet 340 Series Bridge. All associations will be lost and the unit will react as though it had just been powered on.

### *Returning the Unit to the Default Configuration (Default, Reset)*

Use the ***default*** option to return the Aironet 340 Series Bridge configuration to its default factory settings. The unit will erase the currently saved configuration and execute a restart command.

The ***Reset*** command can be used to only return parts of the configuration to the default state. If the argument entered is *ident_save* then all but those parts of the configuration identifying the unit (i.e., the IP address) will be defaulted. If the value is *radio_default*, then only the radio configuration is defaulted. If the value is *filter_default*, then only the filters

are defaults. You may determine into which configuration group each setting resides with the use of the *configuration dump* described in Chapter 3

# Using the Network Menu

Network connection commands are performed using the Network Menu. To access this menu, select **Diagnostics** from the Main Menu the select **Network**.

```
                    Diagnostics Network Menu
        Option          Value         Description
    1 - Connect                     - Start telnet session
    2 - Escape      [ "^X^Y^Z" ]    - Connection escape sequence
    3 - Find                        - Flash LEDs to find unit
    4 - Ping                        - Send an IP PING packet

Enter an option number or name, "=" main menu, <ESC> previous menu
    >_
```

## *Starting a Telnet Session (Connect)*

The *connect* option is used to start a telnet session with a remote unit on the infrastructure to gain access to its Console Menu. The *connect* option can also be used to access any remote node (PC or Server) that supports telnet access.

The connection may be initiated using the remote node's IP address. The connection is completely routable and the destination may be anywhere in the internet.

If the connection is to be made to another Aironet unit which has not been assigned an IP address, start the connection using the MAC level infrastructure address of the unit. This connection uses a proprietary protocol which is not routable. The destination must lie on the local LAN. This is useful when assigning IP addresses to a large number of bridges.

When starting a telnet session with the c*onnect* option:

- Make sure the telnet option on the remote is enabled before connecting to a remote bridge or client. See “Telnet Access” in **Chapter 2**.
- A message is printed on the remote’s Console stating where the connections originated from. The Console is then disabled for the duration of the telnet session to prevent conflicting commands.
- The remote’s Console privilege is set to the highest level that does not have a password.

While the unit is attempting to connect to the remote node, the connection can be terminated by typing “CTRL-C”. This may be required if the incorrect address was entered.

After connecting, you can close a telnet session and return to the local console by:

- Typing the escape sequence of characters as defined by the *escape* option in the Diagnostics Menu. See “Changing the Escape Sequence”.
- If the remote node is an Aironet node, choose the *close* option which is accessible on the Console Port Main Menu during a telnet session only.
- Using the remote node’s logout command.

## *Changing the Escape Sequence (Escape)*

Use the *escape* option to change the sequence of characters that are assigned to close a telnet session to a remote destination. Typically, you would change the sequence if the current sequence has meaning to the remote system.

The sequence may be up to 10 characters in length. To enter non-printable characters in the sequence you may:

- Use the two-character combination of caret (^) and the alphabetic character corresponding to the control character. For example, to enter “control Z”, use the string “^Z”.
- Use a backslash “\” followed by three octal numbers
- Use a dollar sign “$” followed by two hexadecimal numbers

### *Physically Locating a Unit (Find)*

Use the *find* option to blink the amber indicators of the bridge on and off. Find a unit you can telnet to if you are not sure of it's exact location. Type "CTRL-C" to stop the command.

### *Sending a Ping Packet (Ping)*

Use the *ping* option to test infrastructure connectivity from the bridge to other IP nodes. The *ping* option sends an ICMP echo_request packet to a user-specified remote node. If the remote node receives the packet it will also respond with an ICMP echo_response packet.

The Aironet 340 Series Bridge will send the echo_response packet and wait 3 seconds for a response. If none is received, another echo packet is sent. This is repeated up to five times. If a response is received and a message is displayed, the command disappears from the screen. Type "CTRL-C" to stop the command.

# Loading New Firmware and Configurations (Load)

The Aironet 340 Series Bridge code and configuration are stored in a flash memory chip inside the unit. Use the *load* option to load new code versions of the Aironet 340 Series Bridge's firmware or configurations and save them to flash memory.

To load new versions of the firmware, the code must be loaded into main memory first, then programmed into the flash memory. The unit will reboot using the new firmware. The flash memory will retain the new version even if the power is disconnected.

If the file downloaded begins with the string "! CONFIGURATION" then the file is considered to be a text file with lines of commands that are executed as though the were typed at the console.

The new firmware or configuration can be downloaded into the unit using:

- **FTP**: Load the new file into a single unit using either the Xmodem or FTP protocols. Then use the FTP protocol to upload (send) the file from the local unit to other remote units on the infrastructure.
- **Distribute**: Load the new file into a single unit using either the

Xmodem or FTP protocols. Then use the *distribute* option to simultaneously load all of the other units on the infrastructure, whether they are connected wirelessly or via the wired infrastructure.

- **Bootp**: Load the new firmware and configuration revisions into the units each time they power up.

When you select the *load* option, the Diagnostics Load Menu appears:

```
Diagnostics Load Menu                    BR105E_22ef0a
    Option             Value        Description
1 - Xmodem                        - Xmodem load from serial port
2 - Crc-xmodem                    - Xmodem-CRC load from serial port
3 - Ftp              [ menu  ] - Load using FTP
4 - Distribute       [ menu  ] - Distribute the firmware
5 - Bootp/DHCP       [  on   ] - Use BOOTP/DHCP on startup
6 - Class            [BRE105E] - DHCP class id
Enter an option number or name, "=" main menu, <ESC> previous menu
>
```

## *Downloading Using Xmodem Protocol (Xmodem/Crc-xmodem)*

Use the *Xmodem* or *CRC-xmodem* options to load the new firmware version through the Console Port.

Depending on the communications software programs available, choose:

- **Xmodem**: Terminates packets with a “checksum”
- **CRC-xmodem**: Terminates packets with a Cyclic Redundancy Check (CRC).

➔ **To load firmware using Xmodem or CRC-xmodem:**

1. Connect a terminal to the Console Port using a communications software program (Procomm™ or Windows™ Terminal).
2. Select either the *Xmodem* option or *CRC-xmodem* option, depending on your communications software.

The following message appears:

```
Ready for XMODEM download. Use several ^X's to cancel
```

3. Set the communication program to initiate the file transfer to the unit.

4. The unit begins the file download. A message similar to the following appears:

   ```
   XMODEM: received 160450 bytes in 00:03:36; 800 bytes/
   s transfer rate
   ```

After the loaded code for the new firmware is validated, the flash memory is programmed and the unit will restart with the new code.

The firmware consists of the boot block and the application code. During the firmware download, the application code is replaced, but the boot block is not.

When the unit powers up, the boot block checks the integrity of the application code. If it is valid, the boot block will execute the new firmware. If it is invalid, the boot block will display an error message on the Console and the firmware will need to be reloaded.

The only time you should receive an invalid application code is when the flash memory device fails or the power is interrupted while the flash memory is in the process of being programmed.

## *Downloading or Uploading using the File Transfer Protocol (Ftp)*

Use the *FTP* option to download or upload firmware. The Aironet 340 Series Bridge can be an FTP client or FTP server. To upload or download firmware you can initiate a connection from:

- The Aironet 340 Series Bridge console to a remote PC or host and retrieve a new version of the firmware.
- The Aironet 340 Series Bridge console to a remote PC or host and send a copy of the running firmware.
- One Aironet 340 Series Bridge console to another allowing units to send or receive firmware running locally.
- A PC or host system to an Aironet 340 Series Bridge and send a new firmware version.

**NOTE:** Before you download or upload new code versions, make sure you have set the IP address on all units involved.

When you select the *FTP* option, the Diagnostics Load FTP Menu appears:

```
                  Diagnostics Load Ftp Menu
     Option            Value            Description
1 - Get                                - Load a firmware/config file
2 - Put                                - Send a firmware file
3 - Config                             - Send a configuration file
4 - Dest       [ 000.000.000.000 ]     - Host IP address
5 - Username   [       ""        ]     - Host username
6 - Password                           - Host password
7 - Filename   [       ""        ]     - Host filename
Enter an option number or name, "=" main menu, <ESC> previous menu
>
```

## *Downloading a New Firmware/Configuration File (Get)*

Use the *get* option to download (retrieve) firmware or a configuration file. Once the file has been loaded, the unit will check the first characters of the file. If "! CONFIGURATION" is present, the file contains menu configuration commands. Otherwise the file is considered to be firmware and will be loaded in the flash memory and then executed.

**➔To Download Firmware using FTP:**

1. Load the file onto the PC, host, or bridge you will retrieve from.
2. Select the *dest* option and type in the IP address of the host PC or Aironet 340 Series Bridge.
3. Select the *username* option and type in the username required to access the firmware file.

   If downloading from another Aironet 340 Series Bridge, the *username* option must have a value even though the value is not used by the remote Aironet 340 Series Bridge.
4. Select the *password* option and type the password associated with the username.

   If downloading from another Aironet 340 Series Bridge, the login password value must match the console write privilege password on the remote Aironet 340 Series Bridge.

5. Select the *filename* option and type the name of the firmware file you are retrieving (including drive and directory), then press **ENTER**.

   If downloading from another Aironet 340 Series Bridge, the *filename* option must have a value even though the value is not used by the remote Aironet 340 Series Bridge.

6. Select the *get* option.

   The unit will begin an FTP session to the host PC, retrieve the file, program the flash memory, and reboot. A message will appear:

```
220 sun_host FTP server (SunOS 4.1) ready.
230 User sysop logged in.
200 Type set to I.
200 PORT command successful.
150 Binary data connection for apv33.img (163056 bytes).
226 Binary Transfer complete.
221 Goodbye.
FTP: received 161056 bytes in 00:00:10; 15 Kbytes/s transfer rate

rebooting unit.
```

### *Uploading a New Firmware Version (Put)*

Use the *put* option to upload (send) a copy of the currently running firmware to another system. If the system is a:

- **PC or host**: A copy of the firmware will be stored on the system's disk, possibly for downloading to other units later.
- **Aironet 340 Series Bridge**: The remote Aironet 340 Series Bridge will flash the new code and begin running it immediately. You can use one Aironet 340 Series Bridge to upgrade another Aironet 340 Series Bridge.

➔ **To Upload Firmware using FTP:**

1. Select the *dest* option and type the IP address of the remote PC, host or Aironet 340 Series Bridge you are sending to. Press **ENTER**.

2. Select the *username* option and type the username for the remote PC, host, or Aironet 340 Series Bridge you are sending to. Press **ENTER**.

   If uploading to another Aironet 340 Series Bridge, the *username* option must have a value even though the value is not used by the

remote Aironet 340 Series Bridge.

3. Select the *password* option and type the access password for the remote PC, host, or the console. Press **ENTER**.

4. Select the *filename* option type the name of the firmware file you are sending to the PC, host, or Aironet 340 Series Bridge (including drive and directory). Press **ENTER**.

   If uploading to another Aironet 340 Series Bridge, the *filename* option must have a value even though the value is not used by the remote Aironet 340 Series Bridge.

5. Select the *put* option. The unit will begin an FTP session to the remote host PC or Aironet 340 Series Bridge.

### *Uploading the Unit's configuration (Config)*

You may use this option to save the configuration on a remote host or PC in a format suitable for later downloading using FTP or BOOTP.

You are first prompted for the name of the file to be created on the remote system. Once the filename is entered the transfer will begin.

## *Downloading Using the Internet Boot Protocol (Bootp/DHCP)*

The *Bootp/DHCP* option is enabled by default when the Aironet 340 Series Bridge is powered on. The process for downloading firmware files using the Bootp/DHCP parameter is:

1. On power up, the Aironet 340 Series Bridge will issue boot protocol requests to see if there are any Bootp or DHCP servers on the infrastructure that have been configured with the unit infrastructure address.

2. If no response is found, the request is repeated up to 30 times with a 4 second wait after the first request. It then doubles the time between requests for each additional retry. If there is still no response, the unit gives up.

3. If multiple responses are received, the unit will pick a DHCP server over a Bootp server.

4. If a response is received, the IP address assigned to this unit by the server is compared to the configured value. If they are different, the configured value is changed.

5. The downloaded file is examined. If the file is not empty, it is assumed to be a configuration file in the format produced by the "configuration dump" menu command. A Trivial File Transfer Protocol (TFTP) dialogue is used to retrieve the file from the server.

6. The contents of the configuration file is processed as though the commands have been entered by the operator at the console. The commands in the file will modify the currently running configuration.

**NOTE:** The current configuration is not set back to the defaults before the file is processed. Therefore, the file contents do not have to be a complete configuration but may contain just the items you wish to change.

7. Once the configuration has been processed, the name stored in the "diagnostics load ftp filename" parameter is assumed to be the name of the firmware file to download. If the parameter is not empty, the unit will use the TFTP protocol to load the file into RAM.

    - If the firmware is different from the currently running version, the unit will program the flash memory with the new code and restart to execute it.
    - If the new firmware is the same, the unit discards the loaded file and continues normal operation

Use the ***class*** option to enter a class ID for a client node. The entered string is placed in the DHCP discover messages sent to the DHCP servers. The server will determine how to respond based on the class ID.

## *Distributing Firmware or Configuration (Distribute)*

```
            Diagnostics Load Distribution Menu
        Option      Value       Description
    1 - Go                    - Start the distribution
    2 - Type     [firmware]   - What to distribute
    3 - Control  [ "newer"]   - How to control distributions
    4 - Add                   - Change distributable configuration
    5 - Remove                - Remove change
    6 - Show                  - Show changes
    7 - Dump                  - Show Configuration
```

Use the ***distribute*** option to send the firmware or configuration from one Aironet 340 Series Bridge to all other Aironet 340 Series Bridges on the infrastructure (whether they are repeaters or are connected to the wired infrastructure). By using the *distribute* function the time needed to perform firmware upgrades or make global changes to the configuration is greatly decreased.

Once a new version of the firmware has been loaded into a single Aironet 340 Series Bridge, (using Xmodem, CRC-Modem, Ftp or Bootp) or the configuration has changed, use the *distribute* option to upgrade all other units.

### *Controlling Which Units the Distribute Affects*

The ***control*** option controls how the remote units respond to your request to send them a configuration or firmware. Values may be set to:

- **None**: The unit will never respond and cannot be loaded by another unit using the distribute command.
- **Any**: The unit will always respond. It is up to the distributing unit to determine whether to load the local unit.
- **Newer**: The unit will only respond if the version of firmware being distributed has a larger version number than the code currently running. This selection only applies to firmware downloads. For configuration downloads this is equivalent to “any”.
- **None of the Above**: It is interpreted as a password that must have been entered by the operator of the unit doing the distribution. The local unit will not respond to any distributions that do not supply this password.

If the distribution is password protected, only those units that have the same password configured into the *control* parameter will accept the distribution. In this way you may protect your units from unwanted loads. The password may also be used to divide the units into code load groups so the loads to one group will not affect the other groups.

If the distribution is done without a password, the load will be ignored by remote units with a configured password. If the remote unit does not have a password and firmware is being distributed, it will still accept the load based on the version number and code checksum.

## *Controlling Which Parts of the Configuration are Distributed*

By default certain parts of the configuration have been set to being part of the distributable configuration. The distribution always contains all configuration items marked this way.

Each configuration item in the bridge has been assigned a unique id number that will never change over the life of the product. It is these numbers along with the arguments to the configuration commands that are distributed so that menu changes do not affect the configuration.

Use the ***Dump*** command to display the status of the entire configuration. Each line will start with the id number for the item and its arguments. Then in a comment field the string “local” will appear if the item in not distributable and “sent” if it is. Following this will be the current full text for the configuration item.

To change which items are distributable use the ***add/remove/show*** commands. The Add command asks for a configuration id and whether it is to be sent or is local only. The Remove command asks for an id or “all” and returns the item to its default state. The Show command displays the table of changes.

## *Starting The Distribution*

To start the distribution use the ***Go*** option The following message will appear:

```
Finding the other units ...
```

When the command is executed, the local unit will send a special

broadcast message similar to the one below to all other units on the infrastructure. It reports that it has a new firmware file with its assigned version number.

```
BR105E 004096001d45 has code version 3.2a (checksum
1598)
```

The remote units then decide whether to respond based on the value of their rcv_distribute parameter.

When the local unit receives a response to its request, the remote unit is added to a list of units to be loaded. When the response time-out period has expired (approximately 10 seconds), the local unit will begin loading all remote units in parallel using a proprietary protocol. A message similar to the one below will be displayed.

```
Loading 004096001d45
Loading 004096001d45
```

If any remote units timeout during the load, they are removed from the list. Once all units have completed loading, the local unit displays a count of the successful loads. A message similar to the following will be displayed.

```
Completed loading 004096001d45
Completed loading 00409610345f
Loading of 2 Aironet 340 Series Bridges completed
```

# Appendix A - Aironet 340 Series Bridge Specifications

## LAN Interfaces Supported

### *Ethernet*

| Cable | Specifications | Connector |
|---|---|---|
| Thin Ethernet | IEEE 802.3 10Base2 | BNC Connector |
| Thick Ethernet | IEEE 802.3 10Base5 | DB-15 Connector<br>(external Transceiver required) |
| Twisted Pair Ethernet | IEEE 802.3 10BaseT | RJ-45 Connector |

## Radio Characteristics

| Item | Aironet 340 Series Bridge |
|---|---|
| Frequency | 2.400 to 2.497 GHz* |
| Modulation | Direct Sequence Spread Spectrum |
| Antenna | The bridge ships with a single dipole antenna (2.2 dBi gain). Longer range antennas are available. |
| Compliance | The bridge operates license-free under FCC Part 15 and complies as a Class B computing device. Complies with DOC regulations.<br><br>The bridge complies with ETS 300.328, FTZ 2100 and MPT 1349 standards (and others). |

# Physical Specifications

| Item | Description |
|---|---|
| Size | 20 x 15 x 5 cm (7.8 x 5.9 x 1.9 inches) |
| Status Indicators | Top Panel – Radio Traffic activity, Ethernet Traffic activity, Status<br>Back Panel – Ethernet Rx and Tx activity, Polarity, Port connections, Collisions |
| Console Port | DCE with DB-9 female connector |
| Power Supply | Power Pack. The power pack will be either 120VAC/60Hz or 90-264VAC/47-63Hz to 12-18VDC, whichever is appropriate for country of use. |
| Weight | 0.7 Kg (1 lb. 8 oz.) |
| Operating environment | -20$^{o}$C to 50$^{o}$C (-4$^{o}$F to 122$^{o}$F) |

## Console Port Pin-Out

The Console Port is a DCE using a DB-9 female connector. The following table describes the pinouts on the connector and how you should connect the DB-9 pins to the DB-25 on a terminal. Signal names are in terms of the DTE.

| Signal | DB-9 Male Aironet Console Port | DB-25 Female Computer Serial Port |
|---|---|---|
| RxD | 2 | 3 |
| TxD | 3 | 2 |
| GND | 5 | 7 |
| DCD | 1 | 8 |
| DTR | 4 | 20 |
| CTS | 8 | 5 |
| RTS | 7 | 4 |

| Signal | DB-9 Male Aironet Console Port | DB-9 Female Computer Serial Port |
|---|---|---|
| RxD | 2 | 2 |
| TxD | 3 | 3 |
| GND | 5 | 5 |
| DCD | 1 | 1 |
| DTR | 4 | 4 |
| CTS | 8 | 8 |
| RTS | 7 | 7 |

Most terminals and communication programs will only require Txd, Rxd and Gnd to communicate with the Aironet 340 Series Bridge. Some may also require DCD before the connection on-line can be made. If you use hardware flow control, connect all lines.

# Appendix B - Console Menu Tree

The Console system consists of multiple sub-menus that branch off the Main Menu, much like a tree. This Appendix provides you with a detailed listing of all menu, sub-menus and options contained in the Console Port.

**Main Menu**

- Configuration - General configuration
  - Radio - Radio network parameters
    - Ssid - Service set identification
    - Root - Enable root mode
    - Rates - Allowed bit rates in megabits/second
    - Basic_rates - Basic bit rates in megabits/second
    - Distance - Maximum separation in kilometers
    - I80211 - 802.11 parameters
      - Beacon - Beacon period in Kusec
      - Dtim - DTIM interval
      - Extend - Allow proprietary extensions
      - Bcst_ssid - Allow broadcast SSID
      - Rts - RTS/CTS packet size threshold
      - Privacy - Privacy configuration
        - Encryption - Encrypt radio packets
        - Auth - Authentication mode
        - Client - Client authentication modes allowed
        - Key - Set the keys
        - Transmit - Key number for transmit
      - Encapsulation - Configure packet encapsulation
        - Encap - Default encapsulation method
        - Show - Show encapsulation table
        - Add - Add a protocol encapsulation method
        - Remove - Remove a protocol encapsulation method
    - Linktests - Test the radio link
      - Strength - Run a signal strength test
      - Carrier - Carrier busy statistics
      - Align - Antenna alignment test
      - Multicast - Run a multicast echo test
      - Unicast - Run a unicast echo test
      - Remote - Run a remote echo test
      - Destination - Target address
      - Size - Packet size
      - Count - Number of packets to send
      - Rate - Data rate
      - Errors - Radio error statistics
      - Autotest - Auto echo test

```
    Continuous          - Repeat echo test once started
  Extended              - Extended parameters
    Bridge_mode         - Bridging mode
    Parentid            - Parent node Id
    Parent_timeout      - Time to look for specified parent
    Parent_wait         - How long to look for previous parent
    Time_retry          - Number of seconds to retry transmit
    Count_retry         - Maximum number transmit retries
    Refresh             - Refresh rate in 1/10 of seconds
    Roaming             - Type of roaming control packets
    Balance             - Load balancing
    Diversity           - Enable the diversity antennas
    Power               - Transmit power level
    Fragment            - Maximum fragment size
    Options             - Enable radio options
Ethernet                - Ethernet configuration
  Active                - Connection active
  Size                  - Maximum frame size
  Port                  - Port selection
Ident                   - Identification information
  Inaddr                - Internet address
  Inmask                - Internet subnet mask
  Gateway               - Internet default gateway
  Routing               - IP routing table configuration
    Display             - Display Route Table Entries
    Host                - Add a Static Host Route
    Net                 - Add a Static Network Route
    Delete              - Delete a Static Route
  Dns1                  - DNS server 1
  Dns2                  - DNS server 2
  Domain                - Domain name
  Master                - Master unit address
  Nid                   - Network address
  Name                  - Node name
  Location              - System location
  Contact               - System contact name
  Bootp_DHCP            - Use BOOTP/DHCP on startup
  Class                 - DHCP class id
Console                 - Control console access
  Rpassword             - Set readonly privilege password
  Wpassword             - Set write privilege password
  Remote                - Allow remote operators
  Telnet                - Allow telnet connections
  Http                  - Allow http connections
  Display               - Display the remote operator list
```

```
  Add              - Add an operator host
  Delete           - Remove an operator host
  Communities        - SNMP community properties
    Display        - Display SNMP communities
    Add            - Add a community
    Remove           - Remove a community
    Access         - Set community access mode
    Remote           - Allow remote NMS to change community info
  Type             - Terminal type
  Port             - Serial port set-up
    Rate           - Console baud rate
    Bits           - Bits per character
    Parity         - Console parity
    Flow           - Flow control type
  Linemode           - Console expects complete lines
Stp                - Spanning Tree Protocol
  Active           - Protocol enabled
  Display          - Protocol status
  Priority         - Bridge priority
  Hello_time         - Hello message interval
  Forward_delay        - Forwarding delay
  Msg_age_timeout      - Receive hello message timeout
  Port             - Port parameters
    Port           - Protocol enabled for ethernet port
    Priority       - Local ethernet port priority
    Cost           - Local ethernet port cost
    Port           - Protocol enabled for token ring port
    Priority       - Local token ring port priority
    Cost           - Local token ring port cost
    Rport          - Protocol enabled for remote port
    Rpriority        - Remote port priority
    Rcost          - Remote port cost
  Port             - Protocol enabled for ethernet port
  Priority         - Local ethernet port priority
  Cost             - Local ethernet port cost
  Port             - Protocol enabled for token ring port
  Priority         - Local token ring port priority
  Cost             - Local token ring port cost
Mobile-IP            - Mobile IP Protocol Configuration
  AgentType          - Home / Foreign Agent
  Mobile           - Home Agent Active Mobile Nodes
  Visitors         - Foreign Agent Visitor List
  Add              - Add Mobile Nodes
  Remove             - Remove Mobile Nodes
  Display          - Display Home Agent Authorized Addresses
  Setup            - Agent Configuration
    Lifetime       - Max Registration Lifetime
    ReplayProt       - Replay Protection Method
```

```
        Broadcasts          - Broadcast Forwarding
        RegRequired         - Registration Required
        HostRedirects       - Enable ICMP Host Redirects to MN
      Advert                - Advertisement Setup
        AdvertType          - Advertisement type
        AdvertInterval      - Advertisement interval
        PrefixLen           - Advertise prefix length extension
        AdvertRtrs          - Advertise routers
    Time                    - Network Time Setup
      Time_server           - Time protocol server
      Sntp_server           - Network time server
      Offset                - GMT offset in minutes
      Dst                   - Use daylight savings time
    Dump                    - Dump configuration to console
  Statistics                - Display statistics
    Throughput              - Throughput statistics
    Radio                   - Radio error statistics
    Ethernet                - Ethernet error statistics
    Status                  - Display general status
    Map                     - Show network map
    Watch                   - Record history of a statistic
    History                 - Display statistic history
    Nodes                   - Node statistics
    ARP                     - ARP table
    Display_time            - Time to re-display screens
    IpAdr                   - Determine client IP addresses
  Association               - Association table maintenance
    Display                 - Display the table
    Summary                 - Display the table summary
    Maximum                 - Maximum allowed child nodes
    Autoreg                 - Allow automatic table additions
    Add                     - Control node access
    Remove                  - Remove access control
    Staletime               - Backbone LAN node stale out time
    Niddisp                 - Node Ids display mode
  Filter                    - Control packet filtering
    Multicast               - Multicast address filtering
      Default               - Default multicast action
      Show                  - Display the multicast filters
      Add                   - Add a multicast address filter
      Remove                - Remove a multicast address filter
      Radio_mcst            - Where to forward multicasts from radio
    Node                    - Node address filtering
      Ethdst                - Destination address from ethernet
      Tokdst                - Destination address from token ring
      Raddst                - Destination address from radio
      Source                - Source addresses
      Display               - Display the node address filters
```

```
    Ipdisplay          - Display the IP address filters
    Add                - Add a node address filter
    Remove             - Remove a node address filter
  Protocols            - Protocol filters
    Default            - Default action
    Unicast            - Filter unicast packets
    Display            - Display the protocol filters
    Add                - Add a protocol filter
    Remove             - Remove a protocol filter
    Length             - Length of packet data to log
    Monitor            - Protocol monitoring enabled
    Show               - Show forwarded protocol list
    Clear              - Clear forwarded protocol list
  Direction            - Packet direction affected by filters
Logs                   - Alarm and log control
  History              - Log and alarm history
  Clear                - Clear the history buffer
  Printlevel           - Type of logs to print
  Loglevel             - Type of logs to save
  Ledlevel             - Type of logs to light status led
  Statistics           - Set alarms on statistics
  Network              - Log network roaming
  Bnodelog             - Log backbone node changes
  Snmp                 - Set-up SNMP traps
    Trapdest           - IP destination for SNMP traps
    Trapcomm           - Community for SNMP traps
    Loglevel           - Type of logs to cause a trap
    Authtrap           - Enable authentication failure trap
  Syslog               - Unix syslogd address
  Syslevel             - Type of logs to send to syslog
  Facility             - Syslog facility number to send
  Rcvsyslog            - Enable reception of syslogmessages
Diagnostics            - Maintenance and testing commands
  Network              - Network connection commands
    Connect            - Start telnet session
    Escape             - Connection escape sequence
    Ping               - Send an IP PING packet
    Find               - Flash LEDs to find unit
  Linktests            - Test the radio link
    Strength           - Run a signal strength test
    Carrier            - Carrier busy statistics
    Align              - Antenna alignment test
    Multicast          - Run a multicast echo test
    Unicast            - Run a unicast echo test
    Remote             - Run a remote echo test
```

```
    Destination          - Target address
    Size                 - Packet size
    Count                - Number of packets to send
    Pattern              - Packet data pattern
    Rate                 - Data rate
    Errors               - Radio error statistics
    Autotest             - Auto echo test
    Continuous           - Repeat echo test once started
  Restart                - Restart the unit
  Shutdown               - Prepare to power off unit
  Defaults               - Return to default configuration
  Reset                  - Default parts of the configuration
  Load                   - Load new version of firmware
    Xmodem               - Xmodem load from serial port
    Crc-xmodem           - Xmodem-CRC load from serial port
    Ftp                  - Load using FTP
      Get                - Load a firmware/config file
      Put                - Send a firmware file
      Config             - Send a configuration file
      Dest               - Host IP address
      Username           - Host username
      Password           - Host password
      Filename           - Host filename
    Distribute           - Distribute the firmware
      Go                 - Start a distribution
      Type               - What to distribute
      Control            - How to control distributions
      Add                - Change distributable configuration
      Remove             - Remove change
      Show               - Show changes
      Dump               - Show Configuration
    Bootp_DHCP           - Use BOOTP/DHCP on startup
    Class                - DHCP class id
Privilege                - Set privilege level
Close                    - Close the telnet session
Exit                     - Exit the menus
Help                     - Introduction
```

# Appendix C - SNMP Variables

The Aironet 340 Series Bridge supports the Simple Network Management Protocol (SNMP). SNMP provides an industry standard mechanism for the exchange of information in a TCP/IP based internet environment.

The resident SNMP agent is compliant with subsets of the (Management Information Base) MIB-I and MIB-II for TCP/IP based internets as defined in Internet's Request For Changes (RFC) 1156 and 1213. Since the Aironet 340 Series Bridge does not perform any IP routing or forwarding, certain (groups of) managed objects are not meaningful. For SNMP requests pertaining to such managed objects, the node simply returns a "no such name" error status in the response.

The Object ID (OID) prefix for the Aironet 340 Series Bridge resides under the Structure of Managed Information (SMI) tree for private enterprises in the Telxon.arlan.devices (551.2.1) branch. The system object identifier for the Aironet 340 Series Bridge is (1.3.6.1.4.1.551.2.1.76). The resident agent also supports a custom MIB that allows a management station to read/modify most of the parameters that may be set through the Console Menus. For a machine readable version of the custom MIB, contact Aironet Wireless Communications.

## C.1 MIB II Variables

## The System Group

MIBII.system (1.3.6.1.2.1.1.x)

| Object ID | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | sysDescr | string | read |
| 2 | sysObjectID | oid | read |
| 3 | sysUpTime | time | read |
| 4 | sysContact | string | write |
| 5 | sysName | string | write |
| 6 | sysLocation | string | write |
| 7 | sysServices | integer | read |

# The Interfaces Group

MIBII.interfaces (1.3.6.1.2.1.2.x)

| Object ID | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | ifNumber | integer | read |
| 2 | ifTable | Sequence of if | entry |
| 2.1 | ifEntry | Sequence | entry |
| 2.1.1 | ifIndex | integer | read |
| 2.1.2 | ifDescr | string | read |
| 2.1.3 | ifType | integer | read |
| 2.1.4 | ifMtu | integer | read |
| 2.1.5 | ifSpeed | gauge | read |
| 2.1.6 | ifPhysAddress | string | read |
| 2.1.7 | ifAdminStatus | integer | read |
| 2.1.8 | ifOperStatus | integer | read |
| 2.1.9 | ifLastChange | time | read |
| 2.1.10 | ifInOctets | counter | read |
| 2.1.11 | ifInUcastPkts | counter | read |
| 2.1.12 | ifInNUcastPkts | counter | read |
| 2.1.13 | ifInDiscards | counter | read |
| 2.1.14 | ifInErrors | counter | read |
| 2.1.15 | ifInUnknownProtos | counter | read |
| 2.1.16 | ifOutOctets | counter | read |
| 2.1.17 | ifOutUcastPkts | counter | read |
| 2.1.18 | ifOutNUcastPkts | counter | read |
| 2.1.19 | ifOutDiscards | counter | read |
| 2.1.20 | ifOutErrors | counter | read |
| 2.1.21 | ifOutQLen | gauge | read |
| 2.1.22 | ifSpecific | integer | read |

# The Address Translation Group (deprecated by MIB-II)

MIBII.at (1.3.6.1.2.1.3.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | atTable | Sequence of at | entry |
| 1.1 | atEntry | Sequence | entry |
| 1.1.1 | atIfIndex | integer | read |
| 1.1.2 | atPhysAddress | string | read |
| 1.1.3 | atNetAddress | ipaddress | read |

# The IP Group

MIBII.ip (1.3.6.1.2.1.4.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | ipForwarding | integer | read |
| 2 | ipDefaultTTL | integer | write |
| 3 | ipInReceives | counter | read |
| 4 | ipInHdrErrors | counter | read |
| 5 | ipInAddrErrors | counter | read |
| 6 | ipForwDatagrams | counter | read |
| 7 | ipInUnknownProtos | counter | read |
| 8 | ipInDiscards | counter | read |
| 9 | ipInDelivers | counter | read |
| 10 | ipOutRequests | counter | read |
| 11 | ipOutDiscards | counter | read |
| 12 | ipOutNoRoutes | counter | read |
| 13 | ipReasmTimeout | integer | read |
| 14 | ipReasmReqds | counter | read |
| 15 | ipReasmOKs | counter | read |
| 16 | ipReasmFails | counter | read |
| 17 | ipFragOKs | counter | read |
| 18 | ipFragFails | counter | read |
| 19 | ipFragCreates | counter | read |
| 20 | ipAddrTable | Sequence of | ipAd-drEntry |
| 20.1 | ipAddrEntry | Sequence | ipAd-drEntry |
| 20.1.1 | ipAdEntAddr | ipaddress | read |
| 20.1.2 | ipAdEntIfIndex | integer | read |
| 20.1.3 | ipAdEntNetMask | ipaddress | read |
| 20.1.4 | ipAdEntBcastAddr | integer | read |

# The ICMP Group

MIBII.icmp (1.3.6.1.2.1.5.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | icmpInMsgs | counter | read |
| 2 | icmpInErrors | counter | read |
| 3 | icmpInDestUnreachs | counter | read |
| 4 | icmpInTimeExcds | counter | read |
| 5 | icmpInParmProbs | counter | read |
| 6 | icmpInSrcQuenchs | counter | read |
| 7 | icmpInRedirects | counter | read |
| 8 | icmpInEchos | counter | read |
| 9 | icmpInEchoReps | counter | read |
| 10 | icmpInTimestamps | counter | read |
| 11 | icmpInTimestampReps | counter | read |
| 12 | icmpInAddrMasks | counter | read |
| 13 | icmpInAddrMaskReps | counter | read |
| 14 | icmpOutMsgs | counter | read |
| 15 | icmpOutErrors | counter | read |
| 16 | icmpOutDestUnreachs | counter | read |
| 17 | icmpOutTimeExcds | counter | read |
| 18 | icmpOutParmProbs | counter | read |
| 19 | icmpOutSrcQuenchs | counter | read |
| 20 | icmpOutRedirects | counter | read |
| 21 | icmpOutEchos | counter | read |
| 22 | icmpOutEchoReps | counter | read |
| 23 | icmpOutTimestamps | counter | read |
| 24 | icmpOutTimestampReps | counter | read |
| 25 | icmpOutAddrMasks | counter | read |
| 26 | icmpOutAddrMaskReps | counter | read |

## The UDP Group

MIBII.udp (1.3.6.1.2.1.7.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | udpInDatagrams | counter | read |
| 2 | udpNoPorts | counter | read |
| 3 | udpInErrors | counter | read |
| 4 | udpOutDatagrams | counter | read |

## The Transmission Group

MIBII.transmission.dot3 (1.3.6.1.2.1.10.7.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | dot3Table | Sequence of dot3 | entry |
| 1.1 | dot3Entry | Sequence | entry |
| 1.1.1.1 | dot3Index | integer | read |
| 1.1.3.1 | dot3MacSubLayerStatus | integer | write |
| 2 | dot3StatsTable | Sequence of dot3Stats | entry |
| 2.1 | dot3StatsEntry | Sequence | entry |
| 2.1.1.1 | dot3StatsIndex | integer | read |
| 2.1.2.1 | dot3StatsAlignmentErrors | counter | read |
| 2.1.3.1 | dot3StatsFCSErrors | counter | read |
| 2.1.4.1 | dot3StatsSingleCollisionFrames | counter | read |
| 2.1.5.1 | dot3StatsMultipleCollisionFrames | counter | read |
| 2.1.6.1 | dot3StatsSQETestErrors | counter | read |
| 2.1.7.1 | dot3StatsDeferredTransmissions | counter | read |
| 2.1.8.1 | dot3StatsLateCollisions | counter | read |
| 2.1.9.1 | dot3StatsExcessiveCollisions | counter | read |
| 2.1.10.1 | dot3StatsInternalMacTransmitErrors | counter | read |
| 2.1.11.1 | dot3StatsCarrierSenseErrors | counter | read |
| 2.1.12.1 | dot3StatsExcessiveDeferrals | counter | read |
| 2.1.13.1 | dot3StatsFrameTooLongs | counter | read |
| 2.1.14.1 | dot3StatsInrangeLengthErrors | counter | read |
| 2.1.15.1 | dot3StatsOutOfRangeLengthFields | counter | read |
| 2.1.16.1 | dot3StatsInternalMacReceiveErrors | counter | read |

# The SNMP Group

MIBII.snmp (1.3.6.1.2.1.11.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | snmpInPkts | counter | read |
| 2 | snmpOutPkts | counter | read |
| 3 | snmpInBadVersions | counter | read |
| 4 | snmpInBadCommunityNames | counter | read |
| 5 | snmpInBadCommunityUses | counter | read |
| 6 | snmpInASNParseErrs | counter | read |
| 7 | snmpInBadTypes | counter | read |
| 8 | snmpInTooBigs | counter | read |
| 9 | snmpInNoSuchNames | counter | read |
| 10 | snmpInBadValues | counter | read |
| 11 | snmpInReadOnlys | counter | read |
| 12 | snmpInGenErrs | counter | read |
| 13 | snmpInTotalReqVars | counter | read |
| 14 | snmpInTotalSetVars | counter | read |
| 15 | snmpInGetRequests | counter | read |
| 16 | snmpInGetNexts | counter | read |
| 17 | snmpInSetRequests | counter | read |
| 18 | snmpInGetResponses | counter | read |
| 19 | snmpInTraps | counter | read |
| 20 | snmpOutTooBigs | counter | read |
| 21 | snmpOutNoSuchNames | counter | read |
| 22 | snmpOutBadValues | counter | read |
| 23 | snmpOutReadOnlys | counter | read |
| 24 | snmpOutBadGenErrs | counter | read |
| 25 | snmpOutGetRequests | counter | read |
| 26 | snmpOutGetNexts | counter | read |
| 27 | snmpOutSetRequests | counter | read |
| 28 | snmpOutGetResponses | counter | read |
| 29 | snmpOutTraps | counter | read |
| 30 | snmpEnableAuthenTraps | integer | write |

# The Configure STP Group

MIBII.dot1dBridge.dot1dStp (1.3.6.1.2.1.17.2.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | dot1dStpProtocolspecification | integer | read |
| 2 | dot1dStpPriority | integer | write |
| 3 | dot1dStpTimeSinceTopologyChange | integer | read |
| 4 | dot1dStptopChanges | integer | read |
| 5 | dot1dStpDesignatedRoot | string | read |
| 6 | dot1dStpRootcost | integer | read |
| 7 | dot1dStpRootPort | integer | read |
| 8 | dot1dStpMaxAge | integer | read |
| 9 | dot1dStpHelloTime | integer | read |
| 10 | dot1dStpHoldTime | integer | read |
| 11 | dot1dStpForwardDelay | integer | read |
| 12 | dot1dStpBridgeMaxAge | integer | write |
| 13 | dot1dStpBridgeHelloTime | integer | write |
| 14 | dot1dStpBridgeForwardDelay | integer | write |
| 15 | dot1dStpPortTable | Sequence of dot1dStpPortEntry | |
| 15.1 | dot1dStpPortEntry | Sequence | |
| 15.1.1 | dot1dStpPortPriority | integer | read |
| 15.1.2 | dot1dStpPortState | integer | write |
| 15.1.3 | dot1dStpPortState | integer | read |
| 15.1.4 | dot1dStpPortEnable | integer | write |
| 15.1.5 | dot1dStpPortPathCost | integer | write |
| 15.1.6 | dot1dStpPortDesignatedRoot | string | read |
| 15.1.7 | dot1dStpPortDesignatedCost | integer | read |
| 15.1.8 | dot1dStpPortDesignatedBridge | string | read |
| 15.1.9 | dot1dStpPortDesignatedPort | integer | read |
| 15.1.10 | dot1dStpPortForwardTransmissions | integer | read |

MIBII.dot1dBridge.dot1dTp (1.3.6.1.2.1.17.4.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | dot1dTpLearnedEntryDiscards | counter | read |
| 2 | dot1dTpAgingTime | integer | write |
| 3 | dot1dTpFdbTable | Sequence of dot1dTpFdEntry | |
| 3.1 | dot1dTpFdbEntry | Sequency | |
| 3.1.1 | dot1dTpFdAddress | string | read |
| 3.1.2 | dot1dTpFdbPort | integer | read |
| 3.1.3 | dot1dTpFdbStatus | integer | read |

## 3.2 The ARLAN Custom MIB

### The Configure Ethernet Group

ACCESSPOINT.configuration.cfgEthernet (1.3.6.1.4.1.551.2.2.1.1.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | cfgEthEnable | integer | write |
| 2 | cfgEthSize | integer | write |

### The Configure ARLAN Group

ACCESSPOINT.configuration.cfgArlan (1.3.6.1.4.1.551.2.2.1.2.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | cfgArlRoot | integer | write |
| 7 | cfgArlParent | string | write |
| 8 | cfgArlParentTime | integer | write |
| 16 | cfgArlSsid | String | write |

### The Configure Filtering Group

ACCESSPOINT.configuration.cfgFilter (1.3.6.1.4.1.551.2.2.1.3.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | cfgFiltMcst | integer | write |
| 7 | cfgFiltSrc | integer | write |

## The Configure Console Group

ACCESSPOINT.configuration.cfgConsole (1.3.6.1.4.1.551.2.2.1.4.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | cfgConsPrivilege | integer | write |
| 2 | cfgConsReadPwd | string | write |
| 3 | cfgConsWritePwd | string | write |
| 4 | cfgConsType | integer | write |
| 5 | cfgConsBaud | integer | write |
| 6 | cfgConsBits | integer | write |
| 7 | cfgConsParity | integer | write |
| 9 | cfgConsTelnet | integer | write |
| 11 | cfgConsFlow | integer | write |

## The Configure SNMP Group

ACCESSPOINT.configuration.cfgSnmp (1.3.6.1.4.1.551.2.2.1.5.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | cfgSnmpDest | ipaddress | write |
| 2 | cfgSnmpAuth | integer | write |
| 3 | cfgSnmpTComm | string | write |
| 4 | cfgSnmpLog | integer | write |
| 5 | cfgSnmpCommTable | Sequence of cfgSnmp-CommTableEntry | |
| 5.1 | cfgSnmpCommTableEntry | Sequence | |
| 5.1.1 | cfgSnmpCommStatus | integer | write |
| 5.1.2 | cfgSnmpCommIndex | integer | write |
| 5.1.3 | cfgSnmpCommName | string | write |
| 5.1.4 | cfgSnmpCommAccess | integer | write |
| 5.1.5 | cfgSnmpCommIP1 | ipaddress | write |
| 5.1.6 | cfgSnmpCommIP2 | ipaddress | write |
| 5.1.7 | cfgSnmpCommIP3 | ipaddress | write |
| 5.1.8 | cfgSnmpCommIP4 | ipaddress | write |
| 5.1.9 | cfgSnmpCommIP5 | ipaddress | write |
| 5.1.10 | cfgSnmpCommNID1 | string | write |

| | | | |
|---|---|---|---|
| 5.1.11 | cfgSnmpCommNID2 | string | write |
| 5.1.12 | cfgSnmpCommNID3 | string | write |
| 5.1.13 | cfgSnmpCommNID4 | string | write |
| 5.1.14 | cfgSnmpCommNID5 | string | write |

## The Configure Logs Group

ACCESSPOINT.configuration.cfgLogs (1.3.6.1.4.1.551.2.2.1.6.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | cfgLogPrint | integer | write |
| 2 | cfgLogSave | integer | write |
| 3 | cfgLogLed | integer | write |
| 5 | cfgLogClear | integer | write |
| 6 | cfgLogStatusLock | integer | write |
| 7 | cfgLogBnodeLog | integer | write |
| 8 | cfgLogSyslog | ipaddress | write |

## The Configure Association Table Group

ACCESSPOINT.configuration.cfgAssociation (1.3.6.1.4.1.551.2.2.1.7.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | cfgRegAutoReg | integer | write |
| 2 | cfgRegSave | integer | write |
| 3 | cfgRegTable | Sequence of cfgReg-TableEntry | |
| 3.1 | cfgRegTableEntry | Sequence | |
| 3.1.1 | cfgRegTabAddress | string | read |
| 3.1.2 | cfgRegTabName | string | read |
| 3.1.3 | cfgRegTabDevice | string | read |
| 3.1.4 | cfgRegTabRouter | string | read |
| 3.1.5 | cfgRegTabRadDst | integer | read |
| 3.1.6 | cfgRegTabBkbnDst | integer | read |
| 3.1.7 | cfgRegTabSrc | integer | read |
| 3.1.8 | cfgRegTabRegControl | integer | read |
| 4 | cfgRegNvTable | Sequence of cfgReg NvTableEntry | |
| 4.1 | cfgRegNvTableEntry | Sequence | |
| 4.1.1 | cfgRegNvTabAddress | string | write |
| 4.1.2 | cfgRegNvTabStatus | integer | write |
| 4.1.3 | cfgRegNvTabRegControl | integer | write |

| | | | |
|---|---|---|---|
| 4.1.4 | cfgRegNvTabRadDst | integer | write |
| 4.1.5 | cfgRegNvTabBkbnDst | integer | write |
| 4.1.6 | cfgRegNvTabSrc | integer | write |

## The Configure Ident Group

ACCESSPOINT.configuration.cfgIdent (1.3.6.1.4.1.551.2.2.1.9.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | cfgIdIpadr | ipaddress | write |
| 2 | cfgIdImask | ipaddress | write |
| 3 | cfgIdIpGateway | ipaddress | write |

## The Radio Error Statistics Group

ACCESSPOINT.statistics.statRadio (1.3.6.1.4.1.551.2.2.2.1.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | statRadLocalBufferFull | counter | read |
| 3 | statRadDuplicateRcv | counter | read |
| 5 | statRadBadCRC | counter | read |
| 12 | statRadRetries | counter | read |
| 13 | statRadMaxRetries | integer | read |
| 16 | statRadTxFull | counter | read |

## The Logging Group

ACCESSPOINT.logging (1.3.6.1.4.1.551.2.2.3.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | logTable | Sequence of log-TableEntry | |
| 1.1 | logTableEntry | Sequence | |
| 1.1.1 | logTabEntryIndex | integer | read |
| 1.1.2 | logTabEntryTicks | time | read |
| 1.1.3 | logTabEntryText | string | read |
| 1.1.4 | logTabEntryLevel | integer | read |

## The Admin Group

ACCESSPOINT.admin (1.3.6.1.4.1.551.2.2.4.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | adminRestart | integer | write |
| 4 | adminMajVersion | integer | read |
| 5 | adminMinVersion | integer | read |
| 6 | adminBootp | integer | write |
| 7 | adminDistribute | integer | write |
| 8 | adminDistributeCnt | integer | read |
| 9 | adminPing | integer | write |
| 10 | adminPingState | integer | read |
| 11 | adminFallback | integer | write |
| 12 | adminRcvDistribute | integer | write |
| 13 | adminBetaVersion | integer | read |

## The Admin LinkTest Group

ACCESSPOINT.admin.adminLinktest (1.3.6.1.4.1.551.2.2.4.2.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | adminLtMultiTest | integer | write |
| 2 | adminLtDest | string | write |
| 3 | adminLtSize | integer | write |
| 4 | adminLtCount | integer | write |
| 5 | adminLtDstRcv | counter | read |
| 6 | adminLtSrcRcv | counter | read |
| 7 | adminLtSrcXmt | counter | read |
| 8 | adminLtAveTrip | counter | read |
| 9 | adminLtMinTrip | counter | read |
| 10 | adminLtMaxtrip | counter | read |
| 11 | adminLtUniTest | integer | write |
| 12 | adminLtAuto | integer | write |

## The Admin FTP Group

ACCESSPOINT.admin.adminFTP (1.3.6.1.4.1.551.2.2.4.3.x)

| Object Id | Object Name | Object Type | Access |
|---|---|---|---|
| 1 | adminFtpGet | integer | write |
| 2 | adminFtpDest | ipaddress | write |
| 3 | adminFtpUser | string | write |
| 4 | adminFtpPassword | string | write |
| 5 | adminFtpFile | string | write |
| 6 | adminFtpPut | integer | write |

# Appendix D - Cisco Technical Support

If you are a network administrator and need personal technical assistance with a Cisco product that is under warranty or covered by a maintenance contract, contact Cisco's Technical Assistance Center (TAC) at 800 553-2447, 408 526-7209, or tac@cisco.com. To obtain general information about Cisco Systems, Cisco products, or upgrades, contact 800 553-6387, 408 526-7208, or cs-rep@cisco.com.

# Manufacturer's Federal Communication Commission Declaration of Conformity Statement

**Models: BR340, BR342, BRI340, BRI341**

**Manufacturer: Cisco Systems, Inc.**
**170 West Tasman Drive**
**San Jose, CA 95134 USA**

This device complies with Part 15 rules. Operation is subject to the following two conditions:
1) this device may not cause harmful interference, and 2) this device must accept any interference received, including interference that may cause undesired operation.

This equipment has been tested and found to comply with the limits of a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a residential environment. This equipment generates, uses, and radiates radio frequency energy, and if not installed and used in accordance with the instructions, may cause harmful interference. However, there is no guarantee that interference will not occur. If this equipment does cause interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to correct the interference by one of the following measures:

- Reorient or relocate the receiving antenna.
- Increase separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from which the receiver is connected.
- Consult the dealer or an experienced radio\TV technician.

**User Warning**

The Part 15 radio device operates on a non-interference basis with other devices operating at this frequency. Any changes or modification to said product not expressly approved by Aironet could void the user's authority to operate this device.

## *Professional Installation*

Per the recommendation of the FCC, the installation of high gain directional antenna to the system, which are intended to operated solely as a point-to-point system and whose total power exceeds +36dBm EIRP, require professional installation. It is the responsibility of the installer and the end user that the high power systems are operated strictly as a point-to-point system.

Systems operating as a point-to-multipoint system or use non directional antennas cannot exceed +36dBm EIRP power requirement under any circumstances and do not require professional installation.

# Department of Communications—Canada Canadian Compliance Statement

This Digital apparatus meets all the requirements of the Canadian Interference - Causing Equipment Regulations.

Cet appareil numerique respecte les exigences du Reglement sur le material broilleur du Canada.

This device complies with Class B Limits of Industry of Canada. Operation is subject to the following two conditions: 1) this device may cause harmful interference, and 2) this device must accept any interference received, including interference that may cause undesired operation.

The device is certified to the requirements of RSS-139-1 and RSS-210 for 2.4 GHz spread spectrum devices. The use of this device in a system operating either partially or completely outdoors may require the user to obtain a license for the system according to the Canadian regulations. For further information, contact your local Industry Canada office.

# European Telecommunication Standards Institute Statement of Compliance Information to User

This equipment has been tested and found to comply with the European Telecommunications Standard ETS 300.328. This standard covers Wideband Data Transmission Systems referred in CEPT recommendation T/R 10.01.

This type accepted equipment is designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy, and if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications.

## Declaration of Conformity

### Cisco Systems, Inc. Model Numbers:

**AIR-BR340, AIR-BR342,**
**AIR-BRI340, AIR-BRI341**

Radio CE Type Certificate Number:
Radio Type Approval Examination Number:

| | |
|---|---|
| Application of Council Directive: | 89/336/EEC |
| Application of Council Directive: | 72/23/EEC |

Standards which Conformity is Declared:

EN 55022 (B)
EN 55011 (B)
EN 50082-1
EN 60950

Manufacturer: Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706 USA

*The undersigned hereby declares the above specified equipment conforms to the above Directives and standards.*

Michael Smedley

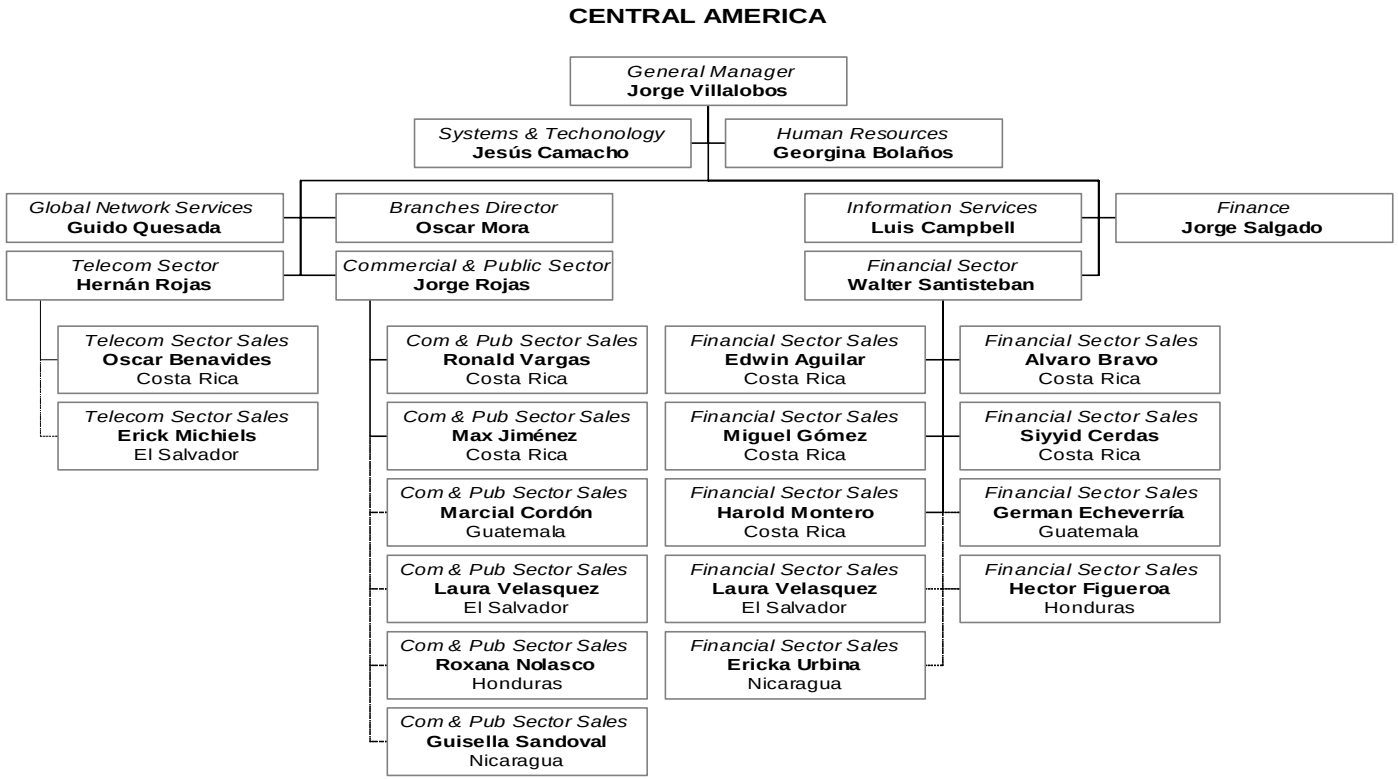
CISCO SYSTEMS

Michael Smedley
Manager, Manufacturing Engineering
Cisco Systems, Inc.

## Anexo 6: Organigramas de Unisys Latinoamérica y Caribe

Este anexo muestra los organigramas de Unisys Latinoamérica y Caribe. El primer organigrama muestra a los directores de departamentos; el segundo organigrama muestra los directores a nivel de Branch y el último muestra la organización del Departamento GNS (que es en el que el estudiante realizó el Proyecto de Graduación).

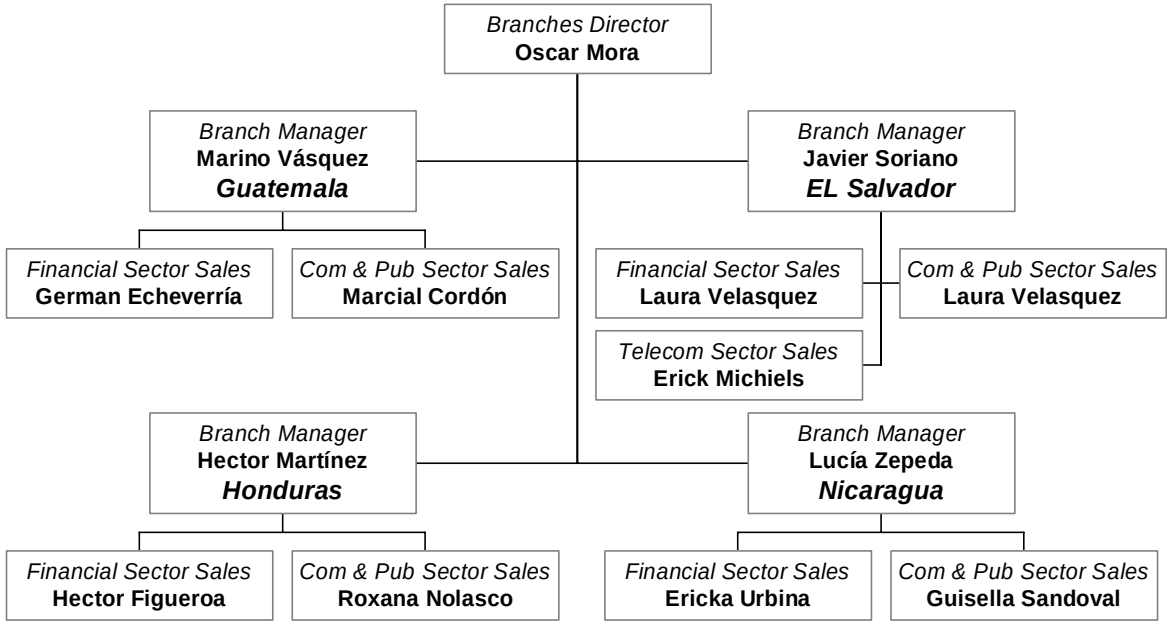
# LATIN AMERICA AND CARIBBEAN GROUP




**Organigrama de Unisys Latinoamérica y Grupo del Caribe**

# LATIN AMERICA AND CARIBBEAN GROUP

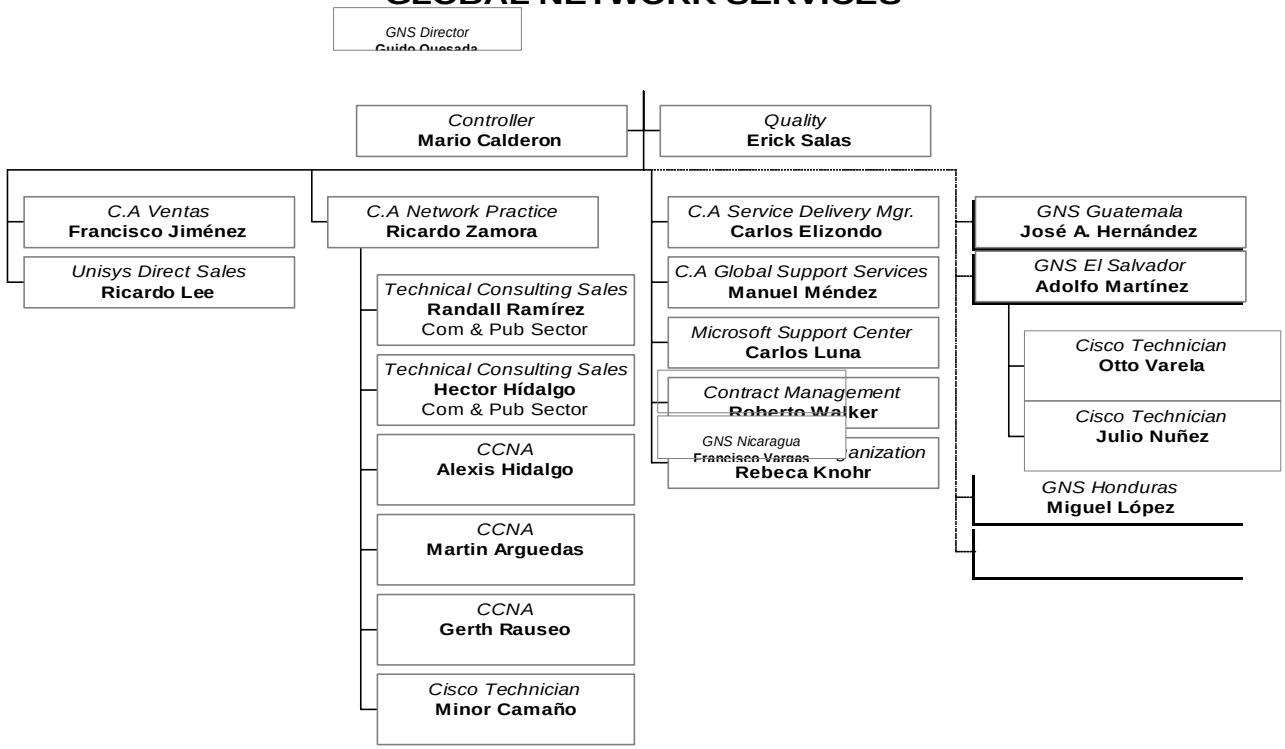
**BRANCHES**




**Organigrama de Unisys Latinoamérica y Grupo del Caribe (Branches)** **497**

# LATIN AMERICA AND CARIBBEAN GROUP

## GLOBAL NETWORK SERVICES




**Organigrama del Departamento GNS de Unisys**

**498**