FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

A LOS ASISTENTES A LOS CURSOS

Las autoridades de la Facultad de Ingeniería, por conducto del jefe de la División de Educación Continua, otorgan una constancia de asistencia a quienes cumplan con los requisitos establecidos para cada curso.

El control de asistencia se llevará a cabo a través de la persona que le entregó las notas. Las inasistencias serán computadas por las autoridades de la División, con el fin de entregarle constancia solamente a los alumnos que tengan un mínimo de 80% de asistencias.

Pedimos a los asistentes recoger su constancia el día de la clausura. Estas se retendrán por el periodo de un año, pasado este tiempo la DECFI no se hará responsable de este documento.

Se recomienda a los asistentes participar activamente con sus ideas y experiencias, pues los cursos que ofrece la División están planeados para que los profesores expongan una tesis, pero sobre todo, para que coordinen las opiniones de todos los interesados, constituyendo verdaderos seminarios.

Es muy importante que todos los asistentes llenen y entreguen su hoja de inscripción al inicio del curso, información que servirá para integrar un directorio de asistentes, que se entregará oportunamente.

Con el objeto de mejorar los servicios que la División de Educación Continua ofrece, al final del curso deberán entregar la evaluación a través de un cuestionario diseñado para emitir juicios anónimos.

Se recomienda llenar dicha evaluación conforme los profesores imparten sus clases, a efecto de no llenar en la última sesión las evaluaciones y con esto sean más fehacientes sus apreciaciones.

Atentamente
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FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

<u>CONMUTACIÓN EN REDES DE TELECOMUNICACIONES</u>

EXPOSITOR: M. en C. ERNESTO E. QUIROZ MORONES
PALACIO DE MINERÍA
JUNIO DE 1999

CONMUTACION EN REDES DE TELECOMUNICACIONES

M. C. Ernesto E. Quiroz M.

Profesor Investigador del CITEDI-IPN

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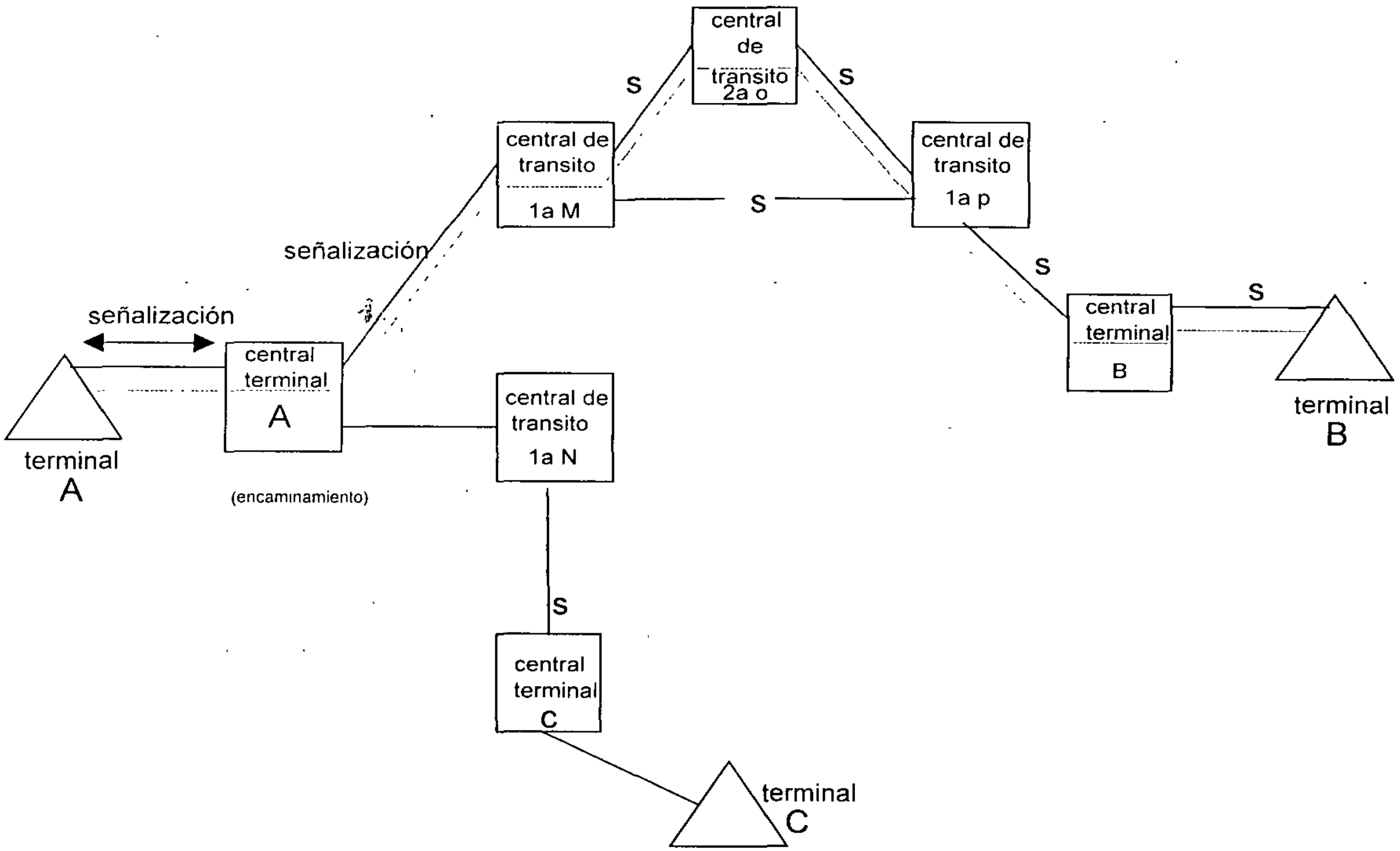
Centro de Investigación y Desarrollo de Tecnología Digital

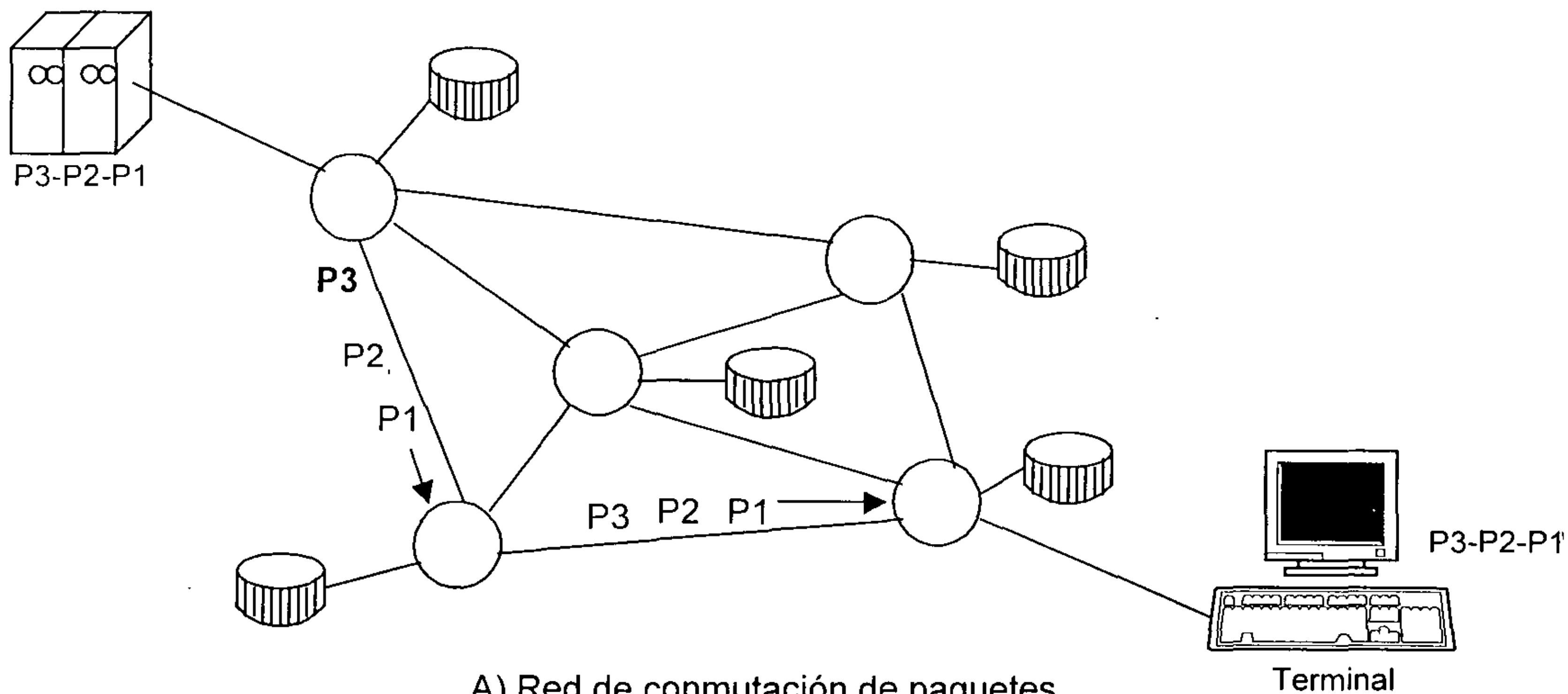
ESQUEMAS DE CONMUTACION

- Conmutacion de circuitos
- Conmutación de paquetes
  - Modo Datagrama
  - Circuito Virtual Permanente
  - Circuito Virtual Conmutado

CONMUTACIÓN DE CIRCUITOS

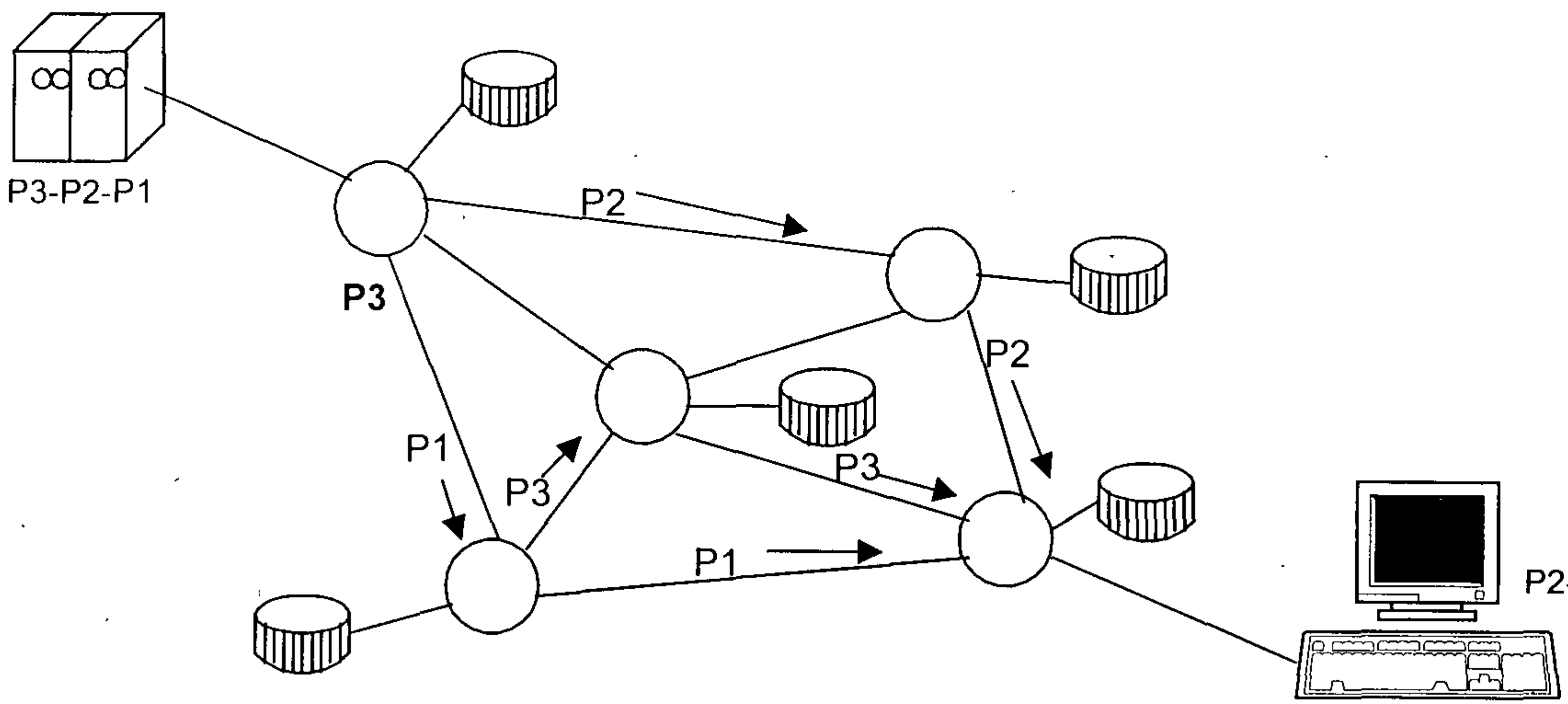


CONMUTACION DE PAQUETES



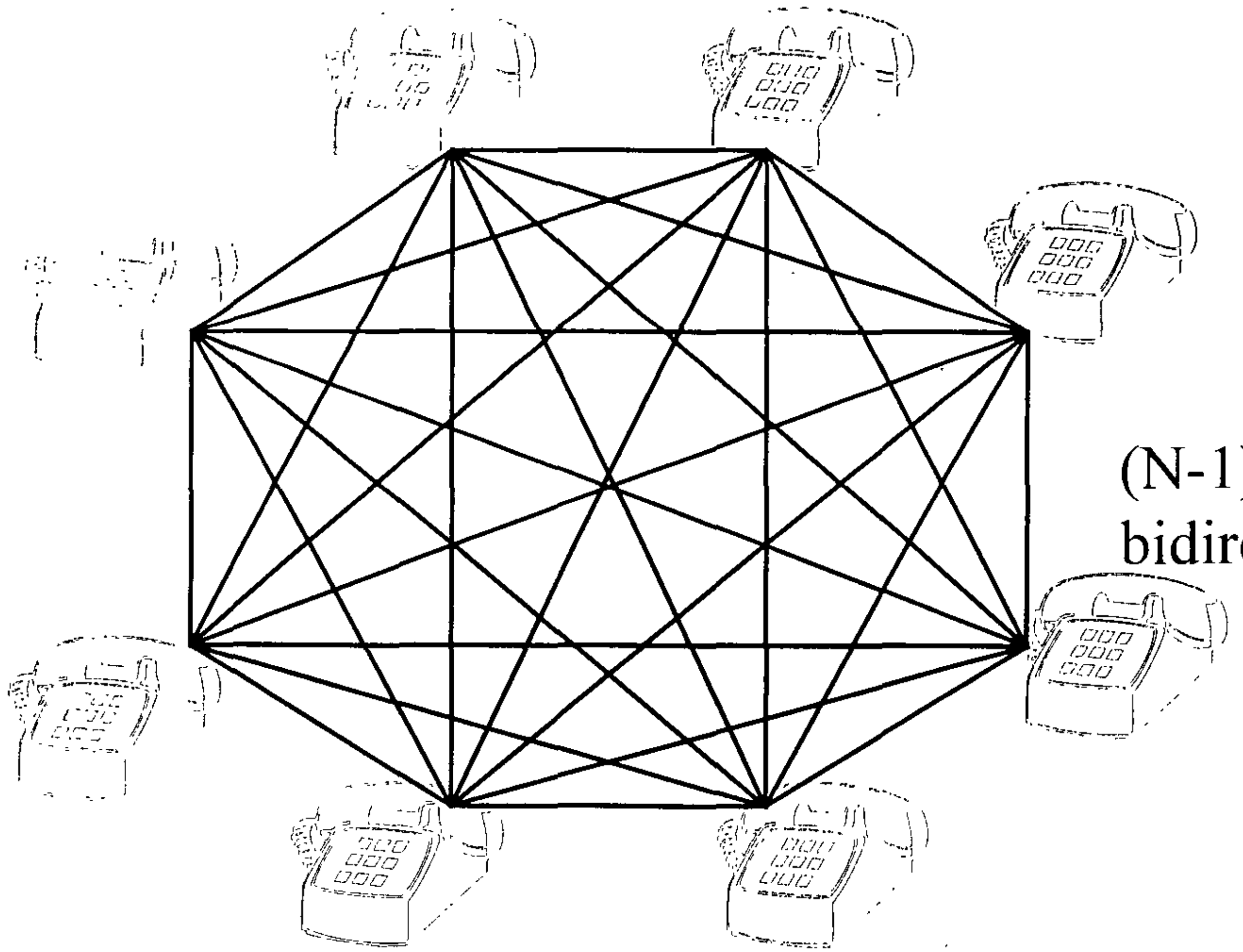
A) Red de conmutación de paquetes.
Modo circuito virtual.

CONMUTACION DE PAQUETES



B) Red de conmutación de paquetes.
Modo datagrama.

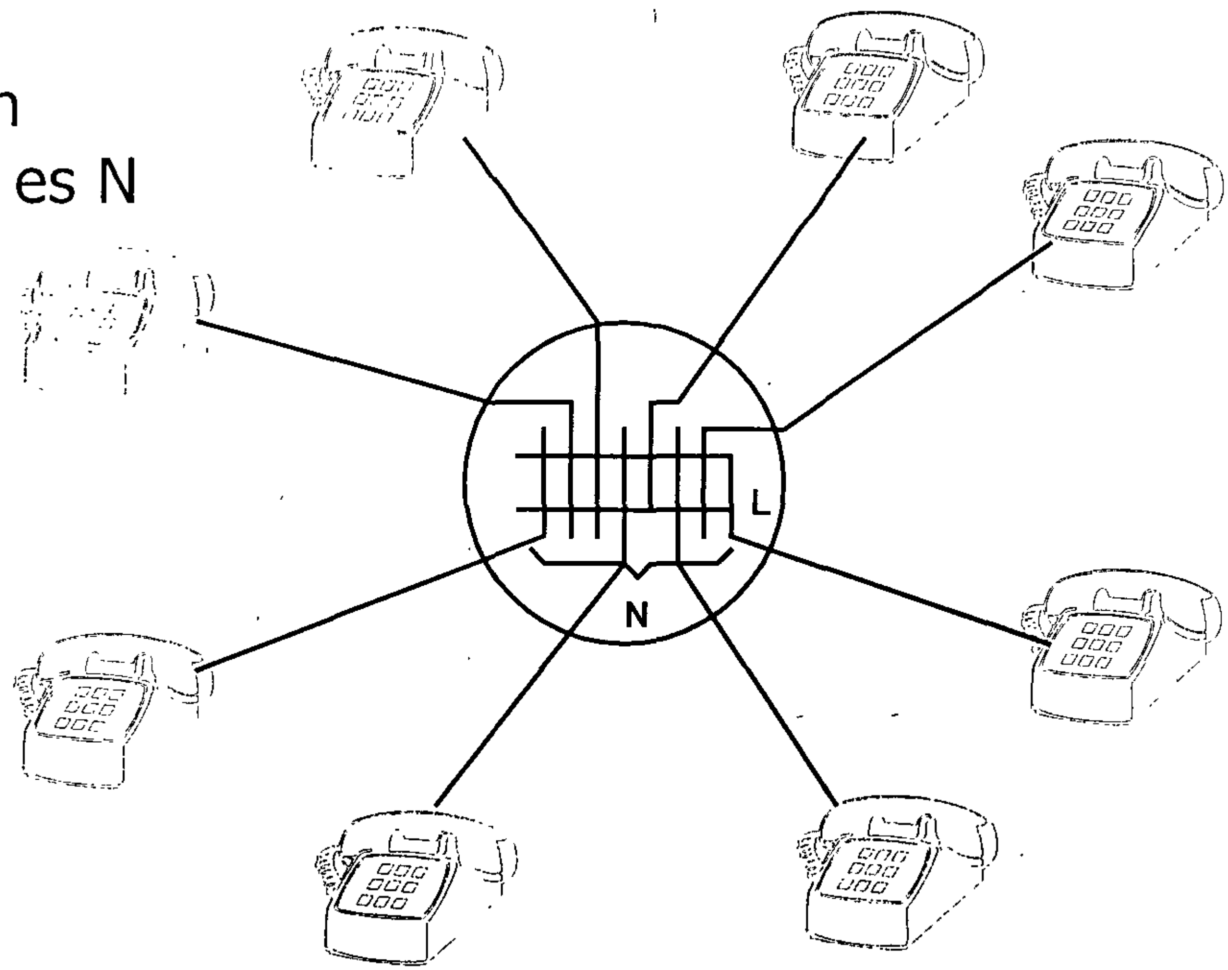
LA NECESIDAD DE LA CONMUTACIÓN



(N-1)N/2 líneas bidireccionales

CONMUTADOR

- El número de líneas de transmisión requeridas es N



FUNCIONES BASICAS DE LOS SISTEMAS DE CONMUTACION DIGITAL

- Interconexión
- Control
- Señalización con las terminales de abonado
- Señalización con otras centrales
- Explotación
- Sincronización
- Temporización

SISTEMA JERARQUICO

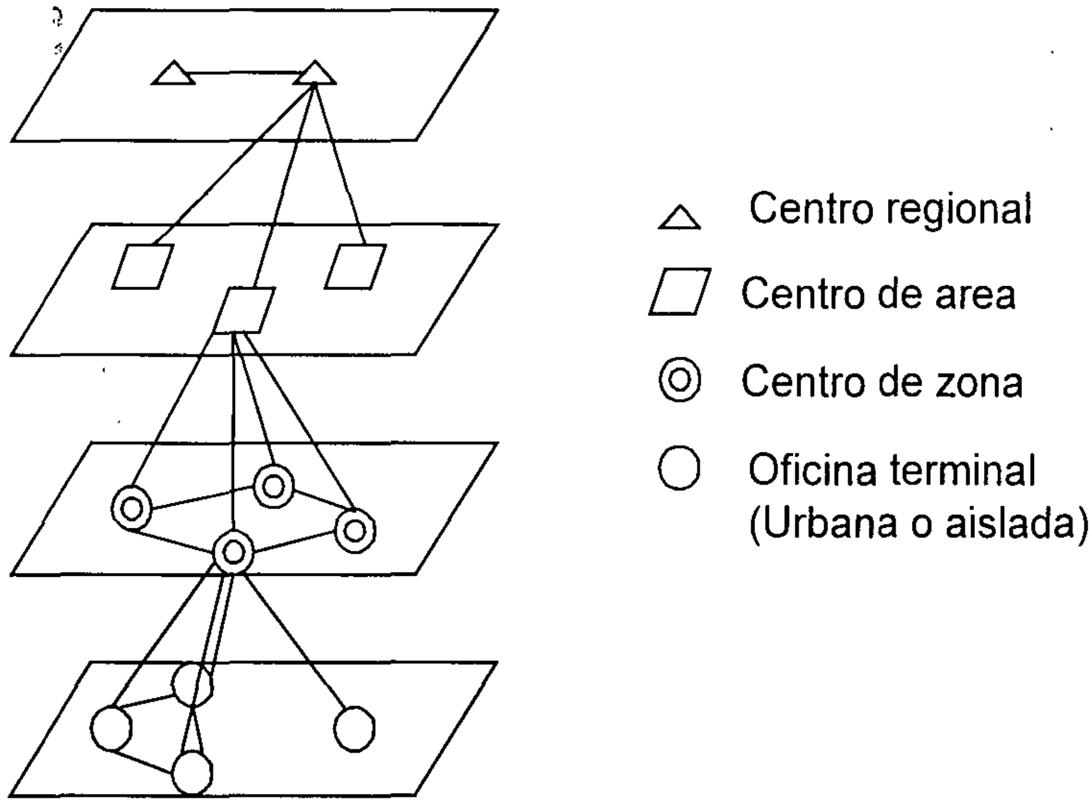
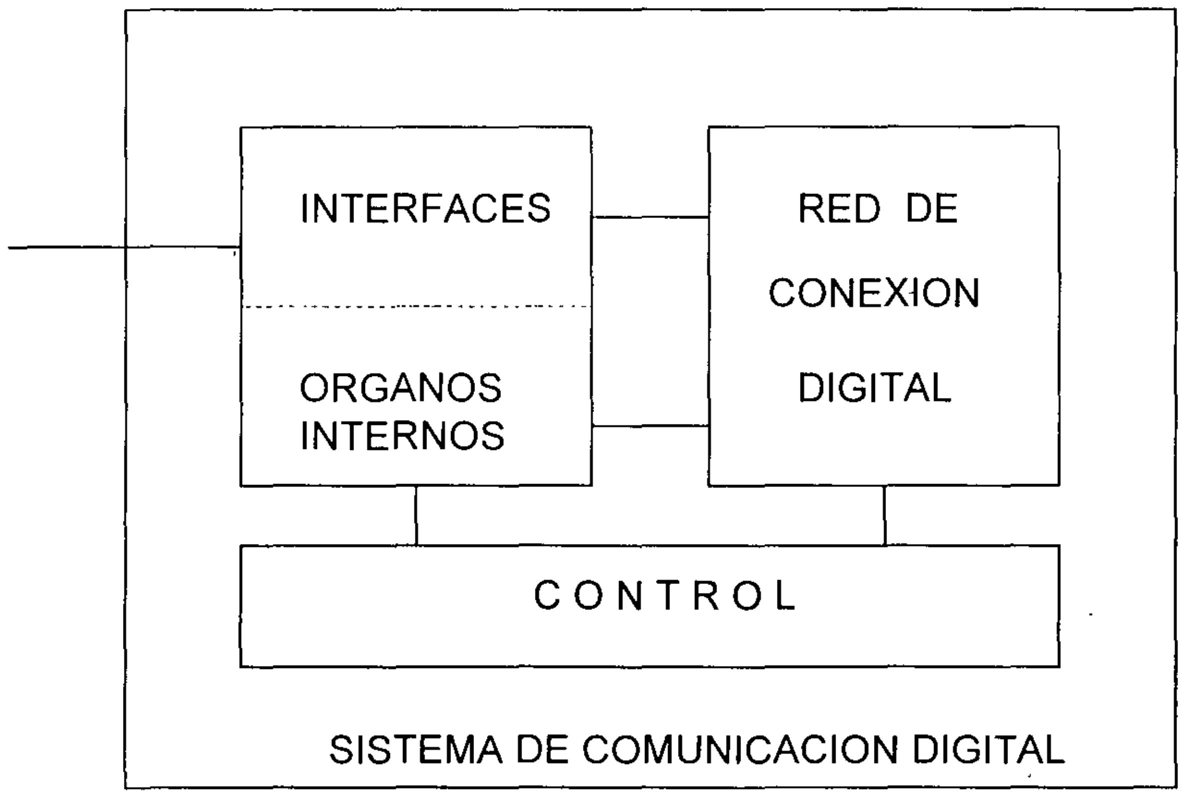


Fig.1.7 Estructura de la red teléfonica

CONMUTACION DIGITAL

Una de las funciones básicas de un sistema de conmutación es la de función de ***interconexión,*** un sistema de conmutación debe ser capaz de suministrar vías de comunicación entre todos los abonado de una central dada, y en coordinación con otras centrales, comunicar abonados más distantes. Esto se lleva a cabo a través de la red de conexión.

COMPONENTES DE UN SISTEMA DE CONMUTACIÓN DIGITAL



INTERFACES

- Conectan al sistema de conmutación con el mundo exterior
- Líneas de abonado
- Troncales
- conexión con operadoras.

ORGANOS INTERNOS

- Soportan funciones para el establecimiento de conexiones y para la explotación del sistema de conmutación digital
- Emisores y receptores de señales para las interfaces
- Organos de conferencia
- Organos de prueba de las interfaces.

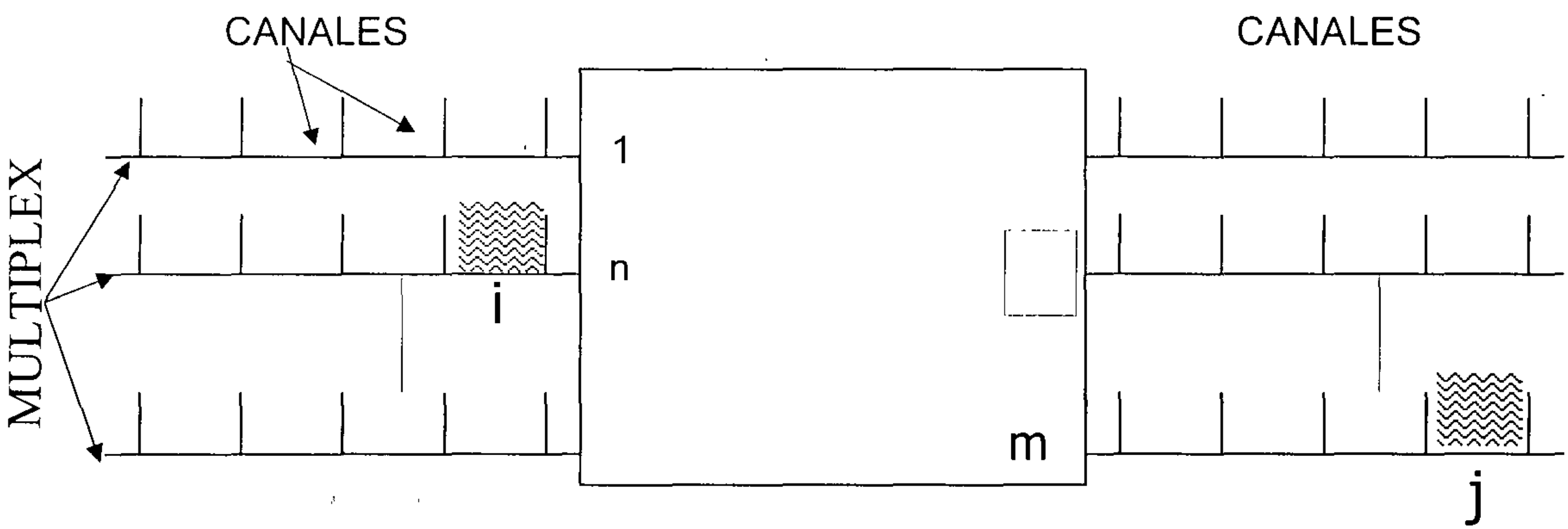
CONTROL

- Es el cerebro del sistema
- Participa en prácticamente todas las funciones del sistema de conmutación
- Constituido por un conjunto de procesadores
- Métodos de redundancia, de reparto de tráfico y de reparto de funciones.

RED DE CONEXION DIGITAL

- Establece las conexiones entre los demás bloques del sistema
- Establece una trayectoria de comunicación para un par de usuarios a través de la central.
- Utiliza dos tipos de etapas de conmutación:
  - Etapa espacial (Space switch)
  - Eapa temporal (Time switch)

CONMUTADOR DIGITAL



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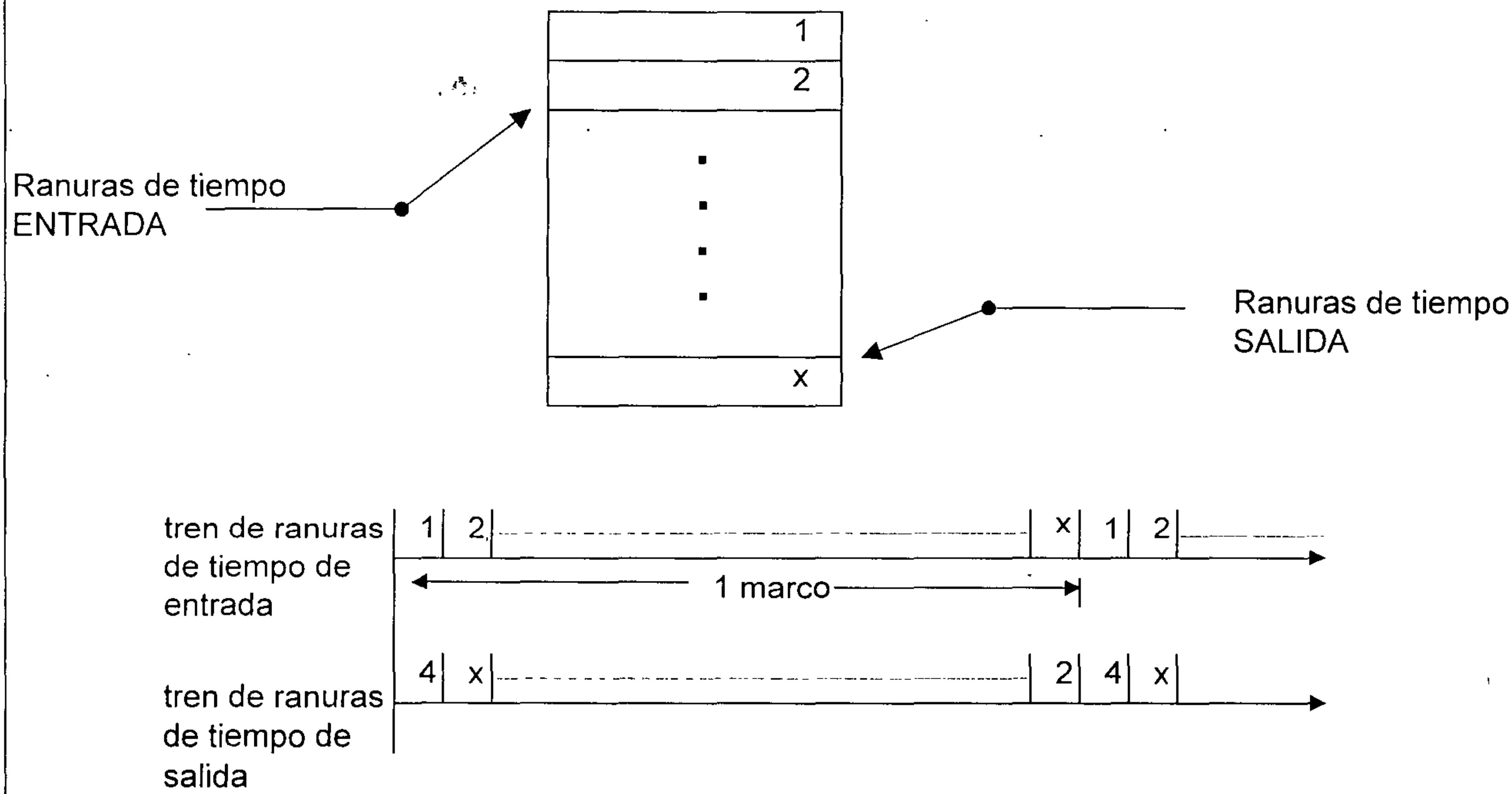
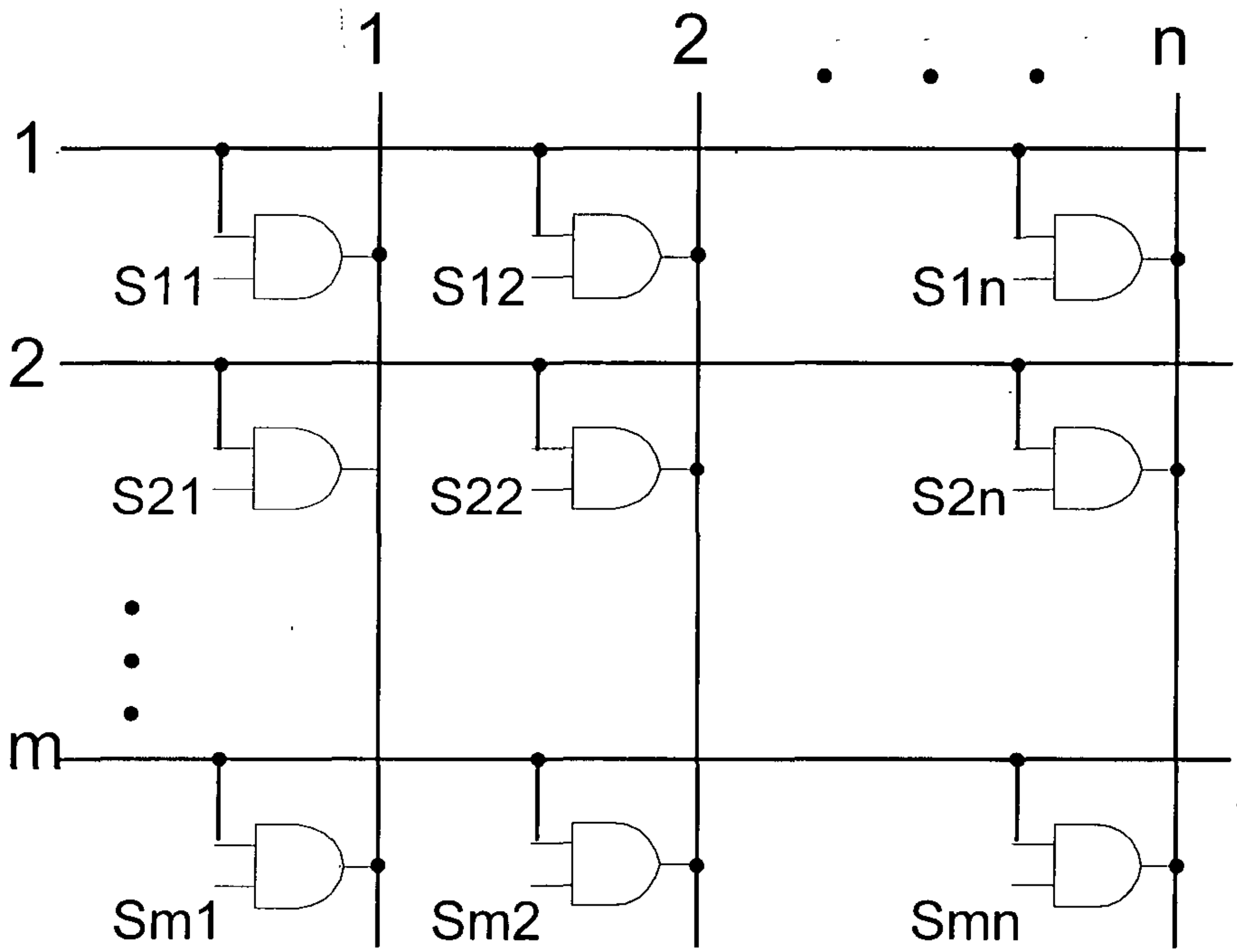
Arquitecturas de red

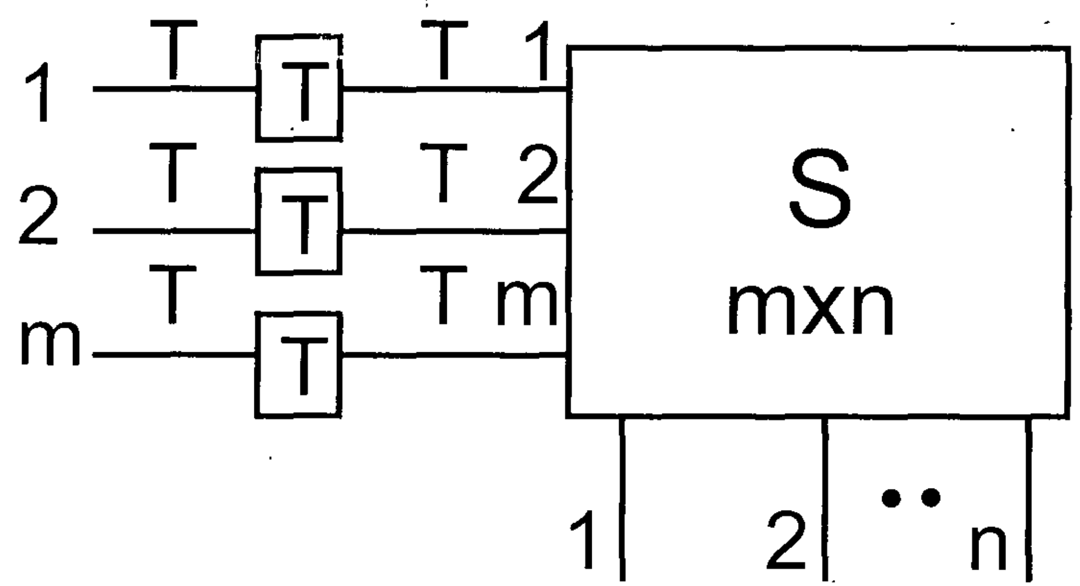
- Pueden contener intercambiadores de ranuras de tiempo, conmutación por división de tiempo, o una combinación de ambos
  - a) Solo T
  - b) Solo S
  - c) T-S
  - d) S-T
  - g) Combinaciones as complejas de S y T



Etapa T: Intercambiador de ranuras de tiempo



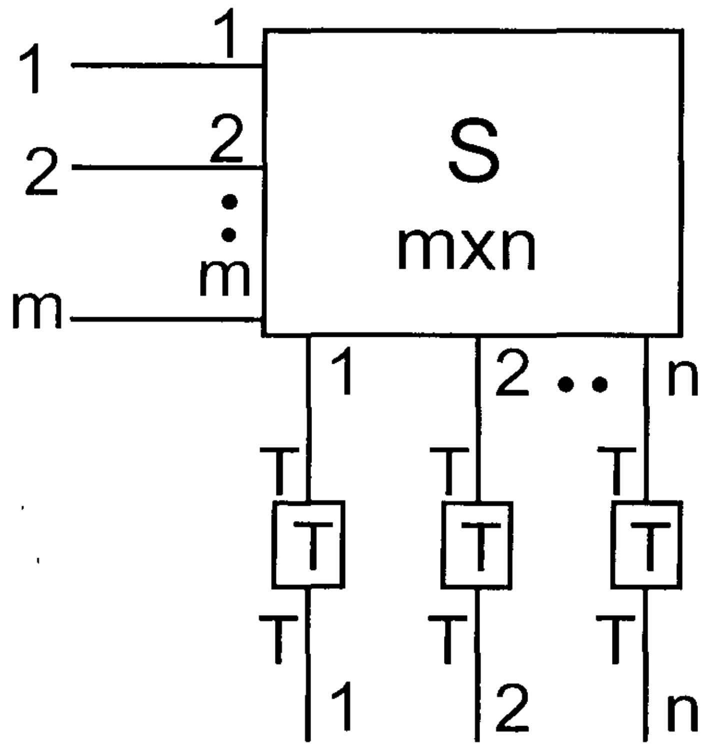
Etapa S: Intercambiador de Múltiplex



mT Ranuras de tiempo entrada
nT Ranuras de tiempo salida

Fig 3.20   Arquitectura de una red tiempo-espacio

Tiempo-espacial



mT Ranuras de tiempo entrada
nT Ranuras de tiempo salida

Fig 3.21 Arquitectura de una red espacio-tiempo

TEMAS RELACIONADOS

- Bloqueo en el Conmutador
- Señalización
  - Señalización de abonado
  - Señalización entre centrales
  - Sistema de Señalización No. 7 (SS7)
- Ingeniería de tráfico

SISTEMAS BASADOS EN CONMUTACION DE CIRCUITOS

- POTS: Plain Old Telephone Network
- PSTN: Public Switched Telephone Network
- Telefonía Celular (Tema 11)
- PCS: Personal Communication Systems (Tema 12)
- ISDN: Integrated Services Digital Network (Tema 6)

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CONMUTACION DE PAQUETES

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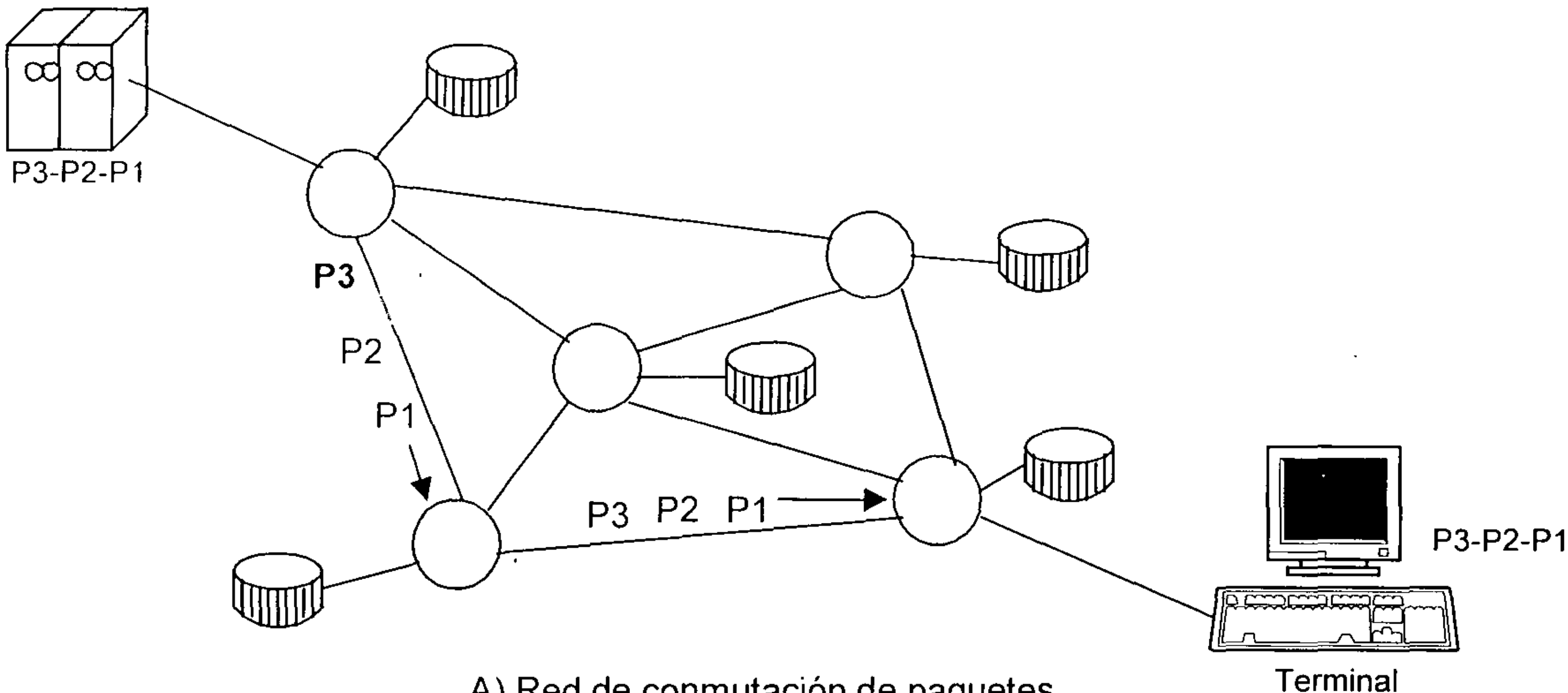
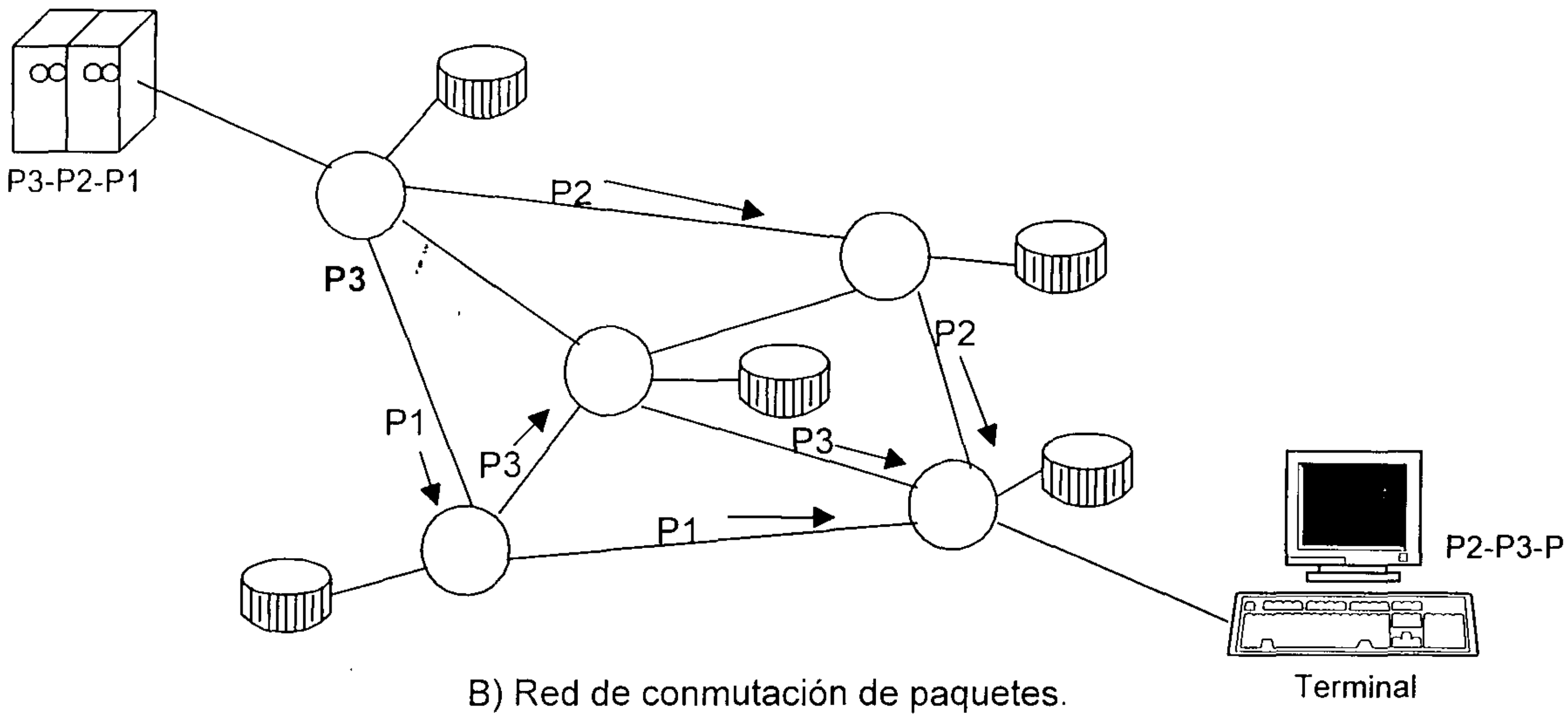
CIRCUITO VIRTUAL



A) Red de conmutación de paquetes.
Modo circuito virtual.

MODO DATAGRAMA



B) Red de conmutación de paquetes.
Modo datagrama.

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	Conmutación de Circuitos	Conmutación de paquetes
Tiempo de estableci-miento de la llamada	Aceptable para servicio telefónico. Relativamente largo para datos.	No existe fase de establecimiento. (Solo en circuito virtual conmutado)
Retardo de transmisión	Prácticamente despreciable.	Existe en toda comunicación (del orden de cientos de msg o de sg). En caso de alto tráfico puede llegar a ser importante.
Asignación de circuitos	Unico y exclusivo para cada comunicación.	Circuito compartido por diversas comunicaciones simultáneas.
Transmisión de información de identificación de destino	Sólo durante la fase de establecimiento.	Es necesario incluir alguna "identificación de destino" en cada paquete de la comunicación.
Necesidad de capacidad de almacenamiento en la red	No.	Sí, localizada en los nodos de conmutación de la red.
Situación con gran cantidad de trafico ofrecido	Incremento de la probabilidad de bloqueo.	No existe bloqueo. Se incrementan notablemente los retardos.
Flexibilidad de la red	Posibilidad de encaminamientos alternativos.	Gran flexibilidad en utilización de la red.

MODELO DE REFERENCIA OSI
(Open Systems Interconnextion)

- Lograr interoperabilidad entre sistemas de datos de distintos fabricantes.
- Es un conjunto completo de estándares (interfaces, servicios, formatos).
- Separa las distintas tareas necesarias para comunicar dos sistemas independientes.
- Define 7 niveles de funciones específicas.

Capas del Modelo OSI

Nivel	Nombre	Función
7	Aplicación	Datos Normalizados
6	Presentación	Interpretación de los datos
5	Sesión	Diálogos de control
4	Transporte	Integridad de los mensajes
3	Red	Enrutamiento de los mensajes
2	Enlace	Detección de errores
1	Físico	Conexión de equipos

CAPA FISICA

- Nivel de comunicación física a través de un canal.
- Se encarga de la transmisión de los bits (0's ó 1's).
- Define las reglas para garantizar la recepción correcta de las señales correspondientes a los símbolos binarios.
- Provee los medios mecánicos, eléctricos, funcionales y de procedimiento para establecer, mantener y liberar conexiones físicas.

CAPA DE ENLACE

- Asegura la integridad de los paquetes que se transportan.
- Basa su operación en el manejo de tramas
- Recibe bits del nivel físico
  - Ordena en tramas
  - Verifica que no contenga errores
  - Revisa la secuencia de las tramas
- Segmenta en tramas los mensajes del nivel superior.

CAPA DE RED

- Responsable de la comunicación entre dos nodos adyacentes de la red.
- Provee direcciones destino a los mensajes.
- Convierte direcciones y nombres lógicos $\Leftrightarrow$ físicos.
- Evalúa la mejor ruta que debe seguir el paquete.

CONMUTACION DE PAQUETES POR X.25

- Estandard de la ITU-T para accesar una red de conmutación de paquetes (RCP).
- Especifica las características de conexión entre un DTE (usuario) y un DCE (red).
- El enlace DTE-DCE se efectúa a través de un circuito dedicado, en forma síncrona, en modo paquete.
- Funciona dentro de las 3 primeras capas del modelo OSI.

Nivel 1

- Especifica un circuito dedicado Full-Dúplex (par de alambres, bidireccional).
- Punto a punto.
- Síncrono.
- Tasa digital = 19.2 Kbps

Nivel 2

- Basado en el protocolo LAPB (Link Access Procedures Balanced).
- Permite iniciar la conexión en cualquier extremo (DTE ó DCE).
- Verifica que las tramas lleguen libres de errores y con una secuencia correcta.
- Utiliza tres formatos de trama.
  - Trama de información
  - Trama de supervisión
  - Trama sin número

Campos de una trama LAPB

BANDERA	DIRECCION	CONTROL
1	1	1
DATOS Variable		
FCS	BANDERA	
1	1	

Nivel 3

- Segmenta la información de usuario en paquetes.
- Cada paquete de datos recibe un número consecutivo.
- Integra a cada paquete la dirección del DTE destino.
- Da acceso a los servicios de la RCP
  - Circuito Virtual Conmutado
  - Circuito Virtual Pemanente
  - Datagrama

DISPOSITIVOS ADAPTADORES

- El PAD (Packet Assembler/Disassembler) es un adaptador de un dispositivo (no X.25) a la RCP.
- También puede actuar como concentrador de varios usuarios a la RCP.
- Los usuarios pueden accesar un PAD por una conexión directa, línea privada o conmutada.
- Provee conversión de velocidad, código y protocolo para acomodar los datos de varios usuarios.

OTRAS ARQUITECTURAS DE CONMUTACION DE PAQUETES

- Frame Relay
- ATM (Asynchronous Transfer Mode)

FRAME RELAY

- Interconexión principalmente de LANs
- Para aplicaciones de datos, fax y voz
- Basado en transmisión de tramas de longitud variable
- Tasas de transmisión hasta 2.048 Mbps
- Poco sobreprocesamiento
- Manejo de ancho de banda en demanda

ATM

- Interconexión de LANs y WANs
- Soporte para una amplia variedad de servicios (datos, voz, video, audio, etc).
- Basado en la transmisión de celdas de longitud fija
- Tasas de transmisión hasta 622.08 Mbps
- Altas velocidades de conmutación
- Manejo de ancho de banda en demanda

CONCLUSIONES

- Las primeras RCP se basaron en el estandar X.25.
- Servicio de datos únicamente.
- Transmisión de tramas de longitud variable.
- Tasa digital de hasta 19.2 Kbps.

Características FR y ATM

FRAME RELAY

- Interconexión de LANs
- Servicios de Datos, Voz y Fax
- Transmisión de "tramas" de longitud variable
- Tasas de transmisión hasta 2.048 Mbps.

ATM

- Interconexión de LANs y WANs
- Servicios de Datos, Voz, Video, Audio
- Transmisión de "celdas" de longitud fija
- Tasas de transmisión hasta 622.08 Mbps.

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PCM: PULSE CODE MODULATION

OBJETIVOS:

- Técnica para digitalización de voz
- Unidad fundamental de transmisión digital: 64 Kbps
- Jerarquía PDH (Plesiochronous Digital Hierarchy)

VENTAJAS DE LA DIGITIZACIÓN DE VOZ

- Se requiere detectar solo dos niveles (bits)
- Regeneración de la señal
- Las señales codificadas son fácilmente cifrables
- Multiplexión
- Conmutación y transmisión con otras fuentes digitales.

Desventajas

- Ocupa mayor ancho de banda que el original analógico.
- Necesidad de sincronización.

Técnicas de Digitalización de voz

- Cuantizadores instantáneos: PCM.
- PCM Adaptivo Diferencial: ADPCM
- Métodos de Codificación Lineal Predictiva: MPE-LPC, CELP, etc.
- Moduladores Delta.
- Codificadores de Sub-banda

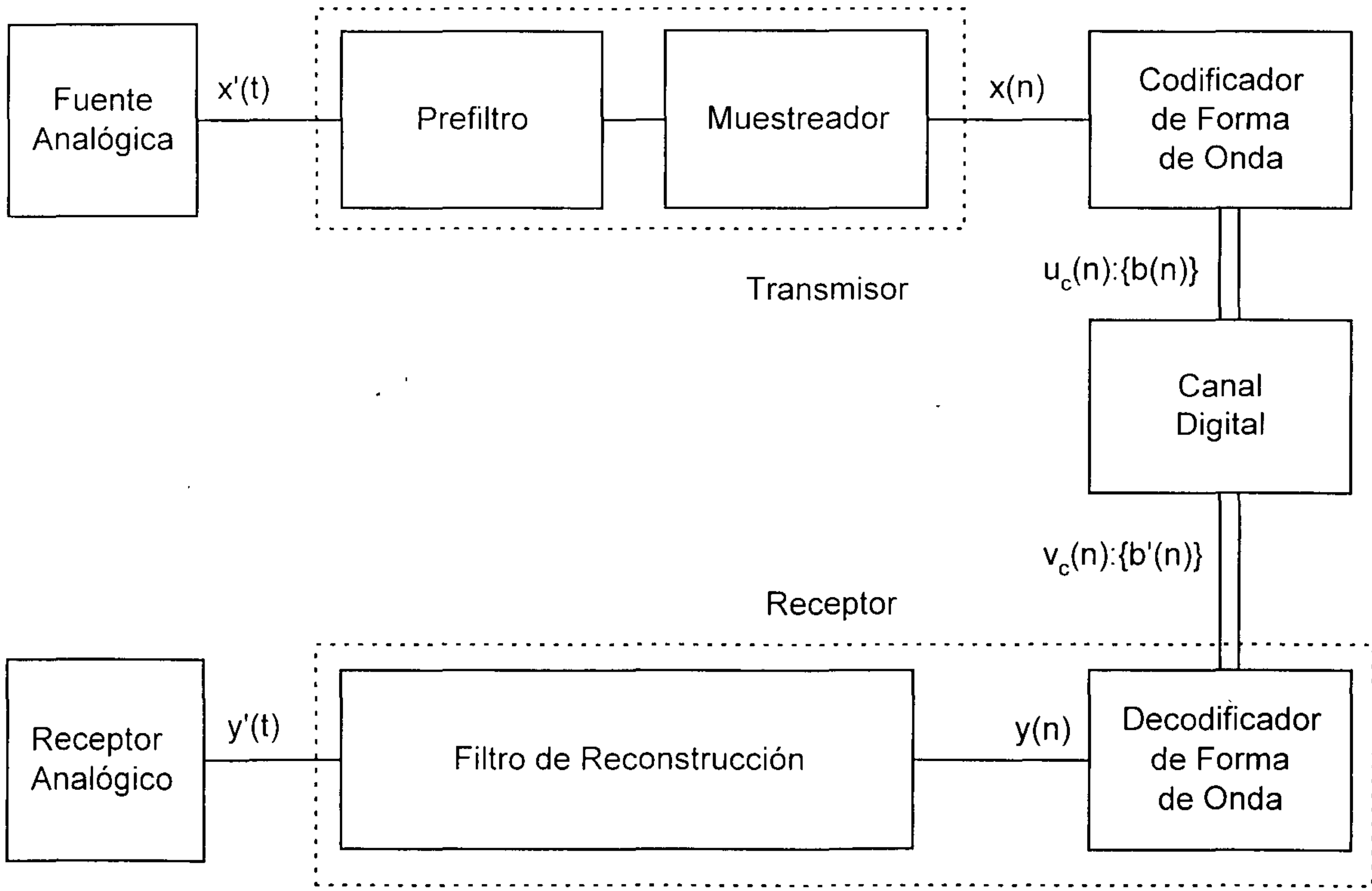
Modulación por codigos de pulsos: PCM

- Primer sistema de codificación de voz desarrollado.
- Sistema sencillo.
- Codificadores PCM instantáneos.
- Estándar para métodos de digitalización.
- Todos los codificadores implican codificación o decodificación PCM.

SISTEMA PCM

- La transformación de una forma de onda continua en una señal digital discreta involucra las operaciones de muestreo, cuantización y codificación.
- La conversión en transmisión se realiza mediante un convertidor analógico/digital (ADC); y el proceso contrario en la recepción medio de un convertidor digital/ analógico (DAC).

Diagrama a bloques de un sistema PCM



MUESTREO

Teorema de muestreo (Nyquist)

- Si una señal x(t) pasabajas limitada en banda ($f_{max} = f_x$) es muestreada a $f_m \geq 2f_x$, es posible reconstruirla completamente a partir de sus muestras instantáneas.

- A la tasa mínima $2\ f_x$ se le denomina frecuencia de Nyquist

Análisis espectral de la señal muestreada

Podemos interpretar el muestreo como la multiplicación de un tren de pulsos periódicos *s(t)* (función muestreo), con la señal del mensaje *f(t)*.

$$\begin{tikzcd} f(t) \arrow[r] & \bigotimes \arrow[r] & f_s(t) = f(t)s(t) \\ & s(t) \arrow[u] & \end{tikzcd}$$

Espectro del producto: $f(t)\ s(t) = f_s(t)$

$$f_s \longleftrightarrow F_s(\omega)$$

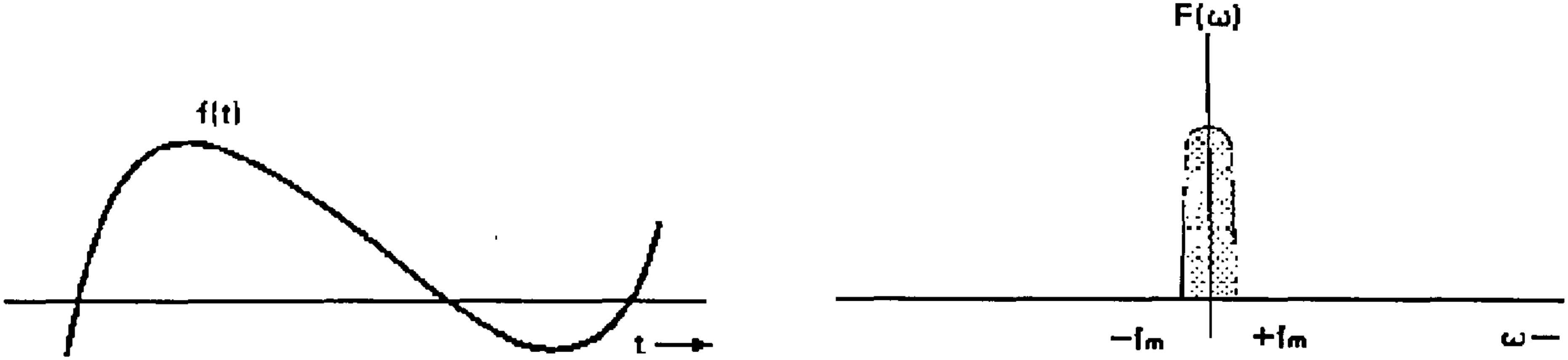
$$f(t)S(t) \longleftrightarrow \frac{1}{2\pi} F(\omega) * S(\omega)$$

$$F_s(\omega) = \frac{1}{2\pi} 2\pi A d F(\omega) * \sum_{n=-\infty}^{\infty} \frac{sen\,\pi d}{n\pi d} \delta(\omega - n\omega_o)$$

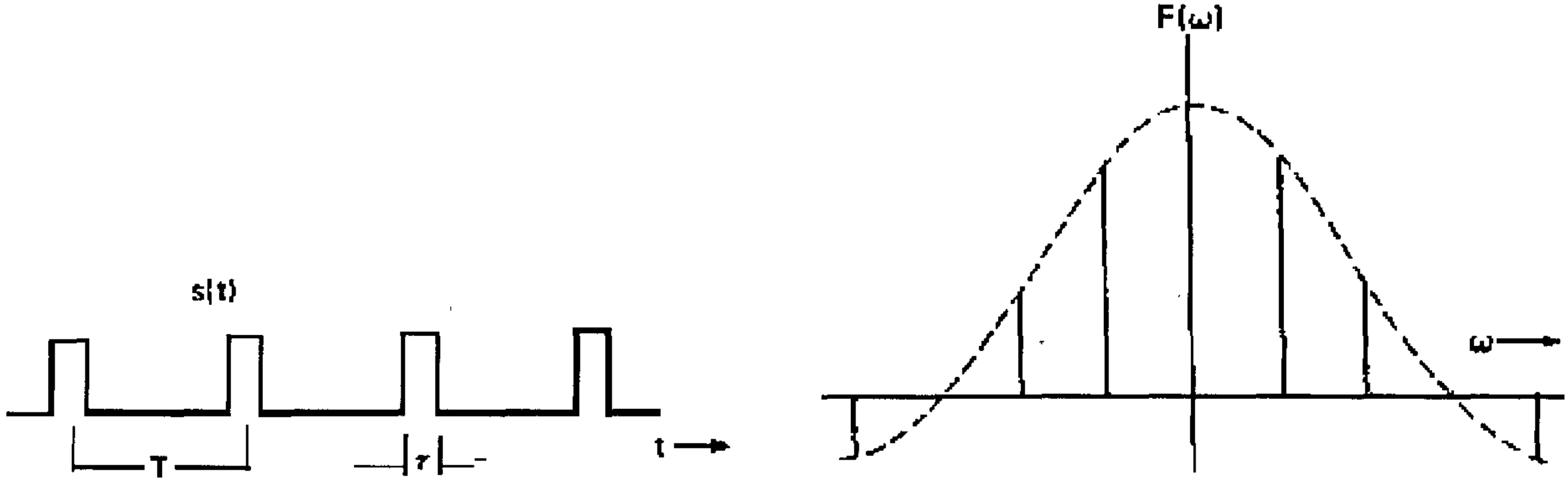
$$= A d F(\omega) * \sum_{n=-\infty}^{\infty} \frac{sen\,\pi d}{n\pi d} \delta(\omega - n\omega_o)$$

$$= A d \sum_{n=-\infty}^{\infty} \frac{sen\,\pi d}{n\pi d} F(\omega) * \delta(\omega - n\omega_o)$$

$$= A d \sum_{n=-\infty}^{\infty} \frac{sen\,\pi d}{n\pi d} F(\omega - n\omega_o)$$



(a)



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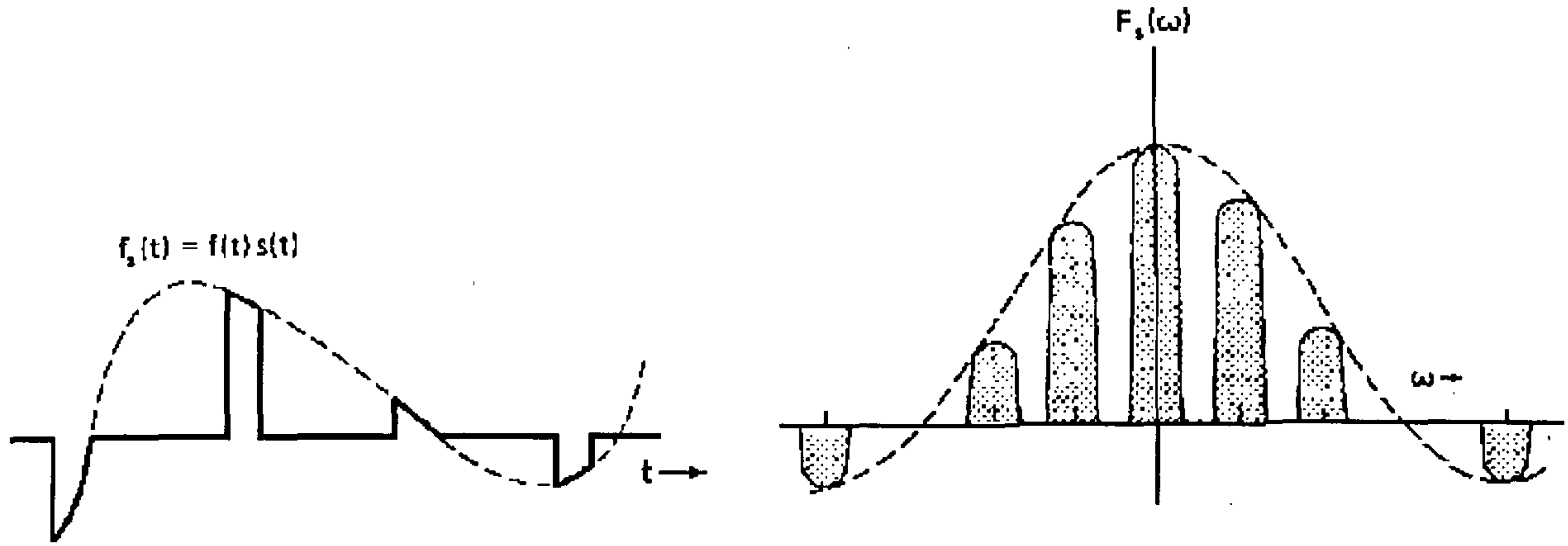
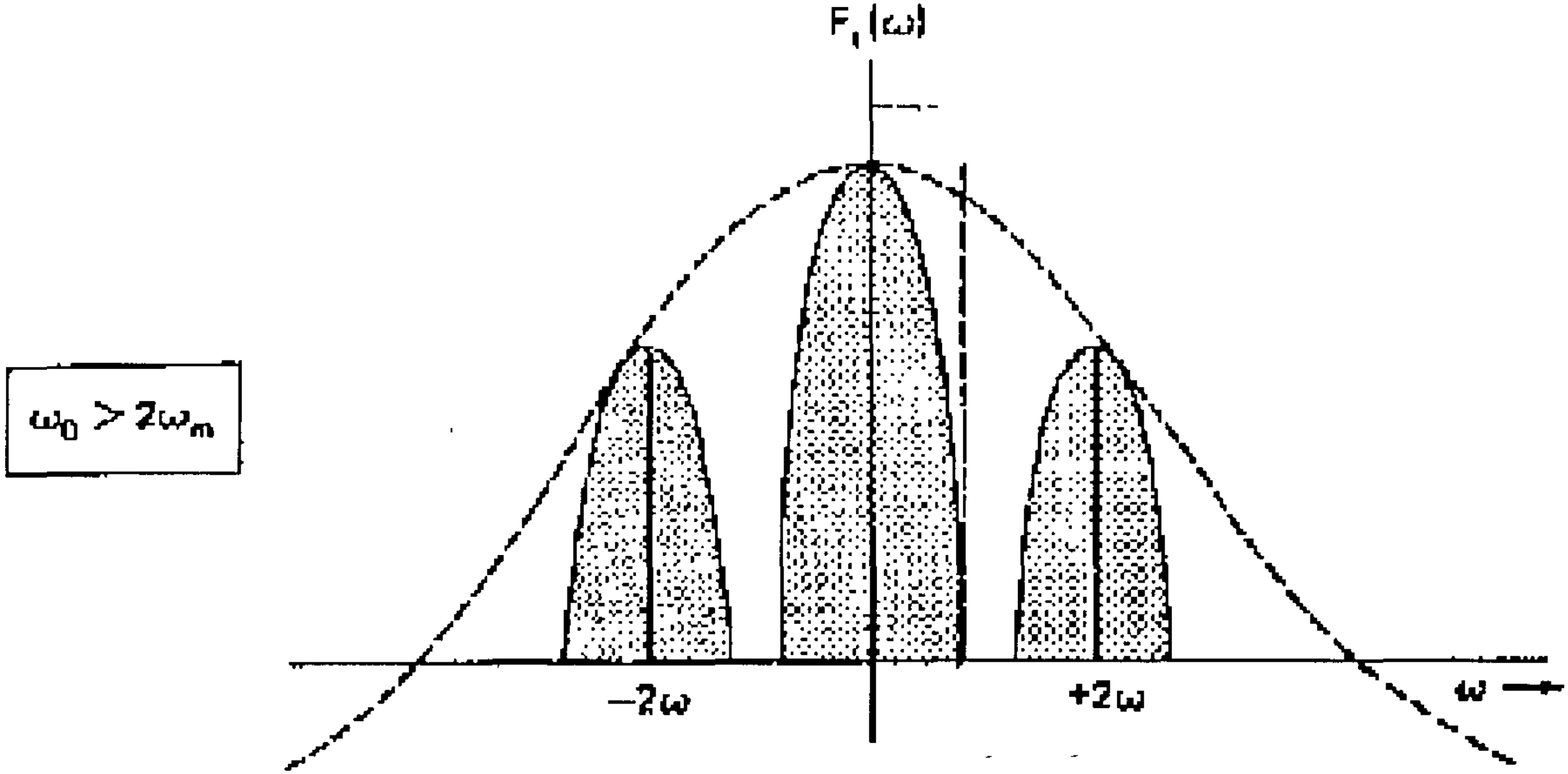
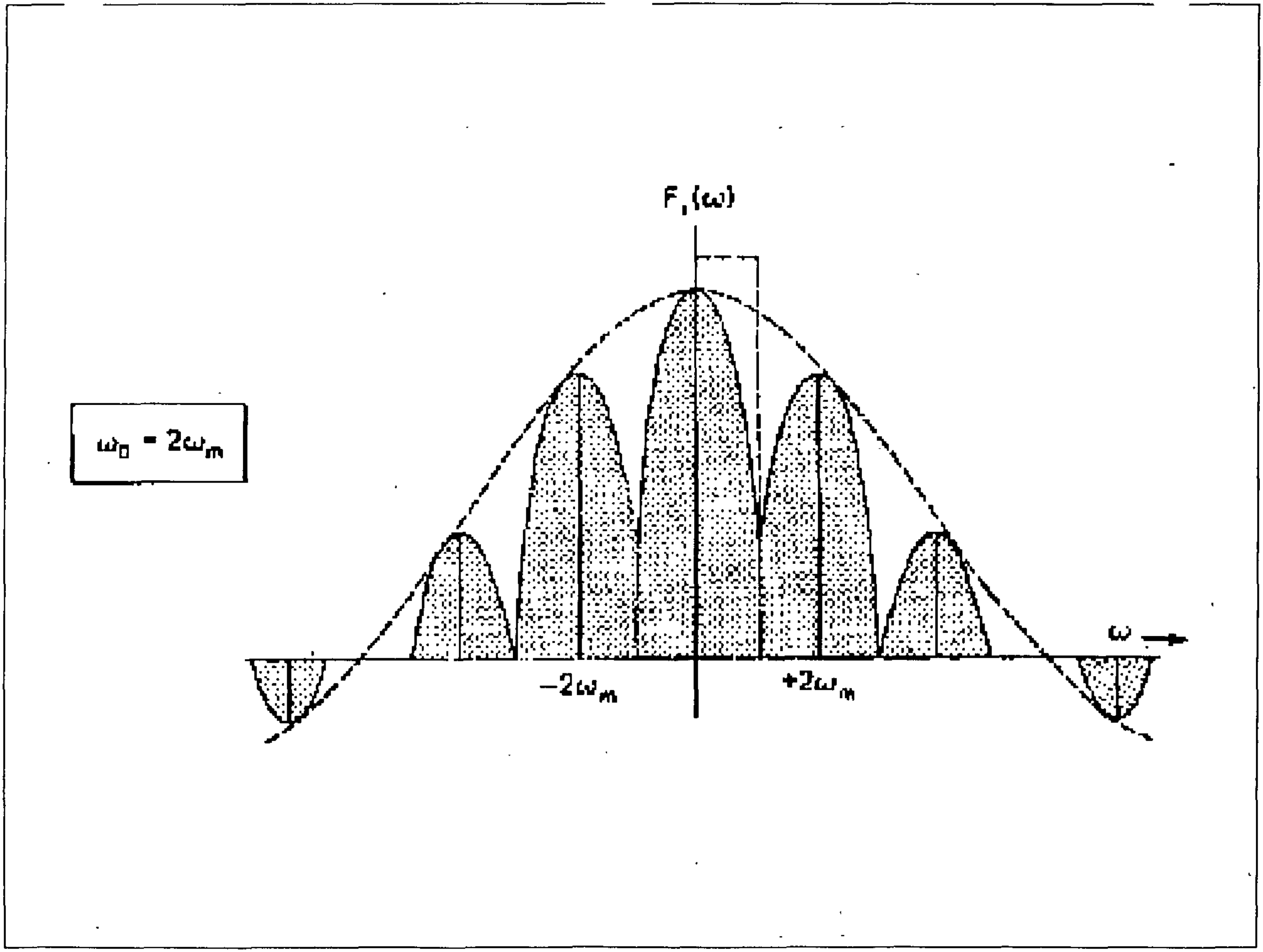


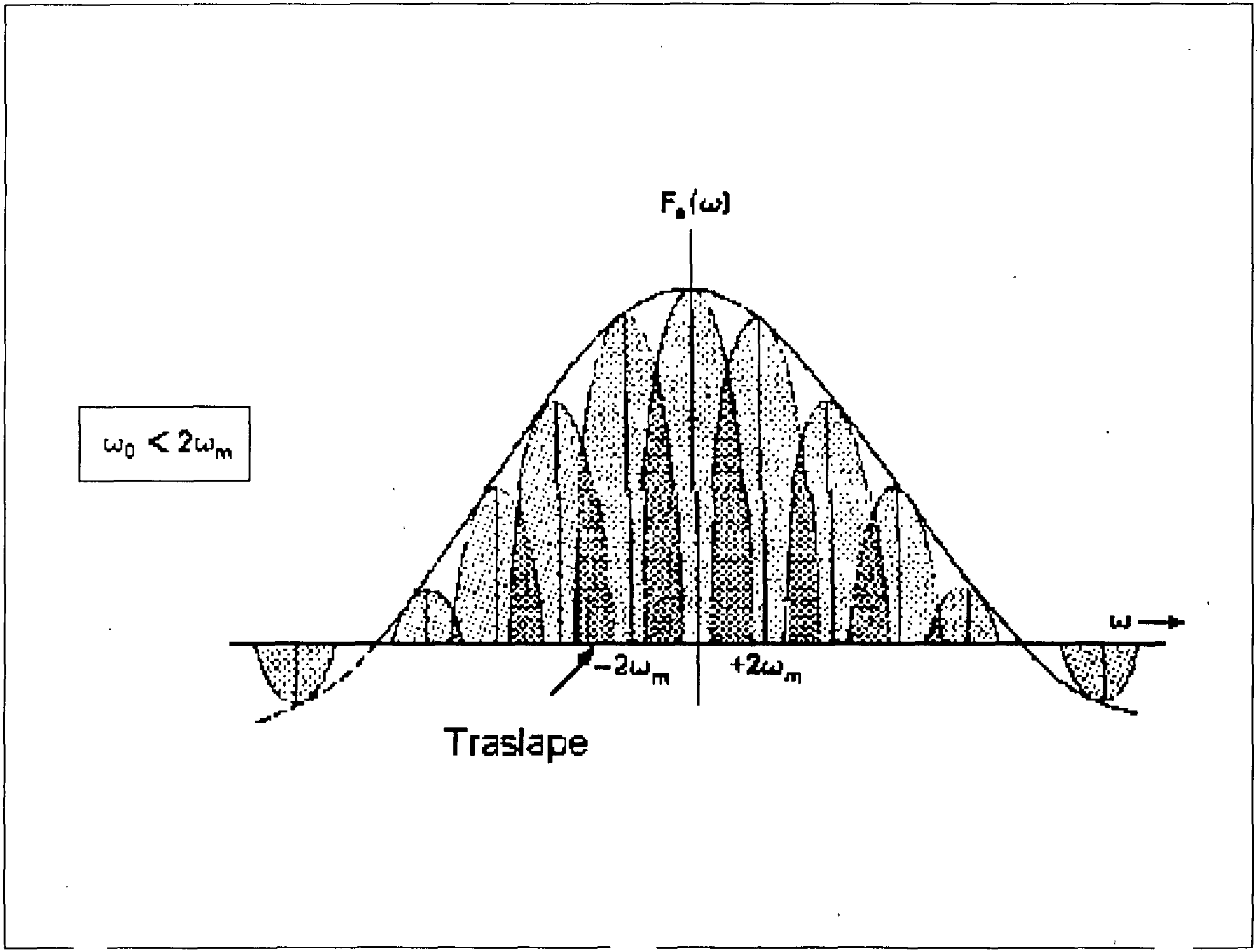
$f_s(t) = f(t)\,s(t)$
t →
$F_s(\omega)$
ω →



$\omega_0 > 2\omega_m$
$F_1(\omega)$
-2ω
$+2\omega$
ω

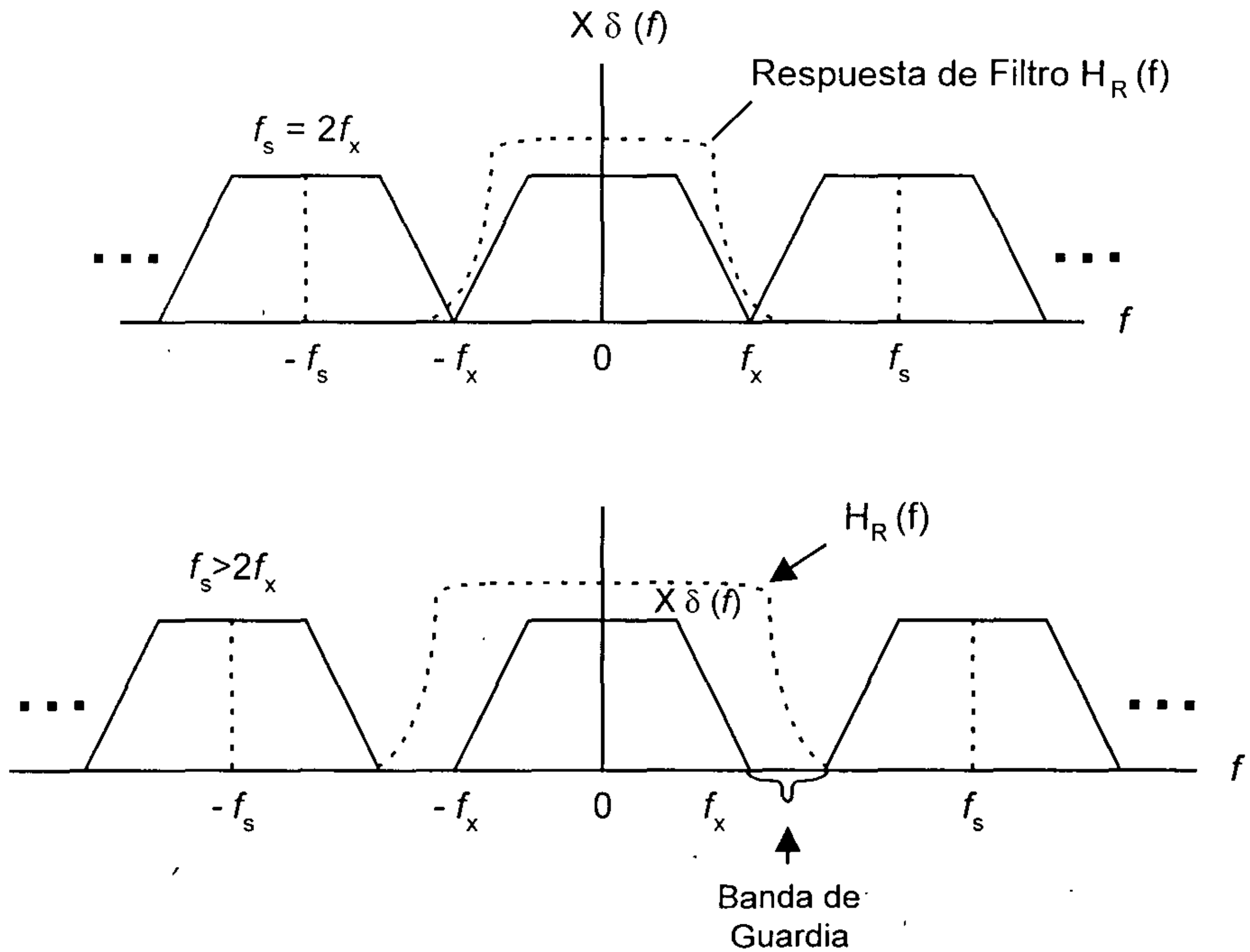


$F_1(\omega)$
$\omega_0 = 2\omega_m$
ω
$-2\omega_m$
$+2\omega_m$



$F_s(\omega)$
$\omega_0 < 2\omega_m$
ω
$-2\omega_m$
$+2\omega_m$
Traslape

Bandas de guardia en el muestreo y recuperación



CUANTIZACION

- Puesto que las muestras de la señal $f_s(t)$ pueden adquirir un continuo de valores entre sus magnitudes mínima y máxima, enviar los valores en forma codificada requeriría una enorme cantidad de bits. En la práctica es necesario representarlas por un número finito de valores predeterminados (256 pasos de cuantización).

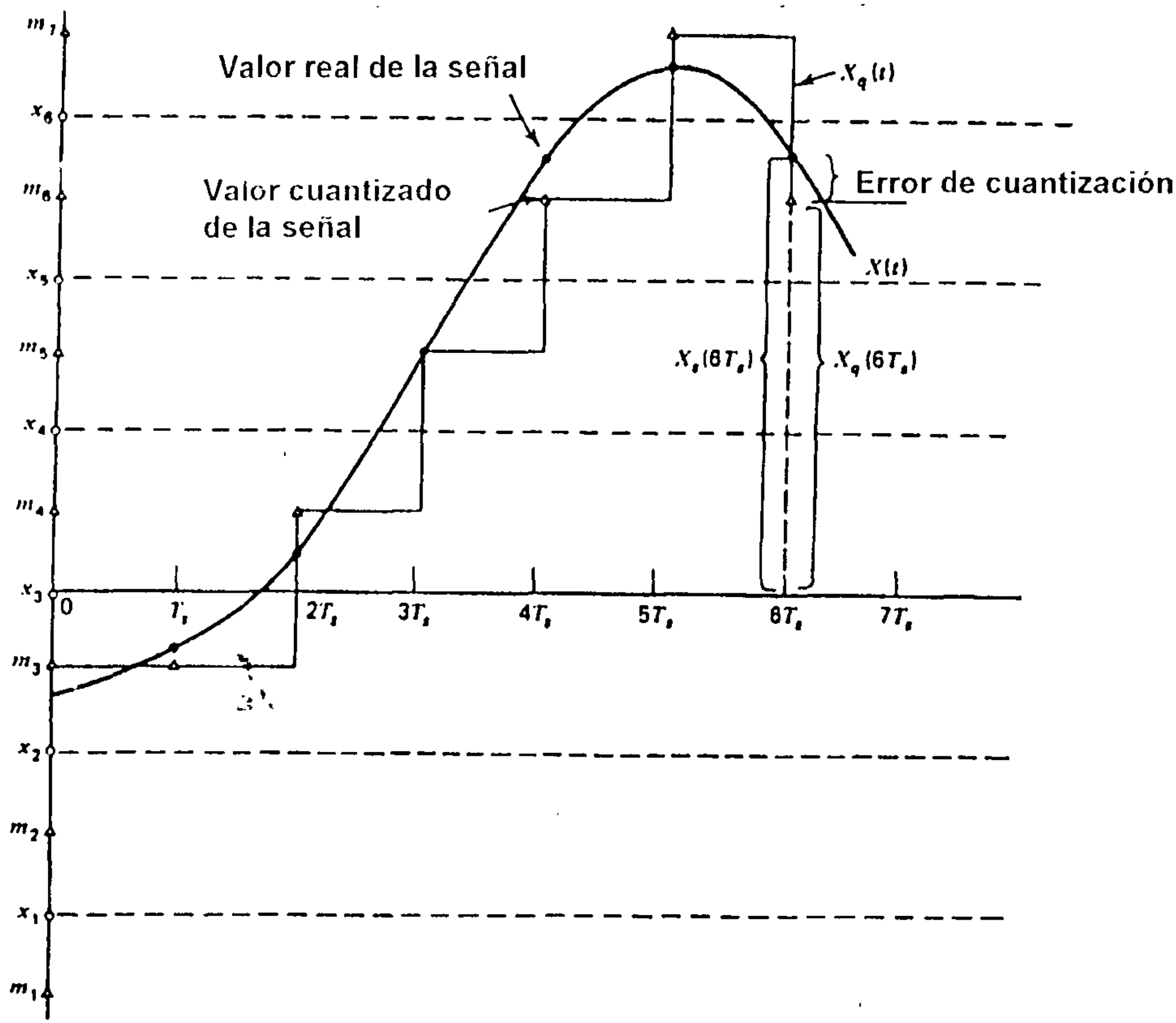
CUANTIZACION

Puesto que las muestras de la señal $f_s(t)$ pueden adquirir un continuo de valores entre sus magnitudes mínima y máxima, enviar los valores en forma codificada requeriría una enorme cantidad de bits. En la práctica es necesario representarlas por un número finito de valores predeterminados (256 pasos de cuantización).

Cuantización uniforme

- El rango de la variable aleatoria continua X se divide en Q intervalos de igual longitud (Δ). El valor cuantizado de X será el punto medio del intervalo. Si "a" y "b" son los valores mínimo y máximo de X :

$$\Delta = \frac{b-a}{Q}$$



Valor real de la señal
Valor cuantizado
de la señal
Error de cuantización
$X_q(t)$
$X(t)$
$X_s(6T_s)$
$X_q(6T_s)$
m_7
x_6
m_6
x_5
m_5
x_4
m_4
x_3
m_3
x_2
m_2
x_1
m_1
0
T_s
$2T_s$
$3T_s$
$4T_s$
$5T_s$
$6T_s$
$7T_s$

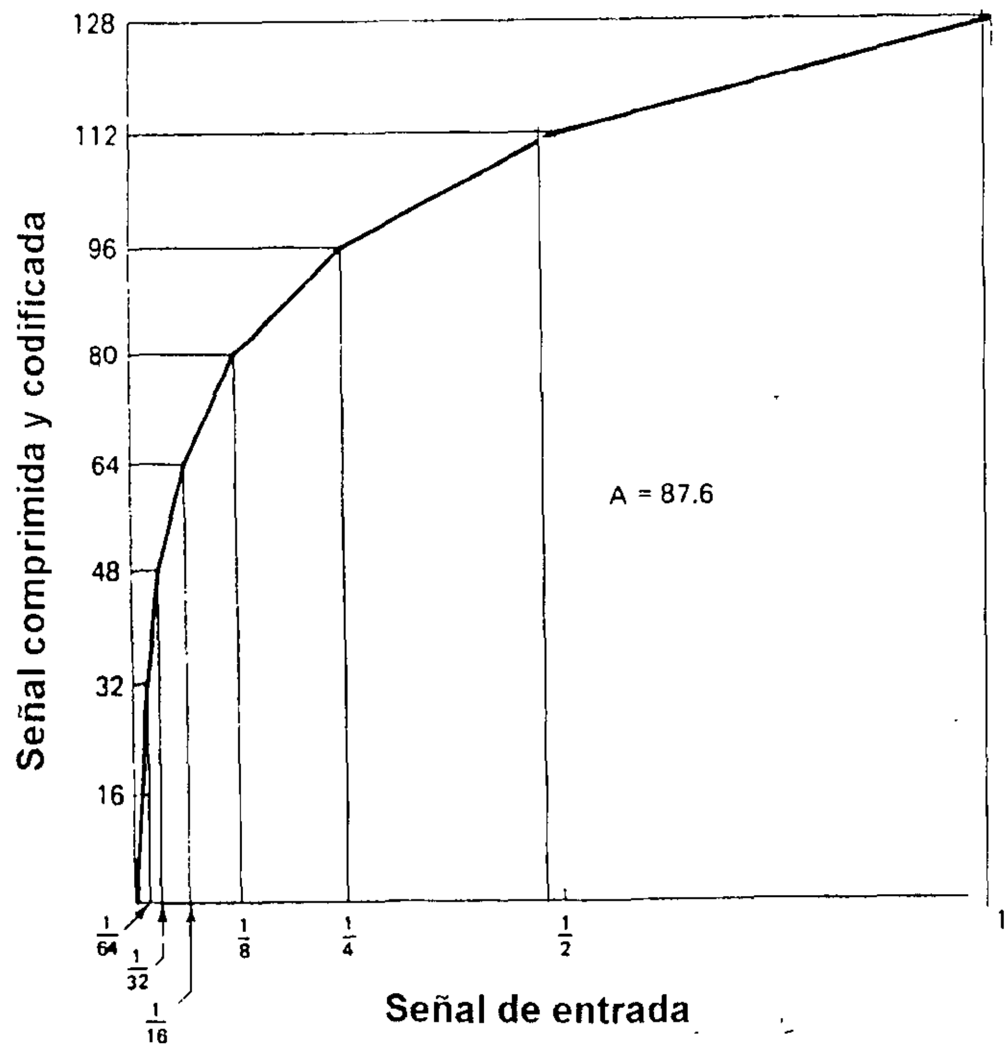
Cuantización no uniforme

- La cuantización uniforme utiliza un tamaño de escalón fijo Δ y una probabilidad de ocurrencia igual para cualquier valor de amplitud en el rango permitido.
- En el caso de voz la probabilidad de ocurrencia de amplitudes pequeñas es mucho mayor que de amplitudes grandes.

- Experimentalmente se ha encontrado que la fdp de la amplitud de voz es aproximadamente exponencial.

- En consecuencia sería apropiado proveer muchos niveles de cuantización (Δ pequeña) en la región de poca amplitud, y unos pocos niveles (Δ grandes) en la región de amplitudes grandes.

Función de transferencia del compresor.



Leyes de compresión

- ***a) Ley*** μ

$$V_c = \frac{\log\,(1+\mu V_i)}{\log\,(1+\mu)}$$

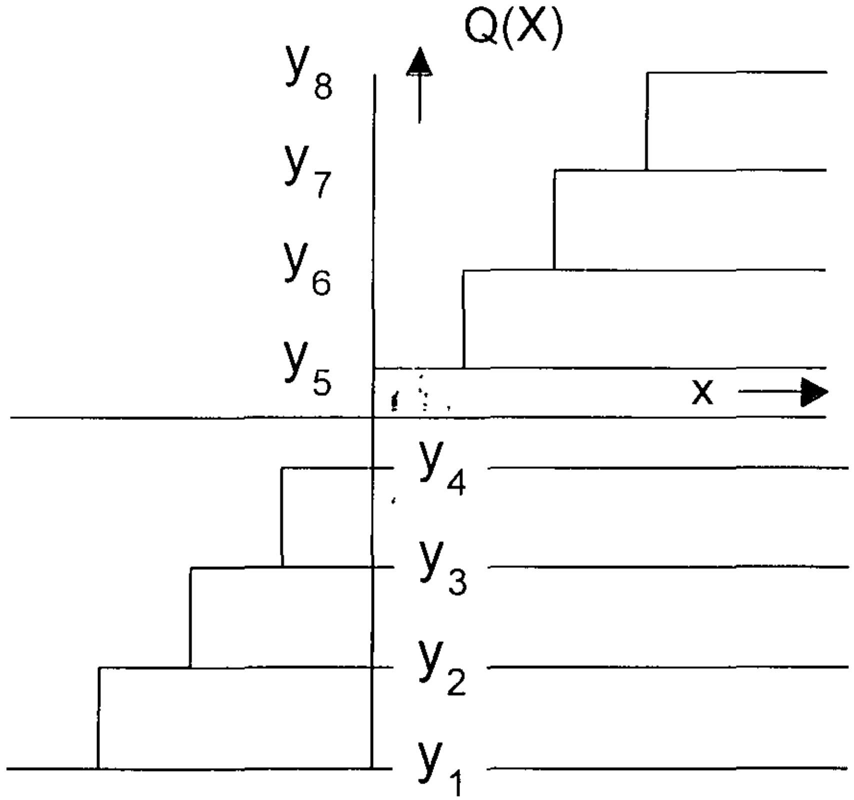
- ***b) Ley A***

$$V_c = \frac{A V_i}{1+\ \log A} \quad , \qquad 0 \leq V_i \leq \frac{1}{A}$$

$$V_c = \frac{1+\ \log\,(A V_i)}{1+\ \log A} \, , \qquad \frac{1}{A} \leq V_1 \leq 1$$

Codificación

La codificación es la operación que asigna en forma biunívoca una palabra-código de un alfabeto binario, a un nivel de cuantización.



NBC	FBC (SMC)	TCC	GC (RBC)
111	111	011	100
110	110	010	101
101	101	001	111
100	100	000	110
011	000	111	010
010	001	110	011
001	010	101	001
000	011	100	000

Código del segmento	Código de Nivel	Nivel de cuantización
000	0000	0
	0001	1
	⋮	⋮
	1111	15
001	0000	16
	0001	17
	⋮	⋮
	1111	31
⋮	⋮	⋮
110	0000	96
	0001	97
	⋮	⋮
	1111	111
111	0000	112
	0001	113
	⋮	⋮
	1111	127

Tasa de transmisión digital PCM

a	Frecuencia máxima de voz	3.4 KHz
b	Frecuencia de muestreo	8 KHz
c	Número de muestras por señal de voz	8,000 / s
d	Número de bits de una palabra PCM	8 bits
e	Velocidad binaria de un canal	(8,000 / s) 8 bits = 64 Kbits / s
f	Período de una trama	1/ 8,000 = 125 *mseg*

Multiplexión PCM

	Características específicas	PCM30	PCM24
a	Nombre comercial	E1	T1
b	Ley de codificación	Ley A	Ley μ
c	Número de intervalos de tiempo de canal por trama	32	24
d	Número de bits por trama	8 bits x 32 = 256 bits	8 bits x (24) + 1 = 193 bits
e	Velocidad binaria de la señal multiplex de tiempo	8.000 / s x 256 bits = 2.048 Kbits / s	8.000 / s x 193 bits = 1.544 Kbits / s
f	Duración de un intervalo de tiempo de canal de 8 bits	$\frac{125\mu s \times 8}{256} =$ aprox. $3{,}9\mu$seg	$\frac{125\mu s \times 8}{193} =$ aprox. $5{,}2\mu$seg

A

FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

<u>SINCRONIZACIÓN DE REDES DIGITALES</u>

EXPOSITOR: ING. GABRIEL FLORES S.
PALACIO DE MINERÍA
JUNIO DE 1999

SINCRONIZACION DE REDES DIGITALES

CURSO DE REDES DIGITALES

ING. GABRIEL FLORES S.

MEXICO.DF.

INDICE

1. OBJETIVOS:

- Establecer y analizar los principales parámetros que afectan la sincronización de la red. tales como la tasa de deslizamientos. la fluctuación de fase. la degradación de los relojes: para mantenerlos dentro de límites aceptables.

- Definir y establecer los métodos de sincronización mas adecuados para mantener una red de telecomunicaciones dentro de especificaciones aceptadas internacionalmente.

- Evitar la progresiva degradación de la información debido a el envejecimiento de los componentes de la red. controlando su envejecimiento.

2.- TERMINOLOGIA.

2.1 Reloj de referencia primario:

- Dispositivo que proporciona una señal de temporización con una desviación de frecuencia a largo plazo mantenida en un valor de 1×10^{-11} ó mejor con verificación respecto al Tiempo Universal Coordinado (UTC).

2.2 Nodo de red síncrona:

- Punto geográfico en que están interconectados uno ó más equipos digitales síncronos.

2.3 Nodo de tránsito:

- Nodo de red síncrona que enlaza con otros nodos y no directamente con el equipo de usuario.

2.4 Nodo local.

- Nodo de red síncrona que enlaza el interfaz directamente con el equipo de usuario.

2.5 Nodo esclavo o subordinado:

- Reloj cuya salida de temporización está enganchada en fase a la señal de temporización recibida de un reloj de calidad superior.

2.6 Incertidumbre:

- Expresa la magnitud de la posible desviación del valor medido con respecto al valor real o nominal de una señal.
- Frecuentemente se distinguen dos componentes, la incertidumbre sistemática y la incertidumbre aleatoria.
- La Incertidumbre sistemática se estima generalmente sobre la base de las características del parámetro y es equivalente al término "Exactitud".
- La Incertidumbre aleatoria se expresa en términos estadísticos como es la desviación típica o standar o por un múltiplo de esta. (Varianza).
- Es equivalente al término "PRECISION".
- La Incertidumbre global comprende ambas partes, la sistemática y la aleatoria y es equivalente a la exactitud total.

2.7 Exactitud:

- Es la capacidad de un reloj para generar una frecuencia tan cercana como sea posible al valor nominal. Está dada por la relación.

$$\left|\frac{\Delta f}{f}\right|$$

En donde:

f= Frecuencia nominal (Hz).
Δf= Variación de la frecuencia (Hz).

2.8 Estabilidad:

- Es el grado con que un reloj produce una misma frecuencia durante un período de tiempo una vez establecida la operación continua. Se mide a intervalos de tiempo, usando la relación.

$$\left|\frac{\Delta f}{f}\right| \cdot \frac{1}{To - T1}$$

En donde:

T0 = Tiempo inicial
T1= Tiempo final

2.9 Deslizamiento:

- Repetición o supresión de un bloque de bits en un tren de bits síncrono o plesíocrono debido a una discrepancia en las velocidades de lectura y de escritura en la memoria de los nodos digitales.

2.10 Tasa de deslizamiento:

- Se define como la cantidad de bits pérdidos o duplicados que ocurren en un cierto intervalo de tiempo y es proporcional a la diferencia de exactitudes de los relojes de los equipos enlazados. Se especifica en deslizamiento/Unidad de tiempo.

2.11 Instante significativo:

- Momento en el que las condiciones significativas de una señal digital (0 ó 1) son reconocidas por un dispositivo apropiado.

2.12 Fluctuación de fase (Jitter):

- Variación a corto plazo de los instantes significativos de una señal digital de su posición ideal en el tiempo. (Numéricamente, para frecuencias mayores a 10 Hz.).

2.13 Fluctuación lenta de fase (wander):

- Es la variación a largo plazo de los instantes significativos de una señal digital de su posición ideal en el tiempo (Para frecuencias menores de 10 Hz).

2.14 Máximo error de intervalo de tiempo (MEIT):

- Es la máxima variación entre crestas del retardo temporal de una señal de temporización dada, con respecto a una señal de temporización ideal comprendida en un periodo de tiempo ideal.

2.15 Intervalo unitario o intervalo unidad (IU):

- Diferencia nominal de tiempo entre instantes significativos consecutivos de una señal isócrona.

2.16 Nodo de sincronización:

- Es un punto de la red de sincronización en donde se originan y/o terminan señales de temporización, se considera inherente a los nodos de conmutación digital o bien un equipo dedicado para tal própósito.

2.17 Red plesiocrona:

- Red en la cual los relojes que controlan los Nodos de Sincronización son independientes y los instantes significativos de las señales que se manejan se mantienen con una variación dentro de límites muy estrechos.

2.18 Red Síncrona:

- Es una red en la cual los relojes están controlados para que idealmente trabajen a la misma frecuencia o al mismo promedio dentro de límites establecidos de diferencia de fase.

2.19 Control unilateral.

- Control establecido entre dos Nodos de sincronización, tal que la frecuencia del reloj de uno de estos Nodos es influenciado por información de temporización derivada del reloj del otro Nodo.

2.20 Método maestro-esclavo (ME):

- En este método existe un Nodo de Sincronización cuyo reloj está actuando como maestro de los demás. Los relojes restantes están subordinados a este reloj.

2.21 Método maestro-esclavo jerárquico (MEJ):

- Método de Sincronización despótico en el que todos los relojes de los nodos de sincronización están dispuestos en una jerarquía y a cada reloj se le asigna una etiqueta de identificación conforme a su posición en ella.

- En caso de fallar el enlace con el reloj maestro, se selecciona automáticamente como nuevo maestro al reloj que se designe como de rango inmediato inferior.

2.22 Memoria elástica:

- Dispositivo de almacenamiento temporal de datos que permite compensar las fluctuaciones de fase.

- Lo anterior se realiza, ya sea aumentando o disminuyendo el tiempo de almacenamiento, según sea la velocidad de los bits entrantes. Cuando la velocidad de los bits entrantes y salientes a la memoria son idénticas, ésta guardará durante un tiempo nominal los bits. Cuando la velocidad de los bits entrantes disminuye el tiempo nominal de almacentamiento se reduce, aumentando la velocidad de salida; el proceso contrario sucede cuando la velocidad de los bits entrantes aumenta.

4.- LINEAMIENTOS GENERALES.

4.1 Objetivos de la tasa de deslizamientos controlados.

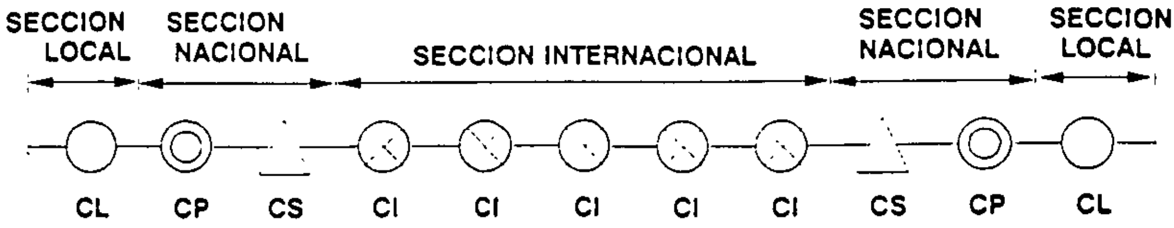
- La calidad de funcionamiento desde el punto de vista de la tasa de deslizamientos de extremo a extremo debe satisfacer las exigencias de los servicios telefónicos y no telefónicos en una conexión digital a 64 kbps para una Red Digital Integrada (RDI).

- La tasa global de deslizamientos para una conexión efectuada a través del número máximo de centrales establecida por los Planes Fundamentales de Conmutación y Transmisión de la RDI- . se indican en la tabla 4.1, para diferentes categorias de calidad.

TABLA 4.1		
CATEGORIA DE CALIDAD	OBJETIVOS DE LA TASA MEDIA DE DESLIZAMIENTOS.	PROPORCION DEL TIEMPO TOTAL $\geq$ 1 AÑO
SATISFACTORIA (S)	$\leq$ 5 DESLIZAMIENTOS EN 24 Hs	> 98.9%
ACEPTABLE (A)	> 5 DESLIZAMIENTOS EN 24 Hs Y $\leq$ 30 DESLIZAMIENTOS EN 1 Hrs.	< 1 %
INACEPTABLE (I)	> 30 DESLIZAMIENTOS EN 1 Hrs.	< 0.1 %

4.1.1 Conexiones de referencia.

- La estructura de red para la RDI- . . se basa en la conexión ficticia de referencia indicada en el Plan fundamental de Conmutación esto se muestra en la figura 4 1.1.



SIMBOLOGIA:
CL:CENTRAL LOCAL
CP:CENTRO PRIMARIO
CS:CENTRO SECUNDARIO
CI:CENTRO INTERNACIONAL

CONEXION DE REFERENCIA DE LA RED DIGITAL.

FIGURA 4.1.1

4.1.2 Distribución de las degradaciones.

- La probabilidad de que, en una red varias secciones experimenten tasas excesivas de deslizamientos que afecten simultáneamente a una conexión, es pequeña. Esto es considerado en el proceso de atribución de objetivos. La tabla 4.1.2 muestra la distribución de los objetivos para las diferentes secciones de una conexión.

TABLA 4.1.2

SECCION DE LA RED	PROPORCION ATRIBUIDA A CADA OBJETIVO	OBJETIVO COMO PROPORCION DEL TIEMPO TOTAL	
	SATISFACTORIA	ACEPTABLE	INACEPTABLE
INTERNACIONAL	8.0 %	0,08 %	0,008 %
NACIONAL	6,0 %	0.06 %	0.006%
LOCAL	40,0 %	0.4 %	0.04 %

- La Tabla 4.1.2 muestra en forma detallada la distribución de los deslizamientos entre secciones y también entre centrales.
- Las aplicaciones, por sección de Red, se muestran en los capítulos respectivos de este documento.

DISTRIBUCION POR SECCIONES

CATEGORIA DE CALIDAD	SECCION LOCAL	SECCION NACIONAL	SECCION INTERNACIONAL	SECCION NACIONAL	SECCION LOCAL
	40 % Dto/Hrs	6 % Dto/Hrs	8 % Dto/Hrs	6 % Dto/Hrs	40 % Dto/Hrs
SATISFACTORIO (S)	1/12 (0.0833)	1/80 (0.0125)	1/60 (0.0166)	1/80	1/12
ACEPTABLE (A)	> 1/12 ≤ 12	> 1/80 ≤ 1.8	> 1/60 ≤ 2.4	> 1/80 ≤ 1.8	> 1/12 ≤ 12
INACEPTABLE (I)	> 12	> 1.8	> 2.4	> 1.8	> 12

OBJETIVOS GLOBALES

(S) ≤ 5 Dto/24 Hrs

(A) > 5 Dto/24 Hrs Y ≤ 30 Dto/Hrs

(I) > 30 Dto/Hrs

DISTRIBUCION ENTRE CENTRALES

CATEGORIA DE CALIDAD	SECCION LOCAL	SECCION NACIONAL	SECCION INTERNACIONAL	SECCION NACIONAL	SECCION LOCAL
	1 CENTRAL Dto/Hrs	2 CENTRALES Dto/Hrs	5 CENTRALES Dto/Hrs	2 CENTRALES Dto/Hrs	1 CENTRAL Dto/Hrs
SATISFACTORIO (S)	1/12	0.0063 1Dto/6.6Dias	0.0033 1 Dto/12.5 Dias	0.0063	1/12
ACEPTABLE (A)	> 1/12 ≤ 12	>0.0063≤0.9	> 0.0033 ≤ 0.48	>0.0063≤0.9	> 1/12 ≤ 12
INACEPTABLE (I)	> 12	>0.9	> 0.48	>0.9	> 12

DISTRIBUCION DE LOS PORCENTAJES DE LA TASA DE DESLIZAMIENTOS PARA SECCIONES DIGITALES Y ENTRE CENTRALES DIGITALES.

TABLA 4.1.2

4.2 Características de los relojes.

- En la RDI-' los relojes se clasifican según se muestra en la tabla 4 2

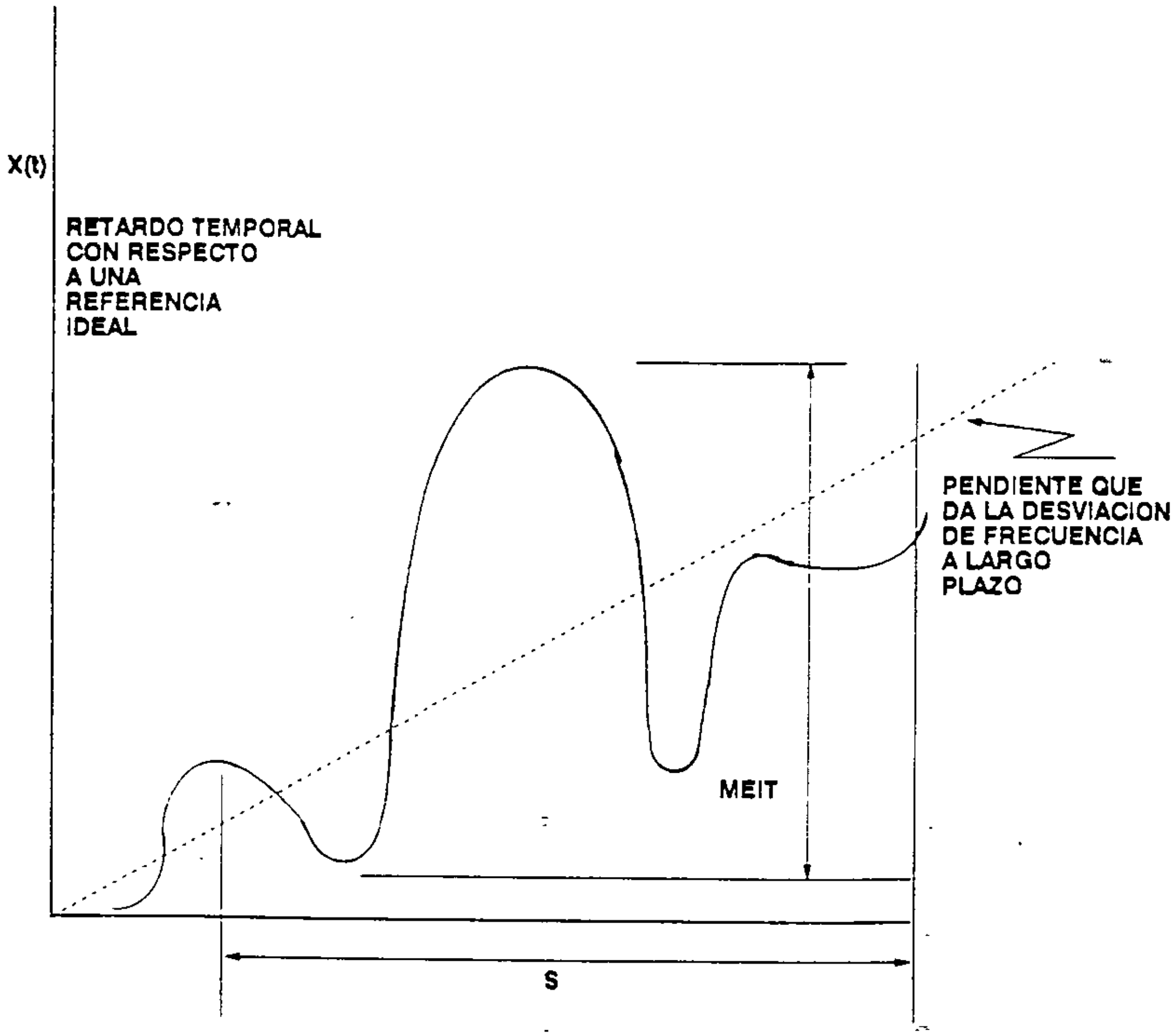
TABLA 4.2		
TIPO DE RELOJ.	EXACTITUD	ESTABILIDAD (1/DIA)
I	1×10^{-11}	1×10^{-12}
II	1×10^{-10}	1×10^{-10}
III	1×10^{-9}	1×10^{-9}

4.3 Caracteristicas de los relojes de referencia primarios (Reloj Tipo I).

4.3.1 Máximo error de intervalo de tiempo (MEIT) (MTIE).

- Es la máxima variación pico-pico del retardo de tiempo de una señal de temporización dada con respecto a una señal de temporización ideal dentro de un periodo de tiempo particular. Esto es: MTIE (s) = Max. X (t) - Min. X (t) para toda T dentro de S.

- La desviación de frecuencia a largo plazo (Δ f/f) o valor de exactitud está determinada por el cociente entre el MEIT y el intervalo de observación S cuando S aumenta, esto es:



DEFINICION DE MAXIMO ERROR DE INTERVALO DE TIEMPO (MEIT)

FIGURA 1/4.3.1

- El MEIT expresa la máxima variación de fase a largo plazo admisible en un reloj de referencia primario (con salida sinusoidal o por impuloss).

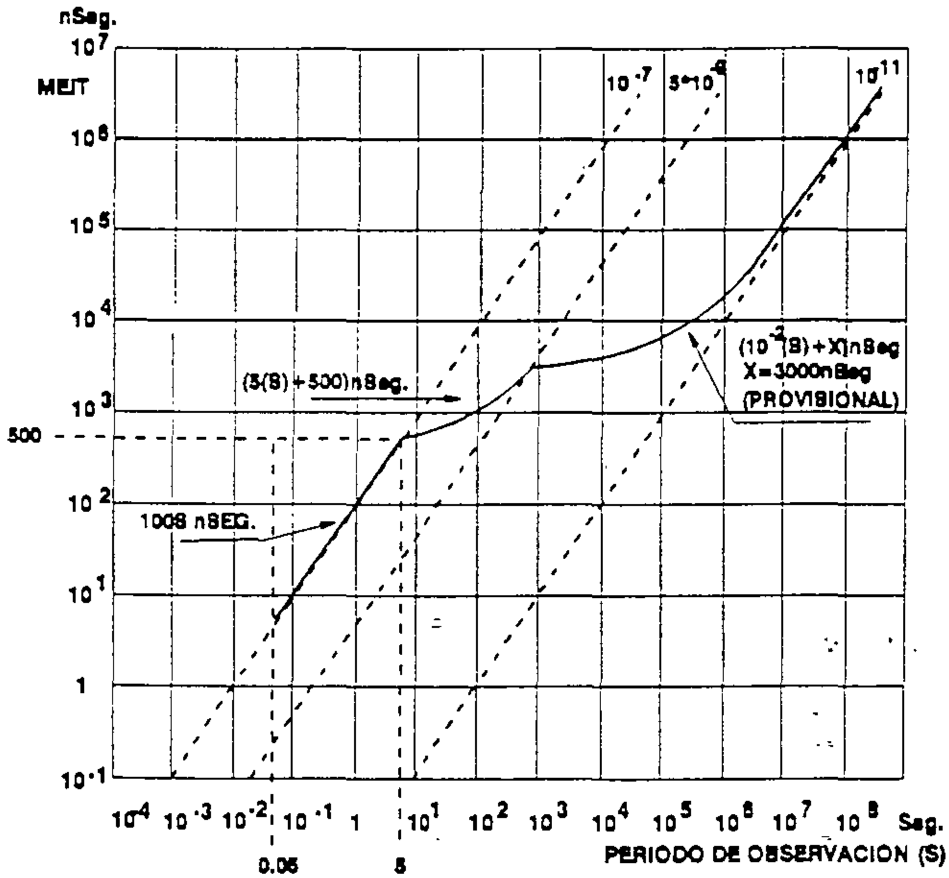
- El MEIT en un período de S segundos no excederá los siguientes límites:

a) (100 S) nseg. para $0.05 < S \leq 5$

b) (5S + 500) nseg. para $5 < S \leq 500$

c) (0.015 + X) nseg. para $S > 500$ y X = 3000 nseg.

La especificación global se muestra en la figura 2/4.3.1.



MAXIMO ERROR DE INTERVALO DE TIEMPO (MEIT) PERMITIDO DEBIDO A VARIACIONES DE FASE A LARGO PLAZO COMO UNA FUNCION DEL PERIODO DE OBSERVACION (S) PARA RELOJ DE REFERENCIA PRIMARIA

FIGURA 2/4.3.1

4.3.2 Desviación de frecuencia a largo plazo.

- El reloj de referencia primario deberá estar diseñado para "Desviaciones de frecuencia a largo plazo" no mayores de 1 x 10^{-11}
- La desviación de frecuencia a largo plazo de 1 x 10^{-11} es cerca de dos órdenes de magnitud mayor que la incertidumbre del Tiempo Universal Coordinado (UTC). Por lo tanto, el UTC deberá ser la referencia para la desviación de frecuencia a largo plazo. (Veáse CCIR reporte 898).
- Para cumplir con lo anterior se requiere que los relojes de referencia primarios sean construidos con tecnología de Haz de Cesio.

4.3.3 Estabilidad de fase.

- Puede describirse por sus variaciones de fase que a su vez se dividen en un cierto número de componentes a saber:

   a) Discontinuidades de fase, debido a perturbaciones transitorias.

   b) Variaciones de fase a largo plazo. Comprende la fluctuación lenta de fase (Wander) y desviación integrada de frecuencia.

   c) Variaciones de fase a corto plazo también conocido como fluctuación de fase (Jitter).

4.3.3a Descontinuidad de fase.

- Debido a que el/los nodo(s) de referencia primarios necesitan una fiabilidad muy alta, se requiere equipo duplicado o triplicado a fin de asegurar la continuidad de salida. Sin embargo, toda conmutación de un reloj a otro en el nodo de referencia o entre nodos de referencia primarios no deberá causar más que un alargamiento o acortamiento de la anchura del intervalo de la señal de temporización y no causará una discontinuidad superior a 1/8 del intervalo unitario a la salida del reloj. Así, si, la señal de salida es de 2,048 KHz., la discontinuidad de fase no deberá ser superior a 61.07 nseg.

4.3.3b Variaciones de fase a largo plazo.

- La variación de fase a largo plazo máxima permitida en la salida de un reloj de referencia primario es expresada como el MEIT, especificado en el inciso 4.3.1 de este documento.

4.3.3c Variaciones de fase a corto plazo.

- Se encuentra en estudio el Jitter del reloj de referencia primario.

4.4 Caracterización de los relojes subordinados (Relojes tipo II y III).

4.4.1 Máximo error relativo de intervalo de tiempo (MERIT) (MRTIE).

- El MERIT es análogo al MEIT definido en el inciso 4.3.1 de este documento, pero está referido a un oscilador práctico de alta calidad en vez del UTC.

4.4.2 Estabilidad de fase.

- Puede describirse por sus variaciones de fase que a su vez se dividen en un cierto número de componentes a saber.

a) Discontinuidad de fase. Debido a perturbaciones transitorias.

b) Variaciones de fase a largo plazo. Comprende la fluctuación lenta de fase (Wander) y la desviación integrada de frecuencia.

c) Variaciones de fase a corto plazo. También conocido como fluctuación de fase (Jitter).

4.4.2a Discontinuidad de fase.

- En los casos, infrecuentes de comprobación o reconfiguración internas en el reloj subordinado, deben satisfacerse las siguientes indicaciones:

a) Las variaciones de fase durante un periodo de hasta 2^{11} IU, no debe exceder 1/8 de IU.

b) Para períodos mayores a 2^{11} IU, la variación de fase para cada intervalo ó 2^{11} IU, no deberá excederse 1/8 de IU, hasta un total de 1µseg.

Donde el valor IU es el inverso de la velocidad binaria.

4.4.2b Variaciones de fase a largo plazo.

- Considerando que la estabilidad de fase de los relojes subordinados deben tomar en cuenta su entorno real, es necesario especificar las categorias de funcionamiento del reloj, que podemos clasificar como:

i) Ideal
ii) Forzado
iii) Mantenido.

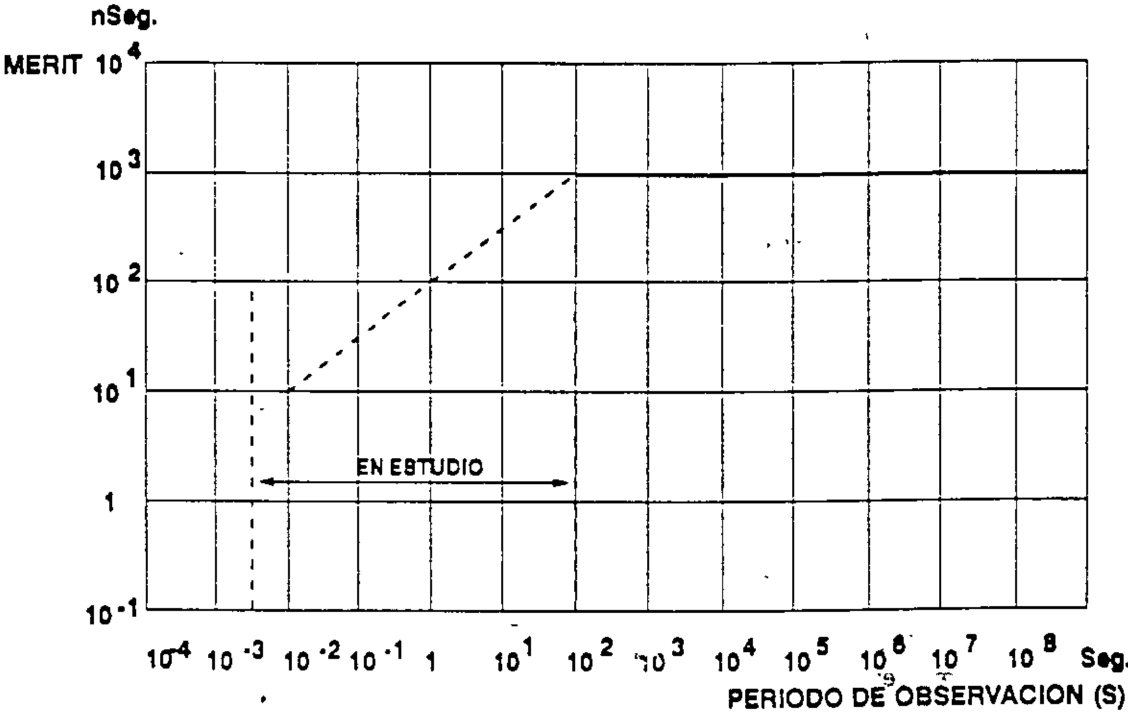
4.4.2.bi **Funcionamiento ideal.**

- Esta categoría de funcionamiento refleja el comportamiento de un reloj en condiciones en que no existen degradaciones de la o las referencias de entrada.

- El MERIT a la salida del reloj subordinado no debe en ningún período de S segundos, exceder los siguientes límites:

1) 0.05 < S < 100
2) 1000 nseg para S $\geq$ 100

- La especificación global se muestra en la fig 4.4.2bi



MAXIMO ERROR RELATIVO DE INTERVALO DE TIEMPO PERMITIDO DEBIDO A VARIACIONES DE FASE A LARGO PLAZO VS PERIODO DE OBSERVACION (S) PARA RELOJES ESCLAVOS BAJO CONDICIONES DE OPERACION IDEALES.

FIGURA 4.4.2.b.I

4.4.2.bii Funcionamiento forzado.

- Esta categoría de funcionamiento refleja el comportamiento real de un reloj considerando la influencia de las condiciones reales (forzadas) de funcionamiento. Las condiciones forzadas incluyen los efectos de la fluctuación de fase, las actividades de conmutación de protección, las ráfagas de errores.

- El resultado de estas condiciones forzadas, son causa de degradaciones de la temporización

4.4.2.biii Funcionamiento mantenido.

- Esta categoría de funcionamiento refleja el funcionamiento del reloj subordinando en las ocasiones infrecuentes que pierde la referencia durante un periodo de tiempo significativo.

- El MERIT a la salida del reloj subordinado no debe, en ningún período de S segundos, exceder los siguientes límites:

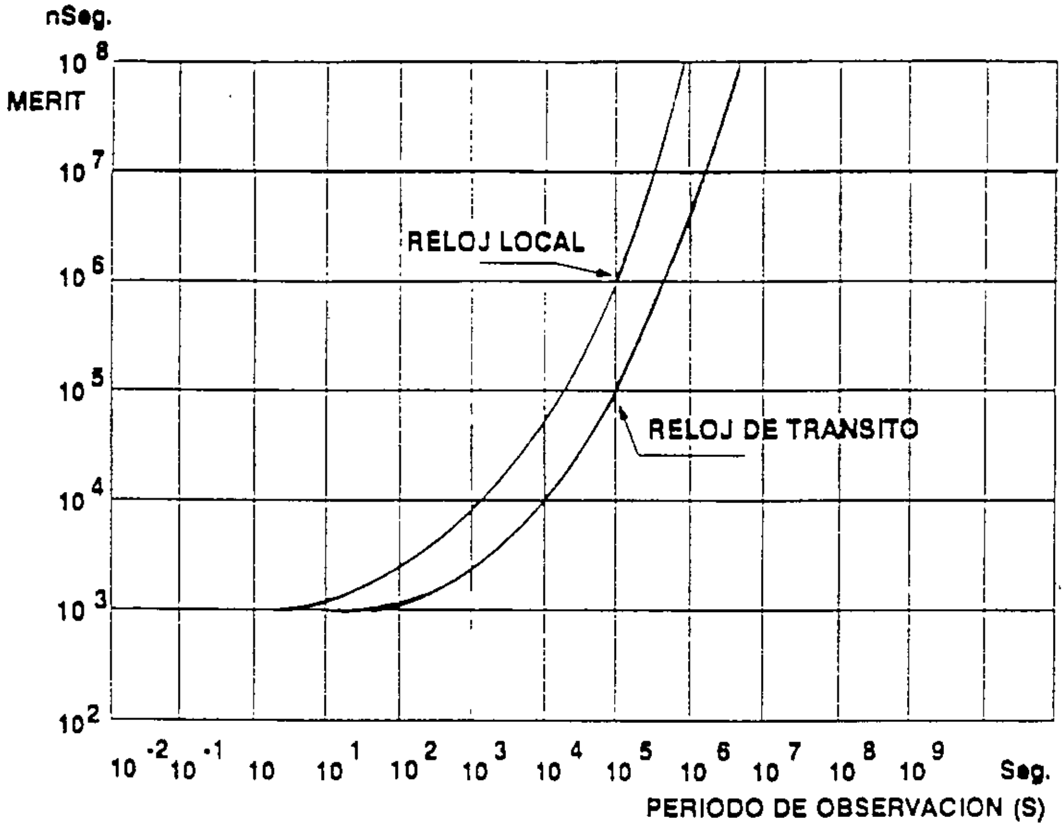
- (aS + b S + c) nseg . para S 100

- Donde las variables a, b y c toman los valores indicados en la tabla 4.4.2.biii

TABLA 4.4.2.b.III				
VARIABLE	RELOJ DE TRANSITO		RELOJ LOCAL	
a	0.5	(1)	10	(3)
b	$1.16X10^{-5}$	(2)	$2.3X10^{-4}$	(4)
c	1000	(5)	1000	(6)

NOTAS:
(1) CORRESPONDE A UN DESPLAZAMIENTO DE FRECUENCIA INICIAL DE $5X10^{-10}$
(2) CORRESPONDE A UN DERIVA DE FRECUENCIA DE $1X10^{-9}$/Dla.
(3) CORRESPONDE A UN DESPLAZAMIENTO DE FRECUENCIA INICIAL DE $1X10^{-8}$
(4) CORRESPONDE A UNA DERIVA DE FRECUENCIA DE $2X10^{-8}$
(5) EFECTO DE LA TEMPERATURA.
(6) TIENE EN CUENTA CUALQUIER MERIT QUE PUEDA HABER EXISTIDO AL COMIENZO DEL FUNCIONAMIENTO "MANTENIDO" Y LOS EFECTOS DE LA RECONFIGURACION INTERNA.(Y DE LA DISTRIBUCION DE LA TEMPORIZACION) EN CUALQUIER CASO,ES NECESARIO UNA TRANSICION GRADUAL ENTRE EL FUNCIONAMIENTO "IDEAL" Y EL "MANTENIDO".

• La especificación global resultante se resume en la figura 4.4 2.biii.



MERIT ADMISIBLE DEBIDO A LAS VARIACIONES DE FASE A LARGO PLAZO EN FUNCION DEL PERIODO DE OBSERVACION (S) PARA UN RELOJ SUBORDINADO EN FUNCIONAMIENTO MANTENIDO.

FIGURA 4.4.2.b.iii

4.5 Fluctuación de fase (FF) y fluctuación lenta de fase (FLF).

- La fluctuación de fase comprende la fluctuación de fase (Jitter) y fluctuación lenta de fase (Wander).

4.5.1 Límites de fluctuación de fase FF en redes digitales.

- En la tabla 4.5.1 se muestran los niveles máximos admisibles de la fluctuación de fase (FF) en interfaces jerárquicos de una red digital. Estos valores son compatibles con la tolerancia mínima de FF que deben proporcionar todos los accesos de entrada del equipo requerido.

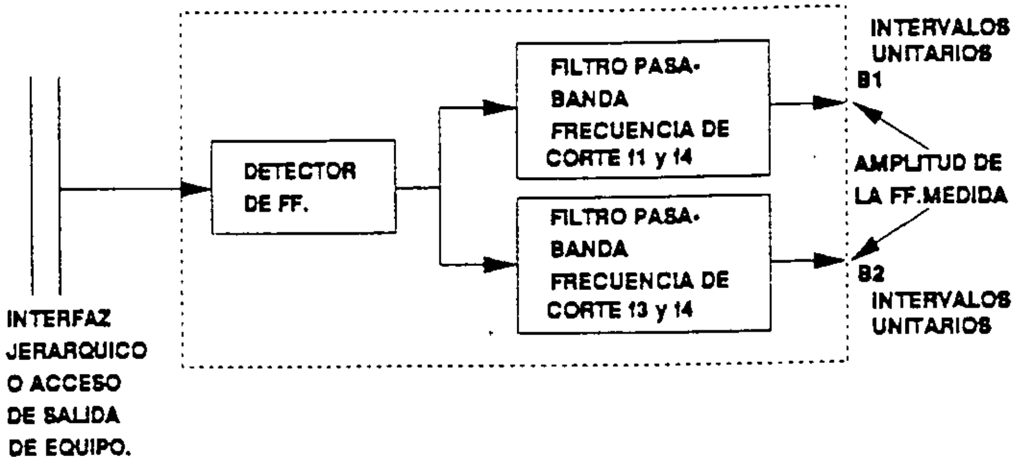
TABLA 4.5.1

VALOR DEL PARAMETRO / VELOCIDAD BINARIA (KBPS)	LIMITE DE RED		ANCHURA DE BANDA DEL FILTRO DE MEDICION		
	B1 IU pp f1 - f4	B2 IU pp f3 - f4	FILTRO PASABANDA CON UNA FRECUENCIA DE CORTE INFERIOR f1 O f3 Y UNA FREC. DE CORTE SUPERIOR f4		
			f1 (Hz)	f3 (KHz)	f4 (KHz)
64	0.25	0.05	20	3	20
2048	1.5	0.2	20	18	100
8448	1.5	0.2	20	3	400
34368	1.5	0.15	100	10	800
139264	1.5	0.075	200	10	3500

IU = INTERVALO UNITARIO, TOMA LOS SIGUIENTES VALORES :

PARA 64 KBPS, 1IU=15.6micro seg.
PARA 2048 KBPS, 1IU=488nseg
PARA 8448 KBPS, 1IU=118nseg
PARA 34368 KBPS, 1IU=29.1nseg.
PARA 139,264 KBPS, 1IU=7.18nseg.

- El montaje para la medición de los valores indicados en la tabla 4.5.1., se muestran en la figura 4.5.1. La respuesta de frecuencia de los filtros asociados a los aparatos de medida deben tener un régimen de decremento de 20Db/decáda. La recomendación 0.171 describe con detalle el aparato de medida.



CIRCUITO PARA LA MEDICION DE LA FF.PROCEDENTE DE UN INTERFAZ JERARQUICO O DE UN ACCESO DE SALIDA DE EQUIPO.

FIGURA 4.5.1

4.5.2 Límites de fluctuación lenta de fase (FLF) en redes digitales.

- El límite de red máximo para FLF en todas las interfaces jerárquicas no se tiene definido estos valores, dependen básicamente de las características del medio de transmisión y del envejecimiento de los circuitos del reloj de la central.

- Los accesos de entrada deben tolerar la FLF de acuerdo con los requerimientos de la tolerancia de entrada indicada en el inciso (3.1.1 Rec. Rev. Doc. 170692).

- Para interfaces en nodos de red, los siguientes límites son aplicables.

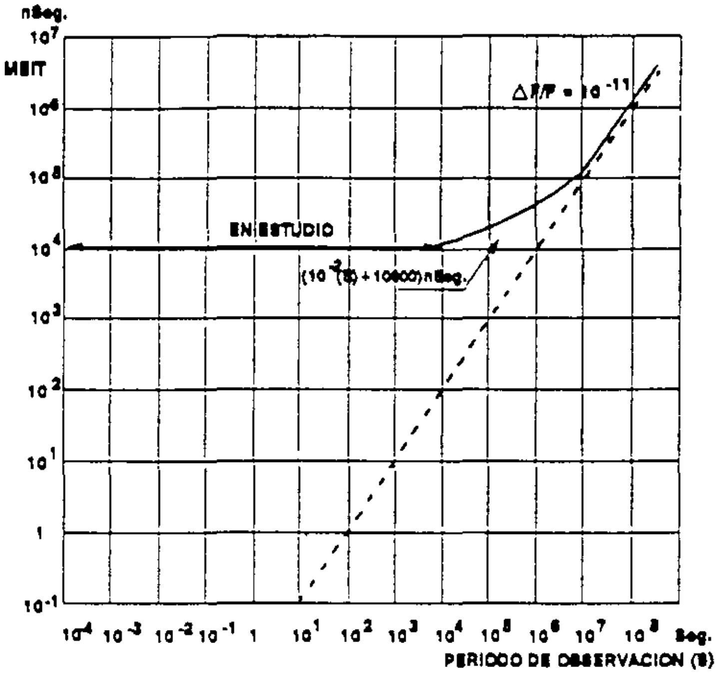
- El MTIE (Rec. G.811) sobre un período de S segundos, no deberá exceder los siguientes valores:

1) $S < 10^4$ seg.

2) 10^{-11} S seg + 10µseg.

- La especificación completa se ilustra en la figura 4.5.2 (Fig. 2/G.823).

- Nota: El MTIE total de 10µseg adicional al tiempo promedio, puede sólo ocurrir en la salida del último nodo en la cadena de nodos.



MAXIMO ERROR DE INTERVALO DE TIEMPO (MEIT) PERMITIDO VERSUS PERIODO DE OBSERVACION (S) PARA LA SALIDA DE UN NODO DE RED.

4.5.3 Limites de fluctuación de fase en equipo digital.

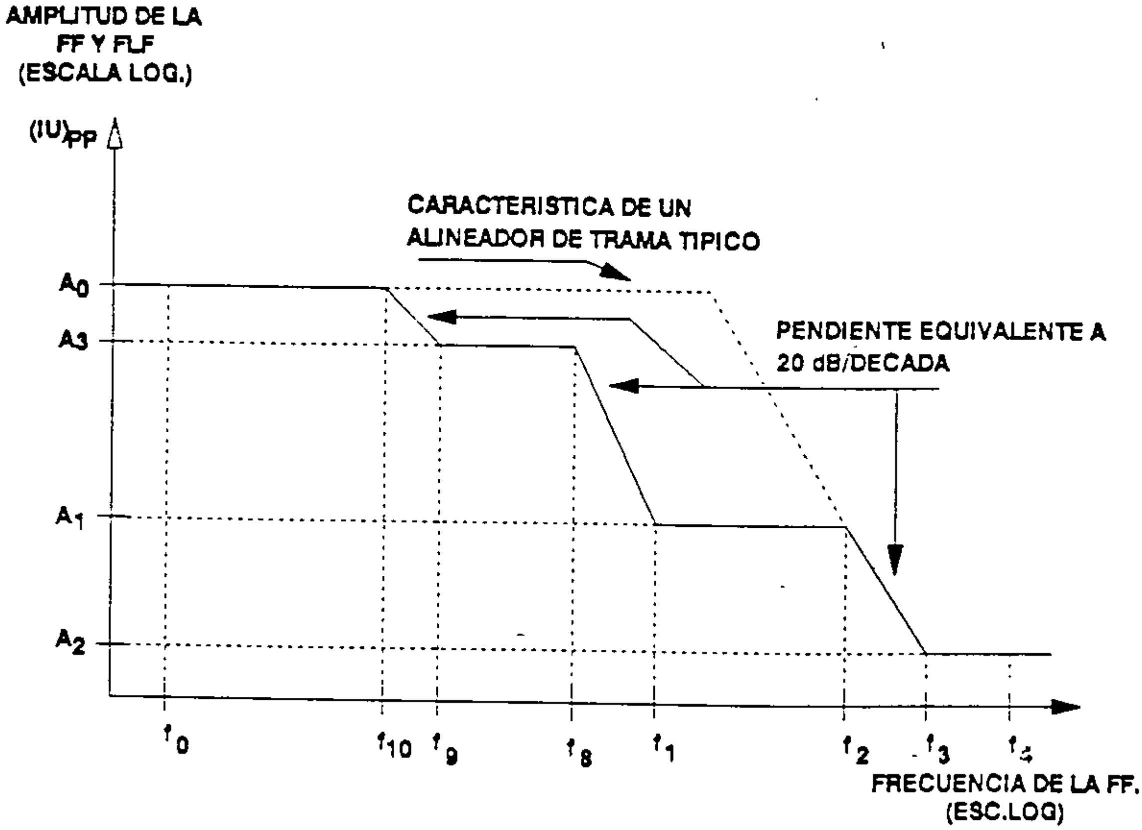
- Para equipos digitales individuales tales como multiplexores, regeneradores radios digitales, etc. es necesario especificar la calidad de funcionamiento respecto a la fluctuación de fase (FF) de tres maneras:

1. Tolerancia de fluctuación de fase en los accesos de entradas digitales, ver 4.5.3.1.
2. Fluctuación de fase máxima a la salida en ausencia de una fluctuación de fase a la entrada, ver 4.5.3.2.
3. Características de transferencia de la fluctuación de fase, ver 4.5.3.3.

4.5.3.1 Tolerancia de fluctuación de fase en los accesos de entradas digitales.

- Por conveniencia para su medición, la tolerancia de FF y FLF requerida se define en función de la amplitud y la frecuencia de una FF sinusoidal que, al modular una señal de prueba, no causa una degradación apreciable del funcionamiento del equipo. Así pues, todos los accesos de entrada digital de los equipos deben estar en condiciones de tolerar una señal digital cuyas características eléctricas cumplen la Rec. G.703 pero modulada por una FLF sinusoidal que tiene una relación amplitud frecuencia definida en la figura 4.5.3.1 y en la tabla 4.5.3.1.

- Lo anterior debe cumplirse cualquiera que sea el contenido de información de la señal digital. Para pruebas, el contenido binario equivalente de la señal modulada por la FF debe ser una frecuencia binaria seudoaleatoria como se indica en la tabla 4 5.3.1.



LIMITE INFERIOR DE LA FF. Y FLF MAXIMA TOLERABLE.

FIGURA 4.5.3.1

TABLA 4.5.3.1													
VELOCIDAD BINARIA (KBPS)	AMPLITUD IU pp				FRECUENCIA FLF. ← → FF.								SEÑAL DE PRUEBA PSEUDO-ALEATORIA
	A_0	A_3	A_1	A_2	f_0 Hz	f_{10} Hz	f_9 Hz	f_8 Hz	f_1 Hz	f_2 KHz	f_3 KHz	f_4 KHz	
64	1.15	*	0.25	0.05		*	*	*	20	0.6	3	20	2^{11}-1 (Rec.O.152)
2048	36.9 (18µS)	18 **	1.5	0.2	1.2 x 10^{-5}	$4.88*10^{-3}$ **	0.01 **	1.667 **	20	2.4 (93HZ)	18 (700Hz)	100	2^{15}-1 (Rec. O.151)
8448	152 (18µS)	*	1.5	0.2		*	*	*	20	0.4 (10.7)	3 (80)	400	2^{15}-1 (Rec. O.151)
34308	618.6 (18µS)	*	1.5	0.15	*	*	*	*	100	1	10	800	2^{23}-1 (Rec. O.151)
139264	2506.6 (18µS)	*	1.5	.075	*	*	*	*	200	0.5	10	3500	2^{23}-1 (Rec. O.151)

*** : VALORES EN ESTUDIO.**

**** : ESTOS VALORES NO SE UTILIZAN CUANDO EL ENLACE TRANSPORTA SEÑAL DE SINCRONIZACION**

IU : INTERVALO UNITARIO

PARA INTERFACES, EN REDES NACIONALES LOS VALORES EN PARENTESIS PARA F_2 Y F_3, PUEDEN SER UTILIZADOS.

VALOR DE LOS PARAMETROS PARA LAS TOLERANCIAS DE ENTRADA DE FF.Y FLF.

- Se considera que los efectos de la FLF son predominantes en frecuencias abajo de f1 En muchos equipos de transmisión, tales como sistema de línea digital y muldex síncronos que utilizan técnicas de justificación, son transparentes a estos cambios de fase de muy baja frecuencia. Sin embargo, es necesario admitir la FLF en la entrada de ciertos equipos (Por ejemplo conmutadores digitales y MULDEX síncronos).

- A diferencia de la parte de la plantilla contenida entre f1 y f4 y que reflejan la FF máxima permisible en una red digital, la parte de la plantilla a bajo de f1, no está destina a representar la FLF máxima admisible que puede producirse en la práctica. Por debajo de la frecuencia f1, la plantilla se establece de forma que, en caso necesario, el valor de el nivel de almacenamiento de la memoria a la entrada de un equipo, facilite la admisión de la FLF generada en una gran proporción de conexiones reales.

- Una entrada que sincroniza a un nodo y otro que no sincroniza el nodo, pueden derivar sus respectivas temporizaciones de el mismo reloj de referencia, pero sobre diferentes trayectorias, y pueden por lo tanto, en un caso extremo tener una desviación con fase opuesta. La esperada desviación de fase relativa máxima es de 18µseg., la cual debe ser absorbida por el equipo.

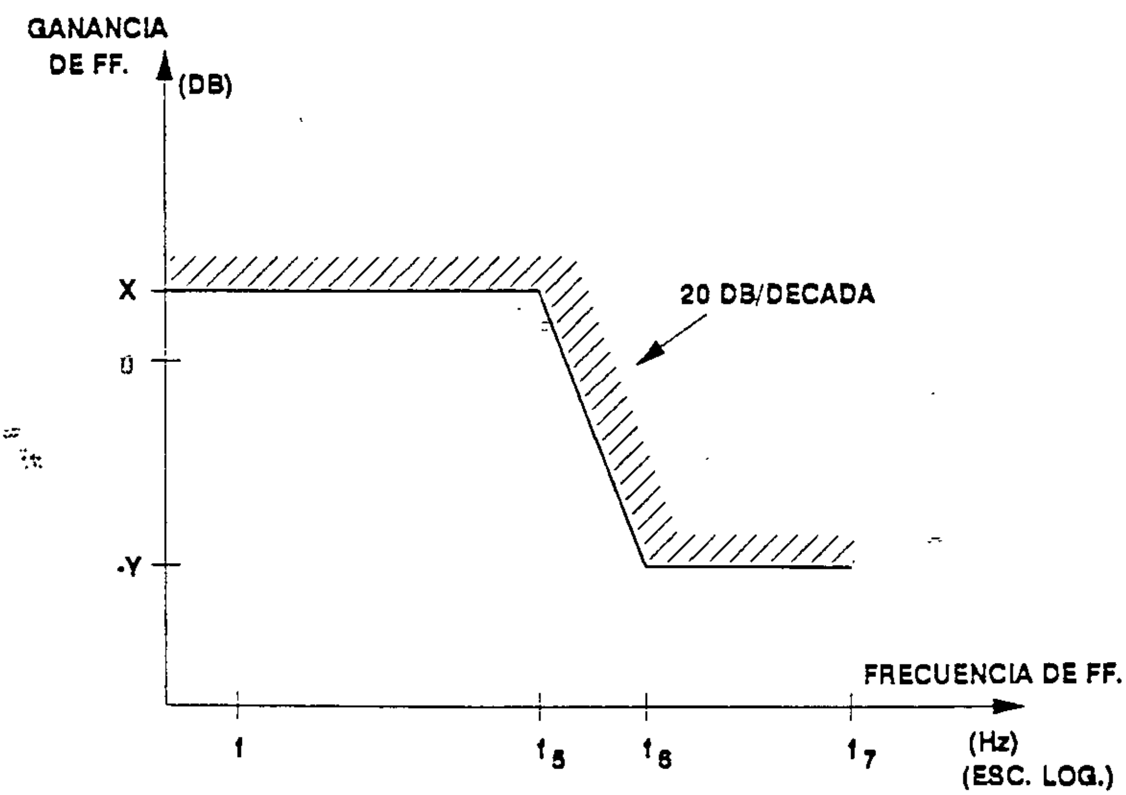
- Un intervalo corto inverso del TIE relativo entre la señal de entrada y la señal de temporización interna de el equipo terminal después de la ocurrencia de un deslizamiento controlado, no debe causar otro deslizamiento. Con el objeto de prevenir tales deslizamientos, el equipo debe ser diseñado con una histéresis adecuada para este fenómeno. Esta histéresis debe ser al menos de 18 microsegundos

4.5.3.2 Fluctuación de fase máxima a la salida en ausencia de una fluctuación de fase a la entrada.

- Es necesario limitar el nivel de la FF; producida dentro de los diferentes equipos. En las recomendaciones sobre sistemas especificos se definen los niveles máximos de FF que pueden generarse en ausencia de una ff a la entrada. Los límites efectivos aplicados dependen del tipo de equipo y deberán respetarse cualquiera que sea el contenido de información de la señal digital. En cualquier caso, los límites no sobrepasan nunca el límite máximo de red permitido (ver tabla 4.5.1).

4.5.3.3 Características de transferencia de la fluctuación de fase y de la fluctuación lenta de fase

- La función de transferencia de la fluctuación de fase se define como el valor de la ganancia de la FF versus. la frecuencia de la FF donde la ganacia es la razón de el valor de entrada y el valor de salida de la amplitud de la FF para una tasa de bit's dado. Cuando la FF está presente en el puerto de entrada del equipo digital, en muchos casos, algunas partes de la FF se transmite a el correspondiente puerto de salida digital. Muchos tipos de equipo digital atenúan inherentemente los componentes de la FF de frecuencia elevada presentes a la entrada.
- Para controlar la FF en una cascada homogénea de equipos digitales, es importante restringer el valor de la ganacia de la FF. La transferencia de la FF para un equipo digital particular, puede ser medido usando una señal digital modulada por la FF sinusoidal.
- La figura 4.5.3.3 muestra la plantilla general de las características de transferencia de la FF.



CARACTERISTICA TIPICA DE TRANSFERENCIA DE FLUCTUACION DE FASE

FIGURA 4.5.3.3

4.5.4 Secciones digitales.

4.5.4.1 Con el fin de asegurar que no se rebase el límite de red maximo dentro de una red digital, es necesario controlar la fluctuación de fase producida por los sistemas de transmisión.

- Los límites de la fluctuación de fase para las secciones digitales se dan en la Rec. G 921 (UIT-T) en estas se incluye lo siguiente:

Tolerancia.

- Límite inferior de la fluctuación de fase admisible a la entrada. Se deben satisfacer los requisitos especificados en la fig. 4.5.3.1 y la tabla 4 5.3.1.

Función de transferencia.

- Características de transferencia de la fluctuación de fase. La ganancia máxima de la función de transferencia de la ff no deberá ser superior a 1dB

- **FF Generada.**

- Fluctuación de fase a la salida en ausencia de fluctuación de fase a la entrada. La fluctuación de fase máxima pico a pico, en ausencia de fluctuación de fase a la entrada, para cualquier condición válida de la señal, no deberá exceder del límite indicado en la tabla 4.5 4

TABLA 4.5.4

VELOCIDAD BINARIA (KBPS)	LONGITUD DE LA S.D.F.R. (Km)	FF MAXIMA A LA SALIDA PARA LONGITUDES DE SECCION DIGITAL NO SUPERIOR A LA DE LA S.D.F.R. IU PP		ANCHURA DE BANDA DEL FILTRO DE MEDICION: FILTRO PASABANDA CON UNA FRECUENCIA DE CORTE INFERIOR f1 O f3 Y UNA FRECUENCIA DE CORTE SUPERIOR f4		
		LIMITE DE BAJA FRECUENCIA (f1-f4)	LIMITE DE ALTA FRECUENCIA (f3-f4)	f_1 (Hz)	f_3 (KHz)	f_4 (KHz)
2048	50	0.75	0.2	20	18 (700Hz)	100
8448	50	0.75	0.2	20	3 (80)	400
34368	50	0.75	0.15	100	10	800
34368	280	0.75	0.15	100	10	800
139264	280	0.75	0.075	200	10	3500

S.D.F.R.:SECCION DIGITAL FICTICIA DE REFERENCIA.
IU :INTERVALO UNITARIO.
FF.:FLUCTUACION DE FASE.

FF MAXIMA A LA SALIDA EN AUSENCIA DE FF A LA ENTRADA, PARA LONGITUDES DE SECCION DIGITAL NO SUPERIORES A LA S.D.F.R.

5

FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

MULTIPLEXAJE

EXPOSITOR: ING. GABRIEL MÉNDEZ HERNÁNDEZ
PALACIO DE MINERÍA
JUNIO DE 1999

MULTIPLEXAJE

ACTUALIDAD Y PERSPECTIVA

MULTIPLEXAJE

- INTRODUCCIÓN
- DESCRIPCIONES
- MULTIPLEXORES PDH
- ACTUALIDAD Y PERSPECTIVA

MULTIPLEXORES PDH
INTRODUCCIÓN

EN LA PRESENTE SECCIÓN SE ABORDARÁ EL TEMA DE MULTIPLEXAJE DIGITAL CONSIDERANDO LAS DIFERENTES JERARQUIAS QUE EXISTEN EN LA ACTUALIDAD EN LA JERARQUIA PLESIOCRONA, ASI MISMO SE PRESENTA UNA SIPNOSIS DE LAS PRINCIPALES CARACTERÍSTICAS TECNICAS QUE DEFINEN A LOS DIFERENTES MULTIPLEXORES.

MULTIPLEXORES PDH
DEFINICIONES

4007 **frame**

F: trame
S: trama

A cyclic set of consecutive time slots in which the relative position of each time slot can be identified.

4008 **multiframe**

F: multitrame
S: multitrama

A cyclic set of consecutive frames in which the relative position of each frame can be identified

MULTIPLEXORES PDH
DEFINICIONES

2024 **jitter**

F: gigue
S: fluctuación de fase

Short-term non-cumulative variations of the significant instants of a digital signal fro their ideal positions in time.

2025 **wander**

F: dérapage
S: fluctuación lenta de fase

Long-term non-cumulative variations of the significant instants of a digital signal fro their ideal positions in time.

MULTIPLEXORES PDH
DEFINICIONES

2.6 Timing

6001 **timing signal**

F: signal de rythme
S: señal de temporización

A cyclic signal used to control the timing of operations.

6002 **timing recovery [timing extraction]**

F: récupération du rythme
S: recuperación de la temporización [extracción de la temporización]

The derivation of a timing signal from a received signal.

6003 **retiming**

F: réajustement du rythme
S: reajuste de la temporización

Adjustment of the intervals between the significant instants of a digital signal, by reference to a timing signal.

6004 **time-slot**

F: créneau temporel [intervalle de temps]
S: intervalo de tiempo [sector de tiempo, celda de tiempo]

Any cyclic time interval that can be recognized and defined uniquely

MULTIPLEXORES PDH
DEFINICIONES

6008 **frame alignment time-slot**

F: créneau temporel de verrouillage de trame
S: intervalo de tiempo de alineación de trama

A time slot occupying the same relative position in every frame and used to transmit the frame alignment signal.

MULTIPLEXORES PDH
DEFINICIONES

6014 **isochronous**

F: isochrone
S: isócrono

The essential characteristic of a time-scale or a signal such that the time intervals between consecutive significant instants either have the same duration or durations that are integral multiples of the shortest duration.

NOTE – In practice, variations in the time intervals are constrained within specified limits.

6015 **anisochronous**

F: anisochrone
S: anisócrono

The essential characteristic of a time-scale or a signal such that the time intervals between consecutive significant instants do not necessarily have the same duration or durations that are integral multiples of the shortest duration.

6016 **synchronous [mesochronous]**

F: synchrone [mésochrone]
S: síncrono [mesócrono]

The essential characteristic of time-scales or signals such that their corresponding significant instants occur at precisely the same average rate.

NOTE – The timing relationship between corresponding significant instants usually varies between specified limits.

MULTIPLEXORES PDH
DEFINICIONES

6019 **plesiochronous**

F: plésiochrone
S: plesiócrono

The essential characteristic of time-scales or signals such that their corresponding significant instants occur at nominally the same rate, any variation in rate being constrained within specified limits.

NOTES

1 Two signals having the same nominal digit rate, but not stemming from the same clock or homochronous clocks, are usually plesiochronous.

2 There is no limit to the time relationship between corresponding significant instants.

6020 **heterochronous**

F: hétérochrone
S: heterócrono

The essential characteristic of time-scales or signals such that their corresponding significant instants occur at different nominal rates.

NOTES

1 Two signals having different nominal digit rates, and not stemming from the same clock or from homochronous clocks are usually heterochronous

2 Terms 6014 to 6020 are based on the following Greek roots

iso = equal
homo = same
plesio = near
hetero = different

MULTIPLEXORES PDH
DEFINICIONES

CODIGO AMI

ES UN CODIGO TEMARIO EN EL CUAL LAS MARCAS SE VAN ALTERNANDO EN POLARUDAD EN FORMA CONSECUTIVA·
1) CADA MARCA "1" EN LA SEÑAL BINARIA ES CODIFICADA ALTERNATIVAMENTE COMO UN PULSO POSITIVO SEGUIDO DE UN PULSO NEGATIVO.
2) CADA ESPACIO ES REPRESENTADO POR UN NO PULSO.

CODIGO HDB3

HDB-3 ES UN CODIGO DE 3 NIVELES EN DONDE CUALQUIER SECUENCIA BINARIA EN LINEA NO INCLUYE MAS DE 3 ESPACIOS CONSECUTIVOS, ES IDENTICO AL CODIGO AMI (INVERSIÓN ALTERNA DE MARCAS) EN DONDE:
EN EL CODIGO HDB3 ADEMAS UNA CADENA DE 4 ESPACIOS EN LA SEÑALBINARIA SE CODIFICA YA SEA COMO 000V O 100V SIENDO V UNA MARCA QUE VIOLA LA REGLA DE INVERSION ALTERNA DE MARCAS TENIENDO LA MISMA POLARIDAD QUE LA MARCA ANTERIOR.

VIOLACIONES SUCESIVAS DEBEN ALTERNARSE EN POLARIDAD PARA QUE NO EXISTA UN COMPONENTE DE DC EXISTA EN LA LINEA.

LA SECUENCIA 100V SE UTILIZA CUANDO UNA MARCA ANTERIOR TIENE LA MISMA POLARIDAD QUE LA VIOLACIÓN PRECEDENTE O ES POR SI MISMA UNA VIOLACIÓN.

LA SECUENCIA 000V SE UTILIZA CUANDO LA MARCA INMEDIANTAMENTE ANTERIOR ES DE LA POLARIDAD OPUESTA A LA DE LA VIOLACIÓN PRESEDENTE Y NO ES UNA VIOLACIÓN POR S I MISMA.

MULTIPLEXORES PDH
DEFINICIONES

CODIGO HDB3

VENTAJAS.

LA MAXIMA DENSIDAD DE POTENCIA DEL ESPECTRO EN FRECUENCIA DE LA SEÑAL SE SITUA ALREDEDOR DE LA MITAD DE LA FRECUENCIA DE SINCRONIA.

LA FRECUENCIA DE SINCRONIA NO APARECE EN EL ESPECTRO.

NO EXISTE UNA COMPONENTE DE DC.

SECUENCIAS LARGAS DE CEROS SON ELIMINADAS PARA FACILITAR LA RECUPERACIÓN DE LA SEÑAL DE SINCRONIA.

SU REDUNDANCIA PERMITE OBTENER LA DETECCIÓN DE ERRORES EN EL EXTREMO DE RECEPCIÓN.

MULTIPLEXORES PDH
DEFINICIONES

CODIGO CMI

INVERSION DE MARCA CODIFICADA, ES UNA CODIFICACIÓN DE 2 NIVELES SIN RETORNO A CERO (NRZ) EN ESTE CASO:

- BINARIO 0 ES CODIFICADO DE FORMA TAL QUE LOS NIVELES DE AMPLITUD A1 Y A2 SE ALCANZAN CONSECUTIVAMENTE, CADA UNA DURANTE UN MEDIO INTERVALO DE TIEMPO.

- BINARIO 1 ES CODIFICADO YA SEA POR EL NIVEL DE AMPLITUD A1 O EL NIVEL DE AMPLITUD A2 DURANTE UN INTERVALO COMPLETO DE UNIDAD DE TIEMPO DE FORMA TAL QUE LOS NIVELES SE ALTERNANN PARA LOS 1 BINARIOS SUCESIVOS.

MULTIPLEXORES PDH
DEFINICIONES

[redacted]

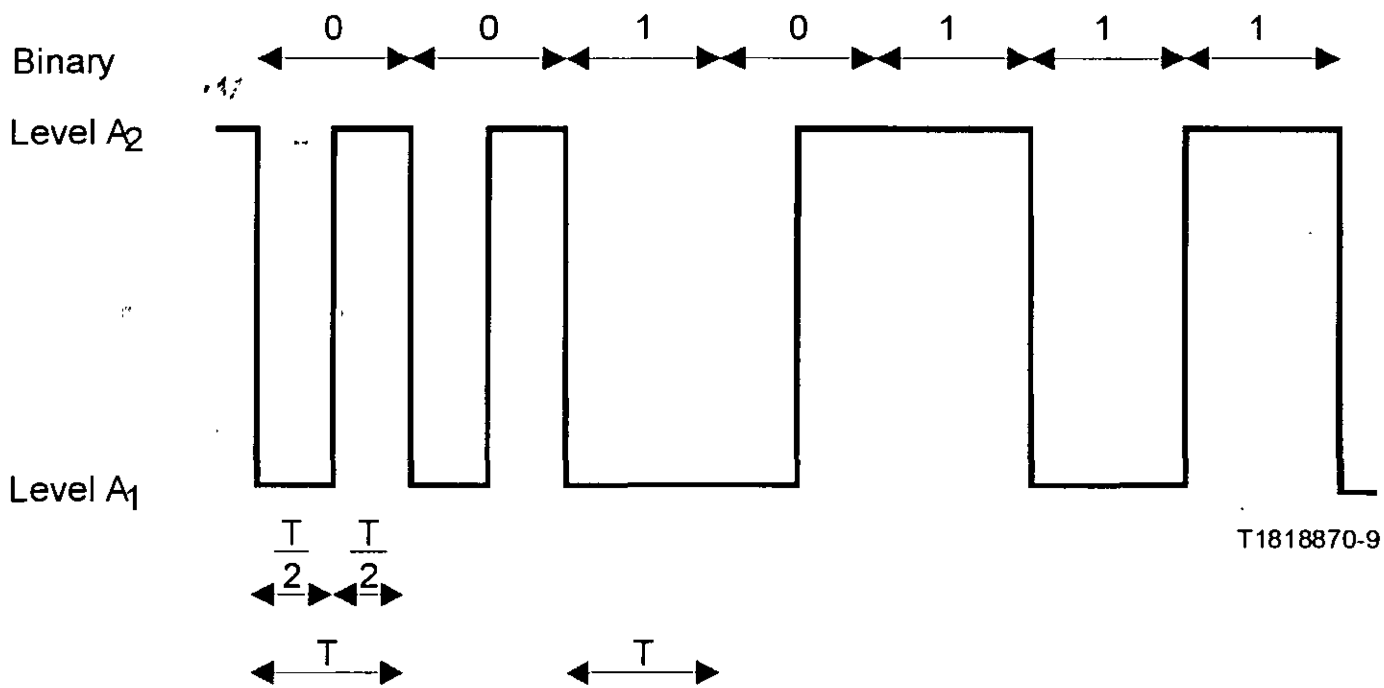


FIGURE 18/G.703

Example of CMI coded binary signal

MULTIPLEXORES DEFINICIONES

MULTIPLEXORES POR DIVISION DE FRECUENCIA:

EN ESTE TIPO DE MULTIPLEXORES SE DIVIDE LA UTILIZACIÓN DEL MEDIO DE TRANSMISIÓN EN FRECUENCIA PARA ENVIAR EN CADA BANDA DE FRECUENCIA LA INFORMACIÓN DE UNO DE LOS CANALES A MULTIPLEXAR.

MULTIPLEXORES POR DIVISION DE TIEMPO:

EN ESTE TIPO DE MULTIPLEXORES SE DIVIDE EL TIEMPO DE UTILIZACIÓN DEL MEDIO DE TRANSMISIÓN PARA ENVIAR EN CADA INTERVALO DE TIEMPO LA INFORMACIÓN DE UNO DE LOS CANALES A MULTIPLEXAR.

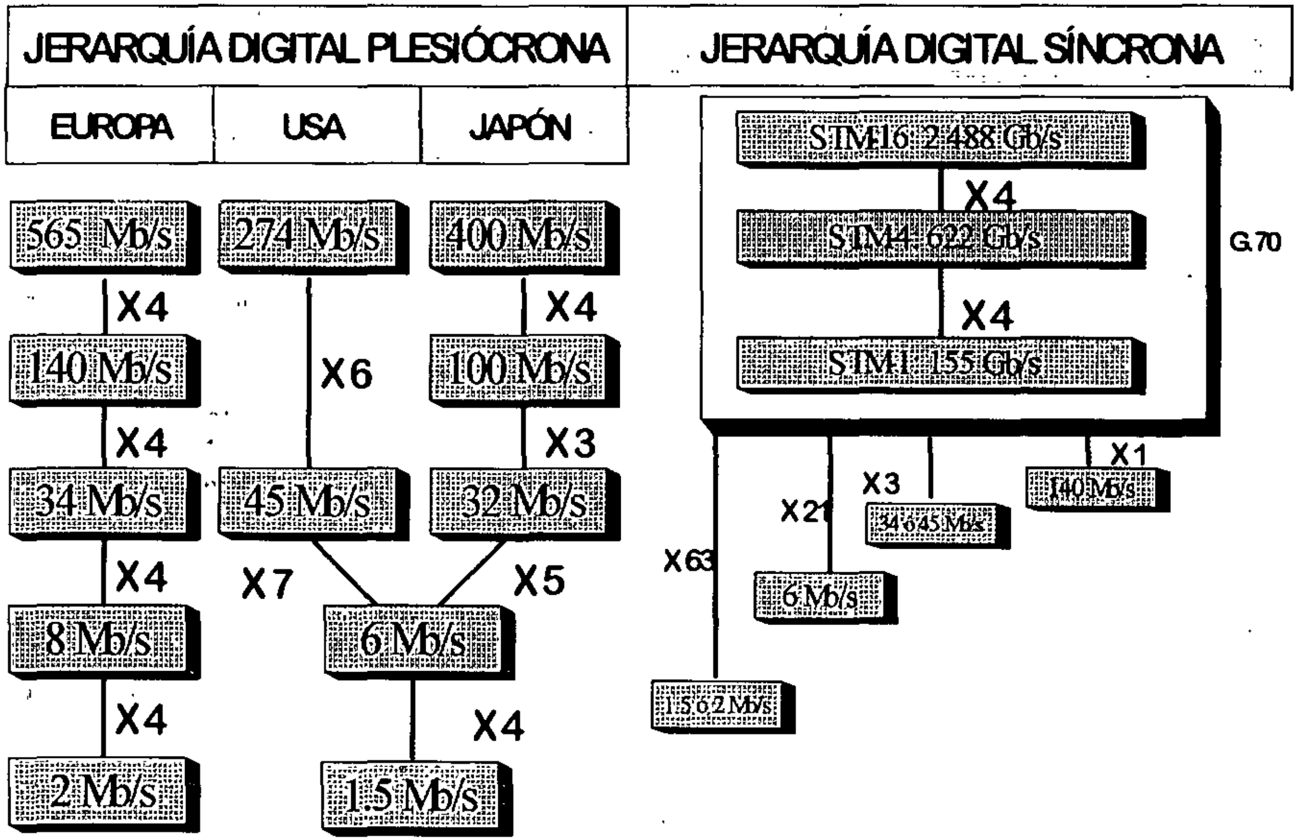
LOS MULTIPLEXORES PDH SON DEL TIPO TDM REALIZANDO LA MULTIPLEXACIÓN DE BIT A BIT DE CADA UNO DE LOS AFLUENTES O TRIBUTARIOS.

LOS AFLUENTES DE LOS MULTIPLEXORES PDH CORRESPONDEN A SEÑALES DIGITALES QUE SE ENCUENTRAN NORMALIZADAS DENTRO DE UNA JERARQUIA DIGITAL, EXISTIENDO 2 VERTIENTES LA AMERICAN BASADA EN SEÑALES DE 1544Kbit/s Y LA EUROPEA BASADA EN SEÑALES DE 2048Kbits/s ESTAS SEÑALES PUEDEN PROVENIR DE MULTIPLEXORES MIC , MULTIPLEXORES DE DATOS CODIFICADORES DE VIDEO U OTROS.

MULTIPLEXORES PDH
JERARQUIAS

Nivel de Jerarquía Digital	velocidades binarias jeraraquia (kbit/s) para redes con la jerarquía digital basada en un primer orden de:		
	1544Kbit/s		2048Kbit/s
1	1544		2048
2	6312		8448
3	32 064	44 736	34368
4	97 728		139264

MULTIPLEXORES PDH
JERARQUIAS



MULTIPLEXORES PDH
NIVELES JERARQUICOS A 2MB/S

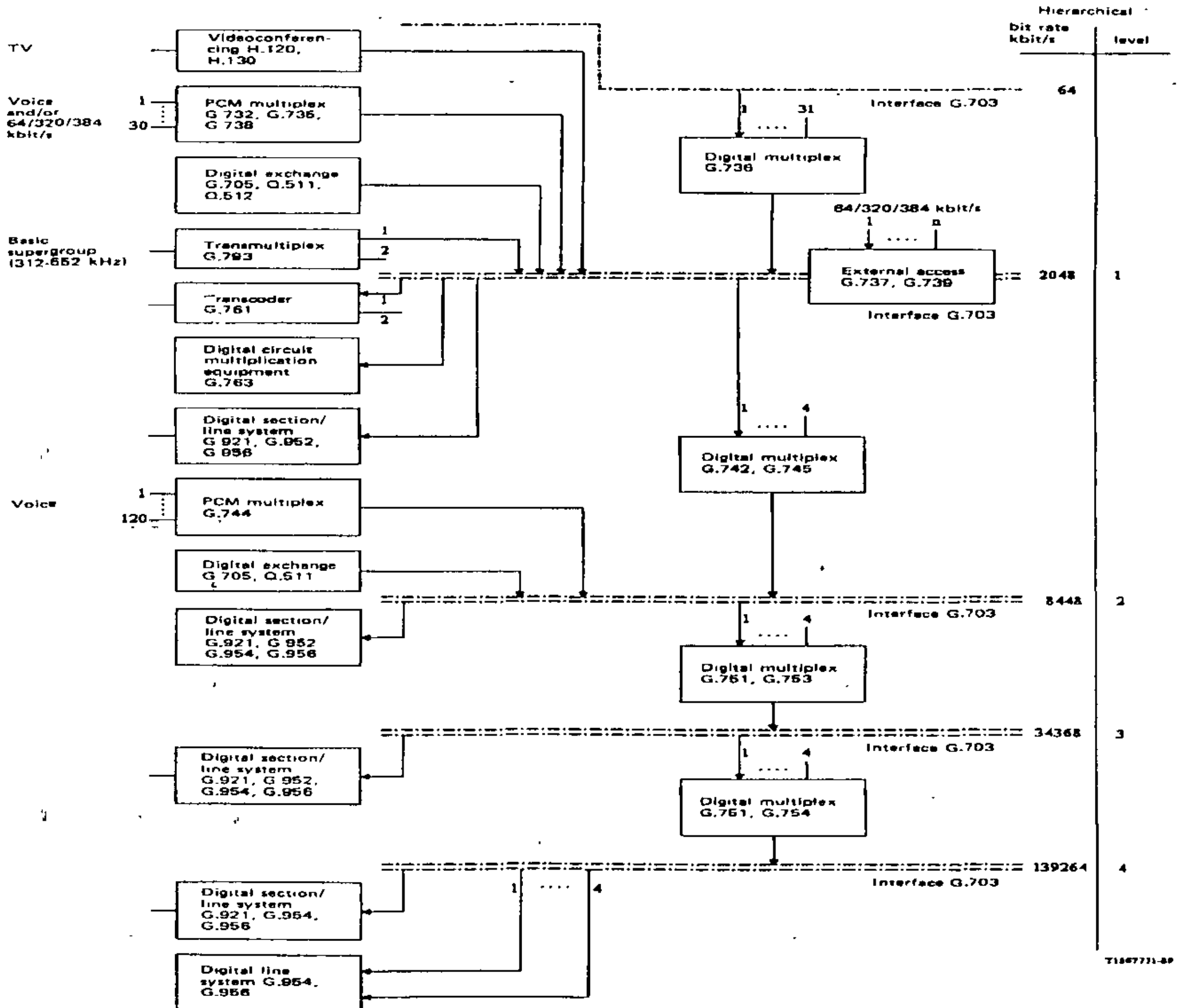


FIGURE 2/G.702

Hierarchical bit rates for networks with the digital hierarchy based on the first level bit rate of 2048 kbit/s

MULTIPLEXORES PDH
NIVELES JERARQUICOS A 1.5MB/S

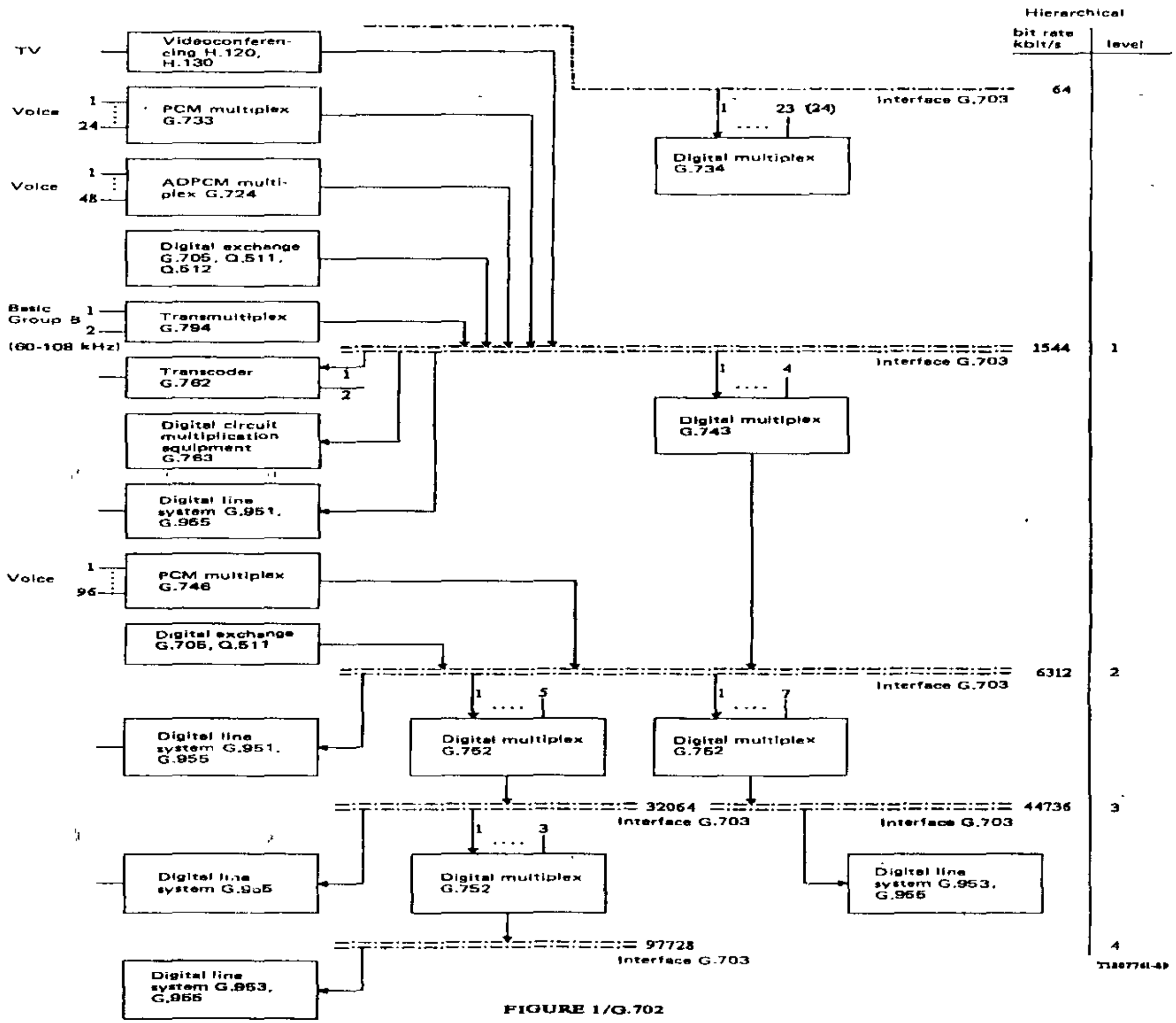


FIGURE 1/G.702

Hierarchical bit rates for networks with the digital hierarchy based on the first level bit rate of 1544 kbit/s

MULTIPLEXORES PDH
MULTIPLEXOR DE PRIMER ORDEN

MULTIPLEX MIC DE PRIMER ORDEN 2048Kbit/s.

ESTE EQUIPO MULTIPLEXA 30 SEÑALES, SEÑALES DE VOZ MODULADAS MEDIANTE TECNICA MIC (PCM) O DATOS EMPAQUETADOS A 64KB/S O NX64KBIT/S, FORMANDO UNA TRAMA DE 30 INTERVALOS DE TIEMPO AL QUE SE LE AÑADEN UN INTERVALO MAS PARA SINCRONIA Y UNO MAS PARA SEÑALIZACIÓN.

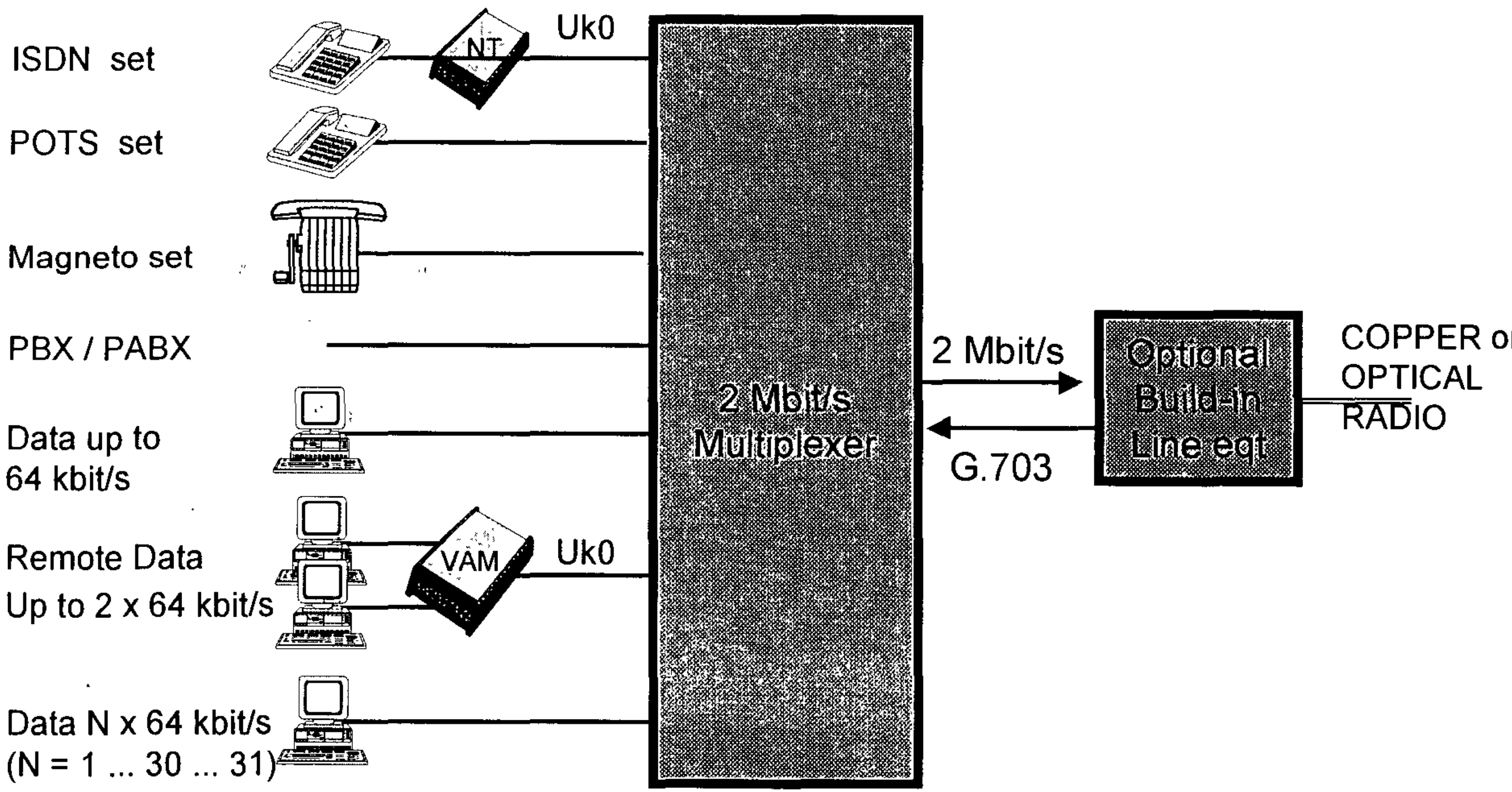
CARACTERÍSTICAS GENERALES

CAUDAL NOMINAL DE LA SEÑAL MULTIPLEXADA	2048 Kbit/s +/- 50×10^{-6}
CAPACIDAD EQUIVALENTE DE CANALES A 64Kbit/s	30
ESTRUCTURA DE TRAMA	CONFORME A REC. G.704
	CONFORME A REC. G.732

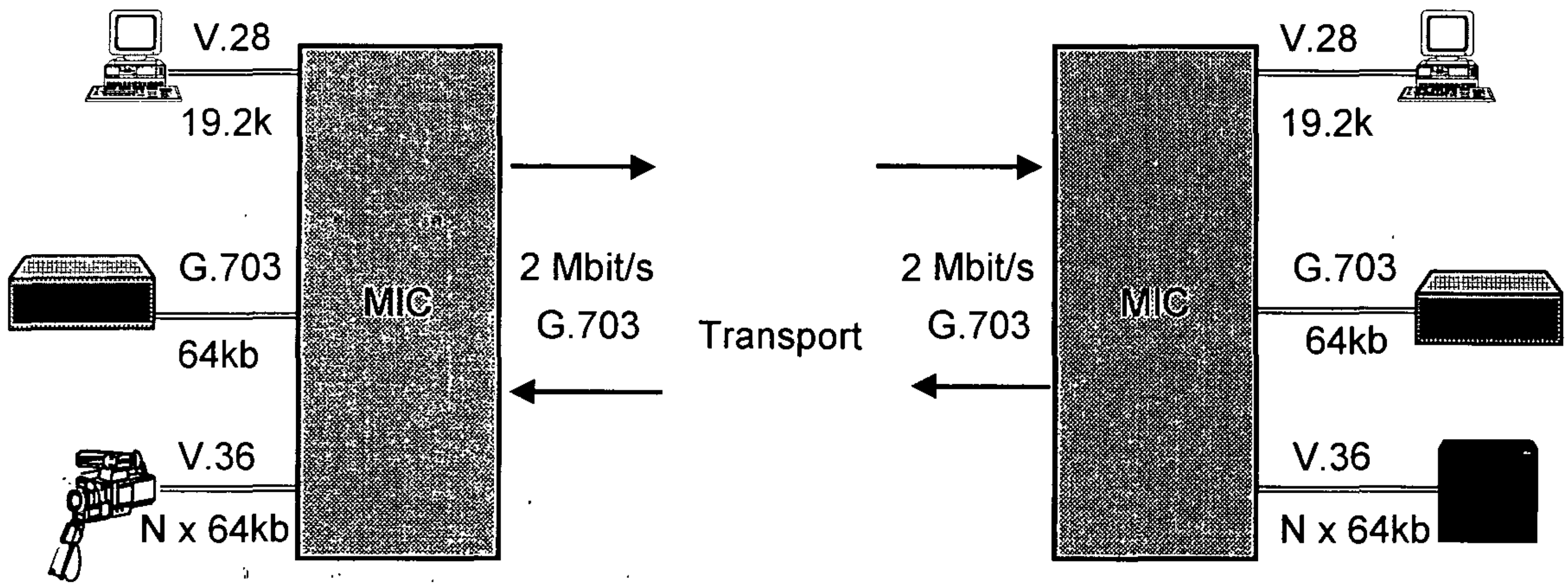
INTERFAZ DIGITAL A 2048Kbit/s

IMPEDANCIA	120 OHMS par simétrico
	75 Ohms cable Coaxial
CODIGO	HDB3
ATENUACION ADMISIBLE A LA ENTRADA A 1024Kbit/s	6dB

MULTIPLEXORES PDH
MULTIPLEXOR DE PRIMER ORDEN



MULTIPLEXORES PDH
MULTIPLEXOR DE PRIMER ORDEN



MULTIPLEXORES PDH
INTRODUCCIÓN

JUSTIFICACIÓN POSITIVA:

EN REDES ASINCRONAS LOS RELOJES DE 2048Kbit/s DE LOS EQUIPOS SON INDEPENDIENTES. LAS FRECUENCIA DE SINCRONIA TIENE EL MISMO VALOR NOMINAL, SIN EMBARGO PUEDEN VARIAR DENTRO DE LIMITES ESPECIFICADOS

PARA PODER REALIZAR UNA MULTIPLEXACIÓN POR DIVISIÓN DE TIEMPO LA SEÑALES DEBEN SER ANTES SINCRONIZADAS, PARA ELLO LAS SEÑALES SON CONVERTIDAS A UNA VELOCIDAD LIGERAMENTE MAYOR DE SU VALOR NOMINAL Y LA DIFERENCIA ES COMPENSADA AÑADIENDO BITS DE JUSTIFICACIÓN O DE RELLENO LO CUAL ES CONOCIDO COMO JUSTIFICACIÓN POSITIVA.

PARA QUE EN LA RECEPCIÓN ESOS BITS PUEDAN SER RECONOCIDOS Y ELIMINADOS PARA PODER RECONSTRUIR LA SEÑAL EN FORMA EXACTA, LOS BITS DE JUSTIFICACIÓN DEBEN TENER UNA POSICIÓN ESPECIFICA DENTRO DE LA TRAMA.

LA PRESENCIA O AUSENCIA DE BITS DE JUSTIFICACIÓN ES DEFINIDA MEDIANTE BITS DE CONTROL DE JUSTIFICACIÓN QUE NORMALMENTE SON 3. UNA SEÑAL 111 INDICA QUE EXISTEN BITS DE JUSTIFICACIÓN MIENTRAS QUE UNA SEÑAL 000 INDICA AUSENCIA.

EN LA RECEPCIÓN SE APLICA UN CRITERIO DE MAYORIA PARA DEFINIR SI EXISTE O NO BITS DE JUSTIFICACIÓN.

MULTIPLEXORES PDH
INTRODUCCIÓN

JUSTIFICACIÓN POSITIVA:

UNA MEMORIA BUFFER ES USADA ADEMAS DE UN COMPARADOR DE FASE PARA PROCESAR CADA SEÑAL. LOS DATOS DE ENTRADA SE ESCRIBEN A UNA VELOCIDAD Fe Y SON LEIDOS A UNA VELOCIDAD LIGERAMENTE MAYOR, LA MEMORIA TENDERÁ A VACIARSE PARA COMPENSAR LA DIFERENCIA EN TEMPORIZACIÓN EL DISPOSITIVO PERIODICAMENTE REALIZARÁ UNA OPERACIÓN DE JUSTIFICACIÓN QUE INVOLUCRA LA REPETICIÓN DE LA LECTURA DE UN BIT. ESTA OPERACIÓN ES REQUERIDA POR EL COMPARADOR DE FASE Y SE REALIZA EN UN TIEMPO ESPECIFICO DENTRO DE LA TRAMA MEDIANTE LA CANCELACIÓN DE UN INTERVALO DE TIEMPO CARACTERISTICO EN LA SEÑAL DE TEMPORIZACIÓN DE LECTURA.

LO ANTERIOR INCLUYE DISCONTINUIDADES DEBIDO A LA ESTRUCTURA DE TRAMA COMO SON LA INSERCIÓN DE LA PALABRA DE ALINEAMINETO DE TRAMA, LOS BITS DE SERVICIO, LOS BITS DE CONTROL DE JUSTIFICACIÓN Y LOS PROPIOS BITS DE JUSTIFICACIÓN.

MULTIPLEXORES PDH
INTRODUCCIÓN

VARIACIONES DE FASE (JITTER):

LAS VARIACIONES DE FASE SON UNA CARACTERÍSTICA DE LAS SEÑALES DIGITALES, SU PRINCIPAL ORIGEN ES DURANTE LA TRANSMISIÓN DE LAS SEÑALES SOBRE LOS SITEMAS DE LINEA.

LAS VARIACIONES DE FASE SE DEBEN AL FENOMENO DE TIEMPO DE ESPERA: DURANTE LA SOLICITUD DE JUSTIFICACIÓN EN LA TRANSMISIÓN PUEDE OCURRIR EN CUALQUIER MOMENTO SIN EMBARGO ESTA SE REALIZA SOLO DURANTE UN TIEMPO ESPECIFICO DURANTE LA TRAMA. EN LA RECEPCIÓN LOS CIRCUITOS DE AMARRE EN FASE QUE RECONSTITUYEN LA SEÑAL PUEDEN REDUCIR LAS VARIACIONES DE FASE PERO NO PUEDEN ELIMINAR LOS COMPONENTES DE MUY BAJA FRECUENCIA

MULTIPLEXORES PDH
INTRODUCCIÓN

SEÑAL DE INDICACIÓN DE ALARMA AIS :

CUANDO UNA FALLA OCURRE EN LA TRANSMISIÓN DE LA SEÑALES ESTAS SON SUBSTITUIDAS POR UNA SEÑAL DE INDICACIÓN DE ALARMA, EL CONTENIDO DE ESTA SEÑAL ES DE TODOS "1".

LA RECEPCIÓN DE UNA SEÑAL AIS INDICA QUE UN DEFECTO HA SIDO DETECTADO EN ALGUN PUNTO DENTRO DE LA TRASNMISIÓN DE LA SEÑAL POR LO QUE SE EVITA QUE EL DEFECTO SEA DETECTADO EN TODOS LOS PUNTOS DE TRANSITO DE LA SEÑAL.

MULTIPLEXORES PDH
INTRODUCCIÓN

MULTIPLEXACIÓN DIGITAL:

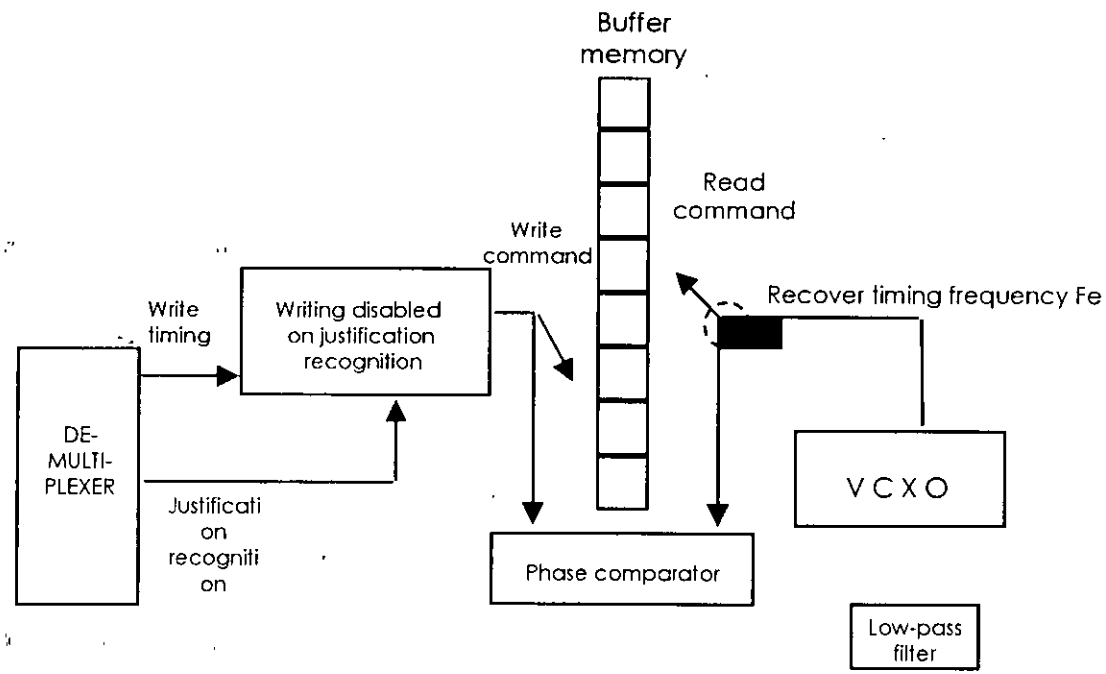
LA MULTIPLEXACIÓN SE REALIZA MEDIANTE LA INTERCALACIÓN CICLICA DE LOS CANALES BITS DE MARCA SE INTRODUCEN A INTERVALOS REGULARES PARA CONFERIR UNA ESTRUCTURA PERIÓDICA PERMITIENDO EN LA RECEPCIÓN LA IDENTIFICACIÓN DE LOS CANALES.

UNA SEÑAL MULTIPLEXADA CONTIENE:

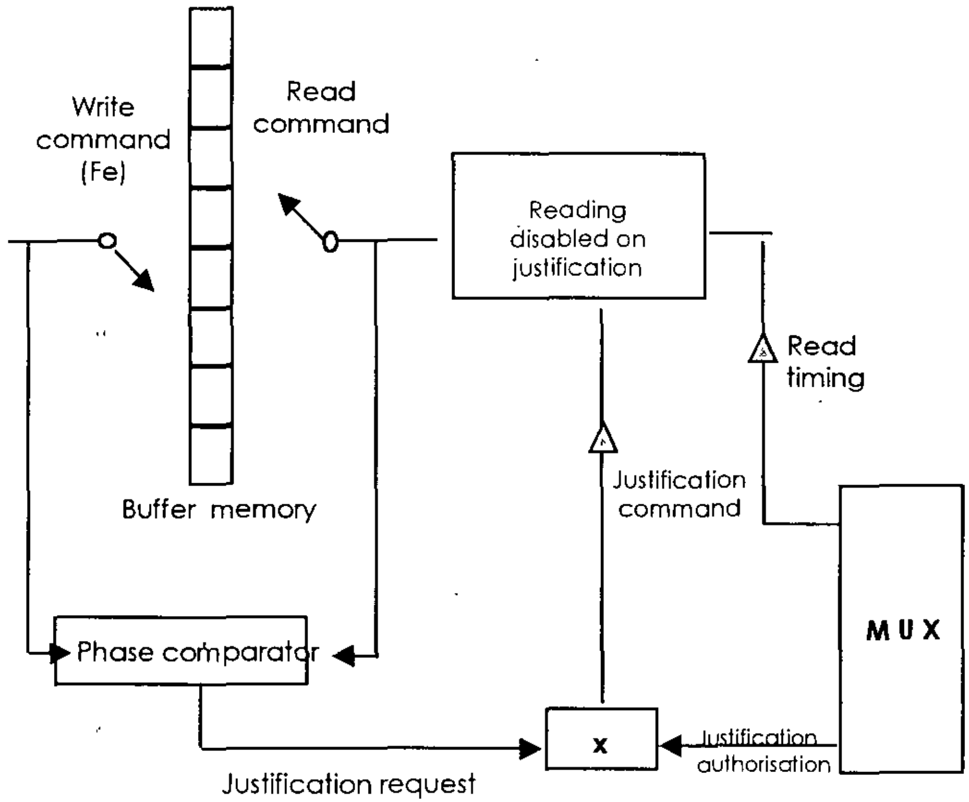
- LOS BITS DE MARCA QUE CORRESPONDEN A LA PALABRA DE ALINEAMINETO DE TRAMA.
- LOS BITS DE SERVICIO.
- LOS BITS DE DATOS DE LAS 4 SEÑALES DE ENTRADA INTERCALADOS
- LOS BITS DE JUSTIFICACIÓN Y LOS DE CONTROL DE JUSTIFICACIÓN.

UNA TRAMA DE UNA SEÑAL DE 8448Kbit/s ESTA FORMADA POR 848 BITS Y SE DIVIDE EN 4 SECTORES DE 212 BITS. EL PRIMER SECTOR CONTIENE LA PALABRA DE ALINEAMIENTO DE TRAMAY LOS BITS DE SERVICIO. LOS SECTORES S2, S3 Y S4 CONTIENEN LOS BITS DE CONTROL DE JUSTIFICACIÓN, ADEMAS EN EL SECTOR S4 SE UBICAN LOS BITS DE JUSTIFICACIÓN.

MULTIPLEXORES PDH



MULTIPLEXORES PDH



MULTIPLEXORES PDH
JERARQUIA MULTIPLEXORES 2MB/S

MULTIPLEX DE 2-8 (SEGUNDO ORDEN)

ESTE EQUIPO MULTIPLEXA 4 SEÑALES AFLUENTES PLESIÓCRONAS A 2048 Kbit/s EN UNA SEÑAL RESULTANTE A 8448 Kbit/s Y EN EL SENTIDO INVERSO DEMULTIPLEXA UNA SEÑAL DE 8448Kbit/s EN 4 SEÑALES AFLUENTES DE 2048Kbit/s.

MULTIPLEX DE 8-34 (TERCER ORDEN)

ESTE EQUIPO MULTIPLEXA 4 SEÑALES AFLUENTES PLESIÓCRONAS A 8448 Kbit/s EN UNA SEÑAL RESULTANTE A 8448 Kbit/s Y EN EL SENTIDO INVERSO DEMULTIPLEXA UNA SEÑAL DE 8448Kbit/s EN 4 SEÑALES AFLUENTES DE 34368Kbit/s.

MULTIPLEX DE 34-140 (CUARTO ORDEN)

ESTE EQUIPO MULTIPLEXA 4 SEÑALES AFLUENTES PLESIÓCRONAS A 34368 Kbit/s EN UNA SEÑAL RESULTANTE A 8448 Kbit/s Y EN EL SENTIDO INVERSO DEMULTIPLEXA UNA SEÑAL DE 8448Kbit/s EN 4 SEÑALES AFLUENTES DE 139264Kbit/s.

MULTIPLEXORES PDH
MULTIPLEXORES DE SEGUNDO ORDEN

MULTIPLEX DE 2-8

ESTE EQUIPO MULTIPLEXA 4 SEÑALES AFLUENTES PLESIÓCRONAS A 2048 Kbit/s EN UNA SEÑAL RESULTANTE A 8448 Kbit/s Y EN EL SENTIDO INVERSO DEMULTIPLEXA UNA SEÑAL DE 8448Kbit/s EN 4 SEÑALES AFLUENTES DE 2048Kbit/s.

CARACTERÍSTICAS GENERALES

CANTIDAD DE AFLUENTES	4
VELOCIDAD NOMINAL DE LOS AFLUENTES	2048 Kbit/S +/- 50 x 10^{-6}
CAUDAL NOMINAL DE LA SEÑAL MULTIPLEXADA	8448 Kbit/s. +/- 30 x 10^{-6}
CAPACIDAD EQUIVALENTE DE CANALES A 64Kbit/s	120.
ESTRUCTURA DE TRAMA	CONFORME A REC. G.742

INTERFAZ DIGITAL A 2048Kbit/s

IMPEDANCIA	120 OHMS par simétrico 75 Ohms cable Coaxial
CODIGO	HDB3
ATENUACION ADMISIBLE A LA ENTRADA A 1024Kbit/s	6dB

INTERFAZ DIGITAL A 8448Kbit/s

IMPEDANCIA	75 Ohms cable Coaxial
CODIGO	HDB3
ATENUACION ADMISIBLE A LA ENTRADA A 4224Kbit/s	6dB

FLUCTUACIONES DE FASE	CONFORME A REC. G.823
COMPORTAMIENTO DE ERRORES	CONFORME A REC G.821.

MULTIPLEXORES PDH
MULTIPLEXORES DE SEGUNDO ORDEN

ESTRUCTURA DE TRAMA 8448Kbit/s	**bits dentro de la trama**
SECTOR S1	Bits 1 a 212
Palabra de alineamiento de trama (1111010000)	1 a 10
Bits de reserva	11 y 12*
Bits de los flujos de entrada	13 a 212
SECTOR S2	213 a 424
Bits de control de Justificación	1 a 4
Bits de los flujos de entrada	5 a 212
SECTOR S3	425 a 636
Bits de control de Justificación	1 a 4
Bits de los flujos de entrada	5 a 212
SECTOR S4	637 a 848
Bits de control de Justificación	1 a 4
Bits de Justificación de los flujos de entrada	5 a 8
Bits de los flujos de entrada	9 a 212
Longitud de trama	848
cantidad de bits por flujo de entrada por trama	206
máxima velocidad de justificación por flujo de entrada (una justificación por trama)	9.962Kbit/s
Frecuencia de solicitud de justificación a máxima justificación	0.424

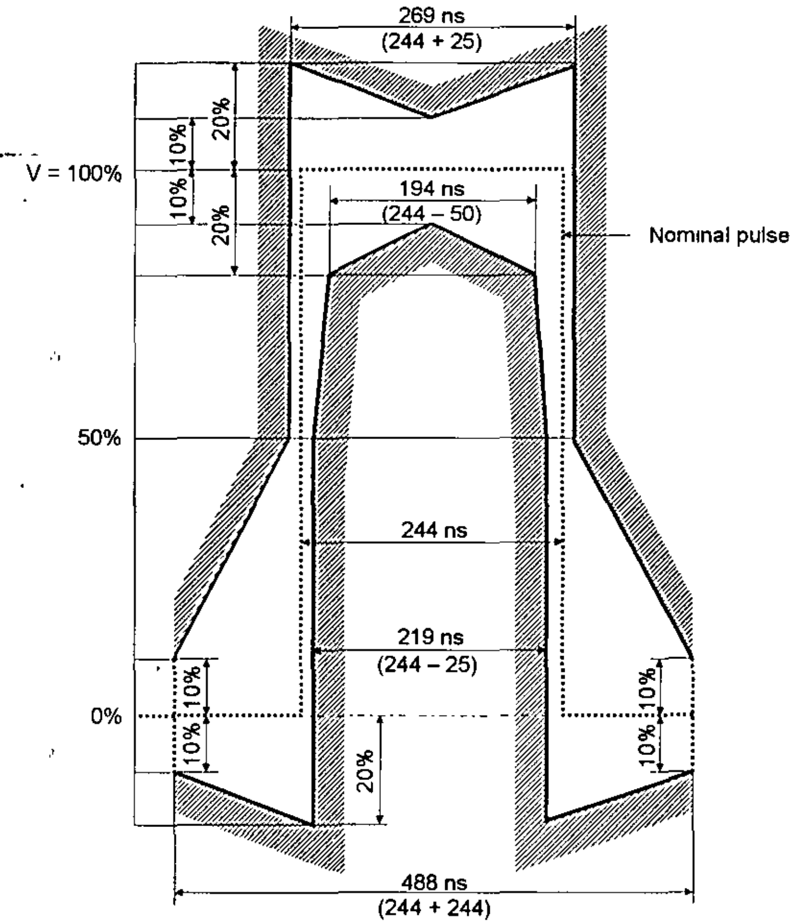
Bit No 11 se utiliza para transmitir una indicación de alarma al equipo multiplexor distante. Tiene un valor de 1 en caso de alarma.
Bit No 12 esta reservado para uso nacional y se fija a 1 en caso de cruce fronteras.

MULTIPLEXORES PDH
MULTIPLEXORES DE SEGUNDO ORDEN

TABLE 6/G.703

Pulse shape (nominally rectangular)	All marks of a valid signal must conform with the mask (se Figure 15/G.703) irrespective of the sign. The value V corresponds to the nominal peak value.	
Pair(s) in each direction	One coaxial pair (see § 6.4)	One symmetrical pair (see § 6.4)
Test load impedance	75 ohms resistive	120 ohms resistive
Nominal peak voltage of a mark (pulse)	2.37 V	3 V
Peak voltage of a space (no pulse)	0 ± 0.237 V	0 ± 0.3 V
Nominal pulse width	244 ns	
Ratio of the amplitudes of positive and negative pulses at the centre of the pulse interval	0.95 to 1.05	
Ratio of the widths of positive and negative pulses at the nominal half amplitude	0.95 to 1.05	
Maximum peak-to-peak jitter at an output port	Refer to § 2 of Recommendation G.823	

MULTIPLEXORES PDH
MULTIPLEXORES DE SEGUNDO ORDEN



Note – V corresponds to the nominal peak value.

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FIGURE 15/G.703

Mask of the pulse at the 2048 kbit/s interface

MULTIPLEXORES PDH
MULTIPLEXORES DE SEGUNDO ORDEN

TABLA 2/G.742 CONDICIONES DE FALLA Y ACCIONES CONSECUENTES

		Acciones consecuentes				
				Aplicación de AIS		
Parte del Equipo	Condición de Falla	Generación de idicación de alarma para mantenimiento inmediato	Generación de Indicación de alarma al multiplexor remoto	Para todas las tributarias	A la señal compuesta	En el intervalo de tiempo de la señal compuesta
Multiplexor Demultiplexor	Falla de Fuente de Alimentación	Si		Si de ser práctico	Si de ser práctico	
Multiplexor solamente	Pérdida de la señal de entrada de tributario	Si				Si
Demultiplexor solamente	Pérdida de señal de entrada a 8448Kbit/s	Si	Si	Si		
	Perdida de la señal de alineamiento	Si	Si	Si		
	Indicación de alarma recibida desde el equipo multiplexor remoto	Si	Si	Si		

MULTIPLEXORES PDH
MULTIPLEXOR TERCER ORDEN

MULTIPLEX DE 8-34

ESTE EQUIPO MULTIPLEXA 4 SEÑALES AFLUENTES PLESIÓCRONAS A 8448 Kbit/s EN UNA SEÑAL RESULTANTE A 8448 Kbit/s Y EN EL SENTIDO INVERSO DEMULTIPLEXA UNA SEÑAL DE 8448Kbit/s EN 4 SEÑALES AFLUENTES DE 34368Kbit/s.

CARACTERÍSTICAS GENERALES

CANTIDAD DE AFLUENTES	4
VELOCIDAD NOMINAL DE LOS AFLUENTES	8448 Kbit/s. +/- 30 x 10^{-6}
CAUDAL NOMINAL DE LA SEÑAL MULTIPLEXADA	34368 Kbit/s. +/- 20 x 10^{-6}
CAPACIDAD EQUIVALENTE DE CANALES A 64Kbit/s	480.
ESTRUCTURA DE TRAMA	CONFORME A REC. G.751

INTERFAZ DIGITAL A 8448Kbit/s	CONFORME A G.703
IMPEDANCIA	75 Ohms cable Coaxial
CODIGO	HDB3
ATENUACION ADMISIBLE A LA ENTRADA A 4224Kbit/s	6dB

INTERFAZ DIGITAL A 34368Kbit/s	CONFORME A G.703
IMPEDANCIA	75 Ohms cable Coaxial
CODIGO	HDB3
ATENUACION ADMISIBLE A LA ENTRADA A 17184Kbit/s	12dB

FLUCTUACIONES DE FASE	CONFORME A REC. G.823
COMPORTAMIENTO DE ERRORES	CONFORME A REC G.821.

MULTIPLEXORES PDH
MULTIPLEXOR TERCER ORDEN

ESTRUCTURA DE TRAMA 34368Kbit/s	bits dentro d la trama
[redacted]	Bits 1 a 384
Palabra de alineamiento de trama (1111010000)	1 a 10
Bits de reserva	11 y 12*
Bits de los flujos de entrada	13 a 384
SECTOR S2	385 a 768
Bits de control de Justificación	1 a 4
Bits de los flujos de entrada	5 a 384
SECTOR S3	769 a 1152
Bits de control de Justificación	1 a 4
Bits de los flujos de entrada	5 a 384
SECTOR S4	1153 a 1536
Bits de control de Justificación	1 a 4
Bits de Justificación de los flujos de entrada	5 a 8
Bits de los flujos de entrada	9 a 384
Longitud de trama	1536
cantidad de bits por flujo de entrada por trama	378
máxima velocidad de justificación por flujo de entrada (una justificación por trama)	22375Kbit/s
Frecuencia de solicitud de justificación a máxima justificación	0.436
Bit No 11 se utiliza para transmitir una indicación de alarma al equipo multiplexor distante. Tiene un valor de 1 en caso de alarma. Bit No 12 esta reservado para uso nacional y se fija a 1 en caso de cruce fronteras.	

MULTIPLEXORES PDH
MULTIPLEXOR TERCER ORDEN

TABLE 7/G.703

Pulse shape (nominally rectangular)	All marks of a valid signal must conform wit mask (Figure 16/G.703) irrespective of the si
Pair(s) in each direction	One coaxial pair (see § 7.4)
Test load impedance	75 ohms resistive
Nominal peak voltage of a mark (pulse)	2.37 V
Peak voltage of a space (no pulse)	0 V ± 0.237 V
Nominal pulse width	59 ns
Ratio of the amplitudes of positive and negative pulses at the centre of the pulse interval	0.95 to 1.05
Ratio of widths of positive and negative pulses at the nominal half amplitude	0.95 to 1.05
Maximum peak-to-peak jitter at an output port	Refer to § 2 of Recommendation G.823

MULTIPLEXORES PDH
MULTIPLEXOR TERCER ORDEN

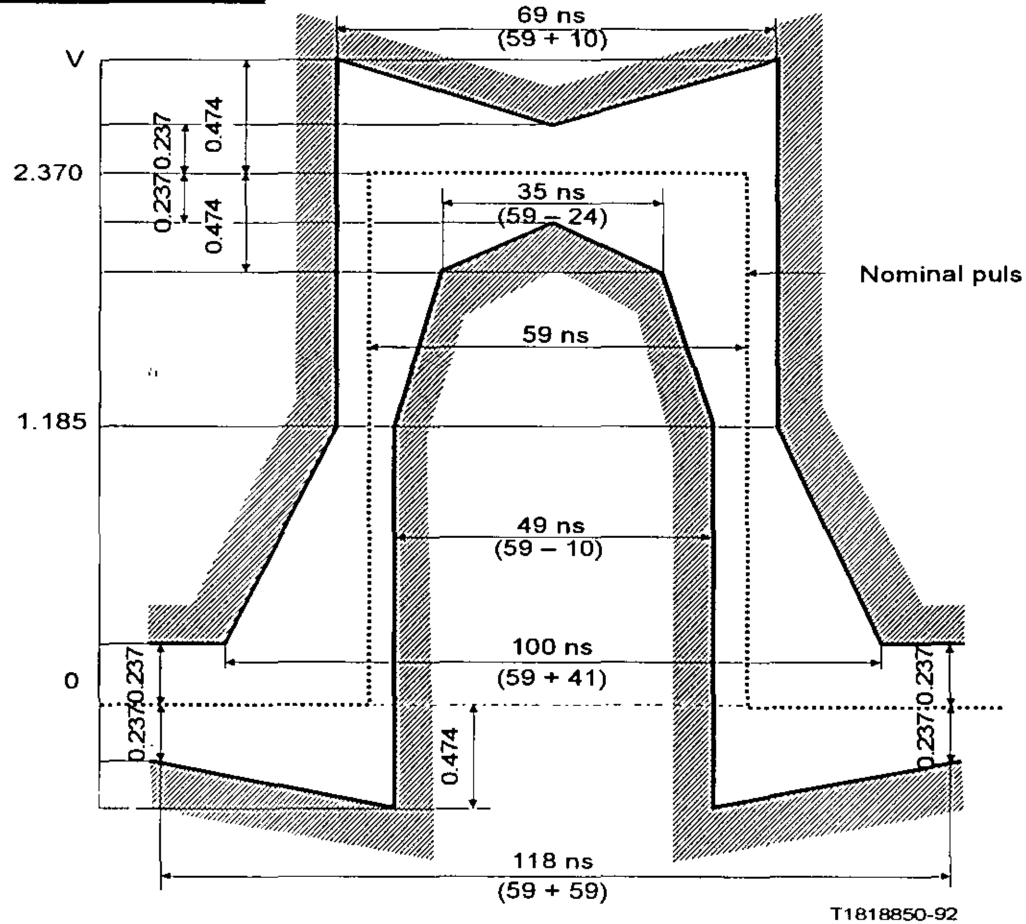


FIGURE 16/G.703

Pulse mask at the 8448-kbit/s interface

MULTIPLEXORES PDH
MULTIPLEXORTERCER ORDEN

TABLA 2/G.753 CONDICIONES DE FALLA Y ACCIONES CONSECUENTES

		Acciones consecuentes				
				Aplicación de AIS		
Parte del Equipo	Condición de Falla	Generación de idicación de alarma para mantenimiento inmediato	Generación de Indicación de alarma al multiplexor remoto	Para todas las tributarias	A la señal compuesta	En el intervalo de tiempo de la señal compuesta
Multiplexor Demultiplexor	Falla de Fuente de Alimentación	Si		Si de ser práctico	Si de ser práctico	
Multiplexor solamente	Pérdida de la señal de entrada de tributario	Si				Si
	Pérdida de señal de entrada a 34368K bit/s	Si	Si	Si		
	Perdida de la señal de alineamiento	Si	Si	Si		
Demultiplexor solamente	Indicación de alarma recibida desde el equipo multiplexor remoto	Si	Si	Si		

MULTIPLEXORES PDH
MULTIPLEXOR CUARTO ORDEN

MULTIPLEX DE 34-140

ESTE EQUIPO MULTIPLEXA 4 SEÑALES AFLUENTES PLESIÓCRONAS A 34368 Kbit/s EN UNA SEÑAL RESULTANTE A 8448 Kbit/s Y EN EL SENTIDO INVERSO DEMULTIPLEXA UNA SEÑAL DE 8448Kbit/s EN 4 SEÑALES AFLUENTES DE 139264Kbit/s.

CARACTERÍSTICAS GENERALES	
CANTIDAD DE AFLUENTES	4
VELOCIDAD NOMINAL DE LOS AFLUENTES	34368 Kbit/s. +/- 20 x 10^{-6}
CAUDAL NOMINAL DE LA SEÑAL MULTIPLEXADA	139264 Kbit/s. +/- 15 x 10^{-6}
CAPACIDAD EQUIVALENTE DE CANALES A 64Kbit/s	1920
ESTRUCTURA DE TRAMA	CONFORME A REC. G.751
INTERFAZ DIGITAL A 34368Kbit/s	CONFORME A G.703
IMPEDANCIA	75 Ohms cable Coaxial
CODIGO	HDB3
ATENUACION ADMISIBLE A LA ENTRADA A 17184Kbit/s	12dB
INTERFAZ DIGITAL A 139264Kbit/s	CONFORME A G.703
IMPEDANCIA	75 Ohms cable Coaxial
CODIGO	CMI
ATENUACION ADMISIBLE A LA ENTRADA A 69632Kbit/s	18dB
FLUCTUACIONES DE FASE	CONFORME A REC. G.823
COMPORTAMIENTO DE ERRORES	CONFORME A REC G.821.

MULTIPLEXORES PDH
MULTIPLEXOR CUARTO ORDEN

ESTRUCTURA DE TRAMA 139264Kbit/s	bits dentro de la trama
SECTOR S1	Bits 1 a 488
Palabra de alineamiento de trama (111110100000)	1 a 12
Bits de servicio cambia a 1 en presencia de una alarma	13
bits de serviciofijos a 1	14 16
Bits de los flujos de entrada	13 a 488
SECTOR S2	499 a 976
Bits de control de Justificación	1 a 4
Bits de los flujos de entrada	5 a 488
SECTOR S3	977 a 1464
Bits de control de Justificación	1 a 4
Bits de los flujos de entrada	5 a 488
SECTOR S4	1465 a 1952
Bits de control de Justificación	1 a 4
Bits de los flujos de entrada	5 a 488
SECTOR S5	1953 a 2440
Bits de control de Justificación	1 a 4
Bits de los flujos de entrada	5 a 488
SECTOR S6	2441 2928
Bits de control de Justificación	1 a 4
Bits de Justificación de los flujos de entrada	5 a 8
Bits de los flujos de entrada	9 a 488
Longitud de trama	2928

MULTIPLEXORES PDH
MULTIPLEXOR CUARTO ORDEN

TABLE 8/G.703

Pulse shape (nominally rectangular)	All marks of a valid signal must conform with the mask Figure 17/G.703), irrespective of the sign
Pair(s) in each direction	One coaxial pair (see § 8.4)
Test load impedance	75 ohms resistive
Nominal peak voltage of a mark (pulse)	1.0 V
Peak voltage of a space (no pulse)	0 V ± 0.1 V
Nominal pulse width	14.55 ns
Ratio of the amplitudes of positive and negative pulses at the center of a pulse interval	0.95 to 1.05
Ratio of the widths of positive and negative pulses at the nominal half amplitude	0.95 to 1.05
Maximum peak-to-peak jitter at an output port	Refer to § 2 of Recommendation G.823

MULTIPLEXORES PDH
MULTIPLEXOR CUARTO ORDEN

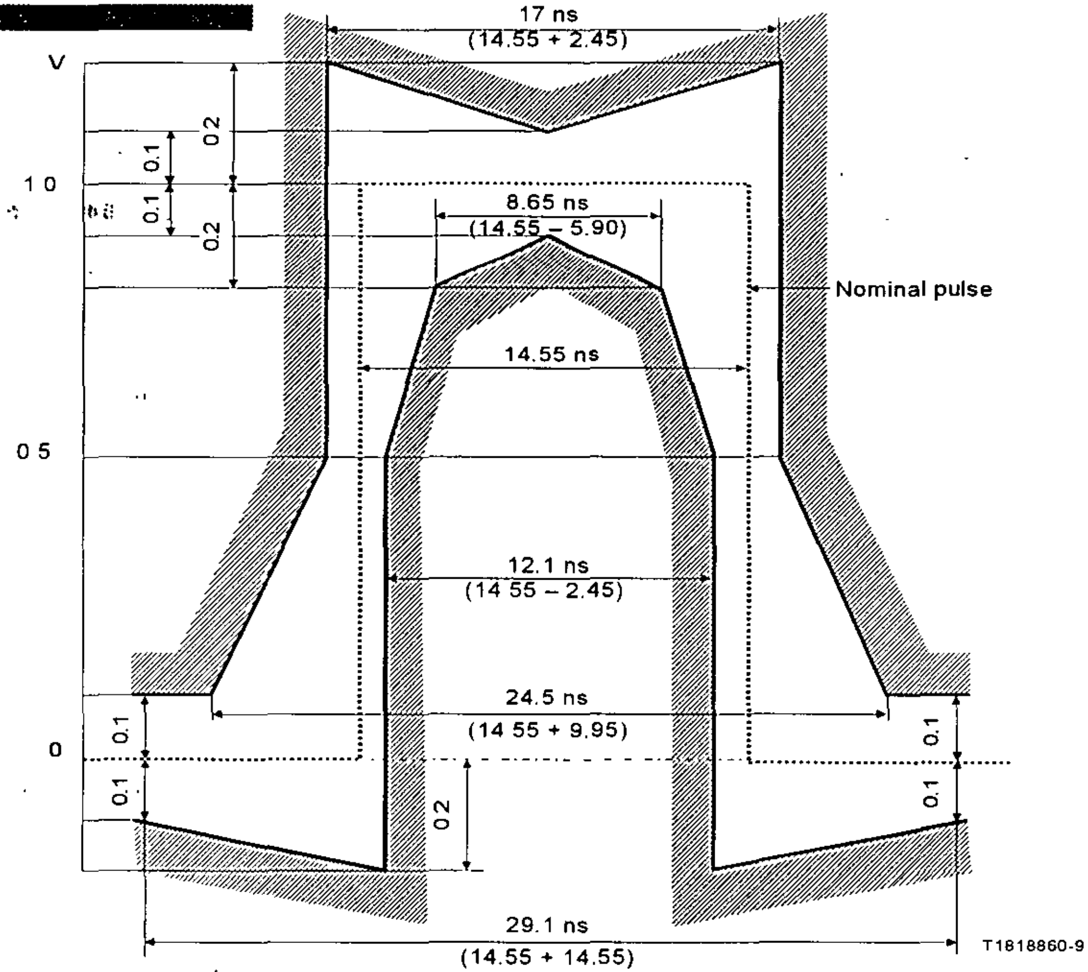


FIGURE 17/G.703

Pulse mask at the 34 368-kbit/s interface

MULTIPLEXORES PDH
MULTIPLEXOR CUARTO ORDEN

TABLA 3/G.751 CONDICIONES DE FALLA Y ACCIONES CONSECUENTES

		Acciones consecuentes				
				Aplicación de AIS		
Parte del Equipo	Condición de Falla	Generación de idicación de alarma para mantenimiento inmediato	Generación de Indicación de alarma en la señal de 139264Kbit/s al multiplexor remoto	En todas las tributarias a la salida del demultiplexor	A la señal compuesta de 139264 Kbit/s a la salida del multiplexor	En el intervalo de tiempo de la señal compuesta
Multiplexor Demultiplexor	Falla de Fuente de Alimentación	Si		Si de ser práctico	Si de ser práctico	
Multiplexor solamente	Pérdida de la señal de entrada de tributario	Si				Si
Demultiplexor solamente	Pérdida de señal de entrada a 139264Kbit/s	Si	Si	Si		
	Perdida de la señal de alineamiento en la señal 139264Kb/s	Si	Si	Si		
	Indicación de alarma recibida desde el equipo multiplexor remoto de 139264 kbit/s					

Para el multiplexor de 4 tributarias de 34368Kbit/s

MULTIPLEXORES PDH
MULTIPLEXOR CUARTO ORDEN

[redacted]

TABLA 4/G.751 CONDICIONES DE FALLA Y ACCIONES CONSECUENTES

		Acciones consecuentes						
					Aplicación de AIS			
Parte del Equipo	Condición de Falla	Generación de idicación de alarma para mantenimiento inmediato	Generación de Indicación de alarma en la señal de 139264Kbit/s al multiplexor remoto	Generación de Indicación de alarma en la señal de 34368kbit/s al multiplexor remoto	En las 16 tributarias de 8448bit/s a la salida del demultiplexor	En las 4 tributarias de 8448kbit/s a la salida del demultiplexor	A la señal compuesta de 139264 Kbit/s a la salida del multiplexor	En el intervalo de tiempo de la señal compuesta
Multiplexor Demultiplexor	Falla de Fuente de Alimentación	Si			Si de ser práctico		Si de ser práctico	
Multiplexor solamente	Pérdida de la señal de entrada de tributario	Si						Si
	Pérdida de señal de entrada a 139264Kbit/s	Si	Si		Si			
	Perdida de la señal de alineamiento en la señal 139264Kb/s	Si	Si		Si			
	Indicación de alarma recibida desde el equipo multiplexor remoto de 139264 kbit/s							
	Perdida de la señal de alineamiento en la señal 34368Kb/s	Si		Si		Si		
Demultiplexor solamente	Indicación de alarma recibida desde el equipo multiplexor remoto de 34368 kbit/s							

Para el multiplexor de 16tributarias de 8448Kbit/s

MULTIPLEXORES PDH
ACTUALIDAD Y PERPECTIVA

HOY EN DIA LOS MULTIPLEXORES PDH DE ALTO ORDEN SON AMPLIAMENTE UTILIZADOS EN LAS ADMINISTRACIONES TELEFÓNICAS A FIN DE CONTAR CON:

UN MEJOR APROVECHAMIENTO DE SU INFRAESTRUCTURA DE FIBRA OPTICA

PARA EL ESTABLECIMIENTO DE ENLACES VIA RADIO.

ULTIMAMENTE SE ESTAN UTILIZANDO PARA EL ESTABLECIMIENTO DE ENLACES DE BAJA CAPACIDAD COMO SON DE 2, 8 Y 34Mb/s Y EN MENOR MEDIDA EN ENLACES DE 140Mbit/s.

MULTIPLEXORES PDH
ACTUALIDAD Y PERPECTIVA

LA NECESIDAD DE CONTAR CON:

UNA MAYOR FLEXIBILIDAD EN EL MANEJO DE BANDA EN ENLACES QUE REQUIERE DE COLECCIÓN EN DIFERENTES PUNTOS DE UNA RED, ASI COMO DE SU DISTRIBUCIÓN.

- UNA ESTANDARIZACIÓN QUE PERMITA LA INTERCONECTIVIDAD ENTRE DIFERENTES FABRICANTES.
- EL CONTAR CON HERRAMIENTAS DE ADMINISTRACION CENTRALIZADA DE LA RED INCORPORADA A LOS EQUIPOS.
- DISPONER DE UN ANCHO DE BANDA SUPERIOR QUE PERMITA EL CRECIMIENTO DE LA DEMANDA DE TRÁFICO Y QUE SIMPLIFIQUE SU ACCESO
- HA DADO ORIGEN A LA APARICIÓN DE NUEVAS TECNOLOGIAS DE MULTIPLEXACIÓN EN LA TRANSMISIÓN QUE SON LA SONET (RED OPTICA SINCRONA PARA EL ESTANDAR AMERICANO) Y LA SDH (JERARQUIA DIGITAL SINCRONA PARA EL ESTANDAR EUROPEO).

FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

INTRODUCCIÓN A LA INTERFAZ V5

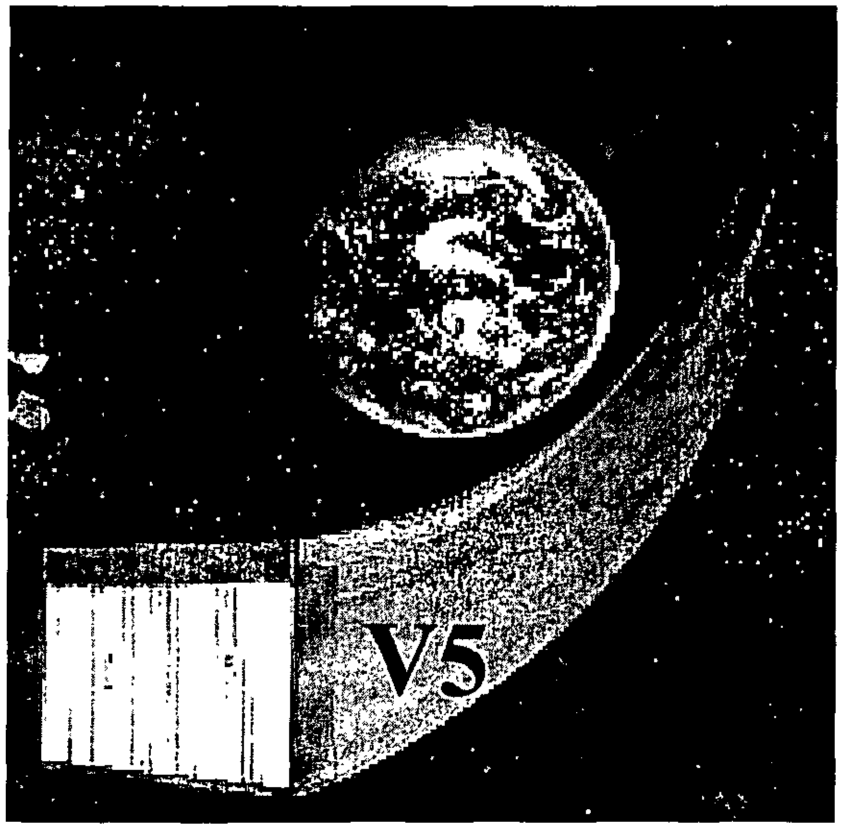
EXPOSITOR: ING. IGNACIO ORTIZ MENDOZA
PALACIO DE MINERÍA
JUNIO DE 1999

Introducción a la Interfaz V5

por

Switching Wireless Access Team
Lucent Technologies - Bell Laboratories
Ing. Ignacio Ortiz Mendoza

Interfaces V5

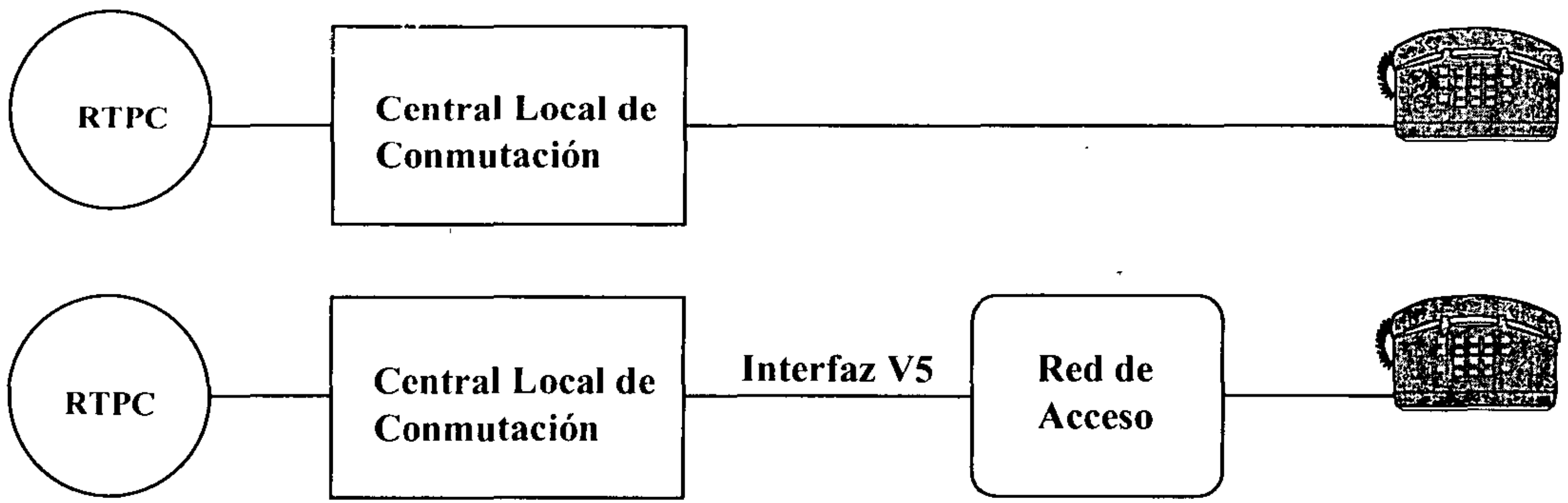


Introducción

- ¿Qué es la interfaz V5?
- Aspéctos Técnicos.
- Resumen y Conclusiones

¿Qué es la interfaz V5?

Diferencias en la implementación de V5



RTPC = PSTN

RTPC = Red Telefónica Pública Conmutada

PSTN = Public Switch Telephone Network

¿Qué es la interface V5?

- Serie de estándares publicados por ETSI/ITU que proveeń una **Interfaz Abierta** entre la Central Local (LE) y la Red de Acceso (AN)
  - El AN es dueño del puerto
  - El LE es dueño del servicio
- Protocolo basado en Mensajes Digitales

¿Qué es la interface V5? (cont.)

- o Ofrece Transparencia de Servicios (PSTN e ISDN).
- o Es independiente del Interfaz de Usuario.
- o Es independiente de la tecnología del AN.
- o Es independente de la tecnología o arquitectura del LE.

Responsabilidades del AN y LE

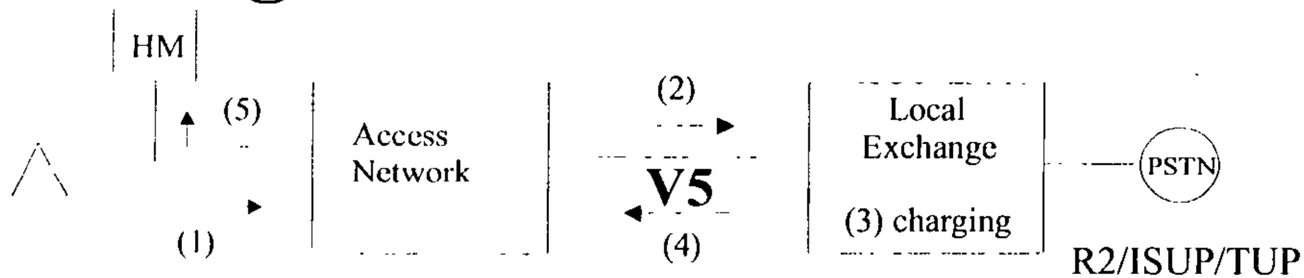
AN

- Terminación de Puntas.
- Puerto Físico del Suscriptor.
- Pruebas y Mantenimiento de la Línea.

LE

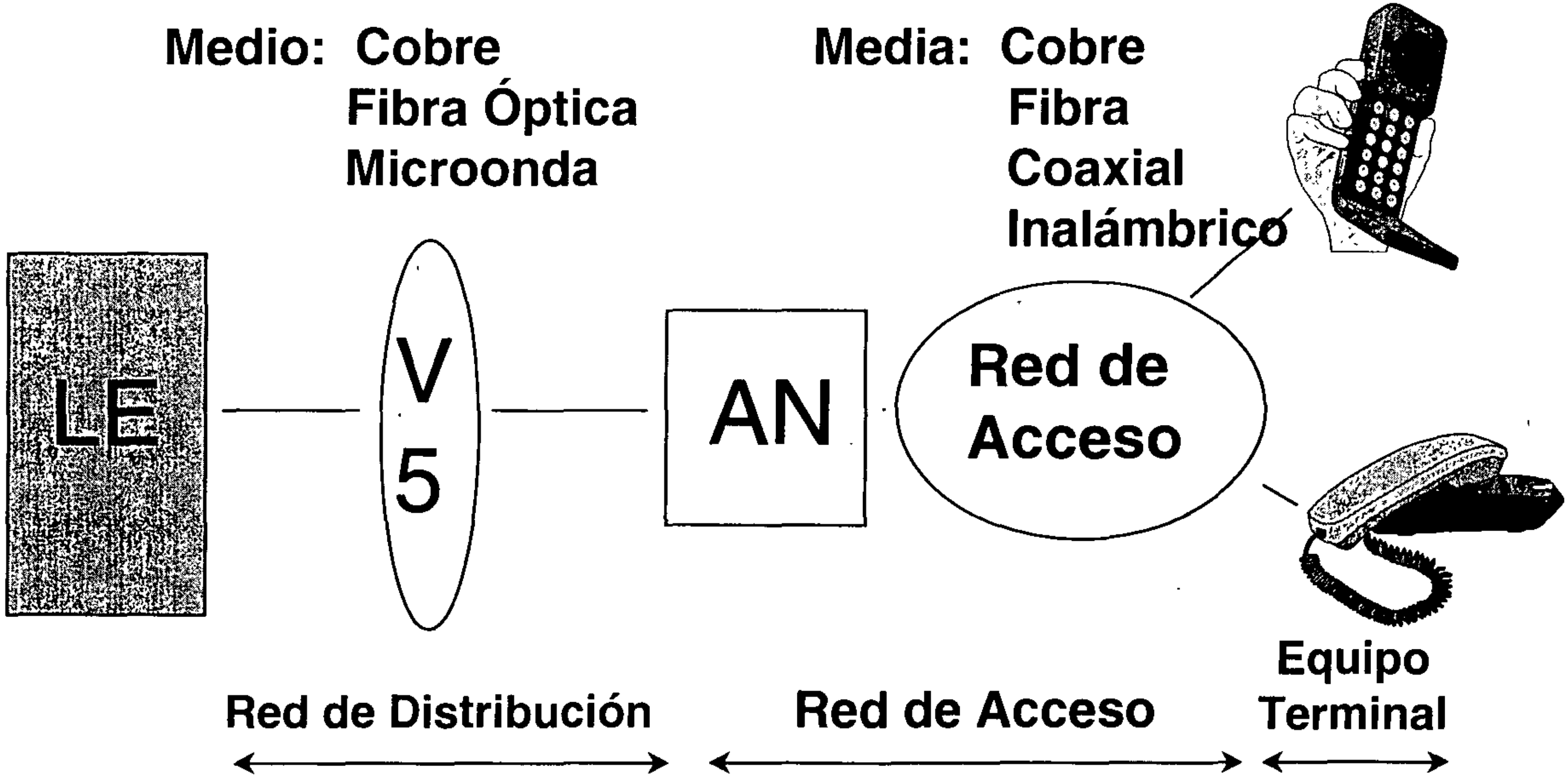
- Procesamiento y enrutamiento de la llamada.
- Conmutación
- Puerto Lógico del Subscriptor.
- Administrar Servicios Suplementarios.
- Tonos y Mensajes
- Tarificación y Facturación
- Señalización entre centrales.

Ejemplo de Registrador de Llamadas



1. Llamada originada a través de una línea PSTN.
2. El AN detecta el flujo de corriente, forma un mensaje digital V5 indicando la petición de originación hacia el LE.
3. El LE determina la razón de tasación.
4. El LE transmite la información al AN por medio de mensajes de V5.
5. El AN es responsable de incrementar el contador de pulsos (utilizando la polaridad correcta).

Tipos de Acceso



Estándares V5

Estándares:

- V5.1 (ITU G.964 and ETS 300 324-1)
- V5.2 (ITU G.965 and ETS 300 347-1)

Servicios Soportados:

- Servicios Suplementarios para PSTN (POTS).
- Acceso Básico ISDN (BRA)
- Acceso Primario ISDN (PRA)
- Conexiones ISDN Semi-Permanentes.
- Conexiones ISDN Permanentes.

Comparación entre V5.1 y V5.2

	V5.1	V5.2
Relación de Concentración	1:1	~8:1[a]
Canales B / E1	~28-30[b]	~200[a]
Máximo número de E1s	1	16
Servicio PSTN (POTS)	✓	✓
ISDN BRA	✓	✓
ISDN PRA		✓

[a] 1% bloquéo, 0.1 Erlang/Sub, y 4 E1s/Interface V5.2.

[b] Depende del número de canales de comunicación proveídos.

- o 1-16 E1s por interfaz V5.2 y 1 E1 por interfaz V5.1.
- o Cada E1 transporta 32 canales.
- o TS0 es utilizado para sincronización.
- o De 1 a 3 ranuras de tiempo o canales son reservados para el control canales. (TS16, TS31, TS15)
- o Los 28 a 30 canales libres son llamados "Canales Portadores (Bearer)" y se utilizan para la transmisión de voz y datos.

Características de V5.1

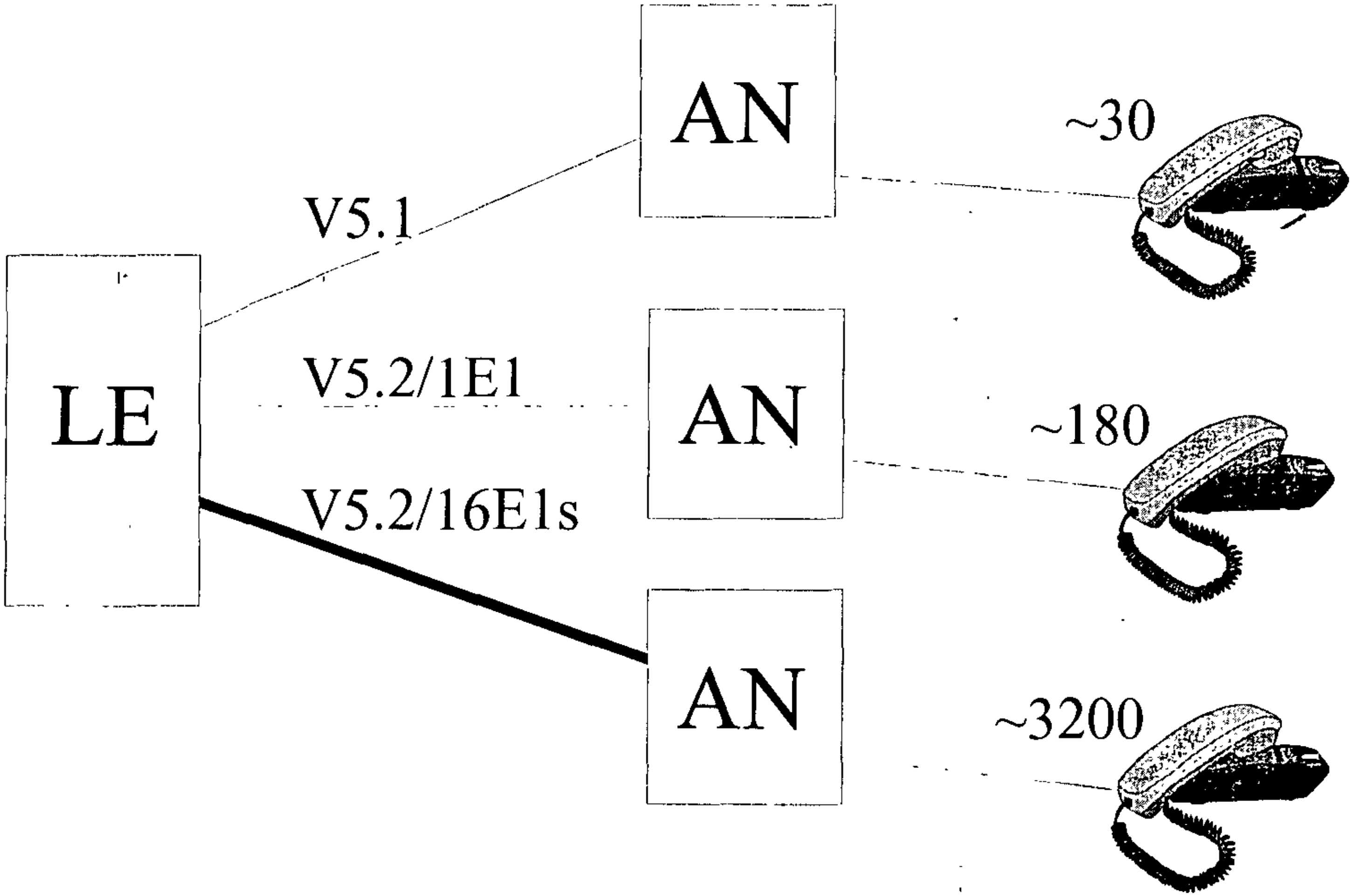
- o No existe concentración. Un solo enlace E1 por interfaz.
- o Canales Portadores (Bearer): PCM a 64 kBits/s y canales B para servicio ISDN Básico.
- o Canales de Comunicación: Señalización POTS, Señalización de Canal-D (ISDN) y Protocolos de Control.
- o Conexiones Analógicas y líneas ISDN Semi-Permanentes
- o Líneas ISDN Permanentes

Características de V5.2

Soporta características de V5.1 junto con las siguientes características:

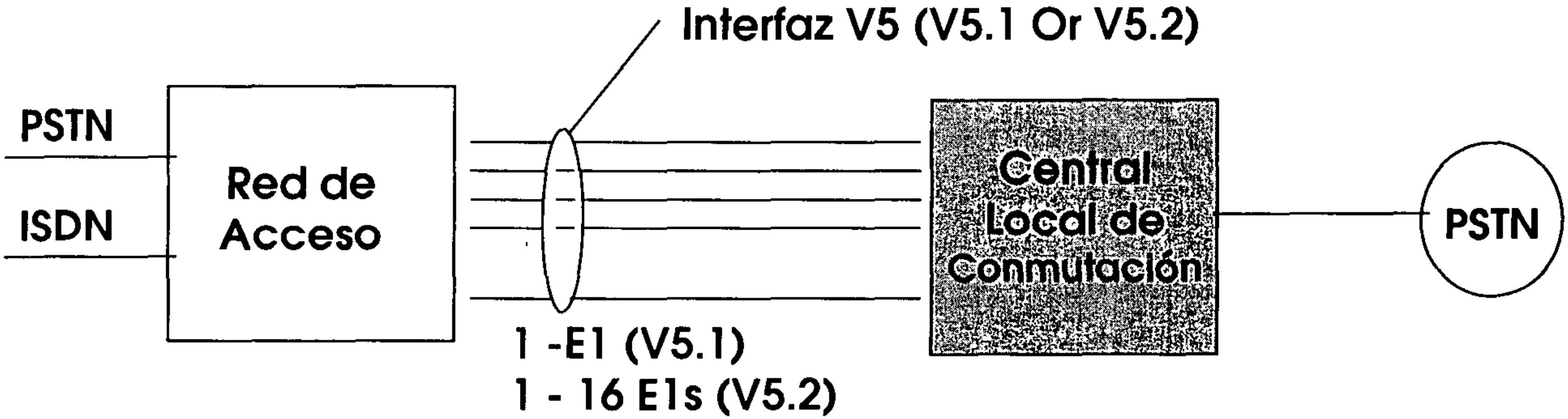
- o Concentración, Multiples enlaces E1
  - — Hasta 16 enlaces E1s por Interface
- o Concentración de Canales Portadores (Bearer Channels)
- o Protocolo de Control de Enlace
- o Protocolo de Protección

Comparasión de Interfaces V5



ASPECTOS TÉCNICOS DE LA INTERFAZ V5

Diseño de la Interfaz V5



- Mensages estructurados en 3 capas
- Definición de dos tipos de canales: Portadores (Bearer) y de Comunicación.
- Transporte de dos tipos de información: Canales Portadores y (Bearer) y de Canales de Control

Diseño de la interfaz V5 (cont.)

Canales Portadores (Bearer)

- 28 - 30 Canales disponibles (dependinedo del número de canales de comunicación)
- Proveé la trayectoria de voz para PSTN e ISDN
- Transportación de tonos tales como DTMF, modem y fax (pero no los tonos / pulsos de marcación)

Diseño de la interfaz V5 (cont.)

Canales de Comunicación (C-channels)

1 - 3 canales por enlace

Trayectos de comunicación (C-paths) son los tipos de información sobre un canal de comunicación (C-channel):

- enlace de datos (datalink) transportador de la señalización PSTN para todo los suscriptores PSTN.
- señalización de canal-D ISDN para uno o mas suscriptores ISDN.
- transmisión de paquetes por el canal-D ISDN para uno o mas suscriptores ISDN.
- enlace de datos tranportador del Protocolo de Control (Común y Puertos).
- enlace de datos transportador del Protocolo de Control de Enlace.
- enlace de datos transportador del Protocolo de Conexión de Canal Portador (BCC).
- cada uno de los dos enlaces de datos transportadores del Protocolo de Protección.
- canales-C lógicos transportadores de uno o mas trayectos de comunicación.
- canales-C físicos (ranuras de tiempo de 64 kbit/s) transportando canales-C lógicos.

Diseño de la interfaz V5 (cont.)

Estructura de mensages en 3 capas

Capa 1 (Capa Física)

- Terminación de los canales portadores (bearer) y Canales-C asociados a un enlace E1.

- Características eléctricas y alineación de tramas para una transmisión libre de errores, ITU-T G.703, G.704, G.706

V5 Interface Design (cont.)

Capa 2 (Capa de Enlace de Datos)

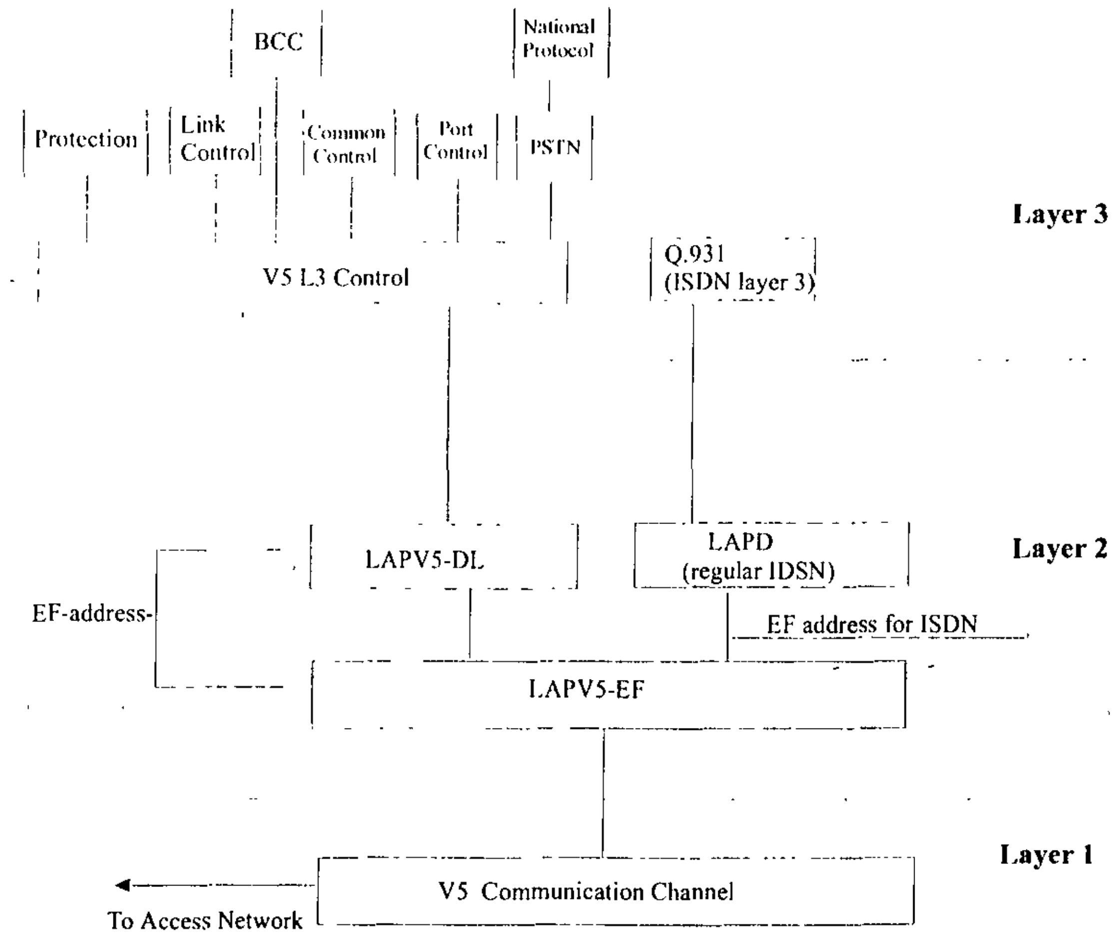
- o Interpretación y clasificación de información enviad o recivida por los canales-C utilizando el subprotocolo LAPV5 el cual:
  - o Asigna la Función de Envolvente EF-addrs (identifica el tipo de trama V5 siendo procesado) y dirección V5DL-addrs (identifica el enlace de datos que transportará la información V5 al protocolo apropiado de la capa 3)
  - o 0-8175 - Señalización ISDN
    - o (V5DL-addrs no asignados - información procesada en el protocolo LAPD en la capa 2)
  - o 8176 - Señalización PSTN
  - o 8177 - Protocolo de Control Común y Puertos de Usuario
  - o 8178 - Protocolo BCC
  - o 8179 - Protocolo de Protección
  - o 8180 - Protocolo de Control de Enlaces

Diseño de la interfaz V5 (cont.)

Capa 3 (Capa de Red)

- Procesa información specífica a un protocolo de información
  - Basado en la asignación de enlace de datos de la capa 2, el subprotocolo V5L3 envía la información específica al protocolo V5 apropiado
  - Ejemplo: información BCC es enviado por el protocolo BCC protocol
  - La información de señalización LAPD ISDN recibida de la capa 2 es procesada por el estandar Q.931

Software Layers for V5.2



Diseño de la interfaz V5 (cont.)

Los Protocolos de control - administran y controlan los puertos PSTN e ISDN de la interfaz V5 como un sola entidad.

Se encuentra subdividida en 5 distintos protocolos

- **Control del Puerto de Usuario (PSTN y ISDN)**
- **Común Control**
- **Conexión de Canales Portadores (Bearer channels)**
- **Control de Enlace**
- **Protección**

Diseño de la interfaz V5 (cont.)

Protocolo de Control de Puerto: administra los diferentes estados del puerto PSTN e ISDN:

-- procedimiento de bloqueo y desbloqueo para puertos de usuario PSTN,

-- bloqueo y desbloqueo así como procedimientos de activación y desactivación para un usuario ISDN.

- Es soportado a través de dos mensages, los cuales se envian y reconocen información de información de control para el puerto de usuario.

Diseño de la interfaz V5 (cont.)

Protocolo Común de Control - utilizado para el intercambio de información referente a la interfaz V5:

-- El LE podrá solicitar al AN o vice-versa la identificación de la interfaz V5.

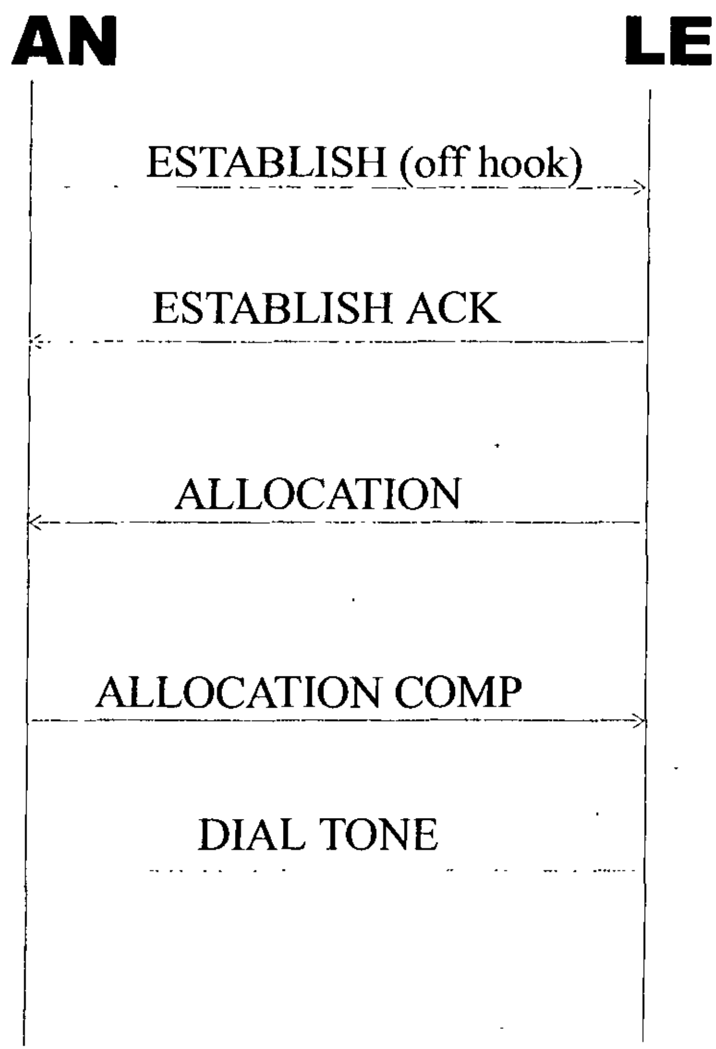
o Se basa en dos mensajes los cuales comunican y confirman funciones de común control (no específicas a un puerto).

Diseño de la interfaz V5 (cont.)

Conexión de Canal Portador (BCC) - provee el algoritmo que permite al LE ordenar al AN el establecimiento y liberación de canales portadores

- Permite concentración en la Interfaz V5.2 por medio de la seleción y liberación de ranuras de tiempo en base a cada llamada.
- Los procedimiento de BCC son aplicados al inicio de cada llamada PSTN o ISDN conmutada.
- Se sustenta por medio de once mensages los cuales comunican y confirman la información del Canal Portador.

Mensages BCC



Diseño de la interfaz V5 (cont.)

Protocolo de Control de Enlace - administra cada enlace en la interfaz V5.2. Este protocolo contola:

- o Estado e Identificación del Enlace. (ya sea por demanda o por temporización) asegura que el LE y AN tengan una visión consistente del mismo enlace.
- o Bloqueo y desbloqueo del enlace el cual puede ser originado por el LE o el AN.
- o Procedimiento de desbloqueo del enlace completamente simétrico.
- o Verificación de la continuidad del enlace.
- o Coordinación de funciones de control.
- o Dos mensages sustentan la comunicación y reconocimiento información del control del enlace.

Diseño de la interfaz V5 (cont.)

Protocolo de Protección - permite la conmutación de canales-C lógicos hacia diferentes canales-C en un evento de fallo.

- La ranura de tiempo 16 de un enlace primario designado, contiene el Protocolo de Protección junto con otros protocolos de canal-C.
- La ranura de tiempo 16 en un enlace secundario como respaldo del enlace primario transporta el Protocolo de Protección.
- Ocho mensages son utilizados para la comunicación y reconocimiento de los procedimientos "switch-over" de enlaces.

Diseño de la interfaz V5 (cont.)

Administrador del Sistema - responsable de vigilar y mantener una interfaz V5 estable utilizando 6 componenetes:

- *Administrador de Interfaz V5 Sincronizado* - responsible de la operación y control de la interfaz V5.
  - -- Por medio del Protocolo Común de Control, mantiene comunicación con su entidad par en el AN.
- *Administrador PSTN del Estado de Puertos* - responsable de los cambios de estados sincronizados de los suscriptores de puertos PSTN.
  - -- Por medio del Protocolo Común de Puertos, mantiene comunicación con su entidad par en el AN.
- *Administrador ISDN del Estado de Puertos* - responsible de los cambios de estados sincronizados de los suscriptores de puertos ISDN.
  - -- Por medio del Protocolo Común de Puertos, mantiene comunicación con su entidad par en el AN.

Diseño de la interfaz V5 (cont.)

Administrador del Sistema (cont.)

- *Administrador de Recursos* - controla la solicitud de canales portadores y negocia el uso de canales con el AN.
  - -- Por medio del protocolo BCC, mantiene comunicación con su entidad par en el AN
- *Administrador de Enlace* - utilizado en la identificación e indicación de estado del enlace
  - -- Por medio del Protocolo de Control de Enlaces, mantiene comunicación con su entidad par en el AN.
- *Administrador de Protección* - responsable del "switch over"
  - -- Por medio del Protocolo de Protección, mantiene comunicación con su entidad par en el AN

PSTN PROTOCOL IMPLEMENTATION

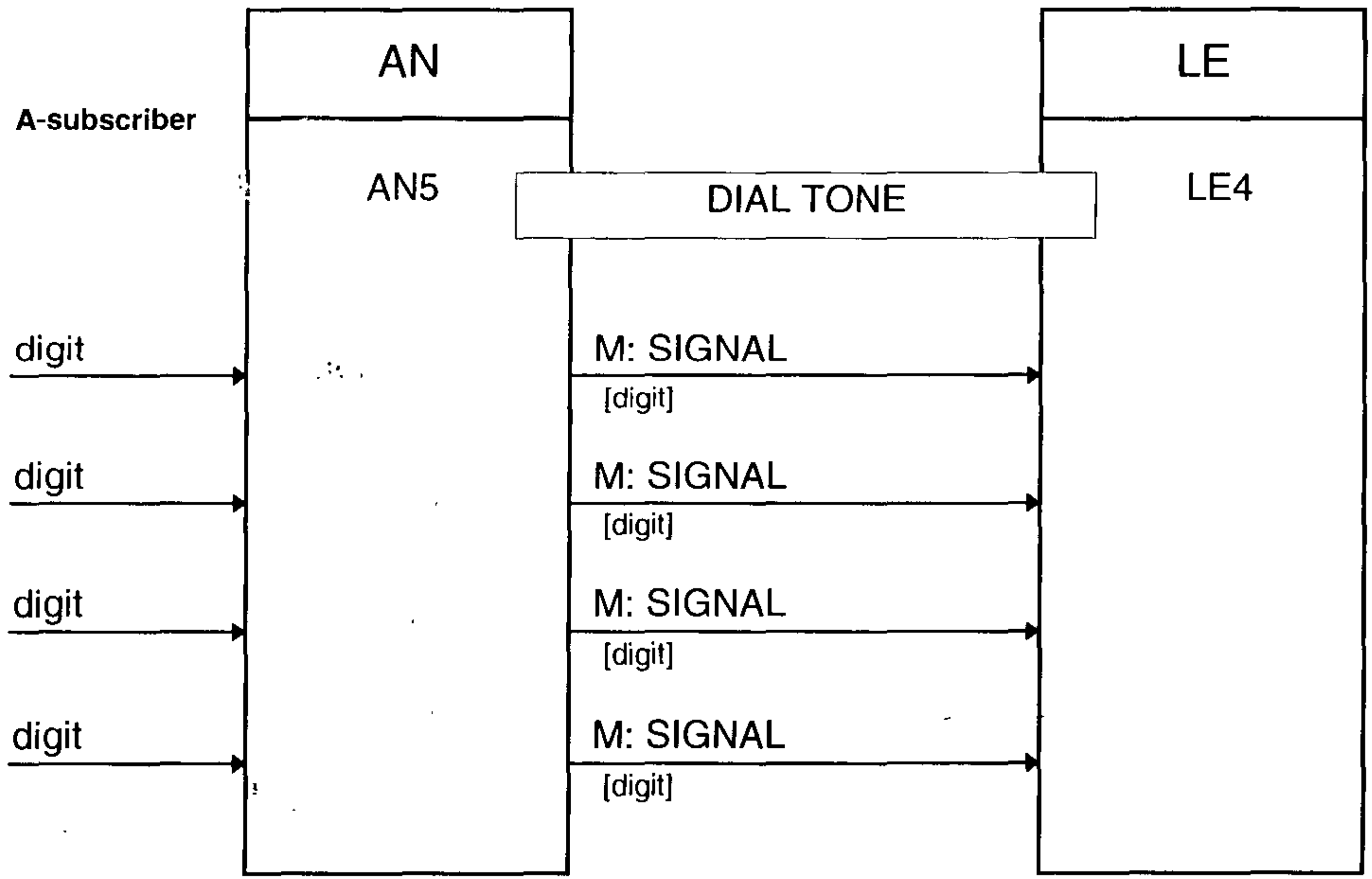
PROTOCOL PSTN - comunica información referenta al estado de las líneas analógicas para el acceso de un suscriptor utilizando mensages específicos V5.

- o Utilizado en conjunto con el Sistema Nacional de Señalización en el LE.
- o Sustentado por nueve mensages los cuales comunican y confirman información referente a un suscriptor:

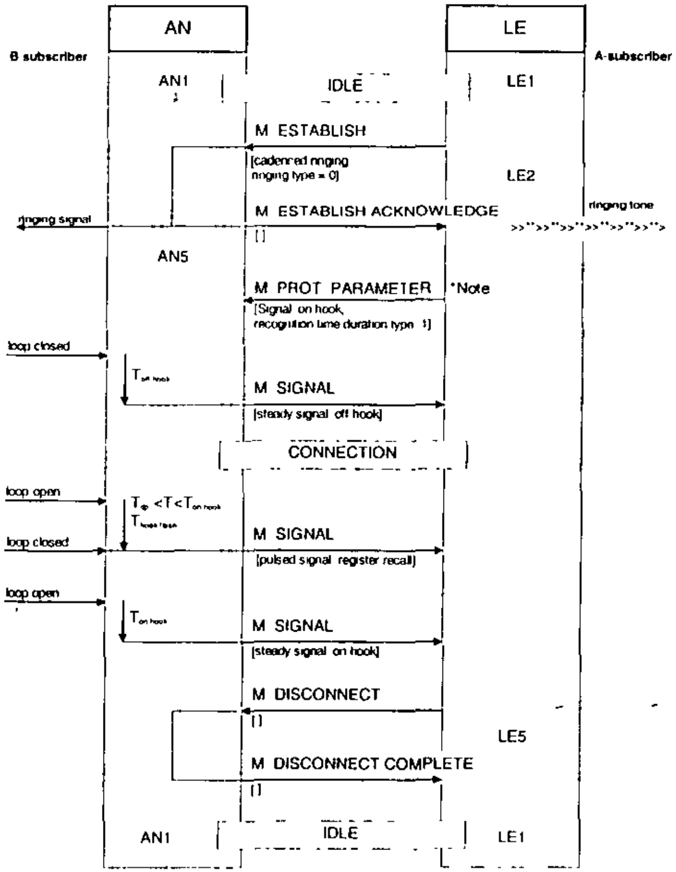
  - ESTABLISH: indica solicitud de originación o terminación de trayecto.
  - ESTABLISH ACK: reconocimiento y ejecucución de solicitud de acción.
  - SIGNAL: comunicar al LE la condición de las líneas PSTN, u ordena al AN: establecer condiciones de línea específicas. También podrá comunicar el pulso de un dígito o un pulso de tasasión solicitado.

PSTN PROTOCOL (cont.)

- **SIGNAL ACK**: reconocimiento de los mensages "SIGNAL" y "PROTOCOL PARAMETER".
- **DISCONNECT**: indica liberación de trayectoria de llamada
- **DISCONNECT COMPLETE**: reconocimiento del mensage "DISCONNECT".
- **STATUS ENQUIRY**: solicitud del estado físico de la entidad del protocolo PSTN en el LE.
- **STATUS**: indica el estado físico de la entidad del protocolo PSTN en el LE.
- **PROTOCOL PARAMETER**: utilizado por LE para cambiar el parámetro de protocolo en el AN (ejemplo. Momento en que es posible recibir el pulso de "flash hook").



Escenario de Marcación por Pulsos



*Note. LE Activates FLASH for AN Customer Based on Line Assignment in LE

Suscriptor AN activa un servicio utilizando "Flash (R)"

CONCLUSIONES

- Lucent Technologies se compromete a cumplir los estándares ETSI/ITU y definidos por nación con respecto evolucionan.
- Lucent Technologies es también proveedor de varios productos de Redes de Acceso, de los cuales varios implementan interfaces V5.1 y/o V5.2.

CONCLUSIONES

- **Lucent Technologies apoya el desarrollo de estudiantes y profesionistas en su carrera.**

- **Lucent Technologies aprecia y agradece a la UNAM su invitación al curso**

"Redes Digitales Actualidad y Perspectiva"

FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

TARIFAS Y EQUIPO RDSI

EXPOSITOR: ING. MA. DEL CARMEN ANGELICA MORENO ARGÜELLO
PALACIO DE MINERÍA
JUNIO DE 1999

ALCATEL

Ejemplos de Tarifas de Conexión y Uso de Líneas RDSI

PAIS	COSTO DE CONEXION			RENTA FIJA MENSUAL		
	ACCESO BASICO	ACCESO PRIMARIO	LINEA ANALOGICA	ACCESO BASICO	ACCESO PRIMARIO	LINEA ANALOGICA (2)
ESPAÑA (*)	299	8,933	168	50	812	20
AUSTRIA	126	1,042	95	32	316	26
BELGICA	87	6,470	71	43	454	34
DINAMARCA	275	2,625	214	28	276	28
FINLANDIA	353	3,993	804	28	206	16
FRANCIA	132	823	49	29	441	18
ALEMANIA (*)	59	118	58	28	318	31
GRECIA	207	2,568	166	41	621	16
IRLANDA	670	6,699	192	56	558	32
ITALIA	259	778	130	32	447	30
HOLANDA	228	4,043	117	25	253	28
NORUEGA	142	3,519	95	19	287	30
PORTUGAL	194	930	92	26	250	24
SUECIA	359	7,180	117	40	598	36
REINO UNIDO	700	4,626	154	55	541	36
USA	150			75	900	

(*) EL TRAFICO ES TRATADO IGUAL QUE EN LAS LLAMADAS NORMALES (PULSOS)
TARIFAS EXPRESADAS EN USD

Introduction ISDN Solutions

Welcome to DIVA—the industry's most complete line of ISDN connectivity solutions for individual users and PC-based servers. Whether you need ISDN for telecommuting, Internet access, mobile computing, or point-to-point file transfers, Eicon offers a DIVA product perfectly tailored to your application

Eicon's decade-long commitment to ISDN has helped DIVA become a leading brand, globally The DIVA T/A PC Card is the best selling PCMCIA ISDN modem worldwide DIVA solutions for desktop computers are the market leaders in eight European countries, where ISDN has experienced its most explosive growth. The entire DIVA line offers more international certifications than any competitor, for the greatest degree of compatibility wherever used Every DIVA product is backed by an extensive five-year warranty, a reflection of our commitment to the highest quality standards, and includes our complete ISDN Software Suite

The DIVA line includes offerings in two basic product groups:

- **Client Solutions:** DIVA offers a broad selection of ISDN modems for PCs and notebooks. Our external, internal and PC Card products offer convenient options for digital-only connections, digital plus analog over ISDN, and combination ISDN and 56 kbps modem functionality
- **Server Solutions:** Eicon's ISDN server cards offer high speed Basic Rate Interface (BRI) and Primary Rate Interface (PRI) ISDN connectivity to multiple, simultaneous remote users and LAN-to-LAN communications. Fully scalable, these ISDN server cards allow you to select the capacity you need today and add capacity economically as your needs grow

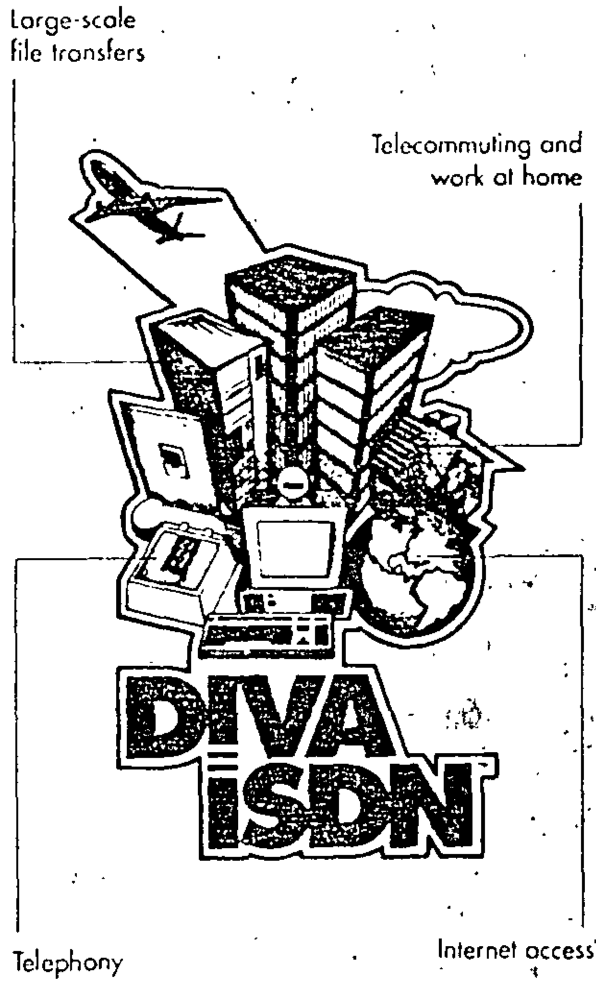
Applications

HOW IS ISDN USED?

Individuals and businesses turn to ISDN because plain old telephone service is inadequate for today's data communications needs. With much faster transmission, multiple communication channels per line, and the ability to connect worldwide through existing infrastructure, ISDN helps millions of people increase productivity and accomplish things they simply couldn't do before. Very importantly, it does this today.

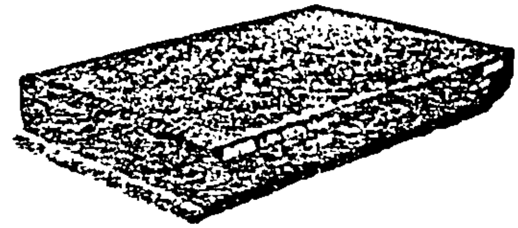
- **Internet access:** Anyone who accesses the Web with a modem understands the limitations of analog communications—and ISDN is the solution. ISDN makes Web graphics appear almost immediately, and reduces file upload and download times by over 80%.
- **Telecommuting and work at home:** Full-time telecommuters and people who finish work at home in the evening benefit tremendously from ISDN. E-mail, database access and file transfers improve so dramatically that it's like being on the LAN, and only one line per home is needed for data, telephone and fax transmission.
- **Large-scale file transfers:** Industries that deal in computer-based imagery, including publishers, hospitals, and police departments, find ISDN the ideal solution for point-to-point file transfers. With its dramatic increase in speed, ISDN makes even multi-megabyte files available to recipients within minutes, not hours.
- **Telephony:** ISDN is often thought of as a data solution, but it's important to remember that it also provides all-digital telephone connections. When plugged into an ISDN modem, a regular telephone has immediate access to advanced calling features including three-way calling, caller ID and call transfer. Even overseas voice calls sound like you're talking to your neighbor.

DIVA makes all these applications possible by connecting individual computers and network servers to high-speed ISDN lines. The following pages provide details on DIVA products; for additional information on ISDN in general, please visit **www.isdnzone.com**



DIVA ISDN

DIVA LAN ISDN Modem



The EASY and complete multi-user communications system for home and small offices. The DIVA LAN ISDN Modem integrates data, fax and phone over one ISDN line and installs in a few minutes.

FEATURES

- **Multi-user DIVA LAN ISDN Modem** allows many users to share one Internet account over a single ISDN line - surf all at the same time!
- **Surf the net AND use the phone** or fax at the same time, eliminating a separate phone line.
- **Share resources** such as printers and scanners between local computers via the built-in 4-port Ethernet 10BaseT hub, expandable up to 50 users by linking additional hubs
- **Plug-and-Play** installation, including color-coded cables and ports, Windows Set-up Wizard and web-based Configuration Wizard. The DIVA LAN ISDN Modem detects ISDN line information
- **Automatic dial-up** - Launch the Internet browser or e-mail application and you're on-line. No more dial-up networking!
- **High-speed Internet & corporate WAN access** - up to 512 kbps with built-in data compression.
- **Supports a virtual private network** over the Internet, eliminating long-distance WAN charges. Transparent PPTP support available on Windows® 95, Windows® 98 and Windows NT® platforms only.
- **Maximize speed and reduce ISDN costs** - Brings up the second ISDN B-channel only when needed. Advanced tariff management features use filters to further reduce ISDN costs.
- **Instant ISP connection without expensive usage fees*** - Uses the inexpensive ISDN D-channel to keep the connection with the Internet Service provider active. Automatically engages expensive B-channels only when needed
- **Total network security** - The DIVA LAN ISDN Modem hides network addresses from the outside world. Passwords prevent tampering
- **Easy software upgrades** - Visit our web site to download the latest software.

TECHNICAL SPECIFICATIONS

- 4-port 10BaseT Ethernet hub, expandable up to 50 users
- Console port for administrative support
- DHCP server and DNS Relay
- Full ISDN BRI compatibility
- On-board STAC data compression & transparent PPTP support
- Interoperable with all major switch types
- Features BOD, NAT, PPP, MLPPP, TCP/IP, AO/DI*, BACP/BAP, Dial-on-Demand, AutoSPID
- Operating system independent
- D-channel Protocols, NI-1, AT&T 5ESS, EDSS-1 or EuroISP (Europe + Australia OnRamp), TPH1962 (Australia Microl

Ships with Accessories

- Ethernet LAN cable
- Console cable
- ISDN cable
- Power supply
- Telephone adapter & terminating resistor where required

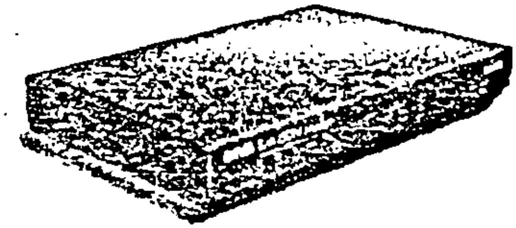
ORDERING INFORMATION

Product Name	Interface	Product Code
DIVA LAN ISDN Modem - Canada	U	310-271
DIVA LAN ISDN Modem - Latin America	S/T	310-270
DIVA LAN ISDN Modem - North America	S/T	310-259
DIVA LAN ISDN Modem - North America	U	310-26[illegible]

* where supported by the Internet or network service provider

This product is available worldwide. For a complete list of ordering codes visit our web site www.eicon.com

DIVA T/A ISDN Modem



Eicon Technology's DIVA T/A ISDN Modem provides corporate telecommuters and SOHO users with fast and secure access to corporate intranets or the Internet, over an ISDN line.

Easy to install and configure, the DIVA T/A ISDN Modem delivers throughput to 512 kbps for blazingly fast e-mail, file transfers, database searches and Web surfing. On-board data compression and support for Multilink PPP and BACP combine to save you time and money.

In addition, the DIVA T/A ISDN Modem eliminates the need for separate analog lines. Two POTS ports let you connect analog telephone and fax machines. Talk on the phone and transmit data simultaneously, on the ne ISDN line.

FEATURES

- **Fast Access:** 128 kbps data connections to the Internet and corporate networks with up to 512 kbps throughput.
- **Plug and Play ISDN:** SPID Wizard with groundbreaking Auto-SPID support to finally make ISDN truly plug and play
- **AO/DI:** Always On/Dynamic ISDN lets you stay continuously connected without the expense of dedicated B-channels
- **Phone Jacks:** Two POTS ports for standard phone and fax machines, and complete support for Caller ID, 3-way conference calls, and more
- **Simultaneous Data, Phone, and Fax:** While downloading a file from the Internet or the office, you can send a fax or place a phone call without having to interrupt the file transfer
- **Bandwidth on Demand:** Use only the bandwidth you need, when you need it, so you save money
- **Security:** Support of authentication already in place on your system, so you don't need to retype your passwords
- **Data Over Voice:** Establishes data calls as voice calls to save money
- **Usable Anywhere in the World:** With support for multinational ISDN protocols, on S/T and U (integrated NT1) interface, and compatibility with more switches than any competitive solution, DIVA T/A ISDN Modem can be deployed anywhere.
- **Multilink PPP:** Combines two ISDN 64 kbps B-channels into a single high-speed pipe for 128 kbps connectivity.
- **Bandwidth Allocation Control Protocol:** Support for BACP means you can establish a Multilink PPP connection by dialing just one number. Additional numbers are automatically dialed for you when required.

OPERATING SYSTEM COMPATIBILITY

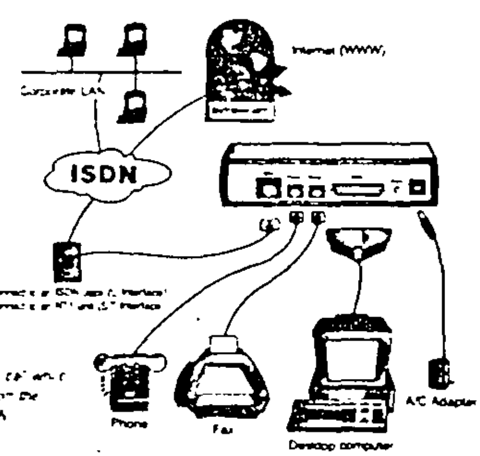
- Windows 95 & Windows 98
- Windows NT 3.51 and 4.0
- Windows 3.X
- NetWare Client
- OS/2
- DOS

SHIPS WITH ACCESSORIES

- Serial cable
- RJ45 cable (ISDN)
- A/C adapter

ORDERING INFORMATION

Product Name	Bus	Interface	Product Code
DIVA T/A ISDN Modem	External	S/T	310 233
DIVA T/A ISDN Modem	External	U	310-191



Send a fax or place a call while downloading a file from the web or corporate LAN

DIVA ISDN

DIVA T/A PC Card

The DIVA T/A PC Card is ideal for notebook computer users looking to take advantage of ISDN communications for reliable, high-speed access to the networked enterprise.

Easy to install and configure, the DIVA T/A PC Card is a direct replacement for your standard PC Card analog modem. Simply remove it and plug in the DIVA T/A PC Card to experience blazingly fast remote LAN access, e-mail, file transfers, database searches and Internet connectivity.

By supporting Multilink PPP, BACP and standards-based data compression, the DIVA T/A PC Card saves you time and money.

FEATURES

- **Fast Access:** 128 kbps data connections to the Internet and corporate networks with up to 512 kbps throughput.
- **Plug and Play ISDN:** SPID Wizard with groundbreaking Auto-SPID support to finally make ISDN truly plug and play.
- **AO/DI:** Always On/Dynamic ISDN lets you stay continuously connected without the expense of dedicated B-channels.
- **Bandwidth on Demand:** Use only the bandwidth you need, when you need it, so you save money.
- **Up to 3 Configurations Profiles:** Simplifies communications when you are on the road by storing up to three ISDN configuration sets, each with a primary and alternate location.
- **Security:** Support of authentication already in place on your system, so you don't need to retype your passwords.
- **Data Over Voice:** Establishes data calls as voice calls to save money.
- **Usable Anywhere in the World:** With support for multinational ISDN protocols, an S/T and U (integrated NT1) interface, and compatibility with more switches than any competitive solution, DIVA T/A PC Card can be deployed anywhere.
- **Multilink PPP:** Combines two ISDN 64 kbps B-channels into a single high-speed pipe for 128 kbps connectivity.
- **Bandwidth Allocation Control Protocol:** Support for BACP means you can establish a Multilink PPP connection by dialing just one number. Additional numbers are automatically dialed for you when required.

OPERATING SYSTEM COMPATIBILITY

- Windows 95
- Windows 98
- Windows NT 3.51 and 4.0
- Windows 3.X
- NetWare Client
- OS/2
- DOS

SHIPS WITH ACCESSORIES

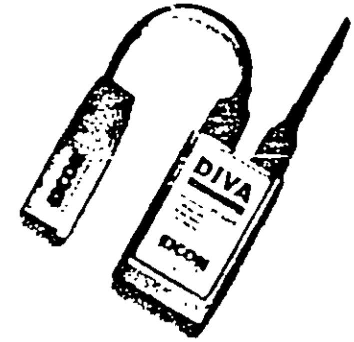
- ISDN cable (S/T or integrated NT1)

ORDERING INFORMATION

Product Name	Bus	Interface	Product Code
DIVA T/A PC Card	PCMCIA	S/T	310-165
DIVA T/A PC Card	PCMCIA	U	310-167

DIVA ISDN

DIVA Mobile V.90 PC Card



The DIVA Mobile V.90 PC Card is a multi-function, easily installed PC Card that is both an ISDN card and V.90 fax/data modem on a single card. It allows connections of up to 128.8 Kbps over ISDN and 56 Kbps over regular phone lines.

The DIVA Mobile V.90 PC Card is ideal for the international traveler who needs instant internet access, remote LAN access, or standard fax/modem connections, using ISDN or regular phone lines from around the world.

FEATURES

- **Digital and Analog over ISDN:** From around the world, connect to your corporate LAN or the Internet, and at the same time transfer data to an analog modem and send/receive faxes - one ISDN line is all you need!
- **V.90 Fax/Data Modem:** For areas around the world where ISDN is not available, the DIVA Mobile V.90 PC Card does double-duty as an analog modem, supporting data communications at up to 56 kbps and fax at up to 14.4 kbps.
- **Fully Plug and Play:** Automatic detection under Windows 95 and Windows 98 means there is less hassle when configuring the DIVA Mobile V.90 PC Card for ISDN and modem use.
- **Express Setup:** Software installation is fast and easy, even for non-technical users.
- **Flash Memory:** Upgrade to a new firmware version without replacing hardware.
- **Ready for Use Worldwide:** Stay connected wherever your travels take you. The DIVA Mobile V.90 PC Card is compatible with more ISDN switches and international analog phone systems than any competitive solution.
- **Multilink PPP:** Combines two ISDN 64Kbps B-Channels into a single high-speed pipe for 128 kbps connectivity.
- **Trace Diagnostic Tool:** Simplifies the management of technical issues by providing status information in the event of ISDN B-Channel or D-Channel problems.

OPERATING SYSTEM COMPATIBILITY

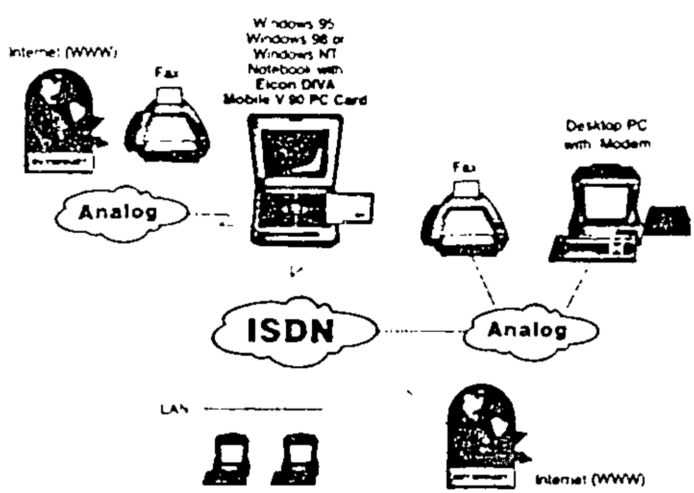
- Windows 95
- Windows 98
- Windows NT 4.0

SHIPS WITH ACCESSORIES

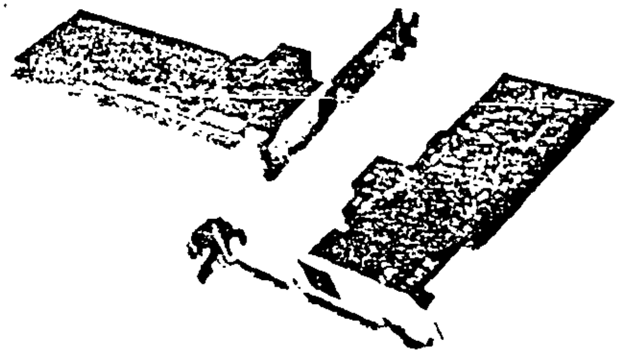
- Modem cable
- ISDN S/T cable
- U dongle
- RJ45 cable
- Teleadapter Pack

ORDERING INFORMATION

Product Name	Interface	Product Code
DIVA Mobile V.90 PC Card - NA	S/T	305 448 01
DIVA Mobile V.90 PC Card - NA	w/NT1	305 449-01



DIVA ISDN



DIVA

Eicon Technology's DIVA is a low-cost, high-performance ISDN interface card for applications that need to move large amounts of data, and do it fast. DIVA supercharges Internet access and minimizes file transfer time by combining two B-channels into a single 128 kbps pipe

Choose DIVA for ISDN connections to corporate resources or the Internet, or to transfer extremely large files for imaging applications in the medical, insurance, law enforcement or pre-press environments

DIVA is fully Plug and Play compliant, and available in both ISA and PCI bus formats With more international certifications than any competing solution, DIVA can reliably be put into service worldwide. And with its low price, DIVA provides substantial savings for enterprises that need to deploy ISDN on a broad basis.

DIVA is one of many quality offerings in the Eicon Technology DIVA family of ISDN products

FEATURES

- **Multilink PPP:** Combines two ISDN 64 kbps B-channels into a single high-speed pipe for 128 kbps connectivity
- **Express Setup:** Software installation is fast and easy, even for non-technical users Perform three simple steps and the DIVA for Windows 95 software is installed (Express Setup option)
- **Trace Diagnostic Tool:** Simplifies the management of technical issues by providing status information in the event of ISDN B-channel or D-channel problems.
- **Usable Anywhere in the World:** With support for multinational ISDN protocols, on S/T and U (integrated NT1) interface, and compatibility with more switches than any competitive solution, DIVA can be deployed anywhere
- **Rate Adaptation:** Rate adaptation permits the use of 56 kbps lines when local switches do not support 64 kbps

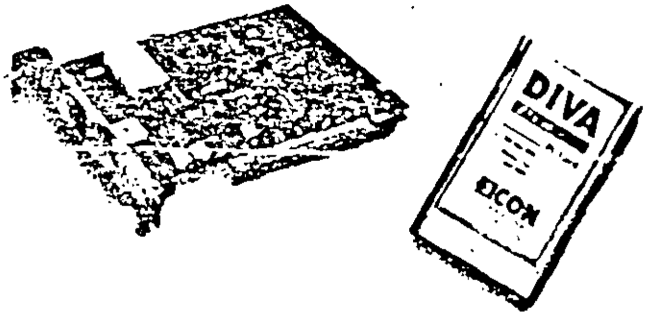
OPERATING SYSTEM COMPATIBILITY

- Windows 95
- Windows 98
- Windows NT 3 51 and 4 0
- Windows for Workgroups 3.11
- DOS

ORDERING INFORMATION

Product Name	Interface	Product Cod
DIVA ISA ISDN	S/T	305-1[illegible]
DIVA ISA ISDN with integrated NT1	U	305-188
DIVA PCI ISDN	S/T	305 189
DIVA PCI ISDN with integrated NT1	U	305 190

DIVA Pro



The DIVA Pro, Eicon's high-end offering for ISDN client PCs, provides everything you expect from an ISDN interface card—plus unique advantages designed for the most demanding communications applications. Featuring Digital Signal Processor (DSP) technology, the DIVA Pro keeps you in touch at top speed with other ISDN devices, and also communicates with analog modems and fax machines over the ISDN line.

The DIVA Pro is specially designed for telecommuters, office workers and remote LAN access users who need the speed of ISDN, but also the ability to communicate th analog resources.

DIVA Pro is available in ISA, PCI and PC Card formats.

FEATURES

- **Digital and Analog over ISDN:** Connect to your corporate LAN or the Internet, and at the same time transfer data to an analog modem and send/receive faxes—one ISDN line is all you need.
- **DSP Programmed for ISDN:** For improved performance, all fax and modem connectivity is handled by the DIVA Pro.
- **Software Upgradable:** Upgrade to new software directly from Exon's Web site. As standards change, DIVA Pro adapts to them.
- **Express Setup:** Software installation is fast and easy, even for non-technical users. Perform three simple steps and the DIVA for Windows 95 software is installed (Express Setup option).
- **Multilink PPP:** Combines two ISDN 64 kbps B-channels into a single high-speed pipe for 128 kbps connectivity.
- **Usable Anywhere in the World:** With support for multinational ISDN protocols, on S/T and U (integrated NT1) interface, and compatibility with more switches than any competitive solution, DIVA Pro can be deployed anywhere.
- **Rate Adaptation:** Rate adaptation permits the use of 56 kbps lines when local switches do not support 64 kbps.
- **Trace Diagnostic Tool:** Simplifies the management of technical issues by providing status information in the event of ISDN B-channel or D-channel problems.

OPERATING SYSTEM COMPATIBILITY

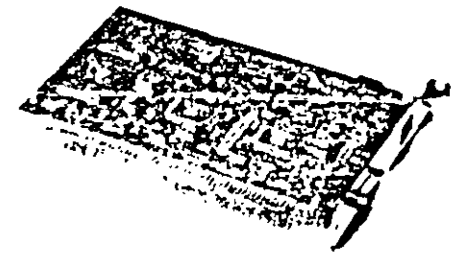
- Windows 95
- Windows 98
- Windows NT 3.51 and 4.0
- Windows for Workgroups 3.11
- DOS

ORDERING INFORMATION

Product Name	Interface	Product Code
DIVA Pro ISA ISDN	S/T	305-191
DIVA Pro ISA ISDN w/ integrated NT1	U	305-192
DIVA Pro PCI ISDN	S/T	305-193
DIVA Pro PCI ISDN w/ integrated NT1	U	305-194
DIVA Pro PC Card ISDN	S/T	305-195
DIVA Pro PC Card ISDN w/ integrated NT1	U	305-196

DIVA ISDN

DIVA Server BRI



Eicon Technology's DIVA Server BRI is a high-performance, active ISDN card that provides both digital and analog connectivity over the ISDN line. It is the perfect choice for corporate servers that need to support high-speed Internet access, remote LAN connectivity, or any digital/analog application with a small user base.

With DIVA Server BRI, a single server can support a full range of call types over one ISDN line. On-board Digital Signal Processors (DSPs) enable simultaneous communications with ISDN devices, analog modems, GSM-compatible mobile phones, and fax machines. And, as requirements grow, additional DIVA Server BRI cards can be added.

FEATURES

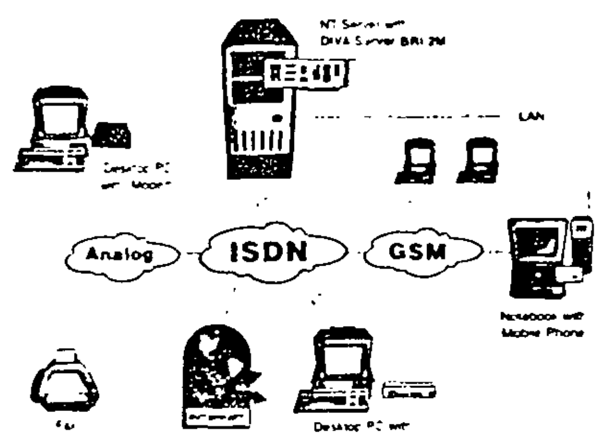
- **Digital and Analog over ISDN:** Connect to the Internet, and at the same time communicate with analog modems, receive calls from GSM-compatible mobile phones, and act as a fax server—one ISDN line is all you need.
- **DSP Programmed for ISDN:** For improved performance, all fax and modem connectivity is handled by the DIVA Server BRI.
- **Software Upgradable:** Upgrade to new software directly from Eicon's Web site. As standards change, DIVA Server BRI adapts to them.
- **Scalable Server Solution:** You can add additional server cards any time you need the extra capacity. DIVA Server BRI cards can be grouped within a server to support multiple BRI lines.
- **Supports Microsoft® BackOffice® Small Business Server:** Lets small businesses take advantage of the flexibility and speed of ISDN for their remote access, Internet access, LAN-to-LAN access and Fax Server needs.
- **Plug and Play:** Automatic detection eliminates the need to manually configure your server.
- **Multilink PPP:** Combines two ISDN 64 kbps B-channels into a single high-speed pipe for 128 kbps connectivity.
- **Rate Adaptation:** Rate adaptation permits the use of 56 kbps lines when local switches do not support 64 kbps.
- **Ready for Use Worldwide:** DIVA Server BRI is compatible with more ISDN switches than any competitive solution, and provides full support for multinational ISDN protocols.

OPERATING SYSTEM COMPATIBILITY

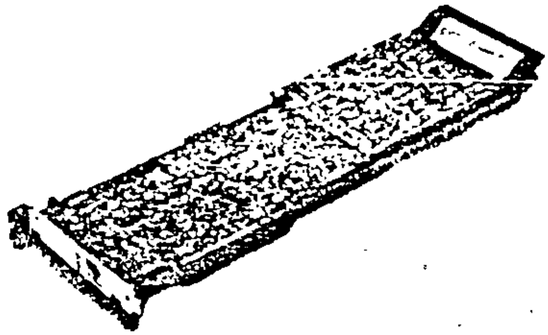
- Windows NT 3.51 and 4.0
- Small Business Server
- NetWare (no analog support)
- Windows 95
- Windows 98
- OS/2
- DOS

ORDERING INFORMATION

Product Name	Interface	Product Code
DIVA Server BRI 2M ISA	S/T	305-21[illegible]
DIVA Server BRI 2M ISA w/ NT1	w/ NT1	305-29[illegible]
DIVA Server BRI 2M PCI	S/T	305-20[illegible]
DIVA Server BRI 2M PCI w/ NT1	w/ NT1	305-44[illegible]



DIVA Server PRI



Eicon Technology's **DIVA Server PRI** solutions are exceptionally powerful, active ISDN cards that offer high-speed Primary Rate Interface (PRI) ISDN connectivity to corporations with a large number of remote users. Featuring channel aggregation, these cards can combine multiple B-channels into a single clear channel for applications that require extremely large throughput.

DIVA Server PRI PCI offers economy for digital-only applications. It supports inbound and outbound ISDN communications with remote ISDN devices on a single PRI line, which equals 23 B-channels in the U.S. and Japan, and 30 B-channels in Europe. DIVA Server PRI PCI is an excellent solution for any ISDN application with a large number of simultaneous, purely digital connections, such as remote access for telecommuters, Internet/intranet access, and LAN bridging and routing.

If dial in and dial out communication with analog modems and fax machine is required, Eicon offers the **DIVA Server PRI-23M** (North America and Japan) and the **DIVA Server PRI-30M** (Europe). With on-board Digital Signal Processors (DSPs), each card allows a single server to support a full range of call types over a single ISDN line. Communicate with:

- ISDN terminal adapters or routers
- analog modems
- fax machines (send and receive)
- GSM-compatible mobile phones in Europe (receive)

The DIVA Server PRI cards are the highest-capacity server solutions in the DIVA family of ISDN server cards.

FEATURES

- **DSP Programmed for ISDN:** For improved performance, all fax and modem connectivity is handled by the DIVA Server PRI.
- **Software Upgradable:** Upgrade to new software directly from Eicon's Web site. As standards change, DIVA Server PRI adapts to them.
- **Digital and Analog over ISDN:** With the DIVA Server PRI-23M and DIVA Server PRI-30M, connect to the Internet, and at the same time communicate with analog modems, receive calls from GSM-compatible mobile phones, and act as a fax server—one ISDN line is all you need.
- **Scalable Server Solution:** You can add additional server cards any time you need the extra capacity. DIVA server PRI cards can be grouped within a server to support multiple PRI lines.
- **Setup Wizard:** The DIVA Setup Wizard guarantees fast and easy setup of the DIVA for Windows NT software.
- **Rate Adaptation:** Rate adaptation permits the use of 56 kbps lines when local switches do not support 64 kbps.
- **Ready for Use Worldwide:** DIVA Server PRI is compatible with more ISDN switches than any competitive solution, and provides full support for multinational ISDN protocols.

OPERATING SYSTEM COMPATIBILITY

- Windows NT 3.51 and 4.0
- OS/2
- NetWare (no analog support)
- DOS

ORDERING INFORMATION

Product Name	Product Code
DIVA Server PRI PCI	3[illegible] 208
DIVA Server PRI-23M PCI	305-210
DIVA Server PRI-30M PCI*	305-211

*[illegible]

The DIV[illegible]mily Hardware Summary Chart

	Client Solutions						Server Solutions	
	DIVA LAN ISDN Modem	DIVA T/A ISDN Modem	DIVA T/A PC Card	DIVA Mobile V.90 PC Card	DIVA	DIVA Pro	DIVA Server BRI-2M	[illegible]
OS Supported	Independent	Independent	Independent	Windows 95 Windows 98 Windows NT	Windows 95 Windows 98 Windows NT Windows for Workgroups 3.11 DOS OS/2	Windows 95 Windows 98 Windows NT Windows for Workgroups 3.11 DOS OS/2	Windows NT Small Business Server NetWare Windows 95 Windows 98 DOS & OS/2	[illegible] [illegible] [illegible] OS/2
Bus Type	External Ethernet 10BaseT	External (com port)	PCMCIA Type II	PCMCIA Type II	ISA PnP 16-bit PCI 32-bit	ISA PnP 16-bit PCI 32-bit PC Card 32-bit	ISA PnP 16-bit PCI 32-bit	PC[illegible] 32
Interface	S/T or U	S/T, U	S/T U	S/T, w/NT1	S/T U	S/T U	S/T, w/NT1	[illegible]
ISDN Port	BRI (2 B-channels + 1 D-channel)	BRI (2 B channels + 1 D-channel)	BRI (2 B-channels + 1 D-channel)	BRI (2 B channels + 1 D-channel)	BRI (2 B-channels + 1 D-channel)	BRI (2 B-channels + 1 D-channel)	BRI (2 B-channels + 1 D-channel)	[illegible]
Processor	Motorola @ 20 MHz	Motorola @ 16 MHz	Motorola @ 16 MHz	Active Rockwell chip-set		DSP with 133 MHz clock, 33 MIPS	2 x 33 MIPS DSP, 32-bit RISC CPU, 16 MIPS	[illegible]
Data Access Speeds	128 kbps	128 kbps	128 kbps	128 kbps	128 kbps	128 kbps	128 kbps	[illegible]
Memory	Upgradable Flash ROM	Upgradable Flash ROM	Upgradable Flash ROM	Upgradable Flash ROM		80 kbyte	1 Mb	[illegible] DRAM
Plug and Play	✓	✓	✓	✓	✓	✓	✓	✓
Analog over ISDN	✓			V.90 - 56 kbps modem K56Flex - 56 kbps 14.4 kbps fax		V.34+ - 33.6 kbps modem, 14.4 kbps fax	V.34+ - 33.6 kbps modem**, 14.4 kbps fax	[illegible]
DSP-Based Fax Support				✓		✓	✓	✓ [illegible]
MLPPP	✓	✓	✓	✓	✓	✓	✓	
Compression	✓	✓	✓	✓*	✓*	✓*	✓*	✓*
Fax Group 3 Support	Via built-in phone port			✓	via 3rd party software	✓Windows 95 ✓Windows 98 Windows NT Windows for Workgroups via 3rd party software	✓Fax server with Microsoft® BackOffice® Small Business Server or other 3rd party software	✓[illegible]
Fax Group 4 Support (CAPI)				✓	✓	✓	✓	✓
CAPI 2.0				✓	✓	✓	✓	✓
V.120		✓	✓	✓	✓	✓	✓**	✓**
V.110						✓	✓**	✓**

* Provided by the RAS client
** Using Windows NT

FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSO INTERNACIONAL EN TELECOMUNICACIONES

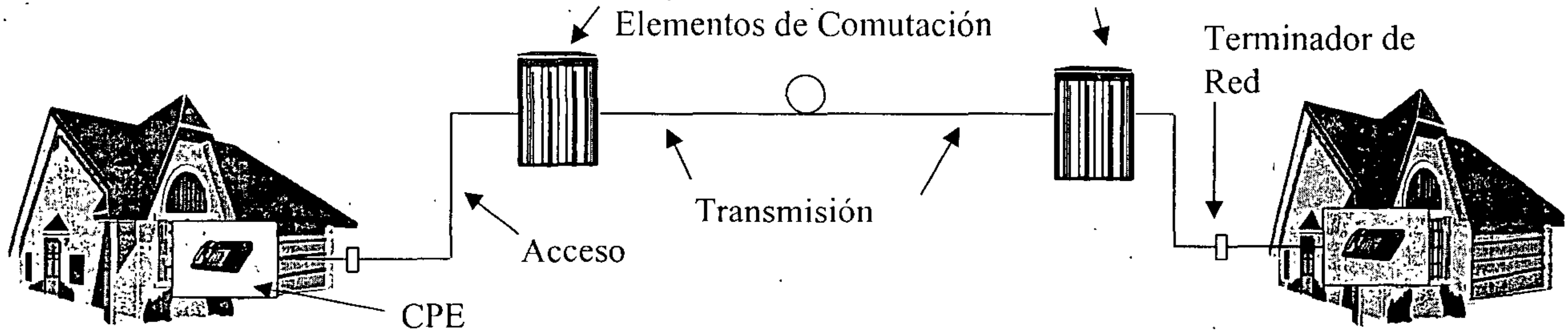
MOD.4.-REDES DIGITALES ACTUALIDAD Y PERSPECTIVAS.

TEMA: COMPLEMENTO DE LA RED.

JUNIO DE 1999.

Palacio de Mineria Calle de Tacuba 5 Primer piso Deleg. Cuauhtemoc 06000 México, D.F. APDO. Postal M-2285
Telefonos: 512-8955 512-5121 521-7335 521-1987 Fax 510-0573 521-4020 AL 26

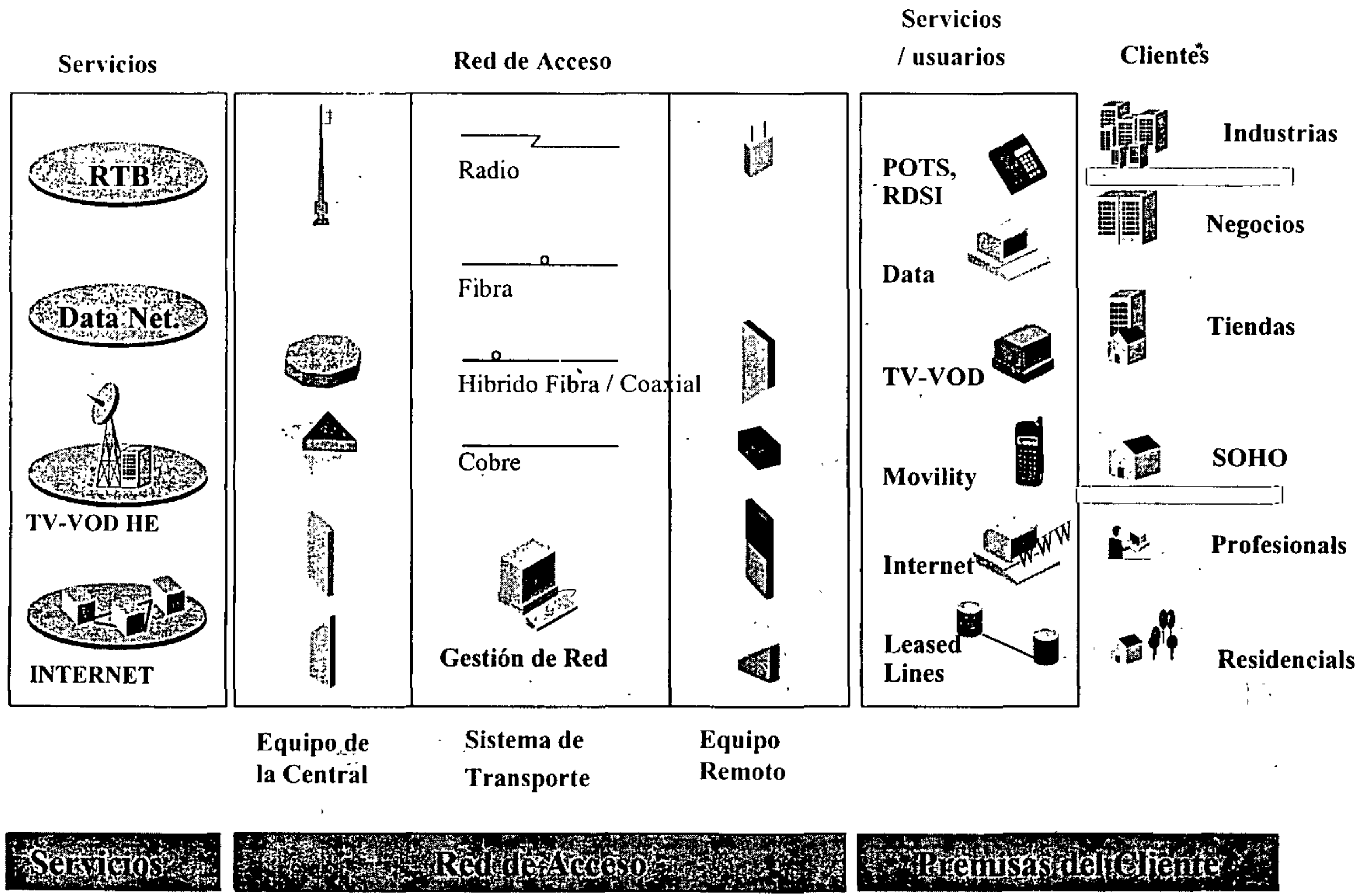
Modelo de Red



- **CPE (Customer Premises Equipment),**
- **Terminador de Red,**
- **Red de Acceso,**
- **Elementos de Conmutación,**
- **Servidores de Servicios,**
- **Transporte,**

ALCATEL

Modelo de Red (2)



Modelo de Red (3)

Servidores (Servicios)	Central (Proceso)	Acceso (Flexibilidad)	Conectividad del Cliente
IN SCP/SMP & SCE	Banda Ancha	FITL	CLI & CLASS CPE
ADSI SV	CLASS	Wireless	Teléfono ADSI
Internet Gateway	Conference Bridge	ISDN	Pagers
Facturación y Atención al Cliente	ISDN	SDH Fibra	PCs, Módems & Software
Correo de Voz y Fax	Transporte	Cobre	Bridges & Routers

7

FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

RDSI CONCEPTOS

EXPOSITORA: ING. MARÍA DEL C. ANGÉLICA MORENO ARGÜELLO
PALACIO DE MINERÍA
JUNIO DE 1999

Contenido

INTRODUCCIÓN

Hasta hoy la mayoria de los sistemas de transmisión entre los nodos (centrales telefónicas) de la red telefónica son digitales. Pero la transmisión y la señalización hacia el subscriptor es todavia analógica. (Véase fig. 1 y 2).

Mundialmente existe una creciente necesidad de mover información entre diferentes partes del mundo y además esta transferencia de información cada vez debe ser más rápida y barata sin importar donde se encuentren localizados los puntos donde se desee dicha información.

Otra situación actual es en los servicios de telecomunicaciones, donde para hacer uso de ellos (telefonia, fax, datos, telex, datos en conmutación de paquetes, etc.) se debe tener un acceso (linea) diferente con un equipo terminal, interfase y red diferente.

Para resolver estos problemas una nueva red que pretende ser universal esta siendo desarrollada y se le conoce como la **R**ed **D**igital de **S**ervicios **I**ntegrados "**RDSI**".

Existen 3 tendencias mundiales que están trabajando en la definición de normas RDSI, que son CCITT Recomendaciones Internacionales, ETSI (European Telecommunications Standards Institute) normas para la Comunidad Europea y (Bellcore-ANSI) para Estados Unidos.

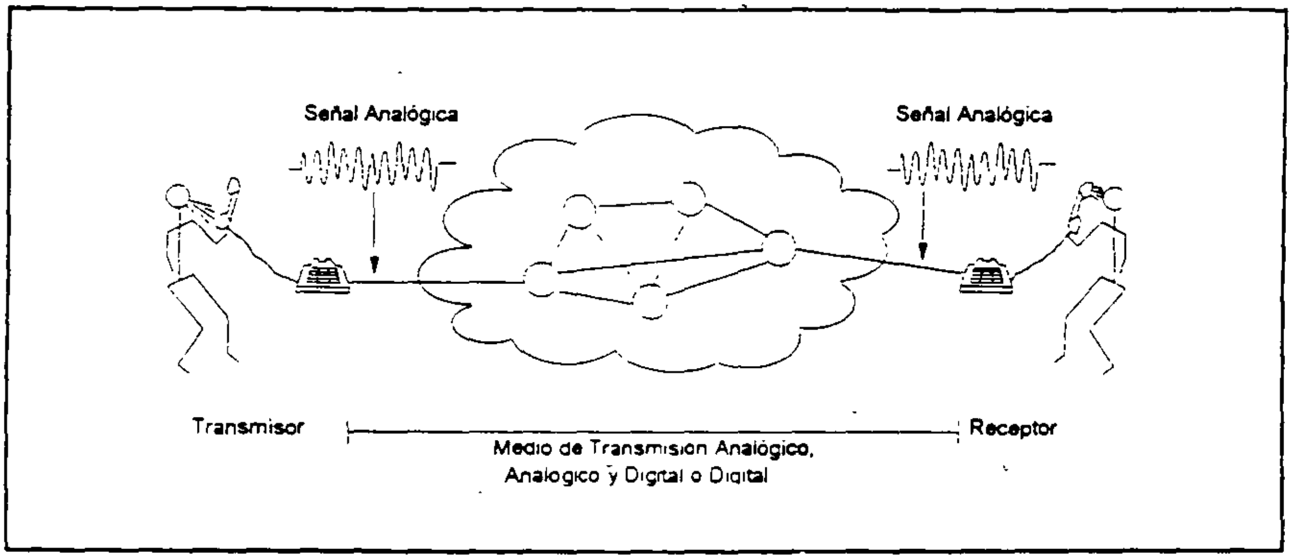


Fig. 1. Linea de usuario en la actual Red Telefonica

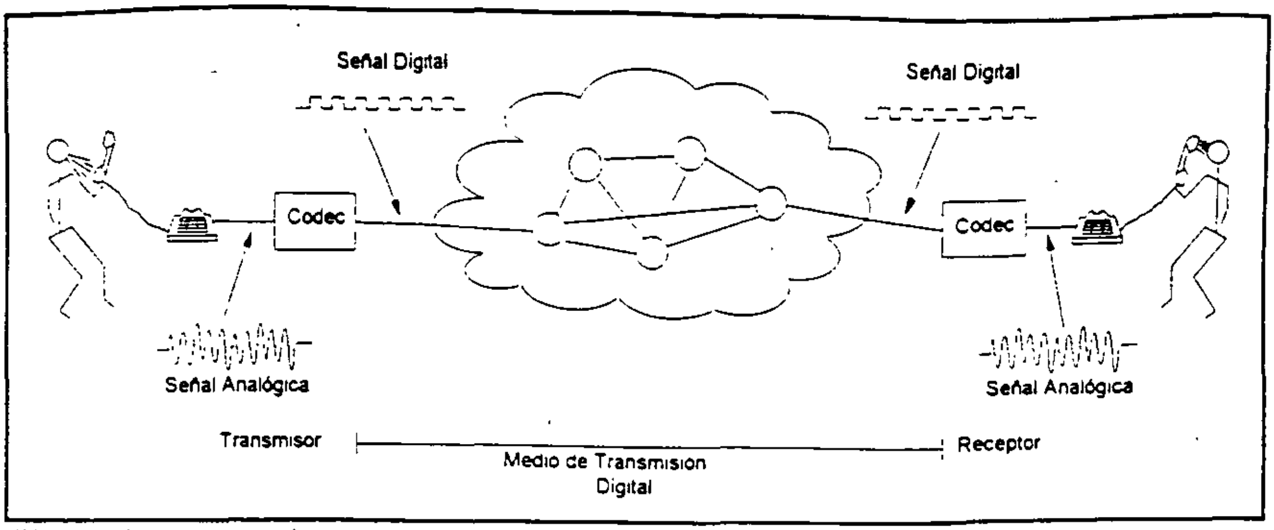


Fig. 2. Linea de usuario con la RDSI

¿QUE ES LA RDSI?

Según el CCITT (Comité Consultivo Internacional de Telefonía y Telegrafía), la RDSI es una red que permite una conectividad digital extremo a extremo para ofrecer una amplia gama de servicios de telecomunicaciones (existentes y por desarrollar) los cuales podrán ser accesados a través de un conjunto reducido y normalizado de interfases, dicha red debe ser una evolución natural de la red telefónica mundial existente.

En las figuras 3 y 4 se puede observar un ambiente donde se hace uso de diferentes servicios de telecomunicaciones en la actualidad y como seria ese mismo ambiente cuando la RDSI exista de forma comercial.

Una de las premisas más importantes bajo la cual fue concebida y diseñada la RDSI es el utilizar al máximo la infraestructura de la red telefónica mundial existente ya que representa en promedio, según datos recopilados por la UIT/CCITT aproximádamente del 0.4 al 1.0% del producto nacional bruto de cada pais.

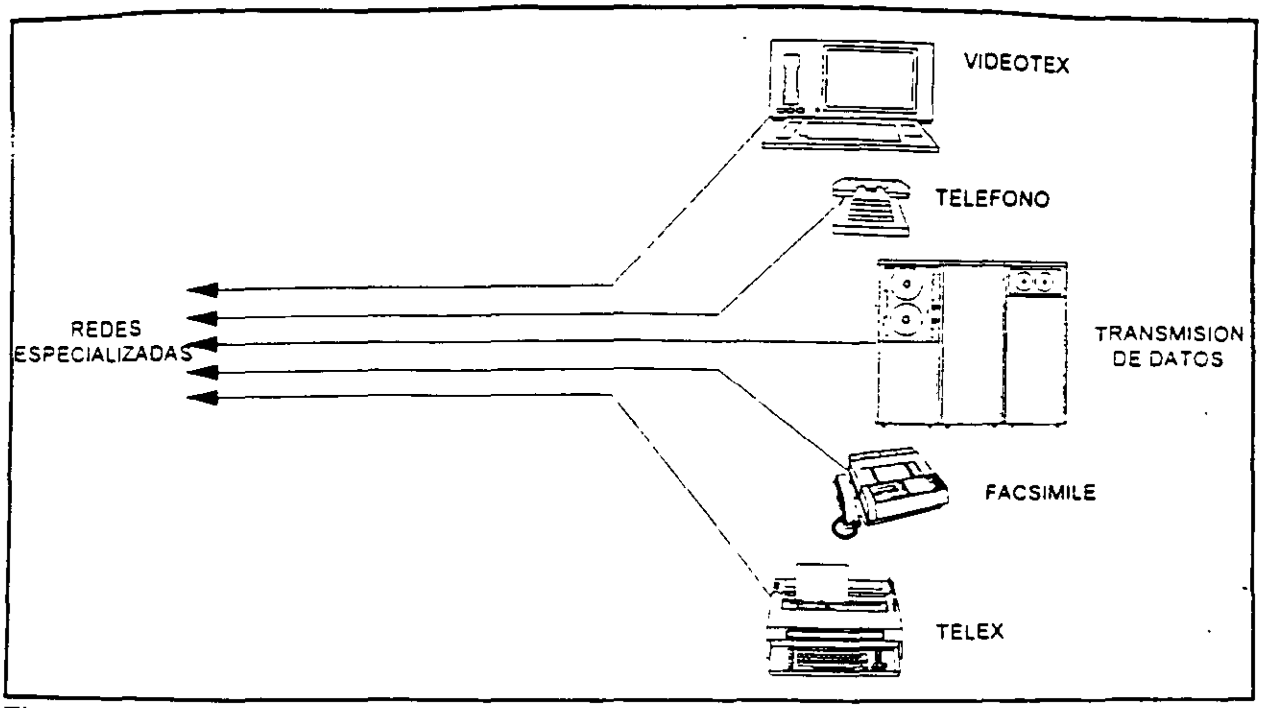


Fig. 3 Acceso a los servicios de telecomunicaciones en la actualidad (sin la RDSI)

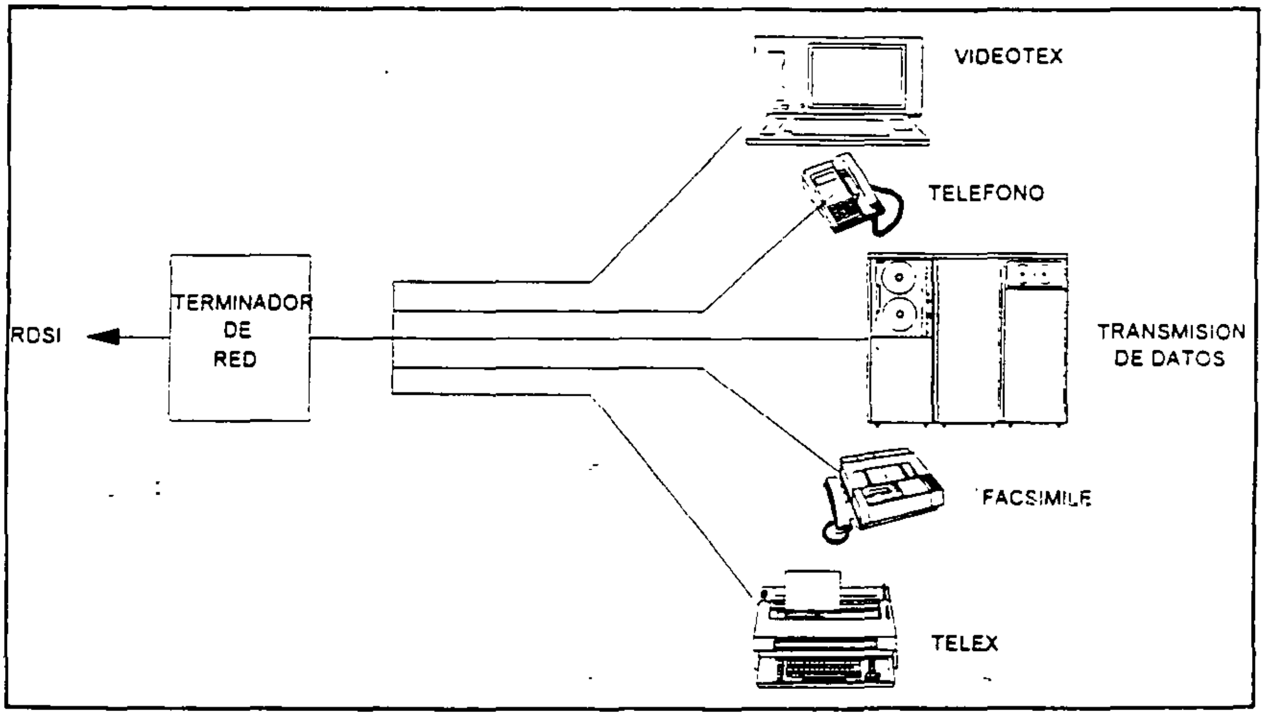


Fig 4. Acceso a los servicios de telecomunicaciones con la RDSI.

Dentro de esta inversión el más alto porcentaje es consumido por la red externa (toda la infraestructura que va desde la central telefónica hasta las instalaciones del usuario), el cual se muestra en la fig. 5.

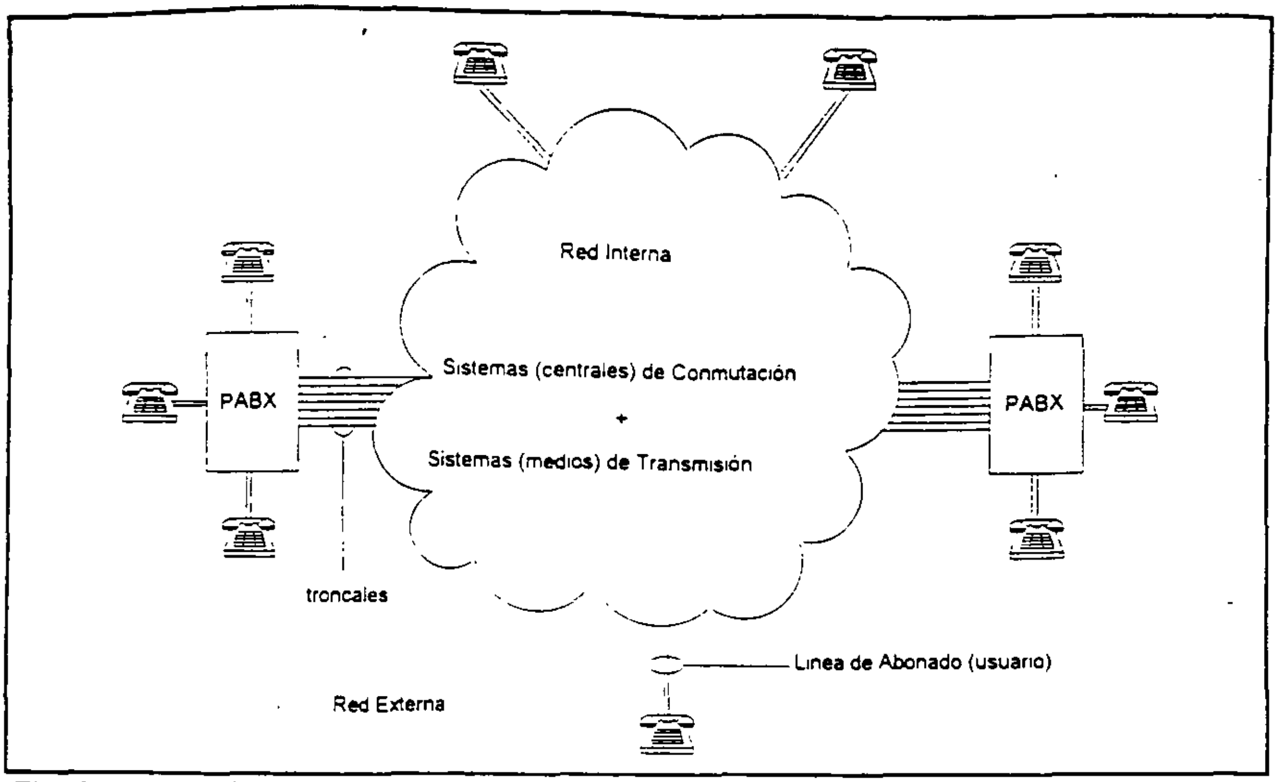


Fig. 5. Red Telefónica.

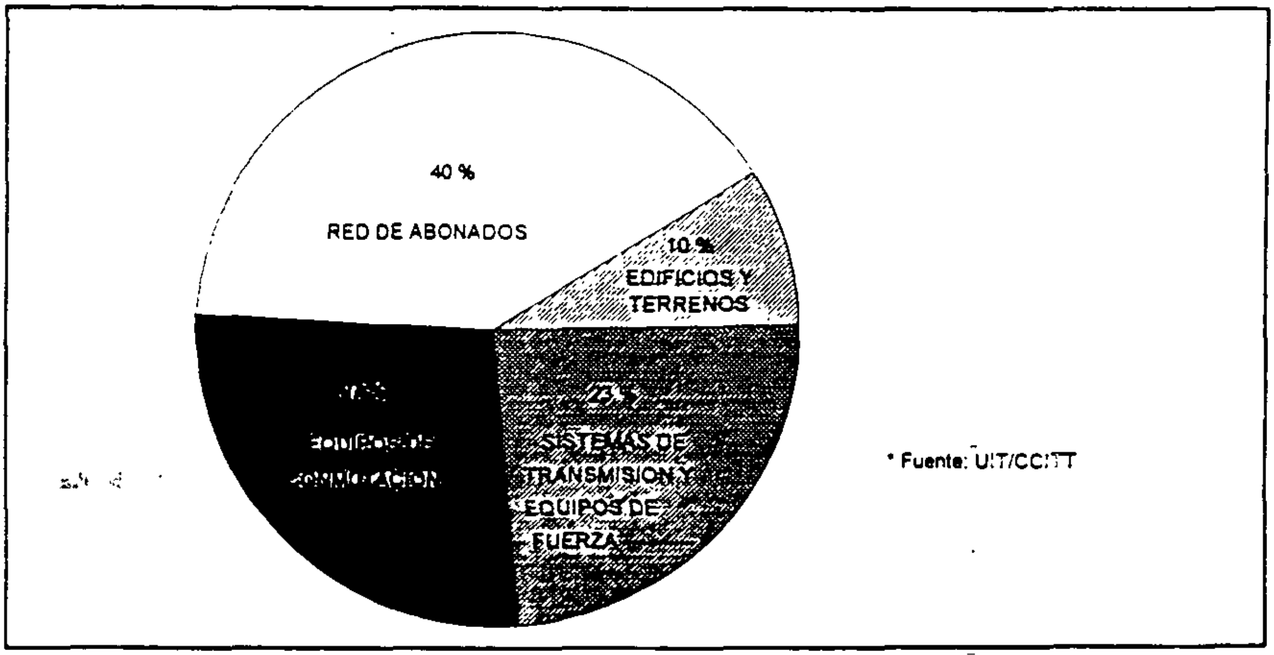


Fig. 6. Inversión en la red telefónica.

Conforme al estudio de la UIT/CCITT, un promedio del 40 al 50% de la inversión total en telecomunicaciones está en la red externa, como se observa en la fig. 6.

Las condiciones de la planta externa son de suma importancia porque determinan la calidad de los servicios ofrecidos a los subscriptores ya que juegan un papel crucial por que están al inicio y al final de toda llamada telefónica ya sea local, interurbana o internacional. Así la introducción de sistemas digitales de conmutación no puede ser eficaz sin el mismo elevado nivel de calidad en la planta externa.

Por esta razón, la planta externa ocupa un lugar destacado en la red telefónica y requiere un diseño y planeación apropiados, así como un buen sistema de operación y mantenimiento.

ACCESO A LA RDSI

El CCITT ha definido 2 formas de acceso o de conectarse a la RDSI y se les conoce como:

a) Acceso Básico y
b) Acceso Primario

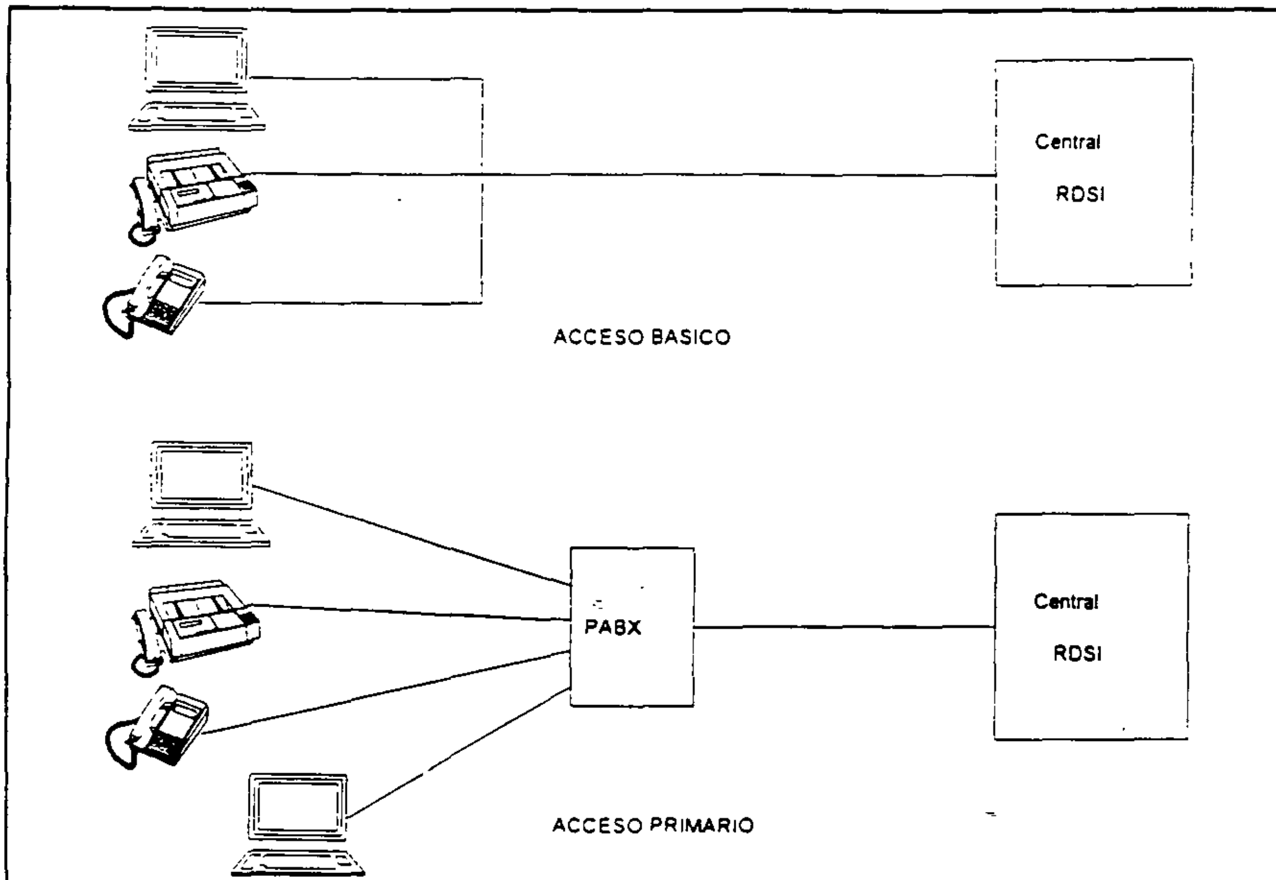


Fig. 7. Tipos de Acceso a la RDSI.

Una forma práctica de identificar la diferencia que existe entre estos dos tipos de accesos se muestra en la fig. 7, donde se puede observar que el Acceso Básico es exclusivamente para conectar y dar servicio a usuarios que tienen una linea telefónica y el Acceso Primario está enfocado a conectar usuarios que actualmente tienen un conmutador (PABX, Private Automatic Branch eXchange) y que están haciendo uso de un sistema de transmisión PCM (Pulse Coded Modulación) de 2.048 Mbps.

CONFIGURACIÓN DE REFERENCIA

Existe un modelo de referencia definido por el CCITT donde se dan los detalles de las interfases que existen en el lado del usuario para conectarse a la red publica RDSI, la cual se muestra en la fig. 8.

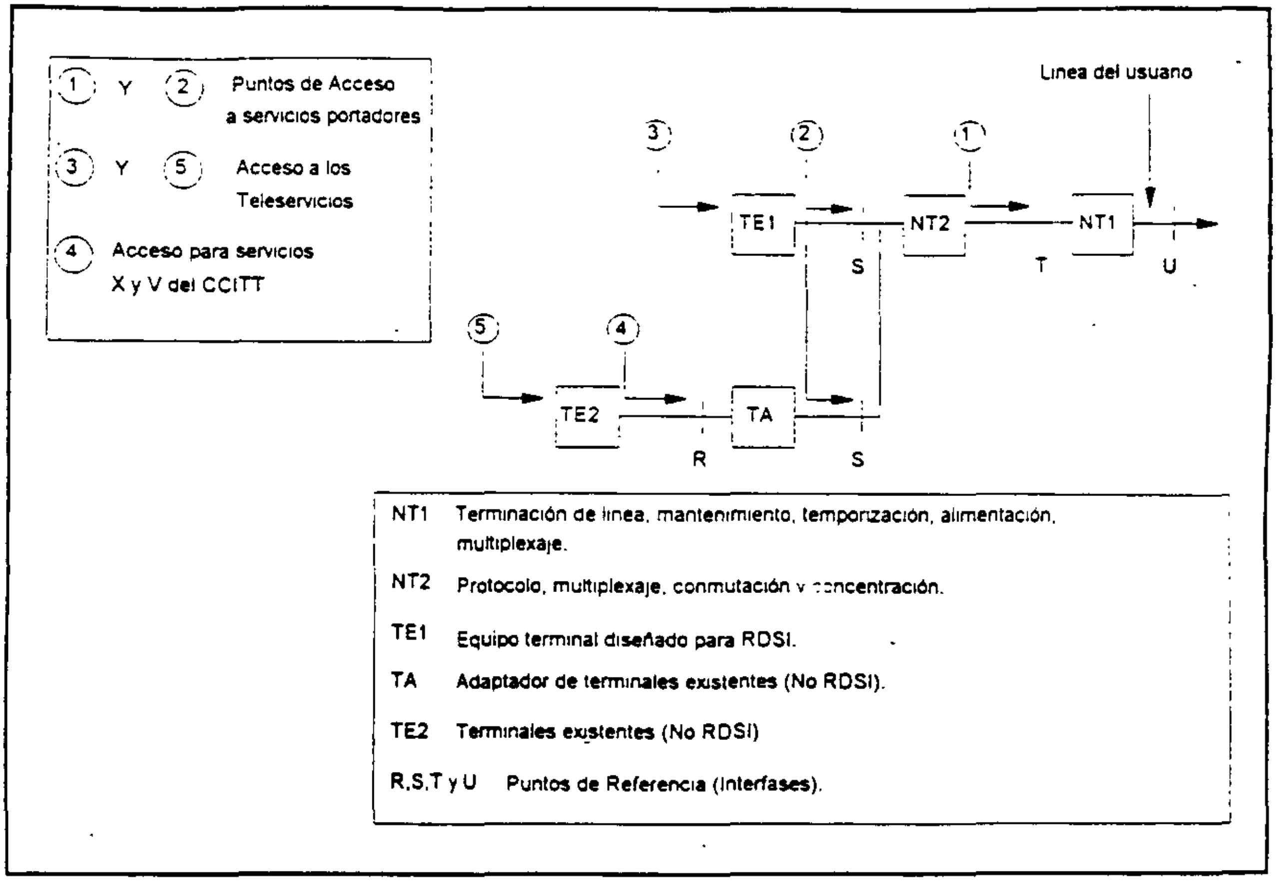


Fig 8. Configuración de Referencia para las interfases usuario-red en la RDSI.

La configuración de referencia (Fig. 8) ubica la interfase Usuario-Red, a través de la cual los usuarios se podrán conectar a la RDSI y tener acceso a los servicios que ofrece ésta.

La interfase Usuario-Red esta ubicada entre los equipos considerados dentro de las premisas del usuario y la central RDSI. Dentro de las premisas de usuario existen básicamente 2 tipos de equipo:

a) Equipo Terminador de Red (NT) y
b) Equipo terminal (TE)

Este equipo es agrupado en **bloques funcionales** los cuales representan una o más partes de equipo. Por ejemplo, algunas veces las funciones de un tipo de equipo están fisicamente ubicadas o implementadas en otro, en casos como este solamente un bloque funcional será mostrado y dependiendo de las necesidades del usuario algún equipo puede o no ser necesario.

Las interfases entre los bloques funcionales son llamados **puntos de referencia**, los cuales son lógicos más que fisicos; esto es, puede no haber una interfase fisica en un punto de referencia dado. (Este es el caso cuando las funciones de un equipo son proporcionados por otro, además de las propias.)

La figura 9 muestra un ejemplo de la forma de conexión por parte de un usuario a la RDSI.

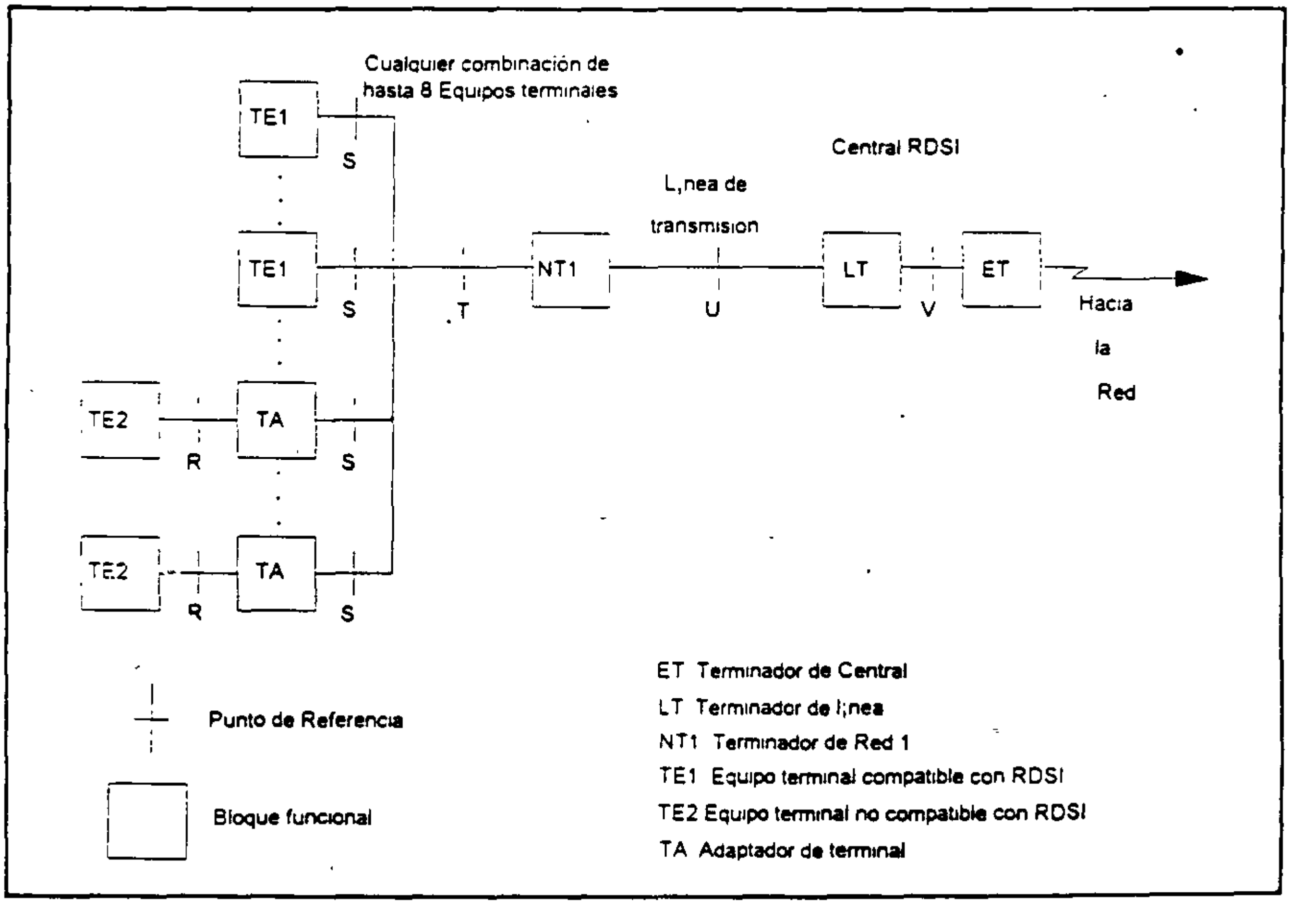


Fig. 9. Ejemplo de la conexión de un usuario a la RDSI

A continuación se describe de forma muy general los bloques funcionales y los puntos de referencia incluidos en la configuración de referencia.

EQUIPO TERMINAL (TE)

El equipo terminal maneja las comunicaciones en el lado del usuario de la interfase Usuario-Red Ejemplos de este tipo de equipos son Terminales de datos, teléfonos, computadoras personales y teléfonos digitales. Los TEs tienen funciones para el manejo de protocolos, de mantenimiento, de interfase y de conexión hacia otros equipos, así como funciones para el manejo de la aplicación propia (teleservicio) del equipo.

EQUIPO TERMINAL DEL TIPO 1 (TE1)

Los TE1s realizan las funciones de los TEs y además tienen integrada la interfase "S", los que los hace compatibles con la RDSI de forma directa. Ejemplos de este tipo de equipos son Terminales multiservicio para voz, datos y video, así como teléfonos digitales RDSI

EQUIPO TERMINAL DEL TIPO 2 (TE2)

Los TE2s también realizan las funciones de los TEs, pero ellos no tiene la interfase "S" que los permite conectarse a la RDSI En lugar de esta interfase tienen otras como la RS232C, V.35, V.24, X.21, etc Sin embargo este tipo de equipos pueden ser conectados a la RDSI a través de un adaptador de terminal (TA) Ejemplos de este tipo de equipos son los teléfonos, fax y computadoras personales existentes.

ADAPTADOR DE TERMINAL (TA)

Este tipo de equipos permite la conexión de TE2s a la RDSI, realizando funciones de conversión en velocidad y protocolos de los equipos TE2 hacia los estándares (interfase S) de la RDSI.

EQUIPO DE TERMINACIÓN DE RED (NT)

El equipo de terminación de red maneja las comunicaciones del lado de la red (central RDSI) de la interfase Usuario-Red.

TERMINADOR DE RED DEL TIPO 1 (NT1)

Los equipos NT1 proporcionan funciones equivalentes al nivel 1 del modelo OSI (Open Systems Interconexion). Estas funciones incluyen conversión de señal, temporización, mantenimiento de la linea de transmisión (interfase "U") y la terminación fisica y eléctrica de la red en las instalaciones del usuario. Algunas veces, el NT1 puede estar integrado en otro equipo y por lo tanto no existir de forma fisica separada.

TERMINADOR DE RED DEL TIPO 2 (NT2)

Los equipos NT2 son más inteligentes que los NT1s y proporcionan funciones adicionales entre las cuales se puede incluir multiplexaje y manejo de protocolos en los niveles 2 y 3 del modelo OSI. Ciertos tipos de NT2s, tales como los PABXs manejan funciones de los nivel 1, 2 y 3, mientras otros, como por ejemplo controladores de terminales, solo proporcionan funciones correspondientes al nivel 1 y 2 del OSI.

EQUIPO DE LA CENTRAL

Este equipo no pertenece a las premisas del usuario, por lo que estrictamente hablando no son parte de la interfase Usuario-Red Sin embargo se incluye por estar en la configuración de Referencia.

TERMINACIÓN DE LINEA (LT)

Estos equipos realizan funciones de terminación de linea en el lado de la central de la linea de transmisión (interfase "U").

TERMINACIÓN DE CENTRAL (ET)

Estos equipos manejan la información de señalización de la interfase Usuario-Red e inician los procedimientos para el manejo de la llamada a través de la red.

PUNTOS DE REFERENCIA

Los puntos de referencia son los puntos de conexión entre los bloques funcionales. Es necesario tener presente que los puntos de referencia son conceptuales y no indican una interfase fisica.

PUNTO DE REFERENCIA R

Este punto corresponde a un interfase (tal como RS232C, V.24, V 35 ó X.21) entre un equipo terminal que no es RDSI (TE2) y un adaptador de terminal (TA).

PUNTO DE REFERENCIA S

Este punto es una interfase a 4 Hilos (1 par para Tx y el otro para Rx) entre un TE1 o un TA y un NT2. Este punto es fisicamente idéntico a la interfase T Hasta 8 equipos TE1s ó TE2 (con sus respectivos TAs) pueden ser conectados a través del punto de referencia S a un NT1. El NT2 efectivamente divide al punto de referencia T en varios puntos de referencia S.

PUNTO DE REFERENCIA T

Este punto es una interfase a 4 Hilos entre un TE1 (o un TA o un NT2) y un NT1 Un par es usado para Tx y el otro para Rx Fisicamente esta interfase es idéntica a la interfase S En algunos casos de PABXs (NT2), el NT1 está integrado al NT2 por lo que no existe el punto de referencia T.

PUNTO DE REFERENCIA U

La interfase U es la línea de transmisión entre la interfase Usuario-Red y la central RDSI Específicamente se encuentra entre el NT1 y la LT. Es una interfase "full-duplex" sobre el par torcido de alambres de cobre (El mismo par se utiliza para Tx y Rx de forma simultánea).

En los EE.UU., el punto de referencia U es el limite entre la interfase usuario-red y la central RDSI. Esto hace que el NT1 pertenezca a las premisas del usuario, mientras que para Europa el límite entre el usuario y la administración telefónica es el punto S/T

PUNTO DE REFERENCIA V

La interfase V divide el equipo LT del ET Esto tampoco ha sido estandarizado y es función directa de la implementación de cada proveedor de equipo de conmutacion (centrales RDSI).

INTERFASE U

Este punto de acceso a la RDSI no esta normalizado por el CCITT, por lo que cada administración define la técnica de transmisión, el código de línea y las características físicas de la interfase

Por razones económicas el actual par de hilos de cobre que llegan a la casa del usuario telefónico deben ser utilizados para transportar la información de los servicios ofrecidos por la RDSI, es por esto que la línea de abonado debe permitir transmitir 160 kbps (144 kbps de los canales 2B+D más bits extras para información de mantenimiento alineación, etc.) en forma "full-duplex".

El diseño de esta interfase se tienen básicamente 2 problemas

1) Transmisión "full-duplex" en 2 hilos de información digital.
2) Velocidad de transmisión en la línea es de 160 kbps.

El primer problema se resuelve utilizando una *técnica adecuada de transmisión* y el segundo tratando de reducir la velocidad con un *código de línea* que además permita aprovechar las características de transmisión que presenta el par de hilos de cobre

TÉCNICAS DE TRANSMISIÓN EN LA LINEA DE ABONADO (INTERFASE U)

TRANSMISIÓN A 4 HILOS

Por supuesto, esta técnica no tiene posibilidades en la práctica ya que todos los subscriptores existentes en la actual red telefónica se conectan con un solo par. Solamente se conectan a 4 hilos cuando la conexión es de 2.048 Mbps (por ejem. la conexión de un PABX) Véase fig. 10.

DIVISIÓN DE FRECUENCIA

Con la técnica de división en frecuencia es posible transmitir en forma "full-duplex", sin embargo las señales digitales codificadas enviadas por la línea se traslapan en su densidad espectral. Para evitar este problema se usan diferentes códigos de línea en cada dirección (por ejem. código bipolar de orden 1 en una dirección y de orden 2 en la otra dirección) ó usando el mismo código en ambas direcciones pero modulando la información transmitida en una de las direcciones.

La separación de la información en el lado de recepción es realizada mediante filtros La distancia que se puede alcanzar esta condicionada por las señales de alta frecuencia que tengan gran cantidad de energía; debido a la diafonía en el lado lejano (FEXT, Far-end crosstalk), la cual es producida por líneas adyacentes de diferente longitud. Las señales de alta frecuencia son transmitidas en la dirección de la central al subscriptor

Una de la ventajas de esta técnica es que la diafonía en el lado cercano (NEXT, Near-end crosstalk) es minimizada debido a que los espectros para transmitir y recibir son diferentes; sin embargo el diseño de los filtros es complejo y su implementación en circuitos integrados digitales presenta problemas. Además no es posible utilizar el mismo equipo en la central y en el subscriptor debido a la asimetria en la transmisión; por lo que esta técnica ha sido abandonada.. Véase fig. 10.

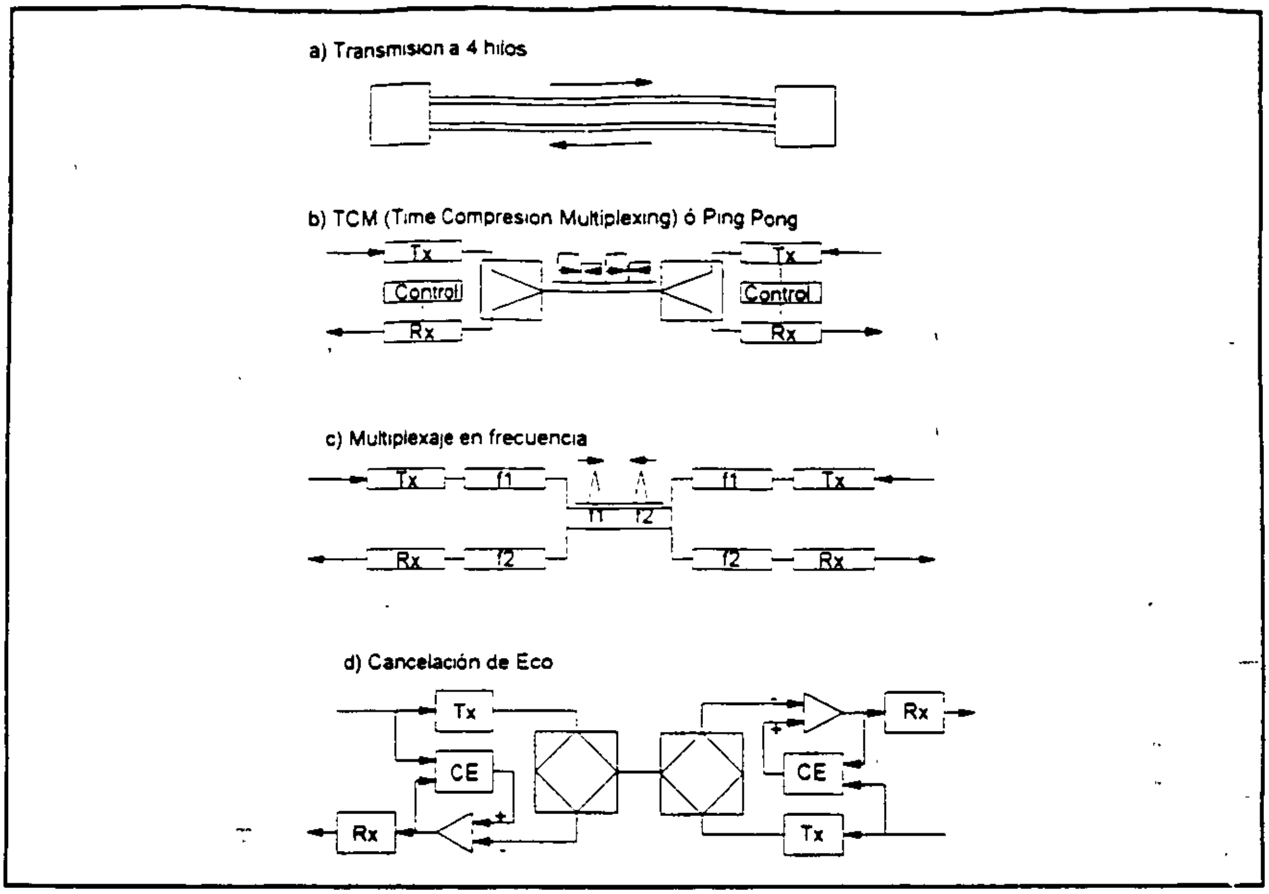


Fig. 10 Métodos de transmisión en la línea de abonado (Interfase U).

TCM (Time Compression Multiplexing) ó PING PONG

Este método también llamado de ráfagas, involucra el cambio alternado de la dirección de transmisión. Esta alternación en la transmisión, no es en el sentido de la transmisión "half-duplex" sino que esta técnica garantiza que efectivamente haya una transmisión "full-duplex", aunque a nivel microscópico esto sea "half-duplex" dado que el transmisor y receptor transmiten en tiempos diferentes. La información binaria es almacenada en forma de bloques en los extremos del enlace y son transmitidos en intervalos de tiempo diferentes. Por lo tanto existen dos fases que no deben traslaparse transmisión y recepción; que pueden ser distinguidas en cada extremo del enlace.

Por lo tanto para una velocidad de información D, la velocidad de línea requerida debe ser mínimo 2D; de hecho considerando la propagación en los cables y el tiempo utilizado entre las diferentes fases dan una velocidad del orden de 2.5D.

La distancia teórica máxima está dada por

$$L_{max} = \frac{V}{2(N/D - 2N/F - 2t_h)}$$

Donde.

V = Velocidad de propagación en los cables (aprox. 200,000 Km/s)
N = Número de elementos binarios en el bloque
F = Velocidad de línea
t_h = Tiempo de guarda (para evitar interferencia entre la transmisión)

Bloques de longitud muy grande reducen el números de veces que se debe alternar la dirección de transmisión y con ello el efecto de la propagación para de esta forma incrementar la longitud teórica, sin embargo para señales de voz el retardo de los octetos produce degradación en la calidad.

Una longitud teórica grande es también obtenida aumentando la velocidad de transmisión pero esta se ve limitada por la atenuación y la diafonia que presenta el par de hilos de cobre.

CANCELACIÓN DE ECO

Este método es utilizado actualmente en transmisión analógica en bajas frecuencias para proporcionar transmisión "full-duplex" por un par, utilizando un acoplador (bobina hibrida) de dos a cuatro hilos con una impedancia balanceada que representa un compromiso entre las impedancias representadas por ambas lineas. De hecho en la hibrida la red balanceada colocada en el lado del medio de transmisión produce un desacoplo y permite que algunas de las señales transmitidas regresen junto con las señales recibidas, a este fenómeno se le conoce como eco local.

La atenuación de la trayectoria del eco para un ancho de banda de aproximadamente 100 kHz es del orden de 10 a 15 dB pero puede caer hasta 6 dB para configuraciones de cable específicas. Un receptor digital solo funciona correctamente para una relación señal a ruido de aproximadamente +25 dB. Dado que se requiere para un sistema de transmisión digital de aproximadamente 45 dB a 100 kHz, la señal remota es atenuada por el valor correspondiente. Por lo tanto es necesario reducir el eco local aproximadamente 64 dB (45dB + 25dB - 6dB) para que los datos sean detectados correctamente. El eco remoto de pequeña amplitud debido al desacoplo de impedancias a lo largo de la línea es sumado al eco local.

Para eliminar la señal producida por dicho desacoplo de impedancias, se ha diseñado un dispositivo que elimina el eco usando la información transmitida, llamado "Cancelador de eco". De hecho el eco es resultado de la configuración intrinseca de la línea de abonado y de las caracteristicas de los simbolos (código de linea) que están siendo transmitidos sobre ella. Este dispositivo hace uso del principio de que no exista una correlación entre el eco y la señal que proviene del lado remoto, para este efecto se usan diferentes aleatorizadores (scramblers) en cada uno de los extremos de la línea. Además el circuito que realiza las funciones de procesamiento de señales debe ser flexible para aceptar todas las posibles configuraciones de una línea de subscriptor en una red telefónica y responder a cualquier variación en sus características con el tiempo.

Existen básicamente dos métodos para estimar el eco; uno usa un filtro transversal y el otro esencialmente usa memorias.

En el primer método el filtro contiene N (el cual puede alcanzar varias decenas) coeficientes variables que representa la respuesta al impulso del eco muestreado. La multiplicación de estos coeficientes con la secuencia de los datos transmitidos producen la perturbación instantánea debida al eco, la cual es calculada cada vez que se transmite un símbolo. Los coeficientes del cancelador de eco son ajustados para reducir el error residual que resulta de una mala estimación del eco real. Se puede demostrar que la diferencia entre el eco real y el eco estimado puede ser expresado estadísticamente, tomando en consideración la no correlación de la señal, como una función de los datos transmitidos y del total de la señal recibida (estos parámetros se obtienen del sistema de recepción). Por lo tanto es posible minimizar este error usando algoritmos de mayor o menor grado de complejidad (del gradiente o tipo de signo) el cual asegura una convergencia progresiva del cancelador de eco. Este método implicitamente asume que el eco del canal es lineal y que cualquier no linealidad está fuera del rango de operación del cancelador, lo cual implica que cualquier no linealidad en la codificación sean excluidas de la trayectoria del eco. Sin embargo otras no linealidades pueden aparecer como. desbalance en el transmisor ó no linealidad del convertidor analógico-digital

El segundo método, usa memorias que contienen el eco que ha sido previamente calculado para todas las posibles secuencias de información con lo cual se puede compensar las no linealidades. Si se asume que el eco puede ser modelado mediante un filtro de N coeficientes para N datos binarios sucesivos, el eco solo puede tomar 2^N valores y por lo tanto es suficiente que los N elementos binarios sean usados para direccionar una memoria cuyo contenido varia en función de error residual de la señal. La gran cantidad de memorias y los grandes tiempos de convergencia son la principales desventajas de este método.

Consecuentemente estructuras intermedias han sido diseñadas, por ejemplo M memorias con $2^{N/M}$ palabras cuyos contenidos son sumados para producir el eco; para esto se debe establecer un compromiso entre robustez a la no linealidad, la velocidad de cálculo y el tiempo de convergencia.

La principal ventaja del cancelador de eco es la preservación de espectro en frecuencia correspondiente en banda base. Sin embargo es importante evitar códigos de línea con mucha energía en las bajas frecuencias para asegurar una buena robustez contra el ruido de la red local, que por lo general ocurre en la banda de 0 a 20 kHz

Por lo antes descrito es conveniente usar códigos de línea para este método de transmisión, que sean líneales y que sean invariantes con respecto al tiempo en el proceso de almacenamiento de las respuestas al impulso. Algunos de los códigos con esta características son el bifase, bipolar, 4B3T y 2B1Q. El código determina la complejidad de su implementación en Circuitos Integrados , por ejem un CI de transmisión que contenga cancelación, equalización, recuperación de la temporización y activación pueden contener hasta 50,000 transistores, pero se puede disminuir esta cantidad realizando una adecuada selección del código.

Después de que el eco ha sido estimado, se elimina (mediante una operación de sustracción) y en ese momento generalmente la señal es manejada como una transmisión a 4 hilos, sin embargo es necesario realizar filtrados adicionales para reducir la interferencia entre símbolos. La velocidad de convergencia del sistema cancelador de eco es un elemento clave en el tiempo de establecimiento de la comunicación. Cuando el sistema ignora por completo las características de la línea, el tiempo de convergencia de arrancando desde un estado aleatorio los coeficientes; puede tomar algunos segundos, sin embargo si los coeficientes son almacenados entre una comunicación y otra, el tiempo de convergencia no excede los 100 ms.. Véase fig. 10.

CARACTERÍSTICAS QUE DEBE TENER UN CÓDIGO DE LINEA PARA RDSI.

El objetivo que se persigue en RDSI en la interfase "U", es bajar lo mas posible la velocidad de la linea, transmitiendo la misma cantidad de información, por lo que el código que cumpla mejor con las siguientes características, será un código adecuado para RDSI.

1. Transparente a la información.
2. Facilidad para recuperar el reloj.
3. Evitar (si es posible) la componente de corriente continua, así como la presencia de grandes cantidades de energía a bajas frecuencias.
4. Redundancia (deseable) para detectar errores en la linea.
5. Espectro limitado en frecuencia para hacer un buen uso de la atenuación y de la diafonía (crosstalk) presentada por el par torcido de cobre.
6. Reducción en la velocidad de transmisión.
7. Eficiencia.
8. Propagación mínima de errores.
9. Insensibilidad a la permutación en los cables del par.

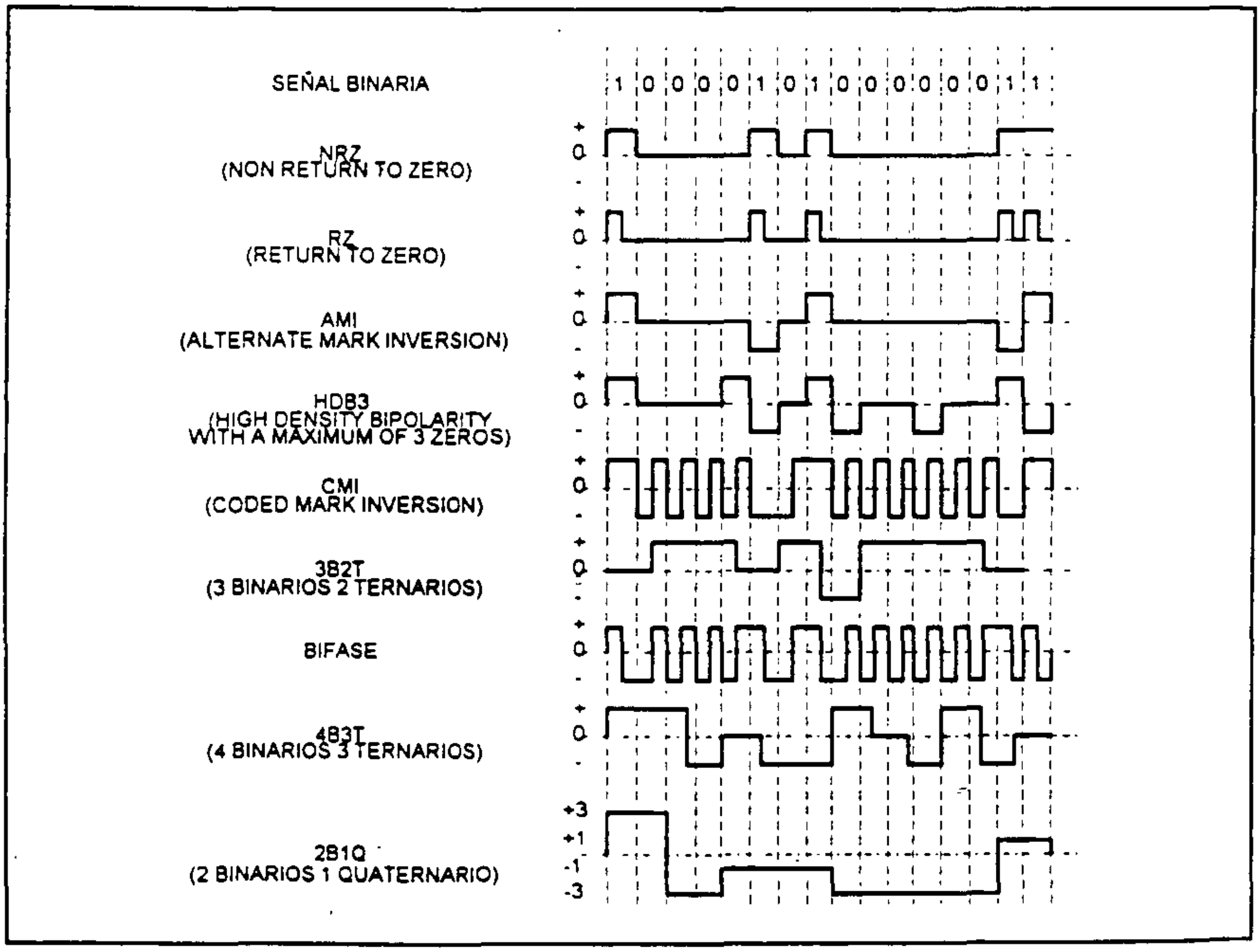


Fig. 11. Códigos de Linea.

En la fig. 11 se muestran los códigos de linea más utilizados en sistemas de transmisión, sin embargo los códigos más utilizados por las Administraciones Telefónicas para RDSI en l interfase U son:

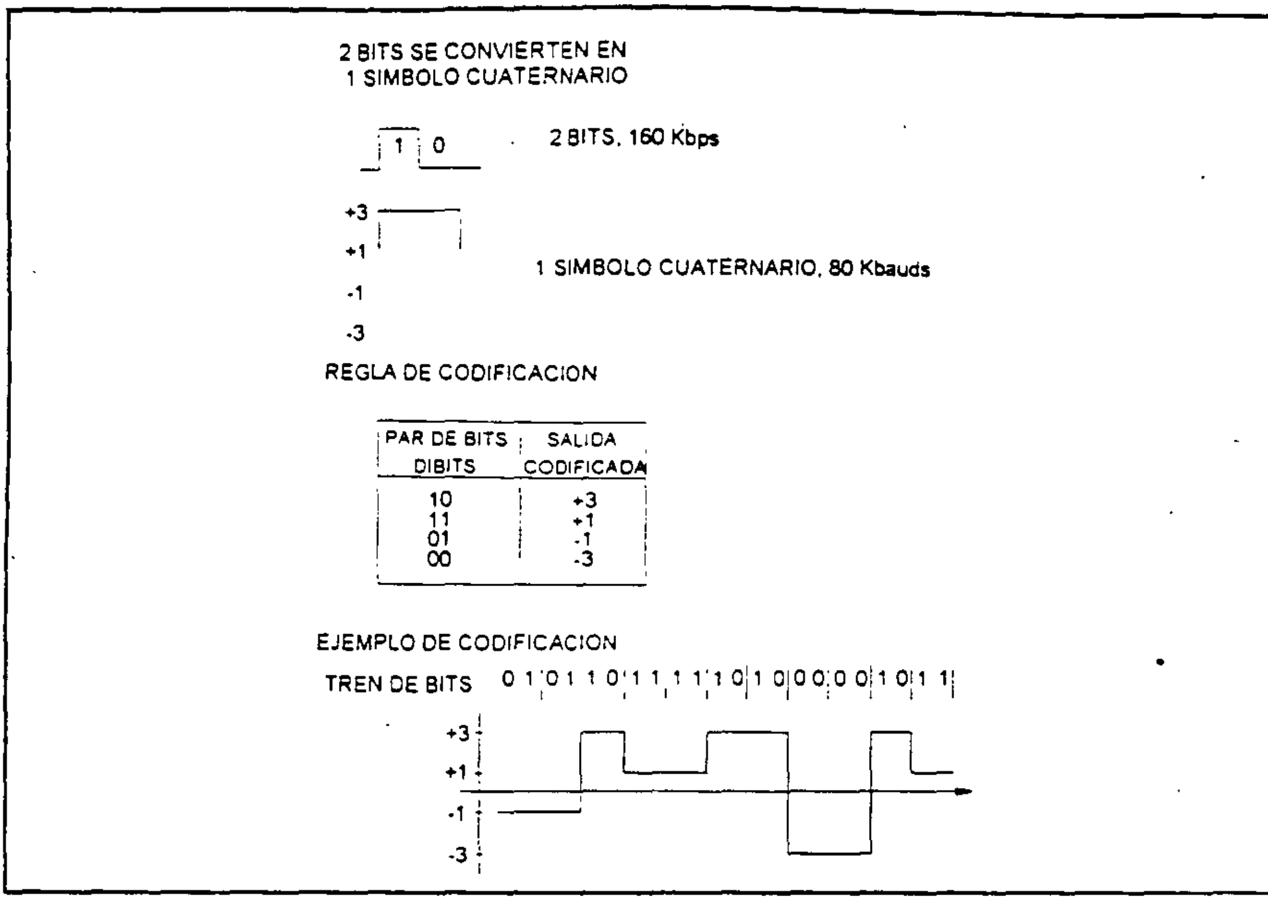


Fig. 12. Código de linea 2B1Q para la interfase U.

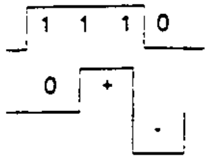
a) 4B3T 4 símbolos binarios son representados mediante 3 símbolos ternarios (3 niveles de voltaje posibles por cada símbolo).

b) 2B1Q 2 símbolos binarios son reprêsentados mediante 1 símbolo cuaternario (4 niveles de voltaje posibles por cada símbolo).

Código de linea 2B1Q

Convierte bloques consecutivos de 2 bits en un pulso de 4 niveles posibles para ser transmitidos a través de la linea de abonado, como resultado de esto la velocidad de símbolos transmitidos (Bauds) se reduce a la mitad de la velocidad de transferencia de información (Bps). Dado que todos los posibles símbolos que proporciona el código son utilizados, se dice que es un código saturado, es decir, 4 posibles valores son representados mediante 2 bits y un símbolo cuaternario solo tiene 4 posibles niveles o valores. (Véase fig 12)

4 BITS SE CONVIERTEN EN
3 SIMBOLOS TERNARIOS



4 BITS. 160 Kbps

3 SIMBOLOS TERNARIOS. 120 Kbauds

TABLA DE CODIFICACION			
BITS	CERO O NEGATIVO	POSITIVO	DISPARIDAD
0000	+ 0 -	+ 0 -	0
0001	- + 0	- + 0	0
0010	0 - +	0 - +	0
0011	+ - 0	+ - 0	0
0100	+ + 0	- - 0	2
0101	0 + +	0 - -	2
0110	+ 0 +	- 0 -	2
0111	+ + +	- - -	3
1000	+ + -	- - +	1
1001	- + +	+ - -	1
1010	+ - +	- + -	1
1011	+ 0 0	- 0 0	1
1100	0 + 0	0 - 0	1
1101	0 0 +	0 0 -	1
1110	0 + -	0 + -	0
1111	- 0 +	- 0 +	0

NOTA: DONDE EXISTE CODIFICACION ALTERNATIVA, LA SELECCION DE LA SECUENCIA SE REALIZA DE TAL FORMA QUE LA DISPARIDAD SE MANTENGA AL MINIMO.

EJEMPLO DE CODIFICACION

TREN DE BITS		0101	1011	1110	1000	0010
SEÑAL ENVIADA		0 + +	- 0 0	0 + -	- - +	0 - +
DISPARIDAD	0	2	1	1	0	0

Fig. 13. Código de linea 4B3T para la interfase U.

Código de linea 4B3T

Este código tiene una compresión menor de velocidad de símbolos (Bauds) que el 2B1Q, por que utiliza señales de 3 niveles en lugar de señales de 4 niveles. Otro factor que no permite bajar más la velocidad de símbolos es que los 16 posibles valores generados por 4 bits son representados mediante 27 posibles combinaciones de 3 símbolos ternarios para ser transmitidos por la linea de abonado. Las 11 combinaciones restantes pueden ser utilizados para otras funciones del código, a lo que se le conoce como código no saturado. (Véase fig. 13)

TIPOS DE CANALES PARA EL TRANSPORTE DE INFORMACIÓN EN RDSI

La información en la interfase usuario-red se transmite entre la NT y el TE a través de "canales". Un canal es una porción especifica del ancho de banda total de la linea de transmisión. Las normas RDSI definen varios canales , pero los más usados son los canales B y D.

Canal B

El canal B (canal de portadora) es un canal digital de 64 Kbps. Este canal no lleva información de señalización; sino, lleva información como voz o datos en conmutación de circuitos o conmutación de paquetes

Canal D

El canal D es un canal separado y su uso es principalmente para transportar información de señalización. Este canal puede ser de 16 Kbps o 64 Kbps. La información de señalización establece, mantiene, y termina la conexiones en la red RDSI.
La naturaleza de las funciones señalización causa que la información de señalización se genere en forma de ráfagas; por lo tanto, cuando el canal D no lleva información de señalización, se puede transmitir información de usuario en conmutación de paquetes, sobre el canal D.

Tabla 1. Canales B y D en la RDSI

TIPO DE CANAL	VELOCIDAD DE TX	USO
B	64 Kbps	Datos o voz en conmutación de circuitos o de paquetes
D	16 Kbps ó 64 Kbps	Información de Señalización para los canales B o información de usuario en conmutación de paquetes, cuando hay no señalización

VELOCIDADES DE ACCESO A LA RDSI

Las Normas RDSI definen el acceso del usuario a la RDSI a través de canales B y D para crear diferentes configuraciones de canales. Estos configuraciones de canal puede pensarse como tubos: Cada tubo lleva varios canales los cuales están multiplexados en tiempo sobre la linea de transmisión. El dos principales configuraciones son la Interfase de Acceso Básico (BRI) y la Interfase de Acceso Primario (PRI). También son conocidas como Acceso Básico (BA) y Acceso Primario (PA)

INTERFASE DE ACCESO BÁSICO (BRI)

Un BRI consiste de dos canales B (64 Kbps cada uno) y un canal D (16 Kbps), el cual es conocido como 2B+D y tiene una capacidad para transportar información de 144 Kbps (64 k+ 64 k+ 16 k). Con bits adicionales de overhead (control) , la velocidad total en la interfase S es de 192 Kbps. El dos canales B pueden usarse independientemente para tipos diferentes tipos de transmisión. Para ejemplo, un canal B puede llevar información de voz y el otro puede llevar datos. De esta manera, voz y datos son integrados sobre los mismos medios de transmisión. .

INTERFASE DE ACCESO PRIMARIO (PRI)

Actualmente, existen dos tipos de Accesos Primarios definidos. En EE.UU., Corea de Sur, y Japón, el PRI es de 1.544 Mbps (23 canales B y 1 canal D a 64 Kbps cada uno más un overhead de 8 Kbps). El PRI Europeo usa 30 canales B y 1 canal D a 64 Kbps cada (más un overhead de 64 Kbps) para una velocidad total de 2 048 Mbps El overhead para ambos PRIs sirve para funciones tales como sincronización de trama y administración de red.

PROTOCOLOS RDSI

Además del equipo, puntos de referencia, y configuraciones de los canales de la interfase usuario-red de la RDSI se han definido los protocolos para la transmisión de datos y funciones de administración. Las normas de RDSI se han desarrollado siguiendo el modelo OSI de siete capas .Las Series I del CCITT describe los protocolos para las primeras tres capas de la RDSI. Hay también números equivalentes en la Serie Q para protocolos de algunas de las capas.

El modelo OSI describe el proceso de comunicacion entre capas, las cuales están formadas por diferentes Entidades. Durante un proceso de comunicación, entidades de la misma capa pero en sistemas diferentes (por ejemplo, en diferentes extremos de una RDSI), éstas deben intercambiar información. Las cuales son llamadas *entidades par* Las entidades par se comunican por medio de las capas inferiores de sus sistemas respectivos Para llevar a cabo esto, las capas adyacentes del mismo sistema interactúan en sus límites comunes de tal forma que las capas inferiores proporcionan servicios a capas superiores Por ejemplo, los servicios usados por la capa 3 están compuestos de los servicios de la capa 2 y de los servicios que provee la capa 1 a la capa 2.

Aplicando estos principios a la comunicación entre dos puntos extremos de una red RDSI, capas adyacentes en el lado originante agregan información de protocolo a la información de usuario que va ha ser enviada. En la capa fisica (capa 1), la información compuesta es enviada sobre el mismo medio de transmisión. En el lado receptor, la información apropiada de protocolo es extraída e interpretada por cada capa. La información sobrante se pasa al próximo nivel superior hasta que la información original de usuario alcanza su destino

Es importante notar que las capas y protocolos involucraron en una transacción particular pueden ser diferentes durante la fase de señalización y la fase de transferencia de información. También, diferentes piezas de equipo RDSI pueden proveer las funciones para una capa dada, dependiendo de los tipos de equipo usados en la configuración particular de la interfase usuario-red.

Generalmente, las funciones de las capas 1 a 3 de la RDSI se construyen una sobre la otra (véase fig. 14) y realizan las siguientes funciones:

CAPA O NIVEL	MODELO OSI	CANALES B	CANAL D
7	APLICACION	DEFINIDOS POR EL USUARIO	SIN DEFINIR TODAVIA
6	PRESENTACION		
5	SESION		
4	TRANSPORTE		
3	RED		I 451 Y TAMBIEN X 25
2	ENLACE DE DATOS		I 441 (Q 921) LAPD
1	FISICO	I 431 PRI I 430 BRI	I.431 PRI I.430 BRI

Fig. 14 Protocolos en la interfase S de la RDSI.

CAPA 1 (CAPA FÍSICA)

La capa 1 determina la características de la transmisión física en un enlace nodo a nodo. Por ejemplo, define el conector físico, las fuentes de alimentación , el código de linea, los niveles de voltaje y la forma de activación y desactivación de la interfase para proveer las características de transmisión necesarias y poder enviar la información sobre el medio de transmisión físico.

En las figuras 15 a 18 se muestran algunas de las características del nivel físico de la interfase S

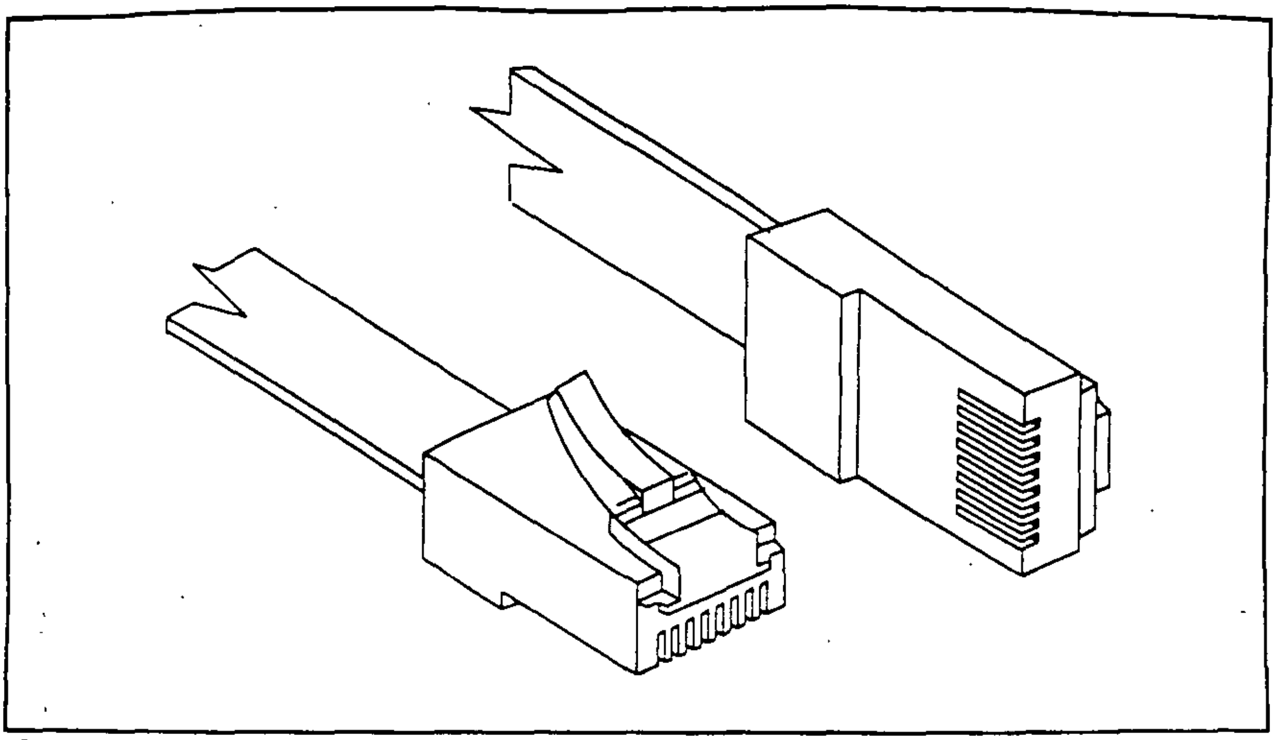


fig 15 Conector ISO-8877 (RJ45) para la interfase S.

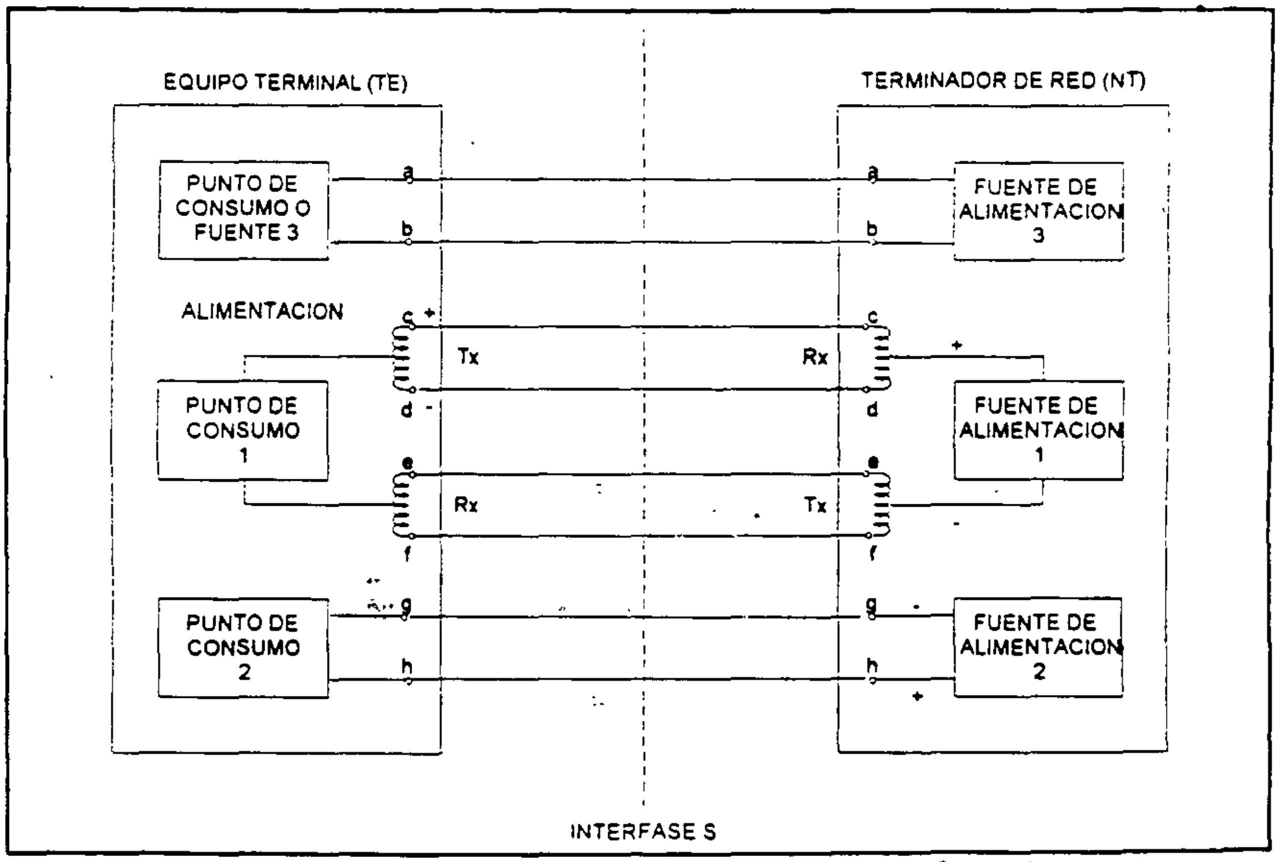


Fig. 16. Fuentes de energia y puntos de consumo en el nivel 1 de la interfase "S".

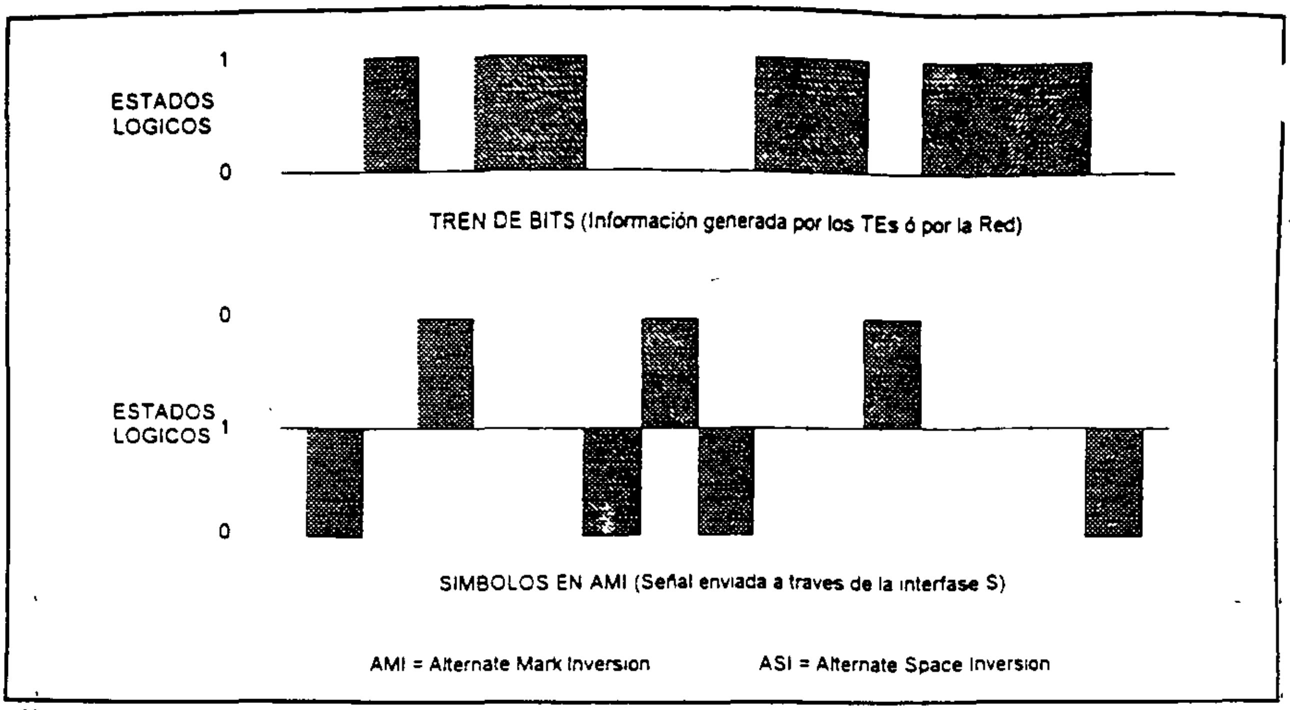


Fig. 17. Código de linea AMI modificado (ASI) usado en la interfase S.

SEÑALES DE NT –> TE	SEÑALES DE TE –> NT
INFO 0 AUSENCIA DE SEÑAL (TRANSMISION DE CEROS BINARIOS)	INFO 0 AUSENCIA DE SE .AL (TRANSMISION DE CEROS BINARIOS)
	INFO 1 SEÑAL CONTINUA CON EL SIGUIENTE PATRON. CERO PSITIVO. CERO NEGATIVO Y SEIS UNOS CONSECUTIVOS. 
INFO 2 TRAMA CON LOS BITS DE LOS CANALES B.D y ECO DEL CANAL D (E) PUESTOS A CERO EL BIT A ES PUESTO A CERO BINARIO. BITS N y L PUESTOS DE ACUERDO A LAS REGLAS DE CODIFICACION NORMALES. 	
	INFO 3 TRAMAS SINCRONIZADAS CON DATOS OPERACIONALES EN LOS CANALES B y D
INFO 4 TRAMAS CON DATOS OPERACIONALES EN EN LOS CANALES B. D y ECO DEL CANAL D (E). BIT A PUESTO A UNO BINARIO.	

Fig. 18. Señales INFO, para la activación y desactivación del nivel fisico de la interfase S

CAPA 2 (CAPA DE ENLACE DE DATOS)

Esta capa lleva la información de la capa 1 y aplica las funciones necesarias para asegurar que la transmisión este libre de errores en cada enlace de la trayectoria de transmisión. La detección y corrección de errores son realizados por el protocolo de capa 2 en cada enlace entre nodos.

CAPA 3 (CAPA DE RED)

La capa 3 define cómo se arma la trayectoria completa de comunicaciones usando los enlaces con protección contra errores proporcionados por la capa 2. La capa 3 usa un protocolo de señalización para determinar la trayectoria o ruta dentro de la red para transportar la información.

Hay que recordar que los canales B llevan solamente información de usuario (aunque hay una variedad de información de usuario, tal como voz, datos, video y facsímile) Por esta razón, el único protocolo especificado para el canal B es la capa física (capa 1) Si la configuración de canales es de un BRI (Interfase de Acceso Básico), el protocolo es I 430. Si la configuración de canales es de un PRI (Interfase de Acceso Primario), el protocolo es I.431. Los niveles restantes del modelo OSI (capas 2 a 7) son definidos por el usuario para el canal B

Dado que los canales D pueden llevar información de señalización o información de usuario y la información de señalización debe de controlar todo el tráfico en el canales B, los protocolos de canal D son más detallados y complejos. La capa 1 del canal D es la misma capa 1 del canal B I 430 para un BRI y I.431 para un PRI. Esto es porque los canales B y D están multiplexados en tiempo sobre la misma linea de transmisión física.

Los niveles 2 y 3 están especificados de tal forma que la señalización se puede realizar en cualquier tipo de interfase en una forma normalizada. La recomendación I.441 del CCITT (Q 921) define la Capa 2.

Este protocolo de capa 2 es también comúnmente conocido como LAPD (Procedimiento de Acceso al Enlace en el canal D). LAPD es semejante al protocolo LAPB usado en X.25 salvo que permite enlaces lógicos múltiples entre puntos extremos. Esta capacidad es necesaria porque el protocolo de capa 2 tiene que proveer los servicios de transporte de nivel de enlace de datos tanto para señalización como para información de usuario al nivel 3. LAPD usa una estructura de trama como el protocolo HDLC. El nivel 3 para el canal D es especificado en la recomendación I 451 (Q 931) del CCITT. X.25 se puede también usar.

El protocolo de señalización del canal D controla el trafico del usuario en los canales B entre el interfase usuario-red y la central RDSI.

Sin embargo, entre las centrales RDSI, se usa protocolo de señalización por canal comun (CCITT#7).

PROTOCOLOS DE CAPA 1 PARA LOS CANALES B Y D

Las recomendaciones del CCITT I.430 (para un BRI) e I.431 (para un PRI) especifican las características físicas de la interfase usuario-red en los puntos de referencia S y T. Estos protocolos de nivel 1 proveen los siguientes servicios al nivel 2:

- Funciones de sincronización y temporización en los canales B y D.

- Los procedimientos necesarios para la activación y desactivación del TE o de la NT.

- Los procedimientos necesarios para permitir a los equipos terminales ganar el acceso al canal D de señalización en una forma ordenada.

- Procedimientos de capa 1 necesarios para realizar funciones de mantenimiento

- Indicación del estado de la capa 1 a las capas superiores

- Capacidad de transferencia de información en modo multipunto a punto así como de Punto a punto.

En las premisas del usuario (interfase usuario-red), la información de usuario y de señalización es transmitida en tramas sobre los cuatro hilos de la linea de transmisión de las interfases S y T a la central RDSI. La estructura de estas tramas depende del tipo de acceso (BRI o PRI)

ESTRUCTURAS DE TRAMA DEL ACCESO BÁSICO EN LOS PUNTOS DE REFERENCIA S ó T

Recordemos que el Acceso Básico consiste de dos canales B (información de usuario a 64 Kbps cada uno) y un canal D (información de señalización o de usuario a 16 Kbps), los cuales son multiplexados en tiempo sobre los cuatro hilos de la interfase S. Un par de hilos es usado para transmitir y el otro par es usado para recibir.

Existen dos tipos de tramas para el Acceso Básico:

- Un tipo de tramas es transmitido del TE al NT (dirección de usuario a central) y

- Otro tipo de tramas es transmitido de la NT al TE (dirección central a usuario)

La sincronía de trama para las tramas de TE a NT es derivada de las tramas de NT a TE, pero con 2 bits de defasamiento (offset), véase fig. 19

Ambos tipos de tramas consisten de 48 bits transmitidas cada 250 microsegundos (4,000 tramas por segundo) Esto equivale a una velocidad de transmisión total de 192 Kbps; sin embargo, algunos de los 48 bits (12 bits) son de overhead (bits adicionales de control) y no de informació de los canales B o D. Los 36 bits de información de los canales B y D son usados como sigue: 16 bits son del primer canal B, 16 bits son del segundo canal B, y cuatro bits del canal D. Esto resulta en una transferencia de datos a una velocidad de 144 Kbps (36 bits x 4,000 tramas por segundo)

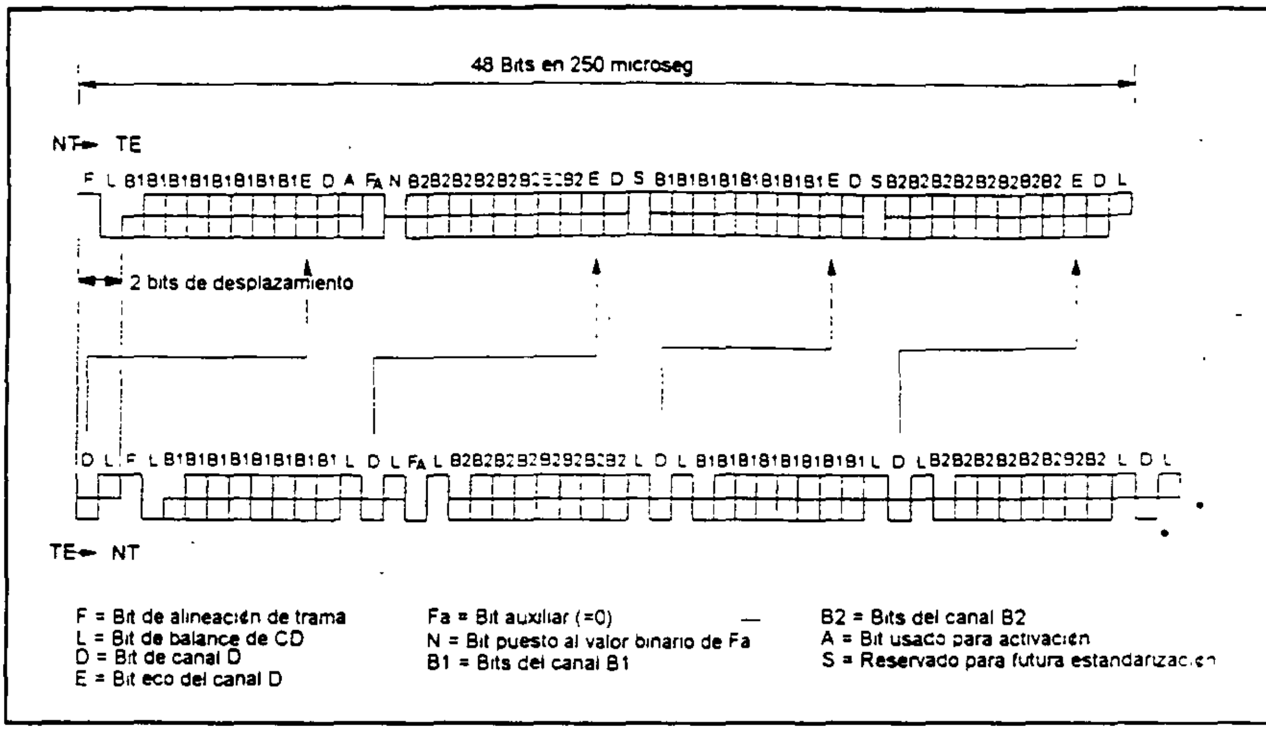


Fig. 19 Trama del Nivel 1 de la interfase S

Aunque ambos tipos de tramas tienen 12 bits de overhead (control), algunos bits son usados dependiendo del tipo de trama. Por ejemplo, dado que un BRI puede ser configurado en punto a punto o punto a multipunto, alguno de los bits de control en las tramas de NT a TE son usados para controlar el acceso al canal D.

Los bits de control que son únicos en las tramas de NT a TE se describen a continuación.

bits A (activación/desactivación)
bits de Activación/desactivación permiten a un equipo terminal estar en linea (activado)
También pueden ponerlo fuera de linea (desactivado) en modo de baja potencia de consumo cuando ni transmite ni recibe.

bits E (eco)
Estos bits de eco regresan los valores de los bits del canal D anteriormente transmitidos en la trama de TE a NT. Son usados para controlar el acceso al Canal D.

bits M
bits de Multitrama.

bits S
bits del canal S.

MÉTODO DE ACCESO AL CANAL D.

Cuando un TE, hace uso del canal D éste transmite su información de señalización en las ranuras correspondientes al canal D en la trama de nivel 1, en la dirección TE a NT, y la NT le envia un eco (le regresa el mismo valor) en la posición del próximo bit E. El TE espera en la posición del próximo bit E recibir el mismo valor (eco) del último bit del canal D enviado. Si no así, el TE supone que en el canal D ha ocurrido una colisión y deja de transmitir Entonces tiene que esperar para volver a usar el canal D, siguiendo las reglas del método de control para el Acceso al canal D, el cual se describe a continuación

Los TEs deben observar los bits E que vienen de la NT. Un cierto número de bits E continuos con un valor binario de 1 indica que el canal D está libre El número especifico de 1s continuos en la posición del canal E que un TE tiene que ver antes de transmitir depende si el TE quiere transmitir información de señalización o de usuario sobre el canal D.

La información de señalización tiene la prioridad alta; por lo tanto, se necesita menos bits con valor 1 continuos en la posición E para poder transmitir información de señalización.
Inicialmente (esto es, para la primera trama enviada por un TE), el número de 1s continuos en los bits E que un TE tiene que ver son 8 para enviar información de señalización y 10 para enviar información de usuario. Después de, transmitir en forma exitosa una trama de capa 2 (esto lleva más de una trama de capa 1), el número de 1s continuos en los bits E que un TE específico tiene que ver debe ser incrementado en uno (tanto para señalización como para información de usuario). Esto permite a otro TE tener acceso al canal D.

Si un TE en particular no tiene información que enviar sobre el canal D, transmite 1s binarios, permitiendo que el proceso anteriormente descrito se lleve a cabo. Una vez que todos los TEs han usado el canal D, el número de bits E es decrementado a su nivel original.

ESTRUCTURAS DE TRAMA DE LA INTERFASE S ó T DEL PRI.

Como se mencionó al inicio, se han definido 2 estándares uno es: el 1.544 Mbps par EE.UU., Corea del Sur, y Japón (23 canales B a 64 Kbps cada uno, más un canal D a 64 Kbps, más overhead) y el otro es el estándar Europeo, también utilizado en México de 2.048 Mbps (30 canales B a 64 Kbps, más un canal D a 64 Kbps, más overhead). A diferencia del Acceso Básico

(que pueda ser usado en configuraciones punto a punto o punto a multipunto), ambos tipos de PRIs son pensados para solamente operación en modo punto a punto.
En operación punto a punto, el PRI permanentemente está activado y no necesita bits de control para activación/desactivación del nivel fisico ni un método para el uso del canal D. Dado que estas funciones no se requieren, las tramas del PRI en ambas direcciones tienen el mismo formato.

Sin embargo existen dos tipos tramas para el PRI. Una trama para el de 1.544 Mbps, y otro tipo de trama para el de 2 048 Mbps. (Véase fig. 20)

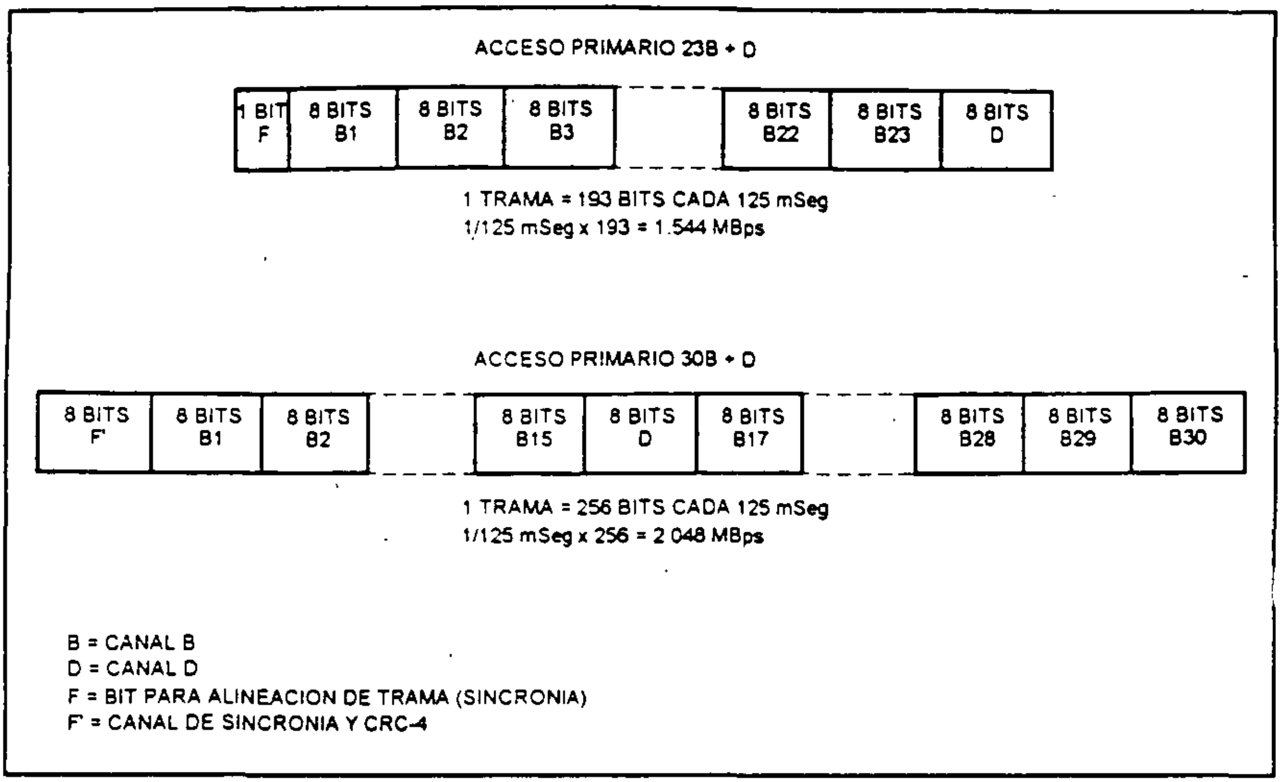


Fig. 20. Estructura de las tramas de Nivel 1 de la Interfase S/t del Acceso Primario.

Para el Acceso a 1.544 Mbps, la trama consiste de 193 bits transmitidos cada 125 microsegundos (8,000 tramas por segundo). Esto da una velocidad total de 1.544 Mbps; sin embargo, la velocidad real de transferencia de datos velocidad es 1.536 Mbps porque uno de los 193 bits es usado para sincronia. Los restantes 192 bits se dividen en 24 ranuras de tiempo, cada una de ocho bits de longitud. Veintitrés de las ranuras de tiempo son para canales B y la ranura restante es para el canal D.

El formato para el Acceso de 2.048 Mbps es semejante al formato de 1.544 Mbps. Estas tramas son transmitidas también cada 125 microsegundos, pero consisten de 256 bits que son divididos en 32 ranuras de tiempo, con una longitud de ocho bits cada una. La ranura cero se usa para sincronía, las ranuras 1 a 15 y 17 a 31 son usadas para los 30 canales B, y la ranura 16 es usada para el canal D. Dado que hay ocho bits de sincronia por trama, la velocidad real de transferencia de datos es de 1.984 Mbps (248 bits x 8000 tramas por segundo).

PROTOCOLO DE SEÑALIZACIÓN DE CAPA 2 DEL CANAL D

Las recomendaciones del CCITT I.430 y I.431 definen el nivel fisico para los Accesos Básico y Primario. y la capa 2 para el canal D es definida en las recomendaciones I.440 (Q.920) e I.441 (Q.921). La recomendación I.440 (Q.920) describe en forma general la capa 2 de RDSI, y La recomendación I.441 (Q. 921) define en forma detallada el nivel 2.

Otro nombre más común para este protocolo es LAPD (Procedimiento de Acceso al enlace en el canal D). Su propósito es controlar el intercambio de información entre las entidades pares de capa 3 a través de la interfase de usuario-red. También controla las interacciones entre el enlace de datos (capa 2) y la capa de red (capa 3) y entre la capa 2 y la capa fisica (capa 1). Para llevar a cabo esto, la capa 2 provee servicios a la capa 3 y recibe servicios de capa 1. Se le conoce como Punto de Acceso al Servicio (SAP) al punto donde la capa 2 proporciona servicios a la capa 3. LAPD puede asociar más de una entidad de capa 3 con una SAP

Para el intercambio de información entre dos o más entidades de capa 3 , una asociación debe ser hecha entre entidades de la capa 3 por el protocolo de capa 2 A esta asociación se le conoce como conexión de enlace de datos

Éstas son algunas de las funciones de LAPD

- Es independiente de la velocidad de transmisión de la capa 1.

- Permite la operación de múltiple equipo terminal en la interfase usuario-red.

- Proporciona para múltiples entidades de nivel 3 (y, por lo tanto, combinaciones múltiples de puntos extremos de enlace de datos) Diferentes conexiones son identificadas mediante el DLCI (Identificador de conexión del enlace de datos) para cada trama de LAPD.

- La delimitación de tramas se realiza mediante el uso de banderas (01111110) y la transparencia a través de la técnica conocida como "relleno de bits" como la usada en el protocolo HDLC. (La técnica de relleno de Bits consiste básicamente en insertar un 0 en la secuencia de bits de datos cuando una serie de cinco 1s es detectada dentro de la trama para impedir que la secuencia de bits se confunda con una bandera)

- Efectúa un control de la secuencia para mantener en orden las tramas a través la conexión de enlace de datos

- Proporciona detección de y recuperación de errores en la conexión de enlace de datos.

- Efectúa Control flujo .

Hay dos tipos de servicios de transferencia de datos que proporciona LAPD: con acuse y sin acuse de recibo.

Sin acuse de recibo, la información de capa 3 se transfiere sin esperar una respuesta del lado receptor. Éste es el método más rápido, pero no provee control sobre la secuencia de las tramas transmitidas para corrección de errores (determinar cuando una trama necesita ser retransmitida)
Existen dos formas del servicio con acuse de recibo: operación de una sola trama y operación de multitrama.

El servicio acuse de recibo permite controlar el orden de las tramas mediante la numeración de las tramas. También provee control de errores dando acuse de recibo para tramas transmitidas de manera exitosa y pidiendo retransmisión de las tramas con errores.
Este servicio es usado solamente en configuraciones de punto a punto.

ESTRUCTURA DE LA TRAMA DEL PROTOCOLO LAPD.

La estructura de la trama para comunicación entre entidades pares a través de una conexión de enlace de datos consta de cinco o seis campos, siendo el de información el único campo opcional. La trama tiene la estructura mostrada en la fig 21

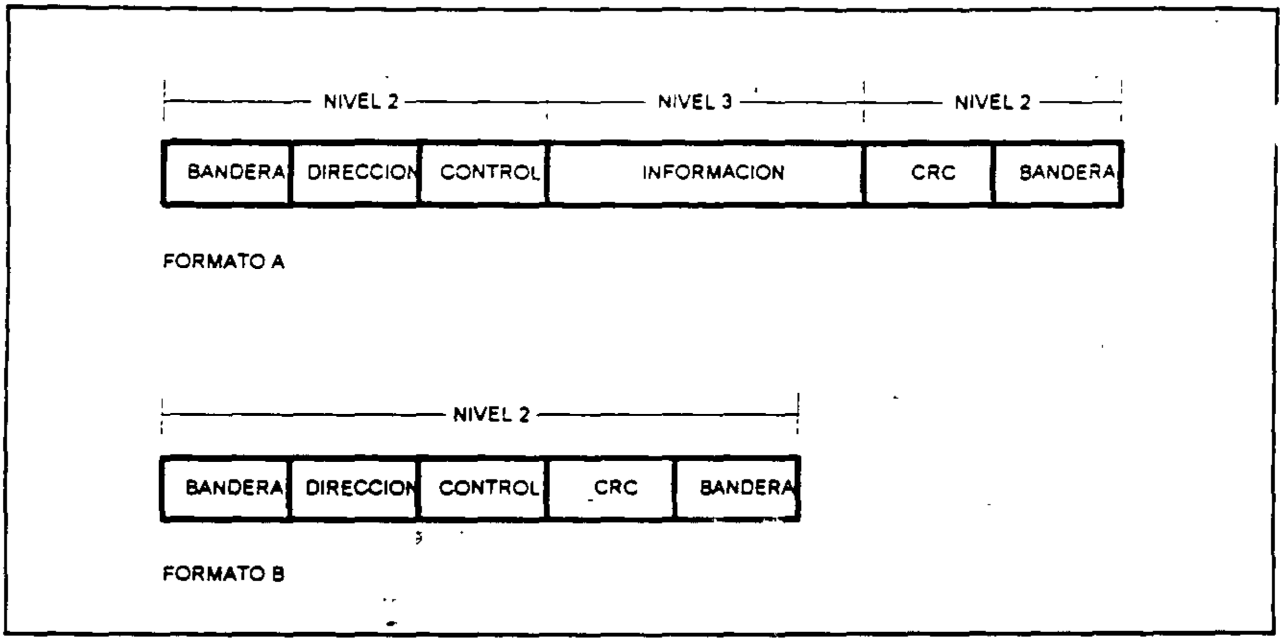


Fig. 21. Tipos de Formato de la trama de Nivel 2 de la interfase S/T.

DESCRIPCIÓN DE LOS CAMPOS DE LA TRAMA. (Véase fig. 22)

Campo de bandera.
Todas las tramas deben iniciar y terminar con un campo de bandera. A la bandera que indica el inicio de la trama se le conoce como bandera de apertura, mientras que a la bandera que indica el fin de la misma se le conoce como bandera de cierre, aunque en algunas ocasiones, ésta también sirve para indicar la apertura de la siguiente trama. El campo de bandera contiene una codificación única, la cual consiste de una secuencia de un cero, seguida por seis unos consecutivos y finalizando con otro cero (01111110).

Campo de dirección.
El campo de dirección identifica al receptor destino de una trama instrucción y al transmisor que Bit de extensión del campo de dirección (EA).

Bit EA
Este bit sirve para indicar que dentro de un campo de dirección existen octetos adicionales colocándolo con un valor de 0, cuando se pone a un valor de 1 se indica que ese octeto es el octeto final del campo de dirección. Para nuestro caso, debido a que el campo de dirección es de dos octetos, el primer bit EA se coloca en un valor de 0 , mientras que el segundo se coloca en un valor de 1.

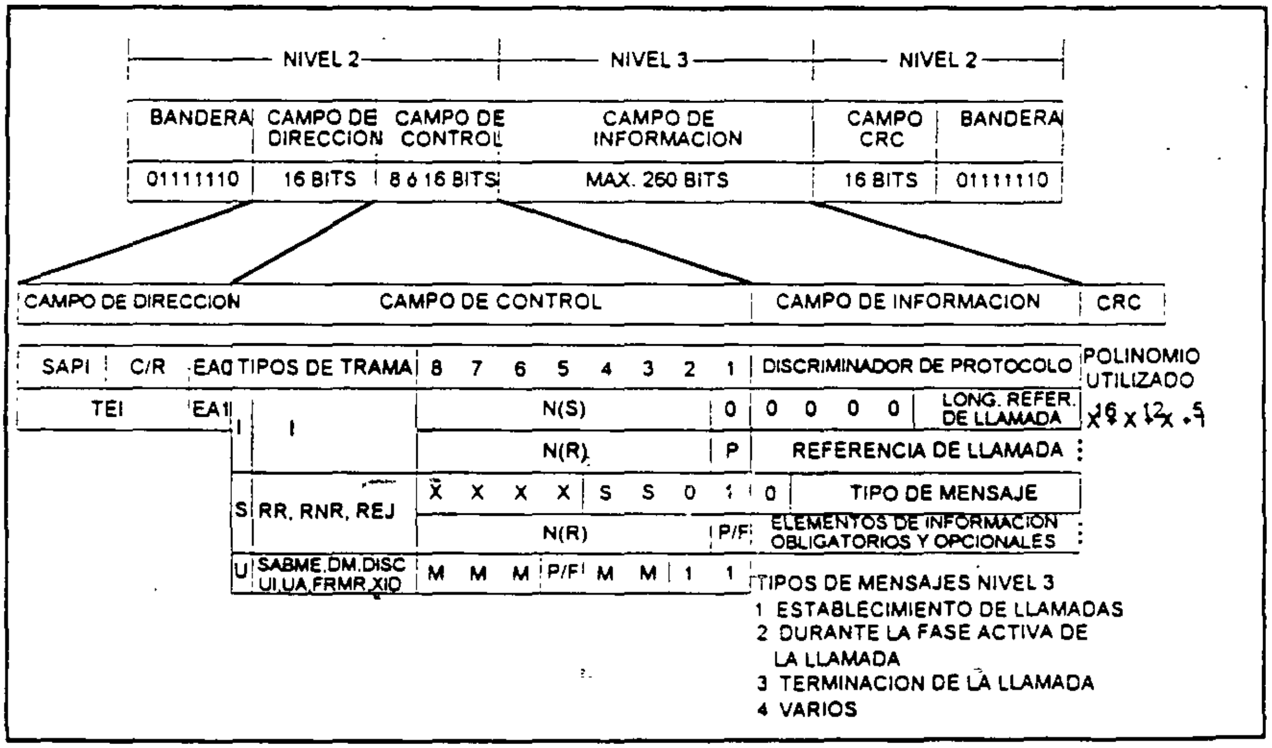


Fig. 22. Trama de Nivel 2 del canal D.

Bit de campo para instrucción/respuesta (C/R).

El bit C/R sirve para identificar una trama como instrucción o respuesta. De acuerdo a las reglas establecidas en el Protocolo HDLC, cuando se desea enviar una instrucción se utiliza la dirección de la entidad de enlace de datos que lo va a recibir, mientras que cuando se trata de una respuesta se utiliza la dirección de la entidad de enlace de datos que la genera.

La fig. 23 muestra los valores que utilizan tanto el lado usuario como el lado red para cualquiera de los casos:

Identificador de punto de acceso al servicio (SAPI).

El SAPI identifica el punto a través del cual una entidad de nivel de enlace de datos proporciona servicios a una entidad de nivel 3 o a una entidad de gestión de capa. En consecuencia el SAPI especifica una entidad de nivel 2 que debe procesar una trama de nivel de enlace de datos y también una entidad de nivel 3 o de gestión de capa que recibirá la información llevada en dicha trama. El subcampo del SAPI consta de 6 bits (del 3 al 8), lo cual permite un total de 64 valores (de 0 a 63), de los cuales solo cuatro están especificados de acuerdo a la fig. 23, quedando los restantes para futura estandarización

Identificador de punto extremo terminal (TEI).

Un TEI para una conexión de enlace de datos puede estar asociado con un solo equipo terminal Un equipo terminal puede contener varios TEIs usados para transferencia de datos punto a punto El TEI para conexiones de enlace de datos en difusión está asociado con todas las entidades de nivel de enlace de datos conteniendo el mismo SAPI. El subcampo de TEI es de 7 bits, lo cual permite hasta 128 valores posibles (de 0 a 127), los cuales están asignados de la manera siguiente:

TEI para conexiones de enlace de datos en difusión.

El valor de TEI para este tipo de conexiones es de 127 (111 1111 en binario), también se le llama TEI de grupo. Dicho TEI es asignado a la conexión de enlace de datos en difusión asociada con el punto de acceso al servicio SAP direccionado.

TEI para conexiones de enlace de datos punto a punto.

El resto de los valores de TEI se utilizan para conexiones de enlace de datos punto a punto asociadas con el SAP direccionado.

Los valores no-automáticos son seleccionados por el usuario, y su asignación es responsabilidad de él mismo. Los valores automáticos son seleccionados por la red, y de igual forma la asignación es responsabilidad de la red.

La asignación de valores de TEI se muestra en la fig 23

CONTENIDO DE LA TRAMA	SENTIDO DE TRANSMISION	VALOR DEL BIT C/R
INSTRUCCION	RED -> USUARIO	1
	USUARIO -> RED	0
RESPUESTA	RED -> USUARIO	0
	USUARIO -> RED	1

SAPI	FUNCION
0	PROCEDIMIENTOS DE CONTROL DE LLAMADA (SEÑALIZACION)
1	COMUNICACIONES EN MODO PAQUETE DE ACUERDO A LOS PROCEDIMIENTOS DE CONTROL DE LLAMADA DE LA REC. Q.931
16	COMUNICACIONES EN MODO PAQUETE DE ACUERDO A LOS PROCEDIMIENTOS DEFINIDOS EN EL NIVEL 3 DE LA REC. Q.931
63	PROCEDIMIENTOS DE GESTION DE CAPA 2
2-15 Y 17-62	PARA APLICACIONES FUTURAS

TEI	FUNCION
0-63	EQUIPO DE USUARIO CON ASIGNACION TEI NO AUTOMATICO (ASIGNACION POR EL USUARIO)
64-126	EQUIPO DE USUARIO CON ASIGNACION DE TEI AUTOMATICO - (ASIGNACION POR LA CENTRAL)
127	EQUIPO DE USUARIO CON CUALQUIER VALOR DE TEI (INFORMACION EN DIFUSION Y ASIGNACION DE LOS TEI 64 A 126)

SAPI = Identificador del Punto de Acceso al Servicio

TEI = Identificador del punto Extremo Terminal

BIT C/R = Indica si la trama es una Instruccion o una Respuesta

Fig. 23. Valores de C/R, SAPIs y TEIs en el nivel 2 del canal D (Campo de Dirección)

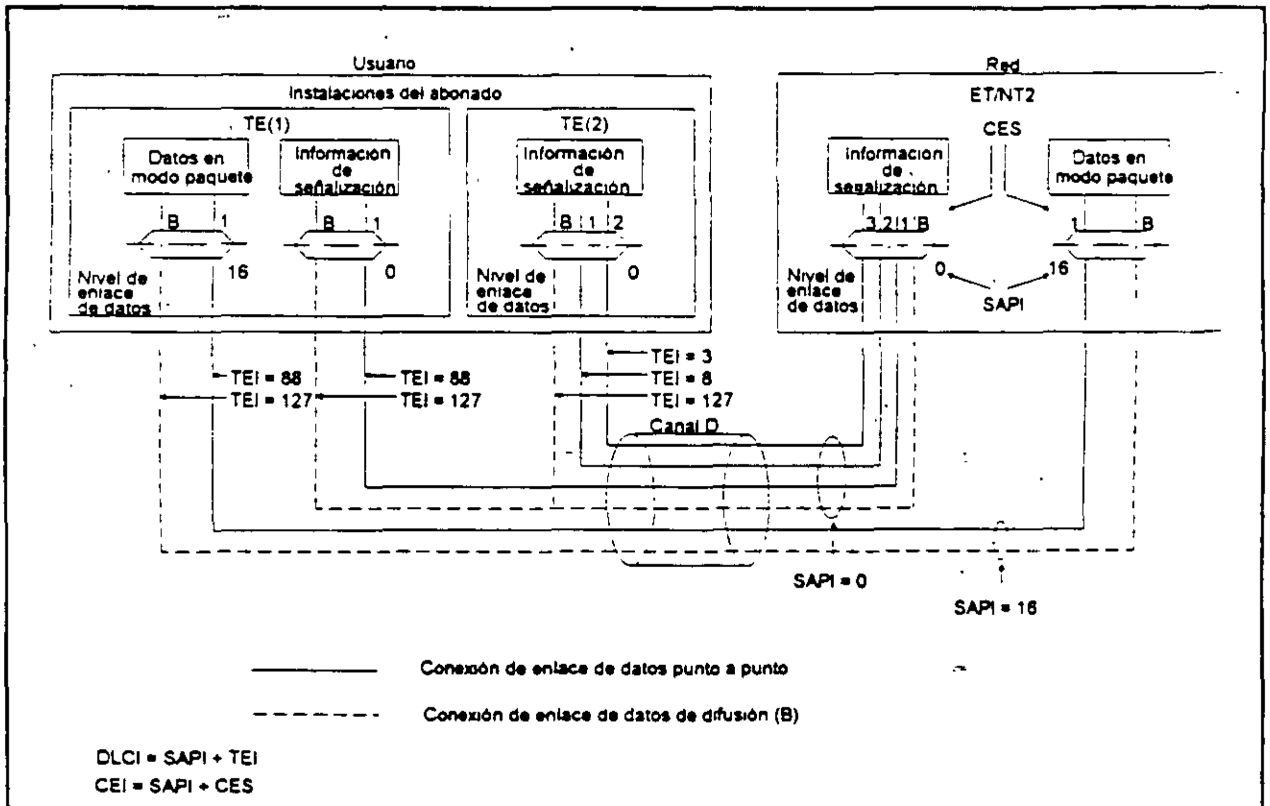


Fig. 24. Descripción general de la relación entre SAPI, TEI y DLCI.

Campo de control.
El campo de control identifica el tipo de trama. El campo de control puede ser de uno o do octetos dependiendo del formato. Los formatos del campo de control se muestran en la fig. 25.

Tres tipos de formatos de campo de control son especificados:

APLICACION	FORMATO	INSTRUCCION/ COMANDO	RESPUESTA	CODIFICACION 8 7 6 5 4 3 2 1
TRAMAS SIN ACUSE DE RECIBO Y MODO MULTITRAMA CON ACUSE DE RECIBO PARA TRANSFERENCIA DE INFORMACION	TRANSFERENCIA DE INFORMACION	I (INFORMATION)		N(S) 0 N(R) P
	SUPERVISION	RR (RECEIVE READY)	RR (RECEIVE READY)	0 0 0 0 0 0 0 1 N(R) P/F
		RNR (RECEIVE NO READY)	RNR (RECEIVE NO READY)	0 0 0 0 0 0 0 1 N(R) P/F
		REJ (REJECT)	REJ (REJECT)	0 0 0 0 0 0 0 1 N(R) P/F
	NO NUMERADAS	SABME (SET ASYNCHRONOUS BALANCE MODE EXTENDED)		0 1 1 P 1 1 1 1
			DM (DISCONNECTED MODE)	0 0 0 F 1 1 1 1
		UI (UNNUMBERED INFORMATION)		0 0 0 P 0 0 1 1
		DISC (DISCONNECT)		0 1 0 P 0 0 1 1
			UA (UNNUMBERED ACKNOWLEDGEMENT)	0 1 1 F 0 0 1 1
			FRMR (FRAME REJECT)	1 0 0 F 0 1 1 1
ADMINISTRACION DE LA CONEXION		XID (EXCHANGE IDENTIFICATION)	XID (EXCHANGE IDENTIFICATION)	1 0 1 P/F 1 1 1 1

Fig. 25. Comandos y respuestas del Nivel 2 del canal D (Campo de control).

Transferencia de información numerada (Formato I). Este formato debe ser usado para llevar a cabo una transferencia de información entre entidades de nivel 3. Cada trama I tiene un número de secuencia N(S), un número de secuencia N(R) mediante el cual se puede o no efectuar un reconocimiento de tramas I adicionales recibidas por la entidad de nivel 2, y un bit P que puede ser puesto a un valor de 0 o 1.

Funciones de supervisión (Formato S). Este formato debe ser usado para llevar a cabo funciones de control de supervisión de enlace de datos como son: reconocimiento de tramas I, solicitud de retransmisión de tramas I y solicitud de una suspensión temporal de transmisión de tramas I Las funciones de N(R) y P/F son independientes, es decir, cada trama de supervisión tiene un número de secuencia N(R) mediante el cual se puede o no efectuar un reconocimiento de tramas I adicionales recibidas por la entidad de nivel 2, y un P/F bit que puede ser puesto a un valor de 0 o 1.

Transferencia de información no numerada y funciones de control (Formato U). Este formato puede ser usado para proporcionar funciones de control de enlace de datos adicionales y para realizar transferencia de información sin acuse de recibo. Este formato no contiene números de secuencia. Incluye un bit P/F que puede ser puesto a un valor de 0 o 1.

Bit Poll/Final (P/F).

Todas las tramas independientemente de su tipo contienen un bit P/F. Este bit proporciona una función tanto en tramas de instrucción como de respuesta. El bit P puesto a 1 es usado por la entidad de nivel 2 para solicitar una trama de respuesta de la entidad de nivel 2 par. El bit F puesto a 1 es usado por la entidad de nivel 2 para indicar la trama de respuesta transmitida como resultado de la recepción de una instrucción con el bit P puesto a 1.

Campo de información.

Este campo es opcional, y aparece dentro de la trama solo cuando se transfiere información con o sin acuse de recibo. Este campo consta de un número entero de octetos que no puede exceder el valor de 260.

Este campo puede ser generado por:

El nivel 3

Lo genera cuando requiere transferir información de señalización sobre las características del enlace que se va a establecer.

estión de capa

Lo genera cuando se requiere de algún procedimiento de administración de TEI (asignación, prueba y supresión).

Secuencia de verificación de trama (FCS).

El campo de la secuencia de verificación de trama FCS debe ser una secuencia de 16 bits. Esta secuencia es calculada de la siguiente manera:

El complemento a unos de la suma en módulo 2 de los residuos de las siguientes divisiones:

a) El residuo de la división módulo 2 de:

$$\frac{(X^k)\,(X^{15}+X^{14}+\ldots\ldots+X^2+X^1+1)}{X^{16}+X^{12}+X^5+1}$$

b) El residuo de la división módulo 2 de:

$$\frac{(X^{16})\,(\text{ Trama de longitud k})}{X^{16}+X^{12}+X^5+1}$$

Donde:

k = Número de bits de la trama entre la bandera de apertura y secuencia FCS, y excluyendo los bits insertados para transparencia (campos de dirección, control y de información, si existe).

Trama de longitud k = Trama contenida entre la bandera de apertura y la secuencia FCS, y excluyendo los bits insertados para transparencia.

$X^{16}+X^{12}+X^{5}+1$ = Polinomio generador V.41 estandarizado por el CCITT.

En el lado receptor debe realizarse el mismo proceso, pero incluyendo el campo de secuencia de verificación de trama FCS, debiendo obtenerse la siguiente secuencia en caso de una transmisión sin errores:

0001 1101 0000 1111

Transparencia.
Una entidad de nivel de enlace de datos transmisora deberá insertar un bit 0 después de cada secuencia de cinco 1's consecutivos entre las secuencias de bandera de apertura y de cierre (campos de dirección, control, información y campo de verificación de secuencia de trama FCS), incluyendo los cinco últimos bits del campo de FCS. Esto para asegurar que una bandera o una condición de aborto no sea simulada dentro de la trama. Una entidad de nivel de enlace de datos receptora deberá examinar el contenido de la trama entre las banderas de apertura y de cierre y descartará cualquier bit 0 que siga en forma directa a una secuencia de cinco 1's consecutivos. A esta técnica se le conoce como "Relleno de Bits"

Instrucciones y respuestas.

Las siguientes instrucciones y respuestas son usadas por las entidades de nivel 2 tanto del lado del usuario como del lado de la red y están representadas en la fig. 25. Cada conexion de enlace de datos deberá soportar el total de las instrucciones y respuestas para cada una de las aplicaciones implementadas. Los tipos de tramas asociadas con cada una de las dos aplicaciones (instrucción o respuesta) están también identificadas en la fig. 25.

Los tipos de tramas asociadas con una aplicación implementada deberán ser descartadas, y ninguna acción se tomará como resultado de esa tr. Para propósitos de los procedimientos del LAPD en cada aplicación, los códigos que no aparecen en la fig. 25 son considerados como campos de control de instrucciones y respuestas no definidos.

Instrucción de información (I).
La función de la instrucción de información es transferir, a través de una conexión de enlace de datos, tramas numeradas secuencialmente conteniendo campos de información proporcionados por el nivel 3. Esta instrucción es usada en la operacion multitrama en conexiones de enlace de datos punto a punto.

Instrucción de establecimiento del modo balanceado asíncrono extendido (SABME).
La instrucción no numerada SABME es usada para establecer a la entidad de nivel 2 del lado del usuario o del lado de la red direccionado, en el modo de operación con acuse de recibo multitrama con un módulo igual a 128.

Ningún campo de información es permitido con la transmisión de una instrucción SABME. Una entidad de nivel 2 confirmará la aceptación de una instrucción SABME mediante la transmisión de una respuesta UA a la brevedad posible. Después de aceptar este comando las variables V(S), V(R) y V(A) serán puestas a 0. La transmisión de una instrucción SABME indica la eliminación de todas las condiciones de excepción existentes.

Las tramas de información I transmitidas previamente que no han sido reconocidas cuando esta instrucción es procesada, permanecen sin serlo y son descartadas. La recuperación de información de esas tramas que son descartadas es responsabilidad del nivel superior (nivel 3 o entidad de gestión).

Instrucción disconnect (DISC).
La instrucción no numerada DISC es usada para terminar n el modo de operación multitrama. Ningún campo de información es permitido con la transmisión de una instrucción DISC. La entidad de nivel 2 que recibe una instrucción DISC confirma la aceptación del mismo mediante la transmisión de una respuesta UA. La entidad de nivel 2 que envia la instrucción DISC termina con la operación del modo multitrama al recibir una respuesta UA o DM.

Las tramas de información I transmitidas previamente que no han sido reconocidas cuando este comando es procesado, permanecen sin serlo y son descartadas. La recuperación de información de esas tramas que son descartadas es responsabilidad del nivel superior (nivel 3 o entidad de gestión).

Instrucción de información no numerada (UI).
Cuando una entidad de nivel 3 o de gestión solicita transferir información sin acuse de recibo, se debe utilizar la instrucción no numerada UI para enviar información a su entidad par sin afectar las variables de nivel 2. Las tramas de instrucción UI no contienen un número de secuencia, y a raiz de esto, pueden perderse sin notificación.

Instrucción/Respuesta listo para recibir (RR).
La trama de supervisión RR es usada por la entidad de nivel 2 para:

a) Indicar que está lista para recibir una trama I.
b) Dar acuse de recibo de tramas numeradas I recibidas previamente incluyendo la trama N(R)-1.
c) Borrar una condición de ocupado que fue indicada anteriormente mediante la transmisión de una trama RNR por la misma entidad de nivel 2.

Además, esta instrucción puede ser usada, poniendo el bit P a un valor de 1, para solicitar a su entidad par de nivel 2 una respuesta acerca de su condición.

Instrucción/Respuesta de rechazo (REJ).
La trama de supervisión REJ es usada por una entidad de nivel 2 para solicitar la retransmisión d tramas I empezando con la trama numerada N(R). El valor de N(R) en la trama REJ da acuse de recibo de tramas numeradas I recibidas incluyendo a N(R)-1. Las nuevas tramas que no han sido transmitidas por primera vez, deberán transmitirse siguiendo a las tramas I retransmitidas.

Solo una condición de excepción para una dirección dada de transferencia de información se puede establecer en un instante. La condición de excepción REJ es borrada después de recibir una trama I con N(S) igual al N(R) de la trama REJ Un procedimiento opcional para la retransmisión de una respuesta REJ es descrita en el apéndice I.

La transmisión de una trama REJ puede también indicar la desaparición de una condición de ocupado dentro de la entidad de nivel 2 que la envía; esta condición de ocupado se reporta mediante la transmisión de una trama RNR por parte de la misma entidad de nivel 2.
Además, esta instrucción puede ser usada poniendo el bit P a un valor de 1, para solicitar a su entidad par de nivel 2 una respuesta acerca de su condición.

Instrucción/Respuesta no listo para recibir (RNR).
La trama de supervisión RNR es usada por una entidad de nivel 2 para indicar una condición de ocupado; es decir, la incapacidad temporal para aceptar nuevas tramas I arribantes. El valor de N(R) en la trama RNR da acuse de recibo de tramas numeradas incluyendo a N(R)-1.
Además, esta instrucción puede ser usada poniendo el bit P a un valor de 1, para solicitar a su entidad par de nivel 2 una respuesta acerca de su condición. .

Respuesta de acuse de recibo no numerado (UA).
La respuesta no numerada UA es usada por una entidad de nivel 2 para dar acuse de recibo de la recepción y aceptación de instrucciones de establecimiento de modo de operación (SABME o DISC). Las instrucciones de establecimiento de modo de operación recibidas no son procesadas hasta que la respuesta UA es transmitida. Ningún campo de información es permitido al transmitir esta respuesta. La transmisión de la respuesta UA indica la eliminación de una condición de ocupado que haya sido reportada mediante la transmisión anterior de una trama RNR por la misma entidad de nivel 2.

Respuesta de modo desconectado (DM).
La respuesta no numerada DM es usada por una entidad de nivel 2, para reportar a su entidad par, que se encuentra en un estado tal que la operación en modo multitrama no puede llevarse a cabo Ningún campo de información es permitido al transmitir la respuesta DM.

Respuesta rechazo de trama (FRMR).
La respuesta no numerada FRMR puede ser recibida por una entidad de nivel 2, como un reporte de una condición de error no recuperable mediante la retransmisión de una trama idéntica. Esta condición de error será al menos una de las siguientes:

a) La recepción de un campo de control de instrucción o respuesta no definido o no previsto.
b) La recepción de una trama no numerada o de supervisión con longitud incorrecta.

c) La recepción de un N(R) inválido.
d) La recepción de una trama I con un campo de información que excede la máxima longitud establecida.

Instrucción/Respuesta intercambio de identificación XID.
La trama XID contiene un campo de información, en el cual la información de identificación está contenida El intercambio de tramas XID es una disposición obligatoria utilizada en la gestión de conexión (es decir, cuando una entidad par recibe una instrucción XID, debe responder con una respuesta XID a la brevedad posible). El campo de control de esta trama no contiene números de secuencia.

El campo de información no es obligatorio Dependiendo si existe o no, la entidad receptora tomará una de las tres acciones siguientes

a) Recepción de una trama conteniendo un campo de información que puede interpretar. En este caso deberá contestar con una trama XID conteniendo un campo de información similar.
b) Recepción de una trama conteniendo un campo de información que no puede interpretar. En este caso deberá contestar con una trama XID conteniendo un campo de información de longitud cero.
c) Recepción de una trama conteniendo un campo de información de longitud cero. En este caso deberá contestar con una trama XID conteniendo un campo de información de longitud cero.

La máxima longitud permitida en el campo de información de esta trama será igual a 260 octetos. La transmisión o recepción de una trama XID no debe tener efecto en el modo de operación de las variables de estado asociadas con las entidades de nivel de enlace de datos.

PROTOCOLO DE CAPA 3 PARA EL CANAL D

La capa 3 corresponde a la capa de RED y es responsable del Establecimiento , Mantenimiento y Terminación de las conexiones de red de los canales D y B. Ademas la capa 3, proporciona funciones de Enrutamiento y Direccionamiento. El protocolo de la capa 3 se describe en las recomendaciones del CCITT I 450 y la I.451. El protocolo de la capa 3 esta contenido en el campo de información de la capa 2 El discriminador de protocolo identifica el protocolo de la capa 3 Este protocolo puede ser el especificado por el CCITT o una versión nacional u otro protocolo como el X 25. El discriminador de protocolo es seguido por el campo Call Reference (CR) el cual es empleado para identificar cada llamada en la interfase local usuario-red. Los valores del Call Reference son asignados por el que origina la llamada y es removido cuando la llamada se completa o después de la suspensión de la misma. Véase fig. 26.

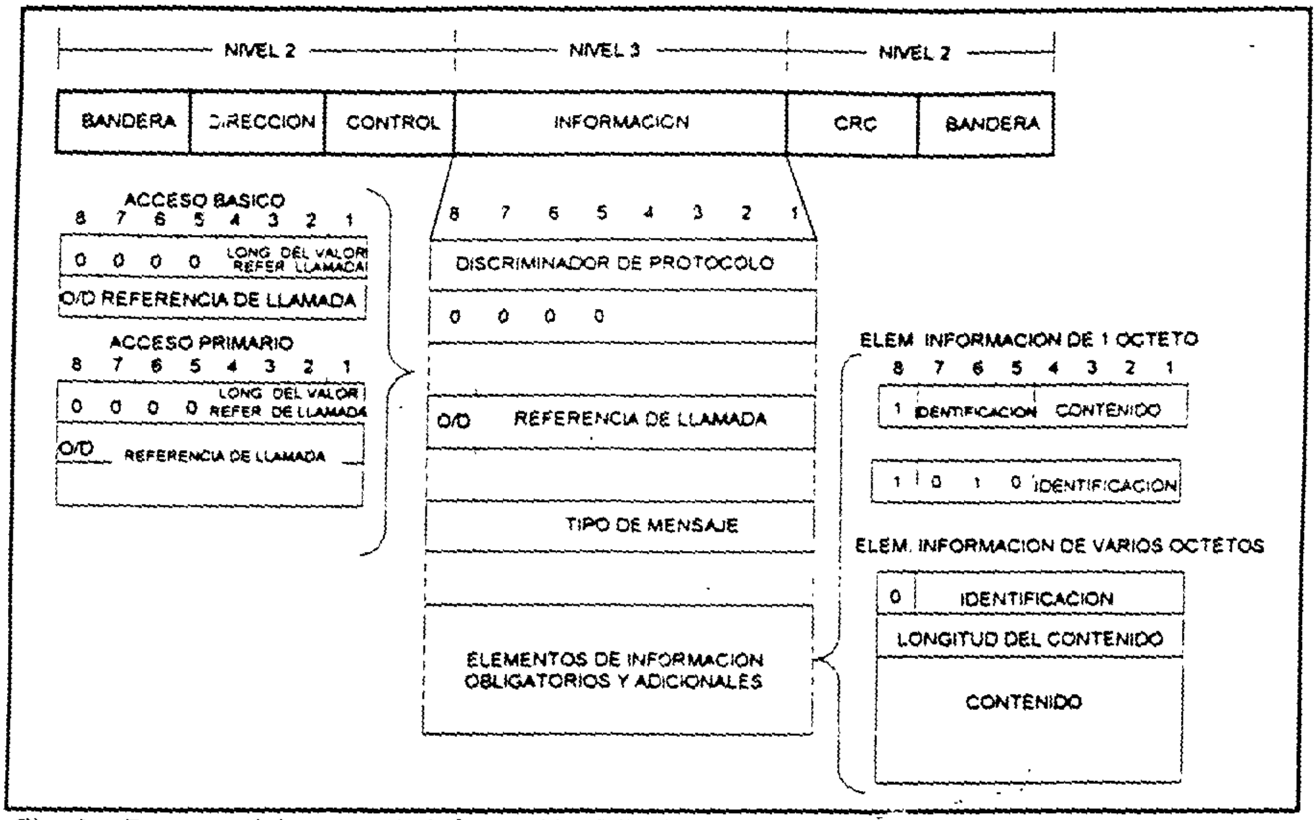


Fig 26 Formato del campo de información (Nivel 3 de canal D).

Los mensajes mas importantes para el control de llamadas se describen a continuación:

SETUP

Se emplea para indicar llamada de establecimiento y puede ser enviado por ambos lados usuario y red Cuando se envia desde la red es un mensaje difundido que da la posibilidad a todos los TE's de contestar la llamada.

87654321	
000-----	MENSAJES PARA EL ESTABLECIMIENTO DE LA LLAMADA
00001	- ALERTing
00010	- CALL PROCeeding
00111	- CONNect
01111	- CONNect ACKnowledge
00011	- PROGress
00101	- SETUP
01101	- SETUP ACKnowledge
001-----	MENSAJES DURANTE LA FASE ACTIVA DE LA LLAMADA
00110	- RESume
01110	- RESUME ACKnowledge
00010	- RESume REJect
00101	- SUSPend
01101	- SUSPend ACKnowledge
00001	- SUSPend REJect
00000	- USER INFOrmation
010-----	MENSAJES PARA LA TERMINACION DE LA LLAMADA
00101	- DISConnect
01101	- RELease
11010	- RELease COMplete
011-----	MENSAJES DIVERSOS
11001	- CONgestion CONtrol
00010	- FACility
11011	- INFOrmation
01110	- NOTIFY
11101	- STATUS
10101	- STATUS ENQuiry

Fig. 27 Mensajes del Nivel 3 del canal D para el control de llamadas en conmutación de circuitos.

CONNECT
Es enviado por el usuario a la red o por la red al usuario llamado para indicarle la aceptación de la llamada.

CONNECT ACKNOWLEDGE
Enviado por la red al usuario para indicarle que la llamada esta localizada en el equipo terminal TE.

DISCONNECT
Invitación a liberar el canal y el call reference ,puede ser enviado por los dos ,usuario y red. Como en este momento el canal y el call reference estan aun activos es puede intercambiar informacion de canal después de liberada.

RELEASE

Puede ser enviado por el usuario o la red como respuesta al mensaje de DISCONNECT si la llamada se concluyo.

87654321	
000-----	MENSAJES PARA EL ESTABLECIMIENTO DE LA LLAMADA
00001	- ALERTing
00010	- CALL PROCeeding
00111	- CONNect
01111	- CONNect ACKnowledge
00011	- PROGress
00101	- SETUP
010-----	MENSAJES PARA LA TERMINACION DE LA LLAMADA
00101	- DISConnect
01101	- RELease
11010	- RELease COMplete
011-----	MENSAJES DIVERSOS
11101	- STATUS
10101	- STATUS ENQuiry

Fig. 28 Mensajes del Nivel 3 del canal D para el control de llamadas en conmutación de paquetes.

RELEASE COMPLETE

Es enviado como respuesta al mensaje de liberar para indicar que ambos canal y call reference están liberados.

BEARER CAPABILITY

Este elemento indica que capacidad de red esta proporcionándose es decir, si transferencia en modo paquete o en circuito, velocidad de información y en el caso de transferencia en paquetes contiene información de protocolos de capa 2 y 3 .

DESTINATION ADDRESS

Identifica la llamada destino .Plan de numeración, direccionamiento y numero llamado .

CHANNEL IDENTIFICATION

Contiene información acerca del tipo de canal que puede ser tipo B o D .

En la fig. 29 se muestra un ejemplo de la señalización por canal D, para el control de una llamada de un usuario "A", a un usuario "B" que tiene conectados al bus "S" dos equipos terminales compatibles (es decir que ofrecen el mismo teleservicio).

Una vez establecida la llamada, los casos presentados son:
CASO 1: el usuario "B" cuelga.
CASO 2: el usuario "A" cuelga

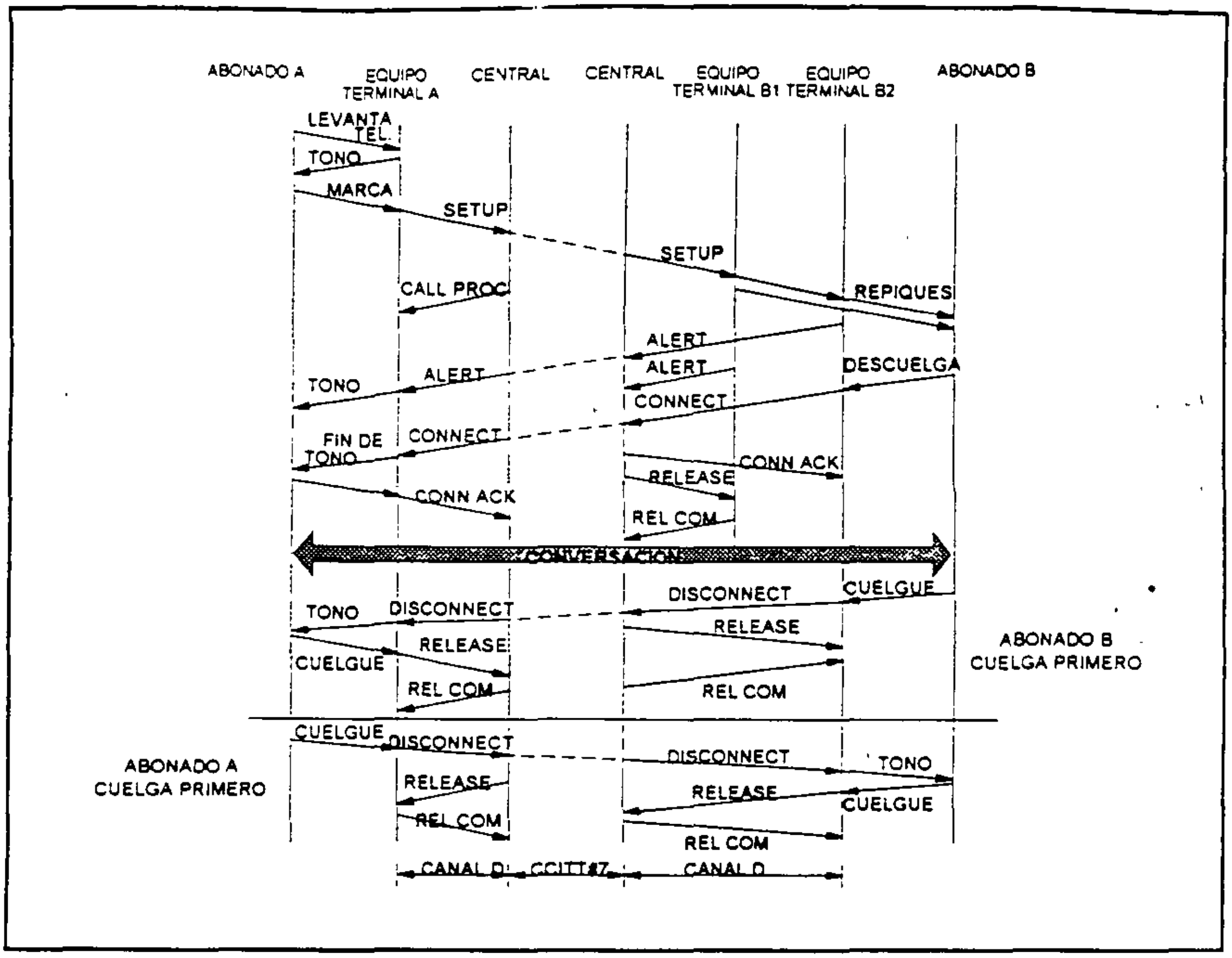


Fig 29. Ejemplo de señalización por canal D a través de la Red.

GLOSARIO DE TÉRMINOS.

DLC - Conexión de enlace de datos.
CCITT - Comité Consultivo Internacional Telegráfico y Telefónico.
DISC - Desconexión.
DM - Modo desconectado
FRMR - Rechazo de trama.
I - Tramas de información numeradas.
DLCI - Identificador de conexión de enlace de datos.
TEI - Identificador de punto extremo terminal.
SAPI - Identificador de punto de acceso al servicio.
OSI - Interconexión de sistemas abiertos.
LAPD - Procedimientos de acceso al enlace en el Canal D
N(S) - Número secuencial en emisión.
N(R) - Número secuencial en recepción.
SAP - Punto de acceso al servicio.
DLCE - Punto extremo de conexión de enlace de datos.
RDI - Red digital integrada
RDSI - Red digital de servicios integrados
REJ - Rechazo.
RNR - No preparado para recibir.
RR - Preparado para recibir.
S - Tramas de supervisión.
SABME - Paso a modo balanceado asincrono ampliado.
FCS - Secuencia de verificación de trama.
U - Tramas no numeradas.
UA - Acuse de recibo no numerado.
UI - Información no numerada.
V(A) - Variable de estado de acuse de recibo.
V(R) - Variable de estado en recepción.
V(S) - Variable de estado en emisión.
XID - Intercambio de identificación.
BRI - Interfase de acceso Básico
PRI - Interfase de Acceso Primario

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Fascículo III.8, I.330 - I.470
Fascículo VI.10, Q.920 - Q.921 ó I.440 - I.441
Fascículo VI.10, Q.930 - Q.940 ó I.450 - I.451

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FACULTAD DE INGENIERIA U.N.A.M.
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CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

RED DIGITAL DE SERVICIOS INTEGRADOS

EXPOSITOR: ING. MA. DEL CARMEN ANGELICA MORENO ARGÜELLO
PALACIO DE MINERÍA
JUNIO DE 1999

Red Digital de Servicios Integrados

ANGÉLICA MORENO ARGÜELLO

Junio, 1999

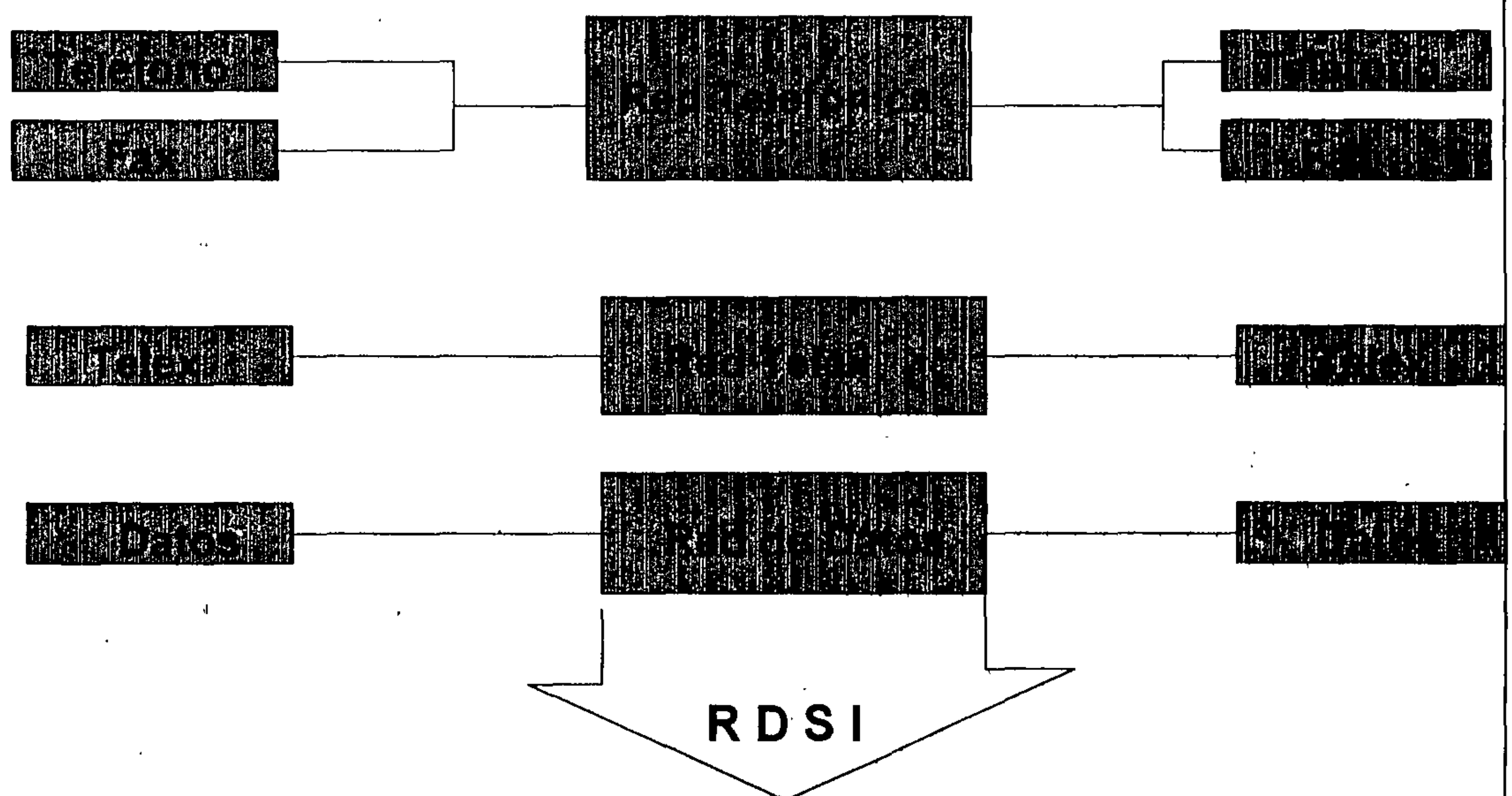
RDSI- ISDN

- ¿Qué es una RDSI?
- Ventajas de una RDSI
- Normalización en RDSI
- Clasificación de RDSI
- ¿Qué necesitamos para tener una RDSI?

Una Red Digital de Servicios Integrados

- Es una Red que permite **Conectividad Digital Extremo a Extremo**, para una amplia gama de Servicios **"Con Voz"** y **"Sin Voz "** en la misma Red. La prestación de esos Servicios deberá hacerse mediante el uso de un <u>conjunto limitado de Tipos de Conexión y Configuraciones de Interfases Usuario- Red.</u>

Antes de RDSI



Ventajas que ofrece RDSI

- Mejor funcionamiento y costo efectivo menor que cualquier red especial actual.

Ventajas que ofrece RDSI

Continuación...

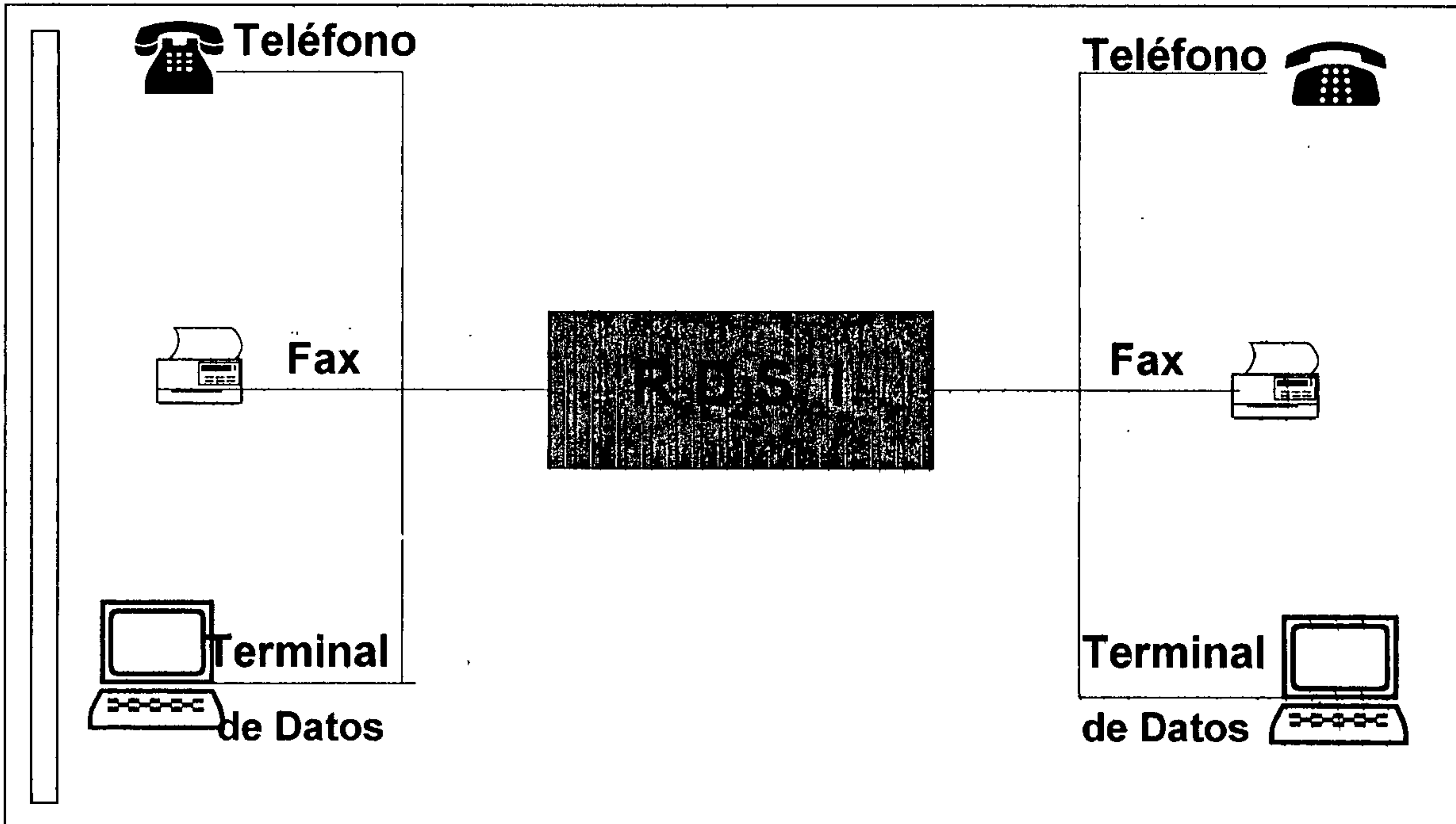
- **Comunicación mas eficiente y amplia, esto se refiere a la posibilidad de emplear terminales multifuncionales y todos los servicios en un enchufe común, una sola linea y un solo número para llamada.**

Ventajas que ofrece RDSI

Continuación.

- **Altas velocidades de transmisión(64Kbps) para la mayoria de los servicios de "no voz"comparados con las de los sistemas comunnmentes disponibles**

Después de RDSI



RDSI- ISDN

- ¿Qué es una RDSI?
- Ventajas de una RDSI
- Normalización en RDSI
- Clasificación de RDSI
- ¿Qué necesitamos para tener una RDSI?

Organizaciones de Estandarización en R D S I

- I T U International Telecommunication Union
- International Standards Organization
- CEPT European Conference of Posts and Telecommunications Administrations
- ETSI European Telecommunications Standards Institute
- ANSI American National Standards Institute
- EIA Electronic Industries Association
- BELLCORE Bell Communications Research

Normalización en RDSI

sus objetivos principales son:

- **"La Normalización de los Servicios Ofrecidos a los Usuarios"** Con el fin de que éstos Servicios sean Compatibles en el plano Internacional

Normalización en RDSI

sus objetivos principales son:

- **"La Normalización de las Interfases Usuario-Red"** Con el fin de que el equipo Terminal sea Transportable, además de Facilitar el aspecto del inciso anterior.

Normalización en RDSI

sus objetivos principales son:

- **"La Normalización de las Capacidades de Red"** Con el fin de hacer posible el Interfuncionamiento **Usuario-Red** y **Red-Red** para lograr los objetivos de los incisos anteriores.

Normalización

- "La Normalización de los Servicios Ofrecidos a los Usuarios"
- "La Normalización de las Capacidades de Red"
- "La Normalización de las Interfases Usuario-Red"

Clasificación RDSI

- **Banda Angosta** (Narrowband) Conocida como **RDSI ó ISDN** ésta ofrece servicios con 64Kbps y emplea el par de cobre utilizado para telefonia convencional.
- **Banda Ancha** (Wideband and Broadband) Conocida como **RDSI-BANCH ó B-ISDN** ésta ofrece servicios mayores a 2Mbps y emplea fibra óptica, radio digital ó coaxial.

Tipos de Canales en RDSI

- **B** Son los canales que están destinados a llevar **Información**
- **D** Son los canales que están destinados a llevar **Señalización**
- **H** Son canales destinados a llevar **Información** en sistemas de Banda Ancha

Tipos de Canales RDSI

- **B** 64Kbps
- **D** 16Kbps / 64Kbps

Tipos de Canales RDSI
en *Banda Angosta*

- H0 384Kbps ≈ 6B
- H11 1536Kbps ≈ 24B
- H12 1920Kbps ≈ 30B

Tipos de Canales RDSI

en *Banda Ancha:*

- **H2** 30 a 45Mbps
- **H21** 30,720Kbps
- **H22** 33,792Kbps
- **H32** 44,160Kbps
- **H3** 60 a 70Mbps
- **H4** 120 a 140Mbps

Un Canal de 64Kbps permite:

- 64Kbps = 8Kbytes/s = 8000 caracteres/s
- Desplegar una pantalla completa (24 lineas de 80 caracteres cada una) en 0.25 segundos
- El contenido de un libro de 200 páginas puede ser transferido en un minuto.

Tipos de Acceso en RDSI

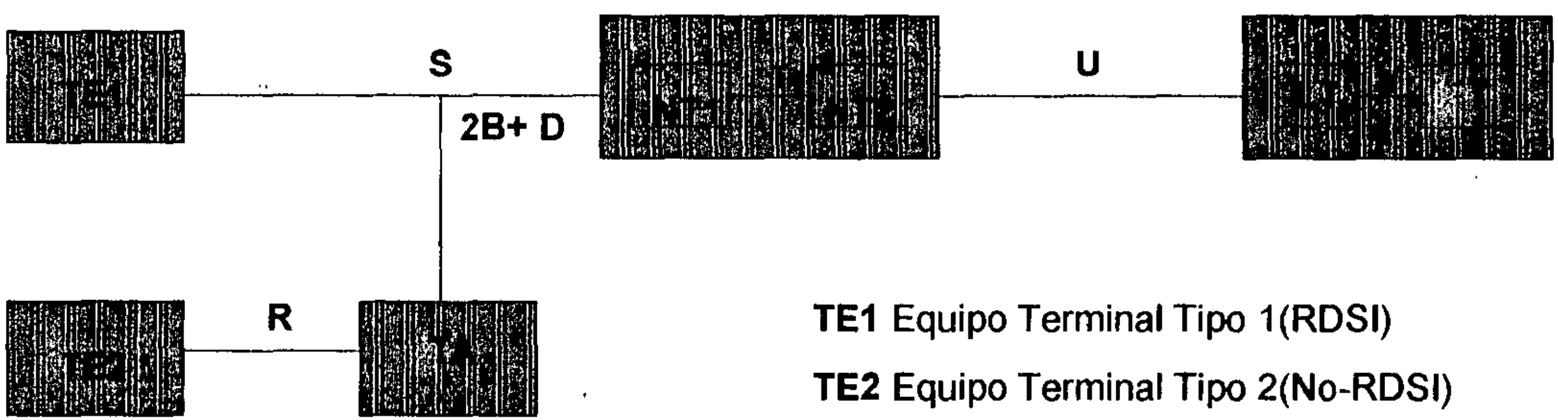
- **Acceso Básico 2B + D**

  canal B = 64Kbps y en éste caso
  canal D = 16Kbps

- **Acceso Primario 30B + D**

  canal B = 64Kbps y en éste caso
  canal D = 64Kbps

Grupos Funcionales y Puntos de Referencia



TE1 Equipo Terminal Tipo 1(RDSI)

TE2 Equipo Terminal Tipo 2(No-RDSI)

TA Adaptador de Terminal

NT1 Terminador de Red Tipo 1

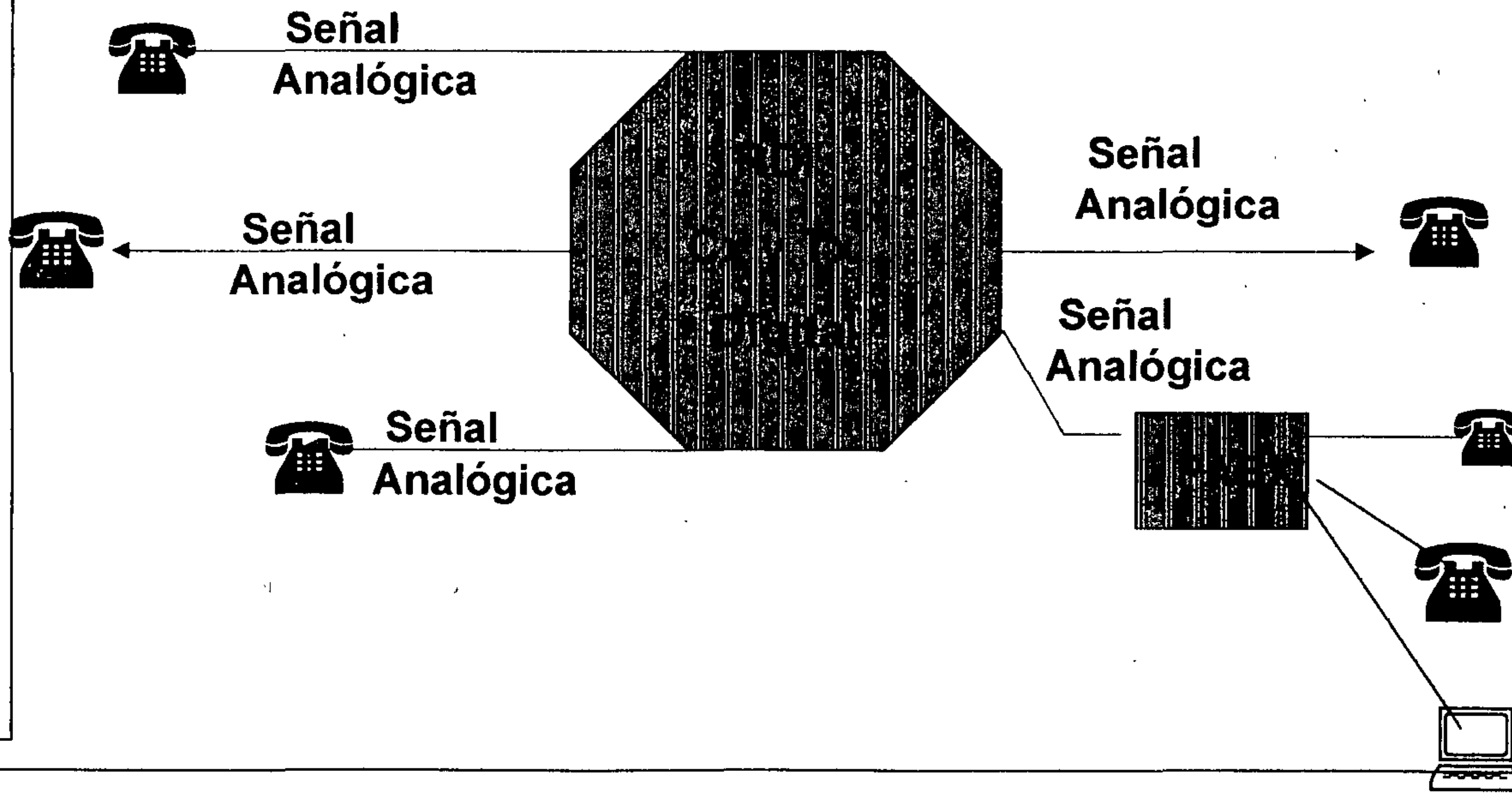
NT2 Terminador de Red Tipo 2 (**PABX ó LAN ; 30B+ D**)

R,S,T,U,V Puntos de Referencia

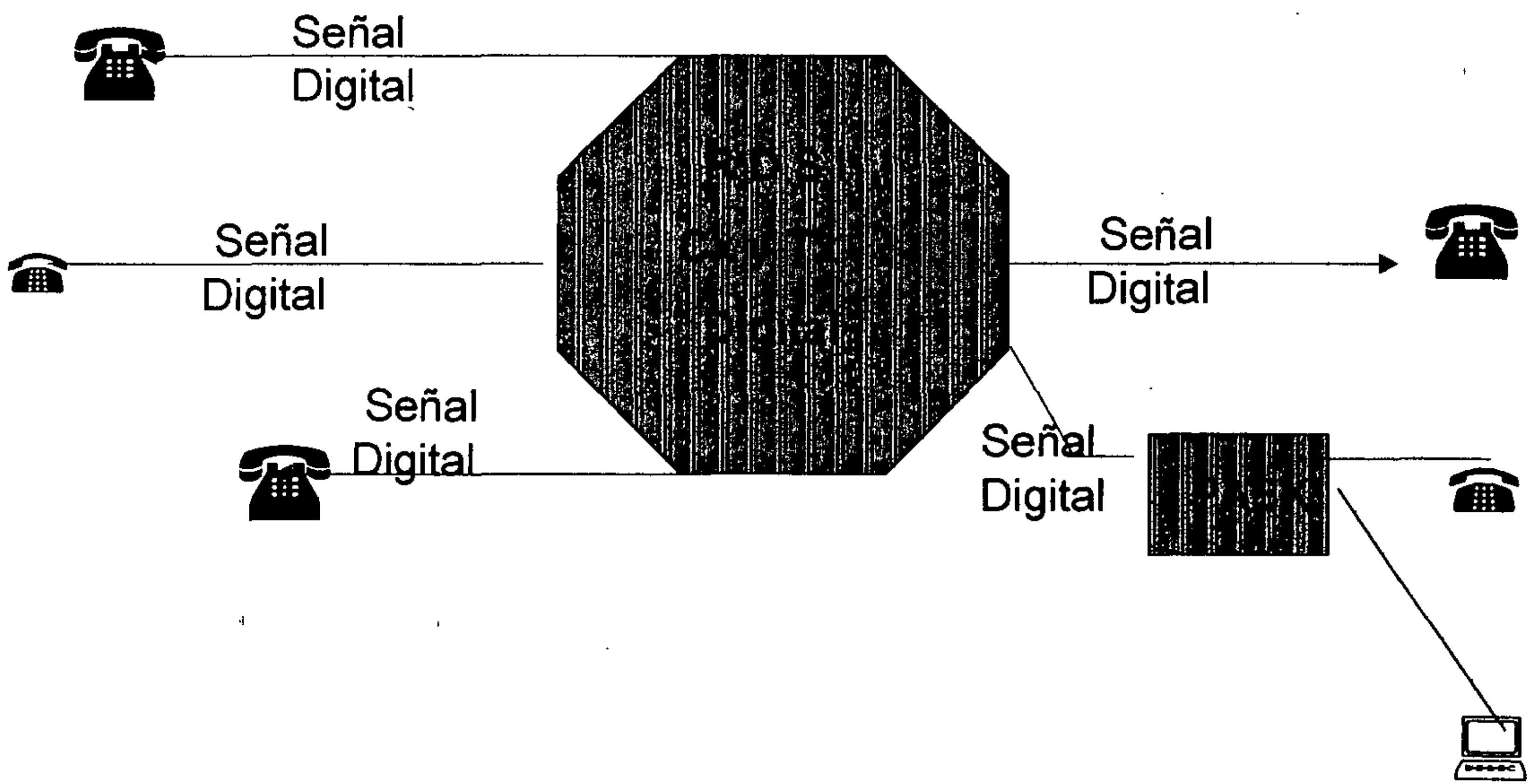
¿Que se necesita para tener RDSI?

- Red Digital Integrada
- Sistema de Señalización por Canal Común
- Conmutación de Circuitos
- Conmutación de Paquetes

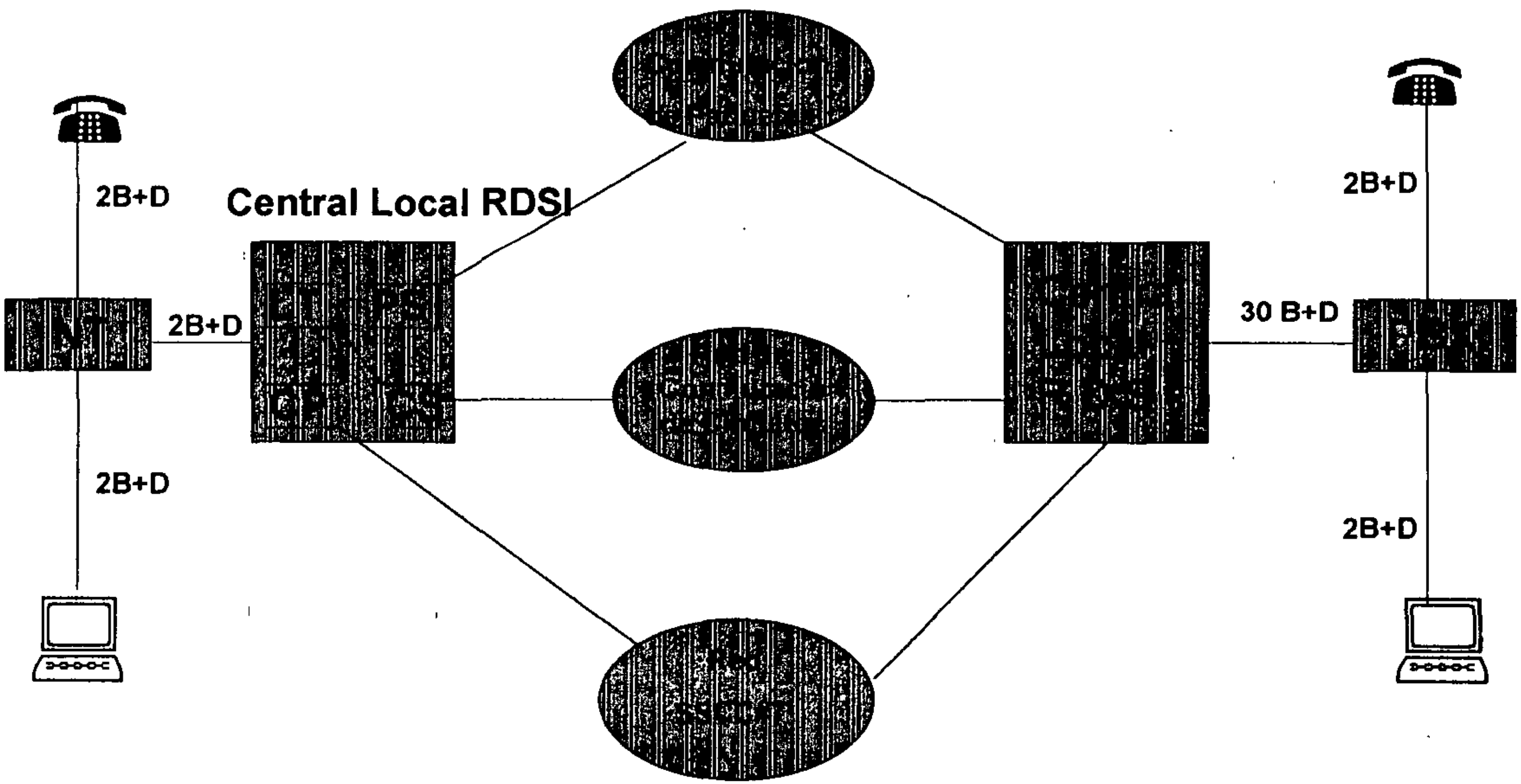
Red Digital Integrada



Red Digital de Servicios Integrados



Modelo RDSI



CURRICULUM VITAE

María del Carmen Angélica Moreno Argüello, Nació en la Ciudad de México. ***Ingeniero Mecánico Electricista*** en el Area de Comunicaciones y Electrónica, egresada de la **F**acultad de **I**ngeniería de la UNAM, cursó los estudios de la Maestría en Ingeniería Eléctrica área Telecomunicaciones en la **D**ivisión de **E**studios de **P**osgrado de la Facultad de Ingeniería **(DEPFI–UNAM).**
Se ha desarrollado en el área de Telecomunicaciones colaborando como Investigador en el **C**entro de **I**nvestigación y **D**esarrollo de **TELMEX**, participando ahí como responsable en la *Primer Videoconferencia en México*. Responsable de Proyectos en Telecomunicaciones como **Canal 40 XHTVM**, en Grupo **IUSACELL-IUSANET** como Gerente de Transmisión de Video y Datos realizó la habilitación de la *Primer Sala Pública de Videoconferencia en México*. Participó en la puesta en marcha de las Oficinas de la Compañía fabricante de Equipos de Videoconferencia **PictureTel** en México como Directora de Mercadotecnia y Soporte Técnico.

Ha realizado estudios en el extranjero en las áreas de ***Comunicaciones en Edificios Inteligentes*** en **España** y ***Equipos de Medición en Red Digital de Servicios Integrados*** en **Alemania**. Cursa actualmente el Diplomado ***"Educación Abierta y a Distancia"*** en la Coordinación de Educación Abierta y Educación a Distancia de la **UNAM**.

En el aspecto Docente ha venido participando como Profesor en la Facultad de Ingeniería de la UNAM en las materias ***Redes de Teleinformática y Temas Selectos de Comunicaciones.***

Es ***Coordinadora Académica*** de los Módulos Redes Digitales y Comunicaciones Vía Microondas en el Curso Internacional de Telecomunicaciones de la División de Educación Continua de la **Facultad de Ingeniería de la UNAM.** Ha colaborado también en Instituciones como **Universidad IberoAmericana (UIA)** Campus Santa Fé (Diplomado de Comunicaciones para Ericsson), **Universidad De Las Americas (UDLA)** Campus Puebla, **Instituto Tecnológico de Estudios Superiores de Monterrey (ITESM)** Campus Ciudad de México y **Fundación Rosenblueth** con Presentaciones de Conferencias sobre Redes de Banda Ancha, Redes Digitales de Servicios Integrados, Redes de Fibra Óptica y Redes de Alta Velocidad.

Es miembro de la **S**ociedad de **E**xalumnos de la **F**acultad de **I**ngeniería **SEFI,** donde ocupa el cargo de **Consejero**. Es miembro activo del **I**nstituto de **I**ngenieros **E**léctricos y **E**lectrónicos **IEEE** donde fué **Presidente de la 5ª Reunión de Otoño de Comunicaciones y Computación ROC&C 1994**, Presidente del Capítulo de Comunicaciones Sección México en 1993 y Presidente del Capítulo de Computación Sección México en 1992 . Es miembro activo de **AIUME** actualmente ocupa el cargo de Protesorero bienio 1999-2001 y ha sido Directora del Instituto de Teleinformática Bienio 1997-1998, Director del Instituto de Telecomunicaciones Bienio 1995-1996, Prosecretario Bienio 1993-1994.

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REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

<u>INTERFASES "S" Y "U"</u>

EXPOSITOR: ING. RODOLFO CASTEÑEDA SEGURA
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JUNIO DE 1999

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INTERFACES "S" Y "U"

Rodolfo Castañeda Segura
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CONTENIDO

1. MODELO DE REFERENCIA OSI

La RDSI ha adoptado un esquema estratificado, basado en el modelo de siete capas de Interconexión de Sistemas Abiertos (OSI) de la Organización Internacional de Normas (ISO), para los protocolos de intercambio de información dentro y a través de la red [CCITT, 1989b] Este modelo se muestra en la Figura 1.1. La estructuración en capas, permite, por un lado, subdividir el problema global de la implementación de protocolos en varias piezas que resultan, obviamente, menos difíciles de realizar que el problema visto como un todo, además de que se obtiene, como consecuencia directa, que cada pieza sea altamente independiente de las demás, de tal forma que se puede alterar el funcionamiento de cualquiera de ellas para aprovechar los nuevos avances en las técnicas de programación o de desarrollo de circuitos sin afectar a las otras capas [Gallardo & Sánchez, 1992].

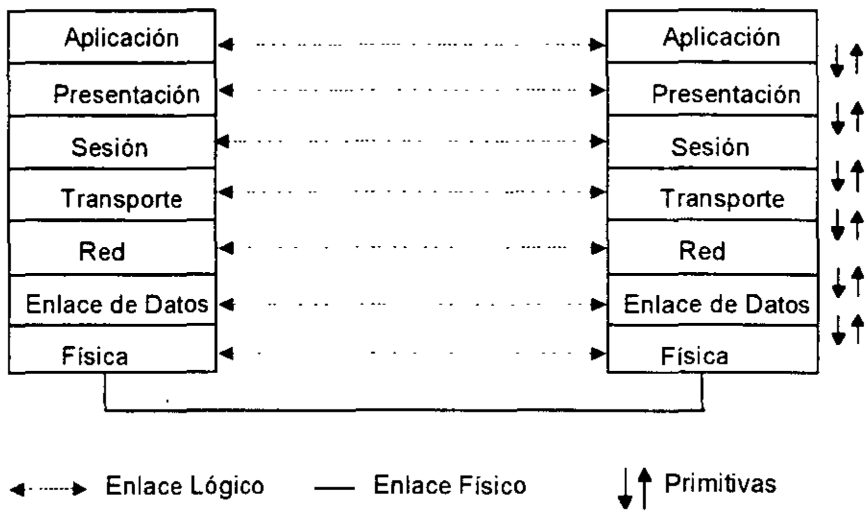


Figura 1.1 Modelo de Referencia OSI

Las características fundamentales de los esquemas estratificados son la definición de procedimientos estandarizados que permiten el intercambio lógico de información entre entidades de un mismo nivel, la creación de fronteras bien delimitadas entre las capas y la posibilidad de interacción directa únicamente entre capas adyacentes

Dos entidades de una misma capa que pertenecen a sistemas diferentes en lados opuestos de la interfaz y que deben intercambiar información para realizar un objetivo común se denominan *entidades par.*

Cada capa puede estar constituida por una o varias entidades que realizan las funciones requeridas. Los mensajes definidos para la comunicación entre entidades de capas adyacentes, de un mismo sistema, se conocen como *primitivas de servicio*. Las primitivas son meramente conceptuales y no está especificado cómo han de realizarse [Gallardo & Sánchez, 1992] Hay cuatro tipos diferentes de primitivas de acuerdo al sentido en que se transmiten y a la función que llevan a cabo, éstas son identificadas con los siguientes nombres petición, indicación, respuesta y confirmación Los tipos de primitivas y su dirección se muestran en la Figura 1 2

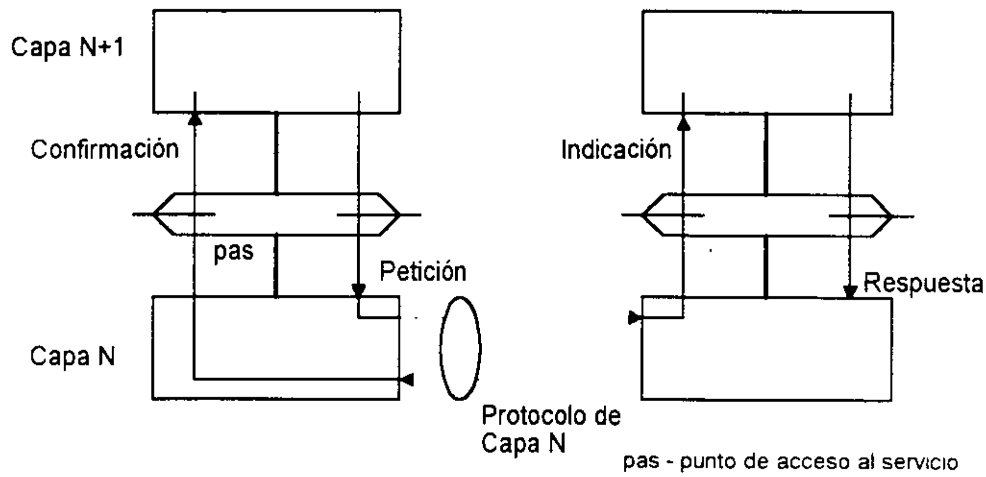


***Figura 1.2**. Tipos de primitivas intercambiadas entre capas adyacentes*

El tipo de *primitiva petición* se utiliza cuando una capa solicita un servicio a la capa inferior El tipo de *primitiva indicación* lo utiliza la capa que proporciona un servicio para notificar a la capa superior cualquier situación relacionada con este servicio, generalmente es el resultado de una actividad desencadenada por una primitiva de tipo petición en la entidad par, o bien, puede implicar la incapacidad de la capa inferior para proporcionar el servicio El tipo de *primitiva respuesta* lo utiliza una capa para acusar recibo de una primitiva de tipo indicación procedente de la capa inferior El tipo de *primitiva confirmación* lo utiliza la capa que proporciona un servicio para confirmar que se ha completado la actividad que le ha sido solicitada mediante una primitiva de tipo petición

La frontera entre entidades adyacentes en un mismo sistema recibe el nombre de *interfaz* y cuenta con un *protocolo de interfaz* que opera a través de ella La interfaz se utiliza para accesar los servicios prestados por la capa inferior a través de un *punto de acceso al servicio* (PAS).

Como se había mencionado antes, la comunicación entre dos entidades del mismo nivel pero de sistemas distintos, se lleva a cabo por medio de *protocolos entre entidades pares*. La comunicación entre entidades pares se realiza utilizando el protocolo de la capa en cuestión pero son necesarios, para lograrla, los servicios de las capas inferiores. Cada capa trata la información procedente de la capa superior como un bloque que no va a procesar, únicamente a transportar. Al construir una trama de salida cada capa añade uno o más campos, que reciben el nombre de *encabezado* [Terpán, 1993]. Estos campos son utilizados para la comunicación con la capa par correspondiente, la cual, al recibir la información procedente de su capa inferior, interpreta y retira el encabezado y transmite el resto de la información hacia arriba hasta que la información original de usuario alcanza su destino.

Los dos tipos de protocolo descritos anteriormente se muestran en la Figura 1.3.

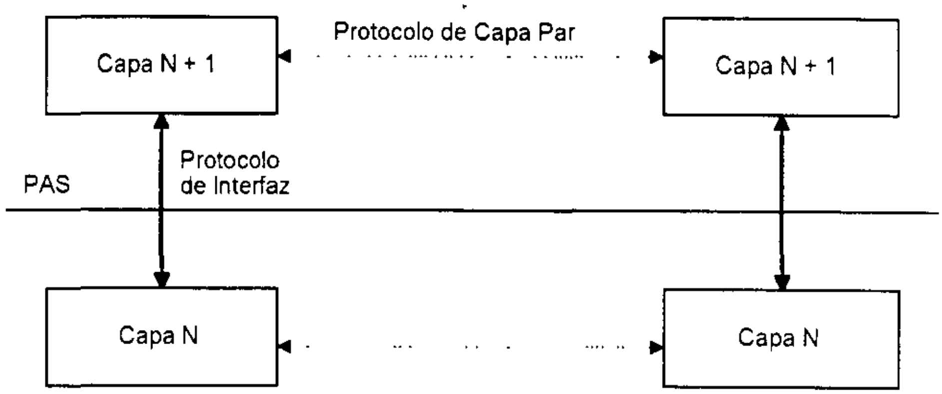


Figura 1.3. *Protocolos de interfaz y de entidades pares en el modelo referencia OSI*

Hasta el momento, el CCITT/UIT ha definido las capas 1, 2 y 3 para la RDSI, las cuales se encuentran íntimamente asociadas con las capas correspondientes del modelo OSI y su relación se muestra en la Figura 1.4. En dicha figura se muestran de una manera separada los protocolos que le corresponden a los canales B y D.

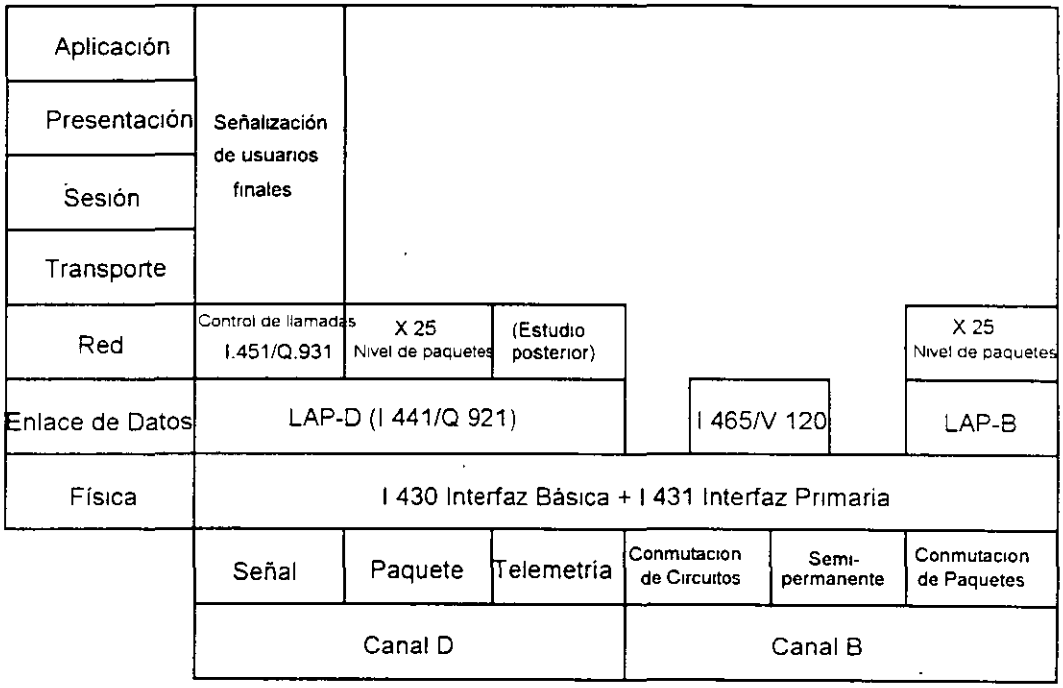


Figura 1.4 *Arquitectura de protocolos RDSI para la interfaz usuario-red*

2. CAPA FISICA DE LA RDSI

La capa física RDSI se presenta al usuario como puntos de referencia S o T. Esta capa es la encargada de todo lo que se refiere a las conexiones eléctricas y mecánicas. se encarga también de las funciones y procedimientos para activar y desactivar las conexiones físicas. Se especifica en las recomendaciones I 430 (Acceso básico). e I.431 (Acceso primario) del CCITT

Las funciones incluidas en la capa físicas (capa 1 de la OSI) son las siguientes:

- Codificación de datos digitales para la transmisión a través de la interfaz
- Transmisión full-duplex de los canales de datos B
- Transmisión full-duplex de los canales de datos D
- Multicanalización de canales para formar la estructura de transmisión de acceso básico o primario
- Activación y desactivación del circuito físico
- Alimentación de energía desde de la terminación de la red hacia la terminal
- Identificación de la terminal
- Aislamiento de terminales con fallas.
- Acceso de contensión al canal D

Los servicios que proporciona a la capa 2 son los siguientes:

- Capacidad de transmisión de los canales B y D. así como funciones de temporización y sincronización.
- Procedimientos de activación/desactivación de ET y/o TR
- Arbitraje de acceso al canal D de los ET en conexiones multipunto
- Procedimiento y funciones de mantenimiento
- Indicación a las capas superiores acerca del estado de la capa 1

2.1. CODIGOS DE LINEA

En la RDSI los datos analógicos o digitales se trasmiten utilizando señales digitales Una señal digital es una secuencia de pulsos de voltaje transmitidos secuencialmente y se utiliza para representar un flujo de datos binarios

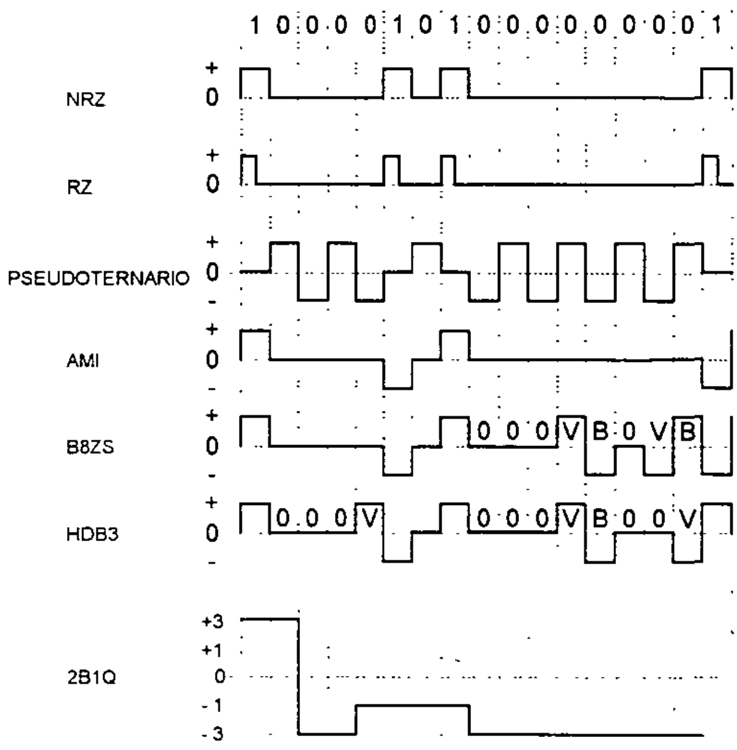
La selección de un código de línea para cualquier sistema de transmisión es crítico para su desempeño Esto es particularmente cierto para la Línea Digital de Abonado (LDA) del acceso básico de la RDSI. En esta aplicación el código de línea afecta a los determinantes del desempeño del sistema de un modo crucial. la principal razón es que el código de línea es un instrumento para determinar tanto las características de transmisión de las señales trasmitidas como de los niveles de ruido de diafonía en el extremo cercano que se añaden de otros pares en el mismo cable. Además se requiere que el desempeño de la LDA tenga una tasa de error (BER) del orden de 10^{-7} para toda la planta externa del par metálico

Para proporcionar accesos básicos de una forma económica la LDA debe de ser utilizada sin acondicionar la planta externa (es decir sin retirar las derivaciones y sin redistribuir los pares).no obstante los efectos perniciosos de las derivaciones y los cambios de calibre. Aún más. no deben asociarse operaciones especiales de ingeniería con las instalación de la LDA Así. para el Acceso básico de la RDSI. una LDA tiene que ser utilizada directamente de la planta telefónica existente.

Uno de los objetivos de la utilización de códigos de línea es reducir al máximo la velocidad de la línea transmitiendo la misma cantidad de información. por lo que el código que cumpla mejor con las siguientes características será un código adecuado para RDSI

- Transparente a la información
- Facilidad para recuperar la señal de reloj
- Evitar (si es posible) la componente de corriente continua. así como la presencia de grandes cantidades de energía a bajas frecuencias
- Redundancia (deseable) para detectar errores en la línea
- Espectro limitado en frecuencia para hacer un buen uso de la atenuación y de la diafonía (crosstalk) presentada por el par torcido de cobre.
- Reducción en la velocidad de transmisión
- Eficiencia
- Propagación mínima de errores
- Insensibilidad a la permutación en los cables del par

En la Figura 2 1 se presentan los códigos de línea utilizados en sistemas de transmisión del tipo RDSI. y en la Tabla 2-1 se presentan los formatos de la codificación de las señales digitales



B = Señal bipolar válida
V = Violación bipolar

Figura 2.1 Códigos de línea

Tabla 2-1 Definición de formatos de codificación de señales digitales

Nonreturn to zero (NRZ)

0 = Nivel alto
1 = Nivel bajo

Bipolar AMI

0 = Ausencia de señal de línea
1 = Nivel positivo o negativo, alternando para unos sucesivos

Pseudoternaria

0 = Nivel positivo o negativo, alternando para ceros sucesivos
1 = Ausencia de señal de línea

B8ZS

Es igual que el código bipolar excepto que cualquier hilera de ocho ceros consecutivos se reemplaza con una hilera que contiene dos violaciones de códigos

HDB3

Es igual que el código bipolar excepto que cualquier hilera de cuatro ceros consecutivos se reemplaza con una hilera que contiene una violación de código

2B1Q

Este código convierte bloques de dos bits consecutivos de la señal en un solo pulso de cuatro niveles para transmisión Como resultado la velocidad de la línea es la mitad de la velocidad de información Como todos los posibles valores de los símbolos transmitidos son utilizados al mapear los dos bits en un símbolo cuaternario, se dice que este es un código saturado. Utiliza el siguiente esquema de codificación

Par de bits DIBITS	Salida codificada
10	+3
11	+1
01	-1
00	-3

2.2. ACCESOS RDSI

La arquitectura RDSI ha definido 3 tipos de interfaz usuario-red para accesar o conectarse a ésta y cubrir la diversidad de aplicaciones requeridas por el usuario [Ibarra, 1993]. La arquitectura RDSI ha definido 3 tipos de interfaz usuario-red para accesar o conectarse a ésta y cubrir la diversidad de aplicaciones requeridas por el usuario. De esta manera en base a los requerimientos del usuario, se le puede asignar una interfaz específica, que cubra sus necesidades, logrando una mejor eficiencia, flexibilidad, baja complejidad y bajo costo [Dicenet, 1987]

Los dos principales tipos de interfaz son la Interfaz de Acceso Básico (BRI) y la Interfaz de Acceso Primario (PRI). Una forma práctica de identificar la diferencia que existe entre estos dos tipos de accesos se muestra en la Figura 2.2, donde se puede observar que el Acceso Básico es exclusivamente para conectar y dar servicio a usuarios que tienen una línea telefónica y el Acceso Primario está enfocado a conectar usuarios que actualmente tienen un conmutador (PABX, Private Automatic Branch eXchange) y que están haciendo uso de un sistema de transmisión PCM (Pulse Coded Modulation) de 2.048 o 1.544 Mbps [A. Moreno, 1995]

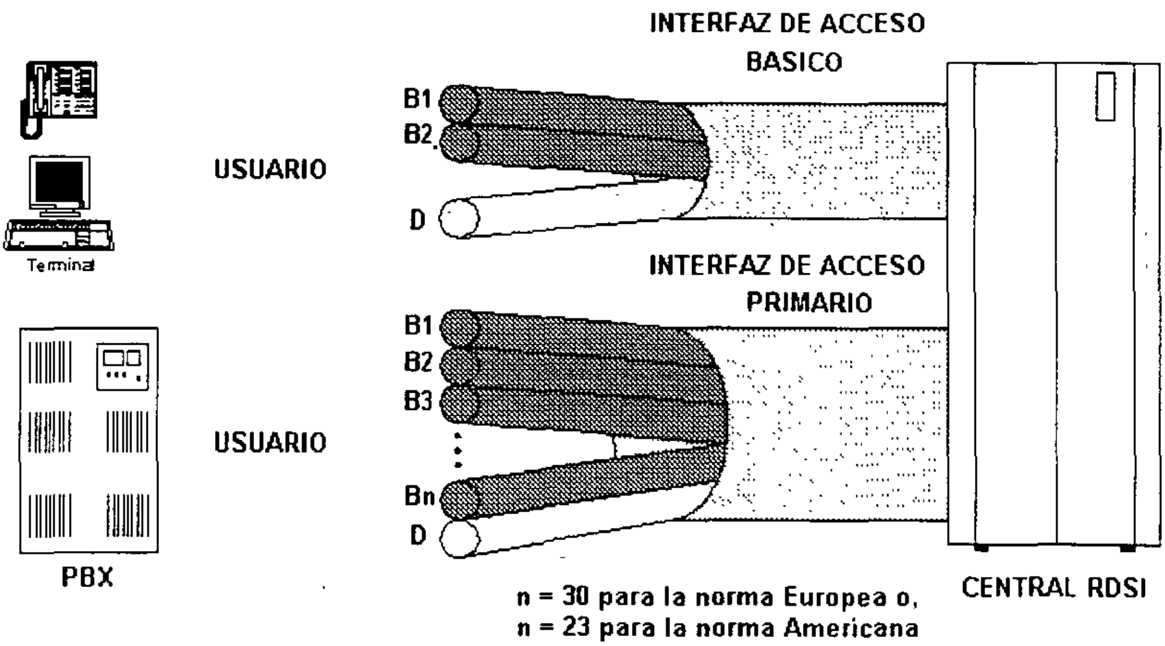


Figura 2.2 Tipos de Acceso a la RDSI.

El tercer tipo de interfaz es la de Acceso de Banda Amplia, ésta proporciona los requerimientos para transmisión de imágenes en movimiento, televisión de alta definición y definición estándar, videoconferencia, etc. Otras aplicaciones incluyen transferencia de archivos a muy alta velocidad y multicanalizadores multimedia que combinen datos de una variedad de fuentes de alta velocidad. La velocidad de datos puede alcanzar varios cientos de Mbps [Ibarra, 1993].

Las característica de la interfaz física y su funcionamiento difieren para el acceso básico y el acceso primario de la interfaz usuario-red

2.2.1. INTERFAZ DE ACCESO BASICO (BRI).

Como se vió anteriormente las normas RDSI definen el acceso del usuario a la RDSI a través de canales B y D para crear las diferentes configuraciones de canales (BRI y PRI) Estas configuraciones de canal se pueden pensar como tubos: cada tubo lleva varios canales los cuales están "multiplexados en tiempo" sobre la línea de transmisión. El circuito de Acceso Básico es normalmente la línea que llega a la casa u oficina del usuario (línea del subscriptor) Este va a reemplazar los circuitos utilizados actualmente por la red telefónica Es una línea digital en la que no se envían tonos de marcación de dígitos. voltajes de timbrado. etc En lugar de enviar éstos. se manda un mensaje que lleva los dígitos marcados. o para indicarle al teléfono que timbre o deje de timbrar

Un BRI consiste de 2 canales B (64 Kbps cada uno) y un canal D (16 Kbps). el cual es conocido como 2B+D y tiene una capacidad para transportar información de 144 Kbps. Con bits adicionales de overhead o control (sincronía. mantenimiento). la velocidad total en la interfaz S/T es de 192 Kbps El protocolo de capa 1 para la interfaz de acceso básico está especificado en la recomendación I 430 [CCITT. 1989a]. la cual define la comunicación entre el ET y el TR a través del punto de referencia S/T

Esta interfaz puede utilizar una configuración punto a punto o punto a multipunto. esta última teniendo dos opciones. ducto pasivo corto y ducto pasivo extendido. y tienen las siguientes características.

- *Configuración punto a punto.* La conexión punto a punto. limitada a 6 dB de atenuación está compuesta por un solo equipo terminal (ET) conectado al terminador de red (TR). del cual. pueden estar separados hasta 1 Km. y puede conectarse sin tomar en cuenta la polaridad

- *Ducto pasivo corto* En esta configuración la ubicación de los terminales está restringido por la dispersión de los pulsos transmitidos simultáneamente en el mismo par Esta configuración permite conectar hasta 8 equipos terminales a un solo terminador de red en un ducto de 100 a 200 mts., según la impedancia del cable. pudiendo estar los ETs y el TR en cualquier punto del ducto

- *Ducto pasivo extendido.* Esta configuración permite que hasta 8 ETs se conecten al final del ducto. agrupadas a no más de 50 mts entre ellas. con cables de conexión menor a 10 mts y pueden ubicarse hasta 500 mts del TR

La impedancia resistiva que debe terminar el ducto es de 100 ohms en cada extremo

La Figura 2.3 muestra la configuración punto a multipunto de la BRI. La conexión física del o los ETs al TR requiere de 2 pares de cables. un par para cada dirección de transmisión

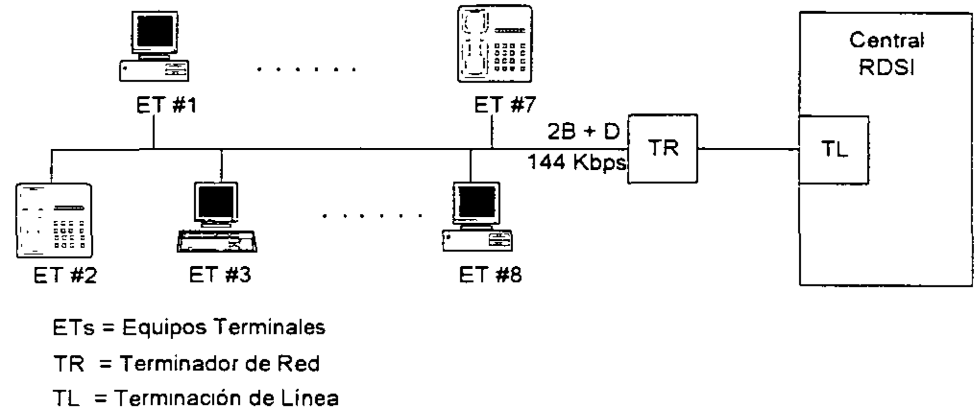


Figura 2.3 *Configuración punto a multipunto de la Interfaz de Acceso Básico.*

Los dos canales B pueden usarse independientemente para diferentes tipos de transmisión. Por ejemplo, un canal B puede llevar información de voz y el otro puede llevar datos. De esta manera, voz y datos son integrados sobre los mismos medios de transmisión.

En la actualidad el BRI es el mismo para todos los países, pero existe una variación en lo que se refiere al contenido del canal B que afecta a equipos que tienen acceso a comunicación de voz como lo es el caso del teléfono, conmutadores privados, y equipos de prueba. La diferencia se basa en el esquema de codificación de la voz que se utilice (ley A o ley μ). La ley μ se utiliza en EUA, Canadá y Japón. La ley A se utiliza en rutas internacionales, Europa, Africa y Latinoamérica.

Para la interfaz de acceso primario sólo se ha recomendado la configuración punto a punto y el nivel físico se encuentra detallado en la recomendación I.431.

A continuación se describirán algunos de los aspectos de la interfaz básica como, conector físico, estructura de trama (incluyendo código de línea), y la forma de activación y desactivación de la interfaz.

2.2.1.1. CODIFICACION DE LINEA

Se utiliza para ambos sentidos de transmisión un *código de línea pseudoternario* (tres niveles de voltaje y solo dos niveles lógicos) con anchura de pulso del 100% (el nivel de voltaje en la línea no varía en el tiempo correspondiente a la duración de un bit). La codificación se efectúa de tal forma que el uno binario se representa por la ausencia de señal (voltaje) en la línea (alta impedancia), mientras que el cero binario se representa por un pulso positivo o negativo de 750 mV ± 10% [Stallings, 1992]. Los ceros binarios se alternarán en polaridad, salvo excepciones necesarias para identificar el inicio y el final de la trama. Un cero que no respeta la alternación de polaridades se conoce como una violación de código [Gallardo & Sánchez, 1992] (Véase Figura 2.4).

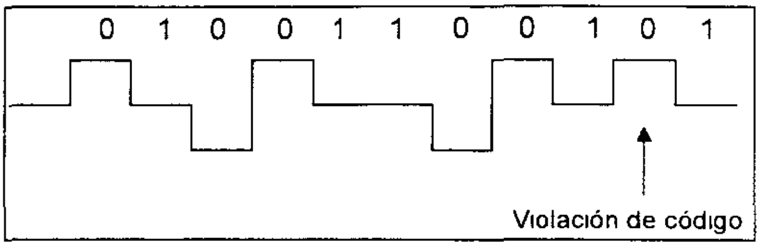


Figura 2.4 Código de linea pseoudoternario con alternación de polaridades en los ceros

El terminador de red (TR) derivará su temporización (tanto de bit. como de octeto y de trama) a partir de la señal recibida de la red y utilizará esta temporización para sincronizar la señal que transmita hacia los equipos terminales (ET's) conectados a él Un equipo terminal deberá obtener sus temporizaciones a partir de la señal recibida desde el terminador de red.

2.2.1.2. CONECTOR FISICO

Esta interfaz utiliza un par metálico simétrico para cada dirección de transmisión y dos pares opcionales para alimentación El conector recomendado (2). corresponde a la norma IS8877 de la ISO y puede verse en la

Figura 2.5 Utiliza obligatoriamente los cuatro terminales centrales para transmitir y recibir la señal en forma balanceada con alimentación en circuito fantasma. esto permite alimentación remota (desde la red) en caso de emergencia

Los 4 terminales externos. son opcionales y se utilizan para alimentación normal en varias configuraciones La utilización del mismo conector para acceso primario. se encuentra en estudio

El ET se basa preferentemente en la detección de las fuentes 1 y 2. para determinar su estado de conexión y envía la correspondiente información de su estado a la entidad de gestión

Los pares 3-4 y 5-6 están destinados a la transmisión bidireccional de la señal digital y pueden proporcionar alimentación en circuito fantasma de TR a ET (fuente 1)

Los pares 1-2 pueden proporcionar energía de TR a ET (fuente 2) o de ET a TR (fuente 3)

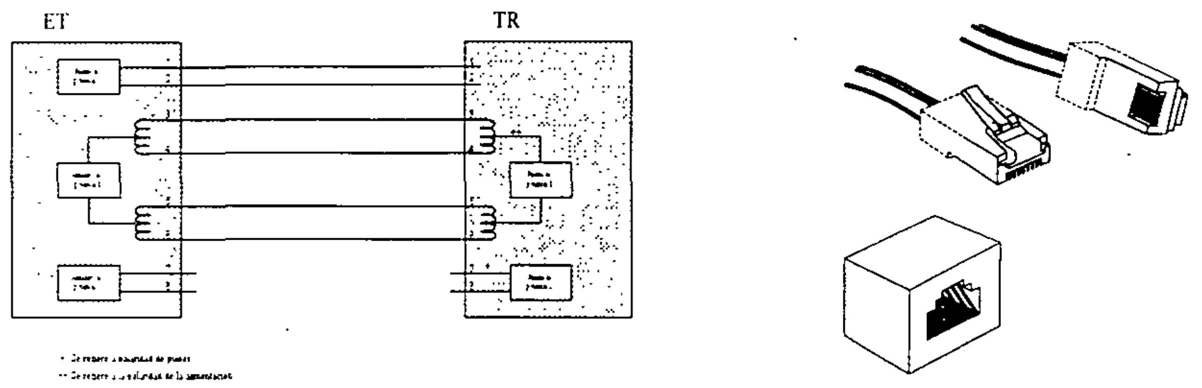


Figura 2.5 Conector físico RDSI.

2.2.1.3. ESTRUCTURA DE TRAMA Y MULTICANALIZACION

Recordemos que el acceso básico consiste de dos canales B (información de usuario a 64 Kbps cada uno) y un canal D (información de señalización o de usuario a 16 Kbps). los cuales son multiplexados en tiempo sobre los cuatro hilos de la interfaz "S". Un par de hilos es usado para transmitir y el otro par es usado para recibir

Las estructuras de trama serán diferentes en cada sentido de la transmisión Un tipo de tramas es transmitido del ET al TR (dirección de usuario a central) y otro tipo de tramas es transmitido del TR al ET (dirección central a usuario). como se ilustra en la Figura 2.6

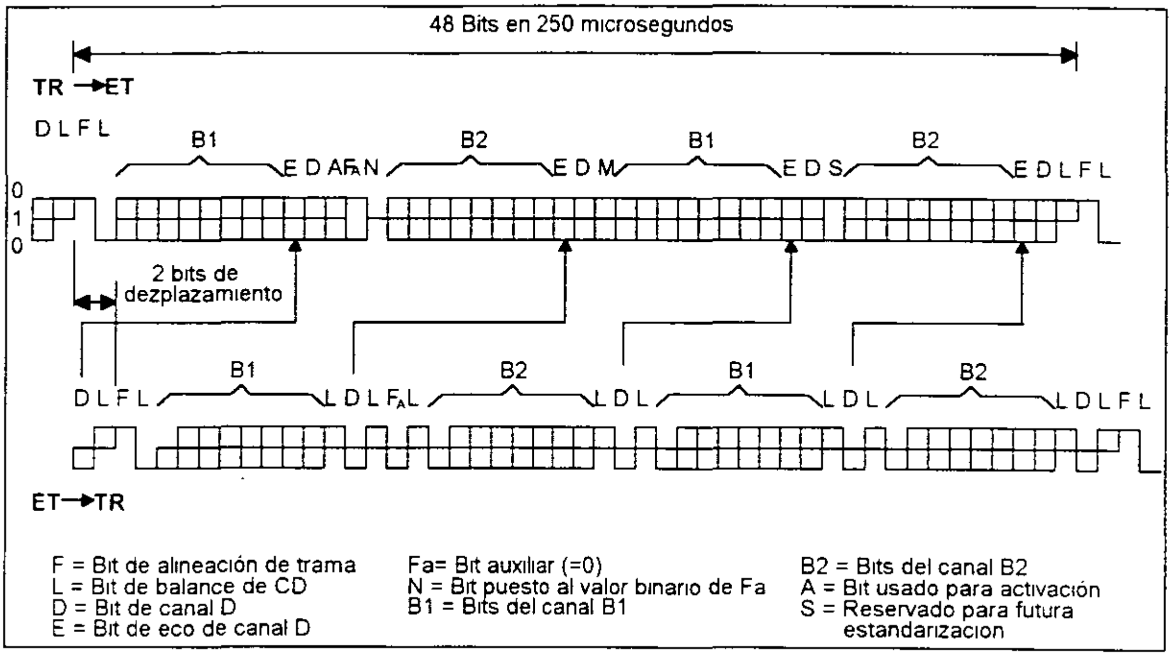


***Figura 2.6** Estructuras de trama para los puntos de referencia S y T a velocidad básica*

El primer bit de cada trama transmitida desde un ET hacia el TR se retardará dos períodos de bit con respecto al primer bit de la trama recibida del TR. Ambos tipos de tramas consisten de 48 bits transmitidas cada 250 µseg (4.000 tramas por segundo). Esto equivale a una velocidad de transmisión total de 192 Kbps, sin embargo, algunos de los 48 bits (12 bits) son de overhead (bits adicionales de control) y no de información de los canales B o D.

Los 36 bits de información de los canales B y D son usados como sigue: 16 bits son del primer canal B, 16 bits son del segundo canal B, y cuatro bits del canal D. Esto resulta en una transferencia de datos a una velocidad de 144 Kbps (36 bits x 4000 tramas por segundo).

El bit F, es un cero binario y siempre se codifica como una violación al código de línea.

El bit L, mantiene el balance de C.D. para un cierto conjunto de bits precedentes. Su valor lógico será un "uno" si los bits que se tratan de equilibrar contienen un número par de "ceros" (paridad par).

Los bits B1, B2 y D, transportan la información de sus respectivos canales.

El bit E, es el eco de lo que TR ha recibido en el último bit D.

El bit A, provee un mecanismo de activación y desactivación por señalización dentro de trama.

El bit Fa, es un auxiliar para alineación de trama En el sentido TR a ET. Fa o N aseguran que existirá una violación al código antes del bit 15, ya que uno de los dos siempre será un cero lógico. En el sentido ET a TR, Fa es normalmente un cero lógico y asegura una violación, excepto cuando se utiliza como bit Q (se explica posteriormente). Fa y L siempre tienen el mismo valor lógico.

El bit N, es siempre el complemento lógico de Fa

El bit M, se utiliza para alineación de multitrama, y se explica posteriormente.

El bit S, se encuentra en estudio y provisionalmente se pone a cero

Se utiliza también una estructura de multitrama, con el objeto de proporcionar un canal extra de 800 b/s para señalización de nivel 1, en la dirección ET a TR, utilizando el bit Fa. Cuando se utiliza este canal, el bit se denomina Q La utilización del bit Q y el bit M son opcionales

Se denomina bit Q, al quinto Fa de cinco tramas consecutivas y se identifican en el ET, cuando TR invierte el valor de Fa. Una estructura adicional, que agrupa 4 bits Q, se logra cuando TR transmite el bit M con valor uno lógico cada 20 tramas Esta estructura de multitrama se muestra en la tabla siguiente.

trama número	ET bit Fa	TR bit Fa	TR bit M
1	Q1	1	1
2	0	0	0
3	0	0	0
4	0	0	0
5	0	0	0
6	Q2	1	0
7	0	0	0
8	0	0	0
9	0	0	0
10	0	0	0
11	Q3	1	0
12	0	0	0
13	0	0	0
14	0	0	0
15	0	0	0
16	Q4	1	0
17	0	0	0
18	0	0	0
19	0	0	0
20	0	0	0
1	Q1	1	1
2	0	0	0

Sólo una terminal a la vez, puede transmitir en un canal B, y en general, el lado RED es el encargado de autorizar el acceso al canal. Cuando un canal B no está en uso, el ET debe transmitir unos binarios

La solicitud de acceso, (descrito en las recomendaciones I.450 e I 451), se realiza a través del canal D

Todas las terminales deben estar sincronizadas, en modo esclavo, al terminador de red, de modo que no se interfieran mutuamente.

Cualquier terminal puede transmitir en el canal D, y debe utilizarse algún mecanismo de contensión, para resolver los casos de conflicto, este mecanismo asegura que aun en caso de colisión un equipo logrará transmitir exitosamente

El mecanismo utilizado para el acceso al canal D se apoya en la utilización de un bit de eco (E), en el que TR repite lo que recibe en su canal D, de modo que antes de transmitir el siguiente bit D, todas las terminales deben haber recibido el eco del bit anterior.

Para comenzar a transmitir una terminal debe verificar que el canal D se encuentra libre, o sea esperar la aparición de una "cantidad determinada" de unos El nivel 2 del protocolo del canal D, asegura que nunca aparezca esa cantidad de unos, durante una transmisión

Una vez que se detecta el canal libre, la terminal puede comenzar a transmitir, pero escuchando su propio eco.

Si existiera alguna discrepancia entre el bit transmitido y el recibido en el canal de eco, se detiene inmediatamente la transmisión (pues es evidencia de que simultáneamente más de una terminal comenzó a transmitir) y se espera nuevamente por el indicador de canal libre.

Las características eléctricas de ducto, hacen que un "cero" binario prevalezca sobre un "uno" binario transmitido De modo que, no ocurra nunca una interferencia destructiva y el protocolo de nivel 2, asegura que como máximo al tercer octeto transmitido sólo una terminal estará usando el canal D y podrá terminar su transmisión exitosamente.

Por medio de una asignación de prioridades (la cantidad de unos para decidir canal libre) se asegura el uso equitativo del canal D, para todas las terminales Una vez que un equipo ha terminado una transmisión exitosa, debe esperar un bit más para transmitir nuevamente, y del mismo modo se asegura que la señalización tenga mayor prioridad sobre otro tipo de información

Prioridad	Contenido	Cuenta Normal	Cuenta Larga
1	señalización	8	9
2	no señalización	10	11

Una vez que se detecta la ocurrencia de la cuenta larga, o sea que todos los ET han tenido oportunidad de transmitir en el canal D, las terminales regresan su prioridad a la cuenta normal y pueden volver a transmitir

Las características de la interfaz de acceso básico pueden resumirse en

- Transmisión en 4 hilos, acoplamiento con transformador
- Velocidad nominal de transmisión 192 Kb/s
- Longitud de trama 48 bits
- Código de Inversión de linea Alternada de Espacios (ASI) con un 100% de ciclo útil

binario	codificado ASI
0	+0.75 V o -0.75 V
1	0 V

- Sincronia de trama por violaciones al código de línea (dos ceros binarios con la misma polaridad) al inicio de cada trama.
- Nivel de los pulsos 750 mV pico, los ceros binarios prevalecen sobre los unos binarios
- Alimentación en varias configuraciones (-40V)
- Consumo (alimentados de la fuente 1 en estado limitado)

  máximo activo: 380 mW
  máximo inactivo: 100 mW

- Activación y desactivación por señalización dentro de la trama (bit A).
- Configuraciones: punto a punto, ducto pasivo corto y ducto pasivo extendido.

Como puede observarse la estructura de la trama no es simétrica, en una dirección TR transmite un bit de paridad al final de cada trama, mientras que en la dirección opuesta, cada ET es responsable de transmitir un bit de paridad en cada campo de la trama que esté utilizando

2.2.1.4. ACTIVACION Y DESACTIVACION

Hay mecanismos de activación y desactivación que permiten minimizar el consumo de potencia de los dispositivos cuando no hay comunicación en curso. Los cambios de estado se dan de acuerdo a ciertos mensajes recibidos por la entidad de capa 1, ya sea mediante primitivas de capas superiores o a través de señales especiales que se transmiten por la linea de interconexión entre el ET y el TR

La comunicación entre la capa 1 y la capa 2 para efectos de activación y de desactivación se establece mediante las primitivas:

- Petición FI-ACTIVACION (FI-AR).
- Indicación FI-ACTIVACION (FI-AI).
- Indicación FI-DESACTIVACION (FI-DI)

La comunicación entre la capa 1 y la entidad de gestión se establece mediante las primitivas

- Indicación GFI-ACTIVACION (GFI-AI).
- Petición GFI-DESACTIVACION (GFI-DR)
- Indicación GFI-ERROR (GFI-EI).

Las señales que se usan para controlar los procedimientos de activación-desactivación, conocidas como señales INFO, se muestran en la Tabla 2-2

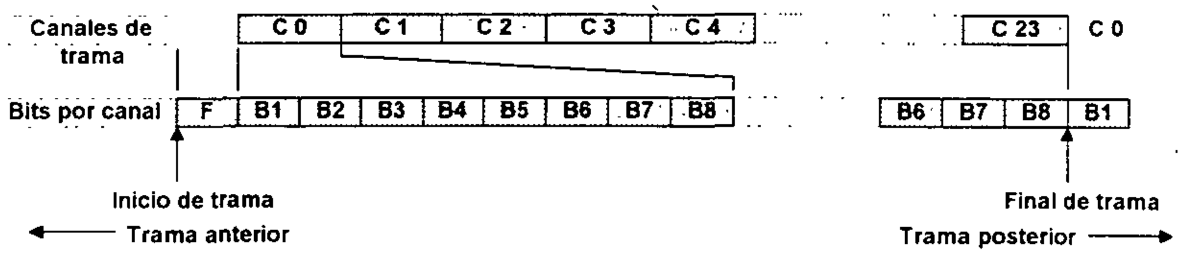
***Tabla 2-2** Señales INFO, para la activación y desactivación del nivel físico de la interfaz "S"*

NOMBRE	DEFINICION	DIRECCION
INFO 0	Ausencia de señal.	ET ↔ TR
INFO 1	Señal continua a una velocidad de 192 Kbps y con el siguiente esquema cíclico cero positivo, cero negativo y seis unos	ET → TR
INFO 2	Trama con todos los bits de los canales B, D y E (eco de canal D) puestos a cero El bit A se pone también a cero	ET ← TR
INFO 3	Trama sincronizada y con datos operacionales en los canales B y D	ET → TR
INFO 4	Trama con datos operacionales en los canales B, D y E (eco de canal D) El bit A se pone a uno	ET ← TR

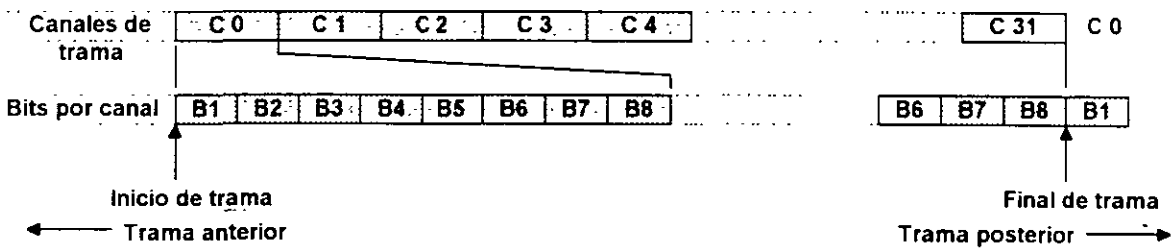
2.2.2. INTERFAZ DE ACCESO PRIMARIO (PRI).

Como se vió anteriormente el acceso básico ofrece un servicio de 64 Kbps ya sea de voz o datos Este limitado ancho de banda no es suficiente para la comunicación entre dos oficinas terminales, o inclusive entre un conmutador privado y una oficina terminal, esto hace necesario la utilización de una interfaz con un mayor ancho de banda, esta interfaz es la que se conoce como Acceso Primario (PRI)

Actualmente, existen dos tipos de Accesos Primarios. El PRI Europeo usa 30 canales B y un canal D a 64 Kbps cada uno (más un overhead de 64 Kbps) para una velocidad total de 2 048 Mbps y se le llama CEPT, en EUA, Corea del Sur, y Japón, el PRI funciona a 1.544 Mbps (23 canales B y un canal D a 64 Kbps cada uno más overhead de 8 Kbps) y se conoce como T1. El overhead para ambos PRIs sirve para funciones tales como sincronización de trama y administración de red



a) Interfaz a 1.544 Mbps (193 bits; 125 μs)



b) Interfaz a 2 048 Mbps (256 bits; 125 μs)

2.3. INTERFAZ U

Este punto de acceso a la RDSI no esta normalizado por el CCITT. por lo que cada administración define la técnica de transmisión. el código de línea y las características físicas de la interfaz

Por razones económicas el actual par de hilos de cobre que llegan a la casa del usuario telefónico deben ser utilizados para transportar la información de los servicios ofrecidos por la RDSI. es por esto que la linea de abonado debe permitir transmitir 160 Kbps (144 Kbps de los canales 2B+D más bits extras para información de mantenimiento alineación. etc.) en forma "full-duplex".

En el diseño de esta interfaz se tienen básicamente 2 problemas

- Transmisión "full-duplex" en 2 hilos de información digital
- Velocidad de transmisión en la línea es de 160 Kbps

El primer problema se resuelve utilizando una técnica adecuada de transmisión y el segundo tratando de reducir la velocidad con un código de línea que además permita aprovechar las características de transmisión que presenta el par de hilos de cobre

2.3.1. TECNICAS DE TRANSMISION EN LA LINEA DE ABONADO (INTERFAZ U)

2.3.1.1. TRANSMISION A 4 HILOS

Esta técnica de transmisión no tiene posibilidades en la práctica ya que todos los subscriptores existentes en la actual red telefónica se conectan con un solo par. Solamente se conectan a 4 hilos cuando la conexión es de 2.048 Mbps (por ejemplo la conexión de un PABX). Véase Figura 2 7

2.3.1.2. DIVISION DE FRECUENCIA

Con la técnica de división en frecuencia es posible transmitir en forma "full-duplex". sin embargo las señales digitales codificadas enviadas por la línea se traslapan en su densidad espectral Para evitar este problema se usan diferentes códigos de línea en cada dirección (por ejemplo código bipolar de orden 1 en una dirección y de orden 2 en la otra dirección) o
usando el mismo código en ambas direcciones pero modulando la información transmitida en una de las direcciones

La separación de la información en el lado de recepción es realizada mediante filtros La distancia que se puede alcanzar esta condicionada por las señales de alta frecuencia que tengan gran cantidad de energía. debido a la diafonía en el lado lejano (FEXT. Far-end crosstalk). la cual es producida por líneas adyacentes de diferente longitud Las señales de alta frecuencia son transmitidas en la dirección de la central al subscriptor

Una de las ventajas de esta técnica es que la diafonía en el lado cercano (NEXT. Near-end crosstalk) es minimizada debido a que los espectros para transmitir y recibir son diferentes. sin embargo el diseño de los filtros es complejo y su implementación en circuitos integrados digitales presenta problemas. Además no es posible utilizar el mismo equipo en la central y en el subscriptor debido a la asimetría en la transmisión. por lo que esta técnica ha sido abandonada. Véase Figura 2 7.

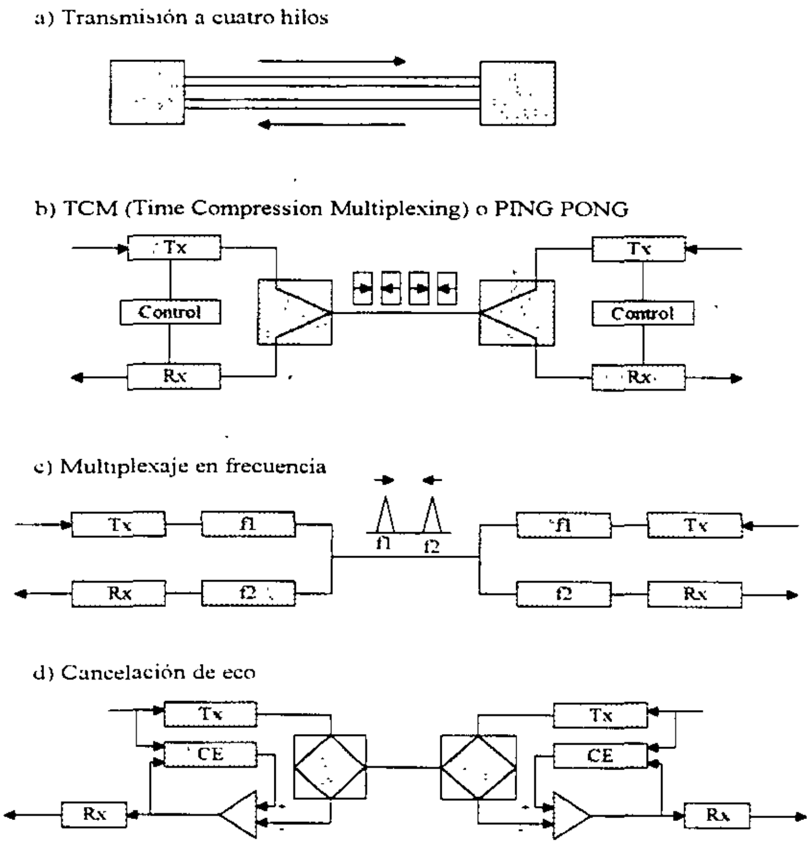


Figura 2. 7 Métodos de transmisión en la línea de abonado (interfaz U)

2.3.1.3. TCM (Time Compression Multiplexing) o PING PONG

Este método también llamado de ráfagas. involucra el cambio alternado de la dirección de transmisión Esta alternación en la transmisión. no es en el sentido de la transmisión "half-duplex " sino que esta técnica garantiza que efectivamente haya una transmisión "full-duplex ". aunque a nivel microscópico esto sea "half-duplex " dado que el transmisor y receptor transmiten en tiempos diferentes La información binaria es almacenada en forma de bloques en los extremos del enlace y son transmitidos en intervalos de tiempo diferentes. Por lo tanto existen dos fases que no deben traslaparse transmisión y recepción; que pueden ser distinguidas en cada extremo del enlace

Por lo tanto para una velocidad de información D. la velocidad de línea requerida debe ser mínimo 2D. de hecho considerando la propagación en los cables y el tiempo utilizado entre las diferentes fases dan una velocidad del orden de 2.5D.

La distancia teórica máxima está dada por:

$$L\max = \frac{V}{2(N/D - 2N/F - 2Th)}$$

Donde.

V = Velocidad de propagación en los cables (aproximadamente 200.000 Km/s)
N = Número de elementos binarios en el bloque
F = Velocidad de línea
th = Tiempo de guarda (para evitar interferencia entre la transmisión

Bloques de longitud muy grande reducen el número de veces que se debe alternar la dirección de transmisión y con ello el efecto de la propagación para de esta forma incrementar la longitud teórica. sin embargo para señales de voz el retardo de los octetos produce degradación en la calidad.

Una longitud teórica grande es también obtenida aumentando la velocidad de transmisión pero esta se ve limitada por la atenuación y la diafonía que presenta el par de hilos de cobre

2.3.1.4. CANCELACION DE ECO

Este método es utilizado actualmente en transmisión analógica en bajas frecuencias para proporcionar transmisión "full-duplex" por un par. utilizando un acoplador (bobina híbrida) de dos a cuatro hilos con una impedancia balanceada que representa un compromiso entre las impedancias representadas por ambas líneas De hecho en la híbrida la red balanceada colocada en el lado del medio de transmisión produce un desacoplo y permite que algunas de las señales transmitidas regresen junto con las señales recibidas. a este fenómeno se le conoce como eco local

La atenuación de la trayectoria del eco para un ancho de banda aproximadamente 100 KHz es del orden de 10 a 15 dB pero puede caer hasta 6 dB para configuraciones de cable específicas. Un receptor digital solo funciona correctamente para una relación señal a ruido de aproximadamente +25 dB. Dado que se requiere para un sistema de transmisión digital de aproximadamente 45 dB a 100 KHz. la señal remota es atenuada por el valor correspondiente Por lo tanto es necesario reducir el eco local aproximadamente 64 dB (45dB + 25dB - 6dB) para que los datos sean detectados correctamente. El eco remoto de pequeña amplitud debido al desacoplo de impedancias a lo largo de la línea es sumado al eco local

Para eliminar la señal producida por dicho desacoplo de impedancias. se ha diseñado un dispositivo que elimina el eco usando la información transmitida. llamado "Cancelador de eco ". De hecho el eco es resultado de la configuración intrinseca de la línea de abonado y de las características de los símbolos (código de línea) que están siendo transmitidos sobre ella. Este dispositivo hace uso del principio de que no exista una correlación entre el eco y la señal que proviene del lado remoto. para este efecto se usan diferentes aleatorizadores (scramblers) en cada uno de los extremos de la línea. Además el circuito que realiza las funciones de procesamiento de señales debe ser flexible para aceptar todas las posibles configuraciones de una línea de subscriptor en una red telefónica y responder a cualquier variación en sus características con el tiempo.

Existen básicamente dos métodos para estimar el eco. uno usa un filtro transversal y el otro esencialmente usa memorias.

En el primer método el filtro contiene N (el cual puede alcanzar varias decenas) coeficientes variables que representa la respuesta al impulso del eco muestreado. La multiplicación de estos coeficientes con la secuencia de los datos transmitidos producen la perturbación instantánea debida al eco la cual es calculada cada vez que se transmite un símbolo. Los coeficientes del cancelador de eco son ajustados para reducir el error residual que resulta de una mala estimación del eco real. Se puede demostrar que la diferencia entre el eco real y el eco estimado puede ser expresado estadísticamente. tomando en consideración

la no correlación de la señal. como una función de los datos transmitidos y del total de la señal recibida (estos parámetros se obtienen del sistema de recepción). Por lo tanto es posible minimizar este error usando algoritmos de mayor o menor grado de complejidad (del gradiente o tipo de signo) el cual asegura una convergencia progresiva del cancelador de eco. Este método implícitamente asume que el eco del canal es lineal y que cualquier no linealidad está fuera del rango de operación del cancelador. lo cual implica que cualquier no linealidad en la codificación sean excluídas de la trayectoria del eco Sin embargo otras no linealidades pueden aparecer como desbalanceo en el transmisor o no linealidad del convertidor analógico-digital

El segundo método. usa memorias que contienen el eco que ha sido previamente calculado para todas las posibles secuencias de información con lo cual se puede compensar las no linealidades. Si se asume que el eco puede ser modelado mediante un filtro de N coeficientes para N datos binarios sucesivos. el eco solo puede tomar 2N valores y por lo tanto es suficiente que los N elementos binarios sean usados para direccionar una memoria cuyo contenido varía en función de error residual de la señal. La gran cantidad de memorias y los grandes tiempos de convergencia son las principales desventajas de este método

Consecuentemente estructuras intermedias han sido diseñadas. por ejemplo M memorias con $2^{N/M}$ palabras cuyos contenidos son sumados para producir el eco. para esto se debe establecer un compromiso entre robustez a la no linealidad. la velocidad de cálculo y el tiempo de convergencia

La principal ventaja del cancelador de eco es la preservación de espectro en frecuencia correspondiente en banda base. Sin embargo es importante evitar códigos de línea con mucha energía en las bajas frecuencias para asegurar una buena robustez contra el ruido de la red local. que por lo general ocurre en la banda de 0 a 20 KHz

Por lo antes descrito es conveniente usar códigos de línea para este método de transmisión. que sean lineales y que sean invariantes con respecto al tiempo en el proceso de almacenamiento de las respuestas al impulso Algunos de los códigos con esta características son el bifase. bipolar. 4B3T y 2B1Q El código determina la complejidad de su implementación en Circuitos Integrados. por ejemplo un CI de transmisión que contenga cancelación. ecualización. recuperación de la temporización y activación pueden contener hasta 50.000 transistores. pero se puede disminuir esta cantidad realizando una adecuada selección del código.

Después de que el eco ha sido estimado. se elimina (mediante una operación de sustracción) y en ese momento generalmente la señal es manejada como una transmisión a 4 hilos. sin embargo es necesario realizar filtrados adicionales para reducir la interferencia entre símbolos. La velocidad de convergencia del sistema cancelador de eco es un elemento clave en el tiempo de establecimiento de la comunicación Cuando el sistema ignora por completo las características de la línea. el tiempo de convergencia de arrancando desde un estado aleatorio los coeficientes; puede tomar algunos segundos. sin embargo si los coeficientes son almacenados entre una comunicación y otra. el tiempo de convergencia no excede los 100 ms. Vease Figura 2.7.

Una vez que ya se tiene un panorama general de lo que es la Red Digital de Servicios Integrados. en la Figura 2 8 se muestra el modelo RDSI en el que se pueden observar los 2 tipos de interfaz de acceso a la RDSI. asi como los grupos funcionales. y los puntos de referencia.

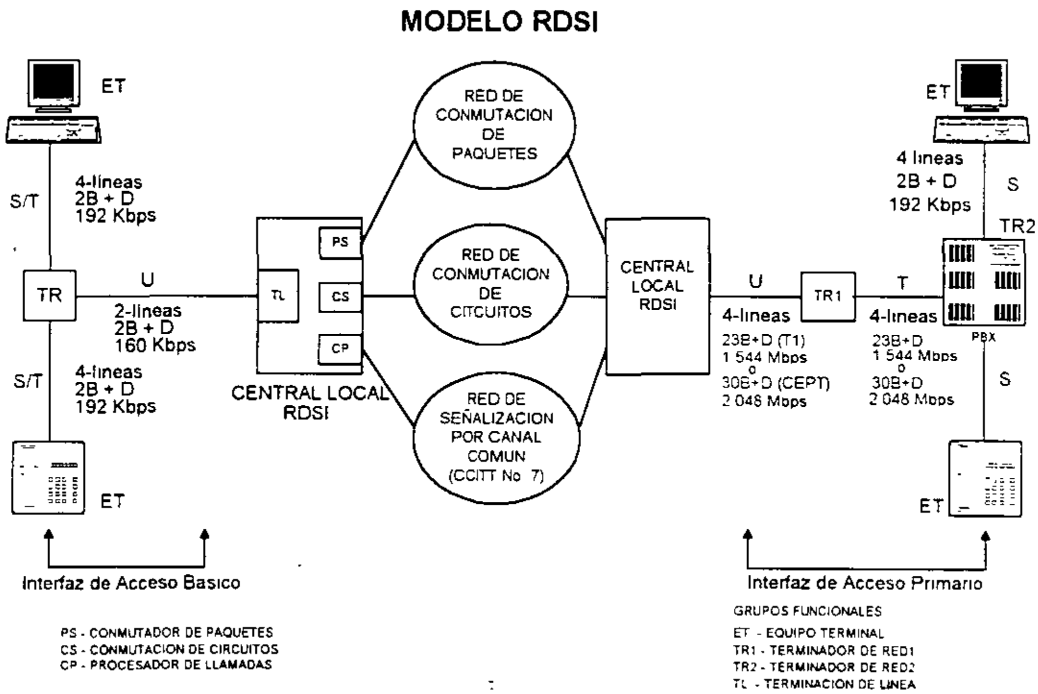


Figura 2 8 Modelo RDSI

3. CAPA DE ENLACE DE DATOS.

La capa 2 para el canal D es definida en las recomendaciones I.440 (Q.920) e I.441 (Q.921) del CCITT Estos protocolos reciben comúnmente el nombre de LAPD (Procedimiento de Acceso al Enlace en el canal D) y tienen como finalidad controlar el intercambio de información entre las entidades pares de capa 3 a través de la interfaz usuario-red. También controlan la interacción de la capa de enlace de datos (capa 2) con la capa de red (capa 3) y la capa física (capa 1). La estructura de trama LAPD para el canal D se muestra en la Figura 3.1. [Terpán, 1993]

Información de Capa de Enlace de Datos

Información de Capa de Red

Bandera	Dirección	Control	Información	SVT	Bandera
0 1 1 1 1 1 1 0 1 octeto	2 octetos	1 ó 2 octetos	Máximo 260 octetos	2 octetos	0 1 1 1 1 1 1 0 1 octeto

***Figura 3.1** Estructura de trama LAPD para el canal D*

El protocolo LAPD provee los siguientes servicios a la capa de red·

- Presta servicios a varias entidades de capa 3, las cuales se diferencian entre sí por medio del campo de dirección de la trama de capa 2.
- Proporciona delimitación de tramas por medio de banderas HDLC (01111110) y transparencia en la transmisión de la información por medio de la inserción y extracción de ceros para asegurarse que no se repita, de manera involuntaria, la secuencia de bandera y ésta se pueda interpretar como un mensaje erróneo. Este procedimiento inserta un cero después de cada 5 unos consecutivos.
- Proporciona un mecanismo de control de secuencia para garantizar el orden de las tramas transportadas a través de la interfaz
- Proporciona procedimientos de detección y recuperación de errores en la conexión de capa 2.
- Proporciona control de flujo manejando tramas que solicitan la suspensión temporal o la reanudación del envío de tramas de información y proporciona control de error a través del acuse de recibo de tramas recibidas exitosamente solicitando retransmisión de tramas recibidas con error.

El campo de dirección de la estructura de trama identifica, en 16 bits, el origen o destino de la trama por medio del Identificador de Punto de Acceso al Servicios (SAPI) y del Identificador de Punto Extremo Terminal (TEI); define, asimismo, si la trama corresponde a una instrucción o a una respuesta (C/R). El SAPI tomará un valor 0 para la interacción con la capa de red y un valor 63 para la interacción con la entidad de gestión. El TEI puede tomar valores entre 0 y 127, siendo los 64 primeros (0-63) asignados de manera no automática; del 64 al 126 asignados automáticamente y el 127 usado para difusión. (en enlaces punto a multipunto) [Gallardo, 1991].

El campo de control puede tener 3 formatos distintos:

- **Tramas I de información numerada:** este formato es utilizado para la transferencia de información proveniente de capa 3. Utiliza contadores para llevar una secuencia de tramas enviadas y una secuencia de tramas recibidas sin error.
- **Tramas S** que manejan funciones de supervisión: con este tipo de tramas se acusa recibo, se pide una retransmisión o se solicita la suspensión temporal del envío de tramas I.
- **Tramas U (no-numeradas):** se utilizan para la transmisión de información no numerada para realizar funciones de control de enlace de datos.

La longitud del campo de control es de 2 octetos para los formatos I y S, siendo de un octeto para el formato U.

El campo de información se encuentra presente en todas las tramas I y en las tramas UI (tramas de Información No-numeradas). Cuando este campo existe es de longitud variable, de un máximo de 260 octetos, y contiene información de capa 3.

El campo de Secuencia de Verificación de Trama (SVT) lleva información del Código de Redundancia Cíclica de 16 bits (CRC-16) definido por el CCITT y calculado de acuerdo al polinomio generador $x^{16} + x^{5} + x^{2} + 1$.

4. CAPA DE RED.

4.1. Aspectos Generales

Las especificaciones generales de esta capa se definen en la recomendación Q 930(I.450) del CCITT La descripción detallada se define en la recomendación Q.931(I 451) del CCITT

El protocolo de capa 3 proporciona los medios para establecer. mantener y terminar conexiones de la red en una RDSI entre entidades de aplicación. Además proporciona funciones de enrutamiento y direccionamiento.

Los tipos posibles de conexiones son:

- Conexiones por conmutación de circuitos utilizando los canales B.
- Conexiones para señalización entre usuarios utilizando el canal D
- Conexiones por conmutación de paquetes usando el canal D. o el canal B

4.2. Funciones de la capa 3

Las funciones soportan procedimientos para el control de una llamada básica. y para el control de una llamada en conjunto con servicios adicionales proporcionados por la red Se dividen en dos categorias. La primera contiene aquellas funciones que controlan directamente el establecimiento de las conexiones. y la segunda incluye las funciones relacionadas con el transporte de mensajes adicionales a las funciones proporcionadas por la capa 2.

Dentro de las funciones más importantes se pueden listar las siguientes

- Proceso de primitivas para comunicarse con la capa 2
- Generación e interpretación de mensajes de capa 3 para la comunicación entre entidades del mismo nivel
- Administrador de temporizadores y entidades lógicas
- Administración de acceso a recursos de la red
- Enrutamiento y retransmisión
- Control de conexión de red
- Detección de error
- Secuenciación
- Reinicio

4.3. Procedimientos para el control de llamadas

4.3.1. Procedimientos para llamadas por conmutación de circuitos.

Este tipo de conexiones se controlan mediante el intercambio de mensajes de capa 3 (ver Tabla 4-1) entre las dos entidades del mismo nivel. Estos mensajes se envían por el canal D, y permiten la asignación de un canal B para el envío del flujo de información.

4.3.2. Procedimientos para llamadas por conmutación de paquetes.

4.3.2.1. Servicio por conmutación de paquetes utilizando el canal B.

La RDSI proporciona un canal B en una conexión semipermanente o conmutada entre una terminal de usuario, y la función de manejo de paquetes de la RDSI.

Para lograr la conexión conmutada primeramente se utiliza la señalización normal RDSI para el establecimiento de un enlace conmutado.

Posteriormente se utilizará el canal B para el envío de paquetes de acuerdo a los protocolos de capa 2 y 3 de X 25

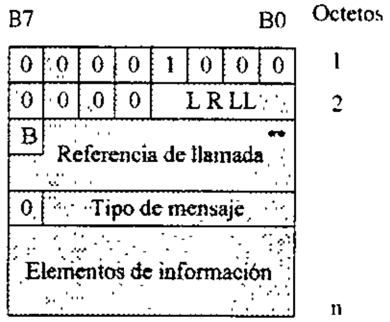
4.3.2.2. Servicio por conmutación de paquetes utilizando el canal D.

El canal D permite a las terminales de usuarios RDSI accesar a la función de manejo de paquetes RDSI estableciendo una conexión de enlace de datos a esa función la cual puede ser utilizada posteriormente para soportar comunicación de paquetes de acuerdo al protocolo de capa 3 de X 25.

4.4. Estructura de los mensajes

La estructura de los mensajes consiste de elementos comunes a todos los tipos de mensajes, y de elementos de información obligatorios y adicionales los cuales son específicos a cada tipo de mensaje.

El campo de información de la trama de capa 2 contiene al protocolo de capa 3. El formato de los mensajes se muestra en la Figura 4 1.



L R LL = Long de referencia de llamada
B = bandera de ref de llamada
0 = Mensaje enviado del lado origen
1 = Mensaje enviado del lado destino

** 1 octeto para acceso básico
2 octetos para acceso primario

Figura 4.1 *Formato de los mensajes de capa 3*

El discriminador de protocolos identifica el protocolo de capa 3. Este protocolo puede ser uno especificado por el CCITT. o cualquier otro protocolo. A este campo le siguen una serie de cuatro ceros. después sigue el campo que indica la longitud que tendrá el campo de referencia

Después aparece el campo de referencia de llamada el cual se utiliza para identificar cada llamada en la interfaz usuario-red local Los valores de este campo los asigna la entidad origen al inicio de cada llamada Este campo se remueve una vez que se ha completado o suspendido la llamada.

El campo de tipo de mensaje es un octeto que permite identificar la función del mensaje que se envía. los diferentes mensajes son los mostrados en las Tabla 4-1 y Tabla 4-2.

Al final aparece el campo de elementos obligatorios o elementos adicionales de información el cual identifica cada uno de los elementos de información posibles que son necesarios en cada mensaje

Tabla 4-1 Mensaje de capa 3 para el control de llamadas en conmutación de circuitos

87654321 000 - - - - 00001 00010 00111 01111 00011 00101 01101	MENSAJES PARA EL ESTABLECIMIENTO DE LA LLAMADA - ALERTing - CALL PROCeeding - CONNect - CONNect ACKnowledge - PROGress - SETUP - SETUP ACKnowledge
000 - - - - - 00110 01110 00010 00101 01101 00001 00000	MENSAJES DURANTE LA FASE ACTIVA DE LA LLAMADA - RESume - RESUME ACKnowledge - RESume REJect - SUSPend - SUSPend ACKnowledge - SUSPend REJect - USER INFOrmation
010 - - - - - 00101 01101 11010	MENSAJES PARA LA TERMINACION DE LA LLAMADA - DISConnect - RELease - RELease COMplete
011 - - - - - 11001 00010 11011 01110 11101 10101	MENSAJES DIVERSOS - CONgestion CONtrol - FACility - INFOrmation - NOTIFY - STATUS - STATUS ENQuiry

Tabla 4-2 Mensaje de capa 3 para el control de llamadas en conmutación de paquetes

87654321 000 - - - - 00001 00010 00111 01111 00011 00101	MENSAJES PARA EL ESTABLECIMIENTO DE LA LLAMADA - ALERTing - CALL PROCeeding - CONNect - CONNect ACKnowledge - PROGress - SETUP
010 - - - - 00101 01101 11010	MENSAJES PARA LA TERMINACION DE LA LLAMADA - DISConnect - RELease - RELease COMplete
011 - - - - 11101 10101	MENSAJES DIVERSOS - STATUS - STATUS ENQuiry

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FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

RDSI
INTERFACES S & U

EXPOSITOR: ING. JUAN CARLOS MALDONADO MEDINA
PALACIO DE MINERÍA
JUNIO DE 1999

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ALCATEL

RDSI

Interfaces S y U

Indice

- Conceptos Básicos
- Puntos de Referencia
- Interfaces
- Ejemplos de Aplicaciones

RDSI: Conceptos Básicos

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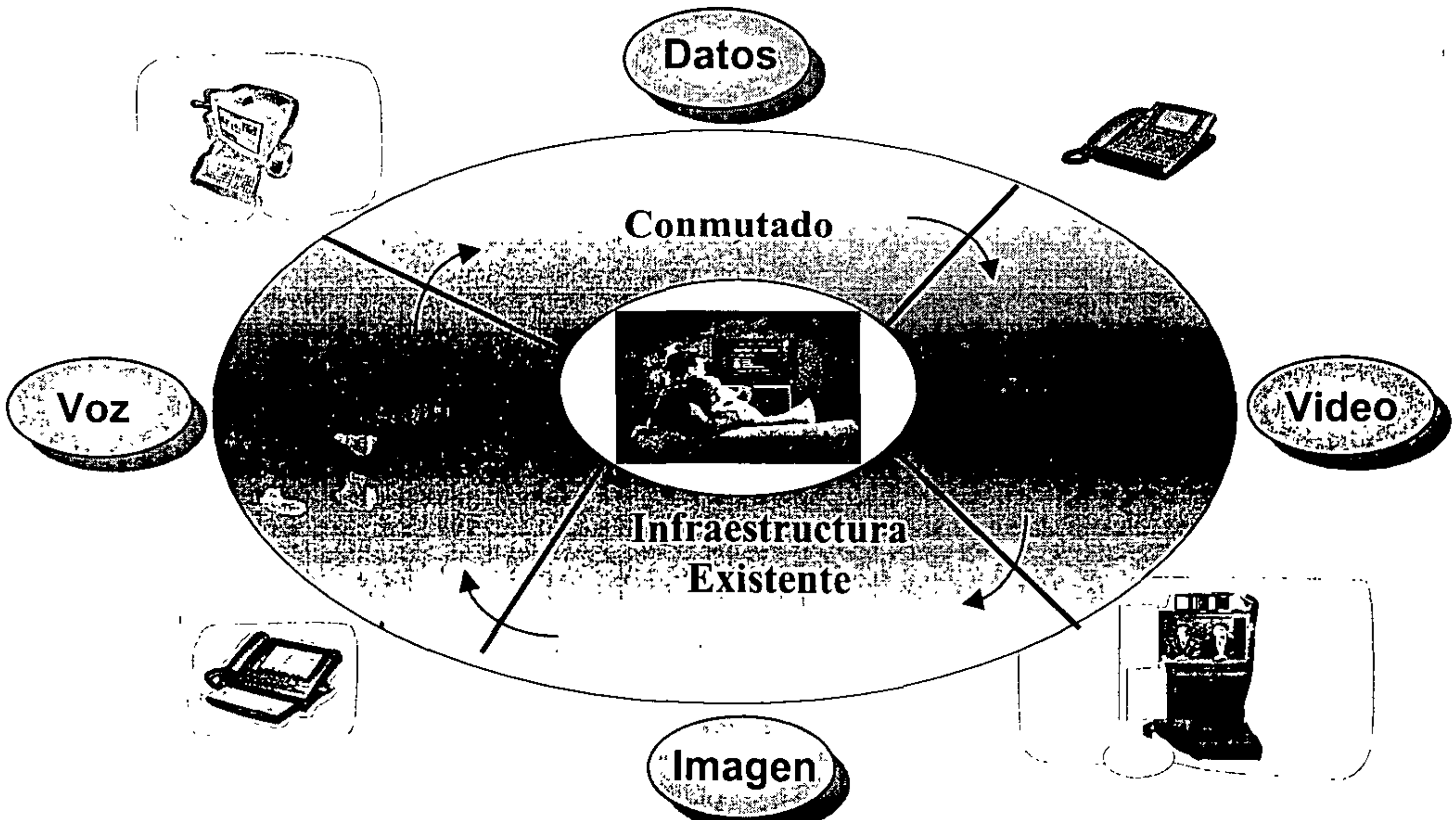
Red Digital de Servicios Integrados

La RDSI es el Medio para Integrar Servicios de Voz, Datos, Video y Audio en forma Conmutada y Totalmente Digital, utilizando la Infraestructura Telefónica Existente.



"Tan Simple como hacer una llamada."

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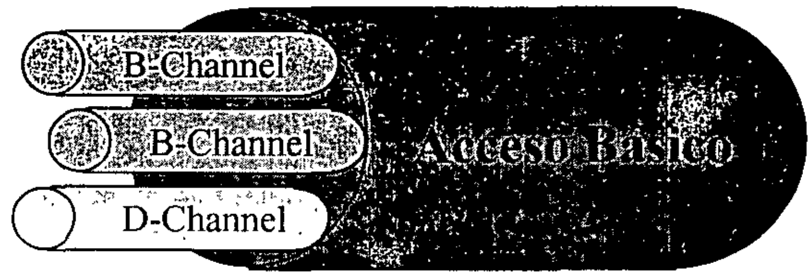
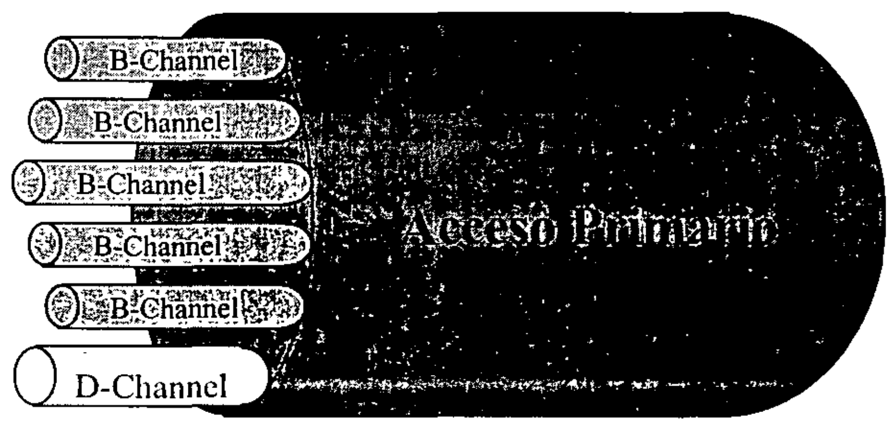
Accesos a la RDSI

- ▼ Acceso Basico
  - 144 Kbit/s
  - 2B + D



- ▼ Acceso Primario
  - ETSI : 30 B+D
    2048 Kbit/s
  - USA : 23 B+D
    1544 Kbit/s



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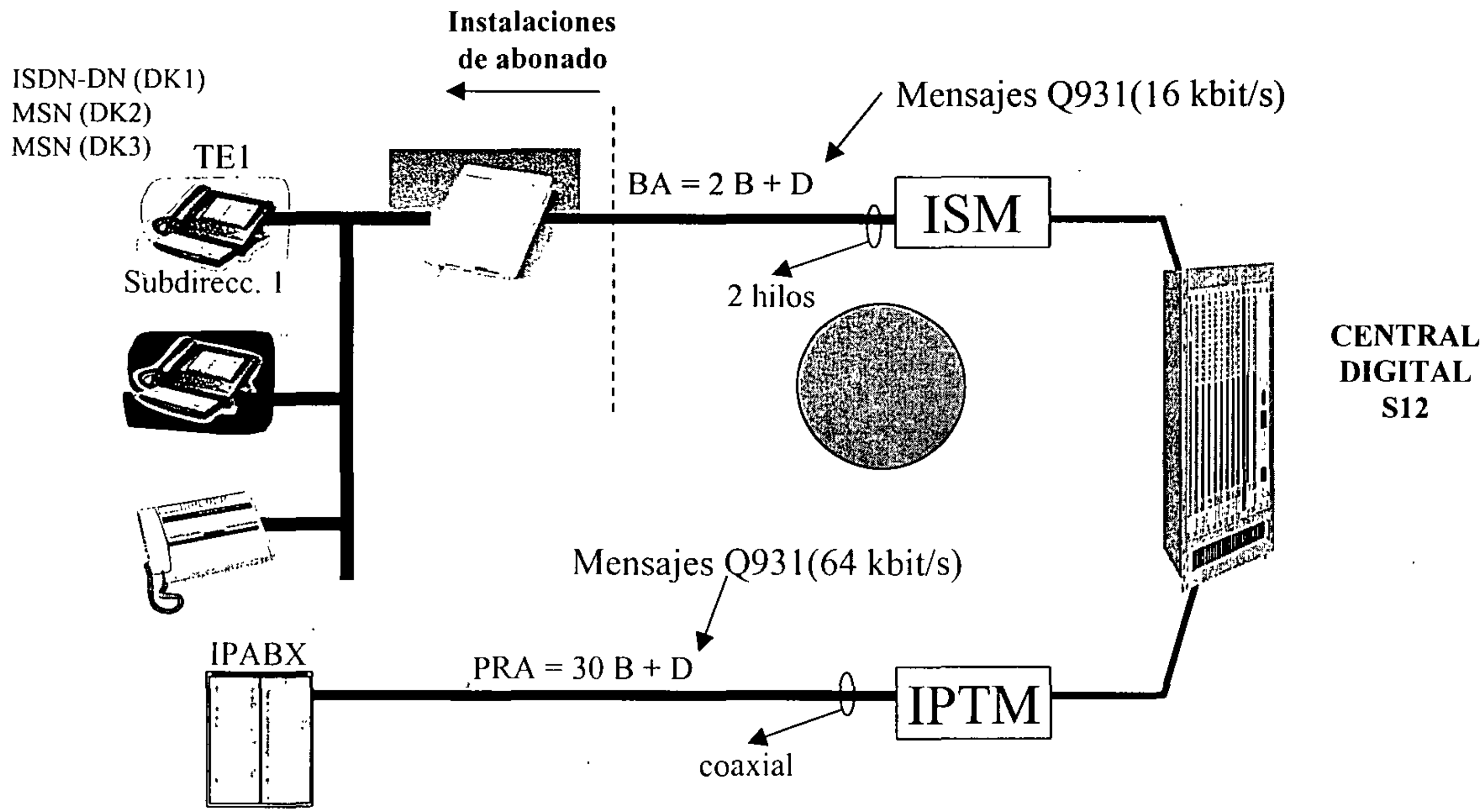
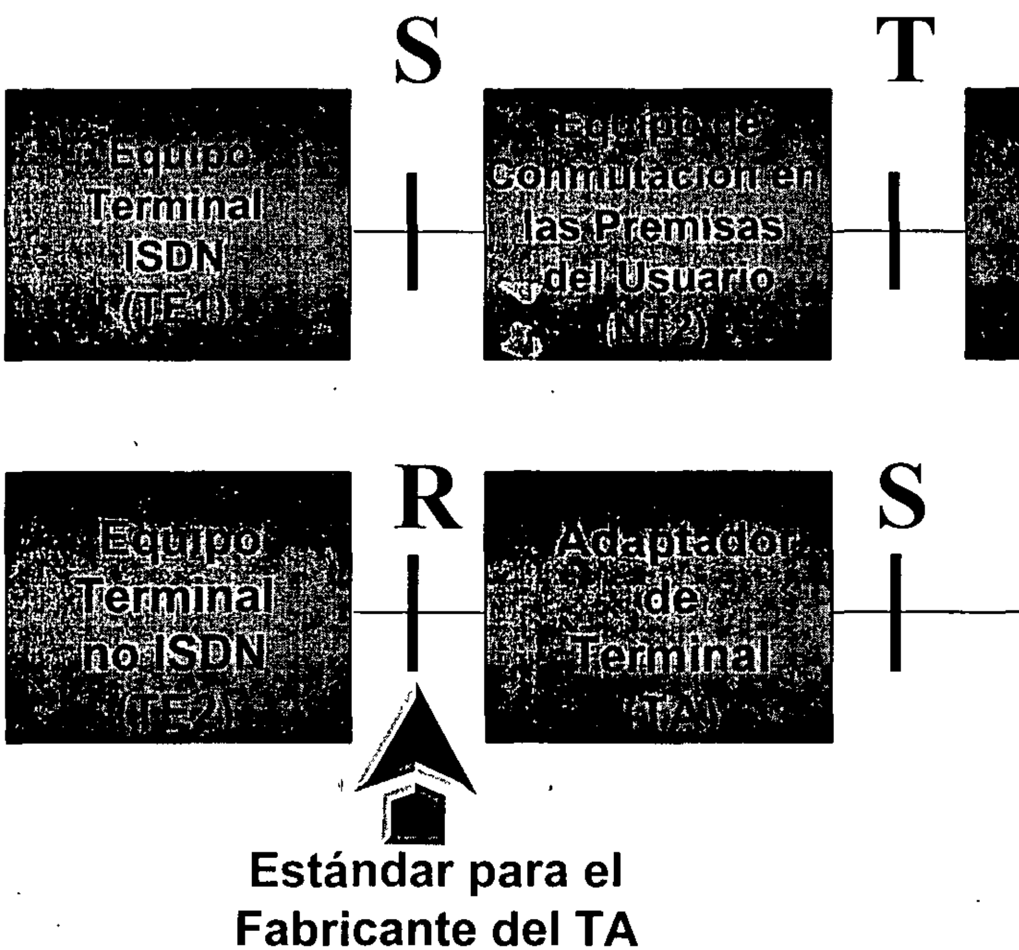
Conexión de Accesos Básico y Primario



RDSI: Puntos de Referencia

Puntos de Referencia RDSI



Puntos de Referencia RDSI

Punto de Referencia R: está entre equipo terminal no-ISDN (TE2) y un TA.

Punto de Referencia S: está entre el equipo de usuario ISDN (TE1 o TA) y el equipo de terminación de red (NT2 o NT1).

Punto de Referencia T: está entre el equipo de conmutación del lado usuario (NT2) y el terminador del bucle local (NT1). En ausencia del NT2, la interfaz usuario-red es comúnmente llamada el punto de referencia S/T.

Punto de Referencia U: está entre el NT1 y la central local ISDN.

Aunque no se muestra en la figura, algunos fabricantes definen un Punto de Referencia V entre el LT y el ET dentro de la central local. Este punto de referencia es una característica dependiente de la implementación de la central, la cual es transparente al usuario.

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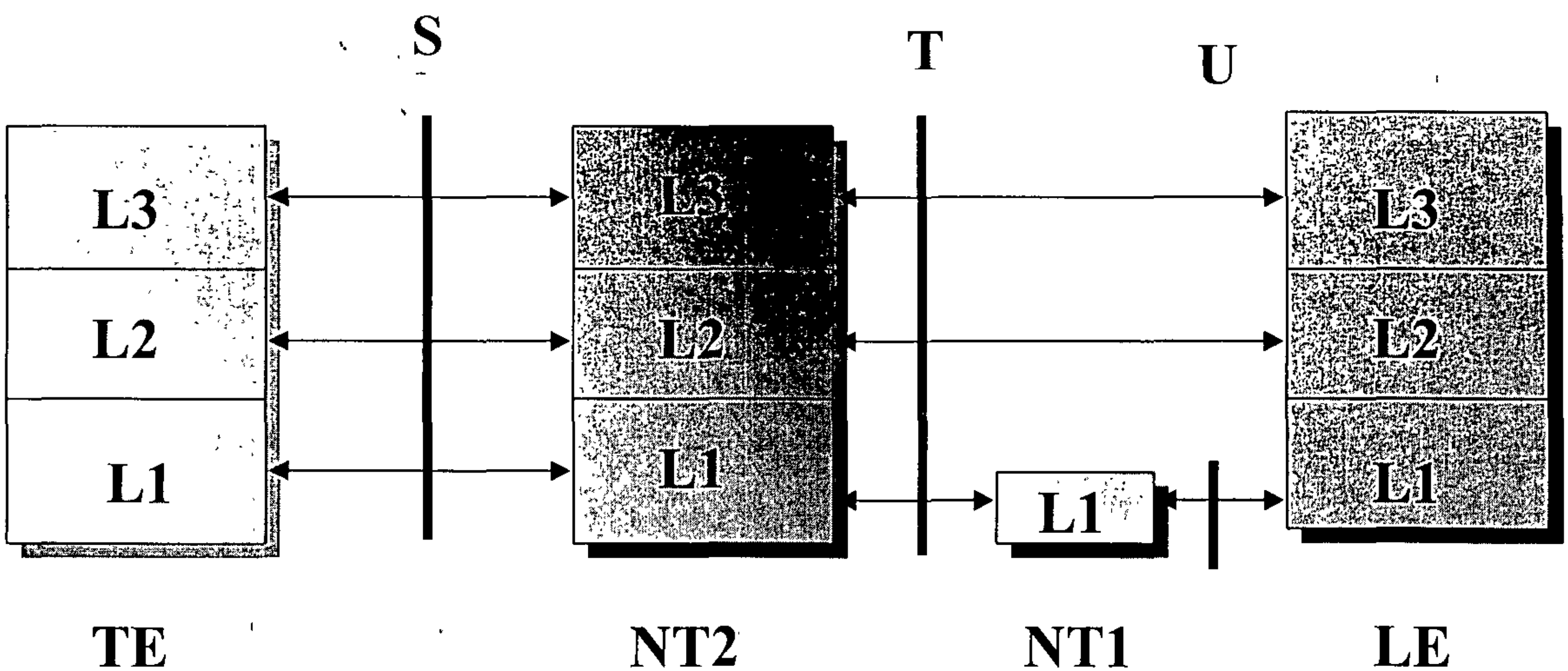
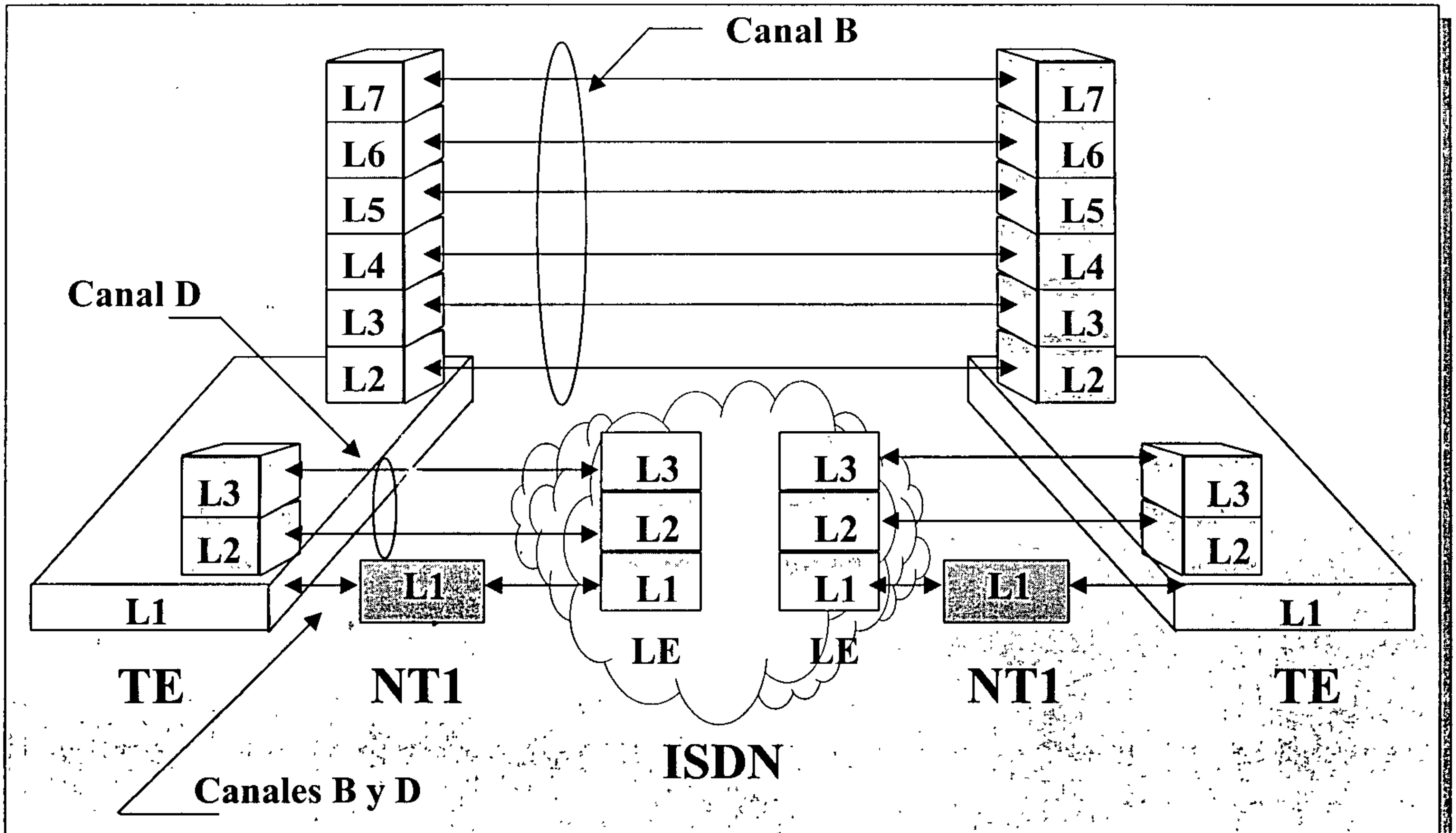
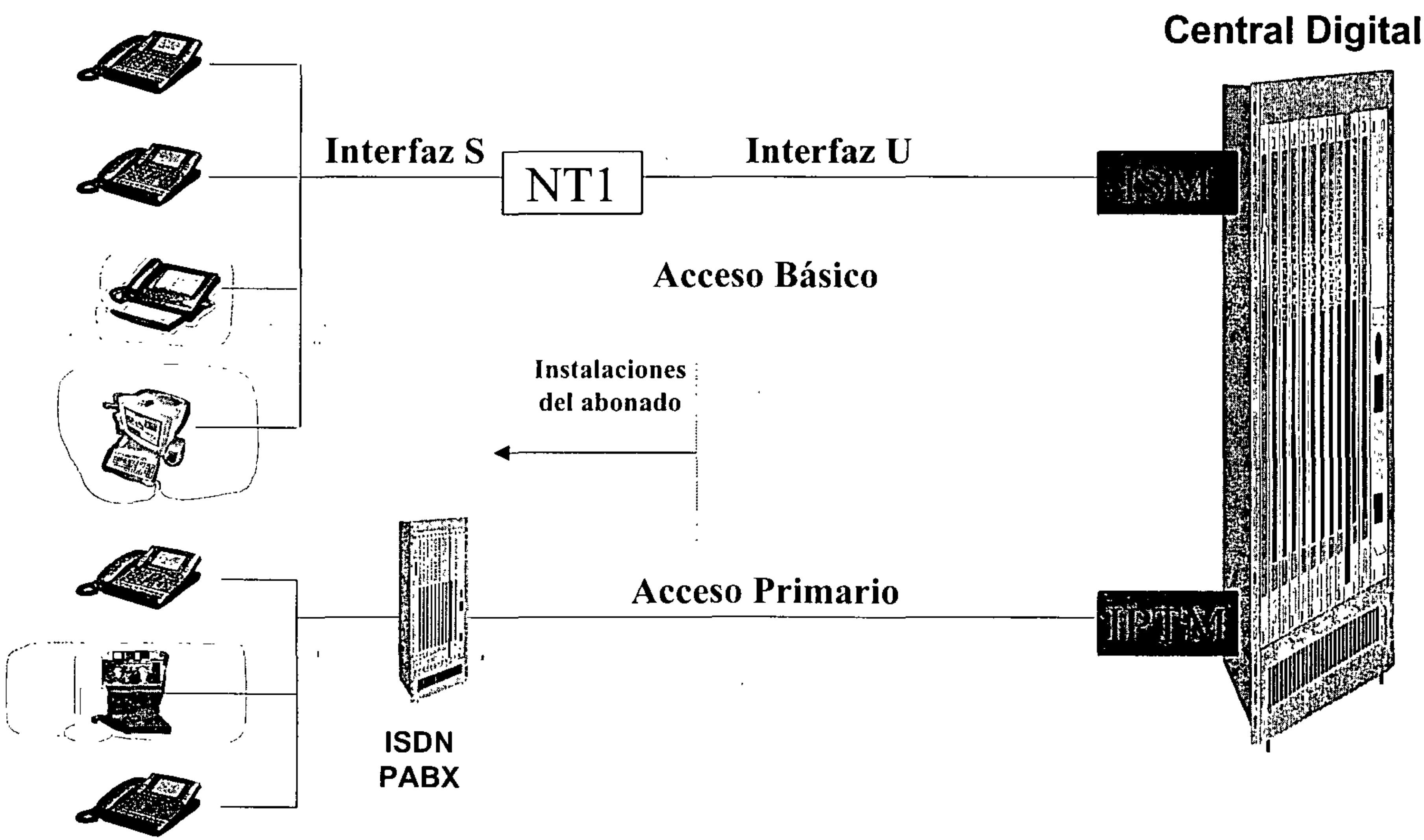
Puntos de Referencia y Arquitectura para el Canal D



Arquitectura de Protocolos de los Canales B y D



Ejemplo de la RDSI Básica



La Interfaz S

- La interfaz S tiene una configuración a cuatro hilos y permite la conexión de hasta ocho terminales RDSI (TE's).
- Una fuente de energía "fantasma" puede ser transportada entre los pares de transmisión y recepción para terminales que requieran energía.
- El código de línea en la interfaz S es el 2B1Q, el cual es una variación del código AMI, donde un "1" digital es representado por un nivel de 0V, mientras que un "0" digital es representado por un pulso, ya sea positivo o negativo.
- La impedancia nominal de la interfaz S es de 100 ohms.
- Generalmente el bus S es terminado con una resistencia de 100 ohms en cada par de hilos.

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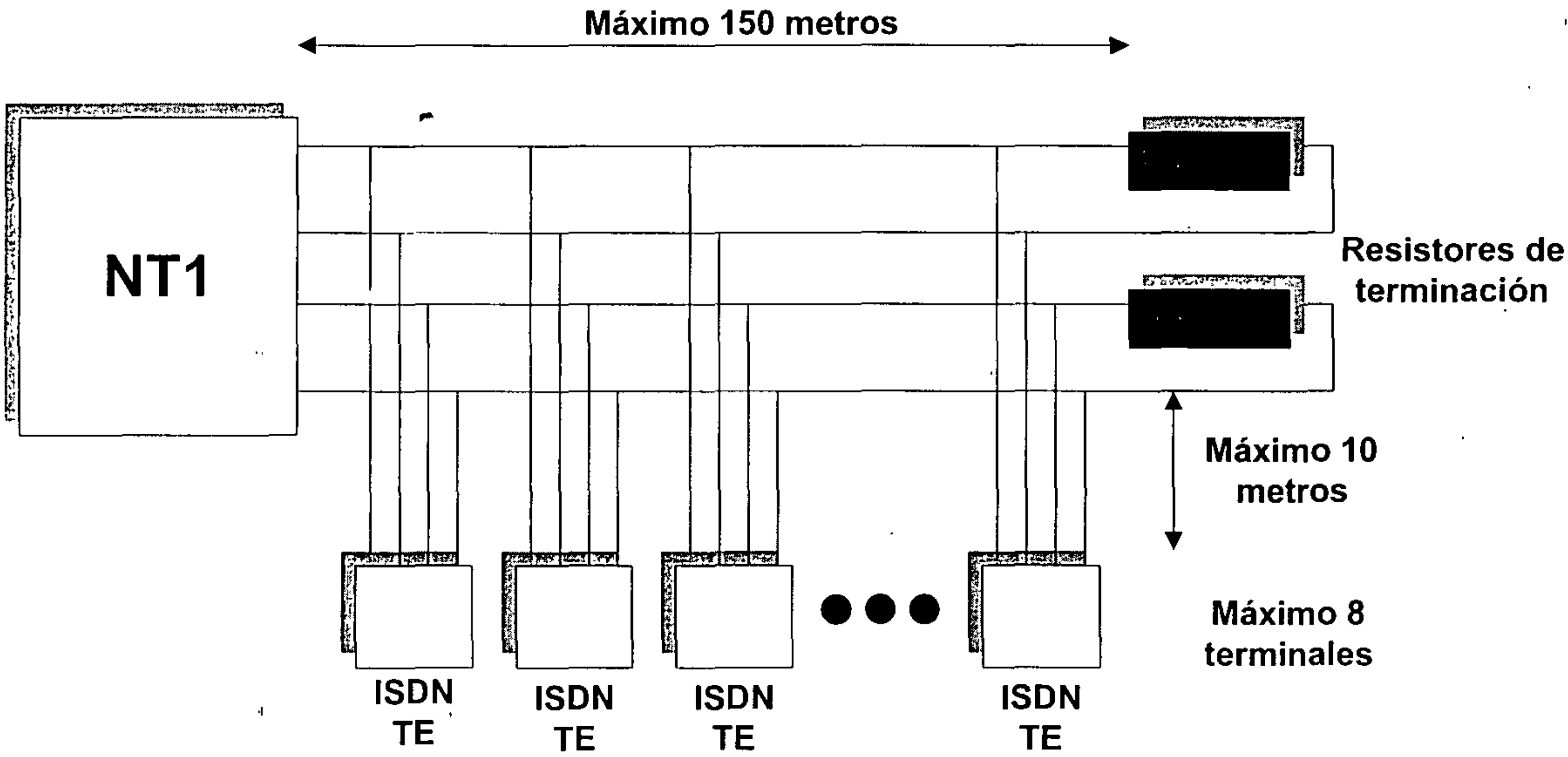
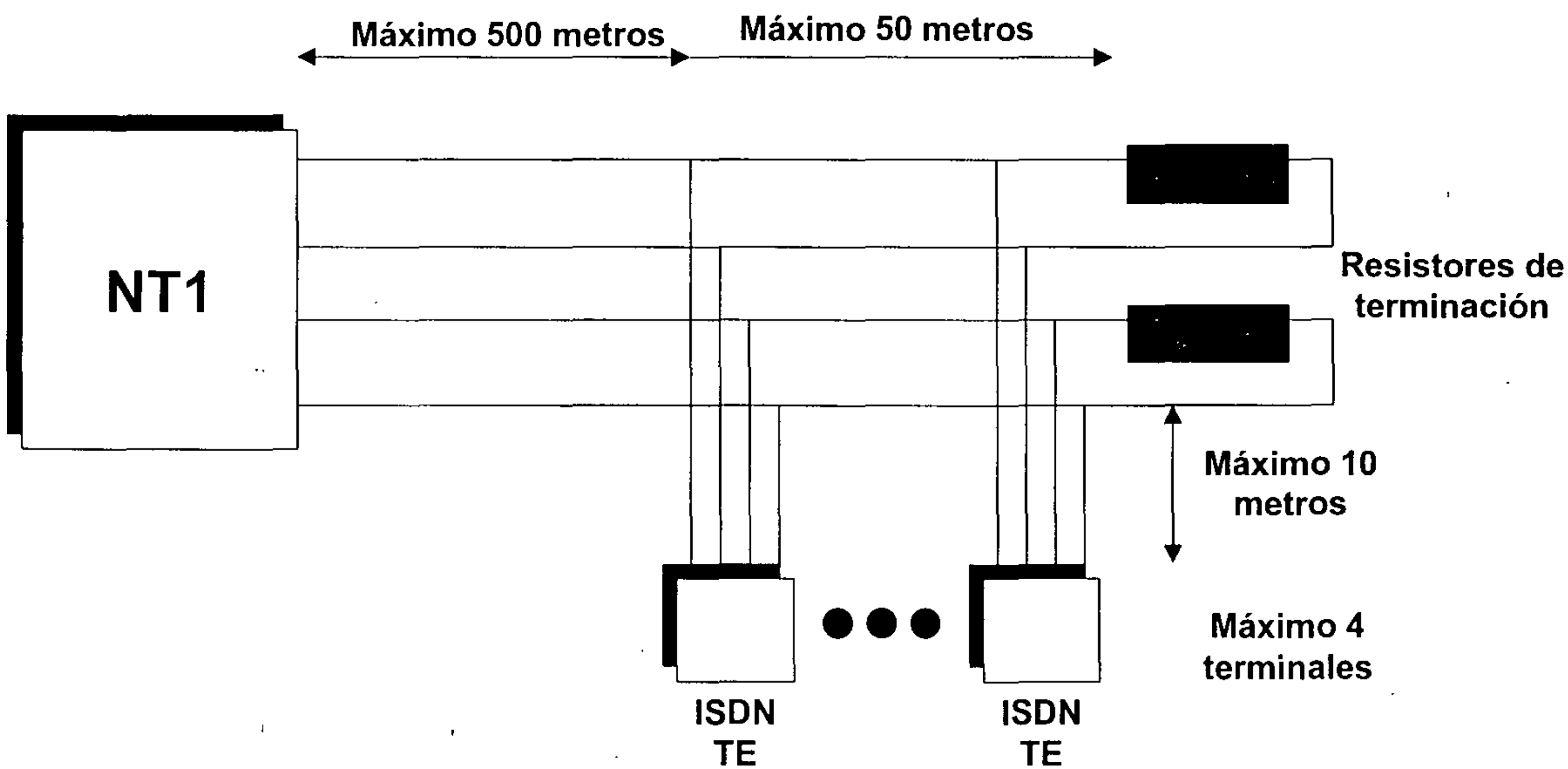
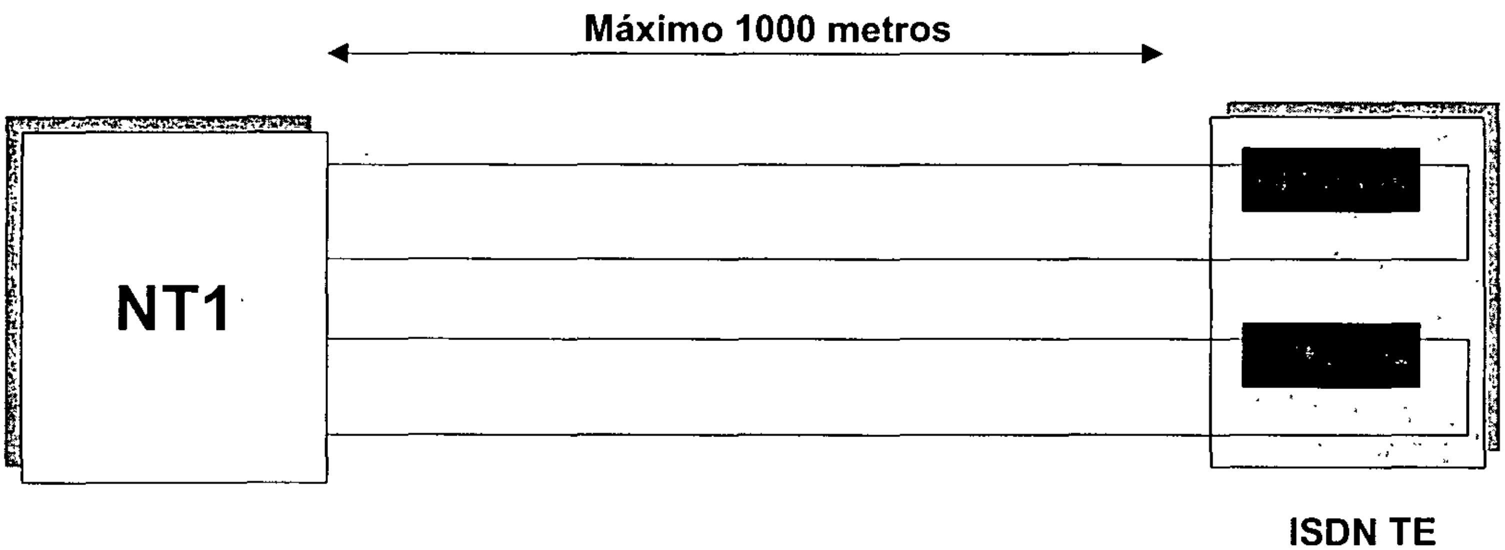
Configuración de Bus Pasivo Corto Punto a Multipunto



Configuración de Bus Pasivo Extendido Punto a Multipunto



Configuración Punto a Punto



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La Interfaz U

- La interfaz U es conocida como 2B+D, la cual contiene 2 canales B (Bearer) y un canal D (Delta) y ha sido diseñada para operar efectivamente utilizando la red de acceso telefónico existente.

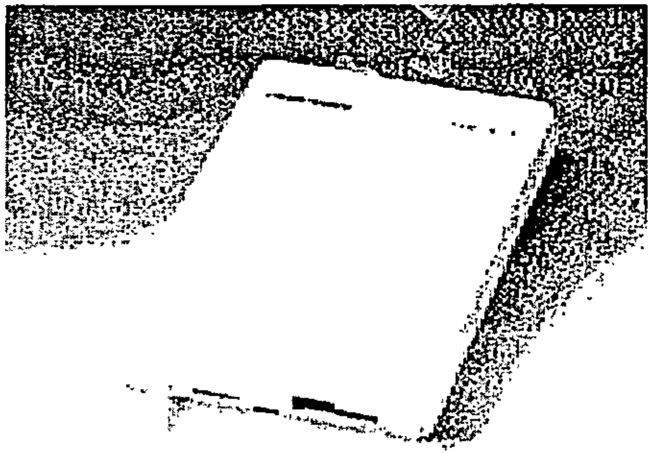
- Los canales B son utilizados para cualquier tipo de comunicación de circuito conmutado, tales como: voz, datos conmutados, video, etc.

- Los canales B son asignados para todo el tiempo que dura la llamada, y cada uno de ellos tiene un ancho de banda de 64 Kbps.

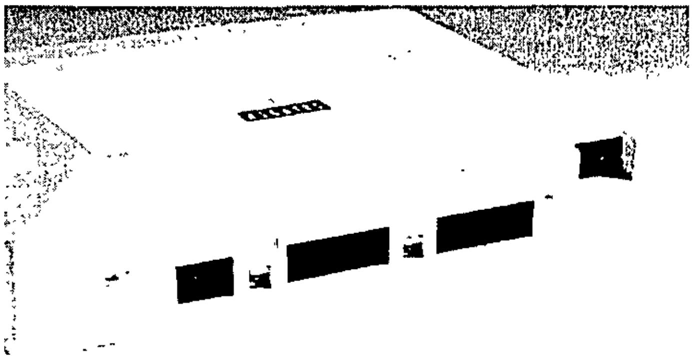
- El canal D es utilizado principalmente para señalización, aunque ya existen aplicaciones como el AO/DI en la cual trsnaporta información de usuario. El canal D tiene un ancho de banda de 16 Kbps.

- El bus U tiene una configuración de dos hilos y proporciona transmisión full duplex..

Ejemplos de Terminadores de Red



NT 1580/ID4
2 Interfaces So RDSI



NT-Twin
2 Interfaces Analógicas
1 Interfaz So RDSI

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Codificación de Línea en el BRA

- El código de línea en la interfaz BRA es 4B3T o 2B1Q.
- Con el 4B3T, 4 digitos binarios (bits) son convertidos en tres digitos ternarios (3 estados).
- Con el 2B1Q, 2 digitos binarios (bits) son convertidos en un digito cuaternario (4 estados).

Código de Línea 4B3T

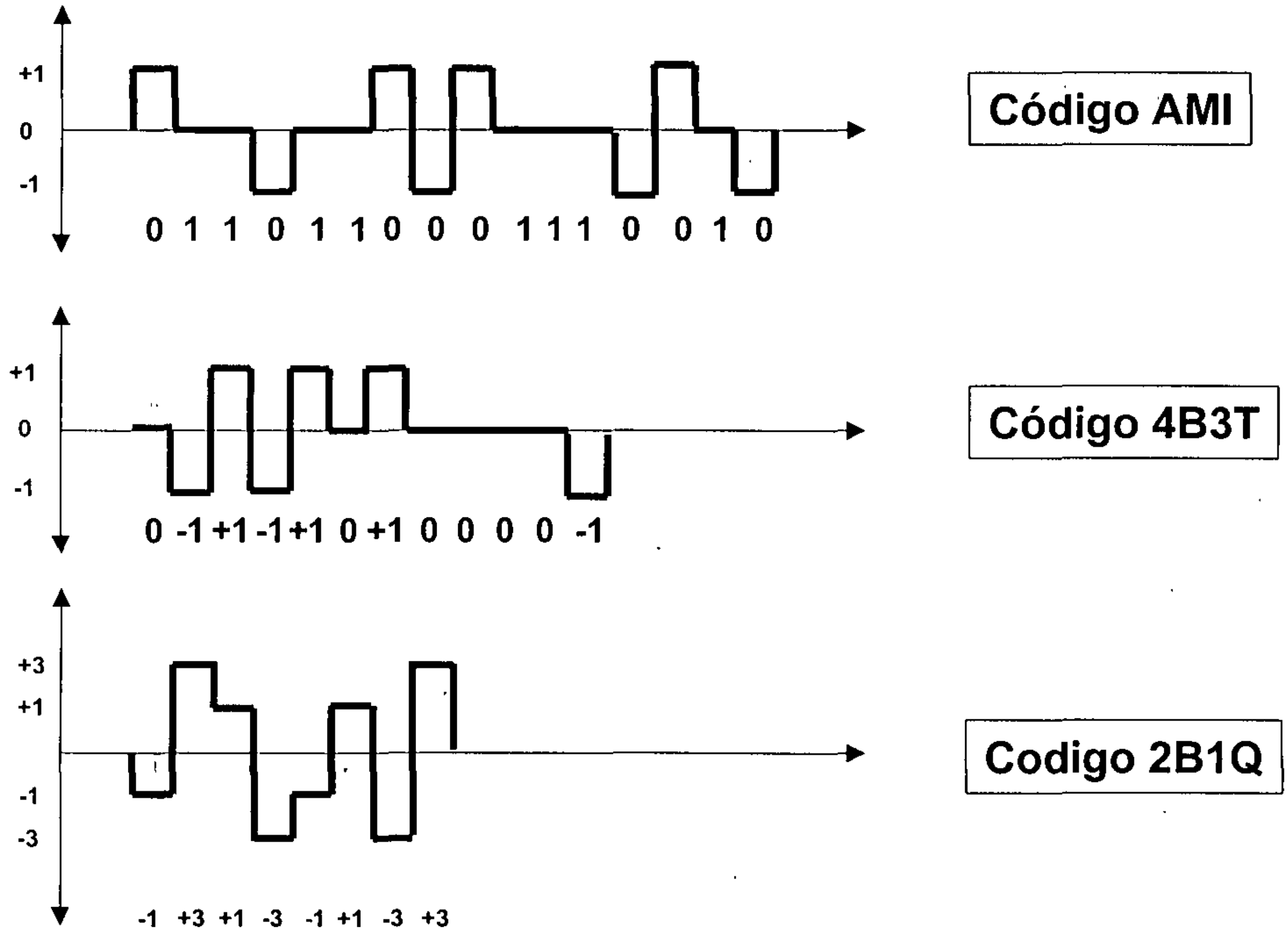
DIGITOS BINARIOS	DIGITO CUATERNARIO
0000	0 0 0
001	0 0 +1
0010	0 0 -1
0011	0 +1 0
0100	0 -1 0
0101	0 +1 -1
0110	0 -1 +1
0111	+1 0 0
•	•
•	•
1011	-1 0 +1

ALCATEL

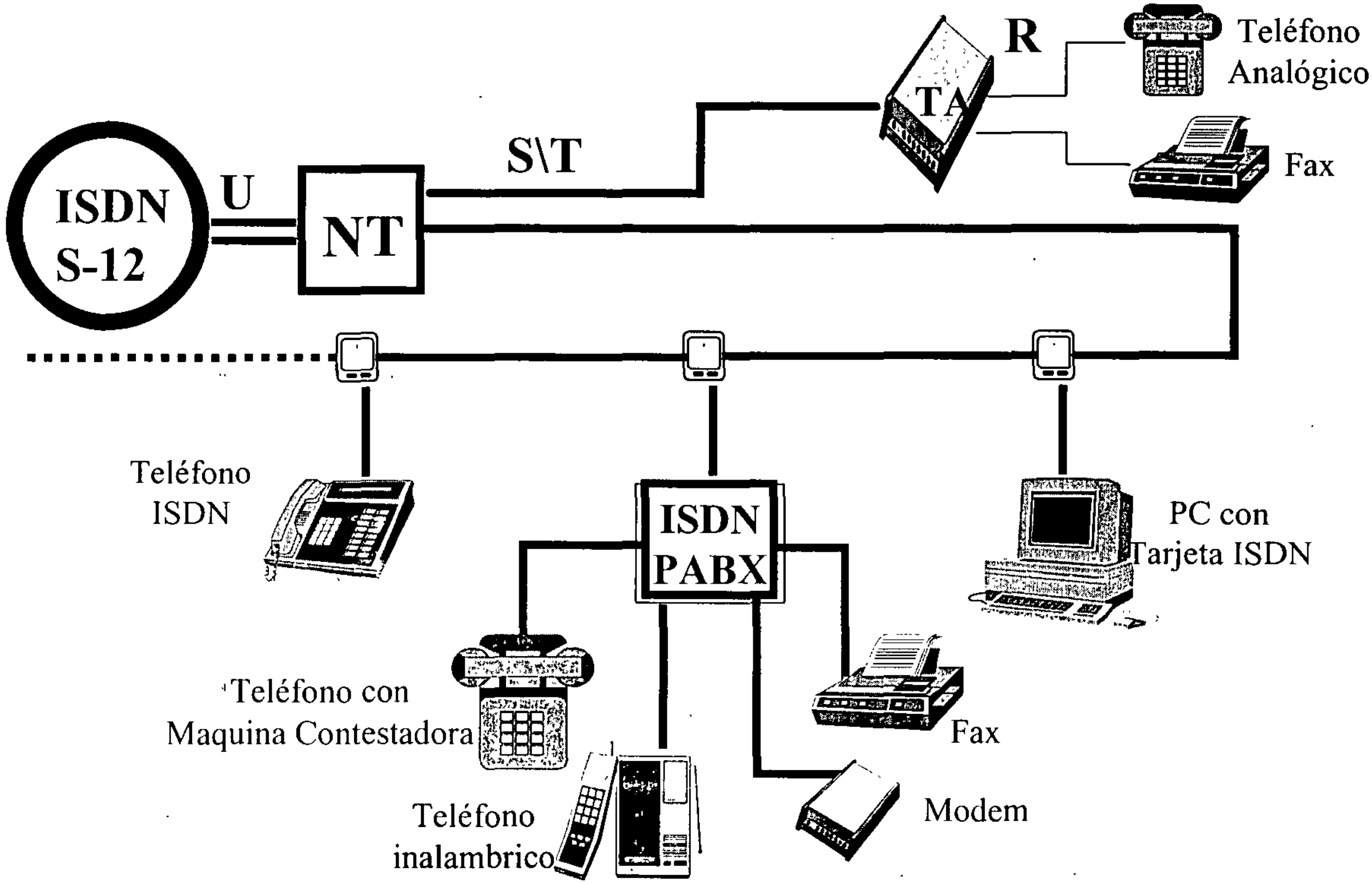
Código de Línea 2B1Q

DIGITOS BINARIOS	DIGITO CUATERNARIO	NIVEL DE VOLTAJE
00	- 3	- 2.5 V
01	- 1	- 0.833 V
10	+ 3	2.5 V
11	+ 1	0.833 V

Ejemplos de Codificación de Línea



Sistema del Bus de Interconexión



RDSI: Ejemplos de Aplicaciones

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Videoconferencia de Salón

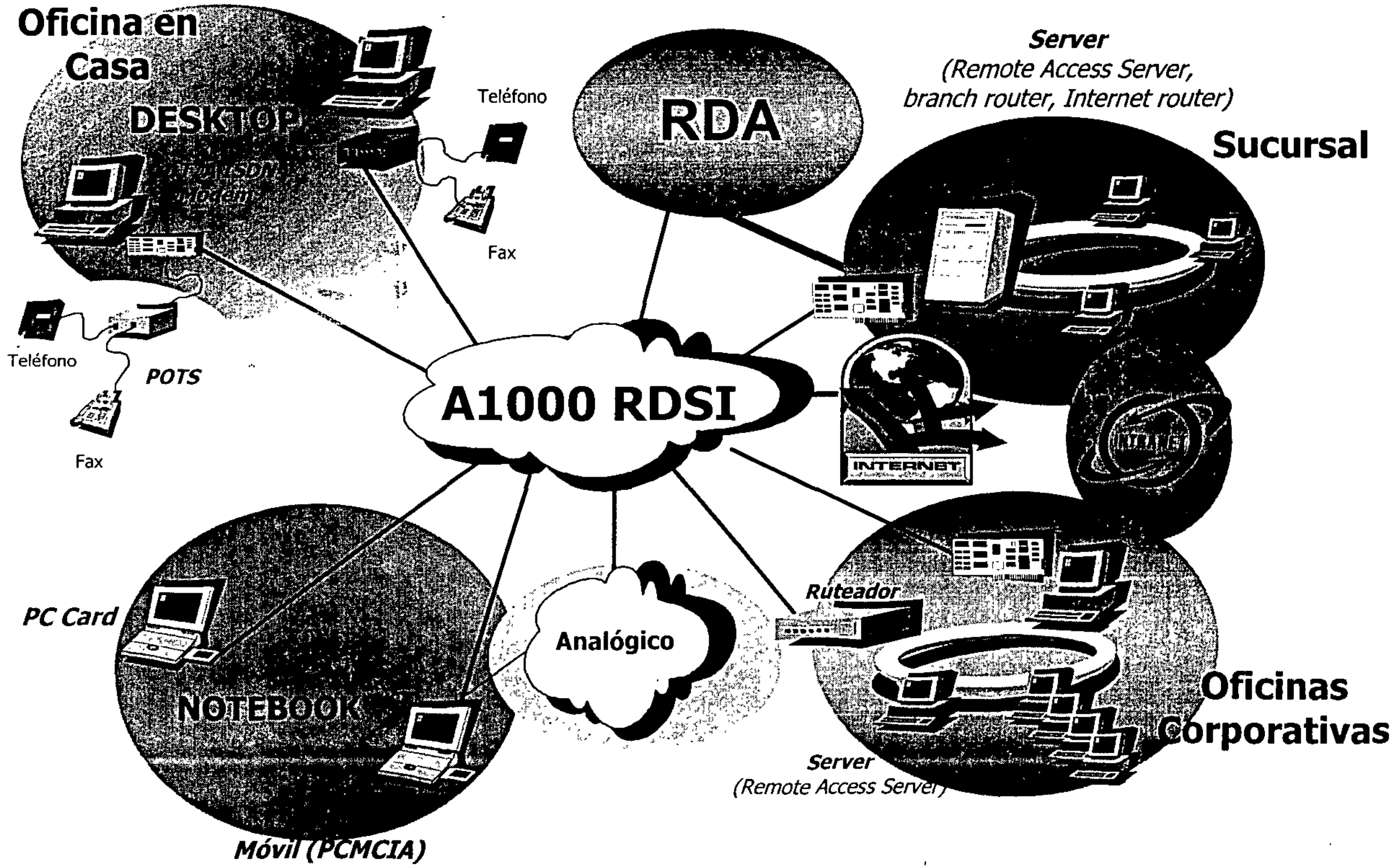
VENTAJAS: AHORRO EN TIEMPO
AHORRO EN GASTOS DE VIAJE
MENORES COSTOS DE OPERACION
ACCESIBLE A TODO EL PERSONAL
SIN RESTRICCION DE TIEMPO
INFORMACION PRECISA Y EN TIEMPO REAL

Videoconferencia de Escritorio

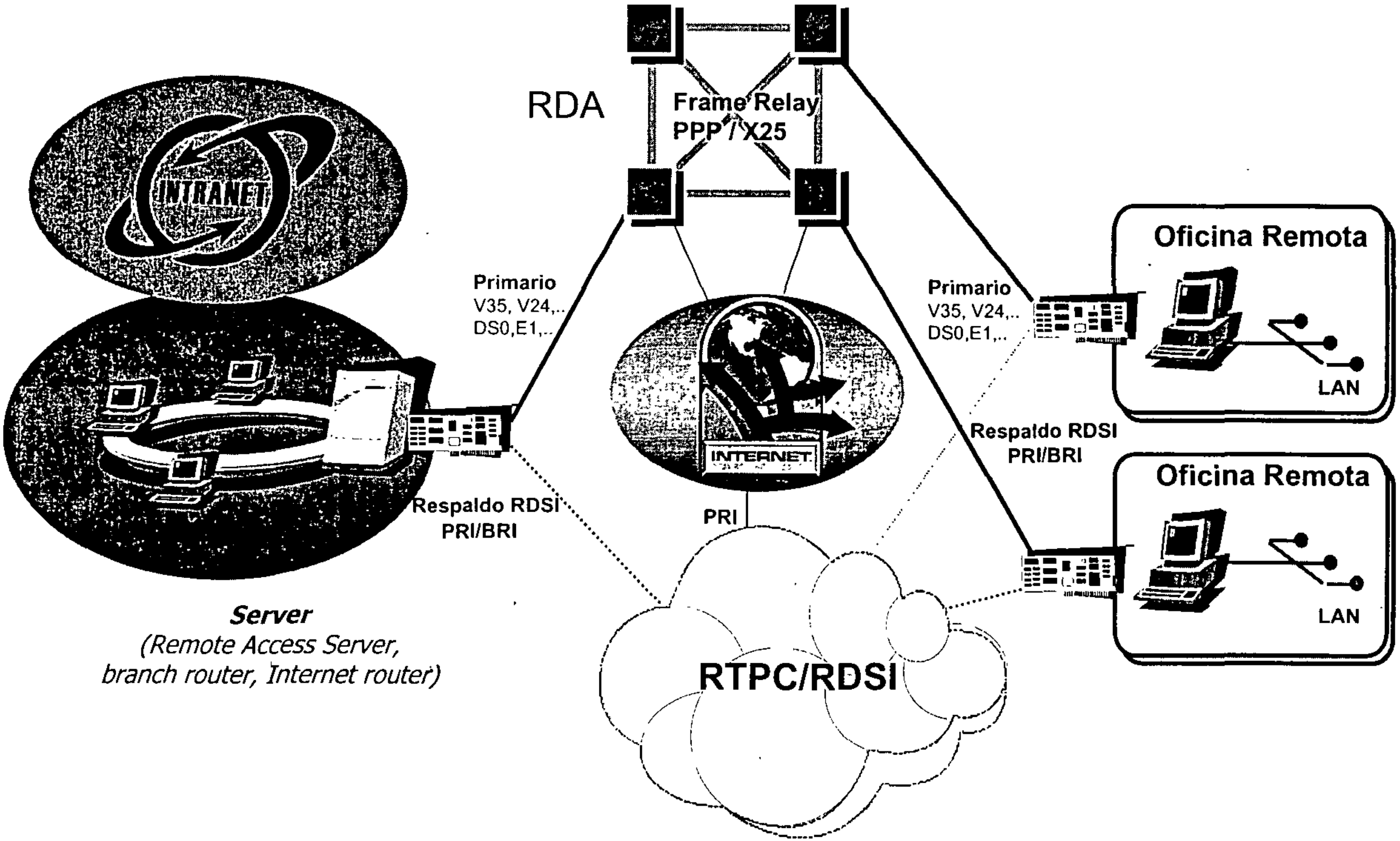
VENTAJAS: AHORRO EN GASTOS DE VIAJE
CONSULTAS INMEDIATAS A CUALQUIER PUNTO
ACCESO A BASES DE DATOS
COMPARTICION Y TRANSFERENCIA DE ARCHIVOS (APLICACIONES)
MENORES GASTOS DE OPERACION
SIN RESTRICCION DE HORARIO

ALCATEL

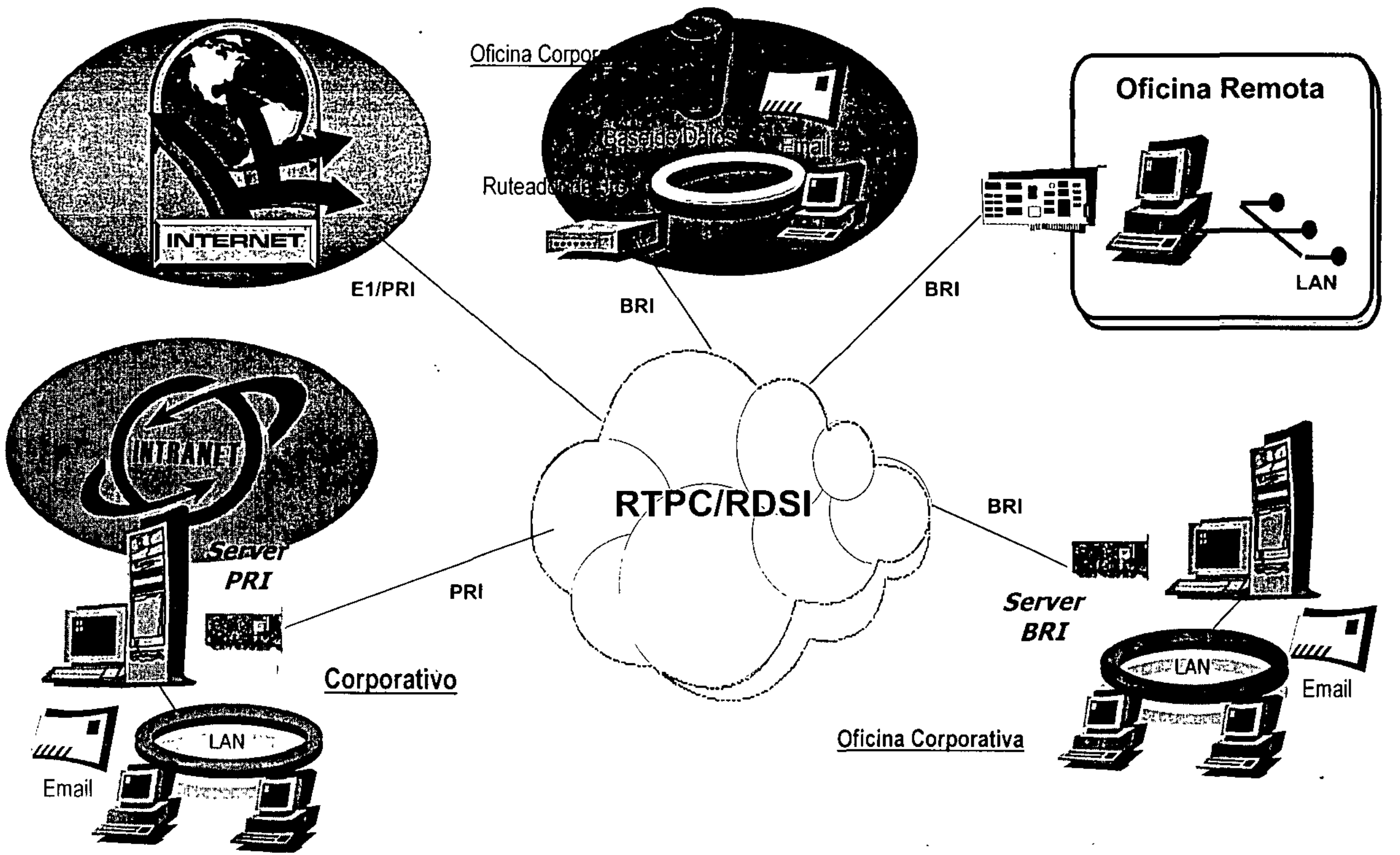
Configuración de Redes de Datos



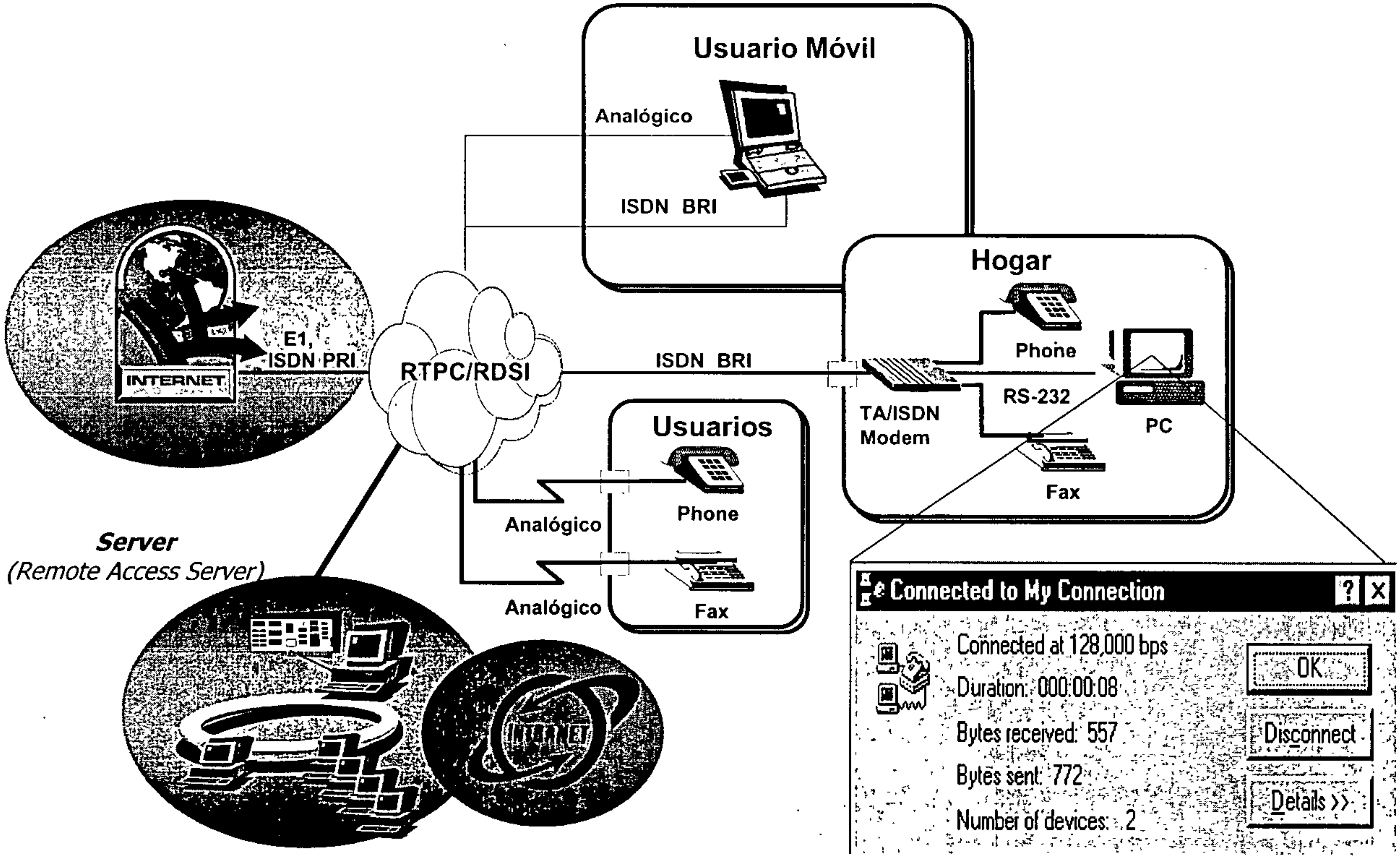
Respaldo a Líneas Privadas



Interconexión de LAN a LAN



Acceso Remoto Fijo y Móvil



FACULTAD DE INGENIERIA U.N.A.M.
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CURSOS ABIERTOS

VIII CURSO INTERNACIONAL
EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES:
"ACTUALIDAD Y PERSPECTIVA"

TEMA

<u>ADLS</u>
<u>LA SOLUCIÓN PARA ACCESO DE ALTA VELOCIDAD</u>

EXPOSITOR: ING. RAÚL DELGADO RIVERA
PALACIO DE MINERÍA
JUNIO DE 1999

Palacio de Minería Calle de Tacuba 5 Primer piso Deleg. Cuauhtemoc 06000 México, D.F. tel.: 521-40-20 Apdo. Postal M-2285

ADSL

La Solución para

Acceso de Alta Velocidad

Servicios On-Line

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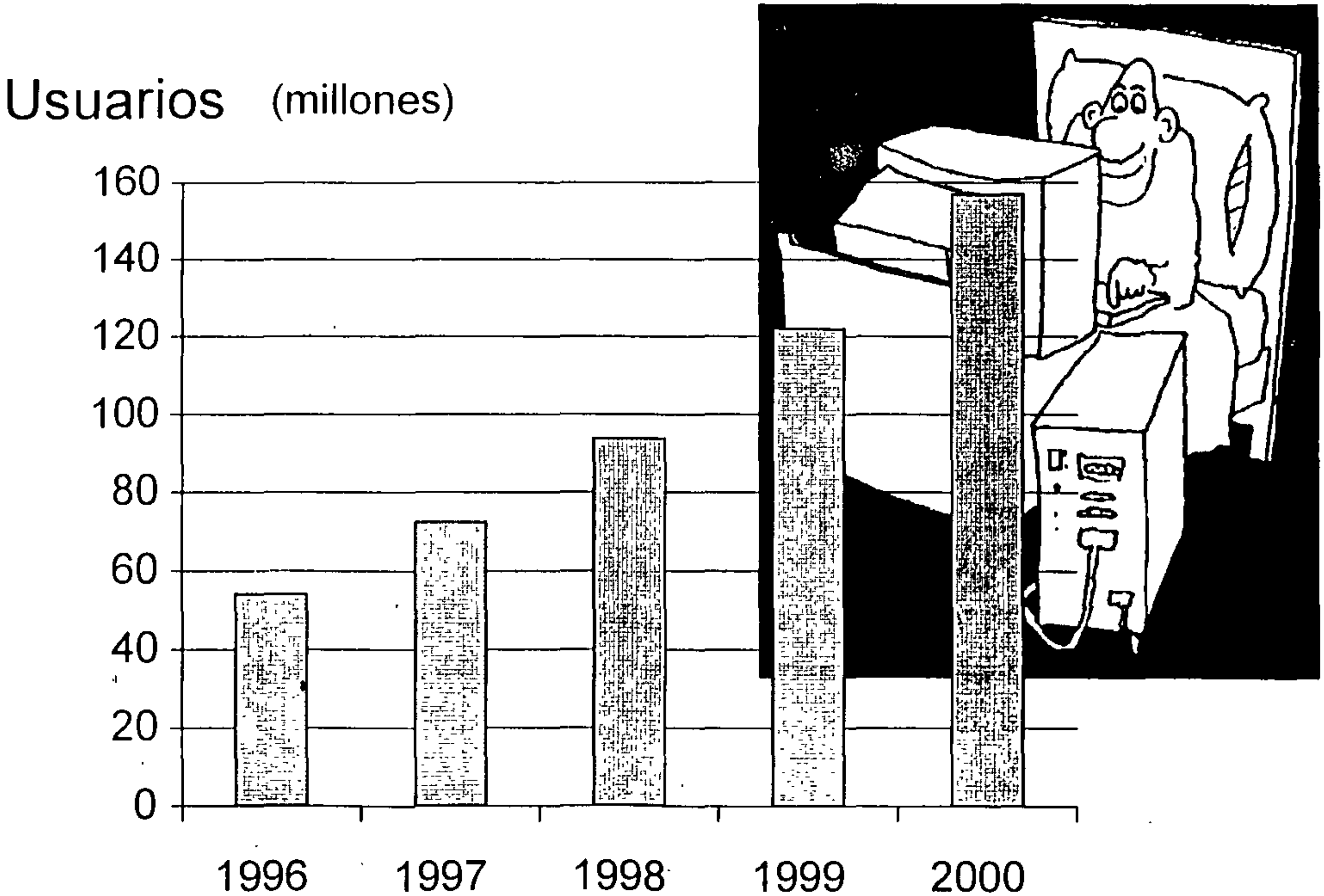
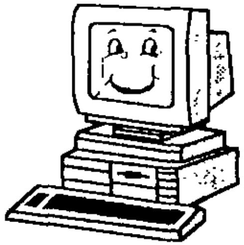
Servicios "On-Line" (OLS)
El Mercado

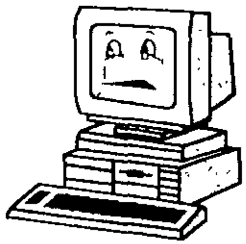


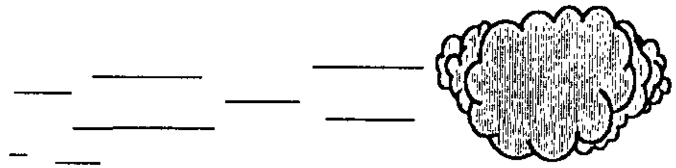
Servicios "On-Line"
Los Factores Clave de Éxito

▼ Uso amigable



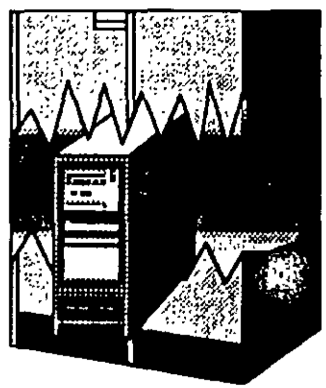
▼ Acceso rápido



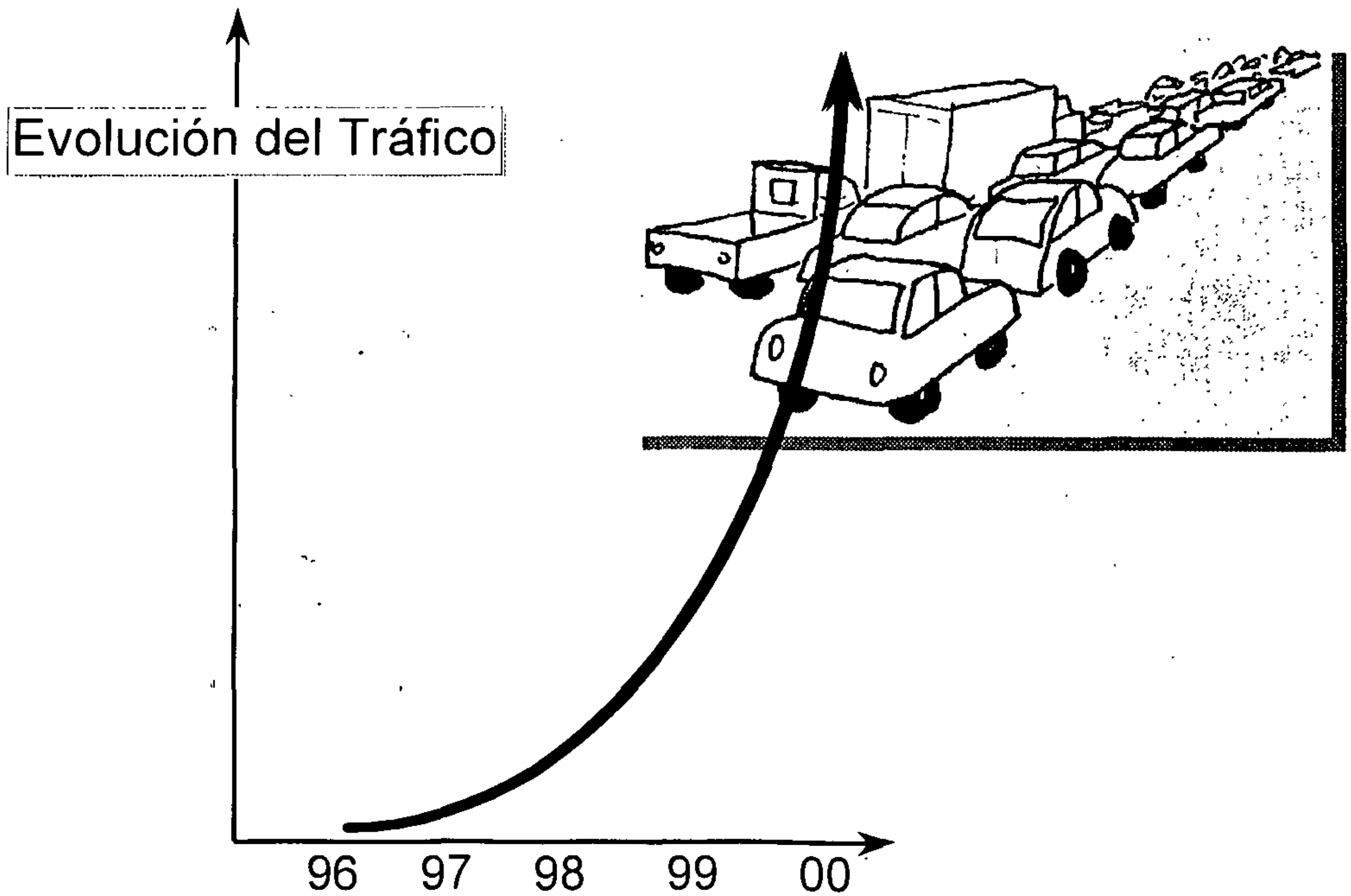


▼ Información útil

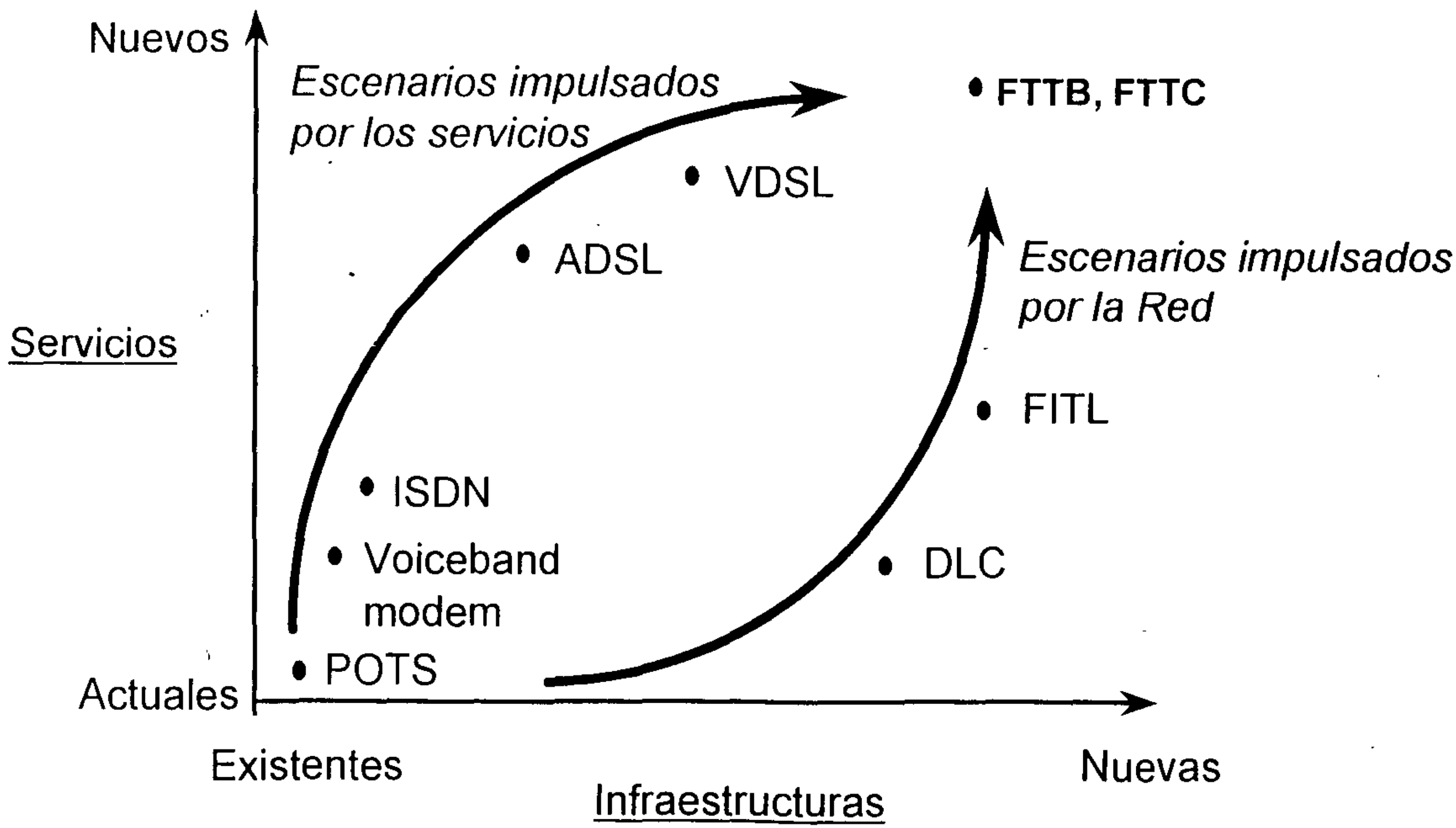
▼ Integración en entornos de TI



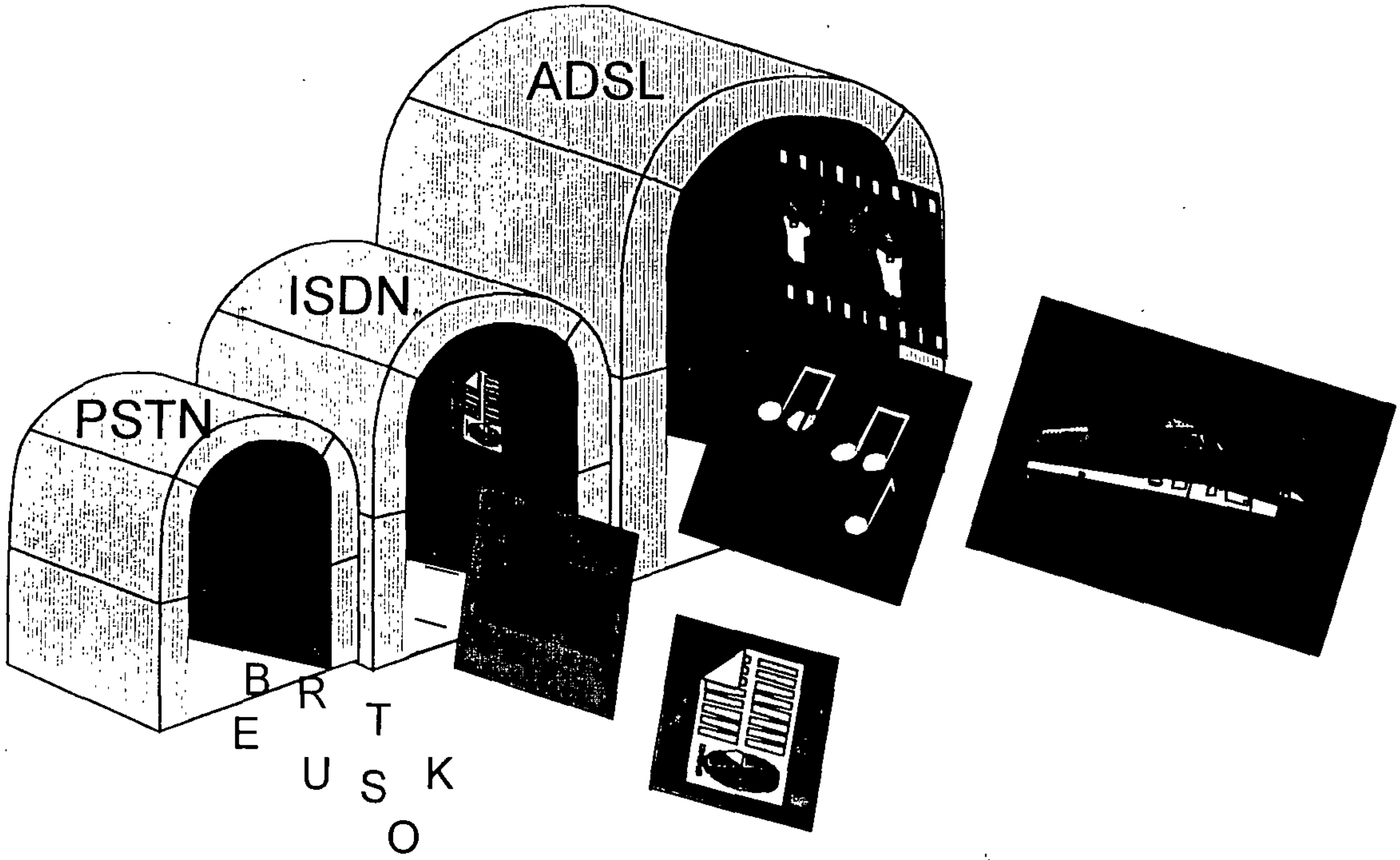
Servicios "On-Line"
Requerimientos de Red



Escenarios de Evolución : Red Telefónica



Banda-ancha en Acceso



ADSL

¿Por qué es importante ADSL?

- ▼ La planta de cobre es el mayor activo de los operadores de Telecomunicaciones
- ▼ La velocidad en el desarrollo del servicio es clave
- ▼ Permite competir con los operadores de cable en los servicios "On-Line" de alta velocidad
- ▼ Los modems ADSL existen HOY
- ▼ No require instalaciones de cable adicionales
  - Bajo costo de instalación
  - Rápida conexión de nuevos usuarios

Características clave de ADSL

▼ Permite obtener ingresos adicionales a los Operadores de Telecomunicación

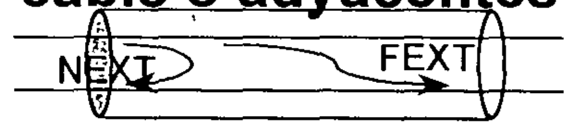
- Ofreciendo acceso de altas prestaciones a servios "On-Line" (Internet, VoD) **sobre el mismo par de cobre**
- Con un coste estríctamente proporcional al número de abonados conectados
  - La baja inversión inicial limita los riesgos financieros
  - Adecuado para un despligue rápido de servicios
- Sin impacto en el servicio telefónico suministrado por la misma linea
- Con capacidad suficiente para servicios de Video
- Compitiendo con los operadores de cable en los servicios On-Line de alta velocidad.

▼ Permite descargar el tráfico Internet de la red de Telefonía

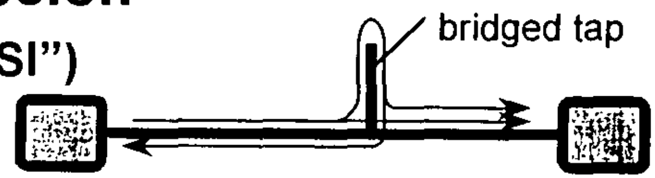
- Desvia el tráfico Internet (On-Line en general) antes de pasar por las centrales

Inconvenientes del Bucle

- ▼ **Atenuación y distorsion dependientes de la frecuencia**
  - ➡ Interferencia entre símbolos ("ISI")
- ▼ **Acoplamiento entre lineas del mismo cable o adyacentes**
  - ➡ Diafonía



- ▼ **Conexiones y transiciones de sección**
  - ➡ Reflexiones : Distorsion de pulsos ("ISI")



- ▼ **Ruido Impulsivo : ráfagas de corta duracción (10 μs - 1 ms)**
- ▼ **Desadaptación de Impedancias entre la híbrida y la linea**
  - ● Reflexiones (eco)
- ▼ **Interferencias de RF : difusion AM & radio-aficionados**
  - ● Transceptor con gran capacidad de adaptación

Solución de ADSL

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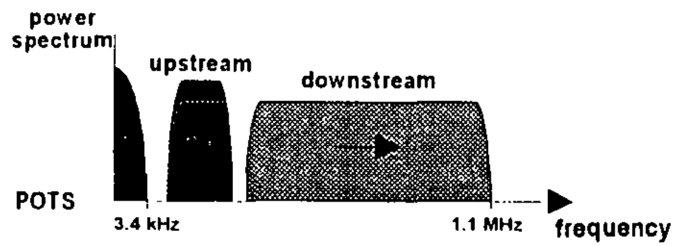
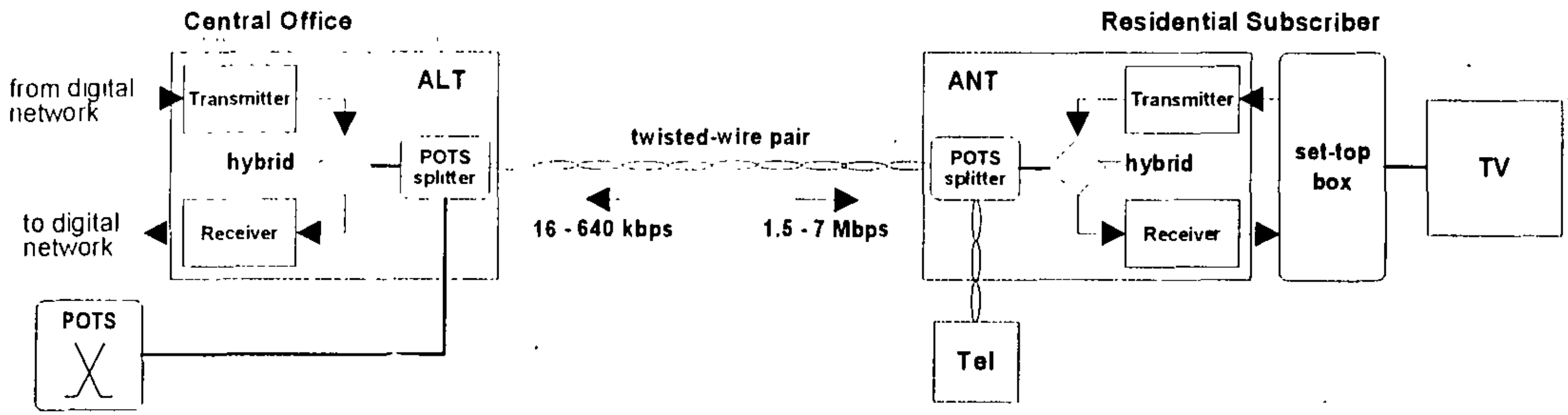
Configuración de referencia ADSL (parámetros ANSI)

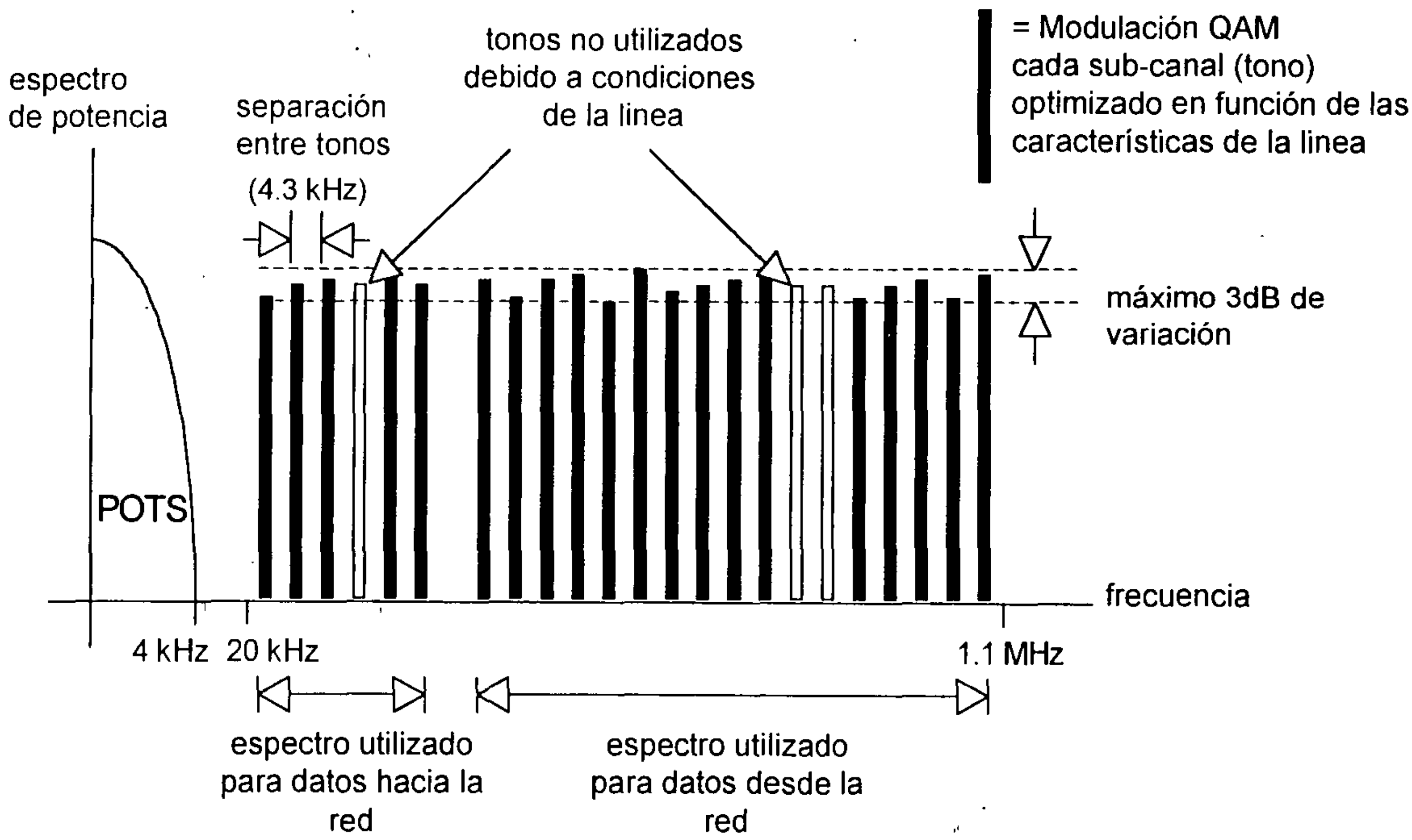




ALT : ADSL Line Termination
ANT : ADSL Network Termination

ALCATEL

Espectro DMT (FDM) - Parámetros ANSI



Modulación DMT

▼ General:

- Definido como estandar ANSI
- Existen varias compañias con desarrollos DMT (Alcatel, Analog Devices, Motorola, Texas Instruments, Metalink)
  - Garantiza mayor competencia -> precios

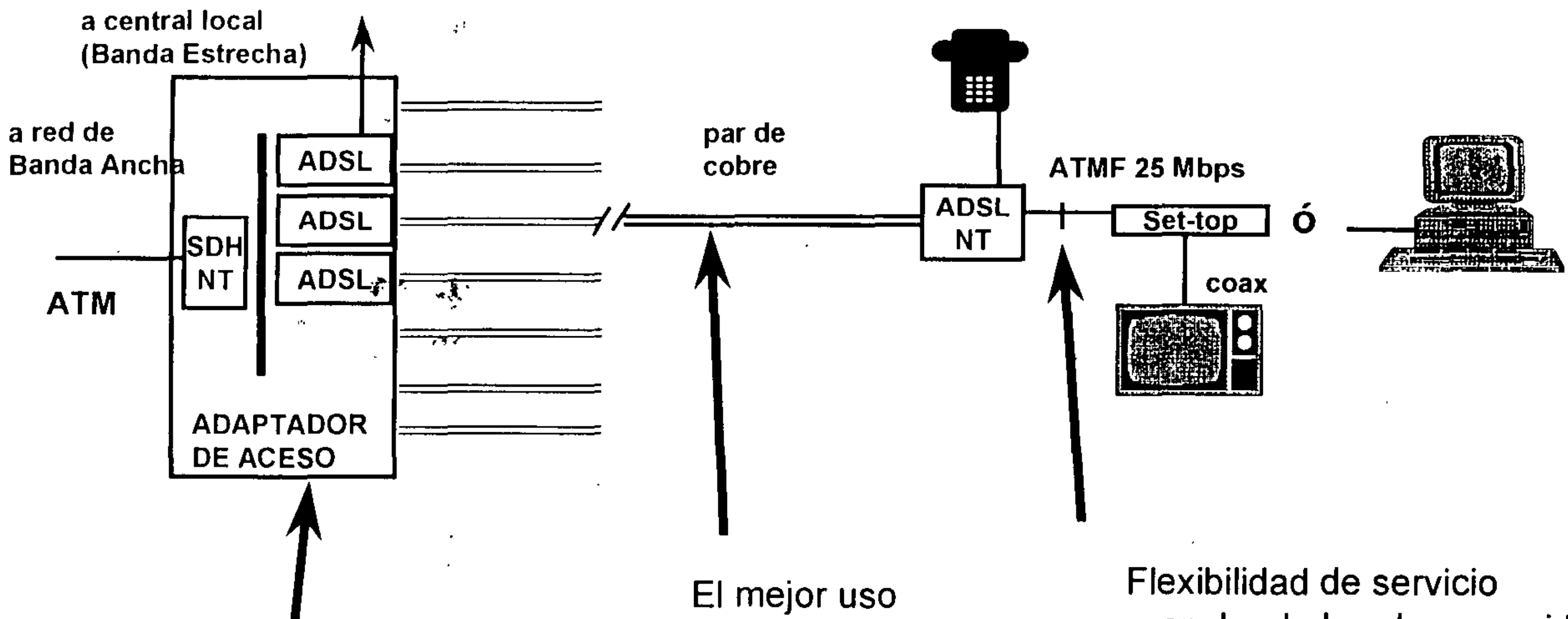
▼ Tecnológicas:

- Permite mayores velocidades para culaquier linea dada
  - Permite la mejor adaptación a las características de la linea, las interferencias de RF y especialmente al ruido impulsivo
- Permite mayores velocidades para bucles cortos
- Permite reconfiguración en operación
  - se obtienen mejores características EMI (Interferencial Electro-Magnéticas

ATM en el Acceso

- Lo más adecuado para adaptarse a los requerimientos de servicio actualmente en evolución:
  - Mezcla eficiente de ancho de banda requerido por diferentes servicios, (sin granularidad)
    - admite evolución en tecnologias de compresión
    - admite evolución en requerimientos de calidad de servicio
  - Transparencia de protocolos
  - Soporte eficiente de servicios con trafico a ráfagas
  - Soporte eficiente de anchos de banda asimétricos
- Lo más adecuado para la capacidad de las diferentes tecnologías de acceso:
  - Llenado eficiente de la capacidad de ancho de banda de las tecnologías de acceso
  - Esquema coherente de Multiplexacion / Demultiplexación y Concentración en los diferentes puntos de flexibilidad.

ADSL
Los beneficios de ATM extremo a extremo



Multiplexación directa ATM desde el "bitrate" ADSL a SDH 155Mbps

El mejor uso de la capacidad de la linea en cada caso

Flexibilidad de servicio
- ancho de banda requerido
- transparencia de protocolos
- asimetría del servicio
- tráfico a ráfagas

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ADSL con "Bitrate" Variable Combinando DMT y ATM

- DMT
  - Permite obtener el máximo "bitrate" de cualquier linea física,teniendo en cuenta la calidad del bucle y las interferencias
  - Determinación automática del máximo "bitrate" durante la inicialización del modem
  - Determinación automática de la forma del espectro de frecuencia para evitar las interferencias con otros servicios
- ATM
  - Aprovecha completamente el "bitrate" físico ofrecido por el modem DMT
  - Como una opción, el "bitrate" ofrecido se puede limitar a nivel ATM para ofrecer varias clases de servicio

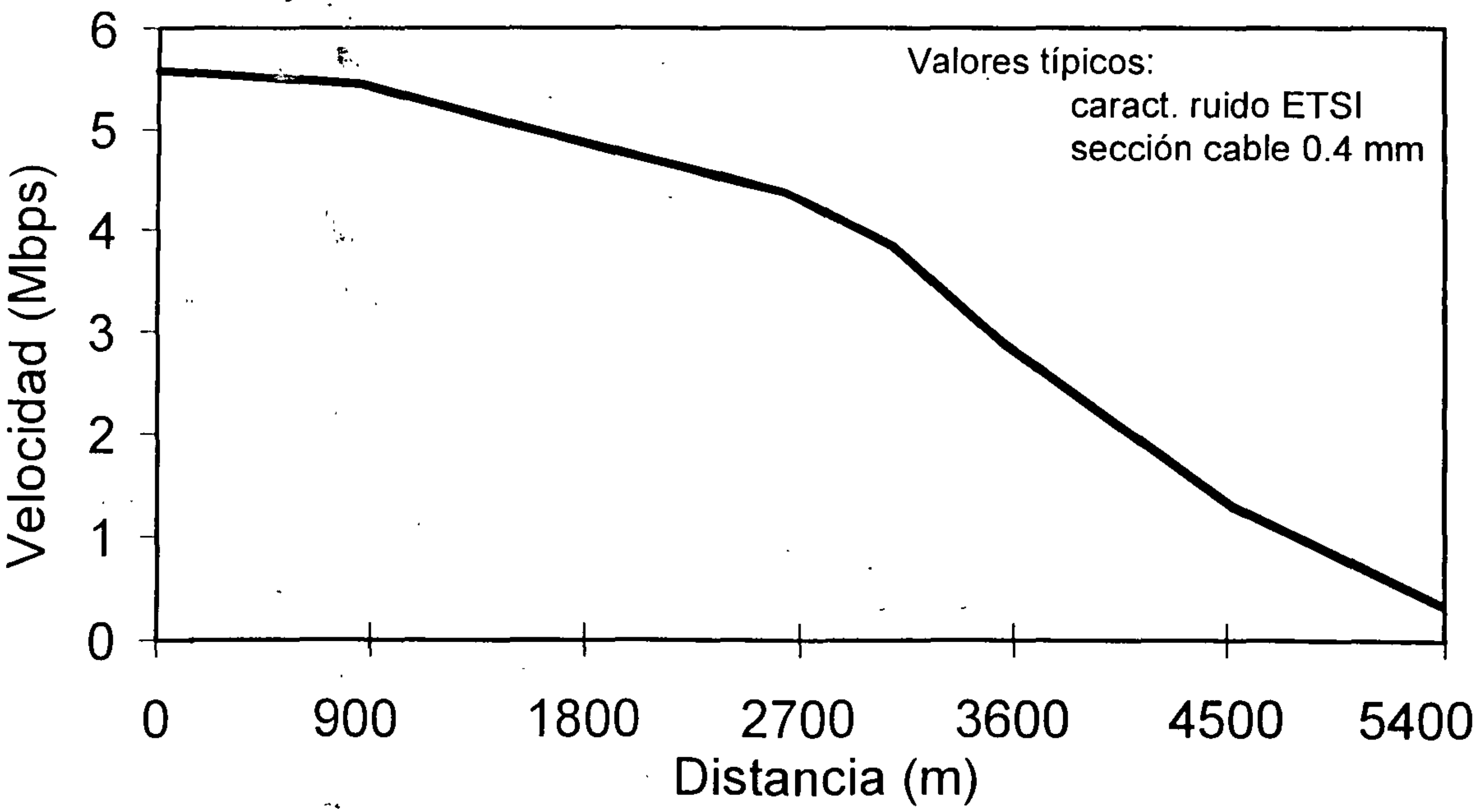
➜ La combinación DMT - ATM proporciona flexibilidad de "bitrate", aprovechando la máxima capacidad de transferencia de la linea

➜ Permite ofrecer una parte del "bitrate" como servicio básico (dependiendo de la estrategia comercial del operador, ej. 1Mbps), ofreciendo el resto en base a disponibilidad

➜ Permite explotar lineas con capacidad inferior a 2Mbps (lineas largas, ruidosas)

ALCATEL

"bitrate" en ADSL de Alcatel



FACULTAD DE INGENIERIA U.N.A.M.
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CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

<u>APLICACIONES ADLS</u>

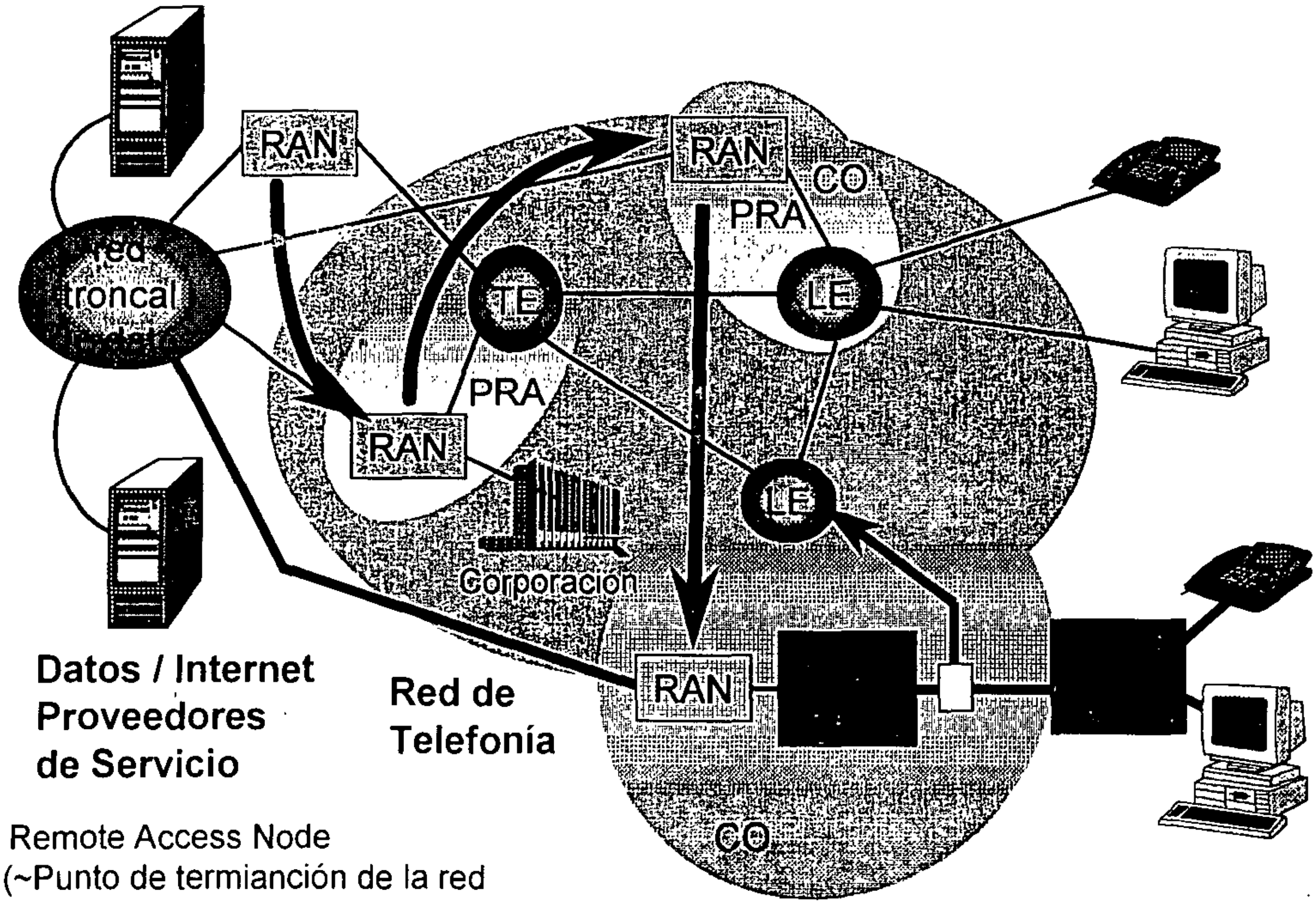
EXPOSITOR: ING. RAÚL DELGADO RIVERA
PALACIO DE MINERÍA
JUNIO DE 1999

Palacio de Mineria   Calle de Tacuba 5   Primer piso   Deleg. Cuauhtémoc 06000   México, D F.   APDO Postal M-2285
Teléfonos: 512-8955   512-5121   521-7335   521-1987   Fax   510-0573   521-4020 AL 26

Aplicaciones ADSL

Evolución de la Arquitectura de Red para servicios "On-Line"

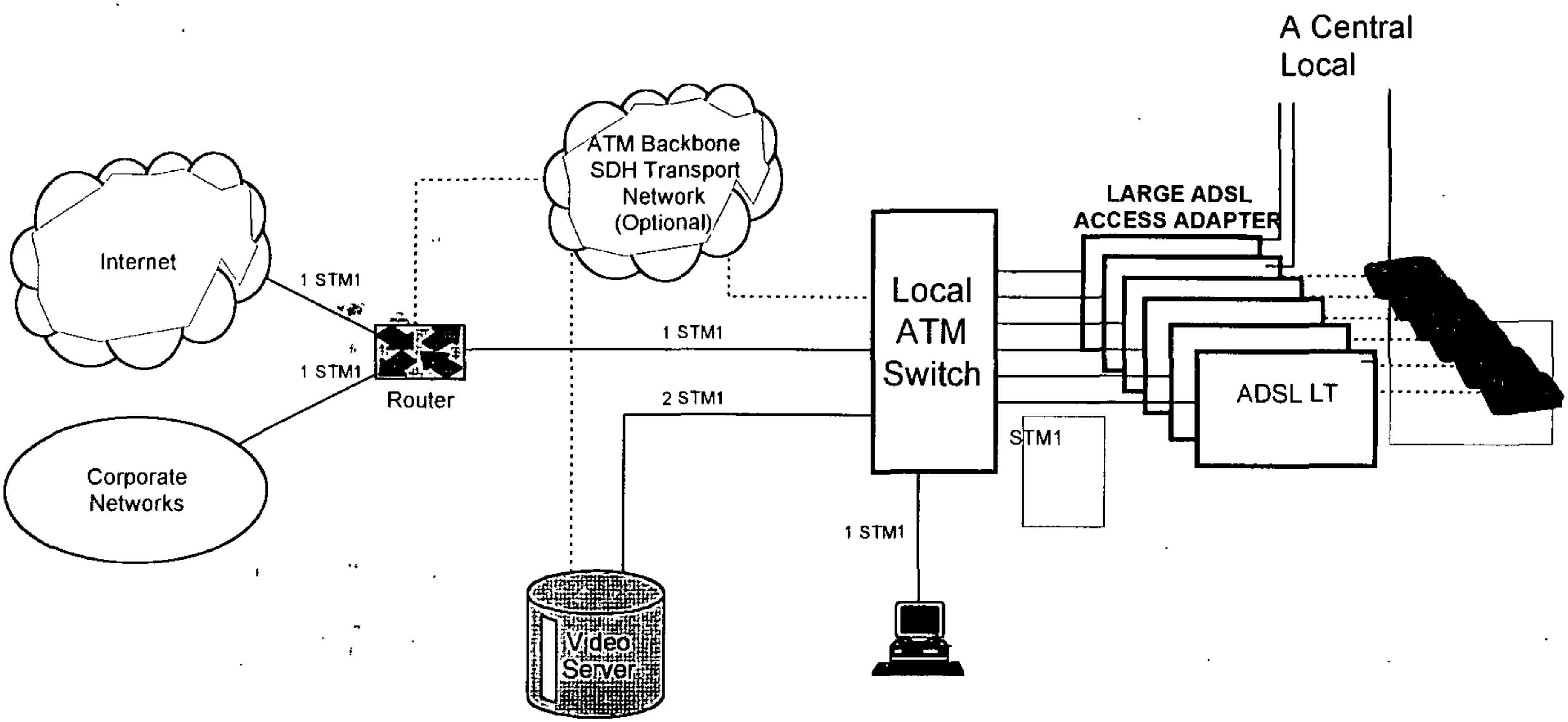
ALCATEL



RAN: Remote Access Node
(~Punto de termianción de la red de acceso a Internet)

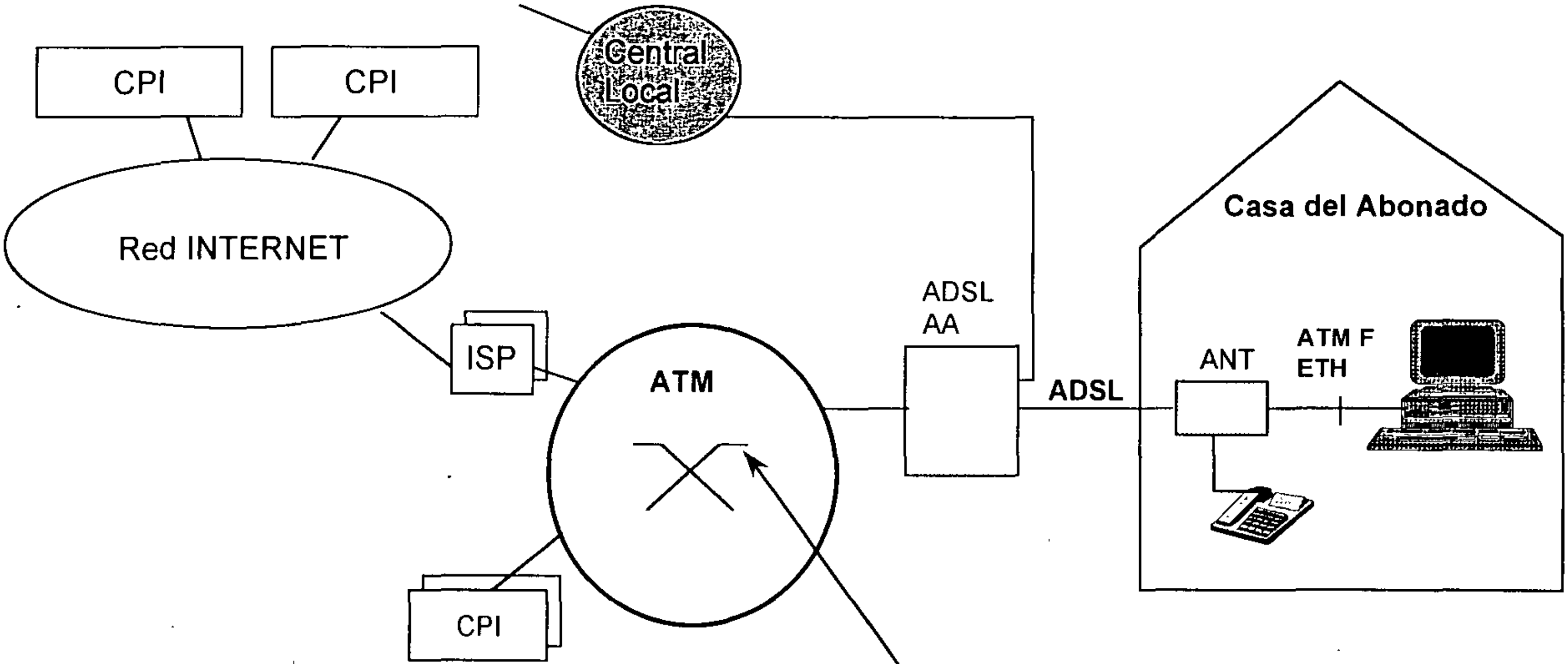
A L A T E L

Aplicaciones ADSL: Internet y Vídeo



A L A T E L

Aplicaciones ADSL: Acceso a Servicios On-Line Internet

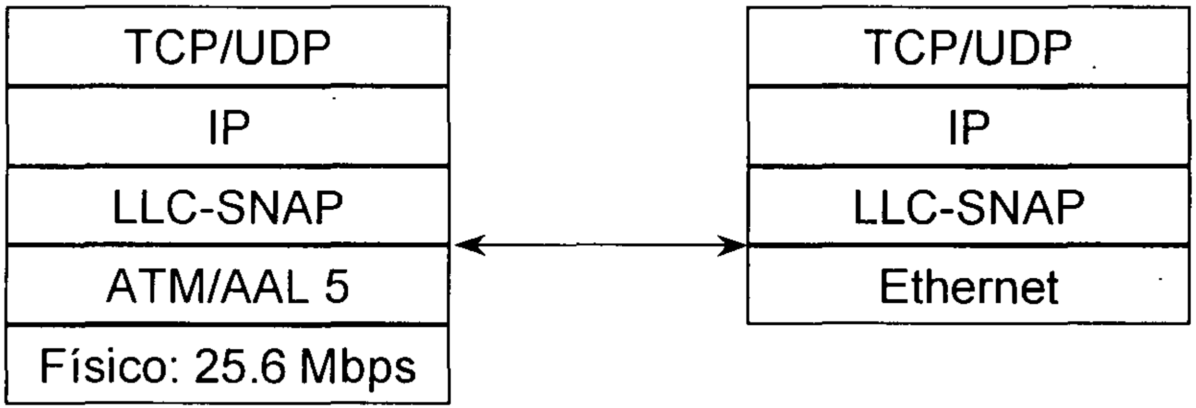


Conexiones virtuale ATM punto a punto
- Permanente
- Semi-permanente
- Conmutada

ADSL como Solución a Internet

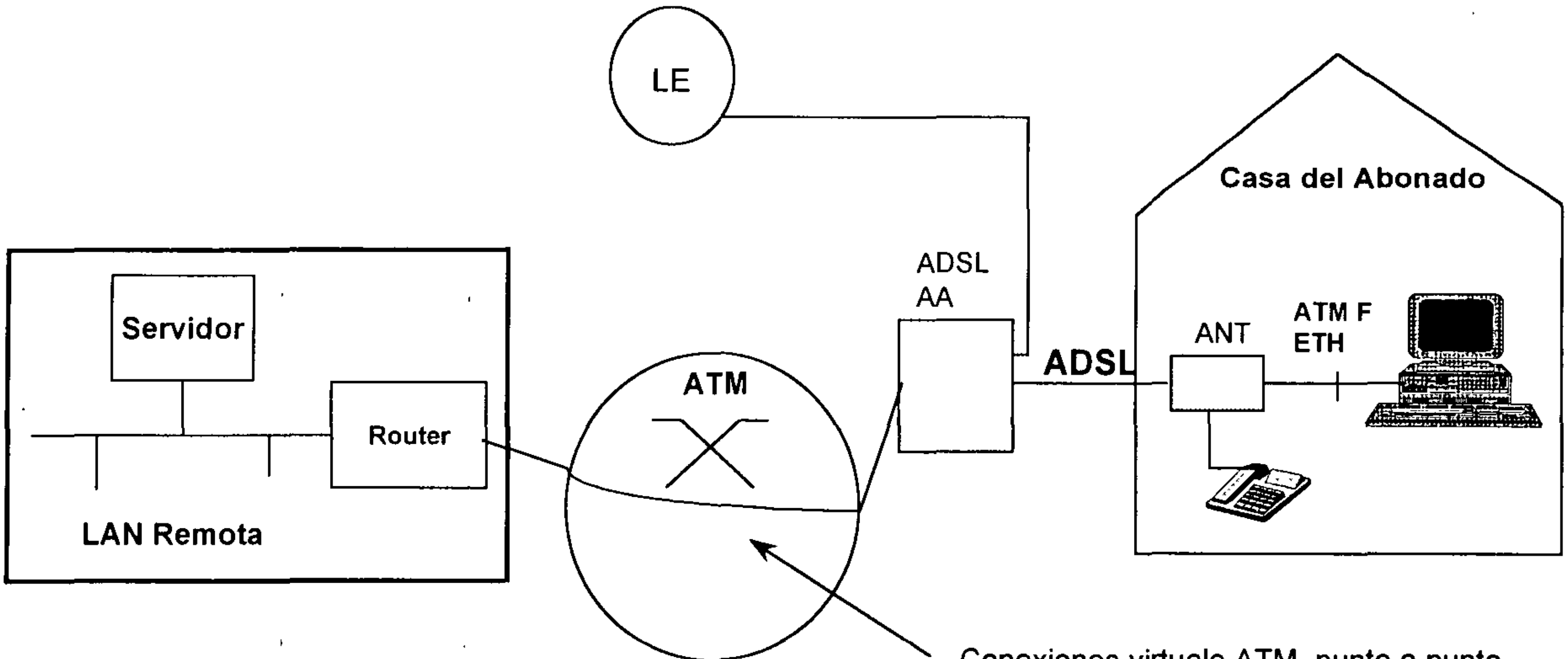
- Multiplica la velocidad de acceso por un factor de 200
  - Rompe la barrera para servicios Multi-Media en Internet
  - Promueve el uso de las telecomunicaciones para Trabajo en Casa
- Saca el tráfico Interent de la Red Telefónica Conmutada
  - Evita la saturación de las centrales con llamadas de larga duración
  - Permite deferentes esquemas de tarificación para servicios de voz y datos
- Proporciona un segundo servicio en la linea sobre el mismo par de cobre

IP sobre ATM



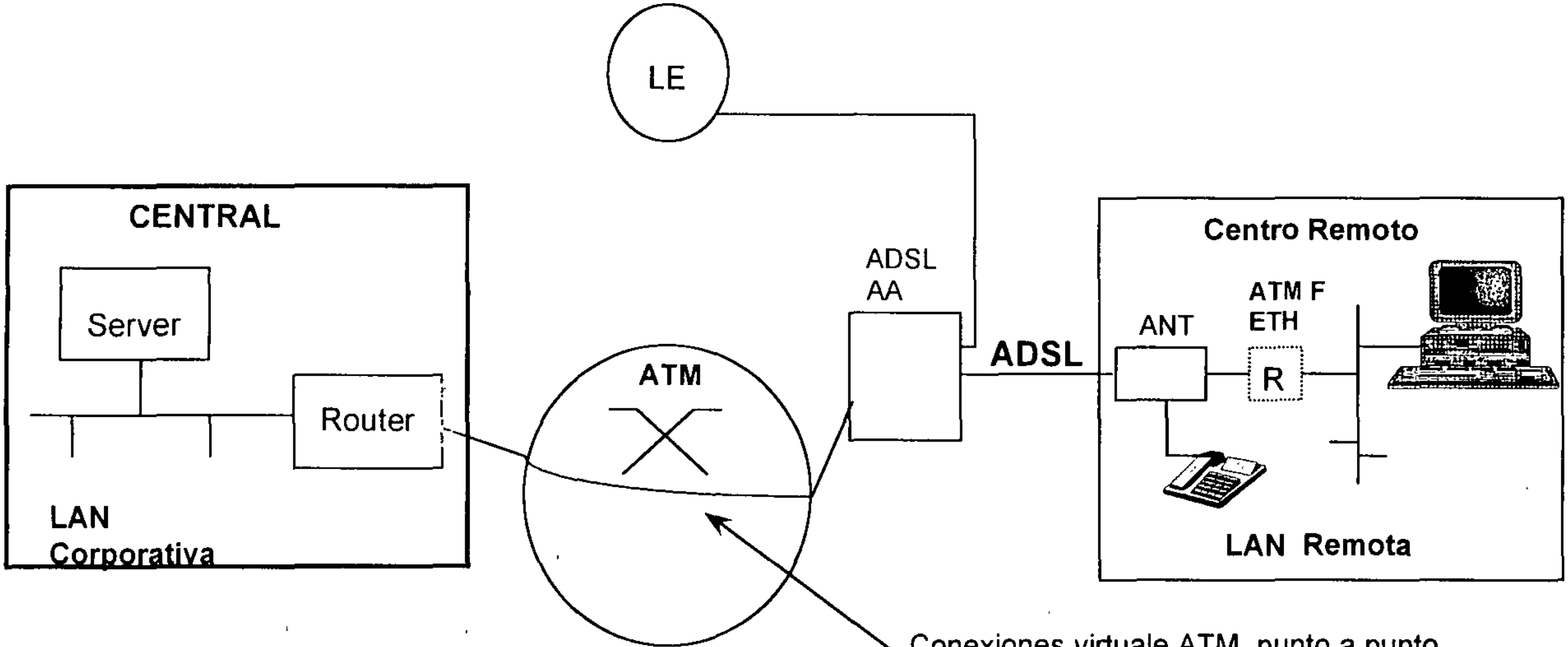
•ATM / AAL5 / Protocolo Punto-a-punto

Aplicaciones ADSL: Trabajo en Casa



Conexiones virtuale ATM punto a punto
•Permanente
•Semi-permanente
•Conmutada

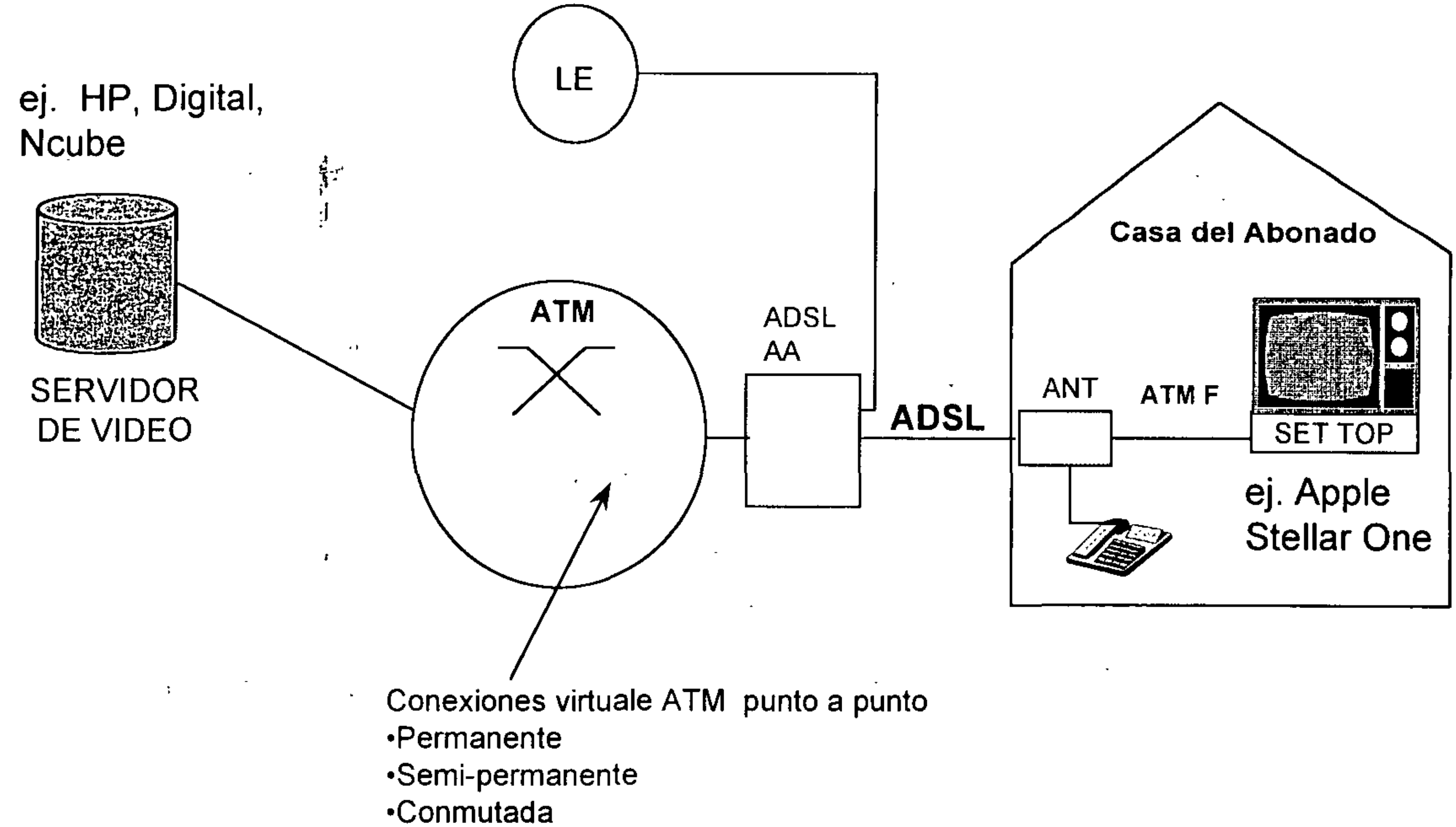
Aplicaciones ADSL: Corporación con Centros Distribuidos



Conexiones virtuale ATM punto a punto
- Permanente
- Semi-permanente
- Conmutada

ALCATEL

Aplicaciones ADSL: Multimedia y Video Bajo Demanda (VOD)



Conclusiones

- ▼ El crecimiento de los servicios Internet y Multi-media impone nuevos requisitos a las redes de telecomunicacion
  - Mayor capacidad de conmutación
  - Mayor ancho de banda en el acceso
- ▼ ADSL proporciona una solución a los operadores de red
  - capacidad 200 veces superior a los modems y 40 veces superior que el Acceso Básico RDSI
  - La tecnología está ya probada
  - Impacto limitado en la infraestructura existente
  - Rápida reacción a los "Cable-modems"
- ▼ La combinación de ATM y ADSL crea las redes para todos los servicios ("Full Service Network")
- ▼ Las soluciones basadas en ADSL/ATM extremo a extremo están disponibles HOY.

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CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

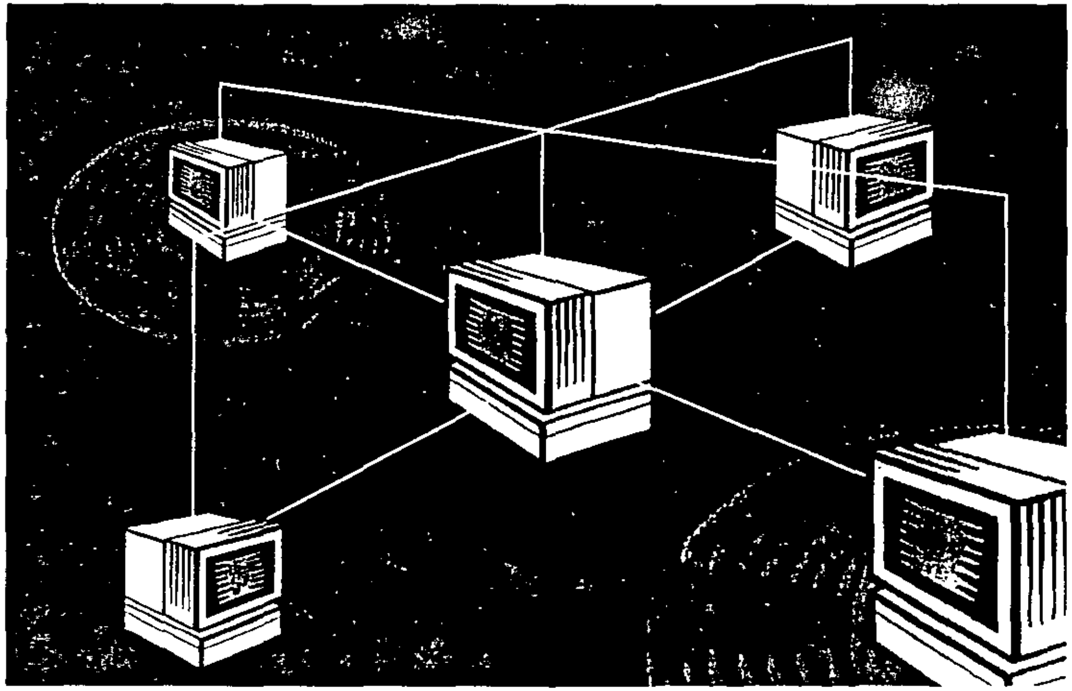
ANALYZING BROADBAND NETWORKS

EXPOSITOR: M. en C. ARTURO ELIE ALALUF OLIVARES
PALACIO DE MINERÍA
JUNIO DE 1999

ANALYZING BROADBAND NETWORKS

Second Edition

ISDN, Frame Relay, SMDS, & ATM



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Analyzing Broadband Networks

Second Edition

ISDN, Frame Relay, SMDS, & ATM

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10 ATM Architecture

ATM is a broadband ISDN technology that has been touted as the next revolution in LAN and WAN communication. Volumes have been written about ATM and its emerging role in corporate networks. References [10-1] through [10-6] provide background on cell-switching and ATM technology. References [10-7] through [10-9] are vendor and analyst reports that discuss trends in the ATM marketplace. References [10-10] through [10-12] provide information on carriers' plans to support ATM. Finally, References [10-13] through [10–19] discuss the integration of LAN and WAN technology. This chapter looks at ATM applications, standards, and interfaces. Chapter 11 discusses the ATM protocols in detail.

10.1 ATM Technology

ATM technology comes from ITU standards that address the worldwide telecommunications infrastructure. Two significant developments preceded ATM: in the early 1980s, the ITU defined ISDN, which is now called Narrowband ISDN (N-ISDN), and in the late 1980s, the ITU enhanced N-ISDN, defining broadband ISDN (B-ISDN). N-ISDN defined two access interfaces: a basic rate operating at 144 Kbps, and a primary rate operating at 1.544 Mbps. These interfaces were designed to carry digital voice, data, and control information. B-ISDN offered transmission rates of up to 622 Mbps. ATM is the technology that implements broadband ISDN.

10.1.1 The ATM Concept

Figure 10-1 illustrates how ATM transmits data. The ATM cellstream starts with signals from individual users or sources. Signals may include constant bit-rate service, such as a DS1 line; variable bit-rate service, such as compressed video; or bursty data, such as LAN traffic. ATM then segments the signals into 48-octet payloads and prefaces them with a 5-octet header containing addressing information. The resulting

53-octet packet is called a *cell*. At that point, ATM takes cells from various signal sources, mixes them with cells from other sources, and sends them to the ATM switch (see References [10-20 and 10-21]). The switch multiplexes the cells together. The cells then contend for vacant slots in the outgoing ATM cellstream.

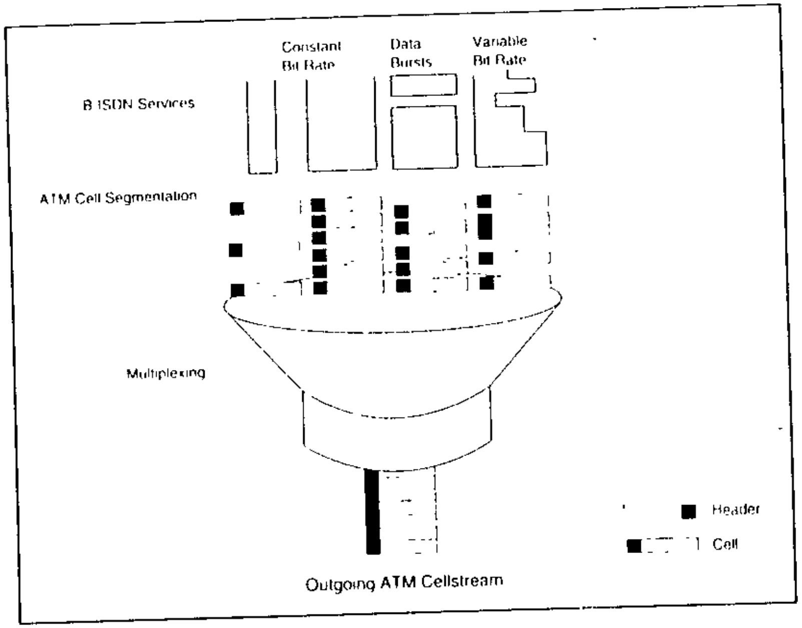


Figure 10-1. The ATM concept.
Courtesy of Hewlett-Packard Company

ATM is a connection-oriented service, meaning it requires an established connection before it can transmit data. There are two types of ATM connections: PVCs and SVCs. Two labels identify the endpoint connections: a VPI and a VCI.

The transmission delay for each cell depends on the traffic load from the other input datastreams, thus the arrival rate (delay) of each datastream is not periodic. Therefore, the cell transfer is referred to as an *asynchronous* operation or ATM. In contrast, a *synchronous* transfer mode has fixed periods for cell transmission and reception.

10.1.2 ATM Example

Image transfer, one of the most frequently touted ATM applications, is one of the key features of the FISHnet, a fiber-optic system designed by Cablevision Systems Corp. of Woodbury, N.Y., that links physicians and researchers at Brookhaven National Laboratory, SUNY-Stony Brook, plus another Cablevision facility in Hicksville, N.Y. (see Figure 10-2a). FISHnet, which stands for Fiber Optic, Island-Wide, Super High-speed Network, currently provides links for medical and environmental researchers at the Brookhaven and Stony Brook locations.

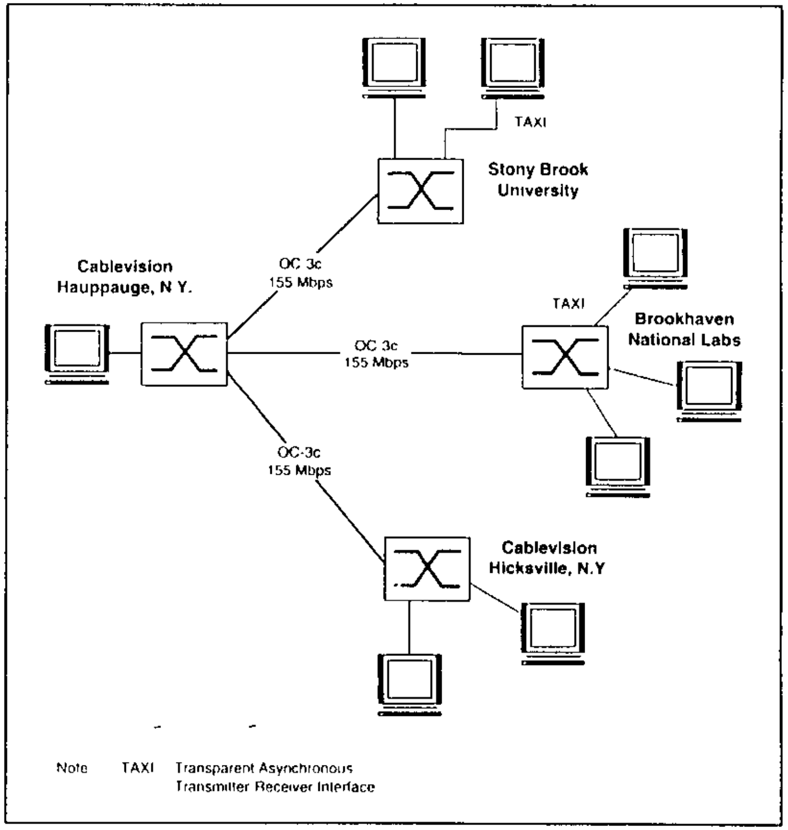


Figure 10-2a. FISHnet phase I

An optical backbone, operating at the OC-3c rate of 155 Mbps, connects ATM switches from Fore Systems Inc. of Warrendale, Pa., at each location. Sun workstations connect to the ATM switches using a 100 Mbps TAXI interface.

Currently, the FISHnet system lets doctors at Stony Brook Hospital read patient X-rays taken at Brookhaven almost as soon as they have been taken. If the doctor needs additional diagnostics, such as a different X-ray position, the X-ray can be retaken immediately.

It is planned to integrate the experimental ATM network into a larger ATM backbone infrastructure that will be deployed to provide frame relay and ATM services to the business community on Long Island (see Figure 10-2b). Many of these organizations have large investments in legacy LAN architectures. To protect this investment, the system will first implement LAN connectivity into ATM on the WAN, and will only gradually migrate to ATM at the desktop. So, ATM-capable hubs and routers will play an interim role in this connectivity plan. ATM switches will be deployed at each premise only when the application requirements can justify the hardware expense.

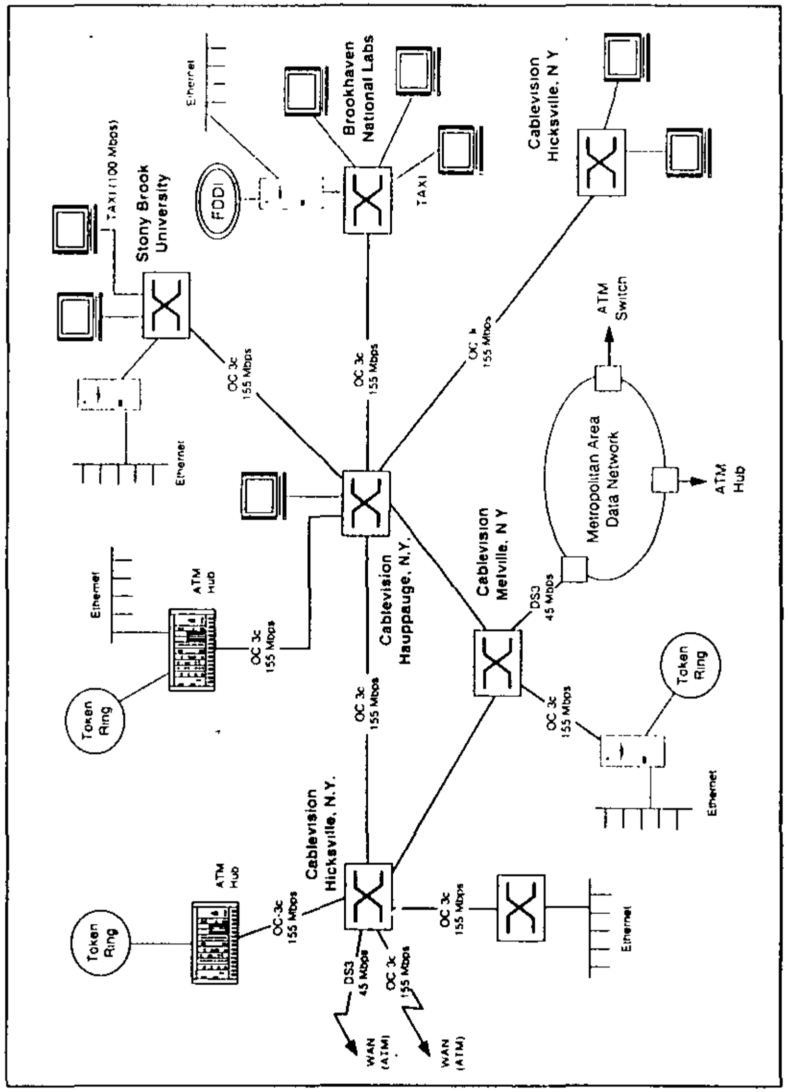


Figure 10-2b. FISHnet phase II.

10.2 ATM Standards

Several sources have defined standards for ATM networks: the ITU-T [10-22], the ATM Forum [10-23], Bellcore [10-24], and ANSI

The key ITU-T standards are as follows

ITU-T Recommendation	Description
I 113	B-ISDN Vocabulary of Terms
I 121	Broadband Aspects of ISDN
I.150	B-ISDN ATM Functional Characteristics
I 211	B-ISDN Service Aspects
I 311	B-ISDN General Network Aspects
I.321	B-ISDN Protocol Reference Model
I.327	B-ISDN Functional Architecture Aspects
I.361	B-ISDN ATM Layer Specification
I 362	B-ISDN ATM Adaptation Layer Functional Description
I 363	B-ISDN ATM Adaptation Layer Specification
I 371	Traffic Control and Congestion Control in B-ISDN
I.413	B-ISDN User-Network Interface
I 432	B-ISDN User-Network Interface Physical Layer Specification
I.555	Frame Relay and ATM Interworking
I 610	B-ISDN Operations and Maintenance Principles and Functions

The key ANSI standards for B-ISDN are.

ANSI Standard	Description
T1 511	Broadband ISDN—ATM Layer Cell Transfer Performance Parameters
T1 624	Broadband ISDN User-Network Interfaces—Rates and Formats Specifications
T1 627	Broadband ISDN—ATM Layer Functionality and Specification
T1.629	Broadband ISDN—ATM Adaptation Layer 3/4 Common Part Functions and Specifications
T1.630	Broadband ISDN—Constant Bit Rate (CBR) Services
T1 635	Broadband ISDN—ATM Adaptation Layer 5 Common Part Functions and Specifications
T1.636	Broadband ISDN—Signaling ATM Adaptation Layer—Overview Description
T1 637	Broadband ISDN—ATM Adaptation Layer Service Specific Connection Oriented Protocol (SSCOP)
T1 638	Broadband ISDN—Signaling ATM Adaptation Layer—Service Specific Coordination Function at the User Network Interface (SSCF at the UNI)
T1.638	Broadband ISDN—Signaling ATM Adaptation Layer—Service Specific Coordination Function at the Network-Network Interface (SSCF at the NNI)
T1 649	Broadband ISDN—Cell Relay Service Description

The ATM Forum does not develop its own standards, but rather fosters consensus between users and vendors regarding the use of standards such as the ITU-T recommendations. This process ensures a higher probability of interoperability between different vendors' products. The framework for the ATM documents is called the Anchorage Accord, illustrated in Figure 10-3, which describes the various documents' dependencies and inter-relationships. The key ATM Forum documents are as follows:

- ATM User-Network Interface (UNI) Specification, version 3.1
- Physical Layer Specifications for DS1 (1 544 Mbps), 6.312 Mbps, 25 6 Mbps, DS3 (44.736 Mbps), 155.52 Mbps, and others
- ATM DXI Specification, version 1.0
- ATM Broadband Inter-Carrier Interface (B-ICI) Specification, version 2.0
- LAN Emulation over ATM Specification, version 1.0
- Broadband Inter Carrier Interface (B-ICI) Specification, version 2 0
- ATM UNI Signaling Specification, version 4.0
- Traffic Management Specification, version 4.0
- Private Network-Network Interface (PNNI) Specification, version 1.0

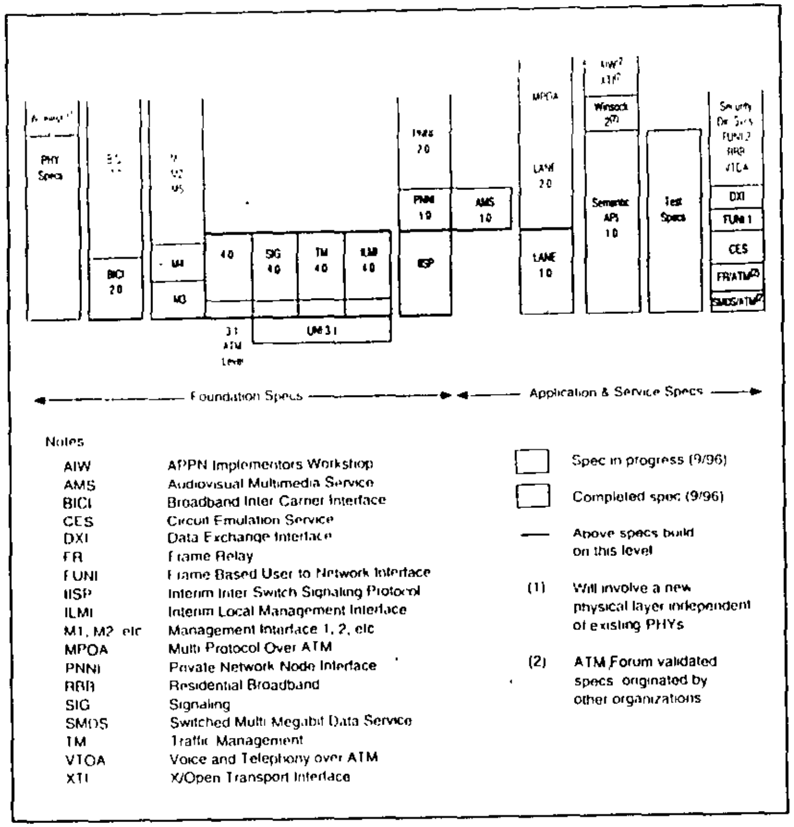


Figure 10-3. Anchorage Accord specification relationships.

Courtesy of the ATM Forum

10.3 Broadband ISDN Architecture

ITU-T Recommendation I.413 provides the reference configuration for the B-ISDN UNI. The reference configuration specifies various functional entities and the *reference points*, which are interfaces between them (see Figure 10-4). All interfaces, except for the R interface, have a designation beginning with the letter B, indicating broadband technology. The R interface may or may not have broadband capabilities (see References [10-25] and [10-26]).

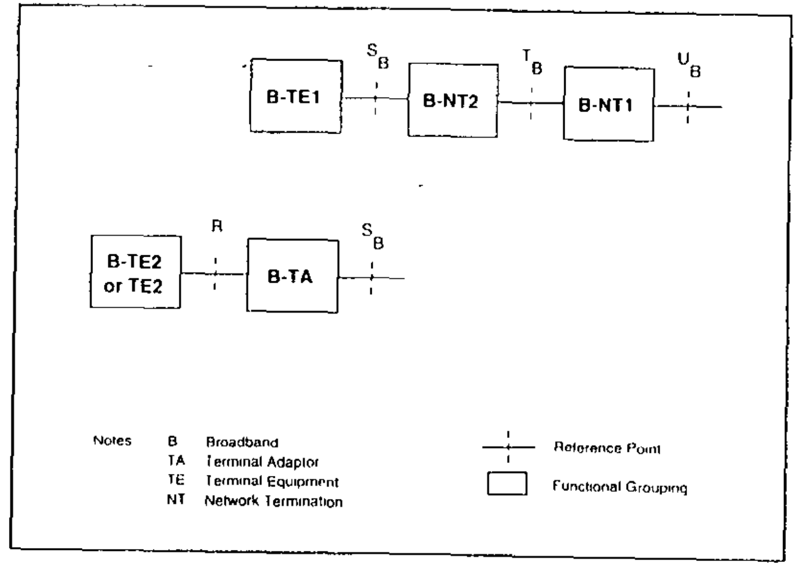


Figure 10-4. B-ISDN functional reference configuration.

Source: TR-NWT-001112, ©1993, Bell Communications Research, Inc., reprinted with permission

10.3.1 B-ISDN Reference Points

The reference points defined for B-ISDN include:

- R, the point between non–B-ISDN equipment (TE2 or B-TE2) and a terminal adapter (TA)
- S, the point between ISDN user equipment (B-TE1 or B-TA) and the customer premises network-termination equipment (B-NT2)
- T, the point between the customer premises network-termination equipment (B-NT2) and the public network termination (B-NT1)
- U, the point between the public network termination (B-NT1) and the public network

10.3.2 B-ISDN Functional Groups

The B-ISDN functions are grouped to indicate the operations they perform. These groups include:

- B-NT1 (broadband network termination 1), which handles the termination of the transmission line as well as the Operations and Maintenance (OAM) functions, such as a SONET line termination
- B-NT2 (broadband network termination 2), which may include higher-layer functions such as buffering, multiplexing, and signaling, as well as other examples such as PBX, LAN, or terminal controllers
- B-TE1 (broadband terminal equipment 1), which supports B-ISDN protocols
- B-TE2 (broadband terminal equipment 2), which supports a broadband interface other than B-ISDN
- TE2 (terminal equipment 2), which supports an interface other than ISDN
- TA (terminal adapter), which lets a B-ISDN user-network interface serve a B-TE2 or TE2

10.3.3 B-ISDN Architecture Model

The B-ISDN protocol architecture model consists of three planes and four layers (see Figure 10-5a). This model differs from the familiar OSI Reference Model in that it uses three, rather than two, dimensions. You can think of the planes as protocol suites.

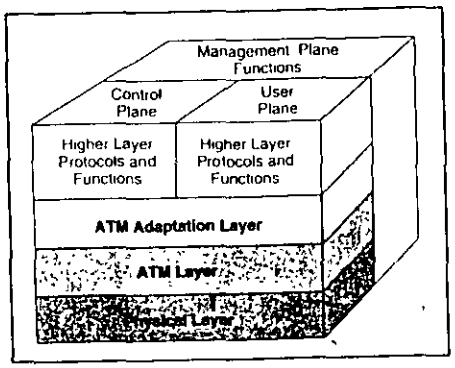


Figure 10-5a. B-ISDN protocol model.
Courtesy of STACKS, The Network Journal

The planes are designated user, control, and management. The *user plane* provides user-to-user information transfer and controls required for that transfer, such as flow control and error recovery. The *control plane* provides call-control and connection-control functions such as signaling. Signaling establishes, supervises, and releases calls and connections.

The *management plane* controls the ATM device, such as a switch or a hub. This plane offers two types of functions: plane management and layer management. Because *plane management* deals with the system as a whole (management of the other planes and coordination between the planes), it does not have a layered structure. *Layer management* deals with the resources and parameters residing at each protocol layer, such as OAM information flow.

The layers include: Physical (PHY), ATM, ATM Adaptation (AAL), and Higher. The next section describes these layers in more detail.

10.3.4 ATM Layers and Sublayers

Figure 10-5b illustrates the ATM layers and sublayers. As you'll see, the Physical layer sends and receives bits on the transmission medium, and it sends and receives cells to and from the next highest layer, the ATM layer. The ATM layer then switches these cells to the appropriate circuit to connect with an end system and its specific application or process. The payload within the cell is generated at, or destined for, the AAL, a layer that interfaces the Higher layer functions and processes with the ATM layer.

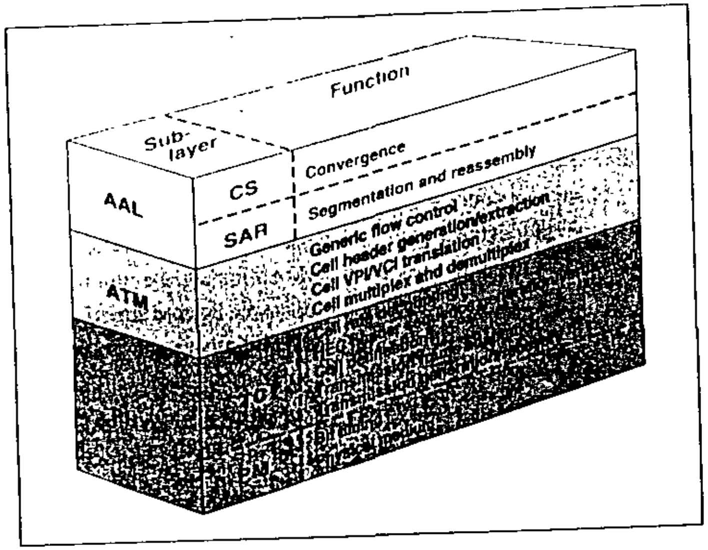


Figure 10-5b. ATM layers and sublayers.

Courtesy of STACKS, The Network Journal

The Physical layer has two sublayers, Physical Medium (PM) and Transmission Convergence (TC). The PM sublayer provides bit-level transmission. Its functions include the electrical or optical interface into the transmission medium, such as a cable, and the timing and recovery of those bits on the transmission medium.

The TC sublayer has five functions: frame generation, frame adaptation, cell delineation, Header Error Correction (HEC) generation, and cell-rate decoupling. Frame generation creates and recovers the data frame sent by the PM sublayer. Next, cells transmitted by the ATM layer must be adapted to the data-frame format required for the PM sublayer. In the receive direction, the frame-adaptation function extracts the cells from the frame. The cell-delineation function identifies the boundaries of cells so the ATM layer can decode them properly. Next, HEC sequence is calculated and added to the ATM header for transmitted frames. For received frames, the cell headers are checked for errors. If errors are found, they are corrected when possible. If they cannot be corrected, the cell is discarded. Finally, cell-rate decoupling inserts or suppresses idle cells, adapting the transmission rate of the valid ATM cells to the payload capacity of the transmission system.

The ATM layer functions independently of the Physical layer and performs four operations on cells: multiplexing, VPI/VCI translation, header generation, and flow control. In the transmit direction, the ATM layer multiplexes cells from individual virtual paths (VPs) and virtual channels (VCs) into a composite cell flow. In the receive direction, demultiplexing directs cells from the composite cell flow to the appropriate VP or VC. Next, the VPI/VCI fields in the incoming cell may require mapping to new VPI/VCI values. Third, the ATM layer generates an ATM header and adds it to the payload for transmission or extracts the payload from a received cell and passes that payload to the next highest layer. Finally, the ATM layer may generate cells to carry Generic Flow Control (GFC) information.

The AAL maps the higher layers (for example, services that define the signal type used) into the ATM layer (for instance, cells). AAL consists of two sublayers: the Segmentation and Reassembly (SAR) sublayer and the Convergence sublayer (CS). The SAR sublayer segments the variable-length higher-layer information to be transmitted into fixed-length ATM payloads, and reassembles the received payloads into the higher-layer information. The CS performs functions required by the AAL type in use, and is therefore service-dependent. In some cases, CS functions may be subdivided into a Common Part Convergence Sublayer (CPCS), or the lower sublayer; and a Service Specific Convergence Sublayer (SSCS), or the upper sublayer.

10.4 ATM Interfaces

A broadband network may include several interfaces (see Figure 10-6): the UNI, the ATM DXI, the network-node interface (NNI), the private network-node interface (PNNI), and the Broadband Inter-Carrier Interface (B-ICI).

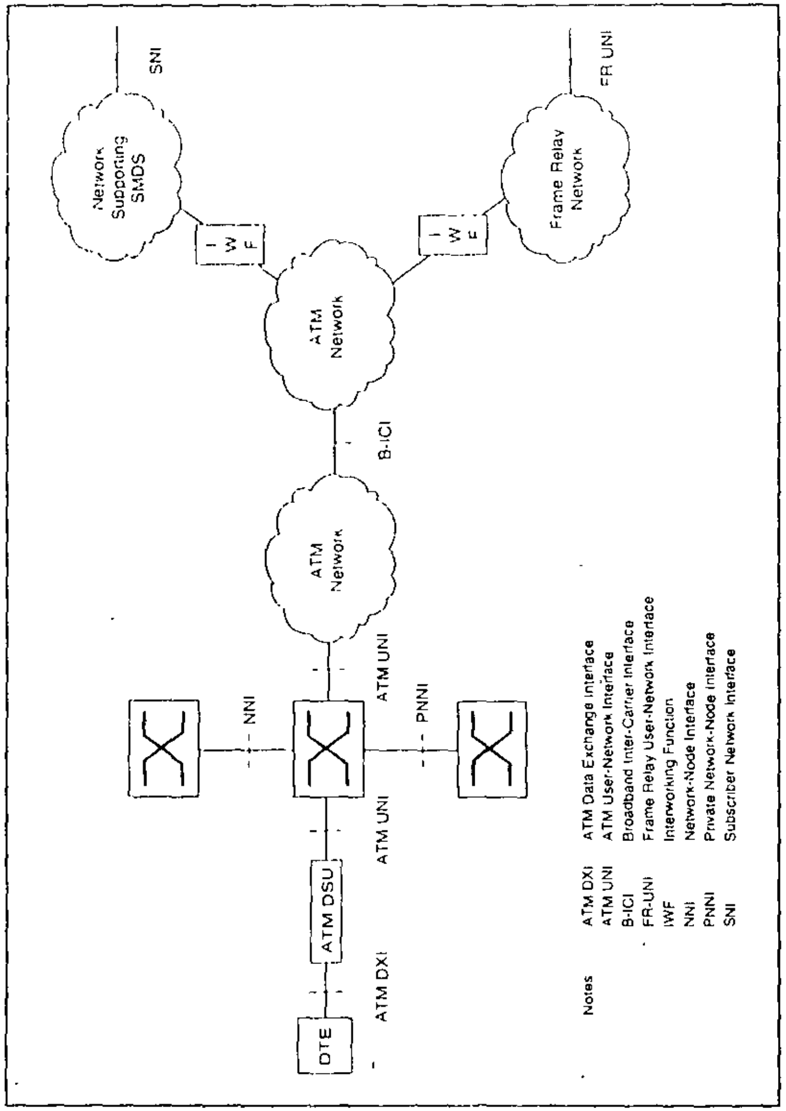


Figure 10-6 ATM interfaces.

The UNI connects the ATM network and premises equipment, which could include an ATM switch. There are two types of UNIs, public and private. A public UNI connects a private ATM switch to a public ATM service provider's network. A private UNI connects ATM users with the ATM switch [10-27].

Some applications divide the ATM protocol functions between the DTE, such as a router, and the hardware interface to the UNI, such as an ATM DSU. The DXI defines the protocol operations between these two devices.

The NNI describes network interconnection within a single carrier's network or between two carrier networks. The PNNI specifies a protocol through which ATM switches that are part of a private ATM network may communicate. The PNNI specification defines two possible configurations: a Private Network-Node Interface, operating between two switches; or a Private Network-to-Network Interface, operating between two groups of switches, or ATM networks. When an NNI interconnects public ATM carriers, it is often referred to as the B-ICI.

When an ATM network connects to either a public or private network, such as frame relay or SMDS, conversions between the two network protocols are required. IWF processes, which the ATM Forum's B-ICI specification defines, perform these conversions.

10.5 ATM Connections: VPIs and VCIs

Whether sent at the UNI or the NNI, each ATM cell contains information that identifies its virtual channel. This identification has two parts, which are both used at the ATM layer, a VPI and a VCI (see References [10-28] through [10-30]).

A *virtual path* is a bundle of virtual channel links, all having the same endpoint. So, the virtual path is like a large telephone cable, where all circuits terminate at a central office. The VPI is either assigned or removed to originate or terminate a virtual path link. These links are concatenated to form a virtual path connection (VPC). Each virtual channel link within a VPC maintains the cell transmission sequence, but does not ensure the integrity of an individual cell.

ITU-T Recommendation I.311 defines the *virtual channel* as "a unidirectional communication capability for the transport of ATM cells." A VCI is either assigned or removed, respectively, to originate or terminate a virtual channel link. Virtual channel links are concatenated to form a virtual channel connection (VCC), an end-to-end cell path at the ATM layer.

The physical transmission path (see Figure 10-7) contains the virtual paths and their VPIs, as well as the virtual channels and their VCIs.

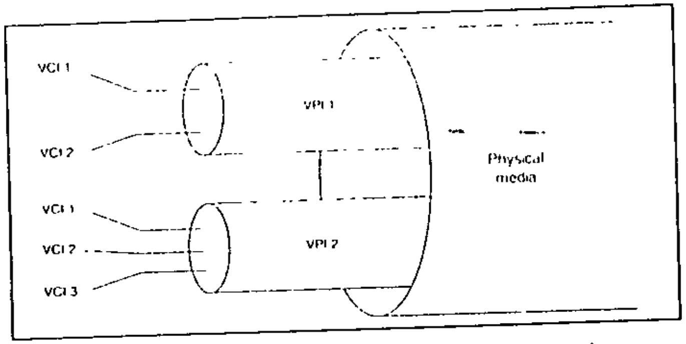


Figure 10-7. The relationship between virtual channels, virtual paths, and physical paths.

Source De Prycker, Peschi, and Van Landegem, "B-ISDN and the OSI Protocol Reference Model," IEEE Network Magazine, March 1993, © 1993 IEEE

The ATM layer provides the logical connection between two AAL processes. The virtual channel link connects a terminal equipment (TE) device with an ATM node (see Figure 10-8). The concatenation of two or more virtual channel links forms a VCC. Similarly, VPCs carry bundles of VCCs on an end-to-end basis.

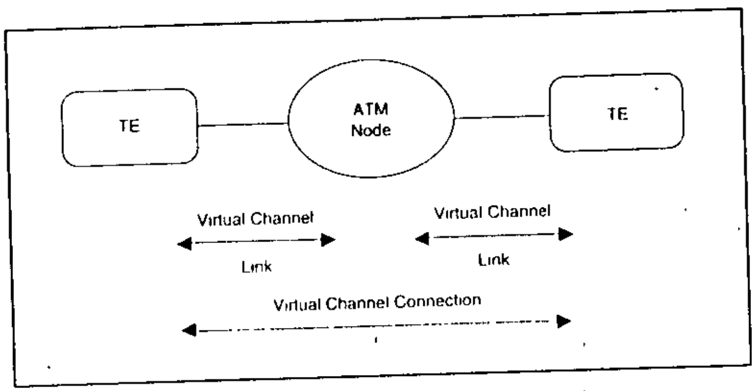


Figure 10-8. Virtual channel link and connection.

Source De Prycker, Peschi, and Van Landegem, "B-ISDN and the OSI Protocol Reference Model," IEEE Network Magazine, March 1993, © 1993 IEEE

An ATM node may also simultaneously support multiple end-user services (see Figure 10-9). Each service may require a different data-transfer mechanism, such as variable bit rate or constant bit rate. Thus, different AAL types have been defined. Each end-user service would be addressed by two VCI/VPI pairs: one to transmit and one to receive.

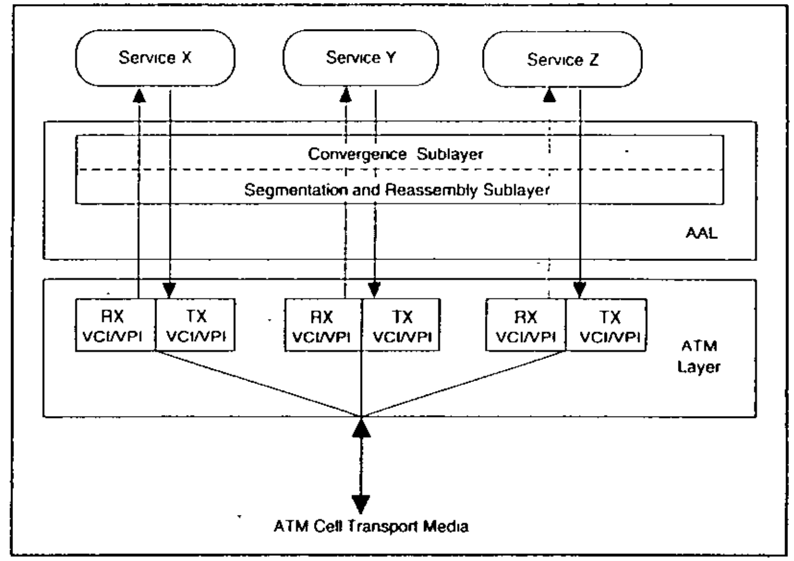


Figure 10.9. Virtual connection end points.

Source Armitage and Adams, "Packet Reassembly During Cell Loss," IEEE Network Magazine, September 1993, © 1993 IEEE

10.6 ATM Protocols

Specific protocols have been defined for each layer of the ATM architecture. The following sections look at the Physical, ATM, and AAL layers. ITU-T Recommendations I.413 and I.432, the ATM Forum's UNI specification [10-27] plus various physical specifications, and Bellcore's TR-NWT-001112 [10-31] are excellent references for the Physical layer. ITU-T Recommendations I.361, I.362, and I.363, plus Bellcore's GR-1113-CORE [10-32], explain the ATM and ATM Adaptation layers.

10.6.1 The Physical Layer: Physical Medium and Transmission Convergence (TC)

The ATM PHY layer contains two sublayers, TC and the Physical Medium Dependent (PMD). The PMD sublayer provides the physical interface to the cable and deals with timing, connectors, and so on. The TC interfaces with the ATM layer. It extracts cells from the incoming PMD bit stream and passes them to the ATM layer, and vice versa.

ITU-T Recommendation I.432 defines two options for the B-ISDN UNI Physical layer. The first specification operates at 155.520 Mbps over two coaxial cables. The second operates at 622.080 Mbps over two single-mode fiber cables.

In the UNI 3.1 Specification, the ATM Forum defines a number of options for the Physical layer interface at either public or private UNIs. These include

- A public or private UNI for the SONET STS-3c interface at 155.520 Mbps, operating over multimode or single-mode fiber
- A public or private UNI for the DS3 interface at 44.736 Mbps, operating over coax pairs
- The Transparent Asynchronous Transmitter/Receiver Interface (TAXI®), at 100 Mbps over multimode fiber (private UNI). This interface was developed by Advanced Micro Devices Inc. and uses 4B/5B encoding, based on the encoding scheme used with FDDI.
- A private UNI operating at 155.520 Mbps with an 8B/10B data-encoding scheme, operating over multimode fiber or shielded twisted-pair cables
- Future development for an interface operating at the E3 rate (34.368 Mbps), based on ITU-T G.703 and G.804
- Future development for an interface operating at the E4 rate (139.264 Mbps), based on ITU-T G.703 and G.804.

Since the publication of the UNI 3.1 Specification, the ATM Forum has developed other interface specifications

- A public UNI for the DS1 interface at 44.736 Mbps, operating over twisted pairs
- A private UNI at 25.6 Mbps, operating over unshielded or shielded twisted pairs
- A UNI at 6.312 Mbps for the DS-2 interface
- A UNI for the STS-1 interface operating at 155 Mbps over unshielded twisted pairs (UTP-3 cable)
- A UNI for the STS-3c interface operating at 155 Mbps over unshielded twisted-pairs (UTP-5 cable).
- A UNI for the OC-12 interface operating at 622.08 Mbps.
- The Universal Test and Operations Physical Interface for ATM (UTOPIA)

The above interfaces are documented in individual documents and are available on the Forum's Web and FTP servers [10-23].

10.6.2 The ATM Layer

You can loosely compare the ATM layer and its associated Physical layer with the OSI Physical layer. Both architectures require a physical transmission medium, including the cable type, connectors, and so on, as shown in the lower portion of Figure 10-10. But the ATM layer also includes the VPIs and the virtual connection that perform multiplexing (see upper portion of Figure 10-10). These multiplexing functions have been described as a virtual physical service, because the VCCs with the VPIs and VCIs act as a "virtual wire" between two end points, as described in Reference [10-26].

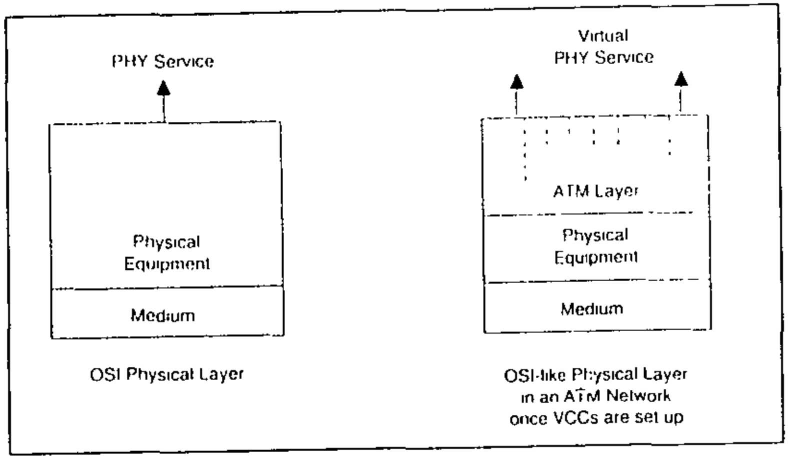


Figure 10-10. OSI and ATM networks: the physical service.

Source De Prycker, Peschi, and Van Landegem "B-ISDN and the OSI Protocol Reference Model," IEEE Network Magazine March 1993 © 1993 IEEE

10.6.3 ATM Layer Service Categories

Many recently published standards, including the ITU-T I 371 [10-33] and ATM Forum documents, such as the Traffic Management specification [10-34], describe service categories for the ATM layer that enable ATM technology to be used for a wide variety of applications (In the next section, we will study the ITU-T's classes of service that apply to the ATM Adaptation Layer, or AAL) Both ATM providers and ATM users benefit when the type of service provided and the type of service supplied are clearly defined.

Attribute	ATM Layer Service Category				
	CBR	rt-VBR	nrt-VBR	UBR	ABR
Traffic Parameters					
PCR and CDVT(4,5)	Specified			Specified(2)	Specified(3)
SCR, MBS, CDVT(4,5)	n/a	Specified		n/a	
MCR(4)	n/a			n/a	Specified
QoS Parameters:					
Peak-to-Peak CDV	Specified		Unspecified		
MaxCTD	Specified		Unspecified		
CLR(4)	Specified			Unspecified	(1)
Other Attributes:					
Feedback	Unspecified				Specified(6)

Abbreviations

Service Categories

CBR	Constant Bit Rate
rt VBR	Real-Time Variable Bit Rate
nrtVBR	Non-Real-Time Variable Bit Rate
UBR	Unspecified Bit Rate
ABR	Available Bit Rate

Traffic Parameters

CDVT	Cell Delay Variation Tolerance
MBS	Maximum Burst Size
MCR	Minimum Cell Rate
PCR	Peak Cell Rate
SCR	Sustainable Cell Rate

QoS Parameters

CDV	Cell Delay Variation
CLR	Cell Loss Ratio
CTD	Cell Transfer Delay

Notes

(1) CLR is low for sources that adjust cell flow in response to control information Whether a quantitative value for CLR is specified is network specific
(2) May not be subject to CAC and UPC procedures
(3) Represents the maximum rate at which the ABR source may ever send The actual rate is subject to the control information
(4) These parameters are either explicitly or implicitly specified for PVCs or SVCs
(5) CDVT refers to the Cell Delay Variation Tolerance CDVT is not signaled In general CDVT need not have a unique value for a connection Different values may apply at each interface along the path of a connection
(6) Defined by the Flow Control Model and Service Model for the ABR Service Category.

Figure 10-11. ATM layer service categories.

Source Traffic Management Specification Version 4 0, The ATM Forum, April 1996 Copyright © 1996, The ATM Forum

The ATM Service Architecture specifies five different service categories: Constant Bit Rate (CBR), Real-Time Variable Bit Rate (rt-VBR), Non-Real-Time Variable Bit Rate (nrt-VBR), Unspecified Bit Rate (UBR), and Available Bit Rate (ABR), as shown in Figure 10-11. Each of these five categories has a number of traffic parameters, including the Minimum Cell Rate (MCR), Peak Cell Rate (PCR), Sustainable Cell Rate (SCR), Cell Delay Variation Tolerance (CDVT), and Maximum Burst Size (MBS). In addition, there are Quality of Service (QoS) parameters, including Peak-to-Peak Cell Delay Variation (CDV), Maximum Cell Delay Variation (maxCDV), and Cell Loss Ratio (CLR). For ABR service, a feedback mechanism is used to control the flow of the cells; it is described in detail in Reference [10-34]. The specific services are defined below:

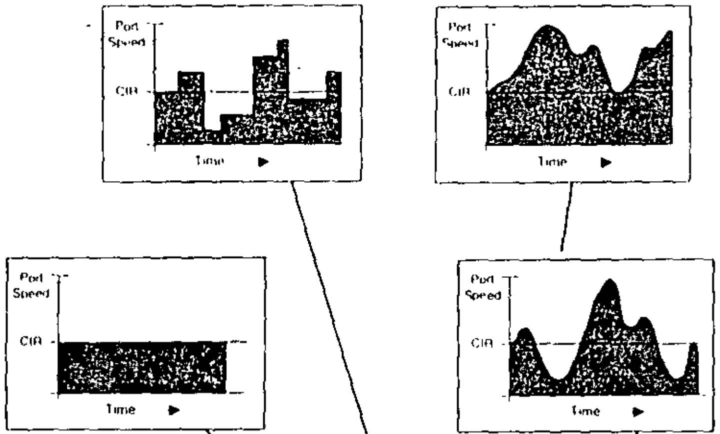
- *Constant Bit Rate (CBR) service* provides a fixed bandwidth for the duration of the circuit connection. In addition, a timing relationship is maintained between the source and destination. This service is intended to be used by real-time applications that require constraints on the CTD and the CDV, such as voice, videoconferencing, television, or circuit emulation service (CES).
- *Variable Bit Rate (VBR) service* is divided into two subclasses: real-time VBR (rt-VBR) and non-real-time VBR (nrt-VBR). The rt-VBR is intended for applications that are time-sensitive, such as multimedia, where the cell delay and cell delay variation must be controlled. The nrt-VBR is intended for applications—such as transaction processing or frame relay interworking—that have bursty traffic, but do not require such stringent controls on cell delay.

- *Unspecified Bit Rate (UBR) service* does not guarantee any traffic or QoS parameters. When excess network capacity exists, service is offered to UBR connections, similar to a zero CIR with frame relay. As such, it is referred to as a best-efforts service. Example applications include data or image retrieval or remote terminal functions.
- *Available Bit Rate (ABR) service* is intended to support applications that are able to increase or decrease their information throughput if network circumstances dictate. To implement ABR service requires an end-to-end, rate-based flow control mechanism for the data. The PCR traffic descriptor is negotiated (the user's commitment not to exceed), as is a Minimum Cell Rate (the network's commitment to provide). ABR service is thus designed for non-real-time applications that do not have delay sensitivity. Applications for ABR would include data transfers, such as LAN emulation and distributed file services.

The topic of ATM traffic management has generated a great deal of research in the last few years. References [10-35] through [10-40] are examples of some of the work that has been done in this area.

10.6.4 ATM Adaptation Layer CS and SAR

The AAL translates data from the higher layer into the cell formats carried in the ATM layer. Recommendation I.362 defines four classes of services that the AAL provides, which depend on three parameters: the timing relation between source and destination (required or not required), the bit rate (constant or variable), and the connection mode (connection-oriented or connectionless). The use of these four classes—A, B, C, and D, described below—minimizes the number of AAL protocols (Figure 10-12).



	Class A	Class B	Class C	Class D
Timing relation between source and destination	Required		Not Required	
Bit Rate	Constant	Variable		
Connection Mode	Connection Oriented			Connectionless
Applications	Voice, Video Circuit Emulation	Compressed Voice or Video	Frame Relay X 25 Traffic	SMDS, LAN Traffic

Figure 10-12. AAL service classes.
Courtesy of AT&T

It should be noted that at the present time, the ATM Forum's ATM layer service categories (CBR, VBR, etc.) are used more frequently than the ITU-T Classes (A, B, etc.) to describe the type of service provided to the application. Nevertheless, the ITU-T classes of service described below are found in many reference documents, and are discussed here for the sake of completeness

- Class A: Connection-oriented, constant bit-rate data with a timing relationship between source and destination. Examples include PCM encoded voice, constant bit-rate video, and DS1 circuits.
- Class B: Connection-oriented, variable bit-rate data with a timing relationship between source and destination Examples include compressed audio or video
- Class C: Connection-oriented, variable bit-rate data with no timing relationship between source and destination Examples include frame relay or X 25 traffic.
- Class D. Connectionless, variable bit-rate data with no timing relationship between source and destination. Examples include SMDS or LAN traffic.

Four different types of AALs are defined in ITU-T I.363. These have been defined to optimize the transmission of the four classes of traffic:

- Class A: AAL Type 1
- Class B. AAL Type 2 (currently being developed)
- Class C: AAL Type 3/4 and AAL Type 5
- Class D: AAL Type 3/4

The associations between the service classes and AAL types are not restrictive, however. Specific implementations may deviate from the list above.

10.6.5 Layer Operation and Interaction

To see the functions of the Physical, ATM, and AAL layers, follow the transmission of a cell from one ATM layer entity to another (see Figure 10-13 from Reference [10-32])

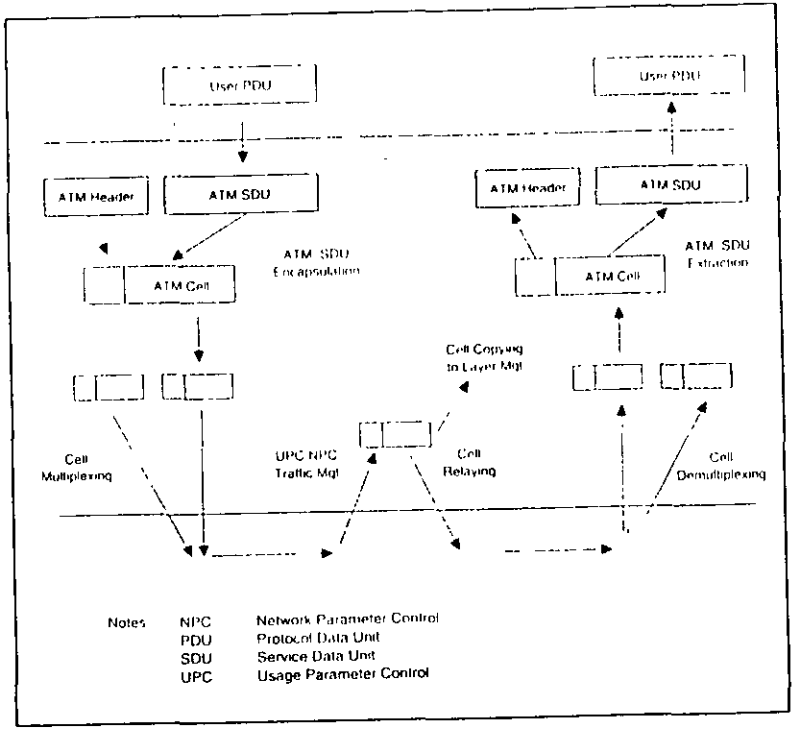


Figure 10-13. ATM_SDU encapsulation and extraction.

Source GR-1113-CORE, © 1994, Bell Communications Research, Inc., reprinted with permission

The originating ATM user, such as the AAL, sends a User PDU to the ATM entity This User PDU becomes the ATM SDU, or ATM payload at the originating ATM entity The ATM_SDU is encapsulated with the ATM header, which adds VCI/VPI and other control information

The resulting ATM cell is multiplexed with other cells and transmitted on the Physical medium. The logical connection between originating and receiving ATM entities is called the ATM peer-to-peer (APP) connection. Two control functions, Usage Parameter Control (UPC) and Network Parameter Control (NPC), monitor the traffic on that connection to ensure conformance with negotiated parameters. Some ATM entities perform network management, and may copy the payload of a cell and send it to the ATM Management (ATMM) entity for further analysis.

The receiving ATM entity performs the processes described above in the reverse order, first demultiplexing the cells, extracting the ATM_SDU, and finally passing the ATM_SDU as a User PDU to the next higher layer.

10.7 The ATM DXI

The ATM DXI lets a DTE, such as a router, and a DCE, such as an ATM DSU, jointly process the ATM protocol suite. The ATM Forum's DXI specification details the division of protocol responsibilities and the DXI operation [10-41].

The objective for this division of labor is to preserve the protocol functions at the ATM UNI, perform most protocol operations in a specialized DSU (typically referred to as an ATM DSU), and let you change the protocol in the router via a software upgrade

The DXI Physical layer uses V.35, EIA/TIA 449/530, or EIA/TIA 612/613 (High-Speed Serial Interface—HSSI) interfaces. The DXI Data Link layer protocol is derived from the high-level HDLC protocol Information from the DTE is encapsulated within the DXI frame and sent to the DCE The DCE converts the frame to the appropriate ATM protocol suite.

The mode of operation (1a, 1b, or 2) and the AAL protocol type (3/4 or 5) used determine the protocols implemented within the DCE (see Figure 10-14).

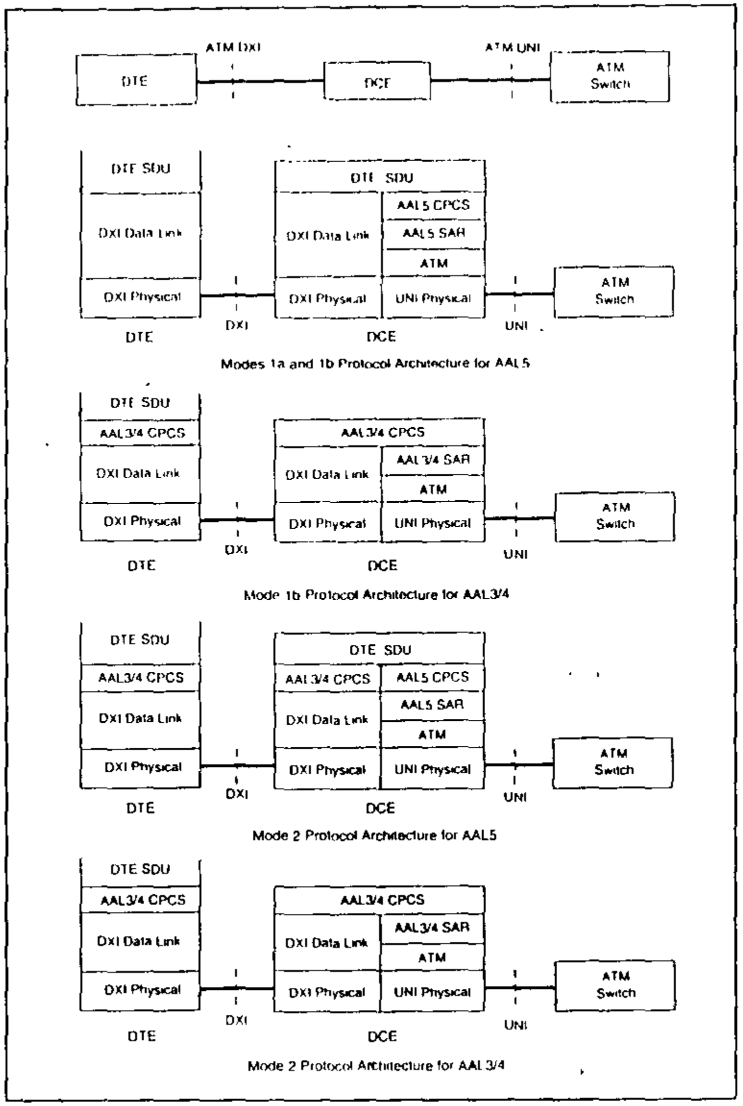


Figure 10-14. The ATM DXI architecture.

Courtesy of the ATM Forum

Mode 1a is used with AAL5 only, and can handle up to 1,023 VCs. The DTE SDU may be up to 9,232 octets long. The DCE implements the AAL5 CPCS and SAR sublayers, in addition to the ATM and Physical layers. A 16-bit FCS is implemented between DTE and DCE.

Mode 1b uses AAL3/4 for at least one VC, and AAL5 for other VCs, up to 1,023 VCs. The DTE SDU may be up to 9,224 octets long for AAL3/4 and 9,232 octets long for AAL5. A 16-bit FCS is implemented between DTE and DCE.

Mode 2 uses AAL3/4 and AAL5, one per VC, for up to 16,777,216 VCs. The DTE SDU may be up to 65,535 octets long. A 32-bit FCS is implemented between DTE and DCE.

10.8 The Frame-Based User-to-Network Interface

The Frame-based User-to-Network Interface (FUNI) is based on the ATM Data Exchange Interface, and defines an interface between DTE and an ATM network that operates at the DS1 (1.544 Mbps) or E-1 (2.048 Mbps) rates.

The user information is carried on a DS1/E1 physical circuit, and may range from a fractional up to a full DS1/E1 bandwidth. Once the user information is inside the ATM network, a conversion function changes the FUNI frame into ATM cells. This conversion function is defined for both AAL3/4 and AAL5 operation.

The Data Link Layer protocol defined for FUNI is identical to the original ATM DXI protocol. Support for operational Mode 1a is required, Mode 1b is optional, and Mode 2 is prohibited. The FUNI frame is delimited by a one-octet flag in the header and trailer. A two-octet FUNI header precedes the User Information (User_SDU), and a two-octet Frame Check Sequence (FCS) follows the User_SDU. Support for a User_SDU up to 4,096 octets long is required, with lengths up to 64K optional. The FUNI header includes a Frame Address (FA) field, which is mapped to the ATM VPI/VCI; a Congestion Notification (CN) field, which is mapped to the ATM PTI field; and a Cell Loss Priority (CLP) field, which is mapped to the ATM CLP field.

The FUNI may be multiplexed with other DS-n signals for transport efficiency. For example, a DS1 FUNI could be multiplexed with other DS1 signals for transport over a DS-3 line, and then demultiplexed back down to a DS1 FUNI signal before connecting to the ATM network. In a similar fashion, a fractional T1 (FT1) FUNI (Nx64 Kbps) could be multiplexed with other FT1 signals, transported over a DS1 ine, and then demultiplexed to a fractional payload of a DS1 before connecting to the ATM network. In this way, FUNI information and other signals, such as voice, may be combined for more efficient transport to the carrier, and then separated before further processing.

For additional details, refer to the ATM Forum's FUNI Specification [10-42] and Reference [10-43]

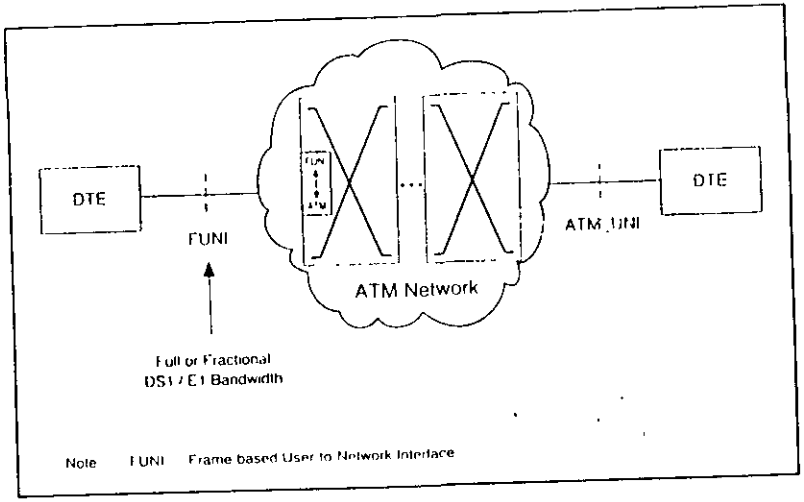


Figure 10-15. FUNI architecture.

10.9 Multiprotocol Encapsulation over AAL5

The popularity of the TCP/IP suite means that new networking technologies, such as ATM, must support these protocols. RFC 1483, "Multiprotocol Encapsulation over ATM Adaptation Layer 5" [10-44], defines two methods of support: LLC encapsulation and VC-based multiplexing. For both cases, the TCP/IP information is carried in the payload field of the Common Part Convergence Sublayer PDU, and the SSCS of AAL5 is empty.

The LLC encapsulation method is based on techniques developed for use with SMDS. This method allows multiplexing of multiple protocols over a single ATM virtual circuit. The receiver uses information contained within LLC and SNAP headers to identify the protocol carried within that PDU. The LLC encapsulation method is used when it is not feasible to have a separate VC for each protocol or when network charges are based on the number of active VCs.

The VC-based multiplexing technique uses ATM VCs to implicitly provide higher-layer protocol multiplexing. In other words, each protocol is carried on a separate VC. This method is used when it is feasible and economical to dynamically create large numbers of virtual circuits.

Reference [10-45] discusses some of the implementation issues in greater detail.

10.10 LAN Emulation

LAN Emulation, or LANE as it is commonly known, is a service that allows existing end-user applications to access an ATM network. More importantly, this access should appear to the application as if it were using more traditional protocols, such as TCP/IP or Novell's Internetwork Packet Exchange (IPX), and running over more traditional LANs such as Ethernet or token ring. One of the design constraints is to account for the differences in protocol design—ATM is connection-oriented, whereas IP and IPX are connectionless. A number of functions, including setting up the ATM connection and translating LAN to ATM addresses, must be hidden from the upper layers, thus making the application think it is operating over a traditional network.

The ATM Forum has defined two different interfaces for LAN Emulation: a LAN Emulation User to Network Interface, called LUNI; and a LAN Emulation Network to Network Interface, called LENNI. Current work has focused on the LUNI. The ATM Forum's LAN Emulation specification [10-46] defines two scenarios that are applicable. In the first, an ATM network may be used to interconnect Ethernets to Ethernets, an Ethernet to an ATM device, or an ATM device to another ATM device. The second scenario replaces Ethernet LANs with token ring LANs under similar conditions. To make either of these systems operate requires the LAN Emulation protocol stack, shown in Figure 10-16. Notice that the LAN host and its applications operate over traditional protocols, such as TCP/IP and IPX, and that a driver, such as NDIS or ODI, provides an interface between the upper-layer software and the MAC layer hardware. The ATM-to-LAN converter sits at the edge of the network running dual protocol stacks: one that communicates with the LAN (on the right) and another that communicates with the ATM switch (on the left). Note that this ATM-to-LAN converter is functioning as a bridge, operating independent of the Network and higher-layer protocols. The ATM switch (or switches) do not participate in LAN emulation other than to switch the ATM connections, as would be the case with any other ATM-based network scenario. An element of LAN emulation is also active on the ATM host (left side of Figure 10-16), masking the ATM functions from the higher-layer processes as well

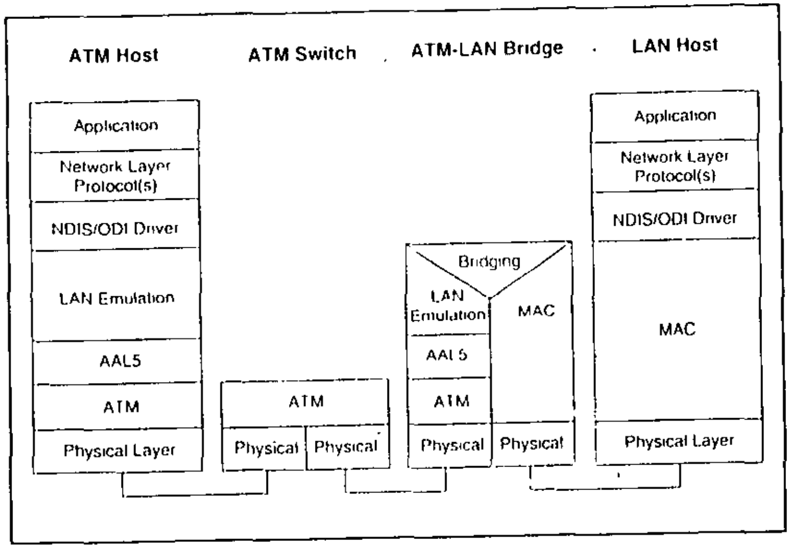


Figure 10-16. LANE architecture.

Courtesy of 3Com Corporation

In summary, the LAN emulation function maps the Ethernet or token ring MAC layer functions into ATM virtual connections, while shielding the application from the connection setup and handshaking functions that the ATM switch requires. The

LANE architecture is designed around a client/server paradigm, such that the LAN Emulation Client (LEC) derives information that it needs from one of several servers: the Configuration Server, the LAN Emulation Server (LES), or the Broadcast and Unknown (BUS) Server. The LEC software may be incorporated into workstation drivers, or it could be incorporated into other internetworking devices such as routers or switches. References [10-47] through [10-50] provide additional details on the operation and implementation of LAN Emulation systems.

10.11 Multiprotocol over ATM

Multiprotocol over ATM, or MPOA as it is commonly known, is a service model for end-to-end internetworking across an ATM network infrastructure. In some cases the attached devices may be running the ATM protocols, and in other cases the devices may be legacy systems, such as Ethernet or token ring LANs, that connect to the ATM network through an edge device.

MPOA is considered to be an evolution of LANE technology. The key difference between LANE and MPOA is in the layers of protocol operation—LANE operates at Layer 2 (bridging only), while MPOA operates at both Layers 2 and 3 (bridging and routing). MPOA uses LANE for its Layer 2 forwarding functions. Thus the scope of LANE is a single Layer 3 subnetwork, while MPOA allows devices to establish direct communication across ATM connections, even if the devices are in different subnetworks.

The MPOA architecture includes several key components: ATM-attached hosts, edge devices, and route servers (Figure 10-17). In addition, devices may be grouped logically into an Internet Address Summarization Group, or IASG. The IASG is defined as "a range of internetwork layer addresses summarized into internetwork layer routing," which is similar to a subnet and its range of addresses [10-51]. In addition, an IASG is protocol-specific, thus, a device that operates two internetworking protocols (such as IP and IPX) would be a member of at least two IASGs.

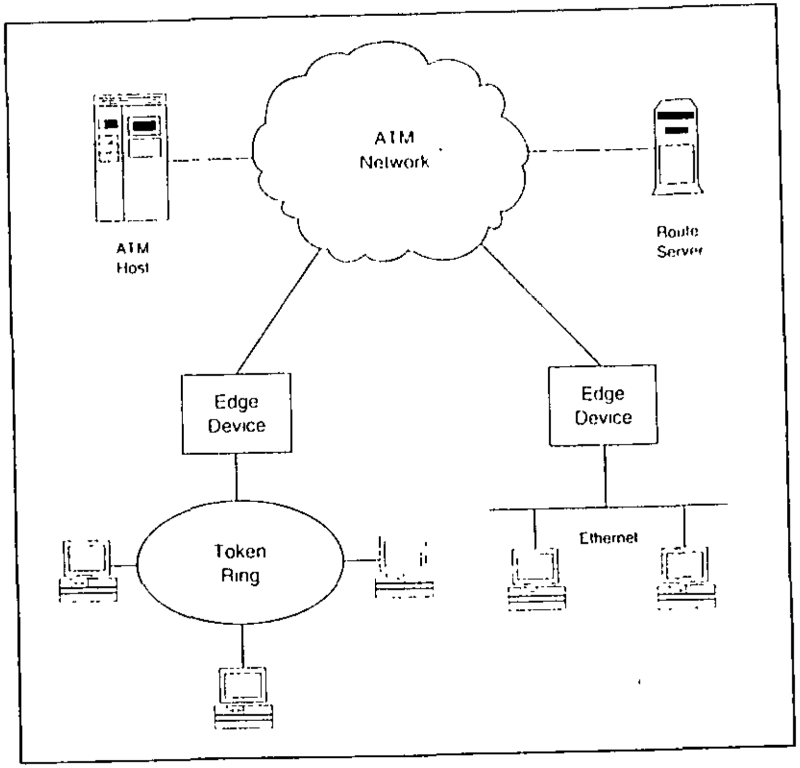


Figure 10-17. MPOA architecture.

An *edge device* provides connectivity between a legacy technology and ATM. An example would be a hub that supported token ring interfaces on one side and an ATM interface on the other. An ATM-attached host can connect directly to the ATM network and include the protocols necessary to participate in the MPOA service. The *route server* is a physical and/or logical device that provides routing information to other devices within the internetwork. This information includes Layer 2, Layer 3, and ATM addresses. If two devices that are attached to the same edge device need to communicate, the edge device forwards the packets using LANE procedures. If the packet needs to go outside its own IASG, the edge device obtains the ATM address from either its internal address cache or from a query and response from the route

server. Route servers, in turn, communicate with each other to discover and update their information regarding addresses and available routes.

The MPOA architecture is expected to be completed in the 1997 timeframe. References [10-52] through [10-55] discuss application and implementation strategies for MPOA.

10.12 ATM Signaling

Because ATM provides a connection-oriented service, it uses signaling to set up and clear the connections. Signaling provides functions such as the ability to establish point-to-multipoint connections, to identify virtual paths and virtual connections, to recover from network errors, to support various ATM address formats, and to communicate end-to-end compatibility parameters (see Figure 10-18). Signaling messages are passed between any combination of three elements: endpoint equipment, such as an ATM switch; a private ATM network; or a public ATM network.

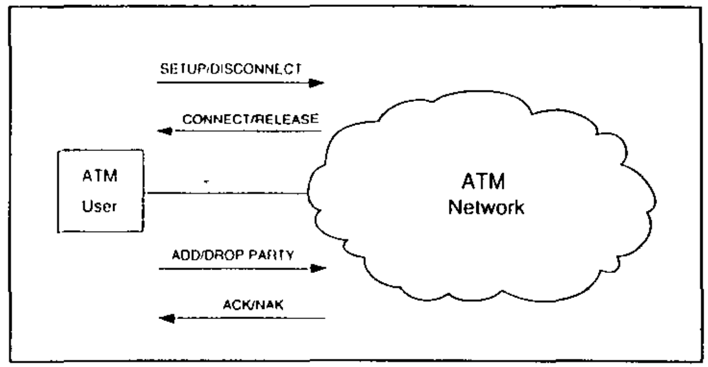


Figure 10-18. ATM signaling.

The ATM signaling protocols are based on ITU-T Recommendation Q.2931, formerly called Q.93B. The ATM Forum's Signaling Specification [10-56], Bellcore's GR-1111-CORE [10-57], and Reference [10-58] also address signaling issues.

The ATM Forum specification addresses signaling between endpoint equipment and a public network (the Public UNI), as well as signaling between endpoint equipment and a private network (the Private UNI). Private ATM networks may use the private UNI signaling. The ATM Forum's B-ICI specification addresses signaling between public ATM networks.

10.13 Interworking

ATM technology must interoperate with other broadband alternatives. The following sections consider interworking between ATM and frame relay, and between ATM and SMDS.

10.13.1 ATM/Frame Relay Interworking

Frame relay is an established broadband networking protocol, while ATM is emerging as the broadband heir apparent. Therefore, many organizations are concerned with preserving their investment in frame relay networking hardware while migrating to ATM. An architecture that solves this problem was proposed by AT&T, Cisco Systems Inc., and StrataCom Inc., and was further developed by the Frame Relay and ATM Forums in the documents entitled "Frame Relay/ATM PVC Network Interworking Implementation Agreement," FRF.5 [10-59], "Frame Relay/ATM PVC Service Interworking Implementation Agreement," FRF.8 [10-60], the ITU-T in Recommendation I.555 [10-61], and Reference [10-62].

PVCs between an ATM UNI and a frame relay UNI or NNI provide logical connections between frame relay and ATM. Figure 10-19 illustrates two PVCs. PVC I connects a frame relay user device with an ATM user device. PVC II connects a frame relay network with an ATM user device. Examples of user devices include terminal equipment and a router or a switch.

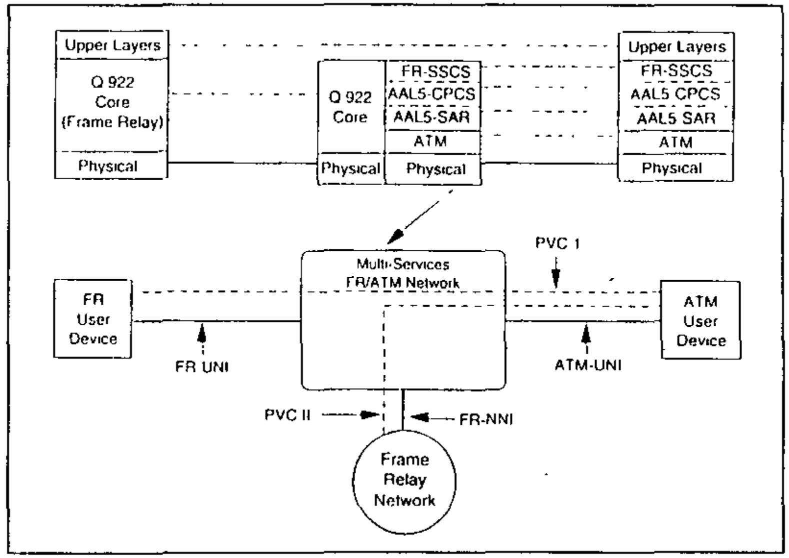


Figure 10-19. ATM/frame relay interworking.

Courtesy of AT&T, Cisco Systems, Inc., and StrataCom Inc

Figure 10-19 illustrates the operations required to convert between the frame relay and ATM protocols. The frame relay user device uses the Q.922 Core protocols while the ATM user device uses ATM, AAL5, and the Frame Relay Service Specific Convergence Sublayer (FR-SSCS). The multiservices FR/ATM network contains both protocol stacks.

The protocol conversion occurs in two steps. First, it makes a correspondence between the two PVC identifiers: the ATM VPI/VCI and the FR DLCI. Second, it maps the protocol data unit between the FR-SSCS protocol and the Q.922 core protocol.

Additional ATM/frame relay interoperability issues that the architecture addresses but that are not shown in Figure 10-19 include:

- conversion between frame relay and ATM protocols
- mapping between frame relay and ATM virtual circuits
- alignment of frame relay and ATM traffic-management parameters, such as the conversions of the frame relay CIR into a meaningful parameter for ATM traffic
- mapping of the local management information (the LMI used at the FR-UNI and the ILMI used at the ATM UNI)

The next two sections will explore the Network Interworking alternatives.

10.13.1.1 Network Interworking

Two types of interworking have been defined: Network Interworking and Service Interworking. The key differences between these two types are the location where the protocol conversions occur and the awareness of the end stations to that protocol conversion.

With FR/ATM *Network Interworking*, two frame relay users are connected via an ATM network. The presence of the ATM network as a transport between the two end users is not visible to those end users. In other words, one protocol is used on the two ends of the connection, and another protocol is used in between.

Two scenarios for Network Interworking are possible. In the first scenario, both ends of the connection are frame relay DTE (Figure 10-20). The FR/ATM protocol conversion is performed just before the data enters that ATM (or B-ISDN) network. In the second scenario (Figure 10-21), one end of the connection is frame relay (need-

ing the conversion to ATM or B-ISDN) and the other end of the connection is an ATM (or B-ISDN) user running a frame relay Service Specific Convergence Sublayer (FR-SSCS), which requires no further conversion.

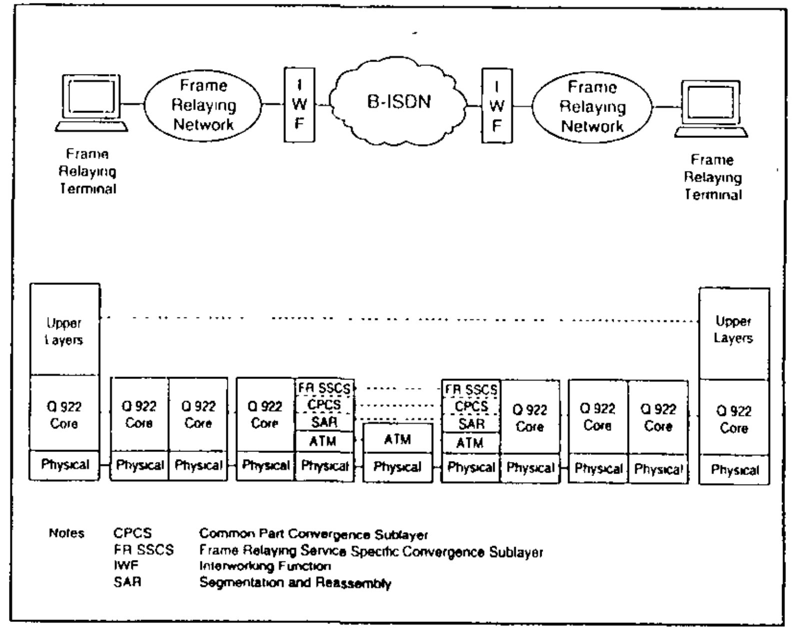


Figure 10-20. Network Interworking between Frame Relaying Bearer Service and B-ISDN (scenario 1).

Source ITU-T I 555

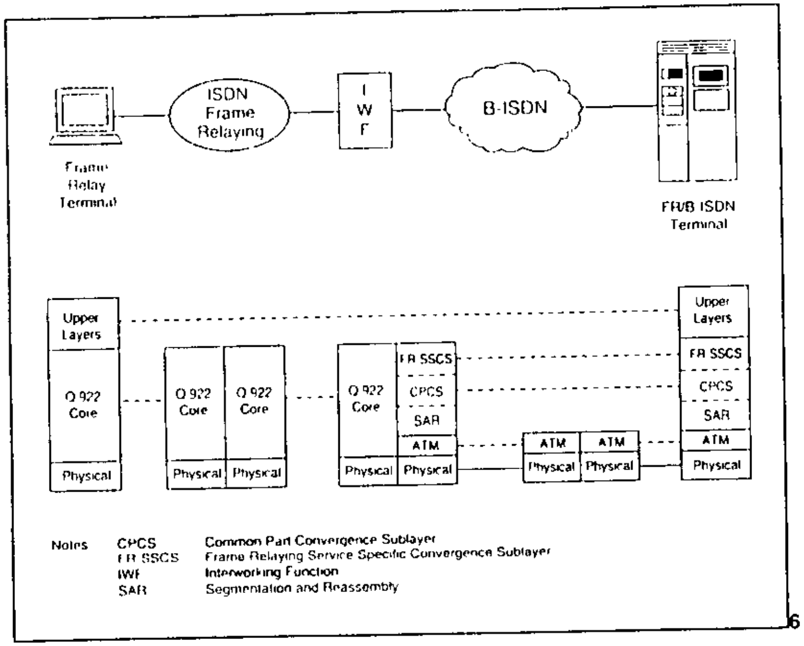


Figure 10-21. Network Interworking between Frame Relaying Bearer Service and B-ISDN (scenario 2).

Source ITU-T I.555

10.13.1.2 Service Interworking

Service Interworking, in contrast, allows users running dissimilar protocols to communicate directly, with the protocol conversion provided by the network. With Service Interworking, the frame relay user performs no ATM service specific functions and the ATM user performs no frame relay service specific functions (Figure 10-22). In this scenario, the ATM device has no knowledge that its remote destination is attached to a frame relay network. All necessary protocol conversions are handled by the interworking function, which is a service provided by the transport network. Hence the term "Service Interworking."

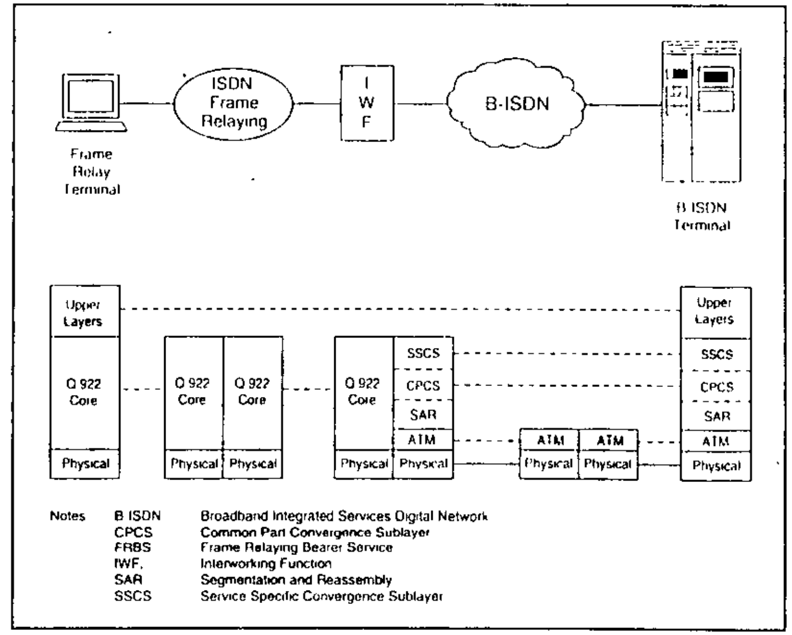


Figure 10-22. Service Interworking between Frame Relaying Bearer Service and B-ISDN.

Source ITU-T I.555

For more information on frame relay/ATM interworking, refer to the Implementation Agreements and ITU-T Recommendation I.555.

10.13.2 ATM/SMDS Interworking

Bellcore defines SMDS as a connectionless data-transport service. It is provided by a number of LECs and IXCs. The company has documented the design requirements for a broadband switching system (BSS) that would support ATM, frame relay, and SMDS services in GR-1110-CORE [10-24]. Bellcore has proposed two scenarios for SMDS and ATM interworking (see Figure 10-23). The first (or top) scenario shows an SMDS user accessing the BSS via an SMDS SNI. At the user's end, the SMDS SIP stack is used and protocol conversions occur in the BSS. The second (or lower) scenario shows the SMDS user accessing the network via the ATM UNI. Here, the SMDS CPE uses different protocols: the SIP Connectionless Protocol (SIP_CLS) and AAL3/4. The functions of SIP Level 3 that are not part of AAL3/4 are placed in SIP_CLS. So, the SMDS user receives the equivalent of SIP Level 3.

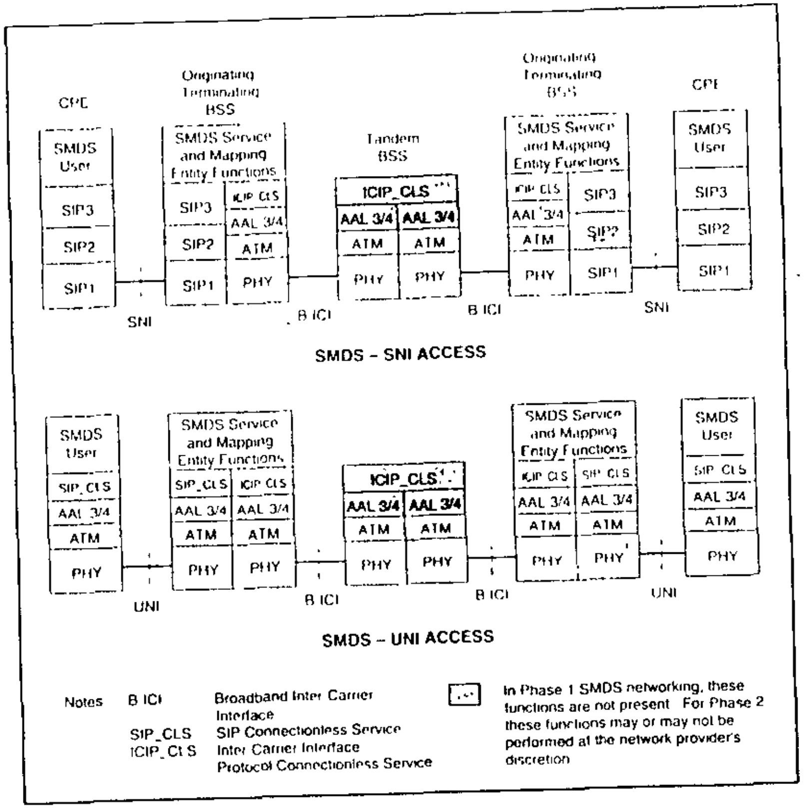


Figure 10-23. ATM/SMDS interworking.

Source GR-1110-CORE, ©1994, Bell Communications Research, Inc., reprinted with permission

10.14 The Interim Local Management Interface

As discussed previously, the B-ISDN architecture defines three planes: user plane, control plane, and management plane. The ATM Forum has developed the Interim Local Management Interface (ILMI) to address the management plane functions.

The ILMI assumes that each ATM device supports at least one UNI and has a UNI Management Entity (UME) for each UNI. The UMEs then communicate network management information (see Figure 10-24). The SNMP/AAL performs the ILMI communication. At the ATM layer, one VCC provides ILMI communication. The default value for this VCC is VPI = 0, VCI = 16.

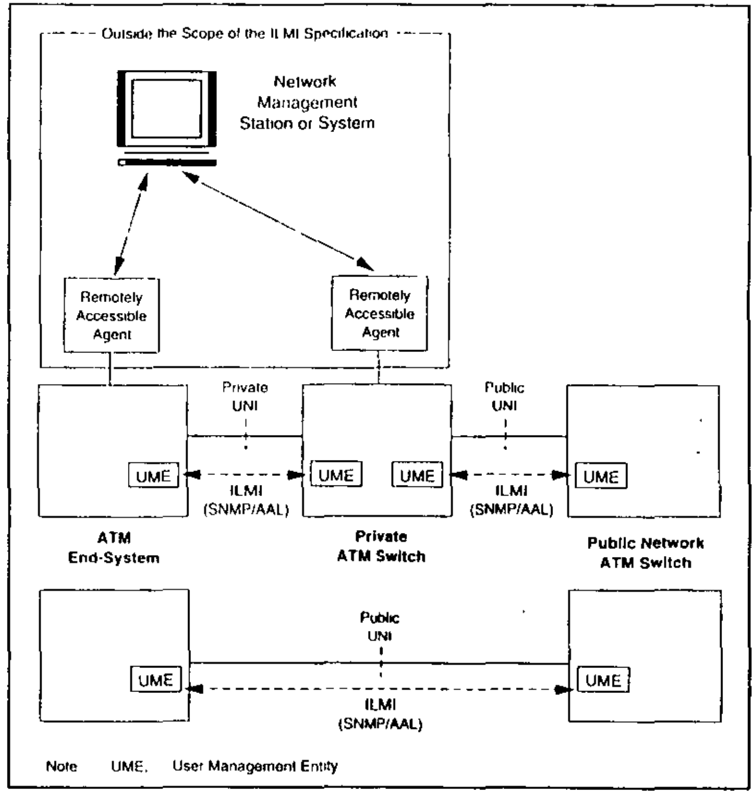


Figure 10-24. Definition and context of ILMI.

Courtesy of the ATM Forum

The management information defined by the ILMI provides status and configuration information from the UME regarding its UNI. This information details the status and configuration of both the ATM and Physical layers at that UNI. This information is organized into a MIB that contains several groups of managed objects.

- Physical layer
- ATM layer
- ATM layer statistics
- VPCs
- VCCs
- address registration information

Within these groups managed objects may pertain to the system as a whole, a physical interface, an ATM layer interface, a virtual path, or a virtual channel. Examples of objects defined in the ATM UNI MIB include:

- transmission type (SONET STS-3c, DS3, and so on)
- media type (coax, single-mode fiber, and so on)
- operational status (in-service, out-of-service, and loop-back)
- maximum number of VCCs
- UNI port type (public or private)
- ATM cells received
- ATM cells dropped
- ATM cells transmitted
- Transmit QOS class
- VPI/VCI value

For further details on ILMI and the ATM UNI MIB, consult the UNI 3.1 specification published by the ATM Forum.

10.15 ATM Customer Network Management

Bellcore has defined a CNM Service for use with Exchange PVC Cell Relay Service (CRS) in document GR-1117-CORE [10-63]. The CRS CNM service provides LEC customers with the ability to manage their access to CRS and ATM UNIs. The LEC provides an SNMP agent within the ATM network, which is accessible by a customer-provided network management station (see Figure 10-25). Because SNMP is used as the communication protocol, CNM may be integrated with other SNMP-based network management platforms, such as the ATM Forum's ILMI. See References [10-64] and [10-65] for other details on CNM.

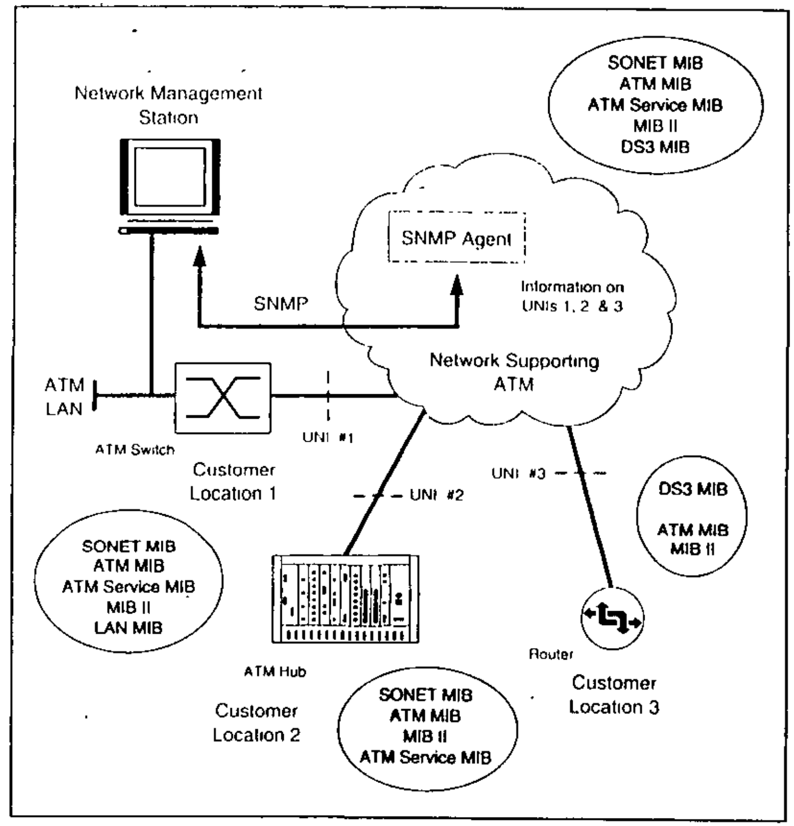


Figure 10-25. ATM CNM agent role (SNMP example).

Source Brown and Kostick, "And CNM for All: Customer Network Management Services for Broadband Data Services." Proceedings of the 18th Annual Conference on Local Computer Networks, © 1993 IEEE.

This concludes our discussion of ATM architecture. Chapter 11 looks at the ATM protocols in detail. Readers interested in ATM applications should consult References [10-66] through [10-69].

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11 ATM Protocols

Because current ATM standards and implementations focus on the operation of the ATM Forum's UNI, this chapter focuses on the UNI protocols and the upper-layer services that use them. It also discusses ATM network management and interworking.

11.1 ATM Protocols and Network Architecture

To begin the study of the ATM protocols, compare the ATM protocols with the OSI Reference Model. Figure 11-1 illustrates that there is an approximate relationship between the ATM layers (PHY, ATM, and AAL) and the OSI Physical and Data Link layers. Note that ATM-specific signaling and upper-layer functions, which may be present for some network configurations, are not shown in the figure.

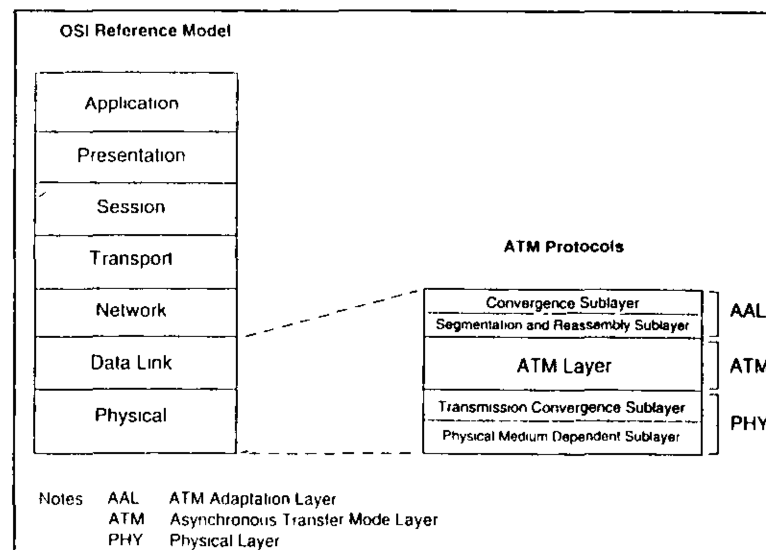

Figure 11-1. Comparing OSI and ATM architectural models.

The ATM network architecture includes the CPE, BSSes, and the interfaces between them (see Figure 11-2). The CPE includes the User layer, which supplies the information to be transmitted, and the three ATM layers (AAL, ATM, and PHY). Examples of User layers include constant bit rate applications, such as DS1, that use AAL1, connectionless services, such as SMDS, that use AAL3/4, and connection-oriented protocol traffic, such as TCP/IP, that uses AAL5 [11-1].

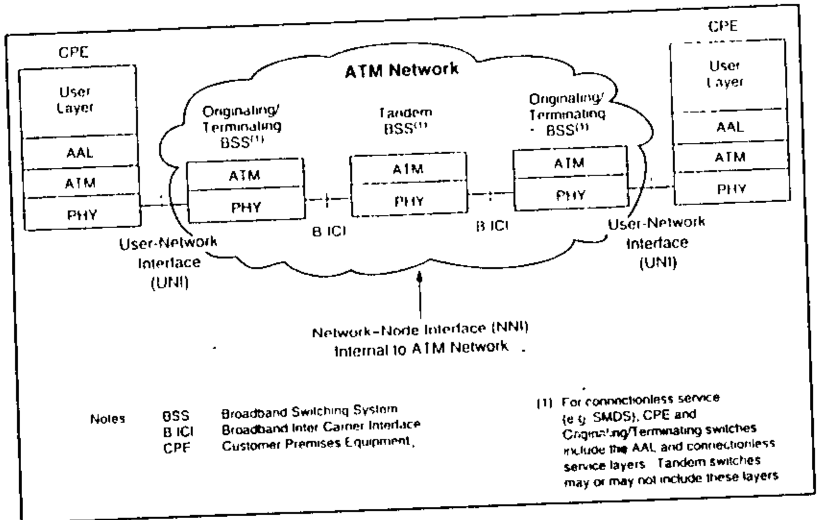

Figure 11-2. ATM network architecture.

Source GR-1110-CORE, ©1996, Bell Communications Research, Inc., reprinted with permission.

The CPE uses the UNI to connect to the ATM network [11-2]. The CPE requires no knowledge of the ATM network's internal architecture and operation. Bellcore views the internal ATM network as interconnected BSSes. An originating/terminating BSS connects to the network side of the UNI, and tandem BSSes provide intermediate switching. The BSS functional layers depend on the transmitted applications.

Bellcore has developed a protocol model of the UNI that illustrates the protocols the BSS will support (see Figure 11-3). It has two categories of UNI protocols: core functions that include the PHY and ATM layers and service-specific functions at the AAL and upper layers.

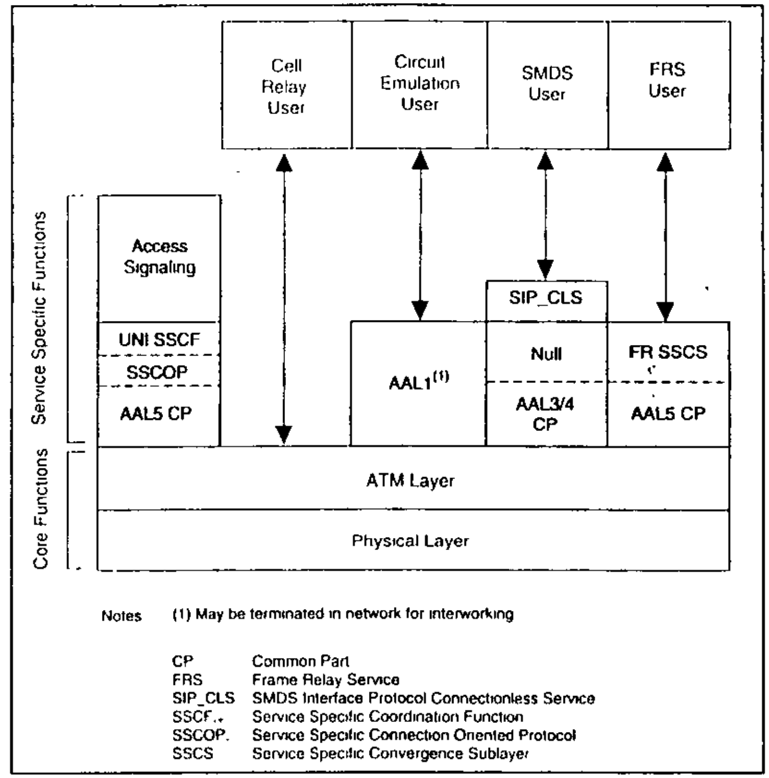

Figure 11-3. Protocol model of the UNI.

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Access signaling is the exchange of call-control messages that set up, maintain, and disconnect virtual channel connections between end users. ITU-T Q.2931 (formerly Q.93B) defines the signaling messages. CRS user information is transmitted directly over the ATM Layer.

Circuit-emulation service, which can carry DS1 traffic, is transmitted over AAL1. The SMDS Interface Protocol Connectionless Service (SIP_CLS) and AAL3/4 transmit SMDS traffic. To the end user, the combination of SIP_CLS and AAL3/4 is equivalent to SIP level 3 service.

Frame relay traffic is transmitted via a frame relay service-specific convergence sublayer (FR-SSCS) over AAL5. These functions are implemented within the device that provides the FR/ATM interworking function.

11.2 ATM Layer Protocols

Each ATM cell is 53 octets long and consists of a 5-octet header and a 48-octet payload. There are two header formats: one at the UNI and the other at the NNI [11-3].

11.2.1 The User-Network Interface

The ATM header at the UNI consists of six fields (see Figure 11-4a):

- *Generic Flow Control (GFC)*, a four-bit field that can provide local functions, such as flow control. This field has local, not end-to-end, significance and is overwritten by intermediate ATM switches. The UNI 3.1 specification provides details regarding the operation of this field.
- *Virtual Path Indicator (VPI)*, an eight-bit field that identifies the virtual path across the interface.
- *Virtual Channel Indicator (VCI)*, a 16-bit field that identifies the virtual channel across the interface. The UNI 3.1 specification defines some VPI/VCI values for specific functions, such as meta-signaling, used to establish the signaling channel; point-to-point signaling; and Operations Administration and Maintenance (OAM) cells.

  Examples of pre-assigned VPI/VCI values are:

Function	VPI	VCI
Unassigned and Idle	0	0
Meta-signaling	0	1
F4 flow (Segment Data)	0	3
F4 flow (End-to-End Data)	0	4
Signaling	0	5
SMDS	0	15
ILMI	0	16

- *Payload Type (PT)*, a three-bit field that identifies the type of information contained in the payload. The field has eight defined values:

PT	Interpretation
000	User data, no congestion, SDU type = 0
001	User data, no congestion, SDU type = 1
010	User data, congestion, SDU type = 0
011	User data, congestion, SDU type = 1
100	OAM segment data, F5, flow related
101	OAM end-to-end data, F5, flow related
110	Reserved, future traffic control and resource management
111	Reserved, future functions

- *Cell Loss Priority (CLP)*, a single bit field that the user or network uses to indicate the cell's explicit loss priority. A cell with a CLP = 1 enters the network, and may be discarded under certain network traffic conditions.
- *Header Error Control (HEC)*, an eight-bit field that detects and/or corrects bit errors occurring in the header.

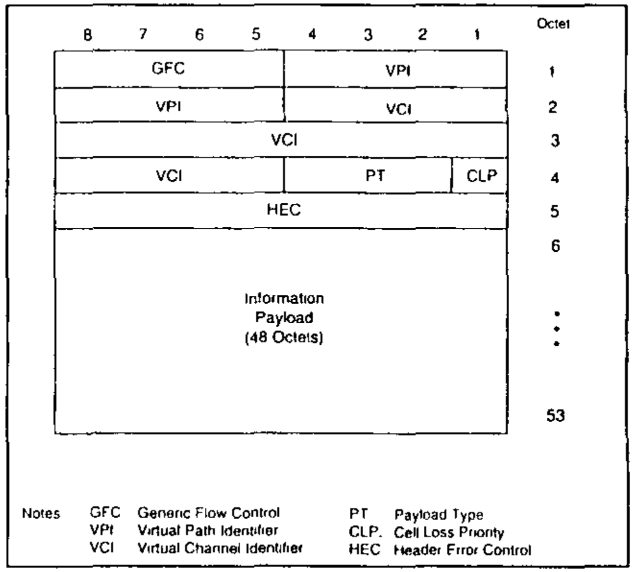


Figure 11-4a. ATM cell format (UNI).

Source: GR-1113-CORE, ©1994, Bell Communications Research, Inc., reprinted with permission

11.2.2 The Network-Node Interface

The ATM header at the NNI is 5 octets long, with a format almost identical to the UNI format except for the first octet (see Figure 11-4b). The NNI, which provides bundles of VCIs between switches, defines an additional 4 bits for the VPI. In other words, the NNI uses 12 bits for the VPI and 16 for the VCI. The UNI uses 8 bits for the VPI and 16 bits for the VCI.

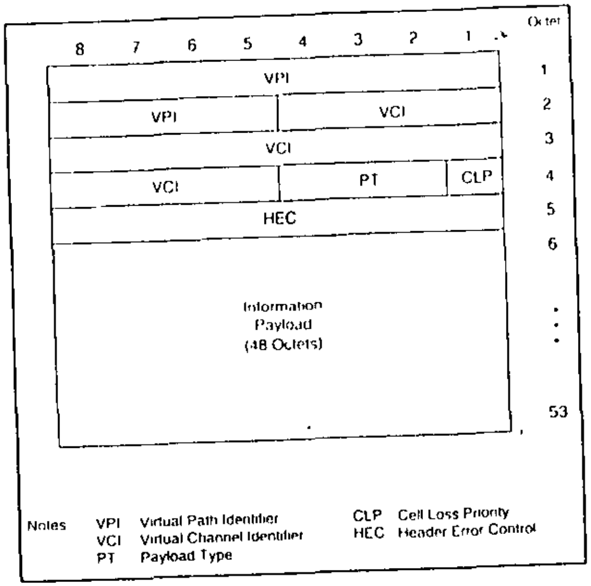


Figure 11-4b. ATM cell format (NNI).

Source GR-1113-CORE, ©1994, Bell Communications Research, Inc., reprinted with permission

11.2.3 Unassigned Cells

At most interfaces, when there are no user-generated (or assigned) cells to send, filler cells, also called *unassigned cells*, are sent to occupy the available bandwidth. These cells have the reserved VPI/VCI value of 0/0 and a fixed-payload pattern. Unassigned cells are generated and discarded at the ATM layer and can be replaced by assigned cells as necessary, such as cell-multiplexing functions.

It is worth mentioning that another type of cell also has a VPI/PCI value of 0/0—the Physical layer cell. When the VPI/VCI value is 0/0, the 4-bit field normally used for the PT and CLP fields is reinterpreted as follows.

Interpretations	Value	
Unassigned	0000	ATM Layer Cell
Idle	0001	Physical Layer Cell
F1 flow (CBPL)	0011	Physical Layer Cell
F3 flow (CBPL)	1001	Physical Layer Cell

Idle cells are used for rate adaptation (cell stuffing) at the Physical layer. Unlike unassigned cells, they take precedence over ATM layer cells and, therefore, cannot be replaced by assigned cells. In North America, idle cells are not normally used; rather, the ITU-T has defined a cell-based Physical layer (CBPL) in I.432 that comprises only cells without a framing structure, such as SONET, in which to transport them. Because of the absence of a framing structure, the F1 and F3 OAM flows have to be carried in Physical layer OAM cells that have a VPI/VCI of 0/0 and PT/CLP field values as shown above. Note that the F2 flow is redundant.

11.2.4 ATM Operations and Maintenance

The ATM network's ongoing performance is key to the success of a broadband implementation. To support good performance, the ITU-T developed Recommendation I.610 [11-4] to define the OAM functions of the Physical and ATM layers and the VP and VC connections. These functions are divided into five phases:

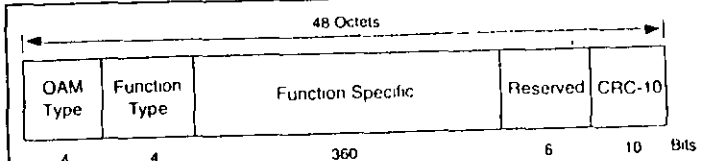
- *performance monitoring*, continuous or periodic checks
- *defect and failure detection*, malfunction detection and alarms
- *system protection*, bypassing a failed component to restore the system
- *failure or performance information*, alarms and reports
- *fault localization*, testing to determine the failed component

OAM functions operate on five levels within the Physical and ATM layers. These functions are called OAM flows, designated F1 through F5. The Physical layer contains three OAM levels: the regenerator section level (F1), the digital section level (F2), and the transmission path level (F3). The ATM layer contains two OAM levels: the virtual path level (F4) and the virtual channel level (F5).

OAM operation is addressed in Bellcore's document GR-1113-CORE [11-5], the ATM Forum's UNI 3.1 specification, ITU-T Recommendation I.610, ANSI Technical Report T1S1.5/92-029R3, and in Stephen Farkouh's paper about managing ATM networks [11-6]. OAM cells are sent on pre-assigned VCIs. For F4 (virtual path) OAM flows, VCI = 3 identifies a segment OAM flow, while VCI = 4 identifies an end-to-end OAM flow. For F5 (virtual channel) OAM flows, the OAM cell is sent with the same VPI/VCI values as the user data; however, the Payload Type (PT) value within the cell header identifies the type of OAM connection as either segment or end-to-end.

Figure 11-4c shows the ATM Layer Management PDU (or OAM cell). The cell payload consists of five fields:

- *OAM Type*, identifies the type of OAM communication (fault management, performance management, or activation/deactivation)
- *Function Type*, defines the function performed by this cell
- *Function specific field*, the detailed OAM cell contents
- *Reserved*, unused bits
- *Error Detection Code*, CRC-10



OAM Type	Value	Function Type	Value
Fault Management	0001	Alarm Indication Signal (AIS)	0000
		Remote Defect Indication (RDI)	0001
		Continuity Check	0100
		Loopback	1000
Performance Management	0010	Forward Monitoring	0000
		Backward Reporting	0001
		Monitoring and Reporting	0010
Activation/Deactivation	1000	Performance Monitoring	0000
		Continuity Check	0001
System Management	1111	Not defined (equipment dependent)	

Figure 11-4c. ATM Layer Management PDU format.

Source: GR-1113-CORE, ©1994, Bell Communications Research, Inc., reprinted with permission

Four function-specific fields perform fault management, performance management, activation/deactivation, and system management functions. Each of these OAM cells contains additional fields to support these functions, which are detailed in ANSI T1S1.5/92-029R3. The fault-management OAM cell performs Alarm Indication Signal (AIS) or Remote Defect Indication (RDI) reporting, continuity checking, OAM cell-loopback testing, and cell-transfer delay measurements. The performance management OAM cell may monitor incoming ATM cell traffic (referred to as inward monitoring) or report on outgoing cell traffic at the distant end (referred to as outward reporting). The activation/deactivation OAM cell enables or disables the performance or continuity check functions. The system management OAM cell is defined by the specific implementation.

11.3 ATM Physical Layer

The ATM Forum defines several options for the Physical layer interface at either public or private UNIs. This section reviews seven of the most commonly used interfaces for ATM implementation in North America, ranging in speed from 1.544 Mbps (the DS1 rate) to 622.080 Mbps (the OC-12 rate). Many of these interfaces are documented in the UNI 3.1 publication; others are documented separately and are available from the ATM Forum's Web and FTP sites (see Reference [10-23]). In addition, Bellcore has described many of these interfaces in their Technical Reference TR-NWT-001112 [11-3].

11.3.1 DS1 Interface

Many of the early ATM definitions assumed that very high bandwidth channels would be required for access to B-ISDN networks. Unfortunately, this assumption did not account for the very large installed base of existing DS1 lines, which have been used for many years to transport voice, data, and LAN traffic. As a result, the ATM Forum developed a public UNI based on the DS1 interface, which is documented in Reference [11-7] and also by Bellcore in Reference [11-3].

The DS1 interface for ATM applies to the public UNI only, and operates over clear channel (or transparent) T1 facilities. The line rate is 1.544 Mbps, and the throughput available for user information cells, signaling cells, and OAM cells is 1.536 Mbps (Figure 11-5). The signal format is based on the ANSI T1.403 standard, which defines the Extended Superframe (ESF) framing format. The physical interface (plugs and jacks), also defined in ANSI T1.403, is used for consistency with other DS1-based systems. Maintenance functions for performance monitoring, failure detection, and alarms, as specified in ANSI T1.408, are employed for the ATM applications.

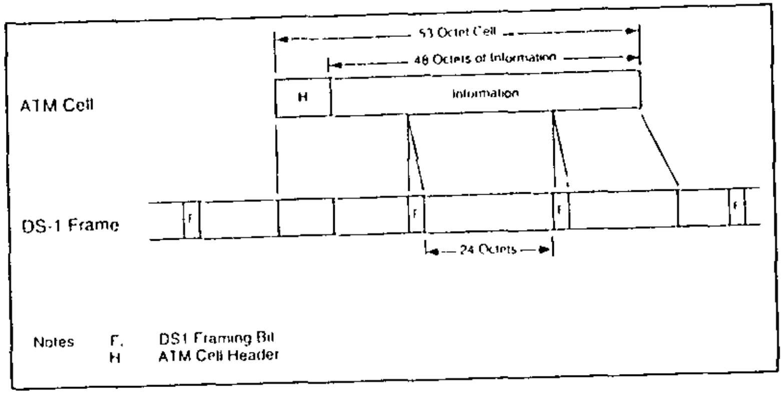


Figure 11-5. Direct ATM cell mapping on the DS1 frame.

Source TR-NWT-001112, ©1993, Bell Communications Research, Inc., reprinted with permission

11.3.2 25.6 Mbps Interface

For many ATM LAN applications, neither the DS1 interface (at 1.544 Mbps) nor the DS3 interface (at 44.736 Mbps) provide an appropriate amount of bandwidth. To address that need, the ATM Forum developed a private UNI, operating at 25.6 Mbps [11-8].

The 25.6 Mbps UNI operates over copper transmission facilities, with impedances of 100 ohms (for unshielded twisted pairs), of 150 ohms (for shielded twisted pairs), or for other copper facilities having an impedance of 120 ohms. The line rate is 32 Mbaud with a 4B5B encoding scheme, yielding the 25.6 Mbps data throughput. Note from Figure 11-6 that the transmitter and receiver implement both the Transmission Convergence (TC) and Physical Medium Dependent (PMD) layers, transmitting and receiving a stream of cells from the user application. The transmission media specified for use by the 25.6 Mbps UNI employ commonly used connectors, such as the eight-pin modular connector (formally specified by the IEC 603-7 and ISO 8877 standards), which is generally referred to as the RJ-45 connector and is frequently used for both Ethernet and token ring LANs. As a result, it is expected that the 25.6 Mbps UNI will achieve widespread acceptance within local ATM networks.

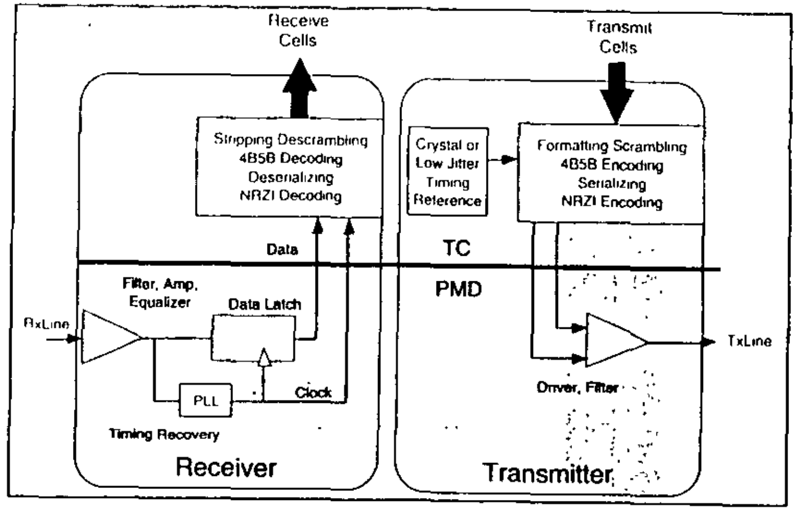


Figure 11-6. TC/PMD components and transmitter timing for the 25.6 Mbps physical interface.

Source AF-PHY-0040.000, © 1995 The ATM Forum

11.3.3 DS3 Interface

Chapter 8 discussed the DS3 PLCP format used with SMDS. The ATM PLCP is based on that work. The DS3 PLCP frame carries 12 ATM cells plus overhead information, which is transmitted every 125 microseconds (see Figure 11-7). The overhead functions consist of the following:

- *A1* and *A2*, framing
- *B1*, bit interleaved parity
- *C1*, cycle/stuff counter
- *G1*, PLCP path status
- *P0* to *P11*, path overhead identifier
- *Z1* to *Z6*, growth octets

For details, see Bellcore document TR-NWT-001112 and the ATM Forum UNI 3.1 specification.

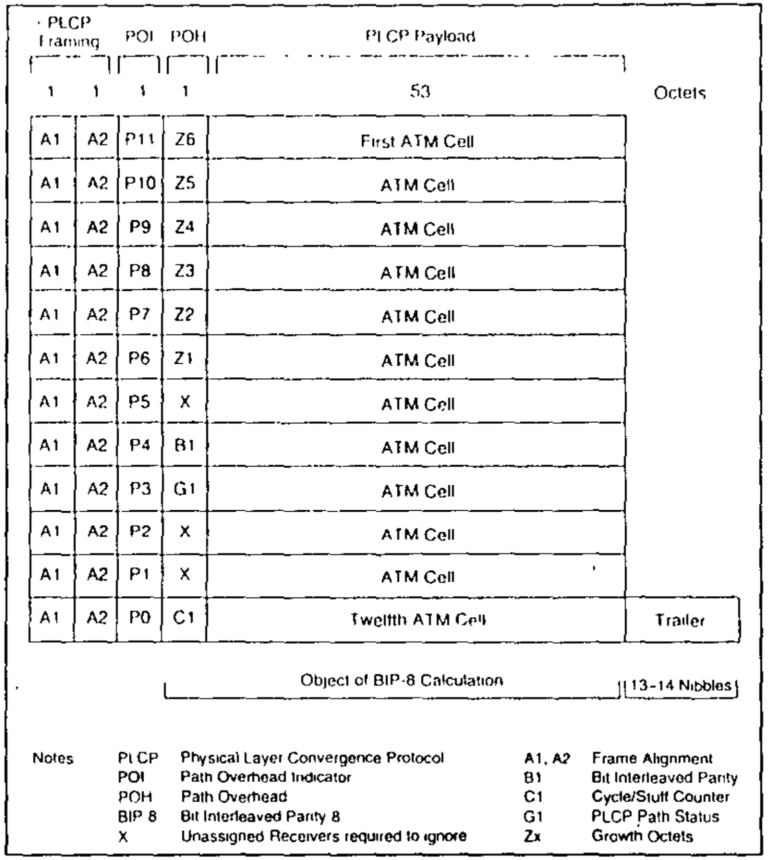


Figure 11-7. DS3 PLCP frame (125μs).

Source TR-NWT-001112, ©1993 Bell Communications Research, Inc., reprinted with permission

11.3.4 STS-1 Interface

A mid-range interface, operating at the STS-1 rate of 51.840 Mbps, is also defined for ATM cell transport. Bellcore has defined a UNI which operates over two single-mode fiber links, and the ATM Forum has defined a private UNI which operates over Category 3 UTP cable using 8-pin modular connectors [11-9]. The frame structure is illustrated in Figure 11-8 and is part of the Synchronous Digital Hierarchy, which will be examined in detail in a following section. For details on the SONET STS-1 signal, see ANSI T1.105.

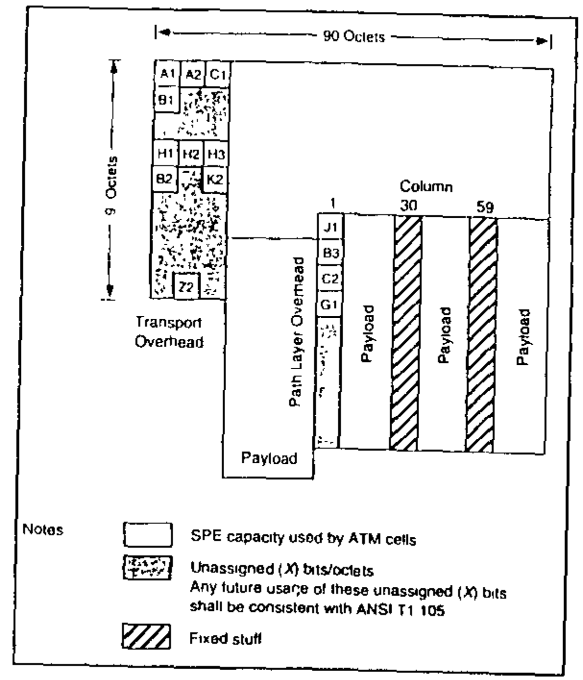


Figure 11-8. Logical frame structure of the 51.840 Mbps UNI.

Source TR-NWT-001112, ©1993, Bell Communications Research, Inc., reprinted with permission

11.3.5 100 Mbps Multimode Fiber Interface

Much current LAN technology is based on previous work defined to support the Fiber Distributed Data Interface (FDDI) standard. The ATM Forum, in the UNI 3.1 document, defines a private UNI operating over multimode fiber. The Physical layer follows the FDDI Physical Medium Dependent (PMD) specification, with a link operating at 125 Mbaud with a 4B5B line coding scheme over 62.5 micron cable. Note that the link operates for cell transport only; therefore, no Physical layer framing structure is required or defined.

11.3.6 SONET STS-3c Interface

The SONET formats are part of the Synchronous Digital Hierarchy (SDH). The DS3 format is part of the Plesiochronous Digital Hierarchy (PDH). Transmission rates for various signals within these two hierarchies are shown below.

SONET/Synchronous Digital Hierarchy (SDH) Rates

Rate (Mbps)	SONET	SDH
51.840	STS-1	
155 520	STS-3	STM-1
466.560	STS-9	STM-3
622 080	STS-12	STM-4
933 120	STS-18	STM-6
1244 160	STS-24	STM-8
2488 370	STS-48	STM-16

Plesiochronous Digital Hierarchy (PDH) Rates

Rate (Mbps)	North America	Europe
0 064	DS0	
1 544	DS1	
2 048		E1
3 152	DS1C	
6 312	DS2	
8 448		E2
34 368		E3
44 736	DS3	
139 264		E4
274 176	DS4	

The Synchronous Transport Signal-Level 1 (STS-1) frame, transmitted at 51.840 Mbps, is the basis for the higher SONET transmission rates. The STS-1 frame consists of nine rows and 90 columns, for a total of 810 octets. Each frame is transmitted within a period of 125 microseconds, occupying a total bandwidth of 51.840 Mbps (810 octets/frame * 1 frame/125 microseconds * 8 bits/octet). Of the 90 data columns, three (or 27 octets) carry transport overhead (TOH), such as framing, error monitoring, management, and payload pointer information. So the STS-1 payload envelope (SPE) is 87 columns wide or 783 octets (87 * 9). The SPE is itself a kind of frame that floats inside the available space in the STS-1 frame. The first column of the 87 column SPE is occupied by an 11-octet transmission path overhead (POH), and the start of the SPE is identified by the payload pointer in the TOH. Note that this mechanism exists to let the information carried in the SPE be timed with a bit clock signal that is not locked to the STS-1 clock signal; pointer movements let the SPE drift toward the STS-1 frame. Because of the POH, the available capacity per STS-1 frame is further reduced to 774 octets (86 * 9), representing a user payload bandwidth of 49.54 Mbps.

To provide SONET at higher rates (above 51.840 Mbps), multiple STS-1 frames are byte (or octet) interleaved. For example, three STS-1 frames may be combined into one STS-3 frame operating at 155.520 Mbps (the three frames must be frame aligned). The STS-3 frame now has 270 columns and nine rows and a TOH of nine columns. The payload of the STS-3 frame now has 261 columns and nine rows and comprises the three STS-1 SPEs (each 87 columns wide) byte interleaved. Note that these SPEs are independent of each other and can have any phase relationship with each other and the STS-3 frame, as determined by the three individual payload pointers that are byte interleaved in the nine-column TOH. The purpose of this multiplexing exercise is to enable the efficient transportation of three independent STS-1 signals long distance over a single optical fiber. Upon arriving at the far end, the STS-1 signals are demultiplexed to become three STS-1 framed signals again.

A special variant of the STS-3 frame is the STS-3c frame (c is for concatenation), in which the three SPEs of an STS-3 frame are merged (concatenated) together to form one large SPE of 261 columns. Because there is only one SPE, there only needs to be one POH, leaving a payload capacity of 260 columns that represents a usable bandwidth (for cells) of 149.76 Mbps (see Figure 11-9). The same approach can be used to generate an STS-12c signal in which the usable payload capacity is 591.04 Mbps (note that three columns of the STS-12c SPE are not used). When SONET is specified as the UNI, STS-3c is the signal used. SONET is a North American standard. Outside North America, most countries use the SDH equivalent to STS-3c, namely STM-1.

You should also note that when STS-1, STS-3c, or STS-12c are sent through optical fiber, the signals become OC-1, OC-3c, or OC-12c, respectively.

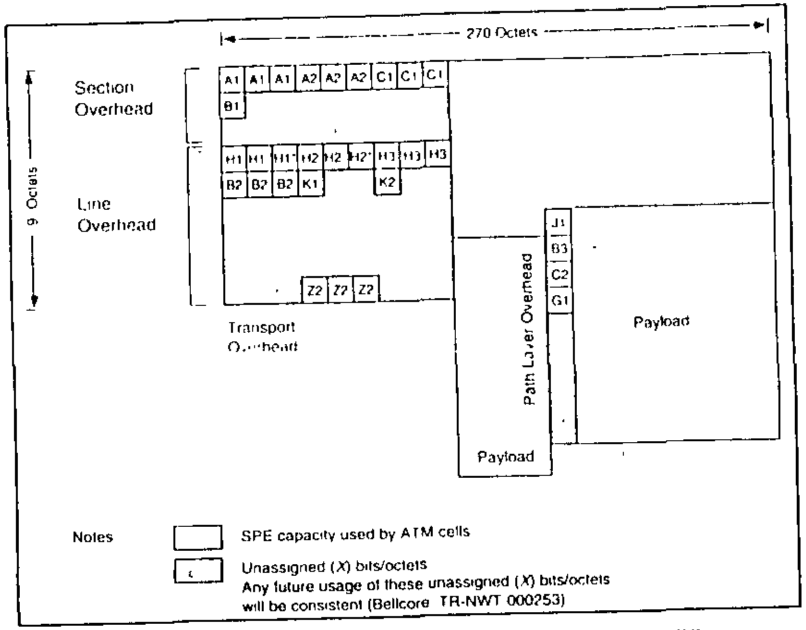


Figure 11-9. Logical frame structure of the 155.520 Mbps UNI.

Source TR-NWT-001112, ©1993, Bell Communications Research, Inc., reprinted with permission

Within the transport overhead, rows one through three contain the SOH, and rows four through nine contain the Line Overhead (LOH). The three original STS-1 TOHs are byte interleaved to create the nine rows of the STS-3c TOH. So, we see A1, A1, A1, A2, A2, A2, C1, C1, C1, and so on in the STS-3c TOH. Notice that overhead octets A1, A2, C1, H1, H2, H3, and B2 are replicated, while octets B1, K2, and Z2 are not. The path overhead (POH) carried in STS-3c SWP includes J1, B3, C2, and G1.

Bellcore's TR-NWT-000253 [11-10] is the baseline reference on SONET. The ATM Forum's UNI 3.1 specification defines the overhead octets and their functions for STS-3c implementations with ATM, as well as an ATM Forum specification for a 155 Mbps private UNI over twisted pair (100 ohm category 5 unshielded, 120 ohms, or 150 ohm shielded) cable [11-11].

Section Overhead

Octet Label	Overhead Function
A1, A2	Frame alignment
C1	STS-1 identification
B1	Section error monitoring

Line Overhead

Octet Label	Overhead Function
B2	Line error monitoring
H1 (bits 1 to 4)	New data flag, Path AIS
H1 and H2 (bits 7 to 16)	Pointer value, Path AIS
H1*, H2*	Concatenation indication, Path AIS
H3	Pointer action, Path AIS
K2 (bits 6 to 8)	Line AIS, Line FERF, removal of Line FERF
3rd Z2	Line FEBE

Path Overhead

Octet Label	Overhead Function
J1	STS path trace
B3	Path error monitoring
C2	Path signal level indicator
G1 (bits 1 to 4)	Path FEBE
G1 (bit 5)	Path RDI (yellow)

AIS: Alarm Indication Signal
FEBE: Far End Block Error
FERF: Far End Receive Failure
RDI: Remote Defect Indicator
STS: Synchronous Transport Signal

ANSI T1.105 [11-12] is an excellent reference about SONET, and References [11-13] and [11-14] provide additional information on SONET implementations for ATM.

11.3.7 OC-12 Interface

The ITU-T, Bellcore, and the ATM Forum [11-15] have defined a UNI operating at the OC-12 rate of 622.080 Mbps. For the ATM Forum's case, a public UNI, a private UNI, and a private NNI are all defined. The transmission consists of two fiber optic links in a point-to-point configuration. Bellcore specifies a single-mode fiber link, while the ATM Forum allows either single-mode or multimode fiber options. The frame structure is illustrated in Figure 11-10 and is based on the SDH hierarchy discussed in the previous section.

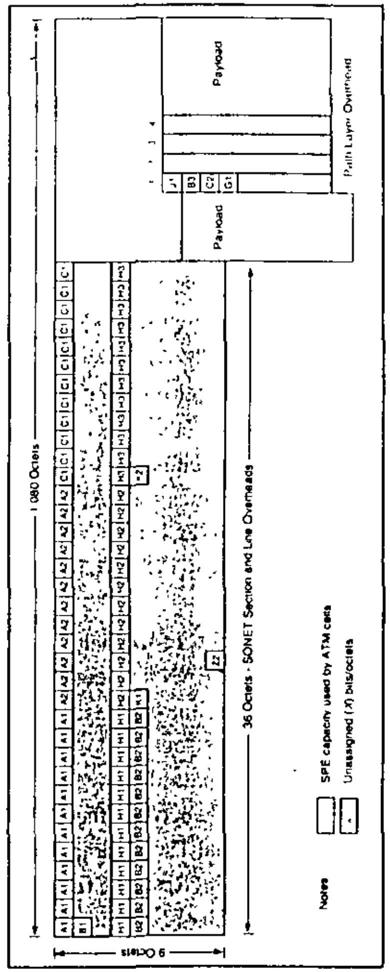


Figure 11-10. Logical frame structure of the 622.080 Mbps UNI.

Source: TR-NWT-001112, ©1993, Bell Communications Research, Inc., reprinted with permission

11.4 ATM Adaptation Layer Services

Recall from Section 10.6.3 and Figure 10-11 that ATM networks may support different types of data traffic. Each traffic type may use a different AAL protocol, as defined in I.363 [11-16]. The following sections explore each of these protocols individually.

11.4.1 AAL Type 1

AAL Type 1 supports Class A traffic, which is sent at a constant bit rate (CBR), is connection oriented and has a timing relationship between the source and the destination. Examples of such traffic include pulse code modulation (PCM), encoded voice, or CBR video.

The AAL consists of at least two sublayers: the Convergence Sublayer (CS) and the SAR Sublayer. For AAL1, the CS takes the User Information, provided at a CBR, and divides it into 47-octet protocol data units (AAL1_CS_PDUs), as shown in Figure 11-11. This AAL1_CS_PDU becomes the SAR_PDU payload. Note that no AAL1-CS protocol control information (PCI) is added to the AAL1_CS_PDU. The SAR sublayer adds a header 1 octet long to the AAL1_CS_PDU, forming a 48-octet AAL1_SAR_PDU.

The AAL1_SAR_PDU header consists of two fields: Sequence Number (SN) and Sequence Number Protection (SNP). The SN field contains two subfields: a Convergence Sublayer Indicator (CSI), which is one bit and is used by service-specific functions of the AAL1_CS; and a Sequence Count (SC), which is 3 bits and contains a binary encoded sequence counter that is passed between peer AAL1_CS entities. One CSI function is to pass timing information between the sender and the receiver. In this method, called the Synchronous Residual Time Stamp (SRTS), the CSI field of successive AAL1_SAR_PDUs carries a 4-bit Residual Time Stamp (RTS). The SC subfield detects lost or misinserted cells. The SNP field contains two subfields: a CRC Control subfield (three bits) and a single Parity bit. These two fields provide error control for the CSI and CS subfields.

The completed AAL1_SAR_PDU is sent to the ATM layer, where the ATM header is added before transmission.

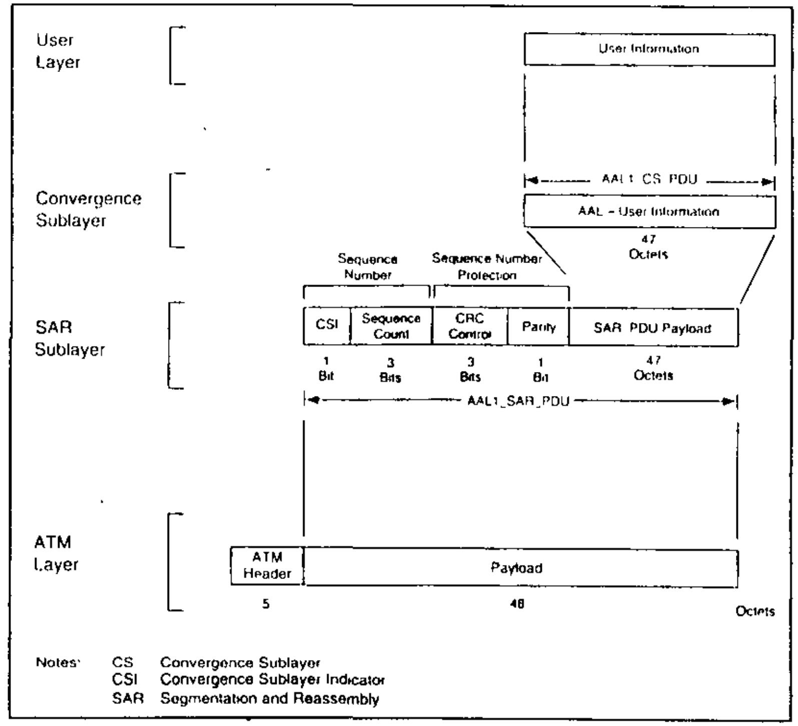


Figure 11-11. AAL Type 1 PDU format.

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11.4.2 AAL Type 2

AAL Type 2 supports Class B traffic, which is sent at a variable bit rate (VBR), is connection oriented, and has a timing relationship between the source and the destination. Examples include VBR voice and video signals.

The ITU-T is still developing AAL2. However, one suggested format for the AAL2_SAR_PDU is shown in Figure 11-12. The SN field is a binary counter that detects lost or misinserted cells. The Information Type (IT) defines one of three message values: Beginning of Message (BOM), Continuation of Message (COM), or End of Message (EOM). The Length Indicator (LI) states how many octets in the SAR_PDU Payload contain data. Finally, a CRC field provides error detection and correction. Error control is especially important for compressed video signals, which could use AAL2. In this case, single bit errors may affect the encoded datastream, which may produce a more severe effect than a single bit error within a cell carrying CBR audio (Class A and AAL1).

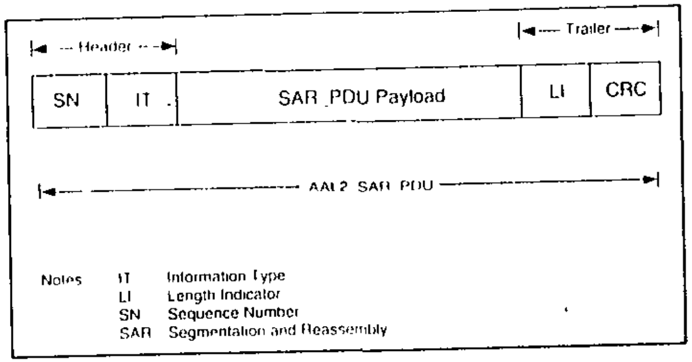


Figure 11-12. AAL Type 2 SAR PDU format.

11.4.3 AAL Type 3/4

AAL Type 3/4 supports Class C or D traffic, which is sent at a VBR with no timing relationship required between the source and destination. At one time, there were two standards: AAL3, which supported connection-oriented traffic, and AAL4, which supported connectionless traffic. These two AALs were later merged into the common AAL3/4. Data traffic that is sensitive to loss but not delay, such as SMDS, would use AAL3/4.

The AAL3/4 Convergence Sublayer is divided into two layers: an SSCS and a CPCS. The SSCS, which may not be present (null), supports the User layer. The AAL3/4 SAR sublayer interacts with the ATM layer.

The CPCS transfers variable-length blocks of data, or AAL3/4_CPCS_SDUs, sequentially between users. Two service modes are defined: Message Mode Service and Streaming Mode Service. Message Mode is used for framed data (see Figure 11-13a). It transfers exactly one Interface Data Unit (AAL3/4_IDU) from the user. This IDU may be fixed or variable in length, up to 65,535 octets long. Streaming Mode is used for one or more IDUs, which may be separated in time (see Figure 11-13b).

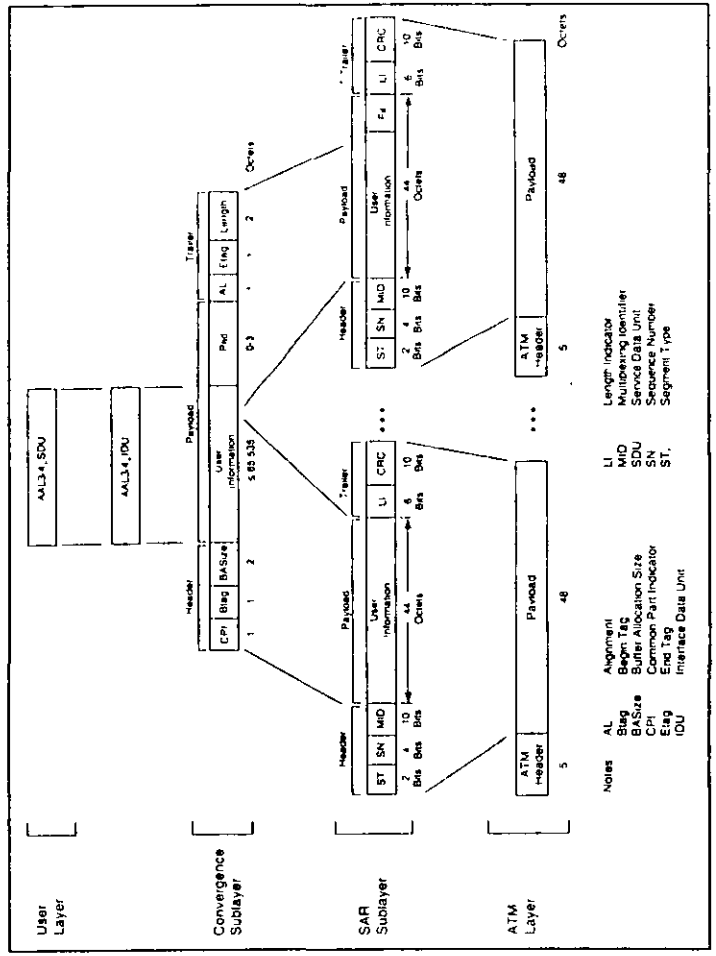


Figure 11-13a. AAL Type 3/4 PDU format (Message Mode Service).

Source: GR-1113-CORE, ©1994, Bell Communications Research, Inc., reprinted with permission.

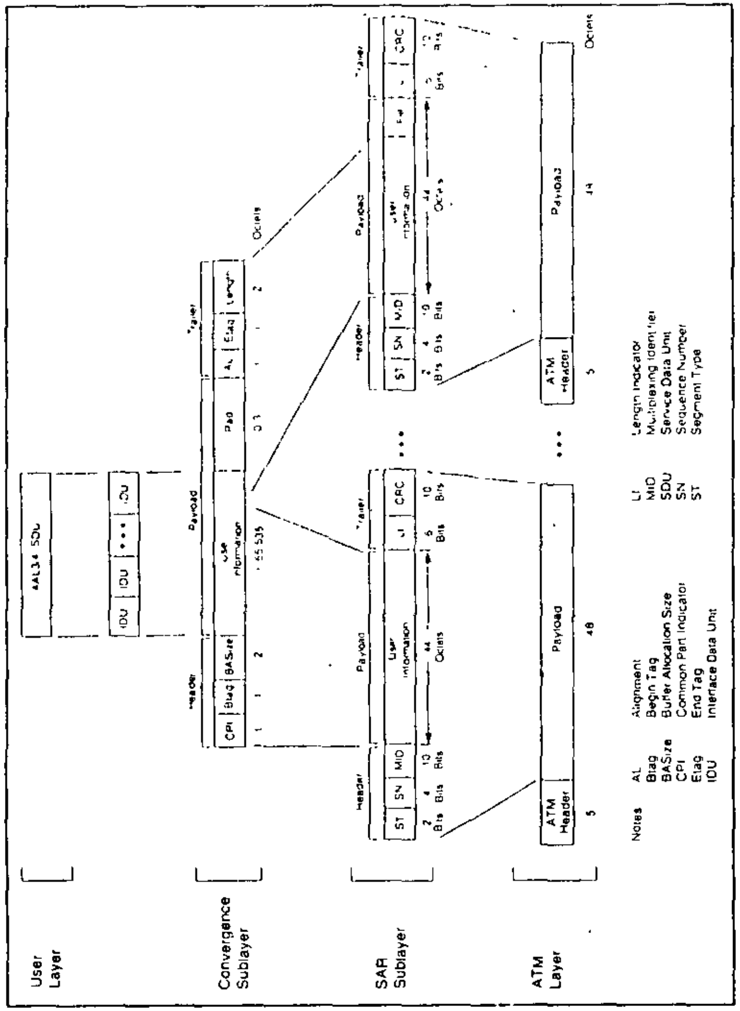


Figure 11-13b. AAL Type 3/4 PDU format (Streaming Mode Service).

Source GR-1113 CORE, ©1994, Bell Communications Research, Inc., reprinted with permission

The AAL3/4 process is initiated when the user information (the AAL3/4_CPCS_SDU) is passed to the CPCS. The CPCS adds a header and trailer,

generating the AAL3/4_CPCS_PDU. This unit is then passed to the SAR sublayer, which segments it into 48-octet SAR PDUs to become the ATM cell payload.

The AAL3/4_CPCS_PDU header contains three fields: Common Part Indicator (CPI), Begin Tag (BTag), and Buffer Allocation Size (BAsize). The CPI field (one octet) identifies the message type and the counting units for the BTag and BAsize fields. This field is currently coded as 00H, which indicates that the counting unit is the octet. The BTag field (one octet) is used in conjunction with the End Tag (ETag) in the trailer to associate the beginning and end of the AAL3/4_CPCS_PDU. The same number is placed in the BTag and ETag fields and is incremented for successive AAL3/4_CPCS_PDUs. The BAsize field (two octets) tells the receiving AAL3/4 process the maximum buffer size it must reserve to reassemble the incoming AAL3/4_CPCS_PDU.

The AAL3/4_CPCS_PDU User Information payload is limited to the maximum value of the BAsize field (65,535) times the counting value contained in the CPI field (typically octets). A Pad field may be placed after the Information field. This Pad may contain zero, one, two, or three octets of filler, which forces the AAL3/4_CPCS_PDU to be 32-bit aligned. Note that for SMDS applications, this field is of zero length because the AAL3/4_CPCS_PDUs are always aligned on 4-octet boundaries.

The AAL3/4_CPCS_PDU trailer contains three fields: Alignment (AL), End Tag (ETag), and Length. The AL field (1 octet) provides 32-bit alignment in the AAL3/4_CPCS_PDU trailer, and is set to 00H. The ETag field (1 octet) is used in conjunction with the BTag, as described above. The Length field (2 octets) indicates the length in counting units of the User Information field.

The SAR process provides the 48-octet payloads carried in the ATM cells. Each AAL3/4_SAR_PDU contains a header (2 octets), User Information (44 octets), and a trailer (2 octets).

The AAL3/4_SAR_PDU header contains three fields: Segment Type (ST), SN, and Multiplexing Identification (MID). The ST field (2 bits) indicates whether the AAL3/4_SAR_PDU is the beginning of a message (BOM, with ST = 10), the continuation of a message (COM, with ST = 00), the end of a message (EOM, with ST = 01), or a single segment message (SSM, with ST = 11). The SN field (4 bits) is a counter that indicates the sequential position of each AAL3/4_SAR_PDU associated with an AAL3/4_CPCS_PDU. The MID (10 bits) identifies the AAL3/4_SAR_PDUs derived from a particular AAL3/4_CPCS_PDU. In other words, several AAL3/4_CPCS_PDUs may be transmitted simultaneously between the same two AAL users. The MID field identifies the AAL3/4_SAR_PDUs from different AAL3/4_CPCS_PDUs, assisting with the interleaving and reassembly process.

The AAL3/4_SAR_PDU payload contains User Information and Fill and is 44 octets long. The User Information field contains up to 44 octets of an AAL3/4_CPCS_PDU. If the User Information field does not contain 44 octets, the Fill field completes it with zeros.

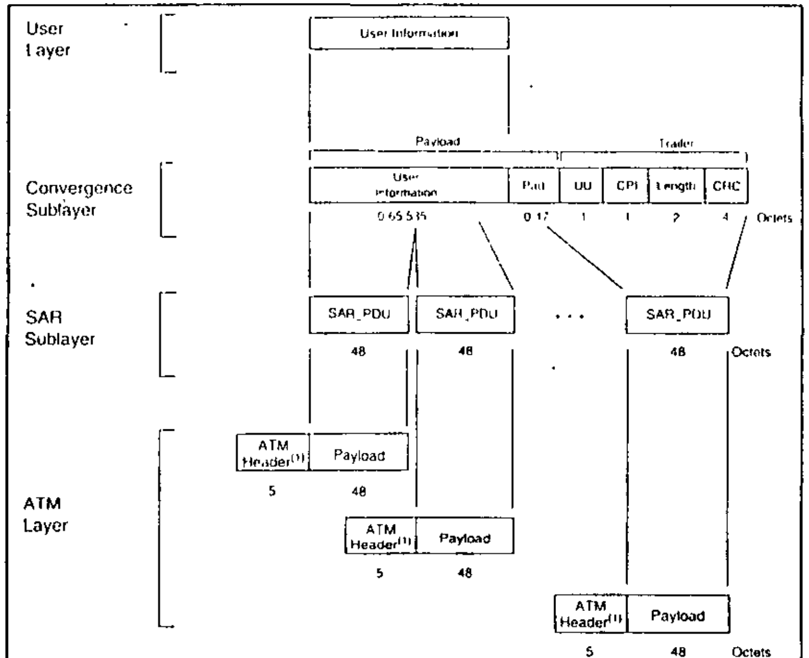
The AAL3/4_SAR_PDU trailer contains two fields, Length Indicator (LI) and a CRC. The LI field (6 bits) contains the length in octets of the User Information field. The type of segment, as indicated in the ST field, restricts the values of the LI field. The BOM and COM segments may be 44 octets long. The EOM segment must be a multiple of four octets (4, 8, 12, 16, . . 44). The SSM must be a multiple of 4 octets and must be at least 8 octets long (8, 12, 16, . . .44).

When the AAL3/4_SAR processing is complete, it hands the AAL3/4_SAR_PDUs to the ATM layer for transmission.

11.4.4 AAL Type 5

AAL Type 5 supports Class C traffic, which is connection-oriented and sent at a VBR with no timing relationship required between the source and the destination. The AAL5 process is considered much simpler than AAL3/4. It removes some of the overhead at the SAR sublayer and supports only Message Mode service. AAL5 is also known as the Simple and Efficient AAL (SEAL).

The User Layer passes User information 0 to 65,535 octets long to the CPCS, as shown in Figure 11-14. The CPCS generates the AAL5_CPCS_PDU, which consists of a payload and a trailer. At the CPCS, a Pad field 0 to 47 octets long is added to the User Information to align the AAL5_CPCS_PDU on a 48-octet boundary.



Notes (1) SDU type subfield of the PT field in the ATM Header detects the beginning and end of the CS_PDU. 0 = first and intermediate cells, 1 = last cell
UU User-to-User
CPI Common Part Indicator

Figure 11-14. AAL Type 5 PDU format.

Source: GR-1113-CORE, ©1994, Bell Communications Research, Inc., reprinted with permission

The AAL5_CPCS_PDU trailer consists of four fields: User-to-User (UU), Common Part Indicator (CPI), Length, and CRC. The UU field (1 octet) contains information to be transferred transparently between AAL5 users. The CPI field (1 octet) aligns the AAL5_CPCS_PDU trailer on a 64-bit boundary. Other uses are under development, and may include identification of layer-management messages. The Length field (2 octets) indicates the length of the AAL5 CPCS payload. The CRC field contains a CRC-32 calculation that detects bit errors in AAL5_CPCS_PDU, including the payload and the first 4 octets of the trailer.

When the AAL5_CPCS_PDU is assembled, the SAR sublayer segments it into 48-octet AAL5_SAR_PDUs, which are then passed to the ATM layer. A single bit in the ATM header PTI field indicates the end of the AAL5_CPCS_PDU. This bit is set to zero for the first and intermediate segments, and to one for the last segment.

11.5 ATM DXI

A number of interfaces and protocols assist with the management of ATM networks (see Figure 11-15a), including the DXI [11-17], the ILMI, and the messages that use these interfaces.

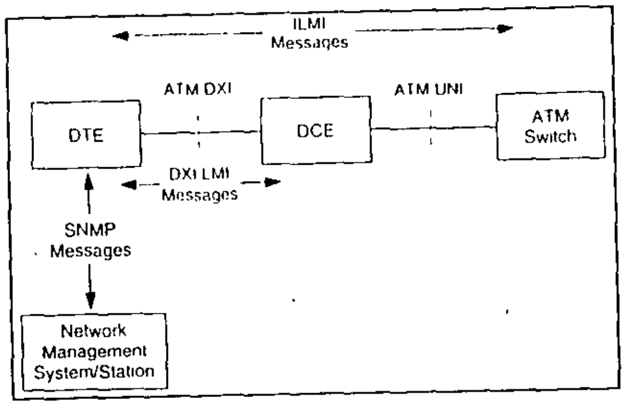


Figure 11-15a. ATM DXI LMI.

Source: AF-DXI-0014 000, © 1993, The ATM Forum

Assume that the DTE is a router connected to an ATM switch via a DCE, such as a DSU. Also connected to the router is a network management console running SNMP. Several interfaces are involved, the ATM UNI between the switch and the DSU; the ATM DXI between the DSU and the DTE (router); and other interfaces, such as a LAN connection, between the router and the network management console.

Several message types may be sent with this configuration. The Network Management System may send SNMP messages to request information contained in the router's MIB. The ATM switch may send ILMI messages to the router requesting

information on a virtual path or virtual channel connection. In some cases, such as when the router requests a count of the dropped received cells, the router must consult the DSU. These consultations are referred to the DSU as DXI LMI messages. Similarly, ATM DXI LMI traps sent from the DSU to the router generate ILMI traps to the switch or SNMP traps to the Network Management Console.

To support these mechanisms, the ATM DXI LMI MIB includes the ATM UNI ILMI MIB, defined in the ATM Forum's UNI 3.1 specification, and the ATM MIB, defined by the Internet Engineering Task Force (IETF).

11.5.1 The DXI Protocol

The DXI protocol provides a way for a DTE, such as a router, and a DCE, such as a DSU, to share the processing of the protocols at the ATM UNI. Three modes of operation are defined:

- *Mode 1a*, AAL5 only
- *Mode 1b*, AAL3/4 for at least one VC, AAL5 for other VCs
- *Mode 2*, AAL5 and AAL3/4, one per VC

The DTE transmits the information field of an AAL_PDU, referred to as a DTE_SDU, to a peer process at another DTE. To do so, the DTE sends the DTE_SDU inside a DXI frame to the DCE. The DCE then completes the processing at the AAL sublayers (CPCS and SAR) and the ATM layers.

The format of the transmission frame sent between DTE and DCE varies depending on the AAL type used. For modes 1a or 1b with AAL5, the DTE sends a DTE_SDU, which may be up to 9,232 octets long, to the DCE in a DXI frame (see upper portion of Figure 11-15b). The DCE receives the DTE_SDU and performs the AAL5 CPCS, AAL5 SAR, and ATM layer functions. The DXI Frame Address (DXA) field maps the VPI/VCI information between the DTE and the DCE.

Mode 1b offers the capabilities of mode 1a plus support for AAL3/4 on individually configurable VCs. The DTE encapsulates an AAL3/4_CPCS_PDU within a DXI frame (see lower portion of Figure 11-15b). In this case, the DTE_SDU is reduced in length to 9,224 octets to accommodate the AAL3/4_CPCS header and trailer. The DCE performs the AAL3/4 SAR and ATM layer functions.

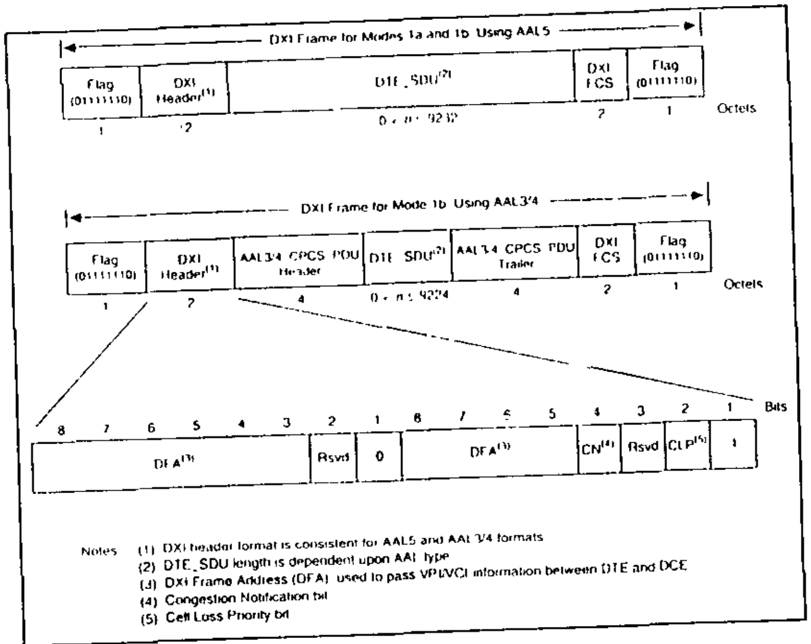


Figure 11-15b. ATM DXI frame formats (modes 1a and 1b).

Source AF-DXI 0014 000, © 1993, The ATM Forum

For mode 2, the DTE encapsulates an AAL3/4_CPCS_PDU within a DXI frame (see Figure 11-15c). In this case, the DTE_SDU may be up to 65,535 octets long. The DCE performs one of the following functions:

- For AAL5 VCs, the DCE removes the AAL3/4 CPCS header and trailer and encapsulates the remainder of the PDU in an AAL5CPCS_PDU The DCE then performs the AAL5 SAR and ATM layer functions.
- For AAL3/4 VCs, the DCE performs the AAL3/4 SAR and ATM layer functions.

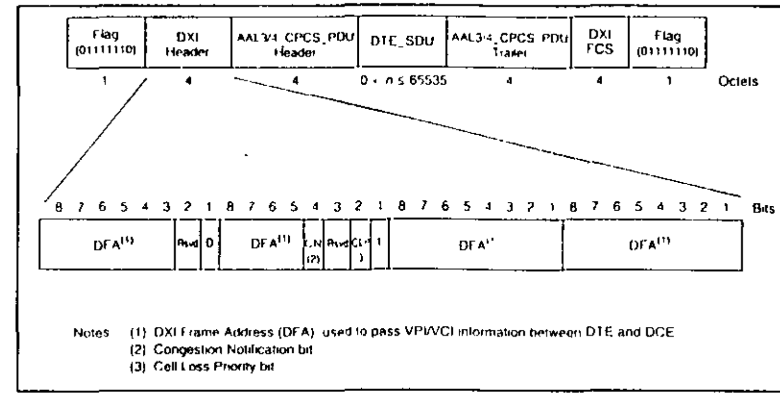


Figure 11-15c. ATM DXI frame format (mode 2).

Source AF-DXI-0014 000, © 1993, The ATM Forum

11.5.2 The DXI LMI

The DXI LMI exchanges management information across the DXI The ATM DXI LMI supports network management systems using SNMP and ATM switches running ILMI. The network management system or the ATM switch may request management information Therefore, the DTE, such as a router, must contain both an SNMP proxy agent and an ILMI proxy agent. If the network management system or the ATM switch sends a request and the DTE does not have the information, it will query the ATM DCE using a DXI LMI PDU (see Figure 11-15d).

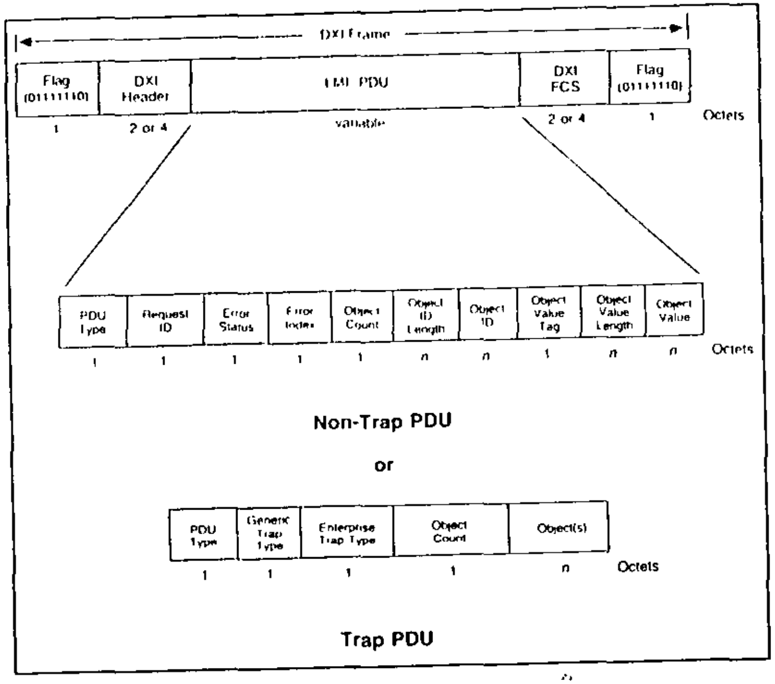


Figure 11-15d. ATM DXI LMI PDU formats.

Source. AF-DXI-0014 000, © 1993, The ATM Forum.

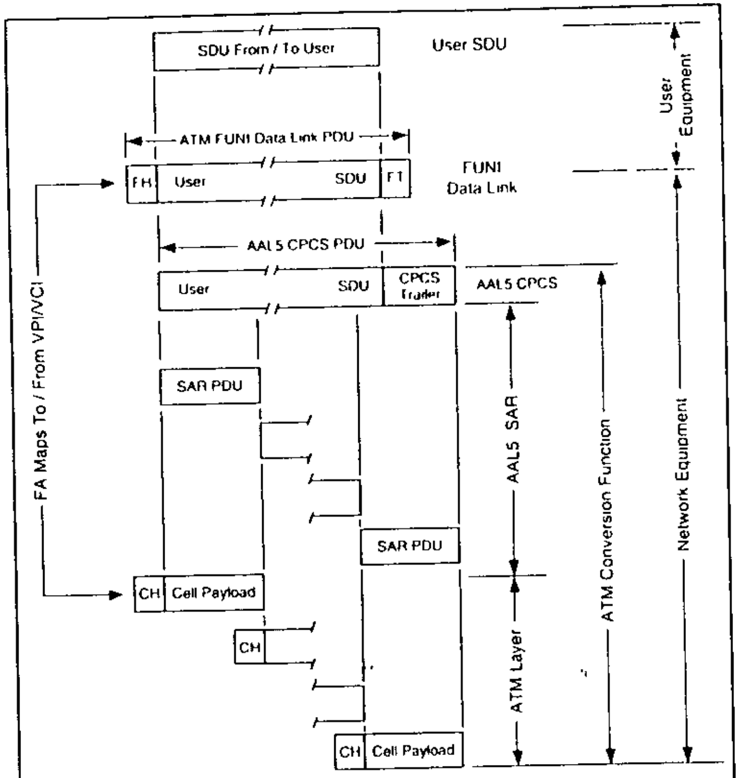
Five LMI messages are defined. The GetRequest, GetNextRequest, and SetRequest originate in the DTE, with a GetResponse returned from the DCE. The Trap message originates in the DCE.

The LMI message formats are similar to those defined for SNMP. The GetRequest, GetNextRequest, SetRequest, and GetResponse messages share a common PDU structure, and the Trap message has a unique PDU format. The ATM Forum's DXI 1.0 specification provides greater details about the PDU formats

11.6 FUNI

The Frame Based User-to-Network Interface, or FUNI, is based on the DXI and provides an interface between end-user equipment and an ATM network at the DS1 (1 544 Mbps) or E1 (2 048 Mbps) rates. Inside the ATM network, a conversion is made between the data carried in the DS1/E1 frame and ATM cells. According to the FUNI specification [11-18], the FUNI shall support DXI mode 1a, may optionally provide support for mode 1b, and shall not support mode 2. The only AALs supported over the FUNI are AAL3/4 and AAL5; therefore, the interface is intended to support VBR and UBR service only. Note that neither AAL1 nor CBR service are supported.

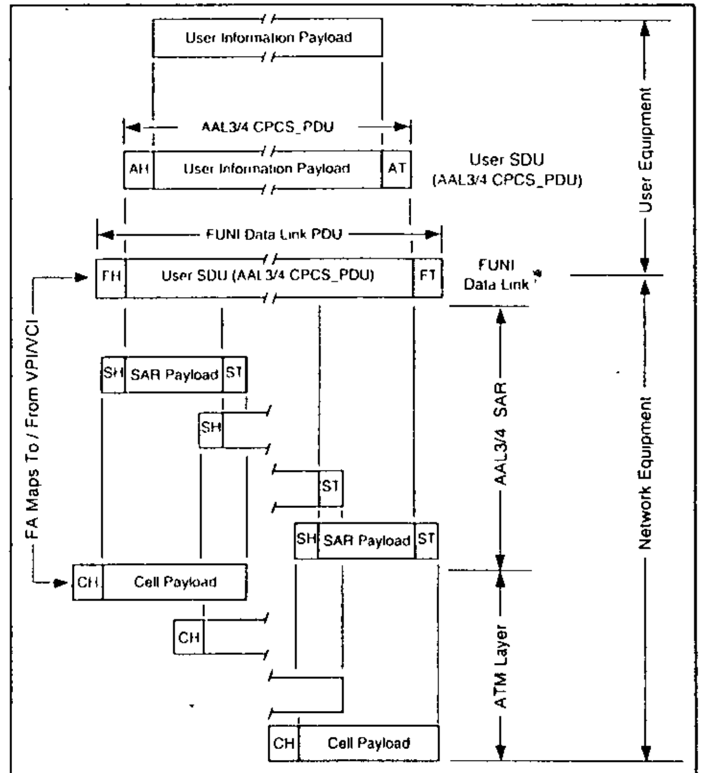
The encapsulation/decapsulation processes are shown in Figure 11-16a (for AAL5) and Figure 11-16b (for AAL3/4). Note that a mapping function translates the frame address (FA) to the VPI/VCI fields to provide the appropriate address conversions. Further details on the FUNI protocols and operation are found in the FUNI specification document [11-18].



Notes
CH Cell Header
CPCS Common Part Convergence Sublayer
FH FUNI Frame Header
FT FUNI/HDLC Frame Trailer
HDLC Flags are not shown

Figure 11-16a. FUNI encapsulation and ATM conversion process for AAL5.

Source AF-SAA-0030 000, © 1995, The ATM Forum



Notes:
AH AAL3/4 CPCS Header
AT AAL3/4 CPCS Trailer
CH Cell Header
FH FUNI Frame Header
FT FUNI/HDLC Frame Trailer
SH SAR Header
ST SAR Trailer
HDLC Flags are not shown

Figure 11-16b. FUNI encapsulation and ATM conversion process for AAL3/4.

Source: AF-SAA-0030 000, © 1995, The ATM Forum

11.7 LAN Emulation

LAN Emulation (or LANE) is a process that allows existing local networks, such as Ethernet/IEEE 802.3 and IEEE 802.5 (token ring) networks, to operate within a switched ATM environment. The term *emulation* is used to describe this process, since many of the core functions present in LANs are not directly available from an ATM system. For example, LAN messages are sent using connectionless transport, which does not require a call setup. ATM is connection-oriented, which does require call setup. LANs are shared media systems, which lend themselves easily to multicast and broadcast transmissions. Within ATM, a means must be defined to intercept these messages and only send them to the stations that actually require that information.

LANE provides functions that are somewhat analogous to bridging and are described in detail in the ATM Forum's LANE Specification [11-19]. This specification details the operation of the various LANE components, including the LANE clients and LANE servers. The LANE client uses one of two frame formats to communicate: a modified Ethernet/IEEE 802.3 frame (Figure 11-17a) or a modified IEEE 802.5 (token ring) frame (Figure 11-17b). Note that the key modification is the addition of a LAN Emulation header (2 octets) which contains either a LAN Emulation Client Identifier (LECID) or the value 0000H. For specific details on LANE operation, refer to the ATM Forum specifications.

LE Header	Destination	Source	Type/Length	Data Unit	Pad	
2	6	6	2	←46-1,500→		Octets

Figure 11-17a. LAN emulation data format for IEEE 802.3/Ethernet frames.

Source. AF-LANE-0021 000, © 1995, The ATM Forum.

LE Header	Access Control	Frame Control	Destination Address	Source Address	Route Info	Information Field	
2	1	1	6	6	0–30	variable	Octets

Figure 11-17b. LAN emulation data format for IEEE 802.5 frames.

Source· AF-LANE-0021 000, © 1995, The ATM Forum

11.8 ATM Signaling

Signaling is the process by which ATM users and the network exchange control information to establish or disconnect VCs, request the use of network resources, or negotiate for the use of circuit parameters such as QOS, VPI/VCI, and so on. Signaling traffic is sent on VPI = 0, VCI = 5.

Meta-signaling is an optional method of establishing signaling channels. Meta-signaling messages are 1 cell long and are sent on VPI = 0, VCI = 1.

Meta-signaling sets up three types of signaling channels: point-to-point, general broadcast, and selective broadcast. There are three meta-signaling procedures: the Assignment procedure establishes a new signaling channel, the Removal procedure disconnects a signaling channel, and the Checking procedure verifies a signaling channel.

11.8.1 ATM Signaling Protocols

The protocols, shown in Figure 11-18, support connection control signaling. ITU-T Recommendation Q.2931 specifies the signaling message format. These messages are sent over the Signaling ATM Adaptation Layer (SAAL), which ensures their reliable delivery. The SAAL is divided into a Service Specific Part and a Common Part. The Service Specific Part is further divided into a Service Specific Coordination Function (SSCF), which interfaces with the SSCF user; and a Service Specific Connection-Oriented Protocol (SSCOP), which ensures reliable delivery. These two protocols are specified in ITU-T Recommendations Q.2130 (formerly designated Q.SAAL.2), and Q.2110 (formerly designated Q.SAAL.1), respectively. The common part of SAAL is AAL5.

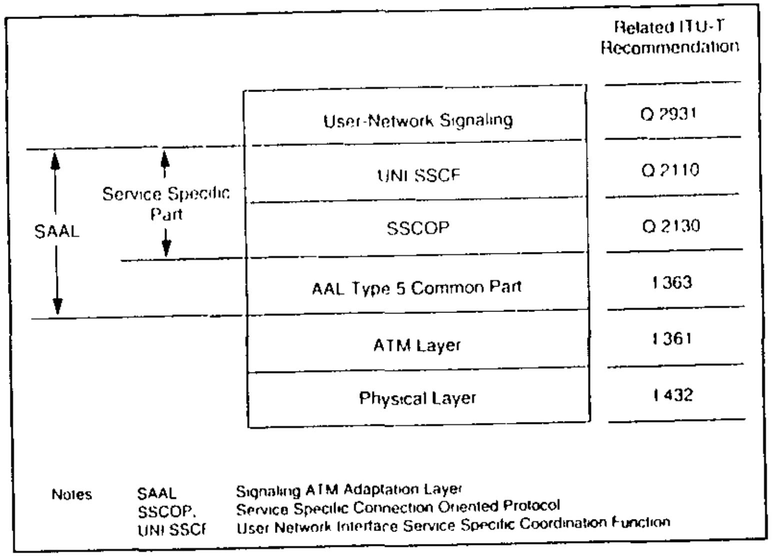


Figure 11-18. SAAL protocol stack at the UNI.

Source GR-1111-CORE, ©1995, Bell Communications Research, Inc., reprinted with permission

The ATM Forum's Signaling Specification v. 4.0 [11-20], ITU-T Q.2931 [11-21], and Bellcore's GR-1111-CORE [11-22] provide further details on the signaling architectures

11.8.2 ATM Address Formats

Before two ATM endpoints can communicate across a private or public UNI, the endpoints must be unambiguously identified. The ATM Forum has defined three Private UNI address formats, each 20 octets long, to provide this identification (see Figure 11-19). The Signaling Specification, version 4.0, states that a Private UNI must be able to accept an initial call setup message containing an ATM address in any of the three formats. A Public UNI must support the native E.164 address format, the three Private UNI address formats, or all of these. These four formats will be discussed below

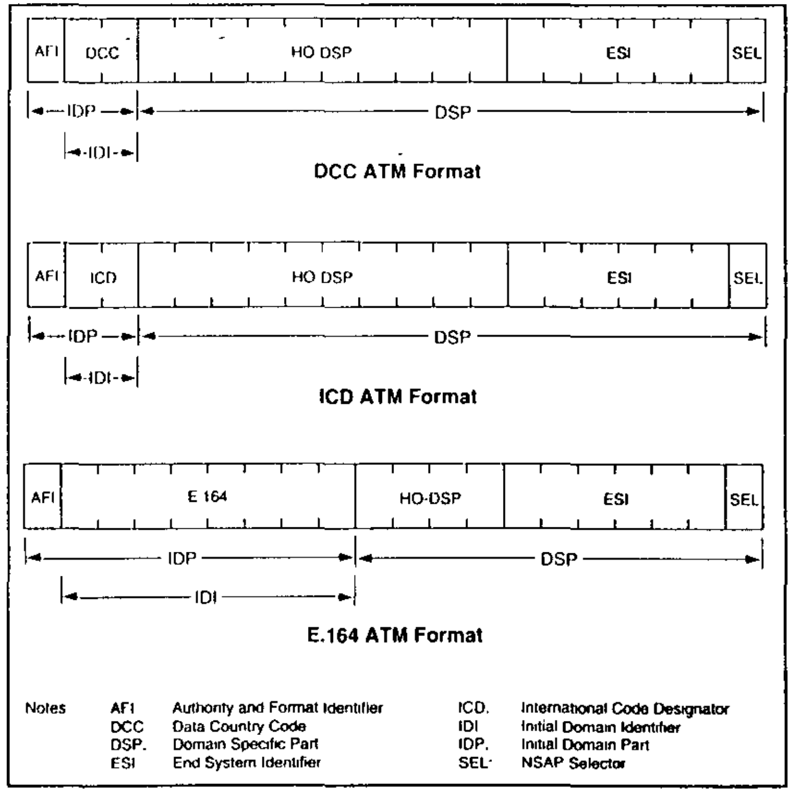


Figure 11-19. ATM address formats.

Source: AF-SIG-0061.000, © 1996, The ATM Forum.

The E.164 address format (not shown) is an ISDN format defined by ITU-T Recommendation E.164 [11-23] and administered by public networks. The address field is 8 octets long and can contain up to 15 Binary Coded Decimal (BCD) digits. A leading pattern of a single 0H character and a trailing pattern of one or more FH characters pad the field to the correct length.

The Data Country Code (DCC) ATM format is divided into an Initial Domain Part (IDP) and a Domain Specific Part (DSP). The IDP contains an Authority and Format Identifier (AFI) and an Initial Domain Identifier (IDI). For the DCC ATM format, the AFI has a value of 39H. The next field contains the DCC (2 octets) that specifies the country where the address is registered. The DSP part of the address contains the High Order DSP (HO-DSP) field, End System Identifier (ESI), and Selector (SEL) fields. The coding of the HO-DSP field is specified by the authority or coding scheme identified by the IDP, and is further defined in Annex 1 of the ATM Forum's Signaling Specification. The ESI is a six-octet number that uniquely identifies the end system, such as an IEEE 802 MAC address. The Selector is a one-octet field which is not used for ATM routing but may be used by the end system.

The International Code Designator (ICD) ATM format is identified by an AFI of 47H. The next field contains the ICD (2 octets), which identifies an international organization and is administered by the British Standards Institute. The remaining address fields are the same as in the DCC ATM format.

The E.164 ATM format is identified by an AFI of 45H. The next field contains the E.164 address (8 octets). The other fields are the HO-DSP, the ESI, and the SEL, which have the same functions as their counterparts in the DCC ATM and ICD ATM formats.

Before an ATM connection can be established at a UNI, both the user and network must be aware of the addresses in effect at that UNI. Address-registration procedures, which are an extension to the ILMI, accomplish this. For Private UNI address formats, the user side of the UNI supplies the user part of the address, the ESI and SEL fields. The network supplies the network prefix, which consists of all the fields that precede the ESI field. When the E.164 address format is used, the network supplies the entire 8-octet address. The address elements are exchanged using ILMI SetRequest messages and are stored in tables at either side of the UNI. After the addresses have been registered, they may be used in the Calling Party Number and Called Party Number information elements transmitted in signaling messages.

11.8.3 ATM Signaling Messages

ATM signaling messages are based on N-ISDN signaling formats specified in Recommendations Q.931 and Q.933. The details of ATM signaling are specified in

B-ISDN Recommendation Q.2931, formerly Q.93B [11-21]. The ATM Forum UNI Signaling Specification version 4.0 builds upon the earlier work with UNI 3.1 and Q.2931, and also adds messages regarding Point-to-Multipoint Call/Connection Control from Q.2971 [11-24].

The ATM signaling messages may be grouped according to their function. Messages for ATM call and connection control include:

- **ALERTING,** sent by the called user to the network or by the network to the calling user to indicate that the called user alerting has been initiated
- **CALL PROCEEDING,** sent by the called user to the network or by the network to the calling user to indicate initiation of the requested call
- **CONNECT,** sent by the called user to the network and by the network to the calling user to indicate that the called user accepted the call
- **CONNECT ACKNOWLEDGE,** sent by the network to the called user to indicate that the call was awarded and sent by the calling user to the network
- **PROGRESS,** sent by the user or the network to indicate the progress of a call in the event of interworking
- **SETUP,** sent by the calling user to the network and by the network to the calling user to initiate a call
- **RELEASE,** sent by the user to request that the network clear the connection or sent by the network to indicate that the connection has cleared
- **RELEASE COMPLETE,** sent by either the user or the network to indicate that the originator has released the call reference and virtual channel
- **RESTART,** sent by the user or the network to restart the indicated virtual channel
- **RESTART ACKNOWLEDGE,** sent to acknowledge the receipt of the RESTART message
- **NOTIFY,** from Q.2971
- **STATUS,** sent by the user or network in response to a STATUS ENQUIRY message
- **STATUS ENQUIRY,** sent by the user or the network to solicit a STATUS message

Messages used with ATM point-to-multipoint call and connection control include.

- **ADD PARTY,** adds a party to an existing connection
- **ADD PARTY ACKNOWLEDGE,** acknowledges a successful ADD PARTY
- **ADD PARTY REJECT,** indicates an unsuccessful ADD PARTY
- **DROP PARTY,** drops a party from an existing point-to-multipoint connection
- **DROP PARTY ACKNOWLEDGE,** acknowledges a successful DROP PARTY
- **PARTY ALERTING,** from Q.2971
- **LEAF SETUP REQUEST,** sent from the Leaf when that Leaf wishes to join a call under the Leaf Initiated Join capabilities
- **LEAF SETUP FAILURE,** sent from the network or Root if the join request from the Leaf could not be completed under the Leaf Initiated Join capabilities

ATM signaling messages, as defined in the UNI Signaling 4.0, use the Q.931 message format (see Figure 11-20) The message consists of five fields:

- *Protocol Discriminator,* distinguishes call-control messages from other traffic
- *Call Reference,* associates this message with a call at the UNI, but does not have end-to-end significance
- *Message Type,* identifies the message function
- *Message Length,* identifies the length of the message contents
- *Information Elements,* parameters required by the message

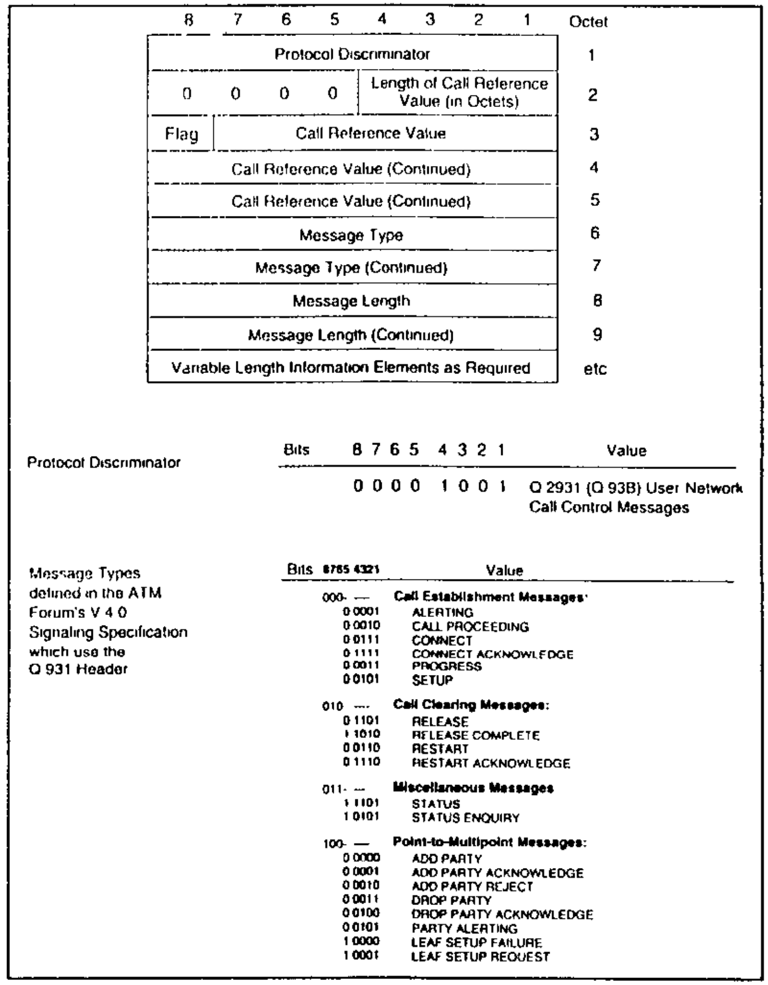


Figure 11-20. Q.931 message format for ATM signaling.

Source: AF-SIG-0061 000, © 1996, The ATM Forum

11.8.4 ATM Information Elements

Information Elements convey details and parameters associated with signaling messages. Figure 11-21 shows the format for the IEs. The first field identifies the IE, and subsequent fields provide control and length information. The IEs defined in UNI Signaling 4.0 are:

- **Narrowband bearer capability,** indicates a requested circuit-mode N-ISDN bearer service to be provided by the network
- **Cause,** identifies the reason for certain messages and provides diagnostic information
- **Call State,** describes the current status of the call, such as call initiated, call present, connect request, or release request
- **Progress indicator,** describes an event that has occurred during the lifetime of a call
- **Notification indicator,** indicates information pertaining to a call
- **End-to-end transit delay,** indicates the nominal maximum end-to-end transit delay acceptable on a per-call basis, and indicates the cumulative transit delay to be expected for a virtual channel connection
- **Connected number,** identifies the connected number (see Q.951).
- **Connected subaddress,** identifies the connected subaddress (see Q.951)
- **Endpoint reference,** identifies the individual endpoints in a point-to-multipoint connection
- **Endpoint state,** indicates the state of an endpoint in a point-to-multipoint connection, such as add/drop party initiated or received
- **ATM Adaptation layer parameters,** indicate the requested AAL end-to-end parameters, such as CPCS_SDU size, CPCS type, or MID size
- **ATM traffic descriptor,** specifies the set of traffic parameters, such as forward or backward peak cell rates or sustainable cell rates
- **Connection identifier,** identifies the local ATM connection, including the VPI/VCI values

- **Quality of service parameter,** requests or indicates the QoS class (zero to four) for a connection
- **Broadband high-layer information,** checks the compatibility of the high-layer information, such as ISO or vendor-specific protocols
- **Broadband bearer capability,** requests a connection-oriented bearer service of the network, such as CBR, VBR, point-to-point, or point-to-multipoint
- **Broadband low-layer information,** checks the compatibility of the low-layer information type, including layer two and three protocols, and packet size
- **Broadband locking shift,** indicates a new active codeset
- **Broadband non-locking shift,** indicates a temporary shift to the specified lower or higher codeset
- **Broadband sending complete,** indicates the completion of the called party number
- **Broadband repeat indicator,** indicates how repeated IEs should be interpreted
- **Calling party number,** identifies the origin of a call
- **Calling party subaddress,** identifies the calling party subaddress
- **Called party number,** identifies the called party
- **Called party subaddress,** identifies the called party subaddress
- **Transit network selection,** identifies one requested transit network
- **Restart indicator,** identifies the class of the facility to be restarted, such as one or all virtual channels
- **Narrowband low-layer compatibility,** provides a means that should be used for compatibility checking by an addressed entity
- **Narrowband high-layer compatibility,** provides a means that should be used by the remote user for compatibility checking
- **Generic identifier transport,** carries an identifier between two users
- **Minimum acceptable traffic descriptor,** specifies the minimum acceptable ATM traffic parameters in the negotiation of traffic parameters during call/connection setup
- **Alternative ATM traffic descriptor,** specifies an alternative ATM traffic descriptor for the negotiation of traffic parameters during call/connection setup

- **ABR setup parameter,** specifies the set of Available Bit Rate parameters during the call/connection establishment
- **Leaf initiated join call identifier,** uniquely identifies a point-to-multipoint call at a Root's interface
- **Leaf initiated join parameters,** used by the Root to associate options with the call when the call is created
- **Leaf sequence number,** used by a joining Leaf to associate a SETUP, ADD PARTY, or LEAF SETUP FAILURE response message with the corresponding LEAF SETUP REQUEST message that triggered the response
- **Connection scope selection,** allows the calling user to indicate to the network that the call/connection shall be processed and progressed within the selected routing range
- **ABR additional parameters,** specify the set of additional ABR parameters during the call/connection establishment
- **Extended QoS parameters,** indicate the individual Quality of Service parameter values on a per-call basis and indicate the cumulative QoS parameter values

The next sections will discuss the use of these messages and information elements.

8	7	6	5	4	3	2	1	Octets
IE Identifier								1
1 ext	Coding Standard		IE Instruction Field: Flag	Res	Spare	Action Indicator		2
Length of IE								3
Length of IE (Continued)								4
Contents of IE								5 etc

Bits 8765 4321		Max Length	Max No of Occurrences
0000 0100	Narrowband Bearer Capability(1, 2)	14	3
0000 1000	Cause(1)	34	2
0001 0100	Call State	5	1
0001 1110	Progress Indicator(1)	6	2
0010 0111	Notification Indicator	(3)	(3)
0100 0010	End-to-End Transit Delay	11	1
0100 1100	Connected Number	25	1
0100 1101	Connected Subaddress	25	1
0101 0100	Endpoint Reference	7	1
0101 0101	Endpoint State	5	1
0101 1000	ATM Adaptation Layer Parameters	21	1
0101 1001	ATM Traffic Descriptor	30	1
0101 1010	Connection Identifier	9	1
0101 1100	Quality of Service Parameter	6	1
0101 1101	Broadband High Layer Information	13	1
0101 1110	Broadband Bearer Capability	7	1
0101 1111	Broadband Low-Layer Information(2)	17	3
0110 0000	Broadband Locking Shift	5	(4)
0110 0001	Broadband Non-Locking Shift	5	(4)
0110 0010	Broadband Sending Complete	5	1
0110 0011	Broadband Repeat Indicator	5	2
0110 1100	Calling Party Number	26	1
0110 1101	Calling Party Subaddress(1)	25	2
0111 0000	Called Party Number	25	1
0111 0001	Called Party Subaddress(1)	25	2
0111 1000	Transit Network Selection	9	1
0111 1001	Restart Indicator	5	1
0111 1100	Narrowband Low Layer Compatibility(2)	20	2
0111 1101	Narrowband High-Layer Compatibility(1)	7	2
0111 1111	Generic Identifier Transport(1)	33	3
1000 0001	Minimum Acceptable Traffic Descriptor	20	1
1000 0010	Alternative ATM Traffic Descriptor	30	1
1000 0100	ABR Setup Parameters	36	1
1110 1000	Leaf Initiated Join Call Identifier	9	1
1110 1001	Leaf Initiated Join Parameters	5	1
1110 1010	Leaf Sequence Number	8	1
1110 1011	Connection Scope Selection	6	1
1110 0100	ABR Additional Parameters	14	1
1110 1100	Extended QoS Parameters	25	1

Notes (1) This IE may be repeated without the broadband repeat indicator IE

(2) This IE may be repeated in conjunction with the broadband repeat indicator IE

(3) The maximum length and the number of repetitions of this IE are network dependent

(4) See § 2 (4 5 3/Q 2931 4 5 4/Q 2931 and 5 6 6/Q 2931) for treatment of these IEs

Figure 11-21. IEs for ATM signaling.

Source. AF-SIG-0061 000, © 1996, The ATM Forum

11.8.5 Call Setup Procedures

Before any of the signaling procedures may be invoked, a user-to-network SAAL connection must be established. Call control signaling is then sent over a permanent signaling virtual channel connection, with VPI = 0 and VCI = 5.

To initiate a call, the calling user sends a SETUP message to the network (see Figure 11-22). The SETUP message is one of the most complex messages; it may contain a number of information elements: AAL Parameters, ATM User Cell Rate, Broadband Bearer Capability, Broadband High-Layer Information, Broadband Repeat Indicator, Broadband Low-Layer Information, Called Party Number, Called Party Subaddress, Calling Party Number, Calling Party Subaddress, Connection Identifier, QOS Parameter, Broadband Sending Complete, Transit Network Selection, and Endpoint Reference.

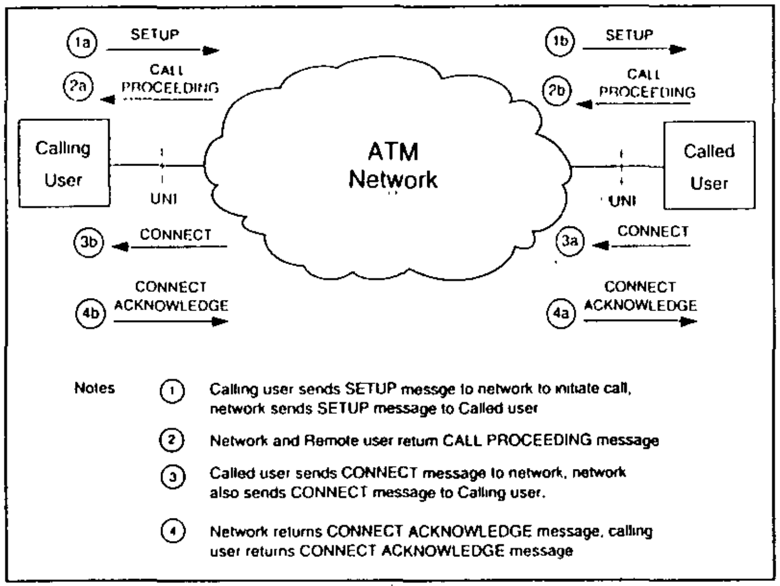


Figure 11-22. ATM call setup procedures.

If the network determines that the requested service is authorized and available, it returns a CALL PROCEEDING message to the calling user and a SETUP message

to the called user. If the network is unable to accept the call, it initiates call clearing as described in the next section. The CALL PROCEEDING message includes Connection Identifier and Endpoint Reference IEs. If the called user wishes to accept the call, it responds with a CALL PROCEEDING message, followed by a CONNECT message. The CONNECT message includes the AAL Parameters, Broadband Low-Layer Information, Connection Identifier, and Endpoint Reference IEs. The network sends a CONNECT ACKNOWLEDGE message to the called user and a CONNECT message to the calling user. The CONNECT ACKNOWLEDGE message conveys no additional parameters. The end-to-end connection is established when the calling user returns a CONNECT ACKNOWLEDGE message to the network.

11.8.6 Call Clearing Procedures

The user or the network may initiate call and connection clearing. (Figure 11-23 illustrates the procedures when the user initiates the clearing; network-initiated procedures are similar.) The user sends a RELEASE message to the network and disconnects the virtual channel. The RELEASE message includes a Cause IE. The network disconnects the virtual channel, initiates procedures to disconnect the remote user, and responds with a RELEASE COMPLETE message. The RELEASE COMPLETE message also includes a Cause IE.

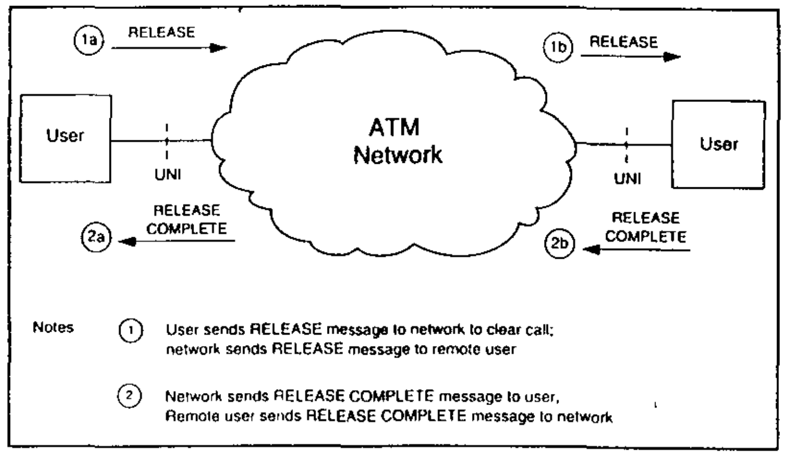


Figure 11-23. ATM call clearing procedures.

11.8.7 Point-to-Multipoint Procedures

Point-to-multipoint connections are a superset of point-to-point connections and use the same signaling channel. The calling user is designated the *Root*, and the called users are designated *Leaves*. The Root sets up the first connection to one Leaf according to the call setup procedures defined for point-to-point calls (see Figure 11-24).

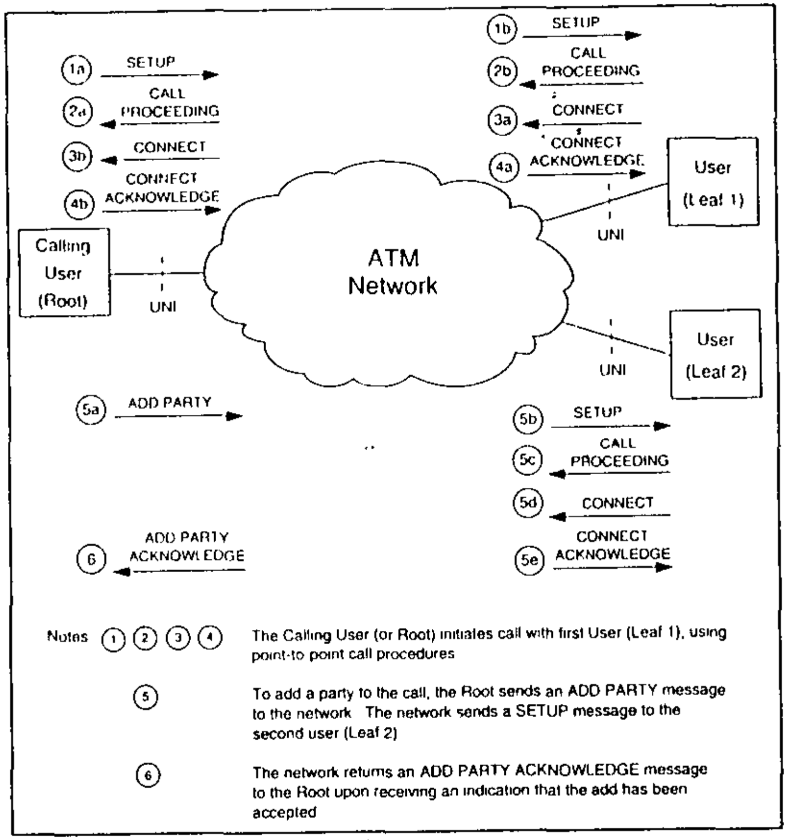


Figure 11-24 ATM point-to-multipoint procedures.

The Root adds a second party by sending an ADD PARTY message to a second Leaf. The ADD PARTY message may include the following IEs: AAL Parameters, Broadband High-Layer Information, Broadband Low-Layer Information, Called Party Number, Called Party Subaddress, Calling Party Number, Calling Party Subaddress, Broadband Sending Complete, Transit Network Selection, and Endpoint Reference. Note that the QOS, Bearer Capability and ATM User Cell Rate IEs are not included in the ADD PARTY message, as these parameters are the same as the originally established (first Root-to-Leaf) call.

When the network receives an ADD PARTY message, it sends a SETUP message or ADD PARTY message across the remote UNI to the Leaf. The SETUP message is sent across the UNI if the link-state is null or clearing, and it initiates the normal CALL PROCEEDING, CONNECT, and CONNECT ACKNOWLEDGE sequence if the user wishes to accept the call.

The ADD PARTY message is sent if the link is in the Active link-state. it initiates an ADD PARTY ACKNOWLEDGE message if the user wishes to accept the call. If the network or called user (Leaf) is unable to accept the ADD PARTY message, it returns an ADD PARTY REJECT message to the Root.

A user or the network may drop a party by sending a DROP PARTY message or a RELEASE message across the interface. The recipient responds with a DROP PARTY ACKNOWLEDGE message or a RELEASE COMPLETE message.

11.8.8 Leaf Initiated Join Procedures

The Leaf Initiated Join (LIJ) procedure is a variation of the point-to-multipoint procedure. In the LIJ procedure, however, the leaf initiates the connection using the LEAF SETUP REQUEST, which contains the LIJ Call Identifier IE and the Root's address (in a Called Party Number IE) (Figure 11-25). If the leaf setup is successful, the network will return a SETUP or ADD PARTY message, and the call will proceed. Otherwise, the network or Root will return a LEAF FAILURE message. Section 6 of the UNI Signaling 4.0 specification is devoted to the LIJ procedures.

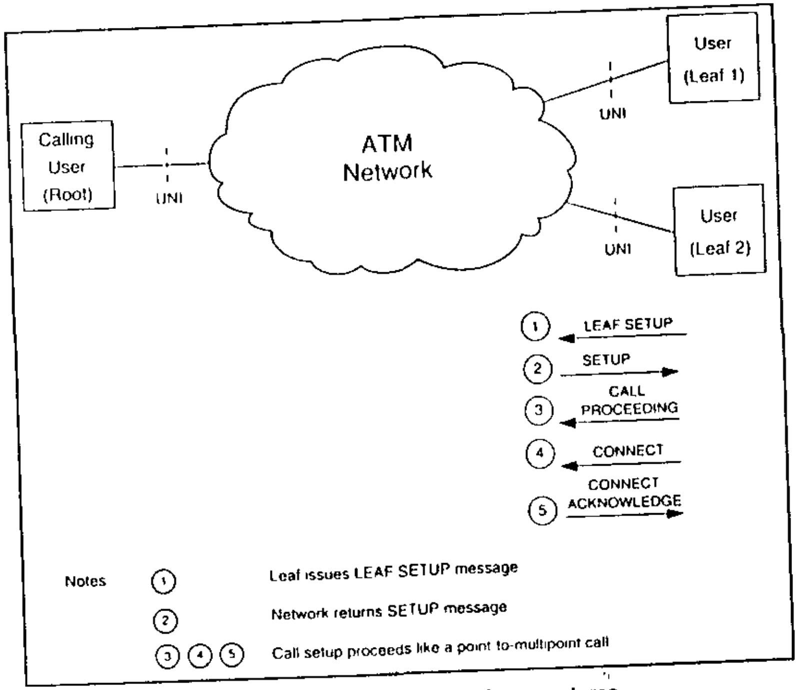


Figure 11-25. Leaf Initiated Join procedures.

11.8.9 Restart Procedures

The restart procedure returns one or all of the virtual channels to the idle condition. It is used when one side of the UNI does not respond to other call-control messages, or after a failure or maintenance action.

A user or the network may send a RESTART message (see Figure 11-26). The RESTART message includes the Restart Indicator IE. If the Restart Indicator IE indicates that only one virtual channel is to be restarted, then a Connection Identifier IE

is included in the message to identify the virtual channel to be returned to the idle condition. The recipient of the RESTART message returns the specified virtual channels to the idle condition, releases all of the call references associated with those virtual channels, and sends a RESTART ACKNOWLEDGE message to the originator. The RESTART ACKNOWLEDGE message includes a Restart Indicator IE, and may also include a Connection Identifier IE.

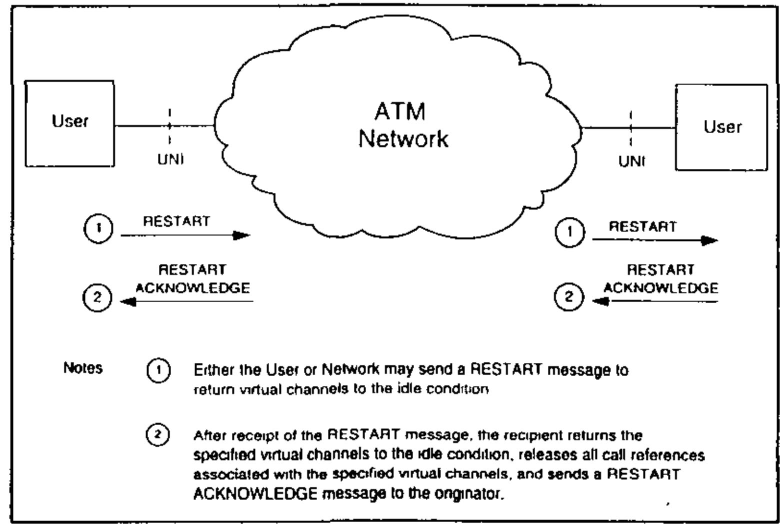


Figure 11-26. ATM restart procedures.

11.8.10 Status Enquiry Procedures

The user or the network may initiate status enquiry procedures to check the status of a call (see Figure 11-27).

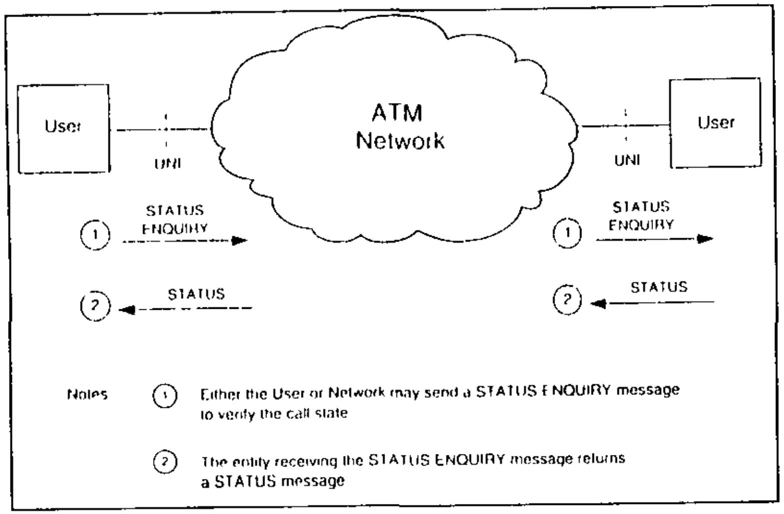


Figure 11-27. ATM status enquiry procedures.

The STATUS ENQUIRY message may optionally include an Endpoint Reference IE. The recipient of the STATUS ENQUIRY returns a STATUS message, which reports on the current call state. The STATUS message includes the Call State, Cause, Endpoint Reference, and Endpoint State IEs.

11.9 ATM Interworking

Using ATM with other protocols in a collaborative manner is called *interworking*. The following sections discuss interworking via multiprotocol encapsulation, frame relay, and SMDS.

11.9.1 Multiprotocol Encapsulation over AAL5

RFC 1483, *Multiprotocol Encapsulation over ATM Adaptation Layer 5* [11-25], defines a method to carry multiprotocol traffic over AAL5. That document describes two methods of support, LLC encapsulation and VC-based multiplexing. In both cases, the higher-layer information, such as TCP/IP or LAN traffic, is carried in the payload field of the Common Part Convergence Sublayer PDU, with the SSCS of AAL5 empty

The LLC encapsulation method, shown in Figure 11-28, is based on techniques developed for use with SMDS. This method is required when a single ATM virtual circuit carries several protocols. Information contained within an IEEE 802.2 LLC header and an IEEE 802.1a SNAP header identifies the protocol carried within that PDU.

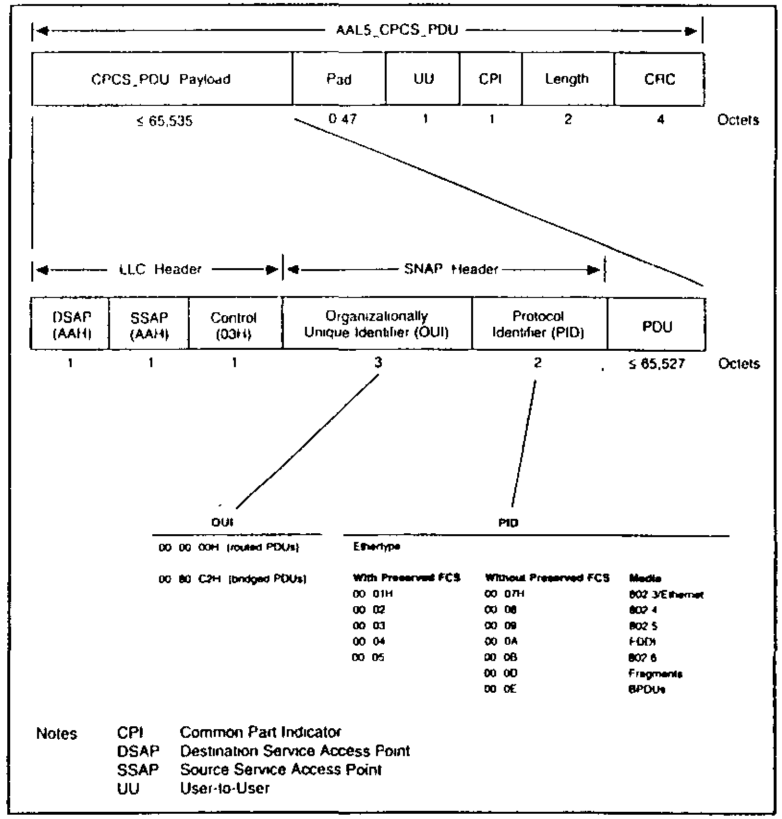


Figure 11-28. Multiprotocol encapsulation over AAL5 (non-ISO routed PDUs or bridged PDUs).

The figure also shows the format used for routed, non-ISO PDUs. (The format for the routed ISO PDUs is slightly different; refer to RFC 1483 for details.) The

DSAP address (1 octet) and the Source Service Access Point (SSAP) address (1 octet) both contain a value of AAH, which indicates that a SNAP header follows. The Control field (1 octet) has a value of 03H, indicating an Unnumbered Information (UI) field. The SNAP header (5 octets) contains two fields: a 3-octet Organizationally Unique Identifier, or OUI; and a 2-octet Protocol Identifier, or PID. The OUI has a value of 00 00 00H for routed PDUs, and a value of 00 80 C2H for bridged PDUs. For routed PDUs, the PID is a 2-octet Ethertype, which for IP would have a value of 08 00H. For bridged PDUs, the PID is a 2-octet field that indicates the type of transmission media used (Ethernet/802.3, 802.5, FDDI, and so on), plus the handling of the FCS. The lower portion of the figure shows PID values defined in RFC 1483. Following the header is the AAL5_CPCS_PDU, which can contain up to 65,527 octets of higher-layer information, such as the MAC LAN frame.

The VC-based multiplexing technique provides higher-layer protocol multiplexing by ATM VCs. Because a separate VC carries each protocol, the AAL5_CPCS_PDU payload does not have to include explicit multiplexing information. For routed protocols, the AAL5_CPCS_PDU payload may be entirely devoted (65,535 octets maximum) to the higher layer information, such as TCP/IP traffic. For bridged frames, only the fields beginning after the PID field are included in the AAL5_CPCS_PDU payload. In other words, the beginning of the PDU would be the MAC Destination Address, followed by the remainder of the MAC frame, any higher-layer information, if applicable, and the LAN FCS.

RFC 1483 provides further details on these encapsulation formats. The specific case of using the Internet Protocol (IP) and the Address Resolution Protocol (ARP) over ATM is discussed in RFC 1577, *Classical IP and ARP over ATM* [11-26], and is also the topic of much discussion and research at the present time.

11.9.2 ATM/Frame Relay

Four organization's documents address interworking between ATM and frame relay: the ITU-T's Recommendation I.555 [11-27], Bellcore's GR-1115-CORE [11-28], the ATM Forum's B-ICI 2.0 Specification [11-29], and the Frame Relay Forum's Implementation Agreements for Network and Service Interworking, References [11-30] and [11-31], respectively.

The logical and physical connection between the frame relay network or device and the ATM network is called an IWF. I.555 defines two functions, encapsulation and protocol mapping, that impact the interworking architectures. *Encapsulation* occurs when "the conversions in the network or in the terminals are such that the protocols used to provide one service make use of the layer service provided by another protocol." In other words, the protocols are stacked at the interworking point.

In contrast, *protocol mapping* occurs when "the network performs conversions in such a way that within a common layer service the protocol information of one protocol is extracted and mapped on protocol information of another protocol." In other words, each end of the connect supports different protocols, but a common layer service in the IWF communicates with both end protocols.

Recommendation I.555 defines two scenarios for connecting networks/devices using B-ISDN (or ATM). Scenario 1 connects two networks/devices via an IWF into and out of a B-ISDN network. In this case, the B-ISDN network is not visible to frame relay users. All mapping and encapsulation functions occur transparently to the end users. This scenario is sometimes called frame relay transport over ATM.

Scenario 2 connects a frame relay network/device with a broadband device using a B-ISDN network. This scenario is also transparent to the end user. In this case, the broadband device supports the frame relay Service Specific Convergence Sublayer (FR-SSCS) function on top of the ATM protocols.

The interworking function maps the frame relay functions to the ATM functions and includes both protocol stacks internally. On the frame relay side is the Q.922 Core and Physical layers. The ATM side includes the FR-SSCS, CPCS, and SAR sublayers for AAL5, plus the ATM and PHY layers.

The B-ICI 2.0 specification and the FR/ATM Network Interworking IA specify that the IWF will support the following frame relay functions: variable length PDU formatting and delimiting, error detection, connection multiplexing, loss priority indication, FECN and BECN indications, and PVC status management.

Figure 11-29 shows the format of the FR-SSCS PDU within AAL5. The B-ICI 2.0 and the FR/ATM IA documents describe details on how the IWF supports each of the above functions.

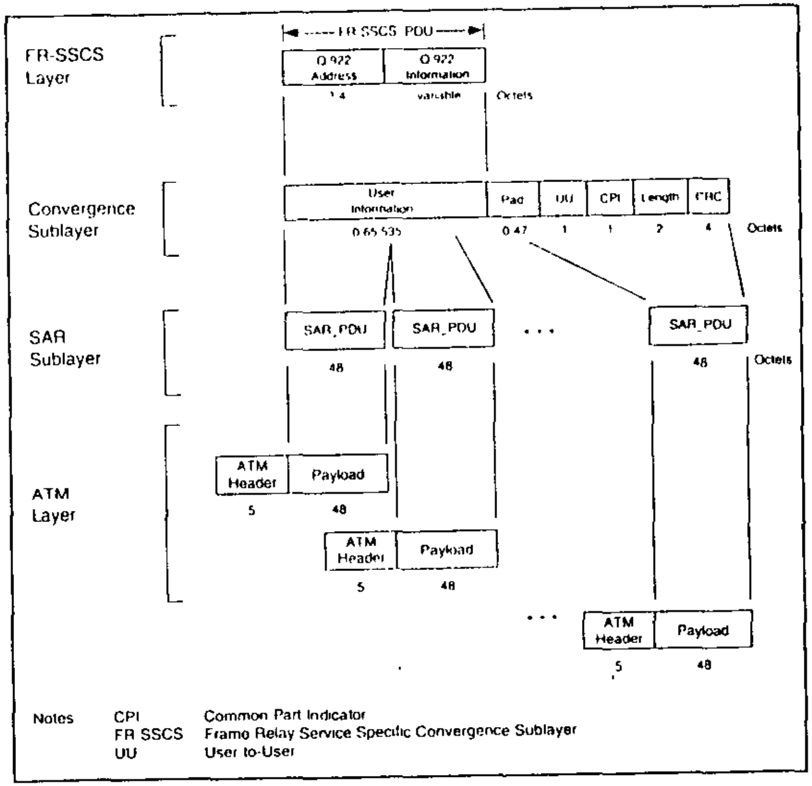


Figure 11-29. FR-SSCS PDU within AAL5.

11.9.3 ATM/SMDS

Interworking between ATM and SMDS has been addressed by three organizations: Bellcore in GR-1110-CORE [11-1], the ATM Forum in the B-ICI 2.0 Specification [11-29], and the SMDS Interest Group (SIG) in a document called Protocol Interface Specification for Implementation of SMDS over an ATM-based Public UNI [11-32].

Under normal conditions, an SMDS CPE accesses the SMDS network at the SNI, using the three layers of Bellcore's SIP. Bellcore and the SIG have defined a method that lets an end user connect to an ATM network using a UNI to access an SMDS service offering. In other words, an end user can use the ATM UNI to access SMDS in the same way that other users would use the SNI or DXI/SNI to access SMDS.

The protocol interface defines a new protocol called SIP Connectionless Service (SIP_CLS). SIP_CLS is a subset of the Connectionless Network Access Protocol, CLNAP, defined by ITU-T Recommendation I.364. SIP_CLS is transported over AAL3/4 (null SSCS, plus CPCS and SAR), as well as the ATM and PHY layers. The combined functions of SIP_CLS and AAL3/4 result in the equivalent of SIP Level 3 functionality. From the customer's perspective, SMDS and the applications that depend on it require no changes. They continue to support features such as multi-CPE arrangements, access classes, and quality of service.

The SIP_CLS_PDU is a subset of the SMDS L3_PDU. To generate a SIP_CLS_PDU, the first and last octets of the SMDS L3_PDU are removed. The resulting PDU includes the Destination Address (DA) through Header Extension fields (32 octets total), the Information field (up to 9,188 octets), and ends with the CRC-32 field (4 octets), as shown in Figure 11-30. The SIP_CLS_PDU may be 32 to 9,224 octets long.

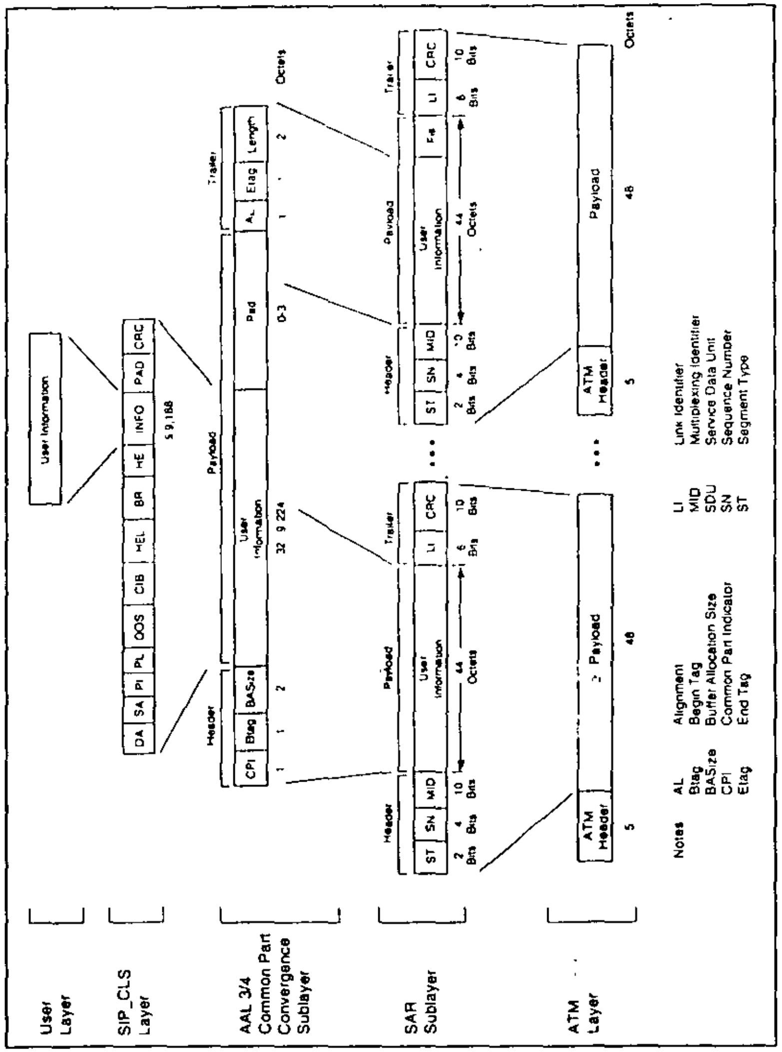


Figure 11-30. PDU format for SMDS on the UNI.

Source. GR-1110-CORE, ©1996, Bell Communications Research, Inc., reprinted with permission

The ATM Forum's B-ICI 2.0 specification supports SMDS/ATM interworking. The interworking function differs from the protocol interface in that a SIP L3_PDU is encapsulated inside another protocol, the Inter Carrier Interface Protocol Connectionless Service (ICIP_CLS). The AAL3/4 then transports the ICIP_CLS_PDU. Mapping functions, which include routing, carrier selection, group address resolution, and others, logically connect SIP Level 3 and ICIP_CLS. Mapping between SMDS and ATM QOS occurs, and performance parameters are also performed.

This chapter has discussed the various ATM protocols and their formats and parameters. Chapter 12 provides case studies that illustrate the operation of these protocols.

11.10 References

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12 ATM Analysis

In Chapter 10, we discussed ATM architecture, in Chapter 11, we covered the ATM protocols. This chapter puts that information to use, examining several case studies illustrating the ATM protocols in action. References [12-1] through [12-3] are recent journal articles that discuss tools and techniques for ATM analysis.

For this section, we used the Hewlett-Packard Broadband Series Test System from HP's Telecom Test Division as the network analyzer (see Figure 12-1). This analyzer has a modular architecture and can be configured with any combination of the following physical interfaces: SONET/SDH (155 Mbps and 622 Mbps), ATM cell-based (155 Mbps), DS3, 4B/5B TAXI, High-Speed Serial Interface (HSSI), and E3 (34 Mbps). It supports a wide range of protocol decodes, including PLCP, ATM, AAL, and SMDS

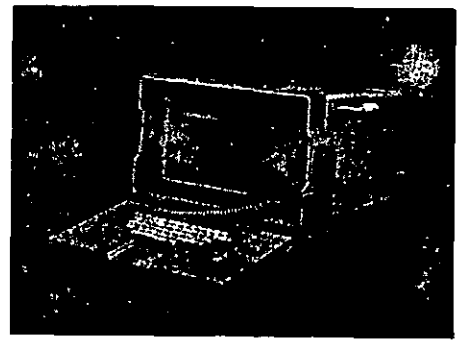

Figure 12-1. Hewlett-Packard Broadband Series Test System.

Courtesy Hewlett-Packard Company

If you are not familiar with the HP Test System's display, some explanation is in order. The first line of the display shows the protocol layer being decoded, such as DS3 PLCP, ATM, AAL3/4 SAR, AAL3/4 CPCS, or CLNAP. Subsequent lines show details of the header and payload for those layers. Details for the Header fields include subfield names and values; details for the Payload field are shown in hexadecimal format. Note that the HP analyzer indicates hexadecimal coding by preceding the value with a 0x. Therefore, a 0xF6 on the analyzer printout would be equivalent to the F6H format for hexadecimal notation used throughout this text.

For the first five case studies in this chapter, the HP analyzer was connected to a DS3 interface, as shown in Figure 12-2.

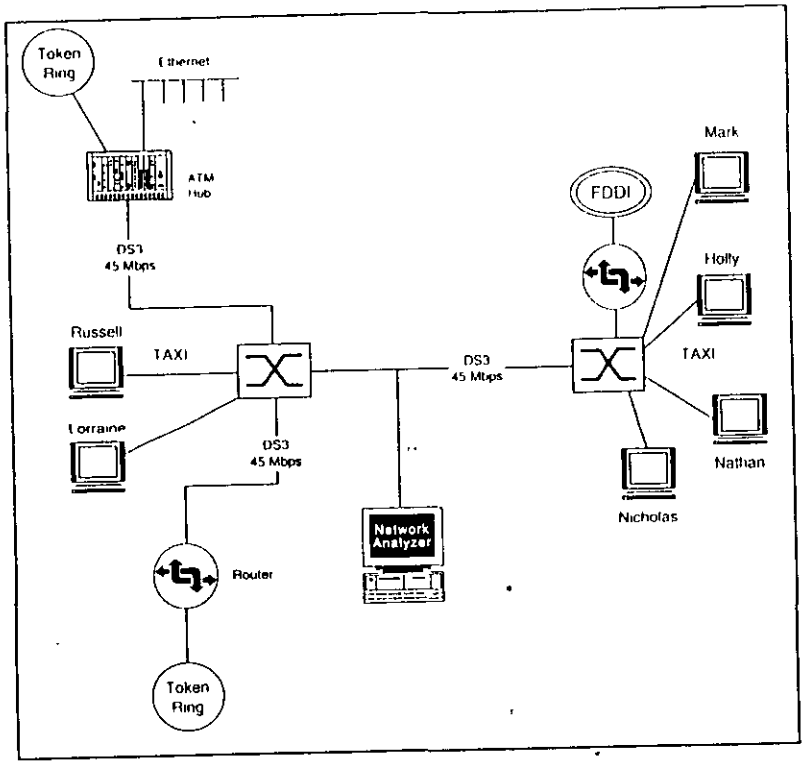

Figure 12-2. Analyzing an ATM network.

12.1 The DS3 Physical Layer Convergence Procedure

The PLCP fills the DS3 frame format with data before transmission at 44.736 Mbps. Each frame contains 48 octets of header information, twelve 53-octet ATM cells, and 13 to 14 nibbles of trailer information, and is transmitted in 125 microseconds (see

Figure 12-3). With overhead, the maximum throughput for ATM cells is 40.704 Mbps. The overhead octets defined in Bellcore's TA-NWT-001112 are as follows:

- A1 Framing, with a pattern of 11110110 (F6H).
- A2 Framing, with a pattern of 00101000 (28H).
- B1 Bit interleaved parity (BIP-8), calculated over the POH field and payload (ATM cells) of the previous PLCP frame.
- C1 Cycle/stuff counter, which provides a nibble-stuffing opportunity and a Trailer length indicator for the PLCP frame. A stuffing opportunity occurs every third frame of a 3-frame (or 375-microsecond) stuffing cycle.

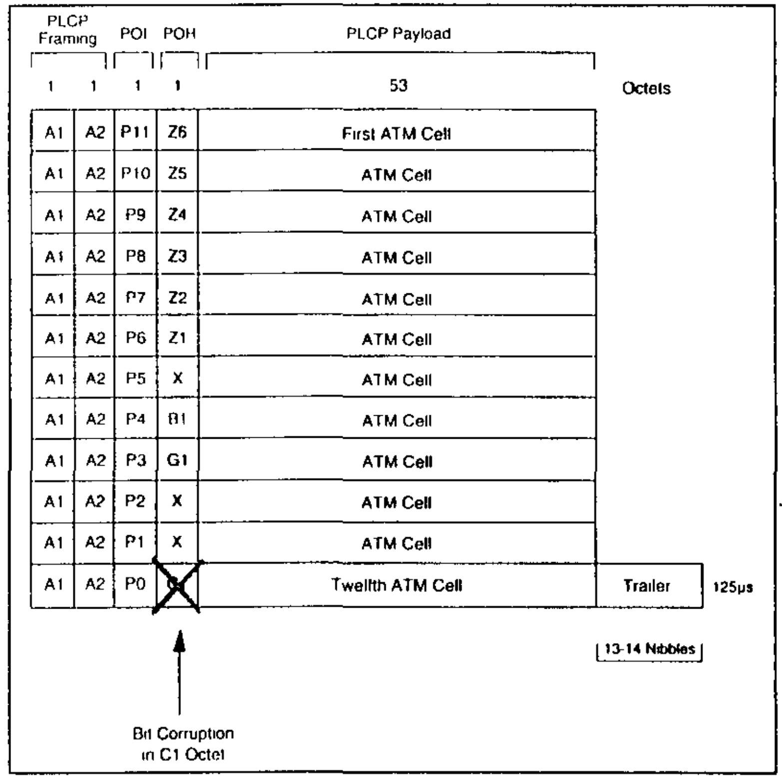

Figure 12-3. DS3 PLCP frame overhead error.

C1 values are.

C1 Octet	Frame Phase	Trailer Length
11111111	(FFH) 1	13
00000000	(00H) 2	14
01100110	(66H) 3 (no stuff)	13
10011001	(99H) 3 (stuff)	14

So, the trailer is 13 nibbles long in the first 125-microsecond frame, 14 nibbles long in the second, and 13 or 14 nibbles long in the third frame, depending on whether or not there has been stuffing for frequency adjustment

G1: PLCP path status, which conveys the received PLCP status and performance to the transmitter at the other end of the link. The G1 octet has three subfields: one 4-bit Far End Block Error (FEBE), a 1-bit Yellow indication, and three reserved bits.

P0–P11: Path overhead identifier, which indexes the adjacent POH octet. The coding of the POI octets are:

POI	POI Code	Associated POH
P11	00101100 (2CH)	Z6
P10	00101001 (29H)	Z5
P9	00100101 (25H)	Z4
P8	00100000 (20H)	Z3
P7	00011100 (1CH)	Z2
P6	00011001 (19H)	Z1
P5	00010101 (15H)	X
P4	00010000 (10H)	B1
P3	00001101 (0DH)	G1
P2	00001000 (08H)	X
P1	00000100 (04H)	X
P0	00000001 (01H)	C1

Z1–Z6 Growth octets, reserved for future use and set to 00000000.

This case study shows the details of a PLCP frame containing 12 ATM cells Trace 12-1 shows the contents of each ATM cell. Each cell printout includes 4 header octets and 48 payload octets. The last line of the trace file shows the contents of the PLCP trailer. In this case, an error occurs on the transmission line, and the C1 octet (the cycle/stuff counter) is invalid. Note that the value of C1 (67H) is identified as an error and is invalid according to the description above

Trace 12-1. DS3 PLCP overhead error

HP Broadband Series Tester Capture Data Record

```
Port 9 1 DS3 PLCP

Header:   A1=0xF6 A2=0x28 P11=0x2C Z6=0x00
Payload   00 00 00 01 52 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A
          6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A
          6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A
          6A 6A 6A 6A 6A

Header    A1=0xF6 A2=0x28 P10=0x29 Z5=0x00
Payload   00 A0 03 20 C2 4A 01 00 01 00 54 00 00 00 00 00
          00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00
          00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00
          00 00 00 10 29

Header:   A1=0xF6 A2=0x28 P9=0x25 Z4=0x00
Payload:  00 A0 03 20 C2 82 01 00 01 00 54 C1 40 35 55 12
          12 FF FF C1 40 34 62 45 45 FF FF 00 0B 00 00 00
          00 00 00 00 00 00 00 00 00 00 00 FF 83 DF 17 32
          09 4E D1 B3 5F

Header:   A1=0xF6 A2=0x28 P8=0x20 Z3=0x00
Payload   00 A0 03 20 C2 06 01 E7 CD 8A 91 C6 D5 C4 C4 40
          21 18 4E 55 86 F4 DC 8A 15 A7 EC 92 DF 93 53 30
          18 CA 34 BF A2 C7 59 67 8F BA 0D 6D D8 2D 7D 36
          C3 E6 5E B0 32

Header    A1=0xF6 A2=0x28 P7=0x1C Z2=0x00
Payload.  00 A0 03 20 C2 4A 01 00 01 00 54 00 00 00 00 00
          00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00
          00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00
          00 00 00 10 29
```

Header A1=0xF6 A2=0x28 P6=0x19 Z1=0x00
Payload 00 00 00 01 52 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A
6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A
6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A
6A 6A 6A 6A 6A

Header A1=0xF6 A2=0x28 P5=0x15 X=0x00
Payload 00 A0 03 20 C2 82 01 00 01 00 54 C1 40 35 55 12
12 FF FF C1 40 34 62 45 45 FF FF 00 0B 00 00 00
00 00 00 00 00 00 00 00 00 00 00 FF 83 DF 17 32
09 4E D1 B3 5F

Header A1=0xF6 A2=0x28 P4=0x10 B1=0x12
Payload 00 A0 03 20 C2 06 01 E7 CD 8A 91 C6 D5 C4 C4 40
21 18 4E 55 86 F4 DC 8A 15 A7 EC 92 DF 93 53 30
18 CA 34 BF A2 C7 59 67 8F BA 0D 6D D8 2D 7D 36
C3 E6 5E B0 32

Header. A1=0xF6 A2=0x28 P3=0x0D G1=0x00
FEBE=0 Yellow signal=0 Link status signal=connected
Payload 00 A0 03 20 C2 4A 01 00 01 00 54 00 00 00 00 00
00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00
00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00
00 00 00 10 29

Header: A1=0xF6 A2=0x28 P2=0x08 X=0x00
Payload 00 A0 03 20 C2 82 01 00 01 00 54 C1 40 35 55 12
12 FF FF C1 40 34 62 45 45 FF FF 00 0B 00 00 00
00 00 00 00 00 00 00 00 00 00 00 FF 83 DF 17 32
09 4E D1 B3 5F

Header A1=0xF6 A2=0x28 P1=0x04 X=0x00
Payload 00 00 00 01 52 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A
6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A
6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A 6A
6A 6A 6A 6A 6A

Header: A1=0xF6 A2=0x28 P0=0x01 C1= 0x67
ERROR The C1 octet is invalid
Payload 00 A0 03 20 C2 06 01 E7 CD 8A 91 C6 D5 C4 C4 40
21 18 4E 55 86 F4 DC 8A 15 A7 EC 92 DF 93 53 30
18 CA 34 BF A2 C7 59 67 8F BA 0D 6D D8 2D 7D 36
C3 E6 5E B0 32
Trailer. 0x C C C C C C C C C C C C C C

12.2 AAL1: Locating a Missing Cell

To verify how the ATM devices being tested will respond to AAL protocol errors, the analyst studies traffic using AAL1, AAL3/4, and AAL5. In this example, test data, consisting of an alternating pattern of ones and zeros, 1012 … 10, or AAAA … AAH, is sent over AAL1. One of the cells is discovered missing. Let's see how AAL1 identifies this problem.

The transmitted data is divided into eight AAL1_SAR_PDUs, which are passed to the ATM layer for transmission in eight ATM cells (see Figure 12-4). The analyzer output shows each ATM header (5 octets) and the corresponding ATM payload (48 octets). Within each payload, the first octet contains the AAL1_SAR_PDU header, and the remaining 47 octets contain the user data (AAAA … AAH).

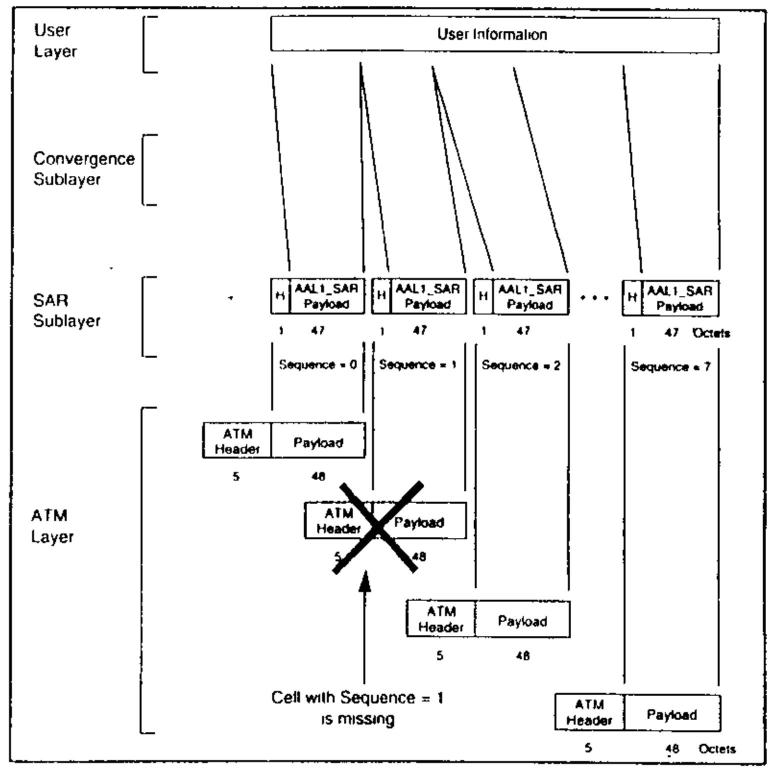

Figure 12-4. Missing AAL1_SAR_PDU.

The top of Trace 12-2, just below the time stamp 14:02 40 34404920, shows the ATM header details. The Generic Flow Control (GFC) = 0, the Virtual Path Identifier (VPI) = 1, and the Virtual Channel Identifier (VCI) = 5. The Payload Type (PT) field has a value of 000, indicating a User data cell with no congestion, and an SDU type (or user indication) of zero. The Cell Loss Priority (CLP) field has a value of zero, indicating a higher priority cell (a CLP = 1 would indicate a lower priority cell). The Header Error Control (HEC) has a value of 40H.

The AAL1_SAR_PDU header contains two fields, Sequence Number (SN) and Sequence Number Protection (SNP). The SN field contains two subfields: a Convergence Sublayer Indicator (CSI) = 0 and a Sequence Count (SC) = 0. (Recall that the CSI field could be used for a Residual Time Stamp, or RTS. But this does not occur in this case, because the values of CSI in each cell have the same value of zero.) The SC field has a value of zero in the first cell, a value of two in the second cell, and then increments from three to seven. Note that the value SC = 1 is not found, indicating a missing cell. The SNP field, a CRC-3, has a value of zero in the first cell, six in the next cell, and so on. The Parity bit is off (P = 0) in the first cell and on (P = 1) in the next cell.

If you combine the four fields (CSI, SC, CRC, and P) into one octet, you can derive the value of the AAL1_SAR_PDU header, shown as the first octet in the ATM payload. Using the second cell as an example, CSI = 0, SC = 010, CRC-3 = 110, and P = 1. This results in a binary value of 00101101, or 2DH, as shown in the payload.

You could make a similar analysis for the ATM and AAL1_SAR_PDU headers in each cell. The one key element is the Sequence Count field, with respective values of 0, 2, 3, 4, 5, 6, and 7. Because SC = 1 is missing, you know that the second cell in the sequence is missing, which means that 47 bits of constant bit rate (CBR) data from the user layer are also missing.

Trace 12-2. Diagnosing a missing AAL1_SAR_PDU

HP Broadband Series Tester Capture Data Record

```
14 02 40 34404920 ATM
 Header    Generic Flow Control          0
           Virtual Path Identifier       1
           Virtual Channel Identifier    5
           Payload Type                  0 (User Data, No Cong, UserInd=0)
           Cell Loss Priority            0 (Higher Priority)
           Header Error Control          0x40
 Payload   00 AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
           AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
           AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA

14 02 40 34404920 AAL-1
 Header    Sequence Number
             Convergence Sublayer Ind    0
             Sequence Count              0
           Sequence Number Protection
             Control Bits (CRC3)         0
             Parity Bit                  0
 Payload   AA AA AA AA AA AA AA AA AA AA

14 02.40 34405950 ATM
 Header:   Generic Flow Control          0
           Virtual Path Identifier       1
           Virtual Channel Identifier    5
           Payload Type                  0 (User Data, No Cong, UserInd=0)
           Cell Loss Priority            0 (Higher Priority)
           Header Error Control          0x40
 Payload:  2D AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
           AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
           AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
```

```
14 02.40 34405950 AAL-1
Header    Sequence Number
            Convergence Sublayer Ind        0
            Sequence Count                  2
          Sequence Number Protection
            Control Bits (CRC3)             6
            Parity Bit                      1
Payload   AA AA AA AA AA AA AA AA AA AA

14 02 40 34406990 ATM
Header    Generic Flow Control            0
          Virtual Path Identifier         1
          Virtual Channel Identifier      5
          Payload Type                    0 (User Data  No Cong, UserInd=0)
          Cell Loss Priority              0 (Higher Priority)
          Header Error Control            0x40
Payload   3A AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
          AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
          AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA

14 02·40 34406990 AAL-1
Header    Sequence Number
            Convergence Sublayer Ind        0
            Sequence Count                  3
          Sequence Number Protection
            Control Bits (CRC3)             5
            Parity Bit                      0
Payload   AA AA AA AA AA AA AA AA AA AA

14.02·40 34408020 ATM
Header.   Generic Flow Control            0
          Virtual Path Identifier         1
          Virtual Channel Identifier      5
          Payload Type                    0 (User Data, No Cong, UserInd=0)
          Cell Loss Priority              0 (Higher Priority)
          Header Error Control            0x40
Payload   4E AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
          AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
          AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
```



```
14 02 40 34408020 AAL-1
Header    Sequence Number
            Convergence Sublayer Ind        0
            Sequence Count                  4
          Sequence Number Protection
            Control Bits (CRC3)             7
            Parity Bit                      0
Payload   AA AA AA AA AA AA AA AA AA AA

14 02 40 34409050 ATM
Header    Generic Flow Control            0
          Virtual Path Identifier         1
          Virtual Channel Identifier      5
          Payload Type                    0 (User Data, No Cong, UserInd=0)
          Cell Loss Priority              0 (Higher Priority)
          Header Error Control            0x40
Payload   59 AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
          AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
          AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA

14 02 40 34409050 AAL-1
Header    Sequence Number
            Convergence Sublayer Ind.       0
            Sequence Count                  5
          Sequence Number Protection
            Control Bits (CRC3)             4
            Parity Bit                      1
Payload   AA AA AA AA AA AA AA AA AA AA

14 02 40 34410080 ATM
Header    Generic Flow Control            0
          Virtual Path Identifier         1
          Virtual Channel Identifier      5
          Payload Type                    0 (User Data, No Cong, UserInd=0)
          Cell Loss Priority              0 (Higher Priority)
          Header Error Control            0x40
Payload.  63 AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
          AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
          AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
```

```
14 02 40 34410080 AAL-1
Header   Sequence Number
           Convergence Sublayer Ind       0
           Sequence Count                 6
          Sequence Number Protection
           Control Bits (CRC3)            1
           Parity Bit                     1
Payload  AA AA AA AA AA AA AA AA AA AA

14 02 40 34411110 ATM
Header   Generic Flow Control             0
         Virtual Path Identifier          1
         Virtual Channel Identifier       5
         Payload Type                     0 (User Data, No Cong, UserInd=0)
         Cell Loss Priority               0 (Higher Priority)
         Header Error Control             0x40
Payload. 74 AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
         AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA
         AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA AA

14 02 40.34411110 AAL-1
Header.  Sequence Number
           Convergence Sublayer Ind       0
           Sequence Count                 7
          Sequence Number Protection
           Control Bits (CRC3)            2
           Parity Bit                     0
Payload  AA AA AA AA AA AA AA AA AA AA
```

12.3 AAL3/4: Identifying SAR Sublayer Errors

The next example (see Trace 12-3) looks at the AAL3/4 process, examining the effects that corruption at the Segmentation and Reassembly layer have on the overall data-transfer process. In this case, 168 octets of user information is sent in the payload of one AAL3/4_CPCS_PDU. This results in four AAL3/4_SAR segments and, therefore, four ATM cells (see Figure 12-5).

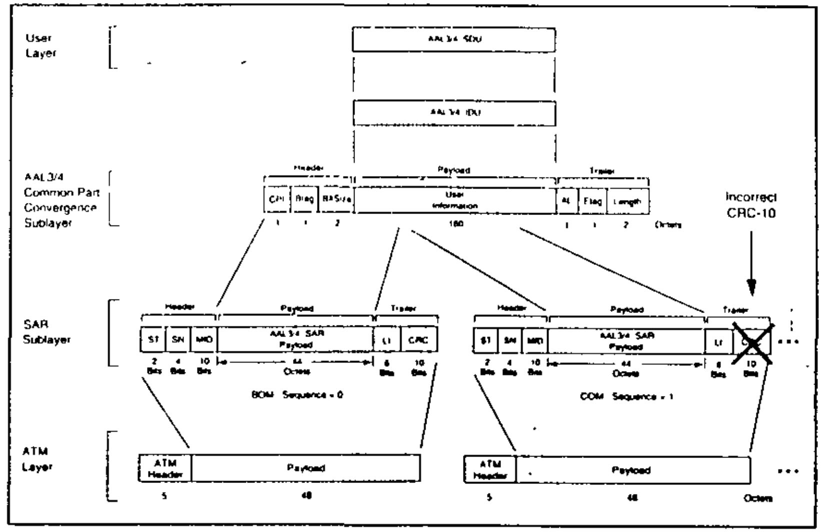

Figure 12-5. AAL3/4_SAR_PDU with SAR Trailer error.

The ATM header fields, including the GFC, VPI/VCI, PT, and so on, are identical to those used in the previous case study. But the overhead associated with AAL3/4 is more extensive than that found in AAL1 because AAL3/4 includes more rigorous error control.

The AAL3/4_SAR header (2 octets) includes three fields. The Segment Type (ST) field, 2 bits long, defines one of four segments: the Beginning of Message (BOM) with ST = 10; the COM with ST = 00; the EOM with ST = 01; and a Single Segment Message (SSM) with ST = 11. This amount of user information requires one BOM, two COMs, and one EOM. The SN, zero through three, keeps the AAL3/4_SAR_PDUs in order. The multiplexing identifier (MID), with a value of one, associates all of the AAL3/4_SAR_PDUs from the same message.

The AAL3/4_SAR trailer (2 octets) includes a Length Indicator field (6 bits), which carries the length of the AAL3/4_SAR_PDU. For BOM and COM segments, the length must be 44 octets; for EOM segments, the length must be a multiple of 4 octets between 4 and 44 octets (4, 8, 12, ..., 44). Note that the EOM segment has a Length = 36 octets, which is a valid number. The second field in the trailer is a CRC-10, used for error control. The analyzer indicates a correct CRC-10 in the BOM segment, but an incorrect value in the first COM.

You can see the effects of the invalid CRC-10 in the first COM segment in subsequent segments. When the receiver finds the CRC-10 to be invalid, that cell (Sequence Number 1) is discarded. When the cell is discarded, the next sequence number (2) becomes invalid. The fourth cell, which is an EOM, arrives next. This further confuses the receiver, which thinks an EOM occurred before a BOM.

The net result is that corruption of one field within one cell caused the entire message to be invalid. To recover, the receiver would have to request a retransmission of that message from its originator. From this example, we can conclude that the rigorous error control incorporated into AAL3/4 provides a solid verification of the integrity of the transmitted message.

Trace 12-3. Effects of an SAR trailer error

HP Broadband Series Tester Capture Data Record

```
13:24:48.81792030 ATM
Header:   Generic Flow Control        0
          Virtual Path Identifier     1
          Virtual Channel Identifier  5
          Payload Type                0 (User Data, No Cong, UserInd=0)
          Cell Loss Priority          0 (Higher Priority)
          Header Error Control        0x40
Payload:  80 01 00 01 00 A0 C1 40 35 55 12 12 FF FF C1 40
          34 62 45 45 FF FF 01 0B 00 00 00 00 00 00 00 00
          00 00 00 00 00 00 00 01 02 03 04 05 06 07 B3 6A

13:24:48.81792030 AAL-3/4 SAR
Header:   Segment Type                2 (BOM)
          Sequence Number             0
          Multiplexing Identifier     1
Payload:  00 01 00 A0 C1 40 35 55 12 12 FF FF C1 40 34 62
          45 45 FF FF 01 0B 00 00 00 00 00 00 00 00 00 00
          00 00 00 00 00 01 02 03 04 05 06 07
```



```
Trailer   Length                      44
          CRC10                       0x36A

13:24:48.81793170 ATM
Header    Generic Flow Control        0
          Virtual Path Identifier     1
          Virtual Channel Identifier  5
          Payload Type                0 (User Data, No Cong, UserInd=0)
          Cell Loss Priority          0 (Higher Priority)
          Header Error Control        0x40
Payload   04 01 08 09 0A 0B 0C 0D 0E 0F 10 11 12 13 14 15
          16 17 18 19 1A 1B 1C 1D 1E 1F 20 21 22 23 24 25
          26 27 28 29 2A 2B 2C 2D 2E 2F 30 31 32 33 B3 44

13:24:48.81793170 AAL-3/4 SAR
Header    Segment Type                0 (COM)
          Sequence Number             1
          Multiplexing Identifier     1
Payload   08 09 0A 0B 0C 0D 0E 0F 10 11 12 13 14 15 16 17
          18 19 1A 1B 1C 1D 1E 1F 20 21 22 23 24 25 26 27
          28 29 2A 2B 2C 2D 2E 2F 30 31 32 33
Trailer   Length                      44
          CRC10                       0x344
ERROR.    CRC10 is incorrect

13:24:48.81794210 ATM
Header    Generic Flow Control        0
          Virtual Path Identifier     1
          Virtual Channel Identifier  5
          Payload Type                0 (User Data, No Cong, UserInd=0)
          Cell Loss Priority          0 (Higher Priority)
          Header Error Control        0x40
Payload:  08 01 34 35 36 37 38 39 3A 3B 3C 3D 3E 3F 40 41
          42 43 44 45 46 47 48 49 4A 4B 4C 4D 4E 4F 50 51
          52 53 54 55 56 57 58 59 5A 5B 5C 5D 5E 5F B3 6B

13:24.48.81794210 AAL-3/4 SAR
Header:   Segment Type                0 (COM)
          Sequence Number             2
          Multiplexing Identifier     1
```

```
Payload     34 35 36 37 38 39 3A 3B 3C 3D 3E 3F 40 41 42 43
            44 45 46 47 48 49 4A 4B 4C 4D 4E 4F 50 51 52 53
            54 55 56 57 58 59 5A 5B 5C 5D 5E 5F
Trailer     Length                       44
            CRC10                        0x36B
PROTOCOL ERROR  Invalid sequence number

13 24 48 81795240 ATM
Header:     Generic Flow Control         0
            Virtual Path Identifier      1
            Virtual Channel Identifier   5
            Payload Type                 0 (User Data, No Cong, UserInd=0)
            Cell Loss Priority           0 (Higher Priority)
            Header Error Control         0x40
Payload     4C 01 60 61 62 63 64 65 66 67 68 69 6A 6B 6C 6D
            6E 6F 70 71 72 73 74 75 76 77 78 79 7A 00 CD 9A
            E6 DD 00 01 00 A0 00 00 00 00 00 00 00 00 90 79

13 24.48.81795240 AAL-3/4 SAR
Header:     Segment Type                 1 (EOM)
            Sequence Number              3
            Multiplexing Identifier      1
Payload     60 61 62 63 64 65 66 67 68 69 6A 6B 6C 6D 6E 6F
            70 71 72 73 74 75 76 77 78 79 7A 00 CD 9A E6 DD
            00 01 00 A0 00 00 00 00 00 00 00 00
Trailer     Length                       36
            CRC10                        0x079
PROTOCOL ERROR  EOM before BOM
```

12.4 AAL3/4: Identifying Higher-Layer Errors

The next case study builds on the previous AAL3/4 example and illustrates the effect that a higher-layer protocol problem has on the transmission of user information.

For this example, shown in Trace 12-4, AAL3/4 is sending Connectionless Network Access Protocol (CLNAP) traffic, which is similar to SMDS (see Figure 12-6). This message also requires four AAL3/4_SAR_PDU segments: one BOM, two COMs, and one EOM, with lengths of 44, 44, 44, and 36 octets, respectively. No errors are detected at the SAR layer, as evidenced by the incrementing sequence numbers (zero, one, two, and three) and the correct CRC-10 fields.

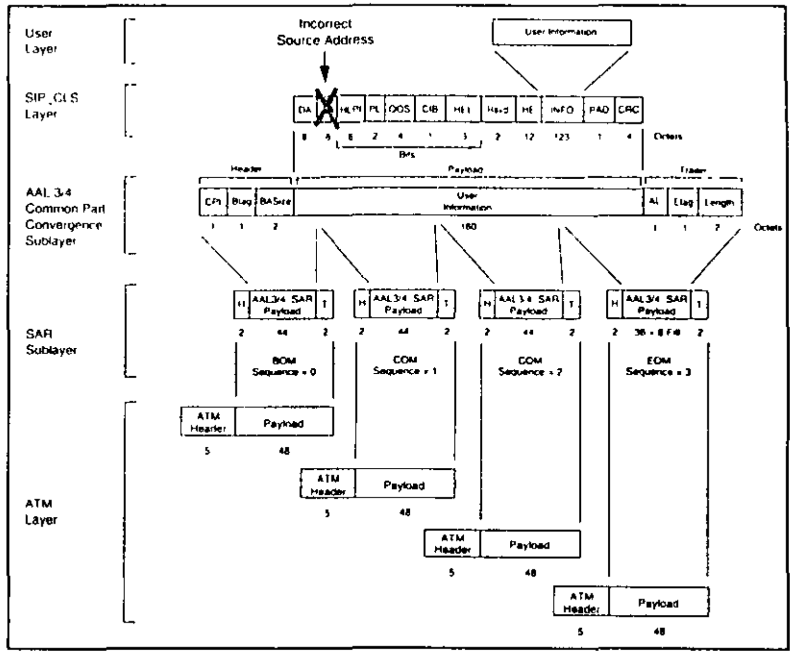

Figure 12-6. AAL3/4_CPCS_PDU with SIP_CLS error.

A decode of the AAL3/4 Common Part Convergence Sublayer (CPCS) header and trailer follows the EOM segment. The AAL3/4_CPCS header (4 octets) includes three fields. The Common Part Indicator (CPI) = 00H indicates that the counting unit is the octet. The BTag field = 01H matches the End Tag (ETag) field in the trailer. The BAsize field = 160 octets indicates the amount of buffer space reserved at the receiver for this message. The values of this header, 00 01 00 A0H, are found in the first line of the BOM payload. Following the CPCS header is the 160 octets of user information, which is distributed between the BOM, COM, and EOM segments.

The AAL3/4_CPCS trailer has an Alignment field (1 octet) filled with 00H; the ETag, as discussed above; and the Length field. Note that the Length field (160 octets) does not equal the sum of the AAL3/4_SAR_PDU payloads (44 + 44 + 44 + 36 = 168 octets) The difference is in the AAL3/4_CPCS header and trailer, which use four octets each. So the Length field measures the length of the AAL3/4_CPCS_PDU payload, not the length of the AAL3/4_CPCS_PDU. The values of the trailer, 00 01 00 A0H, are found at the end of the EOM payload, just before the 8 octets of fill

The CLNAP (SMDS) header begins with the Destination Address field containing the value C1403555121 2FFFF. This is noted as a valid address, and because it begins with an Address Type of CH (1100 binary), we know that this is an individual address. The other permissible value is EH (1110 binary), which represents a group address. The analyzer flags the Source Address as an invalid address because it begins with an FFH. The rest of the CLNAP header and trailer do not indicate any errors. This case study has illustrated that the user information (CLNAP, in this example) is passed transparently by the AAL3/4 processes. Not until the user information reaches the intended receiver is it checked for protocol correctness.

Trace 12-4. Effects of a higher-layer (CLNAP) error

HP Broadband Series Tester Capture Data Record

```
13 03 18 11440800 ATM
Header    Generic Flow Control        0
          Virtual Path Identifier     1
          Virtual Channel Identifier  5
          Payload Type                0 (User Data, No Cong, UserInd=0)
          Cell Loss Priority          0 (Higher Priority)
          Header Error Control        0x40
Payload   80 01 00 01 00 A0 C1 40 35 55 12 12 FF FF FF 40
          34 62 45 45 FF FF 01 0B 00 00 00 00 00 00 00 00
          00 00 00 00 00 00 00 01 02 03 04 05 06 07 B1 7D

13 03 18 11440800 AAL-3/4 SAR
Header.   Segment Type                2 (BOM)
          Sequence Number             0
          Multiplexing Identifier     1
Payload.  00 01 00 A0 C1 40 35 55 12 12 FF FF FF 40 34 62
          45 45 FF FF 01 0B 00 00 00 00 00 00 00 00 00 00
          00 00 00 00 00 01 02 03 04 05 06 07
Trailer   Length                      44
          CRC10                       0x17D
```



```
13 03 18 11441840 ATM
Header.   Generic Flow Control        0
          Virtual Path Identifier     1
          Virtual Channel Identifier  5
          Payload Type                0 (User Data  No Cong  UserInd=0)
          Cell Loss Priority          0 (Higher Priority)
          Header Error Control        0x40
Payload   04 01 08 09 0A 0B 0C 0D 0E 0F 10 11 12 13 14 15
          16 17 18 19 1A 1B 1C 1D 1E 1F 20 21 22 23 24 25
          26 27 28 29 2A 2B 2C 2D 2E 2F 30 31 32 33 B3 16

13 03 18 11441840 AAL-3/4 SAR
Header    Segment Type                0 (COM)
          Sequence Number             1
          Multiplexing Identifier     1
Payload   08 09 0A 0B 0C 0D 0E 0F 10 11 12 13 14 15 16 17
          18 19 1A 1B 1C 1D 1E 1F 20 21 22 23 24 25 26 27
          28 29 2A 2B 2C 2D 2E 2F 30 31 32 33
Trailer:  Length                      44
          CRC10                       0x316

13.03.18 11442870 ATM
Header    Generic Flow Control        0
          Virtual Path Identifier     1
          Virtual Channel Identifier  5
          Payload Type                0 (User Data, No Cong, UserInd=0)
          Cell Loss Priority          0 (Higher Priority)
          Header Error Control        0x40
Payload   08 01 34 35 36 37 38 39 3A 3B 3C 3D 3E 3F 40 41
          42 43 44 45 46 47 48 49 4A 4B 4C 4D 4E 4F 50 51
          52 53 54 55 56 57 58 59 5A 5B 5C 5D 5E 5F B3 6B

13 03 18.11442870 AAL-3/4 SAR
Header.   Segment Type                0 (COM)
          Sequence Number             2
          Multiplexing Identifier     1
Payload   34 35 36 37 38 39 3A 3B 3C 3D 3E 3F 40 41 42 43
          44 45 46 47 48 49 4A 4B 4C 4D 4E 4F 50 51 52 53
          54 55 56 57 58 59 5A 5B 5C 5D 5E 5F
Trailer.  Length                      44
          CRC10                       0x36B
```

13 03 18 11443900 ATM

Header	Generic Flow Control	0
	Virtual Path Identifier	1
	Virtual Channel Identifier	5
	Payload Type	0 (User Data, No Cong, UserInd=0)
	Cell Loss Priority	0 (Higher Priority)
	Header Error Control	0x40
Payload	4C 01 60 61 62 63 64 65 66 67 68 69 6A 6B 6C 6D	
	6E 6F 70 71 72 73 74 75 76 77 78 79 7A 00 AF 0A	
	0F 49 00 01 00 A0 00 00 00 00 00 00 00 00 93 90	

13 03 18 11443900 AAL-3/4 SAR

Header:	Segment Type	1 (EOM)
	Sequence Number	3
	Multiplexing Identifier	1
Payload	60 61 62 63 64 65 66 67 68 69 6A 6B 6C 6D 6E 6F	
	70 71 72 73 74 75 76 77 78 79 7A 00 AF 0A 0F 49	
	00 01 00 A0 00 00 00 00 00 00 00 00	
Trailer	Length	36
	CRC10	0x390

13 03 18 11443900 AAL-3/4 CPCS

Header:	Common Part Indicator	0x00 (BAsize,payld len,Length=payld len)
	Beginning Tag	0x01
	Buffer Allocation Size	160
Payload	C1 40 35 55 12 12 FF FF FF 40 34 62 45 45 FF FF	
	01 0B 00 00 00 00 00 00 00 00 00 00 00 00 00 00	
	00 01 02 03 04 05 06 07 08 09 0A 0B 0C 0D 0E 0F	
	10 11 12 13 14 15 16 17 18 19 1A 1B 1C 1D 1E 1F	
	20 21 22 23 24 25 26 27 28 29 2A 2B 2C 2D 2E 2F	
	30 31 32 33 34 35 36 37 38 39 3A 3B 3C 3D 3E 3F	
	40 41 42 43 44 45 46 47 48 49 4A 4B 4C 4D 4E 4F	
	50 51 52 53 54 55 56 57 58 59 5A 5B 5C 5D 5E 5F	
	60 61 62 63 64 65 66 67 68 69 6A 6B 6C 6D 6E 6F	
	70 71 72 73 74 75 76 77 78 79 7A 00 AF 0A 0F 49	
Pad Characters	not present	
Trailer	Alignment	0x00
	End Tag	0x01
	Length	160

13 03 18 11443900 CLNAP

Header	Destination Address	0xC14035551212FFFF
	Source Address	0xFF4034624545FFFF
ERROR:	Source address type is incorrect	
	Higher Layer Protocol Id	0x00
	Pad Length	1
	Quality Of Service	0x0
	CRC32 Indication Bit	0x1
	Reserved	0x0000
	Header Extension Length	3 (32 bit words)
	Header Extension	0x00 00 00 00 00 00 00 00 00 00 00 00
Payload.	00 01 02 03 04 05 06 07 08 09 0A 0B 0C 0D 0E 0F	
	10 11 12 13 14 15 16 17 18 19 1A 1B 1C 1D 1E 1F	
	20 21 22 23 24 25 26 27 28 29 2A 2B 2C 2D 2E 2F	
	30 31 32 33 34 35 36 37 38 39 3A 3B 3C 3D 3E 3F	
	40 41 42 43 44 45 46 47 48 49 4A 4B 4C 4D 4E 4F	
	50 51 52 53 54 55 56 57 58 59 5A 5B 5C 5D 5E 5F	
	60 61 62 63 64 65 66 67 68 69 6A 6B 6C 6D 6E 6F	
	70 71 72 73 74 75 76 77 78 79 7A	
Pad Characters		0x00
Trailer:	CRC32	0xAF0A0F49

12.5 AAL5: Locating Missing Payload Information

This case study looks at the protocol processes of AAL5, which are considered a subset of AAL3/4. In this example, the user information is divided into five ATM cells (see Figure 12-7). The contents of the user information consist of incrementing hexadecimal numbers: 00, 01, 02, 03, and so on, up to FF, for a total of 256 (16 * 16) octets.

The ATM headers contain the same values for GFC, VPI/VCI, and so on that you have seen in the previous examples. Because AAL5 does not have a header, the payload of the first ATM cell contains 48 octets of user information: 00 01 02 … 2D, 2E, 2F. The second ATM cell contains 60 61 62 … 8D 8E 8F.

Likewise, the third and fourth cells contain 48 octets of data. The fifth cell contains F0 F1 F2 … FD FE FF, 24 octets of zeros, and eight octets containing 00 00 01 00 69 83 2E 15. These last eight octets are the AAL5 trailer, which consists of four fields. The User-User (UU) field = 00. The CPI = 00. The Length field = 0100H, or 256 decimal, which is incorrect. The CRC field is 69 83 2E 15, which is also incorrect.

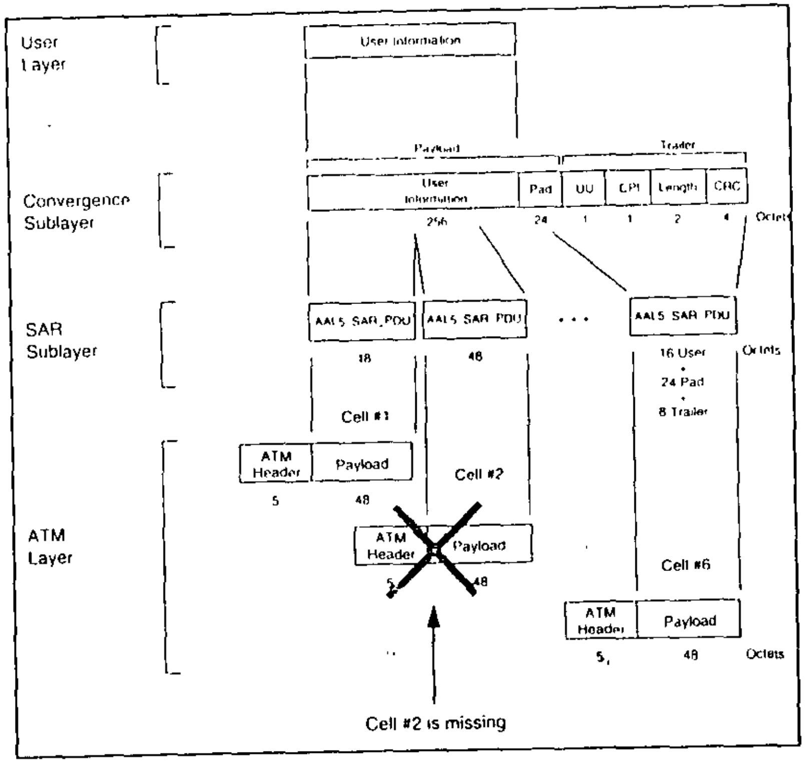

Figure 12-7. Missing AAL5_SAR_PDU.

Knowing that the user information was loaded with 256 octets before transmission, the analyst must determine why the Length and CRC fields were wrong. To find the answer, we looked at the payloads of the five cells:

Cell 1: 00 . . 2F
Cell 2: 60 … 8F
Cell 3: 90 … BF
Cell 4: C0 . EF
Cell 5: F0 … FF, plus 24 octets of zeros

The sequence 30 .. 5F (or 48 octets), which should have been received between Cell 1 and Cell 2, is missing. Because AAL5 has no cell-sequence number, the problem was only identified by the invalid Length and CRC values. Further analysis of the payload information, and the fact that the missing data (30 .. 5F) was easy to identify, lead us to conclude that a cell was missing. Had the missing cell occured under AAL1, the Sequence Count field in the AAL1 header would have identified the problem, with AAL3/4, the Sequence Number and the Segment Type fields would have assisted. As a result, you can conclude that the error control in AAL5 is not as rigorous as that in AAL3/4.

Trace 12-5. Effects of a missing AAL5_SAR_PDU

HP Broadband Series Tester Capture Data Record

```
13 49 18 14378060 ATM
 Header    Generic Flow Control        0
           Virtual Path Identifier     1
           Virtual Channel Identifier  5
           Payload Type                0 (User Data, No Cong, UserInd=0)
           Cell Loss Priority          0 (Higher Priority)
           Header Error Control        0x40
 Payload   00 01 02 03 04 05 06 07 08 09 0A 0B 0C 0D 0E 0F
           10 11 12 13 14 15 16 17 18 19 1A 1B 1C 1D 1E 1F
           20 21 22 23 24 25 26 27 28 29 2A 2B 2C 2D 2E 2F


13 49 18 14379090 ATM
 Header    Generic Flow Control        0
           Virtual Path Identifier     1
           Virtual Channel Identifier  5
           Payload Type                0 (User Data, No Cong, UserInd=0)
           Cell Loss Priority          0 (Higher Priority)
           Header Error Control        0x40
 Payload   60 61 62 63 64 65 66 67 68 69 6A 6B 6C 6D 6E 6F
           70 71 72 73 74 75 76 77 78 79 7A 7B 7C 7D 7E 7F
           80 81 82 83 84 85 86 87 88 89 8A 8B 8C 8D 8E 8F


13 49 18 14380110 ATM
 Header    Generic Flow Control        0
```

	Virtual Path Identifier	1
	Virtual Channel Identifier	5
	Payload Type	0 (User Data, No Cong UserInd=0)
	Cell Loss Priority	0 (Higher Priority)
	Header Error Control	0x40
Payload	90 91 92 93 94 95 96 97 98 99 9A 9B 9C 9D 9E 9F	
	A0 A1 A2 A3 A4 A5 A6 A7 A8 A9 AA AB AC AD AE AF	
	B0 B1 B2 B3 B4 B5 B6 B7 B8 B9 BA BB BC BD BE BF	

13 49 18 14381150 ATM

Header	Generic Flow Control	0
	Virtual Path Identifier	1
	Virtual Channel Identifier	5
	Payload Type	0 (User Data, No Cong, UserInd=0)
	Cell Loss Priority	0 (Higher Priority)
	Header Error Control	0x40
Payload	C0 C1 C2 C3 C4 C5 C6 C7 C8 C9 CA CB CC CD CE CF	
	D0 D1 D2 D3 D4 D5 D6 D7 D8 D9 DA DB DC DD DE DF	
	E0 E1 E2 E3 E4 E5 E6 E7 E8 E9 EA EB EC ED EE EF	

13 49 18 14382180 ATM

Header	Generic Flow Control	0
	Virtual Path Identifier	1
	Virtual Channel Identifier	5
	Payload Type	1 (User Data, No Cong, UserInd=1)
	Cell Loss Priority	0 (Higher Priority)
	Header Error Control	0x4E
Payload	F0 F1 F2 F3 F4 F5 F6 F7 F8 F9 FA FB FC FD FE FF	
	00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00	
	00 00 00 00 00 00 00 00 00 00 01 00 69 83 2E 15	

13 49 18 14382180 AAL-5

Payload	00	01	02	03	04	05	06	07	08	09	0A	0B	0C	0D	0E	0F
	10	11	12	13	14	15	16	17	18	19	1A	1B	1C	1D	1E	1F
	20	21	22	23	24	25	26	27	28	29	2A	2B	2C	2D	2E	2F
	60	61	62	63	64	65	66	67	68	69	6A	6B	6C	6D	6E	6F
	70	71	72	73	74	75	76	77	78	79	7A	7B	7C	7D	7E	7F
	80	81	82	83	84	85	86	87	88	89	8A	8B	8C	8D	8E	8F
	90	91	92	93	94	95	96	97	98	99	9A	9B	9C	9D	9E	9F
	A0	A1	A2	A3	A4	A5	A6	A7	A8	A9	AA	AB	AC	AD	AE	AF
	B0	B1	B2	B3	B4	B5	B6	B7	B8	B9	BA	BB	BC	BD	BE	BF
	C0	C1	C2	C3	C4	C5	C6	C7	C8	C9	CA	CB	CC	CD	CE	CF
	D0	D1	D2	D3	D4	D5	D6	D7	D8	D9	DA	DB	DC	DD	DE	DF
	E0	E1	E2	E3	E4	E5	E6	E7	E8	E9	EA	EB	EC	ED	EE	EF
	F0	F1	F2	F3	F4	F5	F6	F7	F8	F9	FA	FB	FC	FD	FE	FF
	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00
	00	00	00	00	00	00	00	00								

Pad Characters	<none>	
Trailer	User-User Indication	0x00
	Common Part Indicator	0x00
	Length	256
ERROR	Length Field is incorrect	
	CRC32	0x69832E15
ERROR:	CRC32 is incorrect	

12.6 ATM Point-to-Point Call Setup Procedures

The next three case studies investigate the interaction of an ATM switch with end-user devices such as workstations and telephones. In the first of these examples, we will investigate the signaling procedures that are used to set up a point-to-point call. In this example, the HP Broadband analyzer is simulating the function of an end-user device, and actually initiating the call setup to and from an HP workstation, via an ATM switch (Figure 12-8). Note that the HP analyzer has both transmit (Tx) and receive (Rx) connections, which can be noted in the trace files to define the orientation of the cell: either from (Tx) or to (Rx) the analyzer

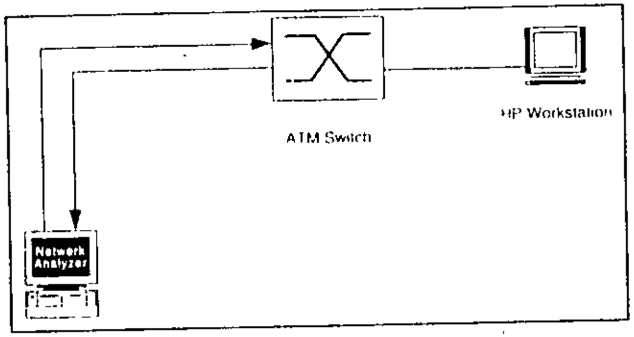

Figure 12-8. Point-to-Point call setup analysis.

Reviewing Figure 11-3, note that the signaling functions (defined by Q 2931 and the ATM Forum's signaling specifications) utilize a Service Specific Coordination Function (SSCF), a Service Specific Connection Oriented Protocol (SSCOP), and AAL5 for communication over the ATM infrastructure.

Trace 12-6a illustrates the operation of the SSCOP for link initialization and subsequent call setup, and has been filtered to only show the highest layer of protocol in operation. The summary information shown in Trace 12-6a includes a timestamp (e.g. 13:44:37:37107640), the Cell Protocol Processor slot number on the analyzer (CPP), the Line Interface slot number on the analyzer (LIF), the direction of transmission (Tx or Rx), the highest layer protocol in use, and a short description of the function of that cell (such as SETUP, CALL PROCEEDING, etc.). Note that the SSCOP link initialization occurs in the first fourteen frames, followed by the UNI signaling functions (the SETUP through RELEASE COMPLETE messages). Traces 12-6b through 12-6g illustrate the details of the individual UNI signaling messages. Review Figure 11-22 for the sequence of events that occur during the call setup procedures.

Trace 12-6a. Point-to-Point Call Setup summary

HP Broadband Series Tester Capture Data Record

```
13 44 37 37107640  CPP 9 LIF 10  Tx  AAL-5  Length = 8
13 44 37 37107640  CPP 9 LIF 10  Tx  SSCOP  BGN
13 44 37 37148090  CPP 9 LIF 10  Rx  AAL-5  Length = 8
13 44 37 37148090  CPP 9 LIF 10  Rx  SSCOP  BGAK
13 44 37.98272150  CPP 9 LIF 10  Tx  AAL-5  Length = 8
13 44 37 98272150  CPP 9 LIF 10  Tx  SSCOP  POLL
```



```
13 44.37 98296530  CPP 9 LIF 10  Rx  AAL-5  Length = 12
13 44 37 98296530  CPP 9 LIF 10  Rx  SSCOP  STAT
13 44 38 02005790  CPP 9 LIF 10  Rx  AAL-5  Length = 8
13 44 38 02005790  CPP 9 LIF 10  Rx  SSCOP  POLL
13 44 38 02089690  CPP 9 LIF 10  Tx  AAL-5  Length = 12
13 44 38.02089690  CPP 9 LIF 10  Tx  SSCOP  STAT
13 44 44 45696230  CPP 9 LIF 10  Tx  AAL-5  Length = 116
13 44 44 45696230  CPP 9 LIF 10  Tx  SSCOP  SD  N(S) = 0
13 44 44 45696230  CPP 9 LIF 10  Tx  UNI Sig  SETUP  3 from
13 44 44 47279460  CPP 9 LIF 10  Rx  AAL-5  Length = 24
13 44 44 47279460  CPP 9 LIF 10  Rx  SSCOP  SD  N(S) = 0
13 44 44 47279460  CPP.9 LIF 10  Rx  UNI Sig  CALL PROCEEDING  3 to
13 44 44 47550200  CPP 9 LIF 10  Rx  AAL-5  Length = 16
13 44 44 47550200  CPP 9 LIF 10  Rx  SSCOP  SD  N(S) = 1
13 44 44 47550200  CPP 9 LIF 10  Rx  UNI Sig  CONNECT  3 to
13 44 44.48468330  CPP 9 LIF 10  Tx  AAL-5  Length = 16
13 44.44 48468330  CPP 9 LIF.10  Tx  SSCOP  SD  N(S) = 1
13 44 44 48468330  CPP 9 LIF 10  Tx  UNI Sig  CONNECT ACKNOWLEDGE  3 from
13 44 45 08271480  CPP 9 LIF 10  Tx  AAL-5  Length = 8
13 44 45 08271480  CPP 9 LIF 10  Tx  SSCOP  POLL
13 44 45.08296590  CPP 9 LIF.10  Rx  AAL-5  Length = 12
13 44.45 08296590  CPP.9 LIF 10  Rx  SSCOP  STAT
13 44 45 12005840  CPP 9 LIF 10  Rx  AAL-5  Length = 8
13 44 45 12005840  CPP 9 LIF 10  Rx  SSCOP  POLL
13 44 45 12091200  CPP 9 LIF:10  Tx  AAL-5  Length = 12
13 44 45 12091200  CPP 9 LIF 10  Tx  SSCOP  STAT
13 44 45 78271340  CPP 9 LIF.10  Tx  AAL-5  Length = 8
13 44 45 78271340  CPP.9 LIF 10  Tx  SSCOP  POLL
13 44 45.78295690  CPP 9 LIF 10  Rx  AAL-5  Length = 12
13 44 45 78295690  CPP.9 LIF 10  Rx  SSCOP  STAT
13 44 45 82005960  CPP 9 LIF 10  Rx  AAL-5  Length = 8
13 44 45 82005960  CPP.9 LIF:10  Rx  SSCOP  POLL
13 44 45 82090020  CPP 9 LIF:10  Tx  AAL-5  Length = 12
13 44 45 82090020  CPP 9 LIF 10  Tx  SSCOP  STAT
13 44 55 67672480  CPP.9 LIF:10  Tx  AAL-5  Length = 20
13 44 55 67672480  CPP 9 LIF.10  Tx  SSCOP  SD  N(S) = 2
13 44.55 67672480  CPP 9 LIF 10  Tx  UNI Sig  RELEASE  3 from
13 44 55 68353670  CPP 9 LIF:10  Rx  AAL-5  Length = 16
13 44 55.68353670  CPP 9 LIF 10  Rx  SSCOP  SD  N(S) = 2
13 44 55.68353670  CPP 9 LIF 10  Rx  UNI Sig  RELEASE COMPLETE  3 to
13 44 56 28272150  CPP 9 LIF 10  Tx  AAL-5  Length = 8
13 44 56 28272150  CPP 9 LIF 10  Tx  SSCOP  POLL
13 44 56 28295970  CPP 9 LIF 10  Rx  AAL-5  Length = 12
```

```
13 44 56 28295970 CPP 9 LIF 10 Rx SSCOP STAT
13 44 56 32006250 CPP 9 LIF 10 Rx AAL-5 Length = 8
13 44 56 32006250 CPP 9 LIF 10 Rx SSCOP POLL
13 44 56 32090500 CPP 9 LIF 10 Tx AAL-5 Length = 12
13 44 56 32090500 CPP 9 LIF 10 Tx SSCOP STAT
```

Trace 12-6b illustrates the details of the SETUP message sent from the end user (analyzer) to the switch (the Tx direction) Note that this message indicates it was sent *from* the call originator (Call Reference Flag = from) and that a Call Reference Value (3) has been assigned In addition, there are seven information elements: ATM Traffic Descriptor, Broadband Bearer Capability, Called Party Number, Quality of Service Parameter, ATM Adaptation Layer Parameters, Broadband Lower Layer Information, and Calling Party Number. In particular, note that the ATM Adaptation Parameters indicate AAL5 will be used for this call.

Trace 12-6b. SETUP message details

```
13 44.44 45696230   CPP 9 LIF 10  Tx  UNI Sig  ATMF UNI 3 1
  1  00001001       Protocol Discriminator      : Q.93B UNI call control
  2  0000----         Spare                     :
     ----0011         Call Reference Length     : 3
  3  0-------         Call Reference Flag       : from
     -0000000         Call Reference Value      : 3
  4  00000000
  5  00000011
  6  00000101       Message Type                : SETUP
  7  1-------         Ext                       : last octet
     -00-----         Spare                     :
     ---0----         Flag                      : not significant
     ----00--         Spare                     :
     ------00         Action Indicator          : clear call
  8  00000000         Message Length            : 102
  9  01100110
  1  01011001       Information Element ID      : ATM Traffic Descriptor
  2  1-------         Ext                       : last octet
     -00-----         Coding Standard           : ITU-T standard
     ---0----         Flag                      : not significant
     ----0---         Reserved                  : reserved
     -----000         Action Indicator          : clear call
  3  00000000         IE Length                 : 9
  4  00001001
  7  10000100         Cell Rate Subfield ID     : forward peak CR(CLP=0+1)
```



```
 7 1  00000101        Forward Peak Cell Rate      : 353207
 7 2  01100011
 7 3  10110111
 8    10000101        Cell Rate Subfield ID       : backward peak CR(CLP=0+1)
 8 1  00000101        Backward Peak Cell Rate     : 353207
 8 2  01100011
 8 3  10110111
 17   10111110        Cell Rate Subfield ID       : best effort indicator
 1    01011110      Information Element ID        : Broadband Bearer Capability
 2    1-------        Ext                         : last octet
      -00-----        Coding Standard             : ITU-T standard
      ---0----        Flag                        : not significant
      ----0---        Reserved                    : reserved
      -----000        Action Indicator            : clear call
 3    00000000        IE Length                   : 3
 4    00000011
 5    0-------        Ext                         : another octet
      -00-----        Spare                       :
      ---10000        Bearer Class                : BCOB-X
 5a   1-------        Ext                         : last octet
      -00-----        Spare                       :
      ---000--        Traffic Type                : no indication
      ------00        Timing Requirements         : no indication
 6    1-------        Ext                         : last octet
      -00-----        Clipping Susceptibility     : not susceptible to clipping
      ---000--        Spare                       :
      ------00        User Plane Connection CFG: point-to-point
 1    01110000      Information Element ID        : Called Party Number
 2    1-------        Ext                         : last octet
      -00-----        Coding Standard             : ITU-T standard
      ---0----        Flag                        : not significant
      ----0---        Reserved                    : reserved
      -----000        Action Indicator            : clear call
 3    00000000        IE Length                   : 21
 4    00010101
 5    1-------        Ext                         : last octet
      -000----        Type of Number              : unknown
      ----0010        Addressing/Numbering Plan.  ISO NSAP addressing
 6etc 01000111        ISO NSAP Address Octets:    0x47 G
      00000000                                    : 0x00 .
      01111001                                    : 0x79 y
      00000000                                    : 0x00 .
      00000000                                    : 0x00 .
```

	00000000		0x00
	00000000		0x00
	00000000		0x00
	00000000		0x00
	00000000		0x00
	00000000		0x00
	00000000		0x00
	00000000		0x00
	00000000		0x00
	10100000		0xA0
	00111110		0x3E >
	00000000		0x00
	00000000		0x00
	00000001		0x01
	00000000		0x00
1	01011100	Information Element ID	Quality of Service Parameter
2	1-------	Ext	last octet
	-00-----	Coding Standard	ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	2
4	00000010		
5	00000000	QoS Class Forward	QoS class 0 - unspecified
6	00000000	QoS Class Backward	QoS class 0 - unspecified
1	01011000	Information Element ID	ATM Adaptation Layer Parameters
2	1-------	Ext	last octet
	-00-----	Coding Standard	ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	9
4	00001001		
5	00000101	AAL Type	AAL type 5
6	10001100	AAL Param Subfield ID	forward maximum CPCS-SDU size
6 1	00000101	Forward CPCS-SDU Size	1516
6 2	11101100		
7	10000001	AAL Param Subfield ID	backward maximum CPCS-SDU size
7 1	00000101	Backward CPCS-SDU Size	1516
7 2	11101100		
8	10000100	AAL Param Subfield ID	SSCS type
8.1	00000000	SSCS Type	null
1	01011111	Information Element ID	Broadband Low Layer Information

2	1-------	Ext	last octet
	-00-----	Coding Standard	ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	9
4	00001001		
7	0-------	Ext	another octet
	-11-----	Layer 3 Id	3
	---01011	User Info Layer 3 Protocol	ISO/IEC TR9577
7a	0-------	Ext	another octet
	-1000000	ISO/IEC TR9577 NLPID	IEEE 802 1 SNAP identifier
	-0------		
7b	1-------	Ext	last octet
	-000000	Spare	
8	1-------	Ext	last octet
	-00-----	SNAP ID	0
	---00000	Spare	
8.1	00000000	OUI Octet	0x00
	10100000		0xA0
	00111110		0x3E >
8 4	00000000	PID Octet	0x00
	00000001		0x01
1	01101100	Information Element ID	Calling Party Number
2	1-------	Ext	last octet
	-00-----	Coding Standard	ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	21
4	00010101		
5	1-------	Ext	last octet
	-000----	Type of Number	unknown
	----0010	Addressing/Numbering Plan	ISO NSAP addressing
6etc	00111001	ISO NSAP Address Octets	0x39 9
	00000000		0x00
	00000000		0x00
	00000000		0x00
	00000000		0x00
	00000000		0x00
	00000000		0x00
	00000000		0x00
	00000000		0x00

00000000	0x00
00000000	0x00
00000000	0x00
00000000	0x00
00000000	0x00
00000000	0x00
00000000	0x00
00000000	0x00
00000000	0x00
00000000	0x00
00000000	0x00

The switch then returns a CALL PROCEEDING message to the analyzer. Note the Rx indication at the top of Trace 12-6c and the Call Reference Flag = to, indicating that this message is being sent to the call originator. Also in this message are the VPCI and VCI assigned to this call (0 and 285, respectively). In the subsequent cell, the switch returns a CONNECT message to the analyzer (Trace 12-6d). We now know that the switch (or ATM network) is prepared to receive information from the analyzer side of the connection. The analyzer then returns a CONNECT ACKNOWLEDGE message (Trace 12-6e), and the transfer of call information may proceed.

Trace 12-6c. CALL PROCEEDING message details

13 44.44 47279460 CPP 9 LIF 10 Rx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	: Q 93B UNI call control
2	0000----	Spare	:
	----0011	Call Reference Length	: 3
3	1-------	Call Reference Flag	: to
	-0000000	Call Reference Value	: 3
4	00000000		
5	00000011		
6	00000010	Message Type	: CALL PROCEEDING
7	1-------	Ext	: last octet
	-00-----	Spare	
	---0----	Flag	: not significant
	----00--	Spare	
	------00	Action Indicator	clear call

8	00000000	Message Length	: 9
9	00001001		
1	01011010	Information Element ID	: Connection Identifier
2	1-------	Ext	last octet
	-00-----	Coding Standard	ITU-T standard
	---0----	Flag	: not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	5
4	00000101		
5	1-------	Ext	last octet
	-00-----	Spare	
	---01---	VP Associated Signalling	explicit indication of VPCI
	-----000	Preferred/Exclusive	exclusive VPCI; exclusive VCI
6	00000000	VPCI	: 0
7	00000000		
8	00000001	VCI	285
9	00011101		

Trace 12-6d. CONNECT message details

13 44.44 47550200 CPP.9 LIF:10 Rx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	: Q 93B UNI call control
2	0000----	Spare	:
	----0011	Call Reference Length	3
3	1-------	Call Reference Flag	: to
	-0000000	Call Reference Value	: 3
4	00000000		
5	00000011		
6	00000111	Message Type	: CONNECT
7	1-------	Ext	: last octet
	-00-----	Spare	:
	---0----	Flag	not significant
	----00--	Spare	
	------00	Action Indicator	: clear call
8	00000000	Message Length	: 0
9	00000000		

Trace 12-6e. CONNECT ACKNOWLEDGE message details

13 44 44.48468330 CPP.9 LIF 10 Tx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	. Q 93B UNI call control
2	0000 ---	Spare	
	----0011	Call Reference Length	3
3	0-------	Call Reference Flag	from
	-0000000	Call Reference Value	3
4	00000000		
5	00000011		
6	00001111	Message Type	CONNECT ACKNOWLEDGE
7	1-------	Ext	· last octet
	-00-----	Spare	.
	---0----	Flag	: not significant
	--- 00--	Spare	
	------00	Action Indicator	. clear call
8	00000000	Message Length	0
9	00000000		

When the end user (the analyzer) has completed all of the information transfer, a RELEASE message is sent to the switch (or network, illustrated in Trace 12-6f and Figure 11-23) Note that the end user uses the Call Reference Value (3) to identify the particular call to be released, and that the RELEASE message contains a Cause information element that defines why the call is being released (normal call clearing). The switch then returns a RELEASE COMPLETE message (Trace 12-6g), which signifies that the call has been cleared and that the network resources allocated for this call are now available for another connection.

Trace 12-6f. RELEASE message details

13·44·55 67672480 CPP.9 LIF 10 Tx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	· Q 93B UNI call control
2	0000----	Spare	
	----0011	Call Reference Length	3
3	0-------	Call Reference Flag	· from
	-0000000	Call Reference Value	. 3
4	00000000		
5	00000011		
6	01001101	Message Type	RELEASE
7	1-------	Ext	· last octet
	-00-----	Spare	:
	---0----	Flag	not significant
	----00--	Spare	

	------00	Action Indicator	clear call
8	00000000	Message Length	6
9	00000110		
1	00001000	Information Element ID	Cause
2	1-------	Ext	last octet
	-00-----	Coding Standard	ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	2
4	00000010		
5	1-------	Ext	: last octet
	-000----	Spare	:
	----0000	Location	· user
6	1-------	Ext	. last octet
	-0010000	Cause Value	· NE normal call clearing

Trace 12-6g. RELEASE COMPLETE message details

13 44 55 68353670 CPP.9 LIF.10 Rx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	Q 93B UNI call control
2	0000----	Spare	
	----0011	Call Reference Length	. 3
3	1-------	Call Reference Flag	· to
	-0000000	Call Reference Value	3
4	00000000		
5	00000011		
6	01011010	Message Type	RELEASE COMPLETE
7	1-------	Ext	: last octet
	-00-----	Spare	.
	---0----	Flag	. not significant
	----00--	Spare	
	------00	Action Indicator	: clear call
8	00000000	Message Length	. 0
9	00000000		

12.7 ATM Point-to-Multipoint Call Setup Procedures

ATM is designed to support a variety of applications, including voice, data, and multimedia For many of these applications, point-to-multipoint connections are required instead of point-to-point connections. Examples of this would be business applications of audio or video teleconferences, or distance learning applications with interactive audio and video connections. This case study will illustrate the protocol interactions required to set up and terminate the point-to-multipoint call. As an early warning to the reader, this case study is rather complex and will involve a number of analyzer trace printouts. However, this complexity also demonstrates the strength of ATM and its ability to support many applications and configurations with equal facility.

In this example, the analyzer was again used to simulate an end-user terminal in order to establish a connection with three voice stations (Figure 12-9). With respect to the point-to-multipoint configuration, the analyzer is acting as the root and the voice stations are acting as the leaves. Reviewing Figure 11-24, recall that the connection with Leaf #1 is set up like a point-to-point connection, and then connections with subsequent leaves are established using the point-to-multipoint procedures. The summary level analyzer output shows both the SSCOP and UNI Signaling protocols in operation as in the point-to-point case; however, additional messages, such as the ADD PARTY and DROP PARTY, are added in this scenario (Trace 12-7a).

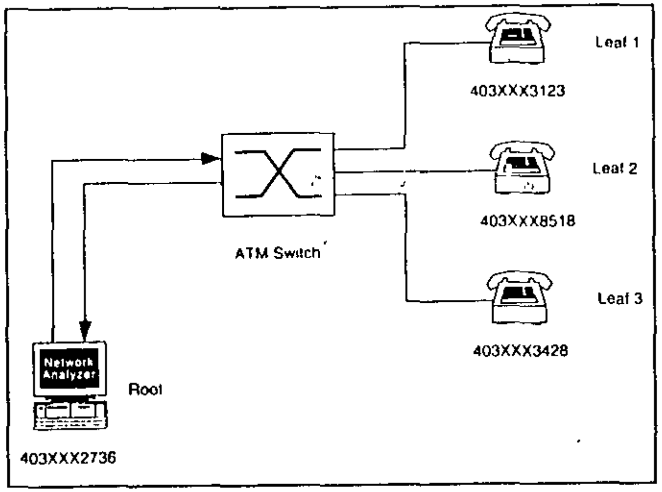

Figure 12-9. Point-to-multipoint call setup analysis.

Trace 12-7a. Point-to-Multipoint Call Setup summary

HP Broadband Series Tester Capture Data Record

```
12:46:41.14861370 CPP 2 LIF 3 Tx SSCOP BGN
12:46:41.14908970 CPP 2 LIF 3 Rx SSCOP BGAK
12:46:41.79362990 CPP 2 LIF 3 Rx SSCOP POLL
12:46:41.79442340 CPP 2 LIF 3 Tx SSCOP STAT
12:46:41.81934830 CPP 2 LIF 3 Tx SSCOP POLL
12:46:41.81964880 CPP 2 LIF 3 Rx SSCOP STAT
12:46:44.21560340 CPP 2 LIF 3 Tx SSCOP SD N(S) = 0
12:46:44.21560340 CPP 2 LIF 3 Tx UNI Sig SETUP 5 from
12:46:44.22105870 CPP 2 LIF 3 Rx SSCOP SD N(S) = 0
12:46:44.22105870 CPP 2 LIF 3 Rx UNI Sig. CALL PROCEEDING 5 to
12:46:44.22287160 CPP 2 LIF 3 Rx SSCOP SD N(S) = 1
12:46:44.22287160 CPP 2 LIF 3 Rx UNI Sig CONNECT 5 to
12:46:44.22561810 CPP 2 LIF 3 Tx SSCOP SD N(S) = 1
12:46:44.22561810 CPP 2 LIF 3 Tx UNI Sig. CONNECT ACKNOWLEDGE 5 from
12:46:44.81935310 CPP 2 LIF 3 Tx SSCOP POLL
12:46:44.81965470 CPP 2 LIF 3 Rx SSCOP STAT
12:46:44.89363820 CPP 2 LIF 3 Rx SSCOP POLL
12:46:44.89442270 CPP 2 LIF 3 Tx SSCOP STAT
12:46:45.51934520 CPP 2 LIF 3 Tx SSCOP POLL
12:46:45.51964880 CPP 2 LIF 3 Rx SSCOP STAT
12:46:45.59363330 CPP 2 LIF 3 Rx SSCOP POLL
12:46:45.59442650 CPP 2 LIF 3 Tx SSCOP STAT
12:46:53.22798170 CPP 2 LIF 3 Tx SSCOP SD N(S) = 2
12:46:53.22798170 CPP 2 LIF 3 Tx UNI Sig. ADD PARTY 5 from
12:46:53.23335660 CPP 2 LIF 3 Rx SSCOP SD N(S) = 2
12:46:53.23335660 CPP 2 LIF 3 Rx UNI Sig ADD PARTY ACKNOWLEDGE 5 to
12:46:53.89363420 CPP 2 LIF 3 Rx SSCOP POLL
12:46:53.89442740 CPP 2 LIF 3 Tx SSCOP STAT
12:46:53.91935050 CPP 2 LIF 3 Tx SSCOP POLL
12:46:53.91965300 CPP 2 LIF 3 Rx SSCOP STAT
12:46:54.59362930 CPP 2 LIF 3 Rx SSCOP POLL
12:46:54.59440900 CPP 2 LIF 3 Tx SSCOP STAT
12:46:54.61934500 CPP 2 LIF 3 Tx SSCOP POLL
12:46:54.61964820 CPP 2 LIF 3 Rx SSCOP STAT
12:46:56.63308440 CPP 2 LIF 3 Tx SSCOP SD N(S) = 3
12:46:56.63308440 CPP 2 LIF 3 Tx UNI Sig ADD PARTY 5 from
12:46:56.63899100 CPP 2 LIF 3 Rx SSCOP SD N(S) = 3
12:46:56.63899100 CPP 2 LIF 3 Rx UNI Sig ADD PARTY ACKNOWLEDGE 5 to
12:46:57.29363360 CPP 2 LIF 3 Rx SSCOP POLL
12:46:57.29442310 CPP 2 LIF 3 Tx SSCOP STAT
```

```
12 46 57 31934660 CPP 2 LIF 3 Tx SSCOP POLL
12 46 57 31964230 CPP 2 LIF 3 Rx SSCOP STAT
12 46 57 99362770 CPP 2 LIF 3 Rx SSCOP POLL
12 46 57 99441910 CPP 2 LIF 3 Tx SSCOP STAT
12 46 58 01935030 CPP 2 LIF 3 Tx SSCOP POLL
12 46 58 01965810 CPP 2 LIF 3 Rx SSCOP STAT
12 47 00 77094350 CPP 2 LIF 3 Tx SSCOP SD N(S) = 4
12 47 00 77094350 CPP 2 LIF 3 Tx UNI Sig DROP PARTY 5 from
12 47 00 77400280 CPP 2 LIF 3 Rx SSCOP SD N(S) = 4
12 47 00 77400280 CPP 2 LIF 3 Rx UNI Sig DROP PARTY ACKNOWLEDGE 5 to
12 47 01 39363740 CPP 2 LIF 3 Rx SSCOP POLL
12 47 01 39442780 CPP 2 LIF 3 Tx SSCOP STAT
12 47 01 41934690 CPP 2 LIF 3 Tx SSCOP POLL
12 47 01 41965760 CPP 2 LIF 3 Rx SSCOP STAT
12 47 02 09363270 CPP 2 LIF 3 Rx SSCOP POLL
12 47 02 09442560 CPP 2 LIF 3 Tx SSCOP STAT
12 47 02 11934740 CPP 2 LIF 3 Tx SSCOP POLL
12 47 02 11965150 CPP 2 LIF 3 Rx SSCOP STAT
12 47 04 05196510 CPP 2 LIF 3 Tx SSCOP SD N(S) = 5
12 47 04 05196510 CPP 2 LIF 3 Tx UNI Sig DROP PARTY 5 from
12 47 04 05490610 CPP 2 LIF 3 Rx SSCOP SD N(S) = 5
12 47 04 05490610 CPP 2 LIF 3 Rx UNI Sig DROP PARTY ACKNOWLEDGE 5 to
12 47 04 69363360 CPP 2 LIF 3 Rx SSCOP POLL
12 47 04 69442310 CPP 2 LIF 3 Tx SSCOP STAT
12 47 04 71934800 CPP 2 LIF 3 Tx SSCOP POLL
12 47 04 71965230 CPP 2 LIF 3 Rx SSCOP STAT
12 47 05 39362750 CPP 2 LIF 3 Rx SSCOP POLL
12 47 05 39441610 CPP 2 LIF 3 Tx SSCOP STAT
12 47 05 41934470 CPP 2 LIF 3 Tx SSCOP POLL
12 47 05 41964750 CPP 2 LIF 3 Rx SSCOP STAT
12 47 09 14326120 CPP 2 LIF 3 Tx SSCOP SD N(S) = 6
12 47 09 14326120 CPP 2 LIF 3 Tx UNI Sig RELEASE 5 from
12 47 09 14645750 CPP 2 LIF 3 Rx SSCOP SD N(S) = 6
12 47 09 14645750 CPP 2 LIF 3 Rx UNI Sig RELEASE COMPLETE 5 to
12 47 09 79363280 CPP 2 LIF 3 Rx SSCOP POLL
12 47 09 79442130 CPP 2 LIF 3 Tx SSCOP STAT
12 47 09 81935020 CPP 2 LIF 3 Tx SSCOP POLL
12 47 09 81965170 CPP 2 LIF 3 Rx SSCOP STAT
12 47 10 49362680 CPP 2 LIF 3 Rx SSCOP POLL
12 47 10 49441380 CPP 2 LIF 3 Tx SSCOP STAT
12 47 10 51934900 CPP 2 LIF 3 Tx SSCOP POLL
12 47 10 51964560 CPP 2 LIF 3 Rx SSCOP STAT
12 47 13 10660400 CPP 2 LIF 3 Tx SSCOP END
12 47 13 10698110 CPP 2 LIF 3 Rx SSCOP ENDAK
```


The SETUP message identifies the Call Reference Value (5) for the call, and also contains six information elements. Of these, the ATM Traffic Descriptor IE indicates a backward peak cell rate of zero (a requirement for this type of point-to-multipoint call), and the Broadband Bearer Capability IE indicates that the User Plane Configuration is, in fact, point-to-multipoint. The Calling Party Number IE identifies the address of the Root (1403XXX2736), while the Called Party Number IE identifies the address of Leaf #1 (1403XXX3123). Note that the telephone numbers have been disguised to maintain the anonymity of the source. The CALL PROCEEDING, CONNECT, and CONNECT ACKNOWLEDGE messages are then exchanged to complete the connection between the Root and Leaf #1 (see Traces 12-7c, 12-7d, and 12-7e, respectively). Note that the CALL PROCEEDING message from the switch (Trace 12-7c) includes an Endpoint Reference IE, which defines an Endpoint Reference Value = 0 for the connection to Leaf #1.

Trace 12-7b. SETUP message details

12 46 44 21560340 CPP 2 LIF 3 Tx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	Q 93B UNI call control
2	0000----	Spare	
	----0011	Call Reference Length	3
3	0-------	Call Reference Flag	from
	-0000000	Call Reference Value	5
4	00000000		
5	00000101		
6	00000101	Message Type	SETUP
7	1-------	Ext	last octet
	-00-----	Spare	
	---0----	Flag	not significant
	----00--	Spare	:
	------00	Action Indicator	: clear call
8	00000000	Message Length	. 63
9	00111111		
1	01011001	Information Element ID	ATM Traffic Descriptor
2	1-------	Ext	last octet
	-00-----	Coding Standard	: ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	8
4	00001000		
7	10000100	Cell Rate Subfield ID	forward peak CR(CLP=0+1)

7 1 00000000	Forward Peak Cell Rate	1000
7 2 00000011		
7 3 11101000		
8 10000101	Cell Rate Subfield ID	backward peak CR(CLP=0+1)
8 1 00000000	Backward Peak Cell Rate	: 0
8 2 00000000		
8 3 00000000		
1 01011110	Information Element ID	Broadband Bearer Capability
2 1-------	Ext	last octet
-00-----	Coding Standard	ITU-T standard
---0----	Flag	. not significant
----0---	Reserved	: reserved
-----000	Action Indicator	clear call
3 00000000	IE Length	. 2
4 00000010		
5 1-------	Ext	last octet
-00-----	Spare	
---10000	Bearer Class	BCOB-X
6 1-------	Ext	last octet
-00-----	Clipping Susceptibility	not susceptible to clipping
---000--	Spare	.
------01	User Plane Connection CFG	: point-to-multipoint
1 01101100	Information Element ID	· Calling Party Number
2 1-------	Ext	. last octet
-00-----	Coding Standard	· ITU-T standard
---0----	Flag	not significant
----0---	Reserved	reserved
-----000	Action Indicator	clear call
3 00000000	IE Length	: 12
4 00001100		
5 1-------	Ext	: last octet
-000----	Type of Number	: unknown
----0010	Addressing/Numbering Plan	: ISDN/telephony numbering plan
6etc 00110001	Address/Number Digits	: 0x31 1
00110100		· 0x34 4
00110000		· 0x30 0
00110011		. 0x33 3
0011xxxx		· 0x3x x
0011xxxx		: 0x3x x
0011xxxx		: 0x3x x
00110010		. 0x32 2
00110111		0x37 7
00110011		0x33 3
00110110		. 0x36 6

1 01110000	Information Element ID	Called Party Number
2 1-------	Ext	. last octet
-00-----	Coding Standard	: ITU-T standard
---0----	Flag	not significant
----0---	Reserved	reserved
-----000	Action Indicator	clear call
3 00000000	IE Length	· 12
4 00001100		
5 1-------	Ext	last octet
-000----	Type of Number	: unknown
----0001	Addressing/Numbering Plan	ISDN/telephony numbering plan
6etc 00110001	Address/Number Digits	: 0x31 1
00110100		0x34 4
00110000		. 0x30 0
00110011		· 0x33 3
0011xxxx		. 0x3x x
0011xxxx		. 0x3x x
0011xxxx		: 0x3x x
00110011		. 0x33 3
00110001		· 0x31 1
00110010		· 0x32 2
00110011		: 0x33 3
1 01011100	Information Element ID	Quality of Service Parameter
2 1-------	Ext	: last octet
-00-----	Coding Standard	. ITU-T standard
---0----	Flag	· not significant
----0---	Reserved	: reserved
-----000	Action Indicator	: clear call
3 00000000	IE Length	: 2
4 00000010		
5 00000000	QoS Class Forward	: QoS class 0 - unspecified
6 00000000	QoS Class Backward	· QoS class 0 - unspecified
1 01010100	Information Element ID	Endpoint Reference
2 1-------	Ext	: last octet
-00-----	Coding Standard	: ITU-T standard
---0----	Flag	: not significant
----0---	Reserved	reserved
-----000	Action Indicator	clear call
3 00000000	IE Length	. 3
4 00000011		
5 00000000	Endpoint Reference Type	locally defined integer

6	0-------	Endpoint Reference Flag	: from the origination side
	-0000000	Endpoint Reference Value	: 0
6 1	00000000		

Trace 12-7c. CALL PROCEEDING message details

12 46 44 22105870 CPP 2 LIF 3 Rx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	Q 93B UNI call control
2	0000----	Spare	
	----0011	Call Reference Length	: 3
3	1-------	Call Reference Flag	to
	-0000000	Call Reference Value	: 5
4	00000000		
5	00000101		
6	00000010	Message Type	: CALL PROCEEDING
7	1-------	Ext	last octet
	-00-----	Spare	:
	---0----	Flag	: not significant
	----00--	Spare	:
	------00	Action Indicator	: clear call
8	00000000	Message Length	16
9	00010000		
1	01011010	Information Element ID	: Connection Identifier
2	1-------	Ext	: last octet
	-00-----	Coding Standard	: ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	: reserved
	-----000	Action Indicator	: clear call
3	00000000	IE Length	: 5
4	00000101		
5	1-------	Ext	last octet
	-00-----	Spare	:
	---01---	VP Associated Signalling	: explicit indication of VPCI
	-----000	Preferred/Exclusive	: exclusive VPCI, exclusive VCI
6	00000000	VPCI	: 0
7	00000000		
8	00000000	VCI	: 35
9	00100011		
1	01010100	Information Element ID	Endpoint Reference
2	1-------	Ext	last octet
	-00-----	Coding Standard	ITU-T standard

	---0----	Flag	not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	: clear call
3	00000000	IE Length	3
4	00000011		
5	00000000	Endpoint Reference Type	: locally defined integer
6	1-------	Endpoint Reference Flag	to the origination side
	-0000000	Endpoint Reference Value	0
6 1	00000000		

Trace 12-7d. CONNECT (Leaf #1) message details

12.46 44 22287160 CPP 2 LIF 3 Rx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	: Q 93B UNI call control
2	0000----	Spare	
	----0011	Call Reference Length	3
3	1-------	Call Reference Flag	: to
	-0000000	Call Reference Value	: 5
4	00000000		
5	00000101		
6	00000111	Message Type	: CONNECT
7	1-------	Ext	: last octet
	-00-----	Spare	:
	---0----	Flag	: not significant
	----00--	Spare	:
	------00	Action Indicator	: clear call
8	00000000	Message Length	: 7
9	00000111		
1	01010100	Information Element ID	: Endpoint Reference
2	1-------	Ext	: last octet
	-00-----	Coding Standard	: ITU-T standard
	---0----	Flag	: not significant
	----0---	Reserved	: reserved
	-----000	Action Indicator	: clear call
3	00000000	IE Length	: 3
4	00000011		
5	00000000	Endpoint Reference Type	: locally defined integer
6	1-------	Endpoint Reference Flag	: to the origination side
	-0000000	Endpoint Reference Value	: 0
6 1	00000000		

Trace 12-7e. CONNECT ACKNOWLEDGE (Leaf #1) message details

12 46 44 22561810 CPP 2 LIF 3 Tx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	Q 93B UNI call control
2	0000----	Spare	
	----0011	Call Reference Length	3
3	0-------	Call Reference Flag	: from
	-0000000	Call Reference Value	5
4	00000000		
5	00000101		
6	00001111	Message Type	CONNECT ACKNOWLEDGE
7	1-------	Ext	last octet
	-00-----	Spare	
	---0----	Flag	not significant
	----00--	Spare	
	------00	Action Indicator	: clear call
8	00000000	Message Length	: 0
9	00000000		

Next, Leaf #2 is added to the connection, as illustrated in Trace 12-7f. Note that the message is an ADD PARTY, which identifies the same Call Reference Value (5) but a different Called Party Number (1403XXX8518). The Endpoint Reference Value (1) indicates Leaf #2. The switch then returns an ADD PARTY ACKNOWLEDGE message confirming the same Call Reference Value (5) and Endpoint Reference Value (1), as shown in Trace 12-7g

Trace 12-7f. ADD PARTY (Leaf #2) message details

12 46 53.22798170 CPP 2 LIF:3 Tx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	: Q 93B UNI call control
2	0000----	Spare	:
	----0011	Call Reference Length	: 3
3	0-------	Call Reference Flag	. from
	-0000000	Call Reference Value	5
4	00000000		
5	00000101		
6	10000000	Message Type	: ADD PARTY
7	1-------	Ext	: last octet
	-00-----	Spare	
	---0----	Flag	: not significant
	----00--	Spare	
	------00	Action Indicator	clear call
8	00000000	Message Length	23

9	00010111		
1	01110000	Information Element ID	Called Party Number
2	1-------	Ext	last octet
	-00-----	Coding Standard	ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	. 12
4	00001100		
5	1-------	Ext	. last octet
	-000----	Type of Number	unknown
	----0010	Addressing/Numbering Plan	ISDN/telephony numbering plan
6etc	00110001	Address/Number Digits	: 0x31 1
	00110100		. 0x34 4
	00110000		: 0x30 0
	00110011		: 0x33 3
	0011xxxx		: 0x3x x
	0011xxxx		0x36 x
	0011xxxx		0x31 x
	00111000		0x38 8
	00110101		. 0x35 5
	00110001		. 0x31 1
	00111000		0x38 8
1	01010100	Information Element ID	: Endpoint Reference
2	1-------	Ext	: last octet
	-00-----	Coding Standard	: ITU-T standard
	---0----	Flag	: not significant
	----0---	Reserved	: reserved
	-----000	Action Indicator	: clear call
3	00000000	IE Length	: 3
4	00000011		
5	00000000	Endpoint Reference Type	. locally defined integer
6	0-------	Endpoint Reference Flag	: from the origination side
	-0000000	Endpoint Reference Value	: 1
6.1	00000001		

Trace 12-7g. ADD PARTY ACKNOWLEDGE (Leaf #2) message details

12 46 53 23335660 CPP 2 LIF 3 Rx UNI Sig. ATMF UNI 3 1

1	00001001	Protocol Discriminator	Q 93B UNI call control
2	0000----	Spare	
	----0011	Call Reference Length	3

3	1-------	Call Reference Flag	to
	-0000000	Call Reference Value	5
4	00000000		
5	00000101		
6	10000001	Message Type	ADD PARTY ACKNOWLEDGE
7	1-------	Ext	last octet
	-00-----	Spare	
	---0----	Flag	not significant
	----00--	Spare	
	------00	Action Indicator	clear call
8	00000000	Message Length	: 7
9	00000111		
1	01010100	Information Element ID	Endpoint Reference
2	1-------	Ext	last octet
	-00-----	Coding Standard	ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	3
4	00000011		
5	00000000	Endpoint Reference Type	locally defined integer
6	1-------	Endpoint Reference Flag	to the origination side
	-0000000	Endpoint Reference Value	1
6 1	00000001		

In a similar manner, Leaf #3 is added to the connection. The ADD PARTY message (Trace 12-7h) includes the same Call Reference Value (5), the address of Leaf #3 (1403XXX3428), and a new Endpoint Reference Value (2). The ADD PARTY ACKNOWLEDGE message from the switch confirms the connection to Leaf #3 (Trace 12-7i).

Trace 12-7h. ADD PARTY (Leaf #3) message details

12 46 56 63308440 CPP:2 LIF 3 Tx UNI Sig. ATMF UNI 3 1

1	00001001	Protocol Discriminator	Q 93B UNI call control
2	0000----	Spare	
	----0011	Call Reference Length	3
3	0-------	Call Reference Flag	from
	-0000000	Call Reference Value	: 5
4	00000000		
5	00000101		
6	10000000	Message Type	ADD PARTY

7	1-------	Ext	last octet
	-00-----	Spare	
	---0----	Flag	not significant
	----00--	Spare	
	------00	Action Indicator	: clear call
8	00000000	Message Length	23
9	00010111		
1	01110000	Information Element ID	Called Party Number
2	1-------	Ext	last octet
	-00-----	Coding Standard	ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	: 12
4	00001100		
5	1-------	Ext	last octet
	-000----	Type of Number	unknown
	----0010	Addressing/Numbering Plan	ISDN/telephony numbering plan
6etc	00110001	ISO NSAP Address Octets	: 0x31 1
	00110100		0x34 4
	00110000		0x30 0
	00110011		0x33 3
	0011xxxx		0x3x x
	0011xxxx		0x3x x
	0011xxxx		0x3x x
	00110011		0x33 3
	00110100		0x34 4
	00110010		0x32 2
	00111000		0x38 8

1	01010100	Information Element ID	Endpoint Reference
2	1-------	Ext	last octet
	-00-----	Coding Standard	: ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	: reserved
	-----000	Action Indicator	: clear call
3	00000000	IE Length	: 3
4	00000011		
5	00000000	Endpoint Reference Type	: locally defined integer
6	0-------	Endpoint Reference Flag	from the origination side
	-0000000	Endpoint Reference Value	2
6 1	00000010		

Trace 12-7i. ADD PARTY ACKNOWLEDGE (Leaf #3) message details

```
12 46 56 63899100 CPP 2 LIF 3 Rx UNI Sig ATMF UNI 3 1
 1   00001001       Protocol Discriminator       Q 93B UNI call control
 2   0000----       Spare                        :
     ----0011       Call Reference Length        . 3
 3   1-------       Call Reference Flag          to
     -0000000       Call Reference Value         5
 4   00000000
 5   00000101
 6   10000001       Message Type                 ADD PARTY ACKNOWLEDGE
 7   1-------       Ext                          last octet
     -00-----       Spare
     ---0----       Flag                         . not significant
     ----00--       Spare                        .
     ------00       Action Indicator             clear call
 8   00000000       Message Length               . 7
 9   00000111
 1   01010100       Information Element ID       Endpoint Reference
 2   1-------       Ext                          last octet
     -00-----       Coding Standard              ITU-T standard
     ---0----       Flag                         · not significant
     ----0---       Reserved                     reserved
     -----000       Action Indicator             . clear call
 3   00000000       IE Length                    : 3
 4   00000011
 5   00000000       Endpoint Reference Type      . locally defined integer
 6   1-------       Endpoint Reference Flag      : to the origination side
     -0000000       Endpoint Reference Value     · 2
 6 1 00000010
```

To disconnect one of the end users from the point-to-multipoint connection, the Root sends a DROP PARTY message (Trace 12-7j). This message includes the Call Reference Value (5) that identifies the call, plus the Endpoint Reference Value (2) that identifies the endpoint to be dropped (Leaf #3). The switch returns a DROP PARTY ACKNOWLEDGE message (Trace 12-7k) containing similar identifiers. In a similar way, Leaf #2 can be dropped from the connection using another DROP PARTY/DROP PARTY ACKNOWLEDGE sequence, this time with Endpoint Reference Value = 1 (identifying Leaf #2), as shown in Traces 12-7l and 12-7m.

Trace 12-7j. DROP PARTY (Leaf #3) message details

```
12.47 00 77094350 CPP 2 LIF 3 Tx UNI Sig ATMF UNI 3 1
 1   00001001       Protocol Discriminator       Q.93B UNI call control
 2   0000----       Spare
     ----0011       Call Reference Length        : 3
 3   0-------       Call Reference Flag          from
     -0000000       Call Reference Value         5
 4   00000000
 5   00000101
 6   10000011       Message Type                 · DROP PARTY
 7   1-------       Ext                          last octet
     -00-----       Spare
     ---0----       Flag                         not significant
     ----00--       Spare                        :
     ------00       Action Indicator             : clear call
 8   00000000       Message Length               · 13
 9   00001101
 1   00001000       Information Element ID       . Cause
 2   1-------       Ext                          last octet
     -00-----       Coding Standard              : ITU-T standard
     ---0----       Flag                         : not significant
     ----0---       Reserved                     · reserved
     -----000       Action Indicator             . clear call
 3   00000000       IE Length                    : 2
 4   00000010
 5   1-------       Ext                          last octet
     -000----       Spare
     ----0000       Location                     . user
 6   1-------       Ext                          : last octet
     -0011111       Cause Value                  : NE unspecified
 1   01010100       Information Element ID       : Endpoint Reference
 2   1-------       Ext                          : last octet
     -00-----       Coding Standard              . ITU-T standard
     ---0----       Flag                         : not significant
     ----0---       Reserved                     reserved
     -----000       Action Indicator             · clear call
 3   00000000       IE Length                    · 3
 4   00000011
 5   00000000       Endpoint Reference Type      . locally defined integer
 6   0-------       Endpoint Reference Flag      from the origination side
     -0000000       Endpoint Reference Value     2
 6 1 00000010
```

Trace 12-7k. DROP PARTY ACKNOWLEDGE (Leaf #3) message details

12 47 00 77400280 CPP 2 LIF 3 Rx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	Q 93B UNI call control
2	0000----	Spare	
	----0011	Call Reference Length	3
3	1-------	Call Reference Flag	to
	-0000000	Call Reference Value	5
4	00000000		
5	00000101		
6	10000100	Message Type	DROP PARTY ACKNOWLEDGE
7	1-------	Ext	last octet
	00-----	Spare	
	---0----	Flag	not significant
	----00--	Spare	
	------00	Action Indicator	clear call
8	00000000	Message Length	13
9	00001101		
1	00001000	Information Element ID	Cause
2	1-------	Ext	last octet
	-00-----	Coding Standard	ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	reserved
	---- 000	Action Indicator	. clear call
3	00000000	IE Length	: 2
4	00000010		
5	1-------	Ext	. last octet
	-000----	Spare	
	----0000	Location	user
6	1-------	Ext	last octet
	-0011111	Cause Value	. NE:unspecified
1	01010100	Information Element ID	Endpoint Reference
2	1-------	Ext	: last octet
	-00-----	Coding Standard	: ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	3
4	00000011		
5	00000000	Endpoint Reference Type	locally defined integer
6	1-------	Endpoint Reference Flag	to the origination side
	-0000000	Endpoint Reference Value	: 2
6 1	00000010		

Trace 12-7l. DROP PARTY (Leaf #2) message details

12 47 04 05196510 CPP 2 LIF 3 Tx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	Q 93B UNI call control
2	0000----	Spare	
	----0011	Call Reference Length	3
3	0-------	Call Reference Flag	from
	-0000000	Call Reference Value	5
4	00000000		
5	00000101		
6	10000011	Message Type	DROP PARTY
7	1-------	Ext	last octet
	-00-----	Spare	
	---0----	Flag	not significant
	----00--	Spare	
	------00	Action Indicator	clear call
8	00000000	Message Length	13
9	00001101		
1	00001000	Information Element ID	Cause
2	1-------	Ext	last octet
	-00-----	Coding Standard	ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	2
4	00000010		
5	1-------	Ext	: last octet
	-000----	Spare	:
	----0000	Location	user
6	1-------	Ext	last octet
	-0011111	Cause Value	NE:unspecified
1	01010100	Information Element ID	. Endpoint Reference
2	1-------	Ext	last octet
	-00-----	Coding Standard	. ITU-T standard
	---0----	Flag	: not significant
	----0---	Reserved	: reserved
	-----000	Action Indicator	: clear call
3	00000000	IE Length	3
4	00000011		
5	00000000	Endpoint Reference Type	locally defined integer
6	0-------	Endpoint Reference Flag	from the origination side
	-0000000	Endpoint Reference Value	1
6 1	00000001		

Trace 12-7m. DROP PARTY ACKNOWLEDGE (Leaf #2) message details

12 47 04 05490610 CPP 2 LIF 3 Rx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	Q 93B UNI call control
2	0000----	Spare	
	----0011	Call Reference Length	3
3	1-------	Call Reference Flag	: to
	-0000000	Call Reference Value	5
4	00000000		
5	00000101		
6	10000100	Message Type	DROP PARTY ACKNOWLEDGE
7	1-------	Ext	last octet
	-00-----	Spare	
	---0----	Flag	not significant
	----00--	Spare	
	------00	Action Indicator	clear call
8	00000000	Message Length	13
9	00001101		
1	00001000	Information Element ID	Cause
2	1-------	Ext	last octet
	-00-----	Coding Standard	ITU-T standard
	---0----	Flag	. not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	: 2
4	00000010		
5	1-------	Ext	last octet
	-000----	Spare	
	----0000	Location	user
6	1-------	Ext	last octet
	-0011111	Cause Value	. NE unspecified
1	01010100	Information Element ID	: Endpoint Reference
2	1-------	Ext	last octet
	-00-----	Coding Standard	: ITU-T standard
	---0----	Flag	not significant
	----0---	Reserved	reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	: 3
4	00000011		
5	00000000	Endpoint Reference Type	: locally defined integer
6	1-------	Endpoint Reference Flag	to the origination side
	-0000000	Endpoint Reference Value	: 1
6 1	00000001		

At this point in time, Leaves #2 and #3 have been disconnected, and only a point-to-point connection between the Root and Leaf #1 exists To terminate this last part, a RELEASE message is sent from the analyzer to the switch (Trace 12-7n). The Call Reference Value (5) is indicated as before, and a Cause IE states the reason for the call termination (normal call clearing). The switch returns a RELEASE COMPLETE message (Trace 12-7o) which completes the call disconnection sequence and allows the network resources to be used for subsequent connections.

Trace 12-7n. RELEASE message details

12.47 09 14326120 CPP 2 LIF.3 Tx UNI Sig. ATMF UNI 3 1

1	00001001	Protocol Discriminator	: Q 93B UNI call control
2	0000----	Spare	
	----0011	Call Reference Length	3
3	0-------	Call Reference Flag	from
	-0000000	Call Reference Value	5
4	00000000		
5	00000101		
6	01001101	Message Type	RELEASE
7	1-------	Ext	last octet
	-00-----	Spare	
	---0----	Flag	: not significant
	----00--	Spare	:
	------00	Action Indicator	: clear call
8	00000000	Message Length	: 6
9	00000110		
1	00001000	Information Element ID	Cause
2	1-------	Ext	: last octet
	-00-----	Coding Standard	ITU-T standard
	---0----	Flag	. not significant
	----0---	Reserved	. reserved
	-----000	Action Indicator	clear call
3	00000000	IE Length	2
4	00000010		
5	1-------	Ext	: last octet
	-000----	Spare	:
	----0000	Location	user
6	1-------	Ext	. last octet
	-0010000	Cause Value	NE normal call clearing

Trace 12-7o. RELEASE COMPLETE message details

12.47.09 14645750 CPP 2 LIF 3 Rx UNI Sig ATMF UNI 3 1

1	00001001	Protocol Discriminator	Q 93B UNI call control
2	0000----	Spare	
	----0011	Call Reference Length	3
3	1-------	Call Reference Flag	to
	-0000000	Call Reference Value	5
4	00000000		
5	00000101		
6	01011010	Message Type	RELEASE COMPLETE
7	1-------	Ext	last octet
	-00-----	Spare	
	---0----	Flag	not significant
	----00--	Spare	
	------00	Action Indicator	clear call
8	00000000	Message Length	0
9	00000000		

12.8 Transmitting TCP/IP Data over ATM

Our final case study looks at the transfer of application data over an ATM infrastructure In this example, a workstation initiates a TELNET connection with a host through an ATM switch (Figure 12-10). Note that the analyzer is monitoring only the data from the workstation to the switch, and does not capture any of the data in the opposite direction. For this reason, the protocol decode will indicate the "Rx" direction (workstation to switch) for the cells, and we never see any cells noted in the "Tx" direction.

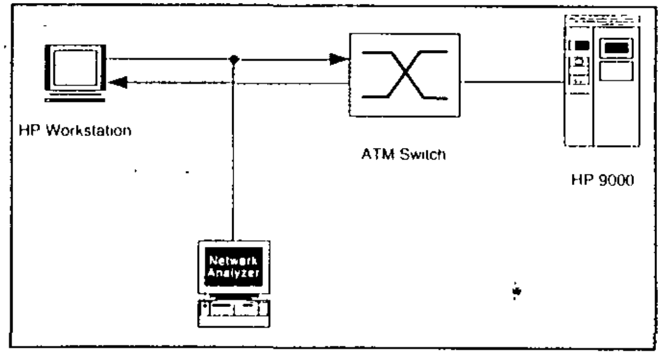

Figure 12-10. ATM and TCP/IP analysis.

The example includes nine different messages that are sent from the workstation to the switch (Trace 12-8a). Note that each of these messages requires two cells to carry the data, as indicated by the timestamps that are placed on the cells (see the left hand margin of Trace 12-8a). For example, the first message contains Address Resolution Protocol (ARP) information and requires two cells for transmission. The first cell arrived at 12:08:29.53426090, and the second cell arrived at 12:08:29.53427120. Note that the analyzer is able to completely decode the higher-layer information after the last (second) cell arrives. The next two lines of the trace indicate that AAL5 is being used to transport an ARP message. The ARP message is further decoded to identify its contents: the Logical Link Control (LLC) header, the Subnet Access Protocol (SNAP) header, and finally the ARP message The other messages from the workstation indicate that a TELNET connection is being established, TELNET parameters are being negotiated, and finally the TELNET login occurs.

Trace 12-8a. TELNET Connection over ATM (summary)

HP Broadband Series Tester Capture Data Record

```
12 08 29 53426090 Port 4·1 Rx ATM VPI/VCI = 0/101
12 08 29 53427120 Port 4 1 Rx ATM VPI/VCI = 0/101
12 08 29 53427120 Port 4 1 Rx AAL-5 Length = 68
12 08 29 53427120 Port 4 1 Rx ARP
          LLC   DSAP=0xAA SSAP=0xAA CTL=0x03
          SNAP  OUI=0x00 00-00 PID=0x0806
          ARP   Hardware=ATM Operation=Request Inverse
12 08 39 43636840 Port 4·1 Rx ATM VPI/VCI = 0/101
12 08 39 43637870 Port 4.1 Rx ATM VPI/VCI = 0/101
12 08 39 43637870 Port 4 1 Rx AAL-5 Length = 68
12 08 39 43637870 Port 4 1 Rx ARP
          LLC   DSAP=0xAA SSAP=0xAA CTL=0x03
          SNAP  OUI=0x00-00-00 PID=0x0806
          ARP   Hardware=ATM Operation=Reply Inverse
12 20 00 15308550 Port 4 1 Rx ATM VPI/VCI = 0/101
12.20 00 15309590 Port 4.1 Rx ATM VPI/VCI = 0/101
12 20 00 15309590 Port 4:1 Rx AAL-5 Length = 52
12 20 00 15309590 Port 4 1 Rx IP
          LLC   DSAP=0xAA SSAP=0xAA CTL=0x03
          SNAP  OUI=0x00-00-00 PID=0x0800
          IP    XXX YYY 125 4 -> XXX.YYY 125 3 Id=25b5
          TCP   telnet -> t2386 Flags= A S Seq=1092224001 Ack=1
12 20 00 20426240 Port 4.1 Rx ATM VPI/VCI = 0/101
12.20 00 20427400 Port 4·1 Rx ATM VPI/VCI = 0/101
12 20 00 20427400 Port 4 1 Rx AAL-5 Length = 48
12 20·00 20427400 Port 4·1 Rx IP
          LLC   DSAP=0xAA SSAP=0xAA CTL=0x03
          SNAP  OUI=0x00-00-00 PID=0x0800
          IP    XXX YYY 125 4 -> XXX.YYY 125 3 Id=25b6
          TCP   telnet -> t2386 Flags= A    Ack=1278080007
12 20 00 27929490 Port 4·1 Rx ATM VPI/VCI = 0/101
12 20 00.27930530 Port 4:1 Rx ATM VPI/VCI = 0/101
12 20 00 27930530 Port 4·1 Rx AAL-5 Length = 51
12 20 00 27930530 Port 4 1 Rx IP
          LLC   DSAP=0xAA SSAP=0xAA CTL=0x03
          SNAP  OUI=0x00-00-00 PID=0x0800
          IP    XXX YYY 125 4 -> XXX YYY 125 3 Id=25c9
          TCP   telnet -> t2386 Flags= AP   Seq=1092224002 Ack=1
          TELNET IAC DO TERMINAL TYPE
```



```
12 20 00 36361750 Port 4·1 Rx ATM VPI/VCI = 0/101
12 20 00 36362780 Port 4 1 Rx ATM VPI/VCI = 0/101
12 20 00 36362780 Port 4 1 Rx AAL-5 Length = 57
12 20 00 36362780 Port 4 1 Rx IP
          LLC   DSAP=0xAA SSAP=0xAA CTL=0x03
          SNAP  OUI=0x00-00-00 PID=0x0800
          IP    XXX YYY 125 4 -> XXX YYY 125 3 Id=25ca
          TCP   telnet -> t2386 Flags=.AP . Seq=1092224005 Ack=1
          TELNET IAC WILL SUPPRESS GO AHEAD <FF><FB>
12 20 00 38805570 Port 4:1 Rx ATM VPI/VCI = 0/101
12 20 00 38806610 Port 4:1 Rx ATM VPI/VCI = 0/101
12 20 00 38806610 Port 4·1 Rx AAL-5 Length = 86
12 20 00 38806610 Port 4·1 Rx IP
          LLC   DSAP=0xAA SSAP=0xAA CTL=0x03
          SNAP  OUI=0x00-00-00 PID=0x0800
          IP    XXX.YYY 125 4 -> XXX YYY 125 3 Id=25cb
          TCP   telnet -> t2386 Flags= AP... Seq=1092224014 Ack=1
          TELNET IAC WILL ECHO <FF><FB><01><FF><FD><01><0D><0A><0D>
12.20 00 39593080 Port 4 1 Rx ATM VPI/VCI = 0/101
12·20.00 39594110 Port 4:1 Rx ATM VPI/VCI = 0/101
12 20 00 39594110 Port 4·1 Rx AAL-5 Length = 51
12 20 00.39594110 Port 4 1 Rx IP
          LLC   DSAP=0xAA SSAP=0xAA CTL=0x03
          SNAP  OUI=0x00-00-00 PID=0x0800
          IP    XXX.YYY 125 4 -> XXX.YYY.125.3 Id=25cc
          TCP   telnet -> t2386 Flags=.AP . Seq=1092224052 Ack=1
          TELNET IAC DONT ECHO
12·20 00 56377560 Port 4 1 Rx ATM VPI/VCI = 0/101
12.20 00 56379630 Port 4 1 Rx ATM VPI/VCI = 0/101
12·20 00 56379630 Port 4 1 Rx AAL-5 Length = 55
12 20.00 56379630 Port 4:1 Rx IP
          LLC   DSAP=0xAA SSAP=0xAA CTL=0x03
          SNAP  OUI=0x00-00-00 PID=0x0800
          IP    XXX YYY 125.4 -> XXX.YYY.125.3 Id=25cd
          TCP   telnet -> t2386 Flags= AP. . Seq=1092224055 Ack=1
          TELNET login
```

The details of one of the workstation messages illustrate how one TELNET message is divided into two cells for transmission (Trace 12-8b) The information from the ATM layer is shown in the first two cells, beginning with the ATM headers. In order to reconstruct the TELNET message, the payloads of the two cells must be combined:

Payload (cell #1)	AA AA 03 00 00 00 08 00 45 00 00 2B 25 C9 00 00 3C 06 DE D3 XX YY 7D 04 XX YY 7D 03 00 17 09 52 41 1A 04 02 4C 2D F4 07 50 18 3F FA 4E E6 00 00
Payload (cell #2)	FF FD 18 6F 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 33 03 C8 DA B7

After the analyzer combines these two payloads, it can next begin to decode the various protocol layers contained within the data. For example, the first three octets are the LLC header: DSAP = 33, SSAP = 33, Control = 03. The next five octets are the SNAP header: OUI = 00 00 00, PID = 08 00. Next comes the IP header, beginning with the Version and Length (45), Type of Service (00), and continuing through and including the Destination Address, XXX.YYY.125.3 (XX YY 7D 03). The TCP header begins with the Source Port number, which identifies port number 23 for TELNET (00 17). The TCP header ends with the Checksum (4E E6) and the Urgent Pointer (00 00). The TELNET data is a simple option (FF FD 18), which is followed by 37 pad characters. The AAL5 trailer completes the reassembled message and includes the User-User indication (00), the Common Part Indicator (00), the length (00 33), and the CRC32 (03 C8 DA B7). Review Figure 11-14 to see the relative position and lengths of the AAL5 message, including the data, pad characters, and trailer fields.

Checking the lengths of all the fields, we note that this message consisted of two payloads of total length 96 octets (48 + 48 = 96). This payload could be further subdivided into the LLC header (3 octets), the SNAP header (5 octets), the IP header (20 octets), the TCP header (20 octets), and the TELNET data (3 octets), which yield 51 octets (the total length). When the padding (37 octets) and the AAL5 trailer (8 octets) are added, the total combined payload of 96 octets is realized. We have thus accounted for all of the headers and data within the two cells that were carrying the TELNET information.

Trace 12-8b. TELNET Connection over ATM (details)

12 20 00 27929490 Port 4 1 Rx ATM

Header:	Generic Flow Control	0
	Virtual Path Identifier	0
	Virtual Channel Identifier	101
	Payload Type	0 (User Data, No Congest, UserInd=0)
	Cell Loss Priority	0 (Higher Priority)
	Header Error Control	0x9C

Payload	AA AA 03 00 00 00 08 00 45 00 00 2B 25 C9 00 00 3C 06 DE D3 XX YY 7D 04 XX YY 7D 03 00 17 09 52 41 1A 04 02 4C 2D F4 07 50 18 3F FA 4E E6 00 00

12 20 00 27930530 Port 4 1 Rx ATM

Header:	Generic Flow Control	0
	Virtual Path Identifier	0
	Virtual Channel Identifier	101
	Payload Type	1 (User Data, No Congest, UserInd=1)
	Cell Loss Priority	0 (Higher Priority)
	Header Error Control	0x92
Payload	FF FD 18 6F 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 33 03 C8 DA B7	

12 20 00.27930530 Port 4 1 Rx AAL-5

Payload:	AA AA 03 00 00 00 08 00 45 00 00 2B 25 C9 00 00 3C 06 DE D3 XX YY 7D 04 XX YY 7D 03 00 17 09 52 41 1A 04 02 4C 2D F4 07 50 18 3F FA 4E E6 00 00 FF FD 18	
Pad Characters	0x6F 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00	
Trailer:	User-User Indication	0x00
	Common Part Indicator	0x00
	Length	51
	CRC32	0x03C8DAB7

12 20 00 27930530 Port 4:1 Rx IP

LLC	DSAP	0xAA (SNAP-SAP)
	SSAP	0xAA (SNAP-SAP)
	Control	0x03 (Unnumbered Information)
SNAP:	OUI	0x00-00-00 (Ethertype)
	PID	0x0800 (Internet Protocol)
IP	Version = 4	
	Header length = 20	
	Type of service = Routine(0)	
	Delay = Normal (0)	
	Throughput = Normal (0)	
	Reliability = Normal (0)	

```
        Packet length = 43
        Id = 25c9
        Fragment offset = 0
        Flags = [Don't Fragment = 0][More = 0]
        Time to live = 60
        Protocol = TCP (6)
        Header checksum = DED3
        Source address = XXX YYY 125.4
        Destination address = XXX YYY.125 3
TCP     Source port = telnet (23)
        Destination port = 2386
        Sequence number = 1092224002
        Ack number = 1278080007
        Data offset = 20
        Flags = [URG=0][ACK=1][PUSH=1][RST=0][SYN=0][FIN=0]
        Window = 16378
        Checksum = 4EE6
        Urgent pointer = 00000000
TELNET. Interpret as Command (IAC)
        DO use option TERMINAL TYPE
```

12.9 Possible ATM Error Conditions

Below are some general guidelines to help you analyze the ATM layer, AAL3/4, and AAL5 protocols [12-4] and [12-5]. Refer to Figures 11-4a, 11-4b, 11-11, 11-13a, 11-13b, and 11-14, respectively, as you study the following sections.

12.9.1 ATM Layer Analysis Guidelines

If the cell is assigned, check that the VCI on the user side of the UNI is not zero (unassigned cells will have a VCI of zero). Verify that the HEC is correct.

For ATM Layer Management PDUs (OAM cells), look for:

- an invalid OAM cell type
- an invalid OAM function type
- an invalid OAM CRC-12

12.9.2 AAL3/4 CPCS Analysis Guidelines

- Verify that the value of the CPI field is not zero
- Verify consistency between the BTag and ETag values.
- Verify that the BAsize field is large enough to contain the PDU.
- Verify that the size of the PDU is not less than the minimum (8 octets) or larger than the maximum (65,544 octets). Note that the maximum is derived by adding the payload (65,535 octets), the pad (1 octet), the header (4 octets), and the trailer (4 octets), which aligns the PDU to a multiple of 4 octets.
- Verify that the Pad length is correct.
- Verify that the value of the Alignment field is zero.

12.9.3 AAL3/4 SAR Analysis Guidelines

- The Segment Type field should not have an unexpected BOM, a COM before a BOM, or an EOM before a BOM.
- Verify that the Sequence Number is correct.
- Look for an Abort_SAR_PDU from I.363 Section 4.3.1.2.2, which terminates the reassembly process but does not start a new reassembly process. The Abort_SAR_PDU is coded with an ST = EOM, Payload = 0, and Length = 63.
- Some errors are similar to those we discussed in Section 9.5 for SIP Layer 2 and are described in I.363. These include: too many reassemblies, reassembly timeout, and reassembly length overrun. For the "too many reassemblies" error, the number of reassemblies is a negotiated parameter, with a default of one. (Recall that for SIP Layer 2, one or 16 concurrent reassemblies are supported.)
- Analysis guidelines for the Length Indication field:
  - The value must be a multiple of 4 octets.
  - The value must be 44 octets for BOM and COMs.
  - The value must be between 4 and 44 octets for EOMs.
  - The value must be between 8 and 44 octets for SSMs.
  - Verify that the CRC-10 value is correct.

9

FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

P.C.M.
ROUTING BASICS
(ANEXO)

EXPOSITOR: M. en C. ERNESTO QUIROZ MORONES
PALACIO DE MINERÍA
JUNIO DE 1999

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CHAPTER 2

Routing Basics

Background

Routing is moving information across an internetwork from source to destination. Along the way, at least one intermediate node is typically encountered. Routing is often contrasted with bridging which seems to accomplish precisely the same thing. The primary difference between the two is that bridging occurs at Layer 2 (the link layer) of the OSI reference model, while routing occurs at Layer 3 (the network layer). This distinction provides routing and bridging with different information to use in the process of moving information from source to destination. As a result, routing and bridging accomplish their tasks in different ways and, in fact, there are several different kinds of routing and bridging. For more information on bridging, see Chapter 3, "Bridging Basics."

The topic of routing has been covered in computer science literature for over two decades, but routing only achieved commercial popularity in the mid-1980s. The primary reason for this time lag is the nature of networks in the 1970s. During this time, networks were fairly simple, homogeneous environments. Only recently has large-scale internetworking become popular.

Routing Components

Routing involves two basic activities: determination of optimal routing paths and the transport of information groups (typically called *packets*) through an internetwork. In this publication, the latter of these is referred to as *switching*. Switching is relatively straightforward. Path determination, on the other hand, can be very complex.

Path Determination

A *metric* is a standard of measurement—for example, path length—that is used by routing algorithms to determine the optimal path to a destination. To aid the process of path determination, routing algorithms initialize and maintain *routing tables*, which contain route information. Route information varies depending on the routing algorithm used.

Routing algorithms fill routing tables with a variety of information. Destination/next hop associations tell a router that a particular destination can be gained optimally by sending the packet to a particular router representing the "next hop" on the way to the final destination. When a router receives an incoming packet, it checks the destination address and attempts to associate this address with a next hop. Figure 2-1 shows an example of a destination/next hop routing table.

Figure 2-1 Destination/Next Hop Routing Table

To reach network:	Send to:
27	Node A
57	Node B
17	Node C
24	Node A
52	Node A
16	Node B
26	Node A
. .	.

S1283a

Routing tables can also contain other information, such as information about the desirability of a path. Routers compare metrics to determine optimal routes. Metrics differ depending on the design of the routing algorithm being used. A variety of common metrics will be introduced and described later in this chapter.

Routers communicate with one another (and maintain their routing tables) through the transmission of a variety of messages. The *routing update* message is one such message. Routing updates generally consist of all or a portion of a routing table. By analyzing routing updates from all routers, a router can build a detailed picture of network topology. A *link-state advertisement* is another example of a message sent between routers. Link-state advertisements inform other routers of the state of the sender's links. Link information can also be used to build a complete picture of network topology. Once the network topology is understood, routers can determine optimal routes to network destinations.

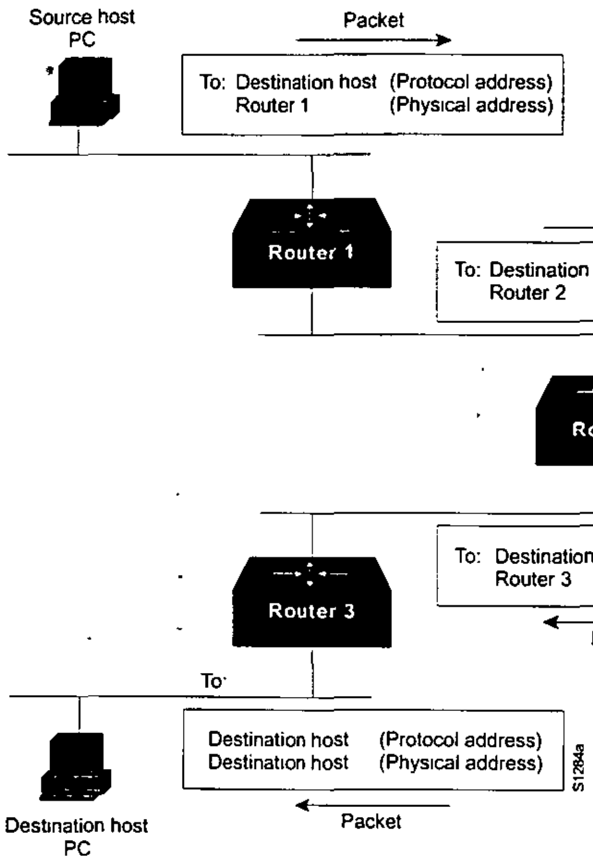
Switching

Switching algorithms are relatively simple and are basically the same for most routing protocols. In most cases, a host determines that it must send a packet to another host. Having acquired a router's address by some means, the source host sends a packet addressed specifically to a router's physical (Media Access Control [MAC]-layer) address, but with the protocol (network-layer) address of the destination host.

On examining the packet's destination protocol address, the router determines that it either knows or does not know how to forward the packet to the next hop. If the router does not know how to forward the packet, it typically drops the packet. If the router knows how to forward the packet, it changes the destination physical address to that of the next hop and transmits the packet.

The next hop may or may not be the ultimate destination host. If not, the next hop is usually another router, which executes the same switching decision process. As the packet moves through the internetwork, its physical address changes but its protocol address remains constant. This process is illustrated in Figure 2-2.

Figure 2-2 Switching Process



The preceding discussion describes switching between a source and a destination end system. The International Organization for Standardization (ISO) has developed a hierarchical terminology that is useful in describing this process. Using this terminology, network devices without the ability to forward packets between subnetworks are called *end systems* (ESs), while network devices with these capabilities are referred to as *intermediate systems* (ISs). ISs are further divided into those that can communicate within routing domains (*intradomain ISs*) and those that communicate both within and between routing domains (*interdomain ISs*). A *routing domain* is generally considered to be a portion of an internetwork under common administrative authority, regulated by a particular set of administrative guidelines. Routing domains are also called *autonomous systems*. With certain protocols, routing domains can also be divided into *routing areas*, but intradomain routing protocols are still used for switching both within and between areas.

Routing Algorithms

Routing algorithms can be differentiated based on several key characteristics. First, the particular goals of the algorithm designer affect the operation of the resulting routing protocol. Second, there are various types of routing algorithms. Each algorithm has a different impact on network and router resources. Finally, routing algorithms use a variety of metrics that affect calculation of optimal routes. The following sections analyze these routing algorithm attributes.

Design Goals

Routing algorithms often have one or more of the following design goals:

- Optimality
- Simplicity and low overhead
- Robustness and stability
- Rapid convergence
- Flexibility

Optimality

Optimality refers to the ability of the routing algorithm to select the "best" route. The best route depends on the metrics and metric weightings used to make the calculation. For example, one routing algorithm might use number of hops and delay, but might weight delay more heavily in the calculation. Naturally, routing protocols must strictly define their metric calculation algorithms.

Simplicity

Routing algorithms are also designed to be as simple as possible. In other words, the routing algorithm must offer its functionality efficiently, with a minimum of software and utilization overhead. Efficiency is particularly important when the software implementing the routing algorithm must run on a computer with limited physical resources.

Robustness

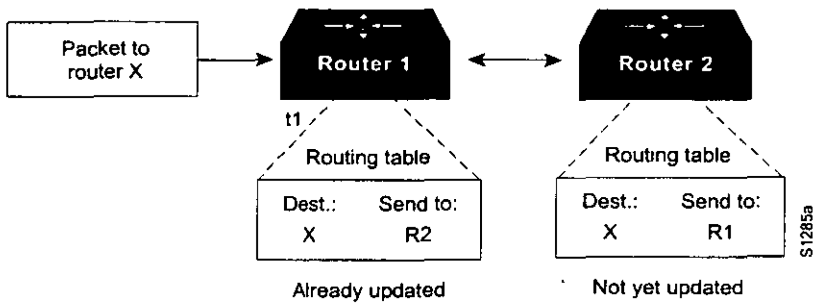
Routing algorithms must be robust. In other words, they should perform correctly in the face of unusual or unforeseen circumstances such as hardware failures, high load conditions, and incorrect implementations. Because routers are located at network junction points, they can cause considerable problems when they fail. The best routing algorithms are often those that have withstood the test of time and proven stable under a variety of network conditions.

Rapid Convergence

Routing algorithms must converge rapidly. Convergence is the process of agreement, by all routers, on optimal routes. When a network event causes routes to either go down or become available, routers distribute routing update messages. Routing update messages permeate networks, stimulating recalculation of optimal routes and eventually causing all routers to agree on these routes. Routing algorithms that converge slowly can cause routing loops or network outages.

Figure 2-3 shows a routing loop. In this case, a packet arrives at Router 1 at time t1. Router 1 has already been updated and so knows that the optimal route to the destination calls for Router 2 to be the next stop. Router 1 therefore forwards the packet to Router 2. Router 2 has not yet been updated and so believes that the optimal next hop is Router 1. Router 2 therefore forwards the packet back to Router 1. The packet will continue to bounce back and forth between the two routers until Router 2 receives its routing update or until the packet has been switched the maximum number of times allowed.

Figure 2-3 Slow Convergence and Routing Loops



Flexibility

Routing algorithms should also be flexible. In other words, routing algorithms should quickly and accurately adapt to a variety of network circumstances. For example, assume that a network segment has gone down. Many routing algorithms, on becoming aware of this problem,,will quickly select the next-best path for all routes normally using that segment. Routing algorithms can be programmed to adapt to changes in network bandwidth, router queue size, network delay, and other variables.

Types

Routing algorithms can be classified by type. For example, algorithms can be:

- Static or Dynamic
- Single-Path or Multipath
- Flat or Hierarchical
- Host-Intelligent or Router-Intelligent
- Intradomain or Interdomain
- Link State or Distance Vector

Static or Dynamic

Static routing algorithms are hardly algorithms at all. Static routing table mappings are established by the network administrator prior to the beginning of routing. They do not change unless the network administrator changes them. Algorithms that use static routes are simple to design and work well in environments where network traffic is relatively predictable and network design is relatively simple.

Because static routing systems cannot react to network changes, they are generally considered unsuitable for today's large, constantly changing networks. Most of the dominant routing algorithms in the 1990s are dynamic.

Dynamic routing algorithms adjust, in real time, to changing network circumstances. They do this by analyzing incoming routing update messages. If the message indicates that a network change has occurred, the routing software recalculates routes and sends out new routing update messages. These messages permeate the network, stimulating routers to rerun their algorithms and change their routing tables accordingly.

Dynamic routing algorithms may be supplemented with static routes where appropriate. For example, a *router of last resort* (a router to which all unroutable packets are sent) may be designated. This router acts as a repository for all unroutable packets, ensuring that all messages are at least handled in some way.

Single-Path or Multipath

Some sophisticated routing protocols support multiple paths to the same destination. These multipath algorithms permit traffic multiplexing over multiple lines; single-path algorithms do not. The advantages of multipath algorithms are obvious; they can provide substantially better throughput and reliability.

Flat or Hierarchical

Some routing algorithms operate in a flat space, while others use routing hierarchies. In a flat routing system, all routers are peers of all others. In a hierarchical routing system, some routers form what amounts to a routing backbone. Packets from nonbackbone routers travel to the backbone routers, where they are sent through the backbone until they reach the general area of the destination At this point, they travel from the last backbone router through one or more nonbackbone routers to the final destination.

Routing systems often designate logical groups of nodes called domains, autonomous systems, or areas. In hierarchical systems, some routers in a domain can communicate with routers in other domains, while others can only communicate with routers within their domain. In very large networks, additional hierarchical levels may exist. Routers at the highest hierarchical level form the routing backbone.

The primary advantage of hierarchical routing is that it mimics the organization of most companies and therefore supports their traffic patterns very well. Most network communication occurs within small company groups (domains). Intradomain routers only need to know about other routers within their domain, so their routing algorithms can be simplified. Depending on the routing algorithm being used, routing update traffic can be reduced accordingly.

Host-Intelligent or Router-Intelligent

Some routing algorithms assume that the source end-node will determine the entire route. This is usually referred to as *source routing*. In source-routing systems, routers merely act as store-and-forward devices, mindlessly sending the packet to the next stop.

Other algorithms assume that hosts know nothing about routes. In these algorithms, routers determine the path through the internetwork based on their own calculations. In the first system, the hosts have the routing intelligence. In the latter system, routers have the routing intelligence.

The trade-off between host-intelligent and router-intelligent routing is one of path optimality versus traffic overhead. Host-intelligent systems choose the better routes more often, because they typically discover all possible routes to the destination before the packet is actually sent. They then choose the best path based on that particular system's definition of optimal. The act of determining all routes, however, often requires substantial discovery traffic and a significant amount of time.

Intradomain or Interdomain

Some routing algorithms work only within domains; others work within and between domains. The nature of these two algorithm types is different. It stands to reason, therefore, that an optimal intradomain routing algorithm would not necessarily be an optimal interdomain routing algorithm.

Link State or Distance Vector

Link state algorithms (also known as *shortest path first* algorithms) flood routing information to all nodes in the internetwork. However, each router sends only that portion of the routing table that describes the state of its own links. Distance vector algorithms (also known as *Bellman-Ford* algorithms) call for each router to send all or some portion of its routing table, but only to its neighbors. In essence, link state algorithms send small updates everywhere, while distance vector algorithms send larger updates only to neighboring routers.

Because they converge more quickly, link state algorithms are somewhat less prone to routing loops than distance vector algorithms. On the other hand, link state algorithms require more CPU power and memory than distance vector algorithms. Link state algorithms can therefore be more expensive to implement and support. Despite their differences, both algorithm types perform well in most circumstances.

Metrics

Routing tables contain information used by switching software to select the best route. But how, specifically, are routing tables built? What is the specific nature of the information they contain? How do routing algorithms determine that one route is preferable to others?

Routing algorithms have used many different metrics to determine the best route. Sophisticated routing algorithms can base route selection on multiple metrics, combining them in a single (hybrid) metric. All of the following metrics have been used:

- Path Length
- Reliability
- Delay
- Bandwidth
- Load
- Communication Cost

Path Length

Path length is the most common routing metric. Some routing protocols allow network administrators to assign arbitrary costs to each network link. In this case, path length is the sum of the costs associated with each link traversed. Other routing protocols define *hop count*, a metric that specifies the number of passes through internetworking products (such as routers) that a packet must take en route from a source to a destination.

Reliability

Reliability, in the context of routing algorithms, refers to the reliability (usually described in terms of the bit-error rate) of each network link. Some network links may go down more often than others. Once down, some network links may be repaired more easily or more quickly than other links. Any reliability factors can be taken into account in the assignment of reliability ratings. Reliability ratings are usually assigned to network links by network administrators. They are typically arbitrary numeric values.

Delay

Routing delay refers to the length of time required to move a packet from source to destination through the internetwork. Delay depends on many factors, including the bandwidth of intermediate network links, the port queues at each router along the way, network congestion on all intermediate network links, and the physical distance to be travelled. Because it is a conglomeration of several important variables, delay is a common and useful metric.

Bandwidth

Bandwidth refers to the available traffic capacity of a link. All other things being equal, a 10-Mbps Ethernet link would be preferable to a 64-kbps leased line. Although bandwidth is a rating of the maximum attainable throughput on a link, routes through links with greater bandwidth do not necessarily provide better routes than routes through slower links. If, for example, a faster link is much busier, the actual time required to send a packet to the destination may be greater through the fast link.

Load

Load refers to the degree to which a network resource (such as a router) is busy. Load can be calculated in a variety of ways, including CPU utilization and packets processed per second. Monitoring these parameters on a continual basis can itself be resource intensive.

Communication Cost

Communication cost is another important metric. Some companies may not care about performance as much as they care about operating expenditures. Even though line delay might be longer, they will send packets over their own lines rather than through public lines that will cost money for usage time.

Routed vs. Routing Protocols

Confusion about the terms *routed* protocol and *routing* protocol is common. Routed protocols are protocols that are routed over an internetwork. Examples of such protocols are the *Internet Protocol* (IP), *DECnet*, *AppleTalk*, *NetWare*, *OSI*, *Banyan VINES*, and *Xerox Network System* (XNS). Routing protocols are protocols that implement routing algorithms. Put simply, they route routed protocols through an internetwork. Examples of these protocols include *Interior Gateway Routing Protocol* (IGRP), *Enhanced Interior Gateway Routing Protocol* (EIGRP), *Open Shortest Path First* (OSPF), *Exterior Gateway Protocol* (EGP), *Border Gateway Protocol* (BGP), *OSI Routing*, *Advanced Peer-to-Peer Networking*, *Intermediate System to Intermediate System* (IS-IS), and *Routing Information Protocol* (RIP). Routed and routing protocols are discussed in detail later in this publication.

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CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

APLICACIONES ATM EN MÉXICO
CONSIDERACIONES PARA UN PROYECTO DE INGENIERÍA
DRAF

EXPOSITOR: ING. ARTURO ALALUF OLIVARES
PALACIO DE MINERÍA
JUNIO DE 1999

Palacio de Minería Calle de Tacuba 5 Primer piso Deleg. Cuauhtémoc 06000 México, D.F. tel.: 521-40-20 Apdo Postal M-2285

Aplicaciones ATM en México

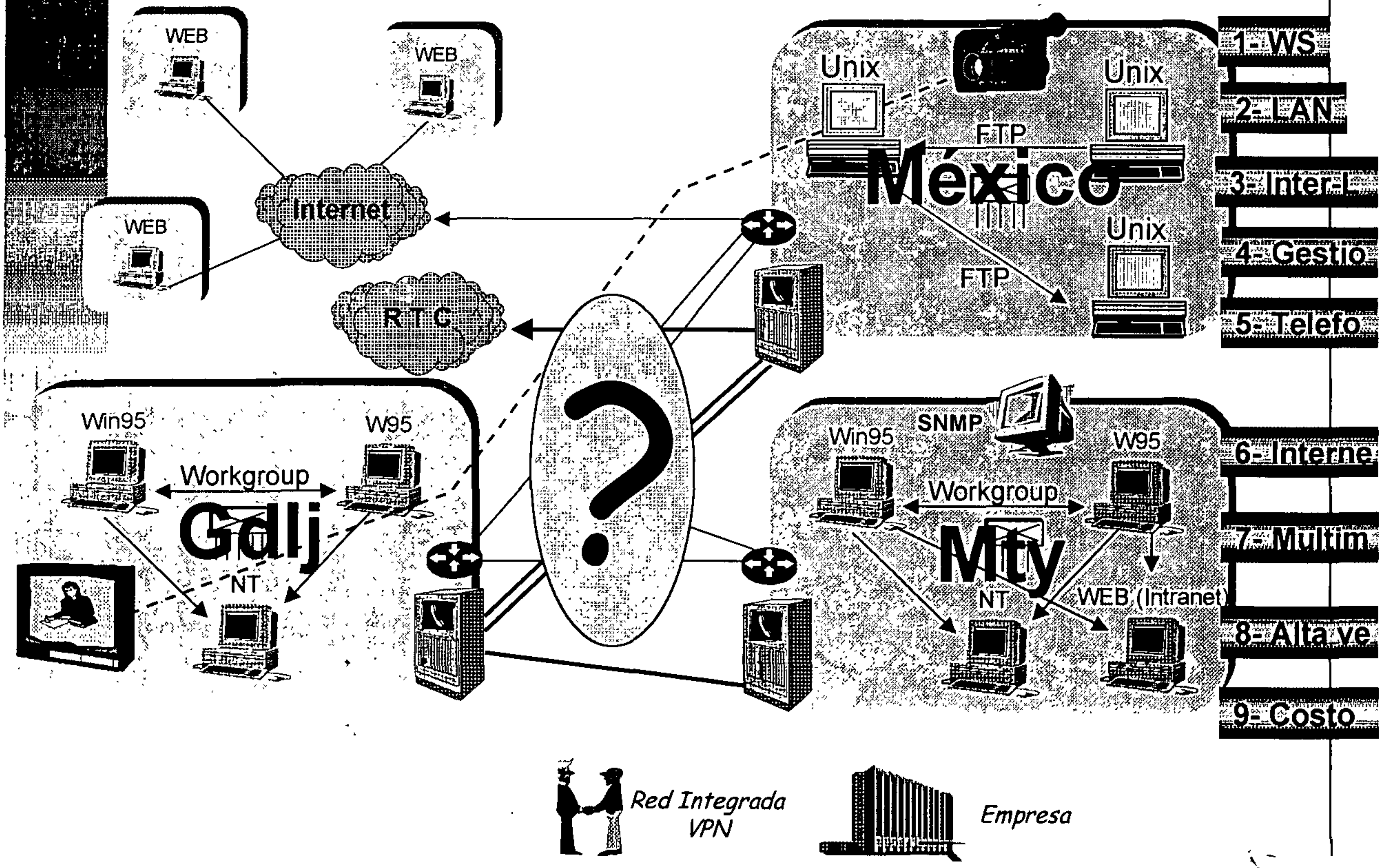
Consideraciones para un Proyecto de Ingeniería

DRAFT

May 15,1999

DRAFT

stemas de información
itan comunicarse entre sí?



ATM, una técnica prometedora

Ventajas - Puntos débiles

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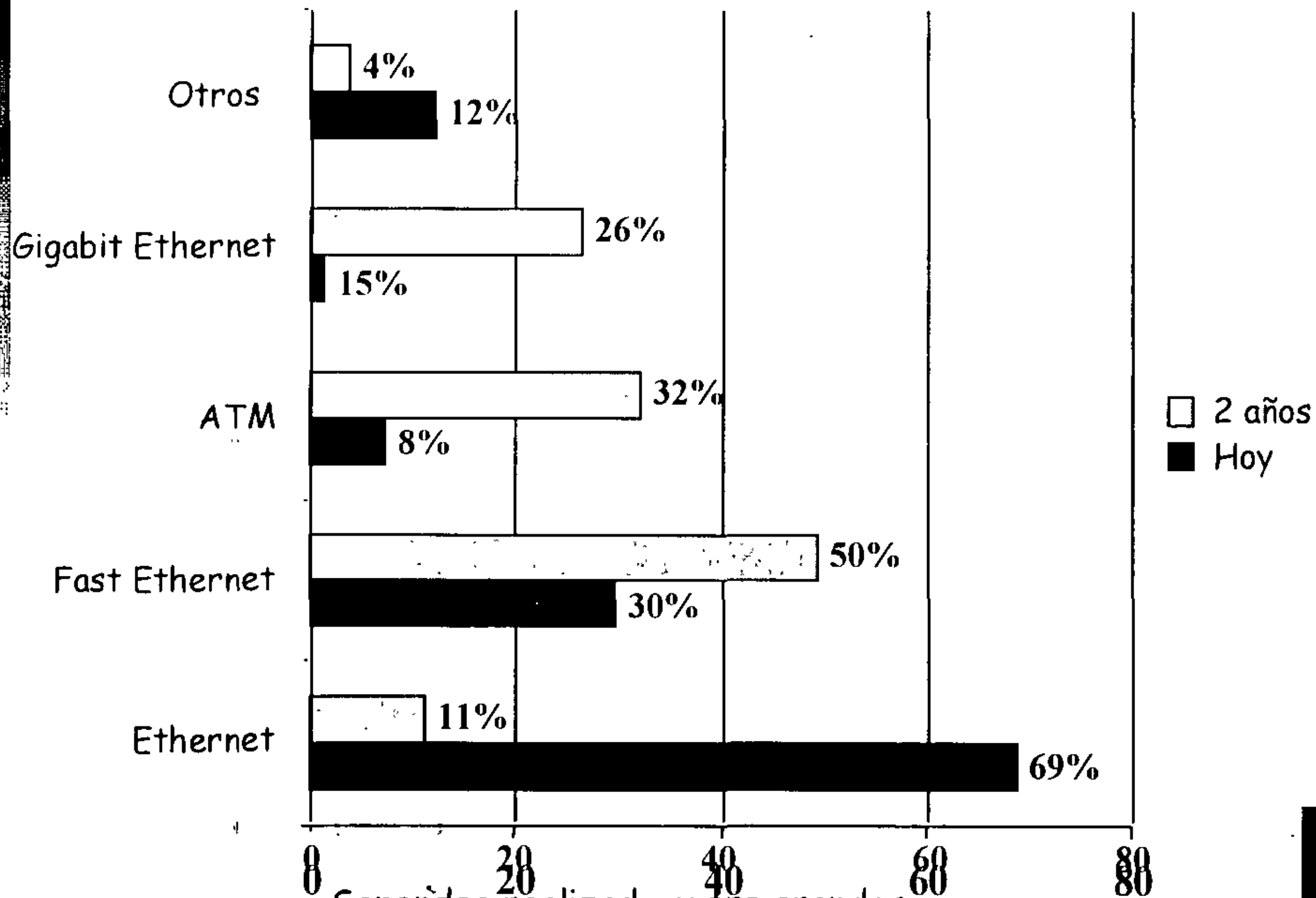
Mercado del LAN « Backbone »
Proporción de utilización por tecnología



Un mercado ATM en fuerte crecimiento

Sonondeo realizado entre grandes empresas

Source: CNS Registrations, 1998

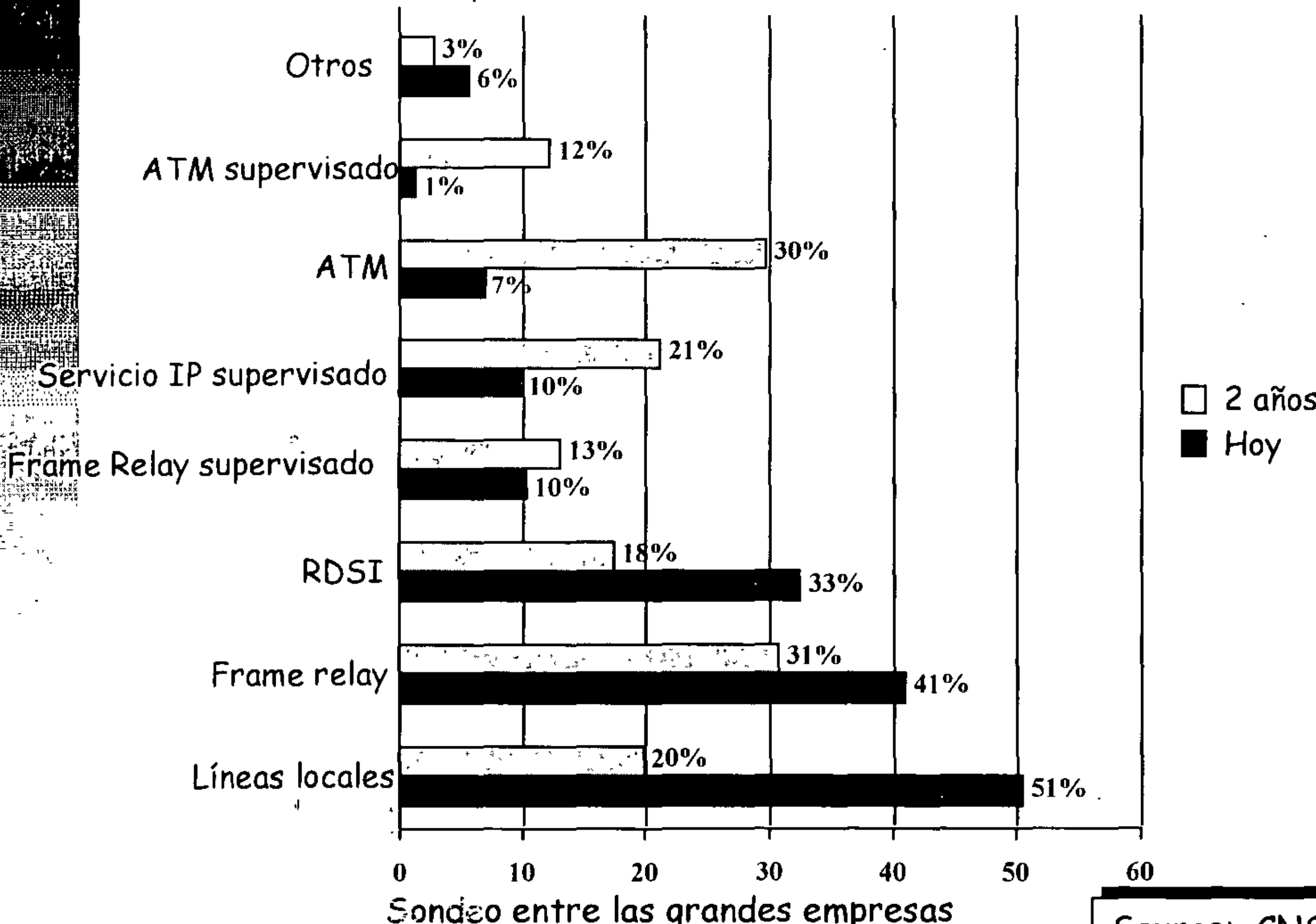
Operadores

Integradores potenciales

Empresas

DRAFT

...ado del WAN « Backbone »
...orción de utilización por tecnología



Un mercado ATM en fuerte crecimiento

THE YANKEE GROUP

Source: CNS Registrations, 1998

Operadores

Intéegradores Patenciales

Empresas

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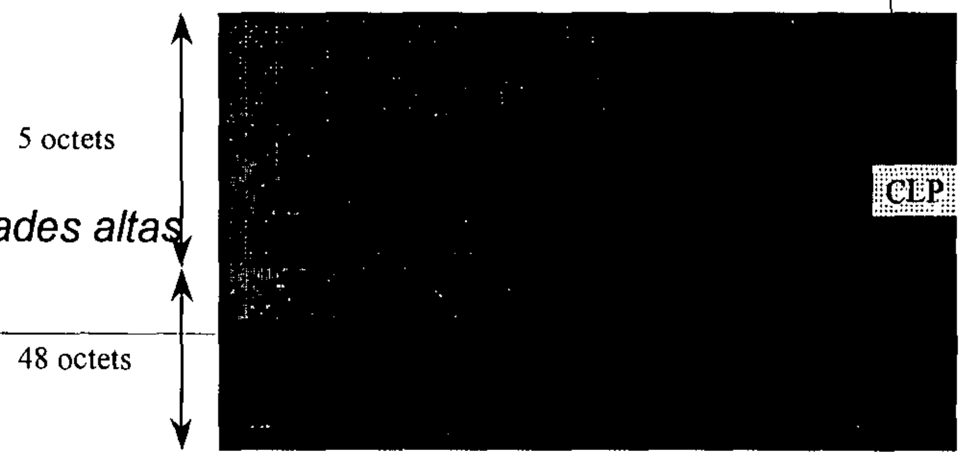
ATM :una tecnología adecuada para altas velocidades

- ***paquetes de tamaño fijo:*** *células de 53 octetos*
  - *La implementación hardware es totalmente factible*

- ***modo conectado***
  - *adecuado para velocidades altas*



ATM : tecnología adecuada para perfiles de tráfico muy diferentes - Contratos de tráfico

- ***Principio :***
  - *el usuario declara los descriptores de tráfico de la conexión: define el contrato de tráfico*
  - *la red decide si la conexión puede ser aceptada*
    - *Connection Admission Control* ***(CAC)***
  - *si acepta la conexión , la red se compromete en un nivel de calidad de servicio*

DRAFT

ATM : tecnologia adecuada para perfiles de trafico muy diferentes - Contratos de trafico

- ***una vez que es aceptada la conexión, la red verifica la conformidad del tráfico con el contrato***
    - *Usage Parameter Control **(UPC)***
    - *La red puede descartar celdas (células) no conformes con el contrato de tráfico definido (con o sin tagging -bit CLP)*

Distintos contratos de tráfico (ATC)

- ***Contratos con velocidad constante***
  - ***DBR*** *(Deterministic Bit Rate) o CBR (Constant Bit Rate)*
  - *el Peak Cell Rate es garantizado*
- ***Contratos con velocidad variable***
  - ***SBR*** *(Statistical Bit Rate) o VBR (Variable Bit Rate)*
  - *se especifican y se garantizan una velocidad máxima y una velocidad promedio*
  - *posibilidad para el operador de red de gestionar su red con multiplex estadístico*

Distintos ***contratos de tráfico (ATC)***

- ***Contratos con control en bucle (loop)***
  - ***ABR*** *(Available Bit Rate) para fuentes capaces de adaptar su velocidad en función de la disponibilidad de los recursos de la red (Ressource management cells)*

- ***Contratos de tipo Best Effort***
  - ***UBR*** *(Unspecified Bit Rate)*

Con distintos niveles de calidad

- *Cada tipo de tráfico está asociado con un compromiso de nivel de calidad:*
  - *en términos de integridad del mensaje (CLR) Cell Loss Rate*
  - *en cuanto al tiempo de tránsito o retardo (CTD) Cell Time Delay*

… …. lo cual permite manejar distintos niveles de precios

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Compatibilidad con los sistemas de existentes y los servicios

- ***Posibilidad de mapeo de ATM sobre sistemas de transmision PDH o SDH***
- ***homogeneidad del transporte de la información***
  - ***para las necesidades de los servicios :*** *soporte de voz, de video, de datos , de IP ....*
  - ***para adecuarse a las necesidades de gestión de red:*** *mecanismos de aceptación de conexión en función de los recursos de la red, del contrato de servicio del cliente (CAC y policing)*
  - ***coexistencia entre los mecanismos de protección*** *de la capa de transporte y los mecanismos de reenrutamiento dinámicos ATM.*

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Compatibilidad con los sistemas de existentes y los servicios

- ***flexibilidad en las velocidades ofrecidas***
  - *ofrece granularidad fina dede algunos kbit/s hasta centenas de Mbit/s*
  - *un solo nivel de multiplexado*
  - *multiplexado estadístico*
- ***flexibilidad temporal***
  - *conexiones permanentes*
  - *conexiones permanentes con posibilidad de modificar el ancho de banda en función de un calendario (périodico o no)*
  - *conexión inmediata sobre pedido (PVP /PVC a partir del sistema de administración de red) o por conmutación (por señalización SVC)*

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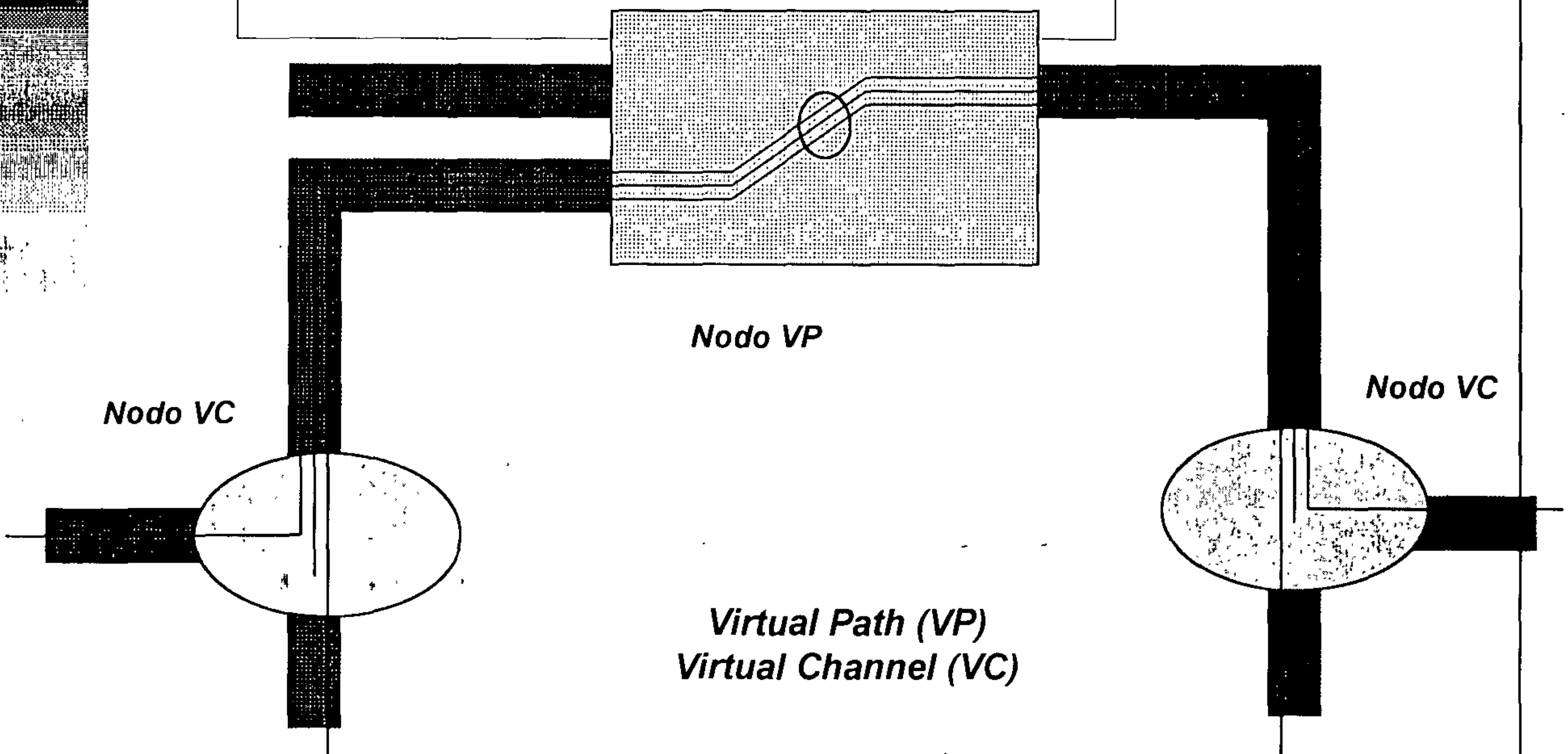
- *ATM : flexibilidad para la gestión de las conexiones*



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Equipos ATM

[illegible]	crosconect VP	Crossconects de VC
[illegible]	Conmutador VP	Conmutador VC

- *Proveedores de dos orígenes distintos*
  - *Proveedores de los operadores de telecomunicaciones : Ejm..Ericsson, Alcatel, Siemens, Lucent...*
  - *Proveedores del mundo de la data y redes privadas : Newbridge, GDC, Ascend, Cisco...*
- *Convergencia entre crossconects y equipo de conmutación en un solo equipo multiplexor/conmutador de VP/VC*
  - *Matriz de conmutación de 1 Gbit/s para los equipos del cliente (CPE) y de varias decenas de Gbits para los equipos de los Operadores.*

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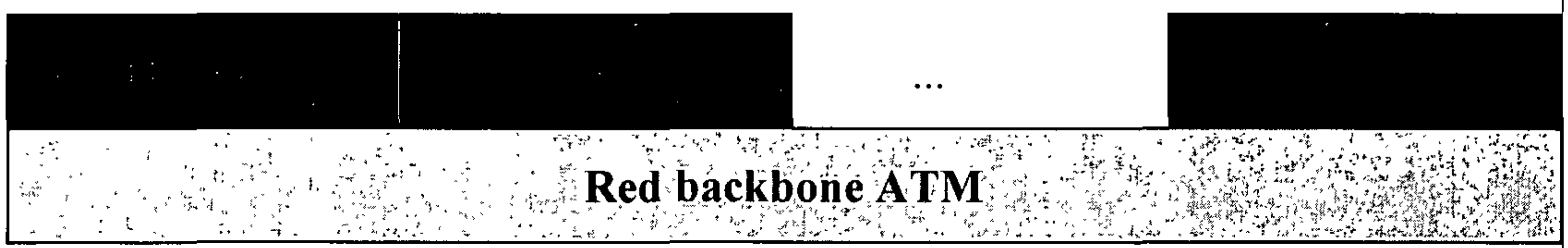
Hacia una red integrada de banda ancha

ATM es la tecnología adecuada para construir una red integrada, única capaz de soportar diferentes servicios (Datos, Video, Voz...)



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ATM, una técnica prometedora

Ventajas - Puntos débiles

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Debilidades del ATM

- *Complejidad*
- *Normalización « lenta » (ITU-T . ATM Forum)--> las necesidades de interfuncionamiento entre las redes públicas de telecomunicaciones y las redes de datos ocupará una buena parte del esfuerzo de normalización.*
- *No existe un « estandar de facto »*
- *La implantación de la conmutación de voz en forma masiva (usando SVC´s) aún tomará tiempo. El asunto se resolverá con potencia y velocidad de procesamiento. (No obstante los equipos actuales logran ya establecer o levantar de 1000 a 3000 SVC´s facturados por segundo lo cual significa que ya es tiempo de considerar ésta tecnología en las redes telefónicas para un número importante de « llamadas »)*

Debilidades del ATM

- ***Otras tecnologias reconocen los méritos del ATM y tratan introducir los mismos conceptos***
  - *calidad diferenciada*
  - *calidad garantizada*
  - *red multi servicios (datos, voz, video)*

- ***Por lo cual evolucionan también hacia una mayor complejidad ...y costos*** *(IP con QoS….)*

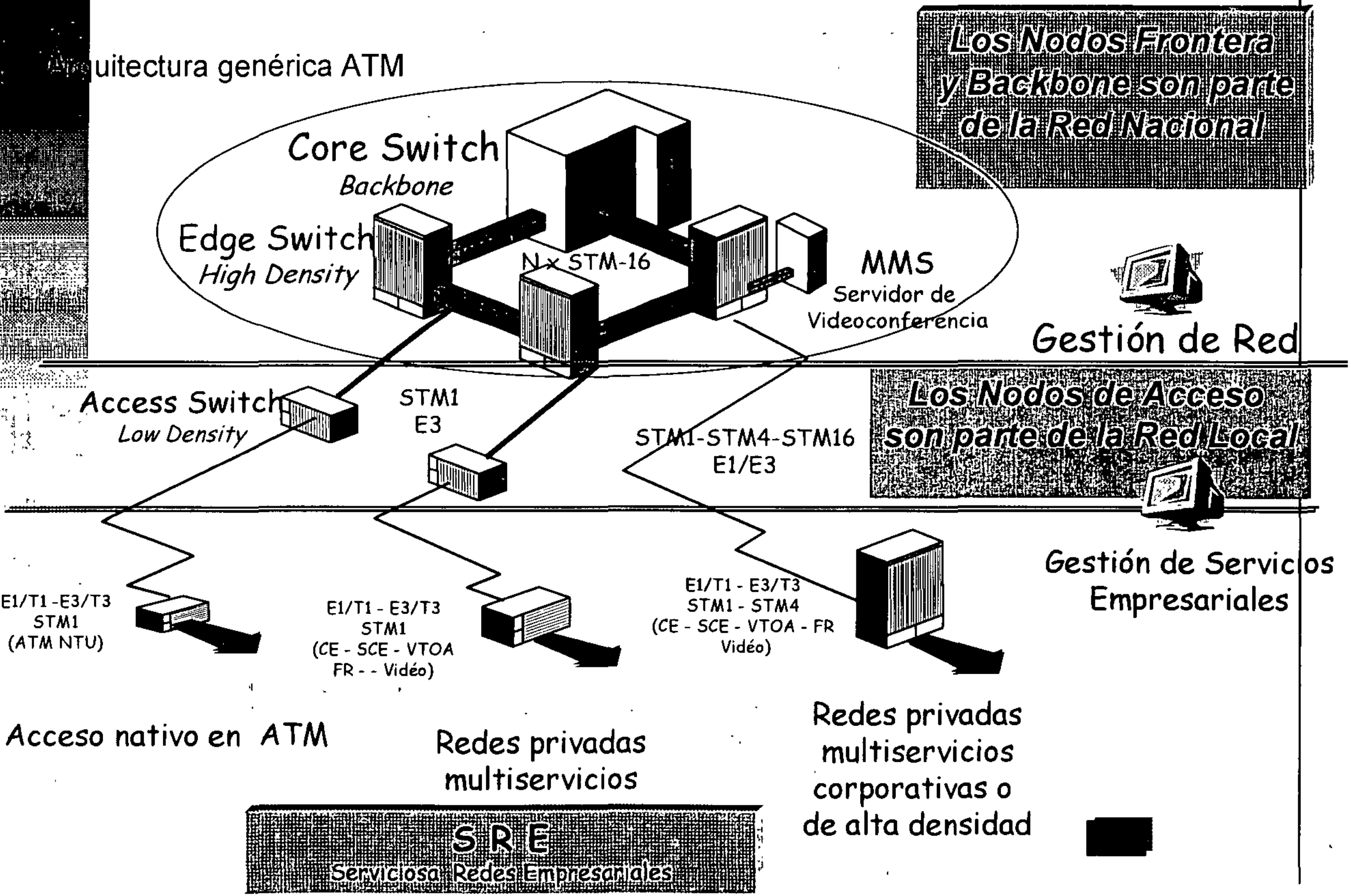
Arquitectura Genérica de una Red ATM para la convergencia de las Redes Telefónicas con las Redes de Datos

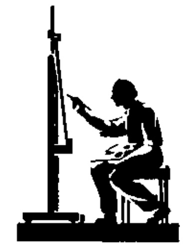
Servicios - Arquitectura - Equipos

Gestión y Explotación

DRAFT

Arquitectura genérica ATM



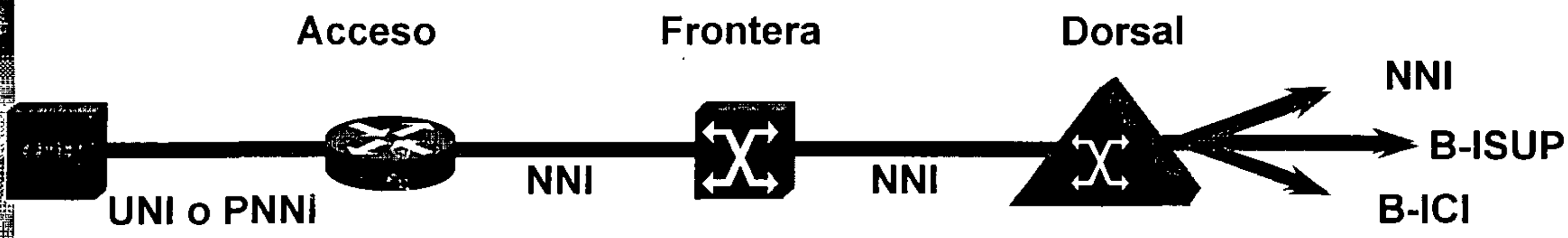


Escenarios de interfuncionamiento entre Redes Telefónicas y Redes de Datos utilizando ATM como integrador de tecnologías

Servicios - Arquitectura - Equipos

Gestión y Explotación

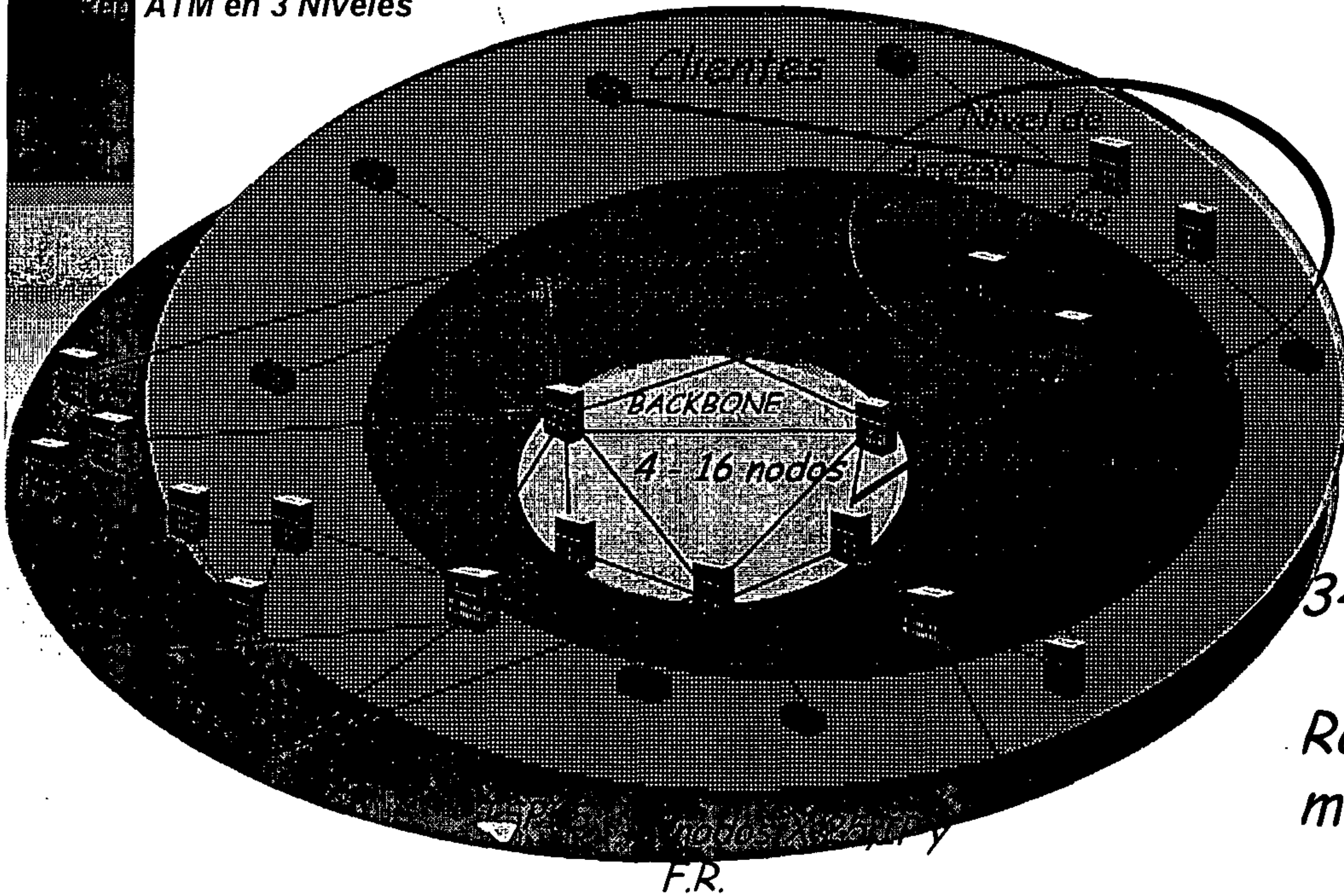
Diagrama de conexión



- Para el caso de interconexión con otras redes de ATM inicialmente se utiliza el Protocolo NNI pudiendo evolucionar a B-ICI.
- La interconexión con centrales telefónicas se realiza con B-ISUP.
- La interconexión con los Giga Routers se realiza con NNI.

DRAFT

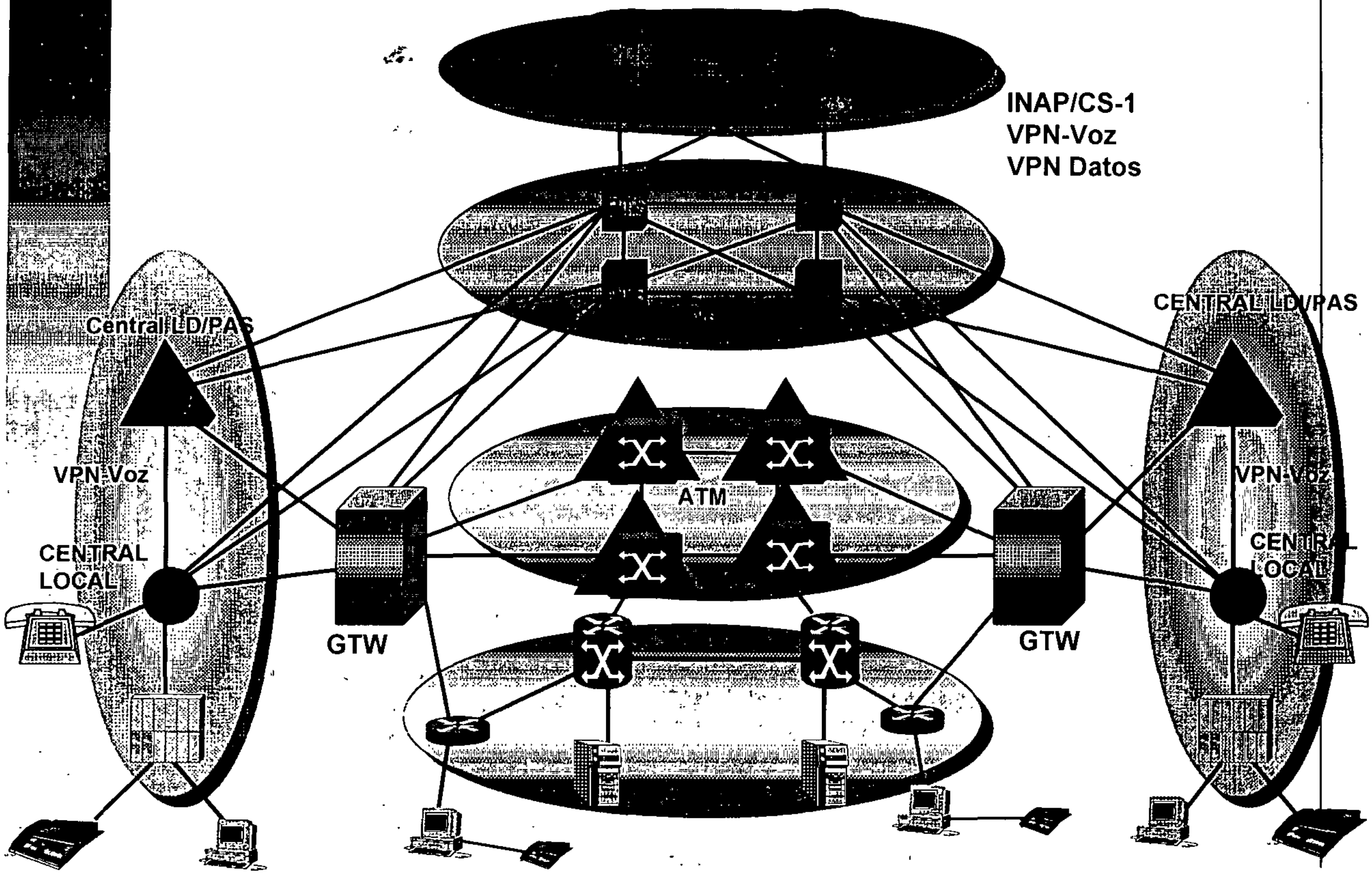
Red ATM en 3 Niveles



34 - 98 nodos

Red multiservicios

Interconexión de una Red Inteligente con las redes de datos



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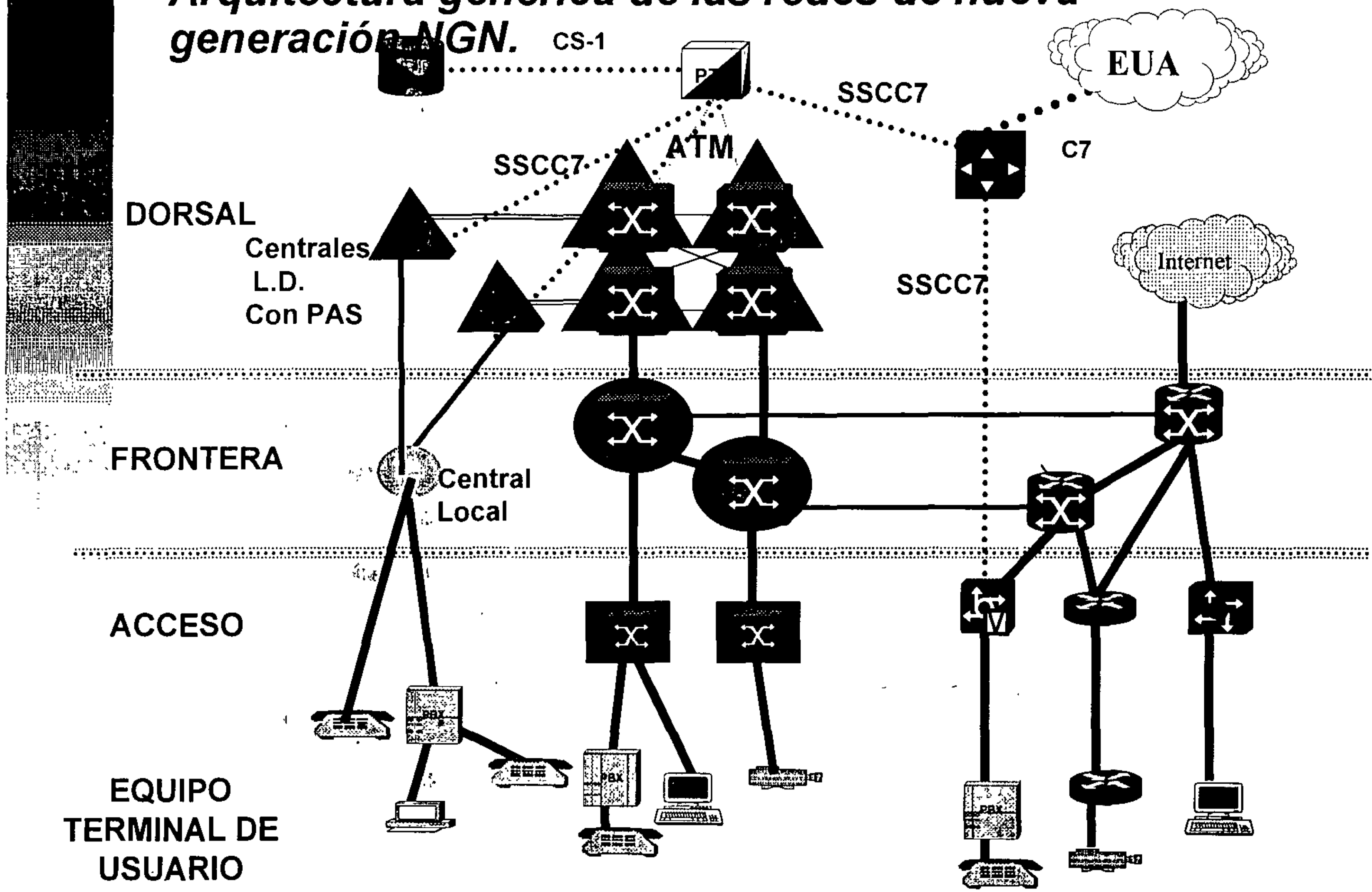
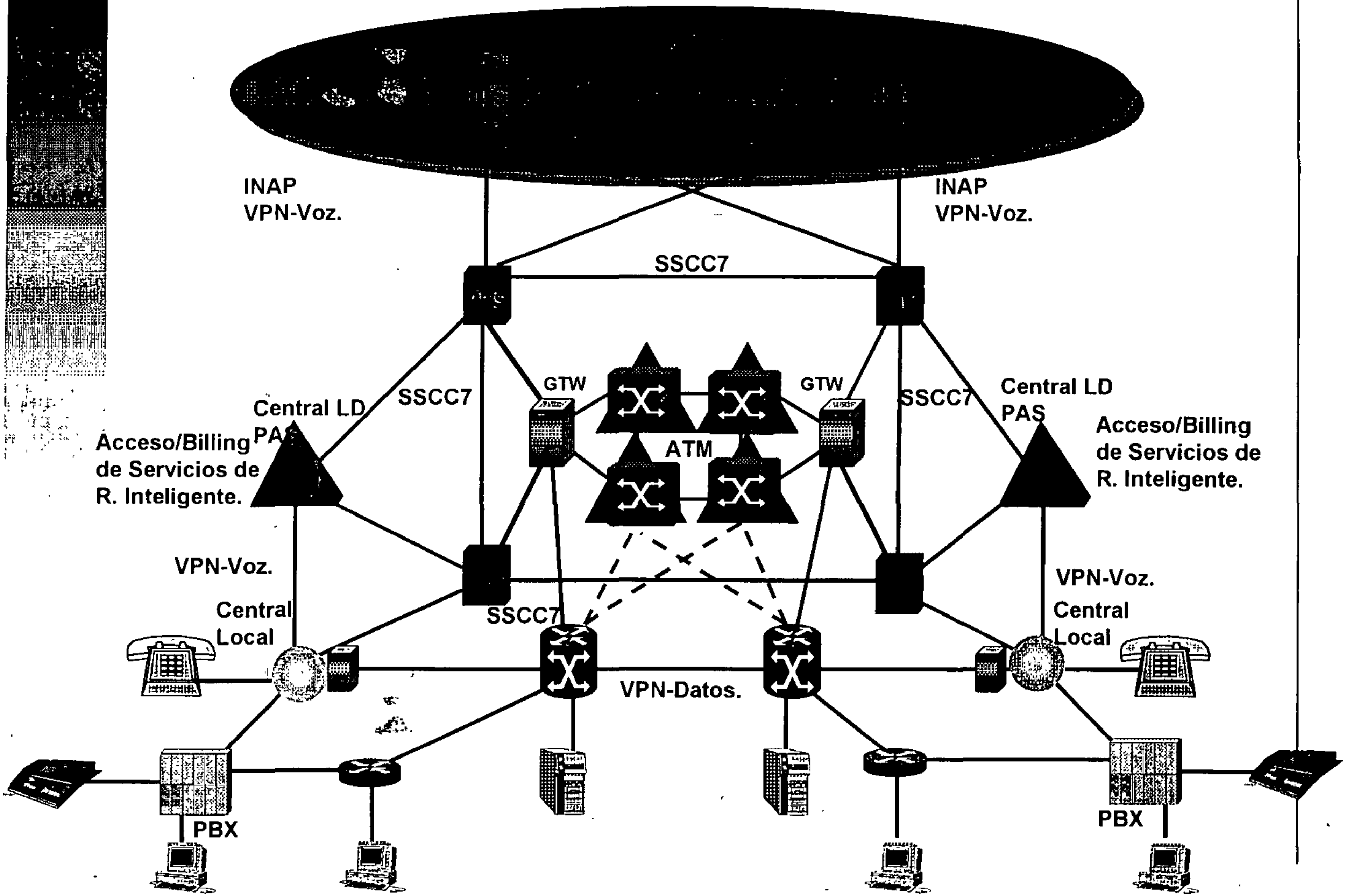
Arquitectura genérica de las redes de nueva generación NGN.



Ejemplos típicos de interfuncionamiento con ATM en la evolución hacia la NGN



Ejemplos de Servicios de un Backbone ATM

- *Oferta nacional*
  - *conexiones VP y VC*
  - *Conexiones permanentes*
  - *CoS : IMS 13H en HO, GTR 4H, Parametros I.356 Classe 1 (DBR)*
  - *Contratos de tráfico DBR con « stringent CDV »*
  - *Contratos de tráfico VBR con MBS= 38 cells*

Ejemplo de Servicios listos para soportarse desde 1999

- *Oferta Multi-accesos ATM (OMA)*
- *Transporte de IP*
- *Emulación de circuitos*

DRAFT

Red Backbone ATM

Un solo backbone para soportar los distintos servicios .

1999 *:*

- *Oferta Multi-servicios sobre ATM*
- *Video :*
- *IP baja velocidad*

2000:

- *IP baja y alta velocidad*
- *Enlaces dedicados asimétricos (ADSL)*
- *Conexión de la red ATM con la Red de Paquetes*
- *Voice Trunking en baja escala*

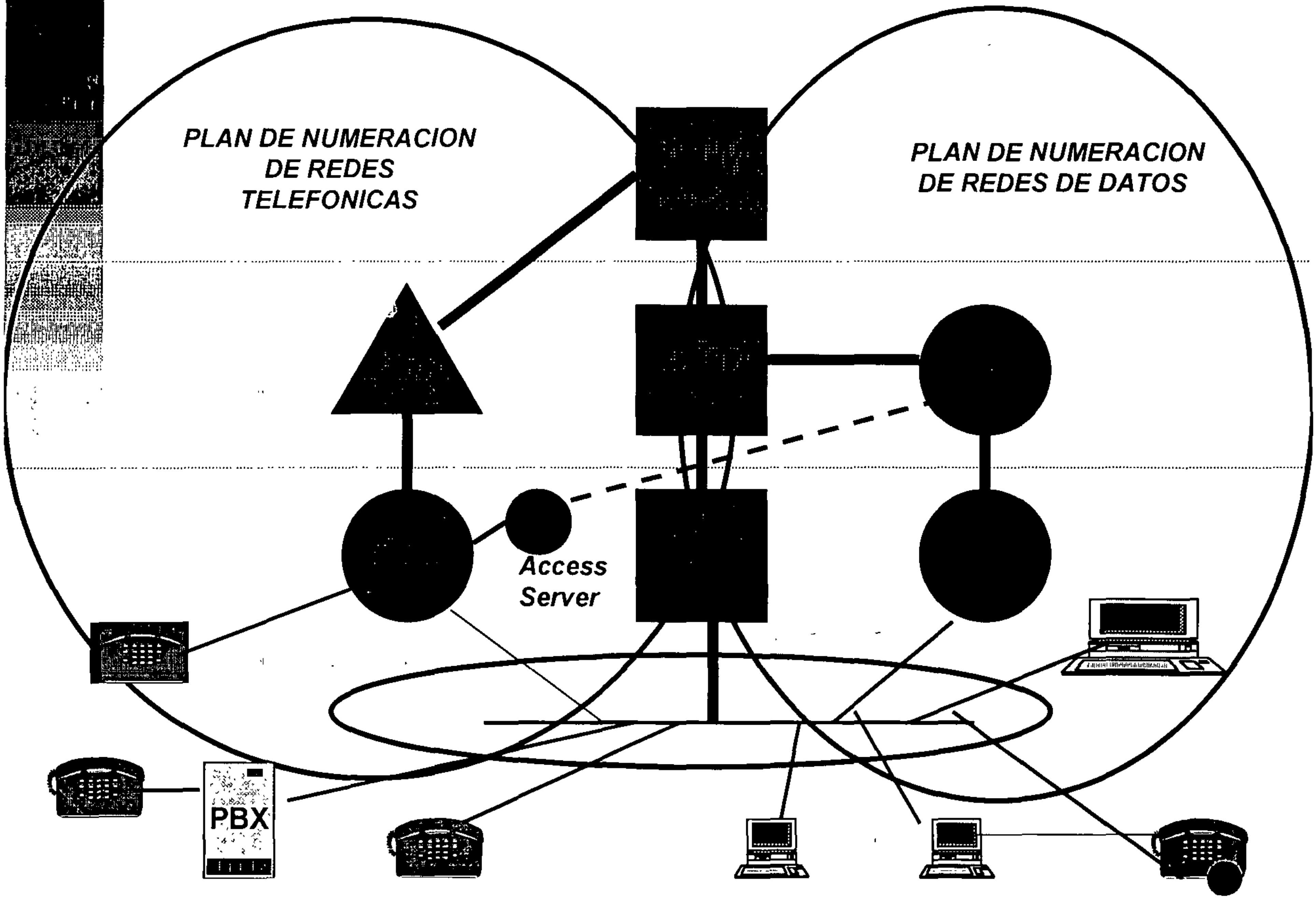
[redacted]

DRAFT

OMA : Oferta Multiservicios con ATM

- *Cobertura geográfica:*
  - *Las grandes ciudades de México y aquellas poblaciones con gran actividad económica*
  - *Ejm. (México D.F. Monterrey, Guadalajara, Puebla, Tijuana,Hermosillo, Mérida, Querétaro, Chihuahua y 58 ciudades más).*
- *Servicios :*
  - *Interconexión LAN, Interconnexión PABX, Servicios ATM via interfaces Frame Relay, Ethernet,* ***ATM***
  - *VP/VC DBR y VP/VC VBR*
- *Servicios a inicio de 2000 :*
  - *Más flexibilidad : VP/VC reservado , ancho de banda flexible,*
  - *Contratos de trafico todavía más flexibles (VBR con MBS importante, ejm MBS=2000)...*

Concepto de Red de Nueva Generación



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FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

<u>ATM UN TECNOLOGÍA COMPETITIVA PARA REDES DE NUEVA GENERACIÓN</u>

EXPOSITOR: ING. ARTURO ALALUF OLIVARES
PALACIO DE MINERÍA
JUNIO DE 1999

Palacio de Minería Calle de Tacuba 5 Primer piso Deleg. Cuauhtémoc 06000 México, D.F. tel.: 521-40-20 Apdo. Postal M-2285

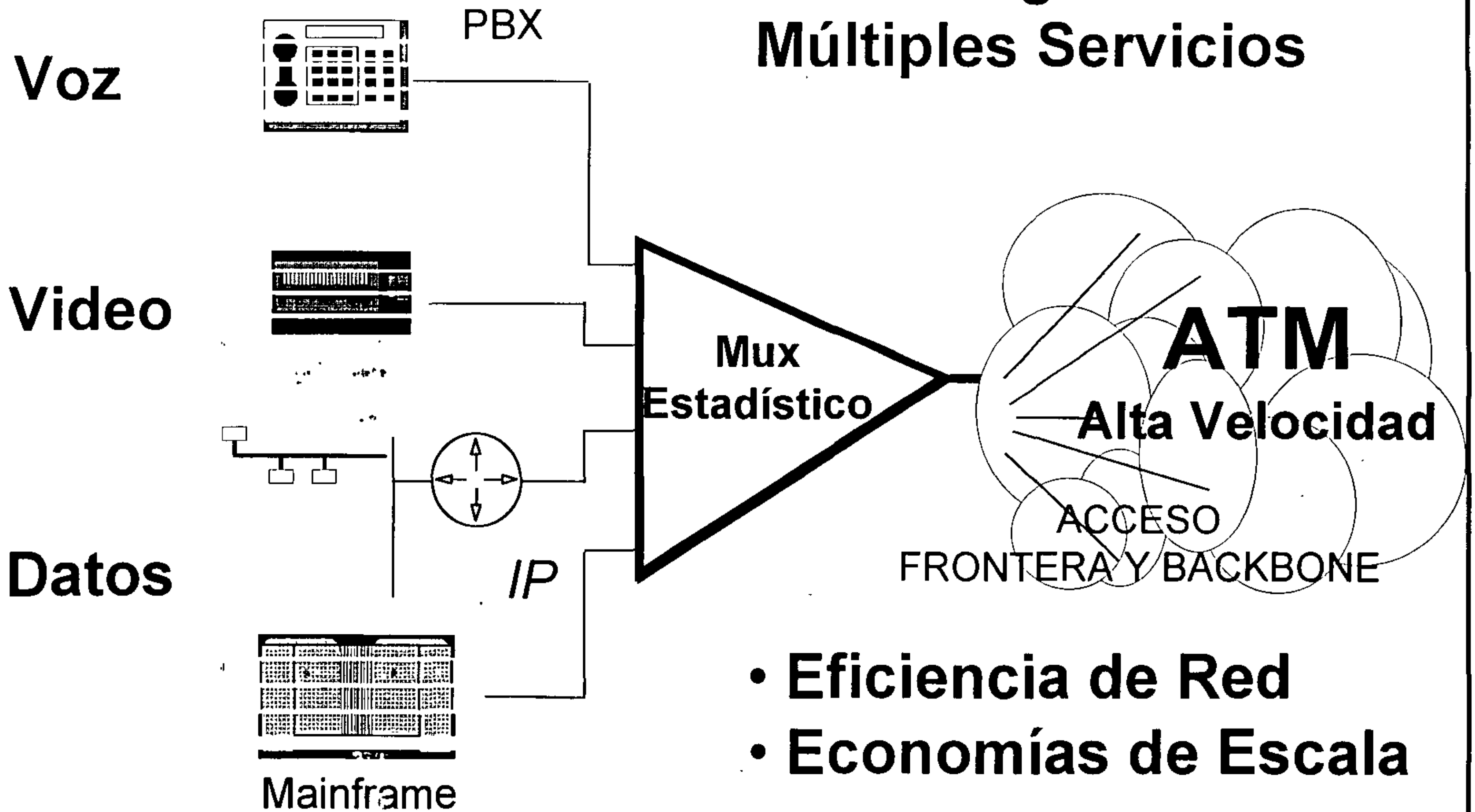
ATM UNA TECNOLOGIA COMPETITIVA PARA REDES DE NUEVA GENERACION.

- ATM PERMITE CONSOLIDAR REDES DIGITALES DE DATOS Y DE TELEFONIA A TRAVES DE UNA PLATAFORMA COMUN.

- COMBINANDO LOS AVANCES DE LA ELECTRONICA Y DE LA FOTONICA EN LA RED, LAS EMPRESAS DE TELECOMUNICACIONES ASEGURAN SU LIDERAZGO EN SERVICIOS DE VOZ, DATOS Y VIDEO.

Propósito de las Redes ATM

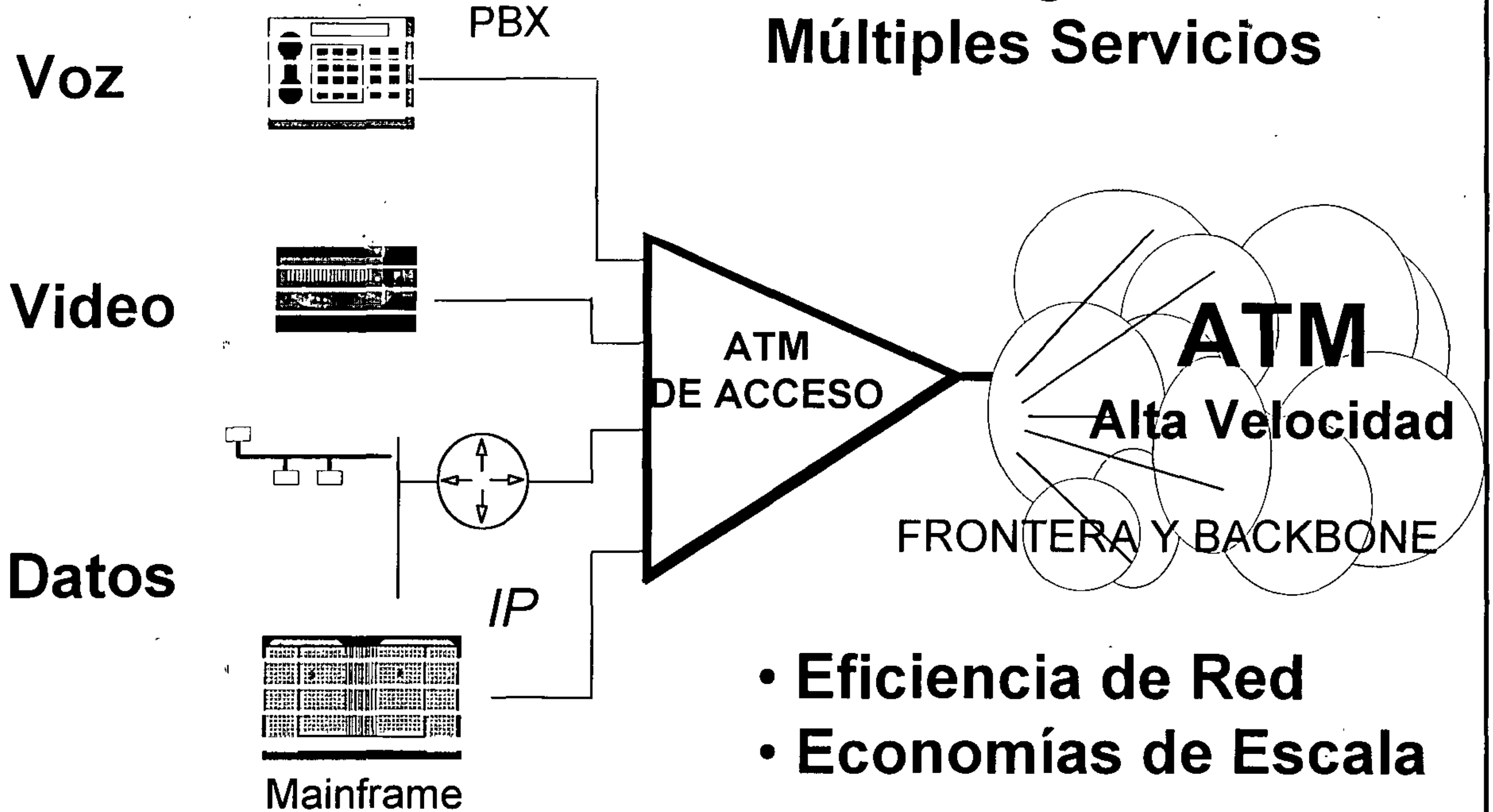
Integrar
Múltiples Servicios



- **Eficiencia de Red**
- **Economías de Escala**

Propósito de las Redes ATM

Integrar
Múltiples Servicios

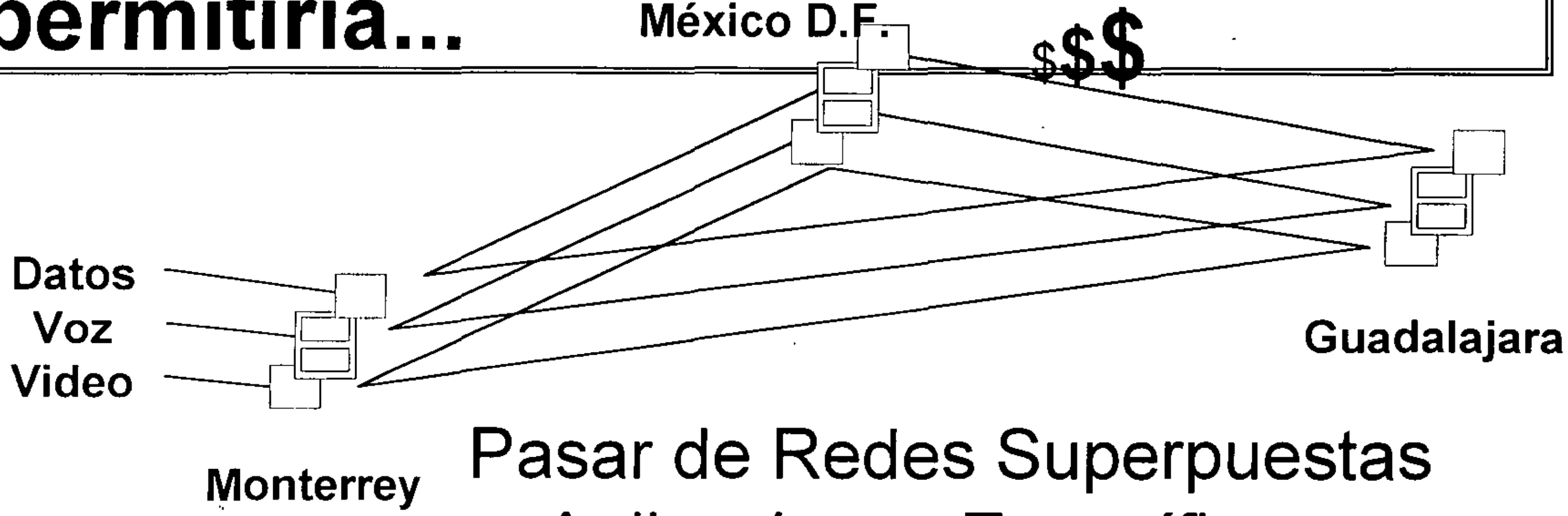


- Eficiencia de Red
- Economías de Escala

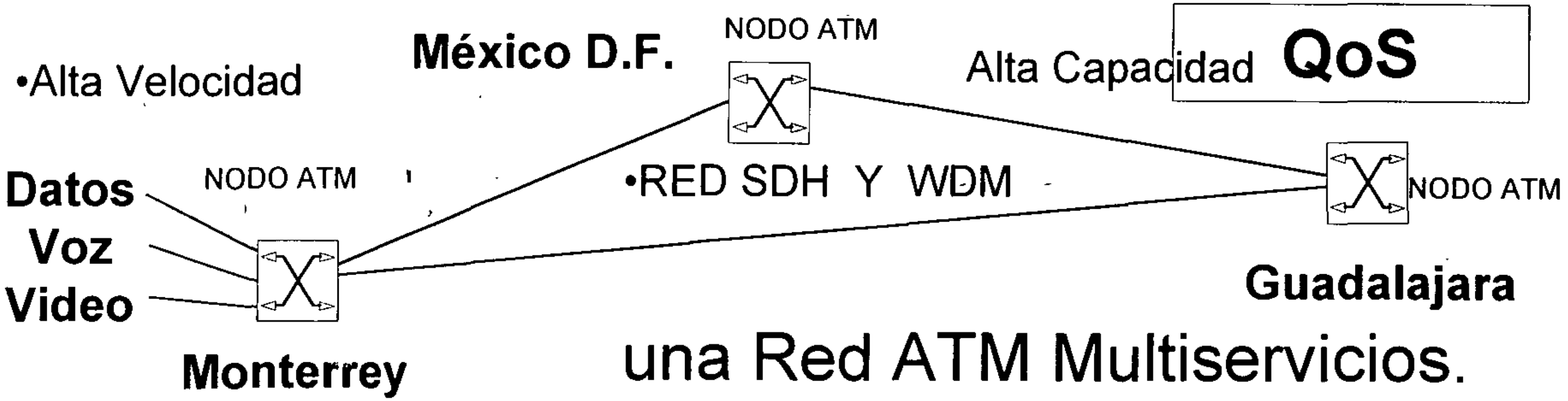
COMO CUIDAR LAS INVERSIONES?...

- Invirtiendo en Tecnología altamente competitiva...
- Que optimice el uso y Evolución de la Red,
- Que reduzca los costos de Operación y Administración,
- Que permita atender con oportunidad, amplitud y calidad los servicios definidos por sus áreas comerciales.
- ATM CUMPLE CON ESTOS REQUISITOS

Por ejemplo, un Proyecto ATM nos permitiría...



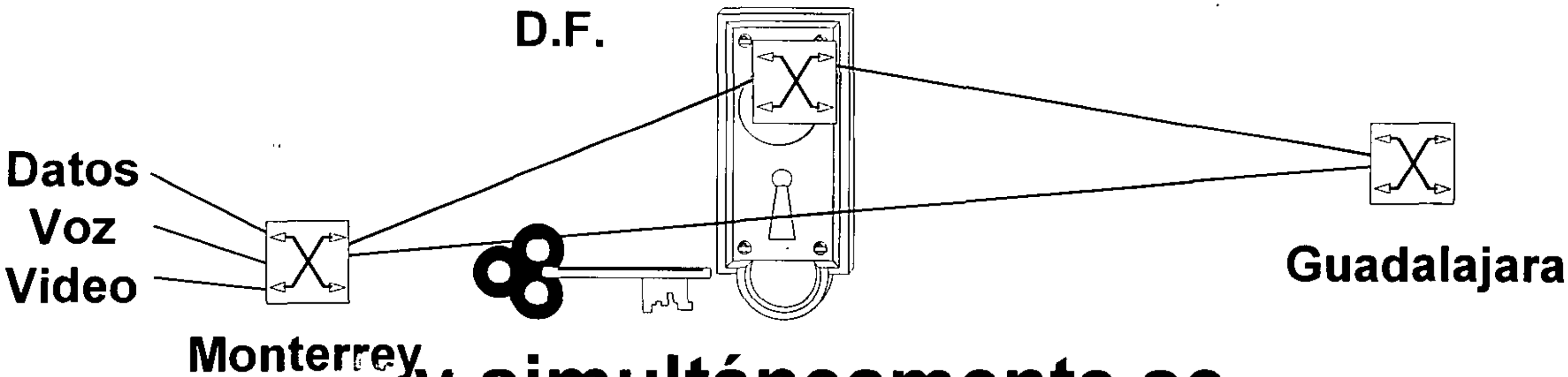
Pasar de Redes Superpuestas con Aplicaciones Específicas, a....



una Red ATM Multiservicios.

Con una Plataforma ATM ...

Se Entrega la
Calidad de Servicio (QoS) Acordada
con cada Cliente

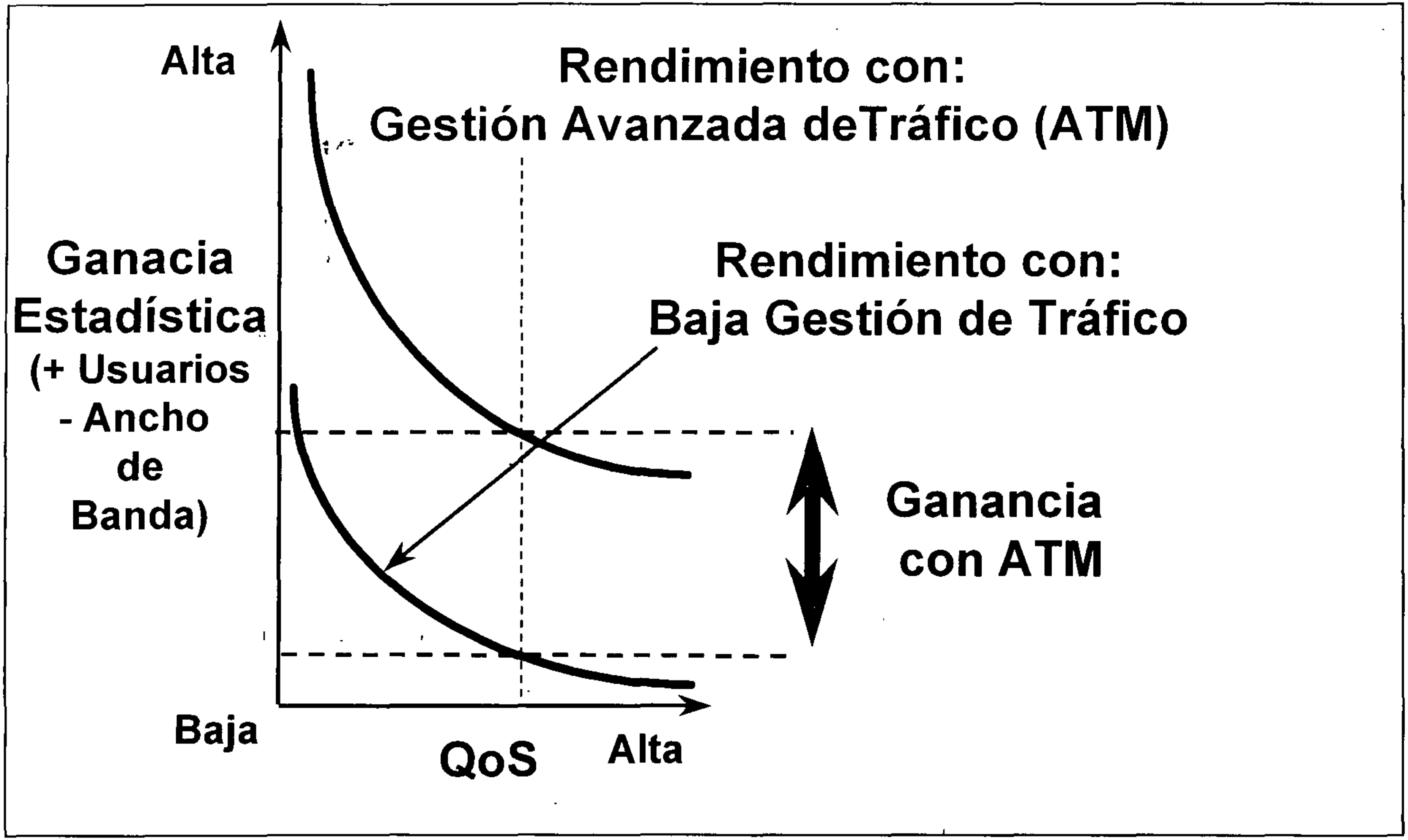


y simultáneamente se
optimiza el Ancho de Banda
- Eficiencia de la Red-

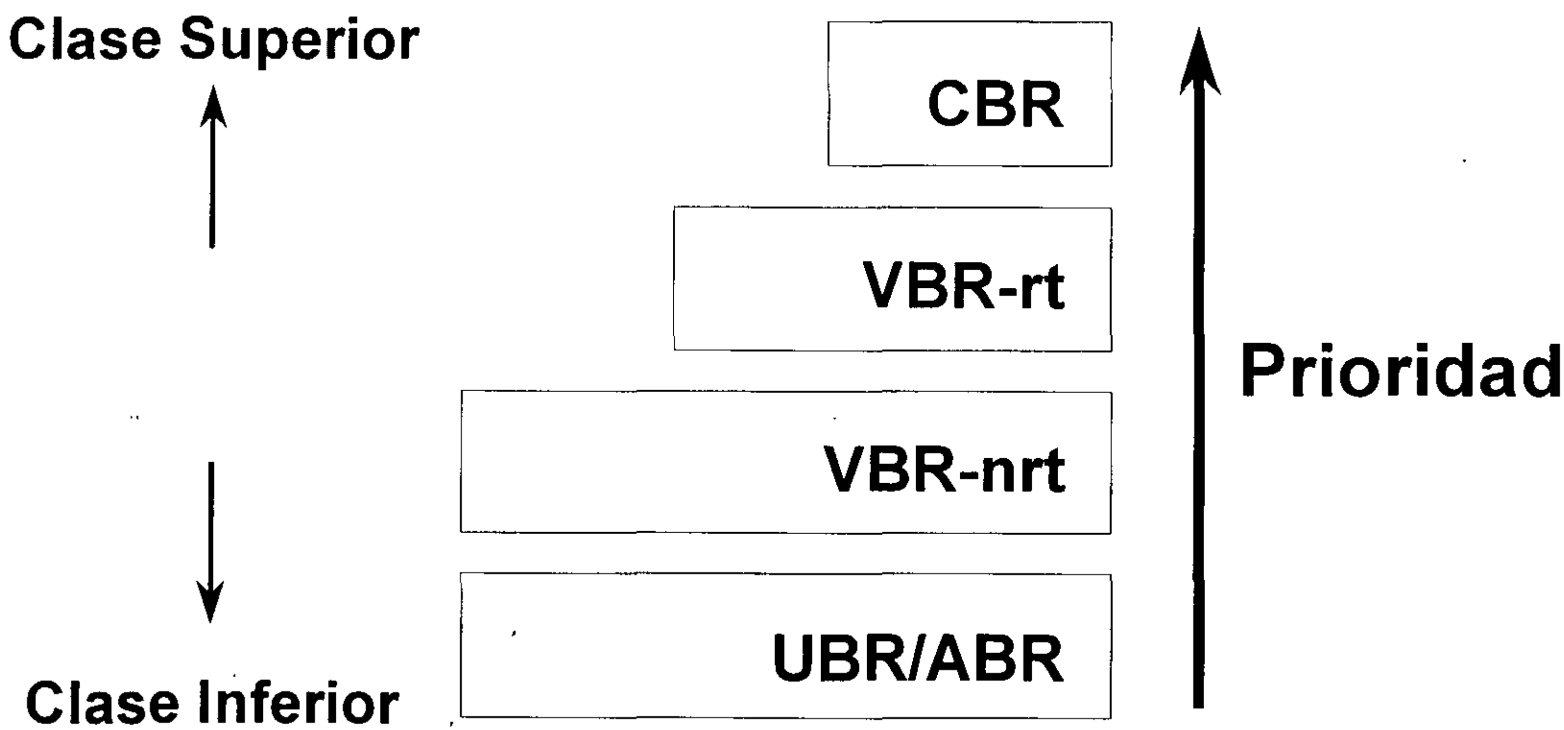
Gestión Avanzada de Tráfico de Red

CON UNA PLATAFORMA ATM ES POSIBLE OFRECER....

Mayor Rendimiento usando Gestión Avanzada de Red

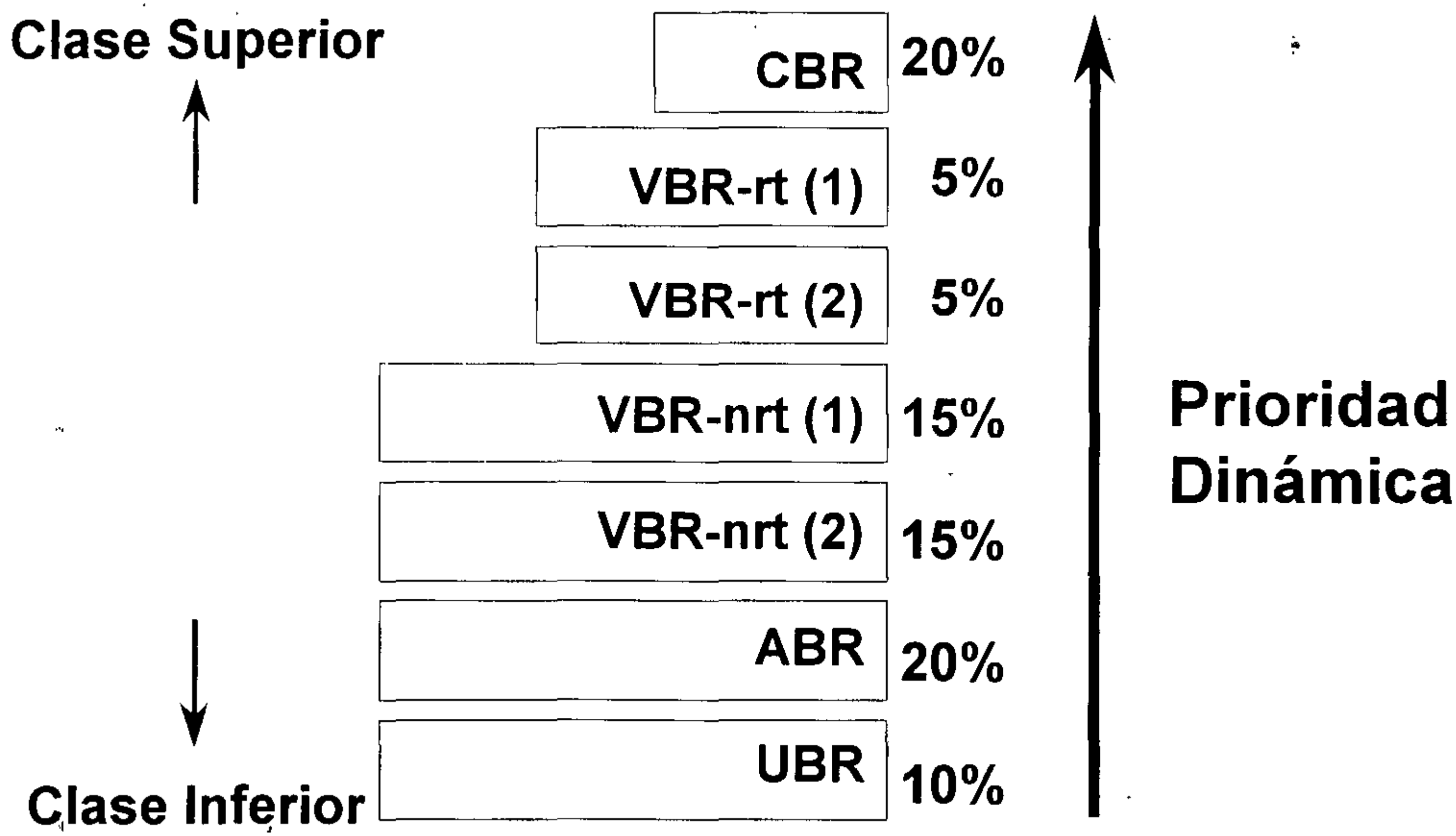


El Modelo Tradicional de Gestión de Tráfico de *IP* necesita evolucionar.....



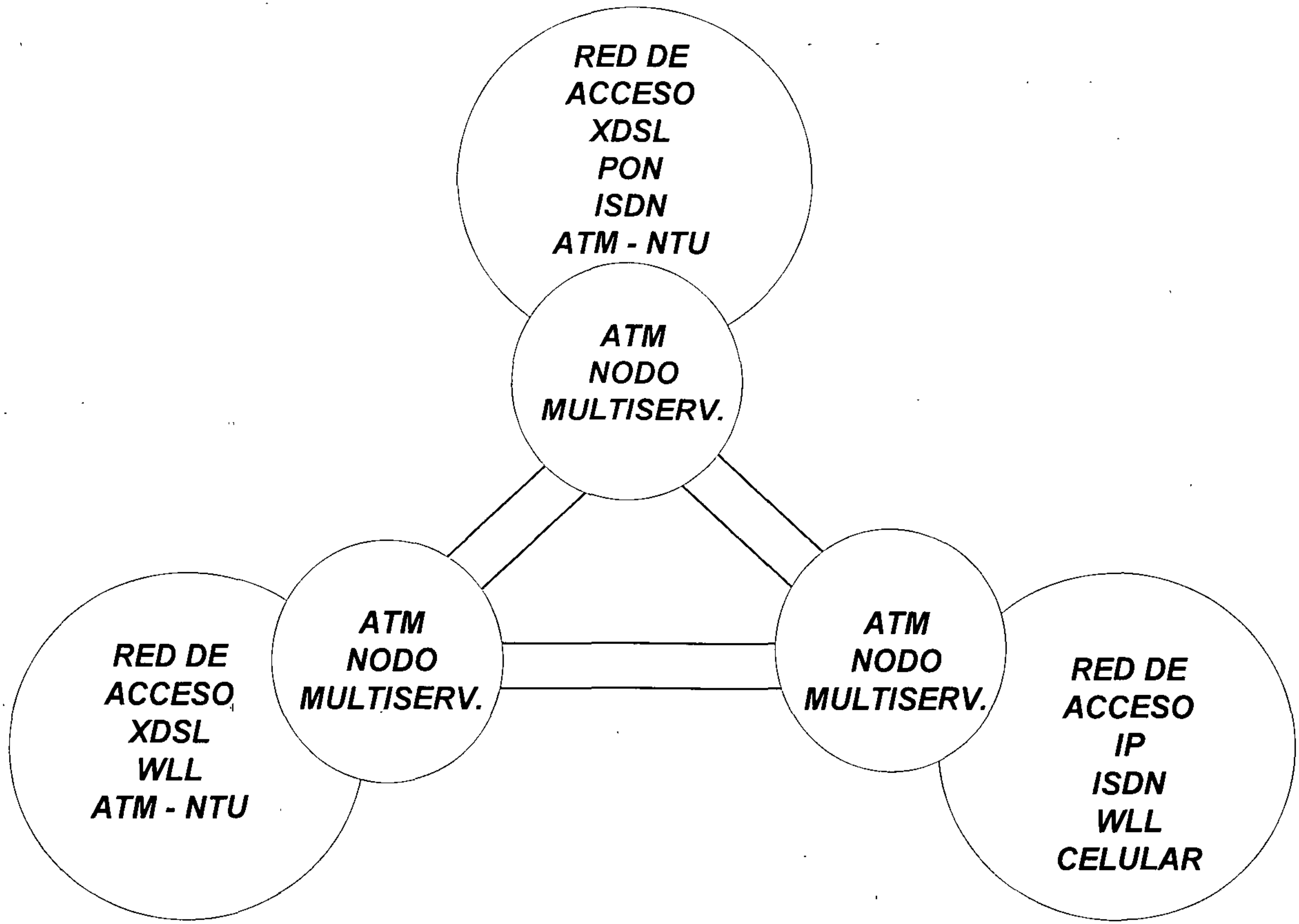
- Servicios de clase superior pueden llegar a complicar la disponibilidad de servicios de clase inferior
- Puede escalarse a un problema de congestión

Gestión Avanzada de Tráfico con una Plataforma ATM.....



- Garantiza siempre un Mínimo de Ancho de Banda
- Elimina Ausencia de Servicio a Clases Inferiores
- Reduce Retransmisiones que Provocan Congestión

ATM INTEGRACION GLOBAL DE LA RED



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MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

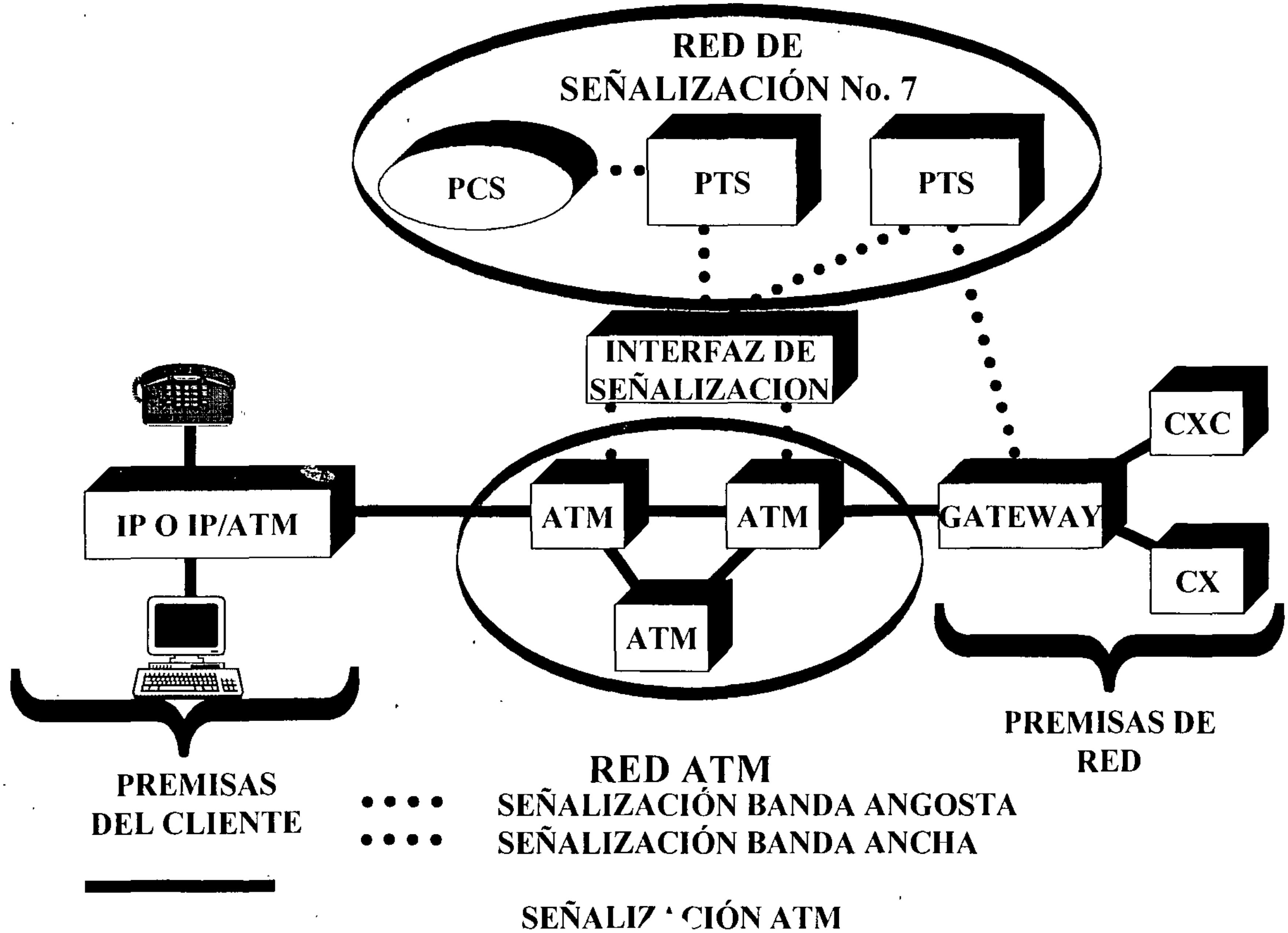
TEMA

<u>RED DE NUEVA GENERACIÓN</u>
<u>NGN</u>

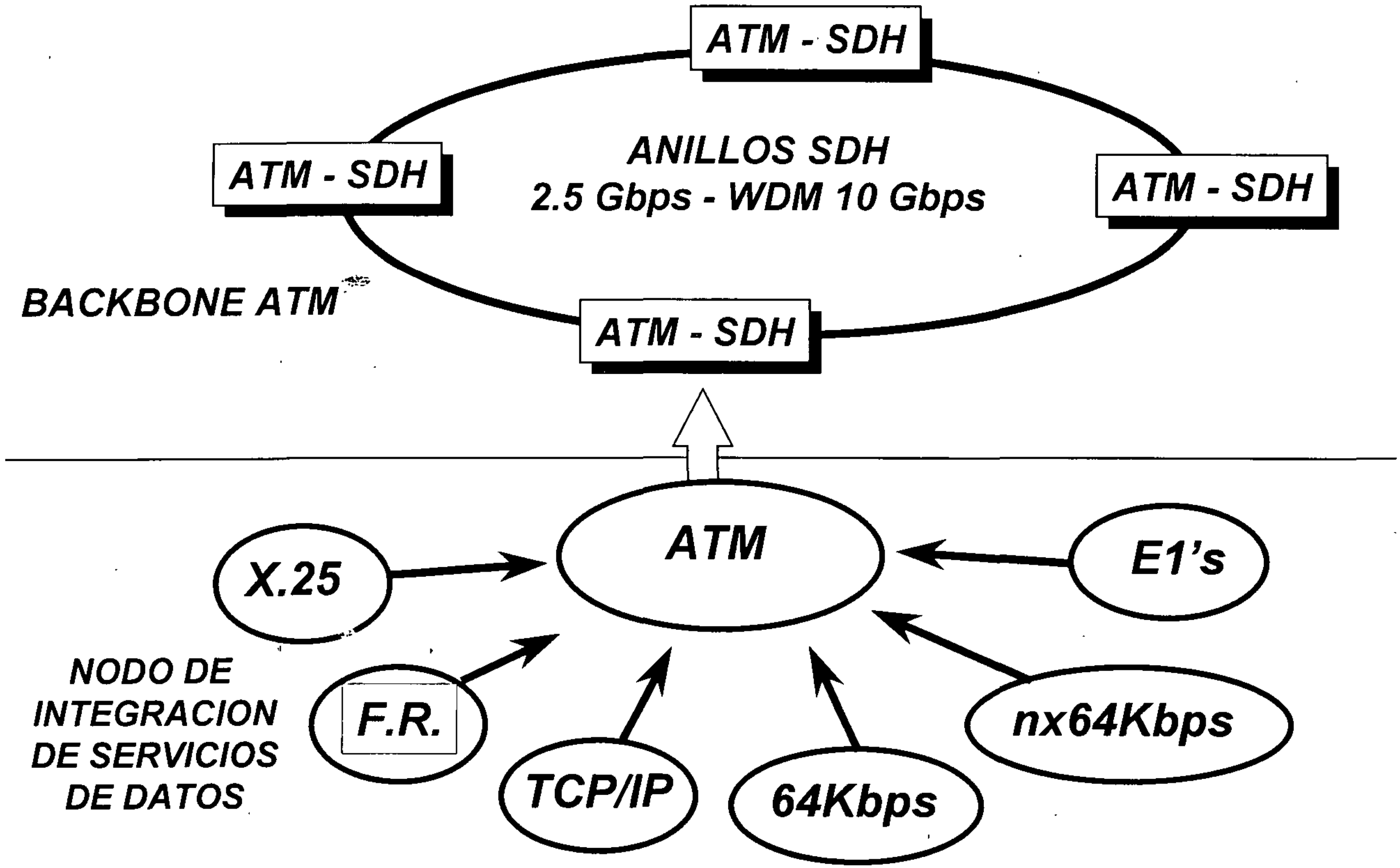
EXPOSITOR: ING. ARTURO ALALUF OLIVARES
PALACIO DE MINERÍA
JUNIO DE 1999

RED DE NUEVA GENERACION NGN

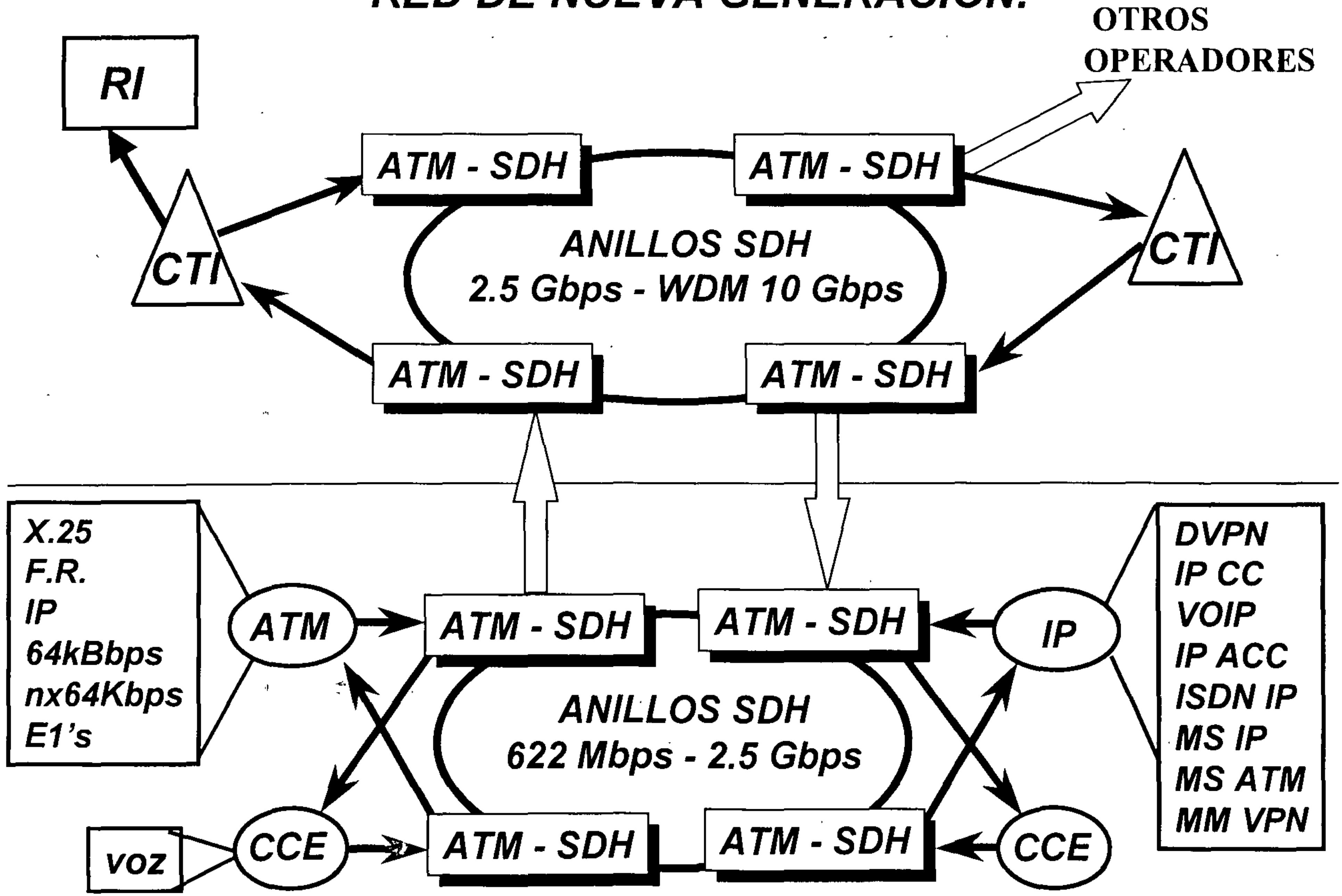
RED DE NUEVA GENERACION



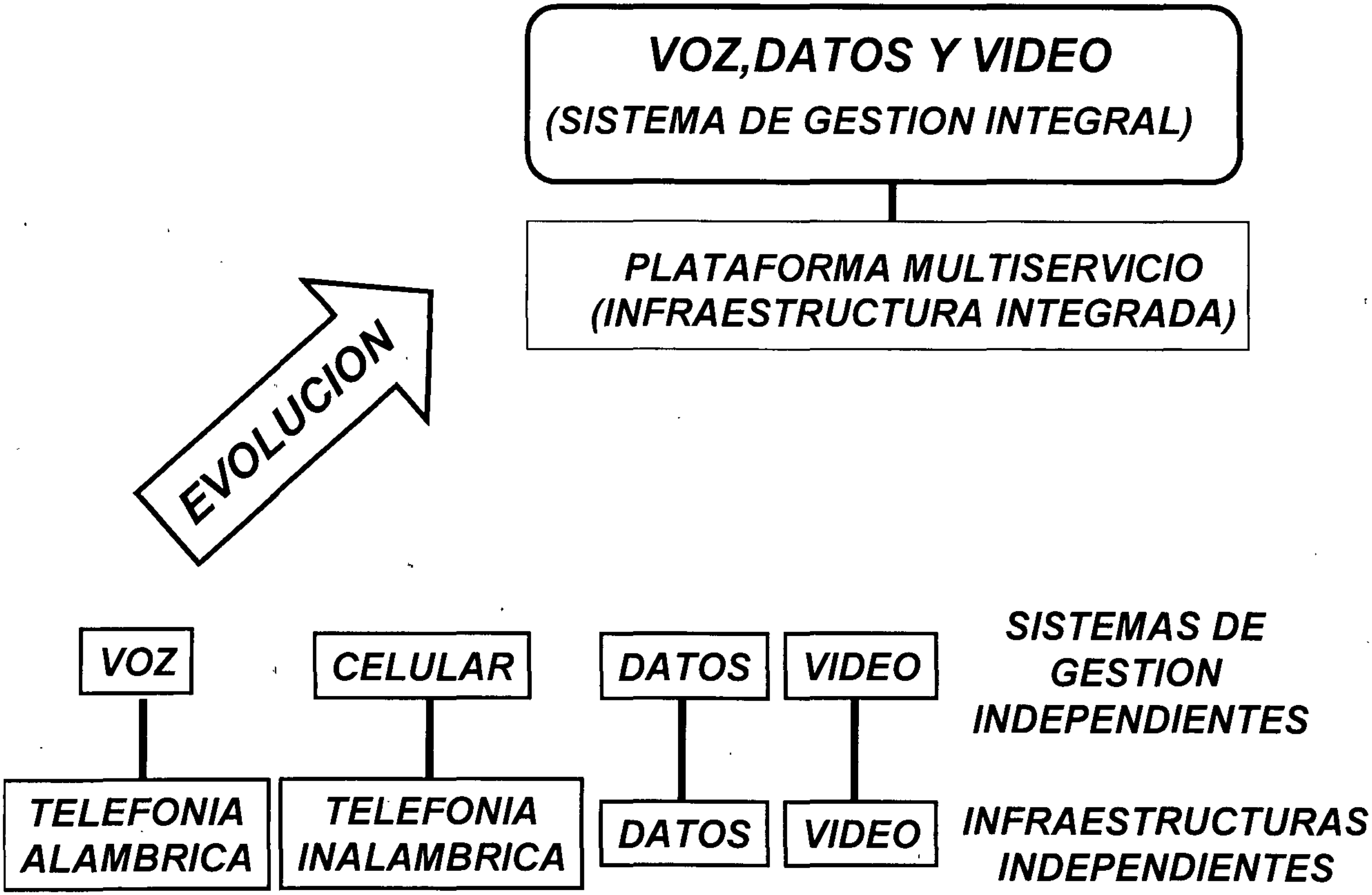
RED DE NUEVA GENERACION



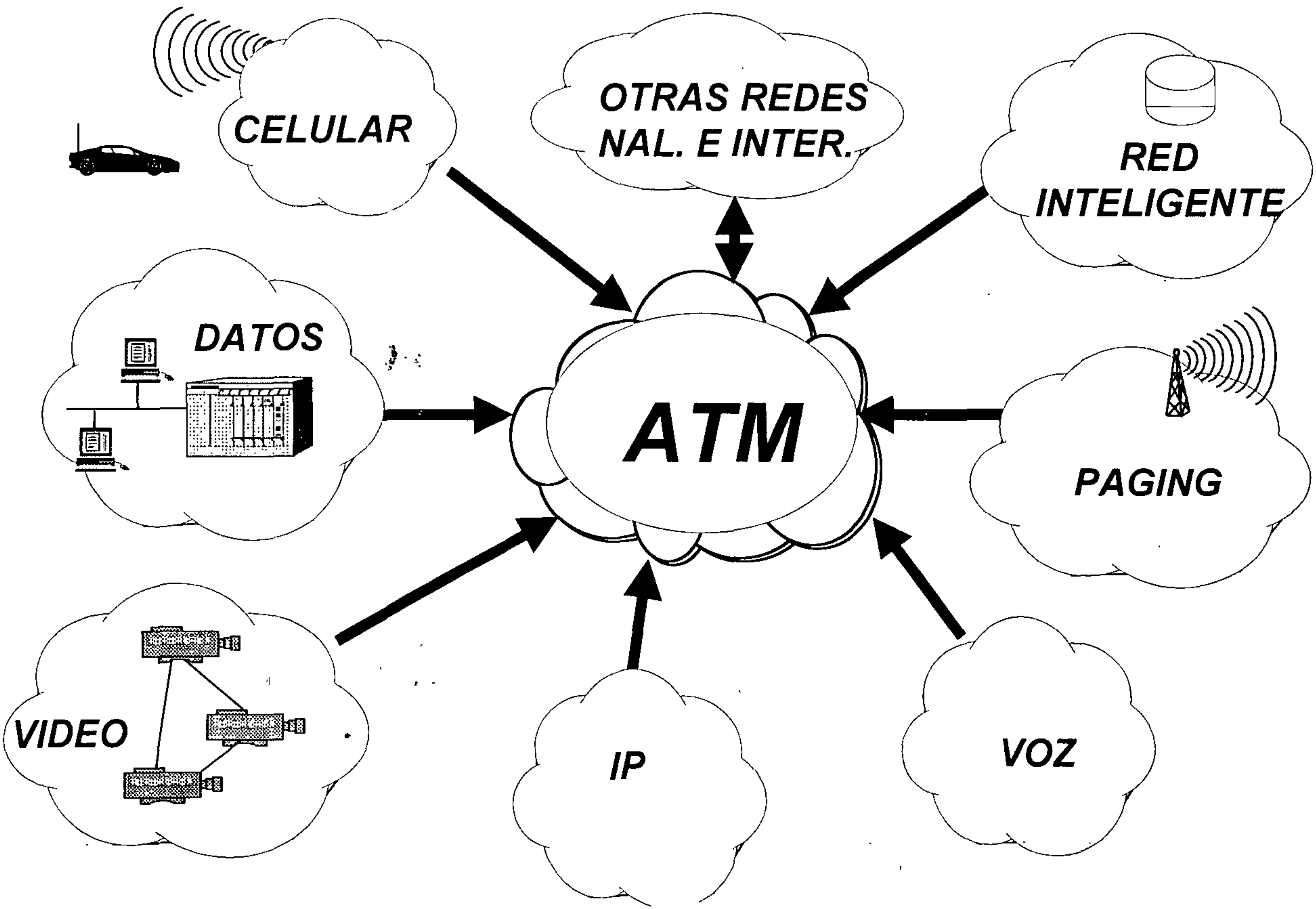
RED DE NUEVA GENERACION.



RED DE NUEVA GENERACION
GESTION



RED DE NUEVA GENERACION

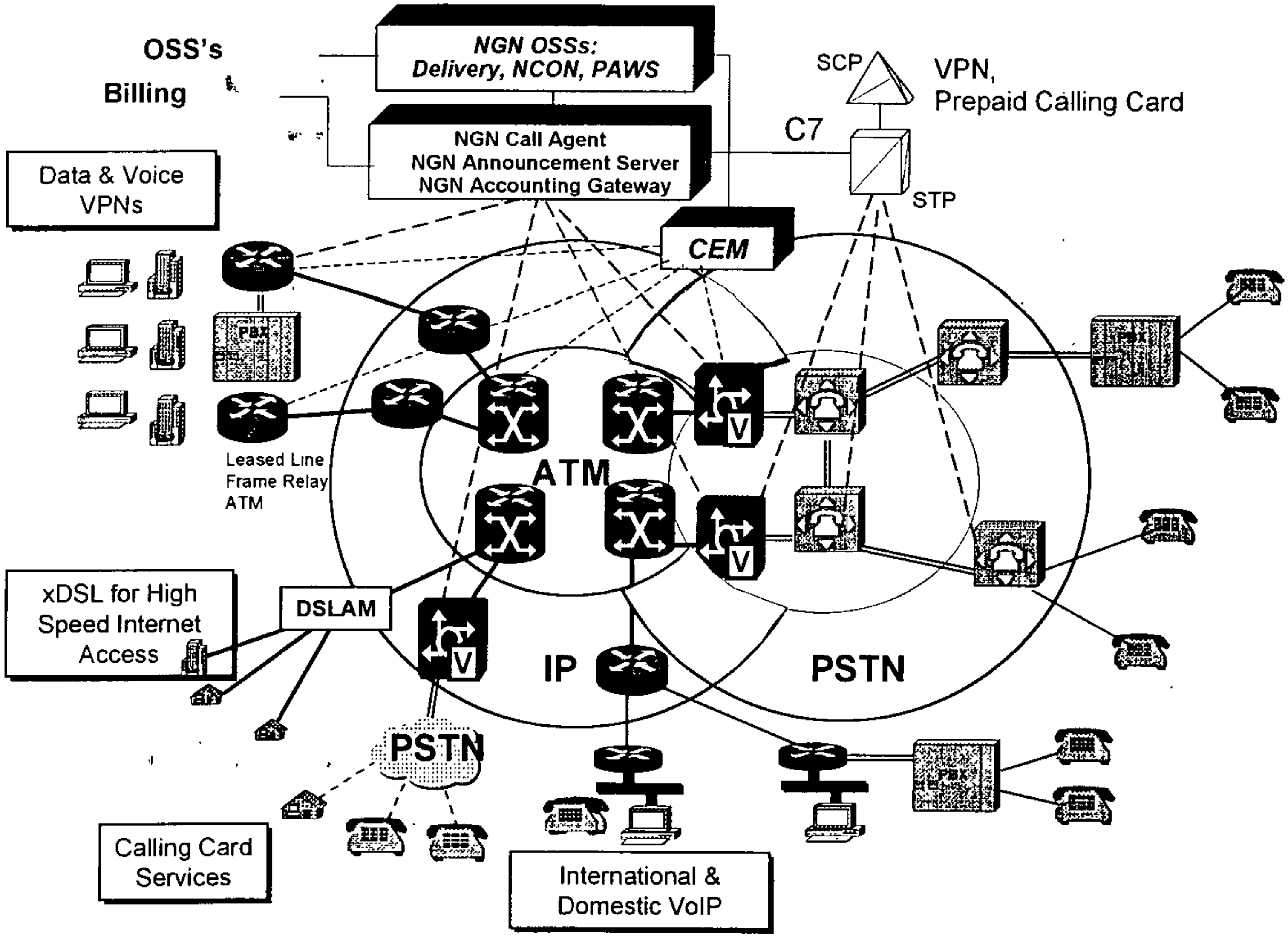


ALCANCE DE LA NGN

- INCLUYE TODOS LOS SERVICIOS ACTUALES Y QUE SE HAN PREVISTO PARA LAS REDES DE DATOS Y DE TELECOMUNICACIONES.
- ESTA BASADA EN UNA RED ATM/IP PARA EL PROCESAMIENTO E INTERFUNCIONAMIENTO CON LAS REDES ACTUALES.

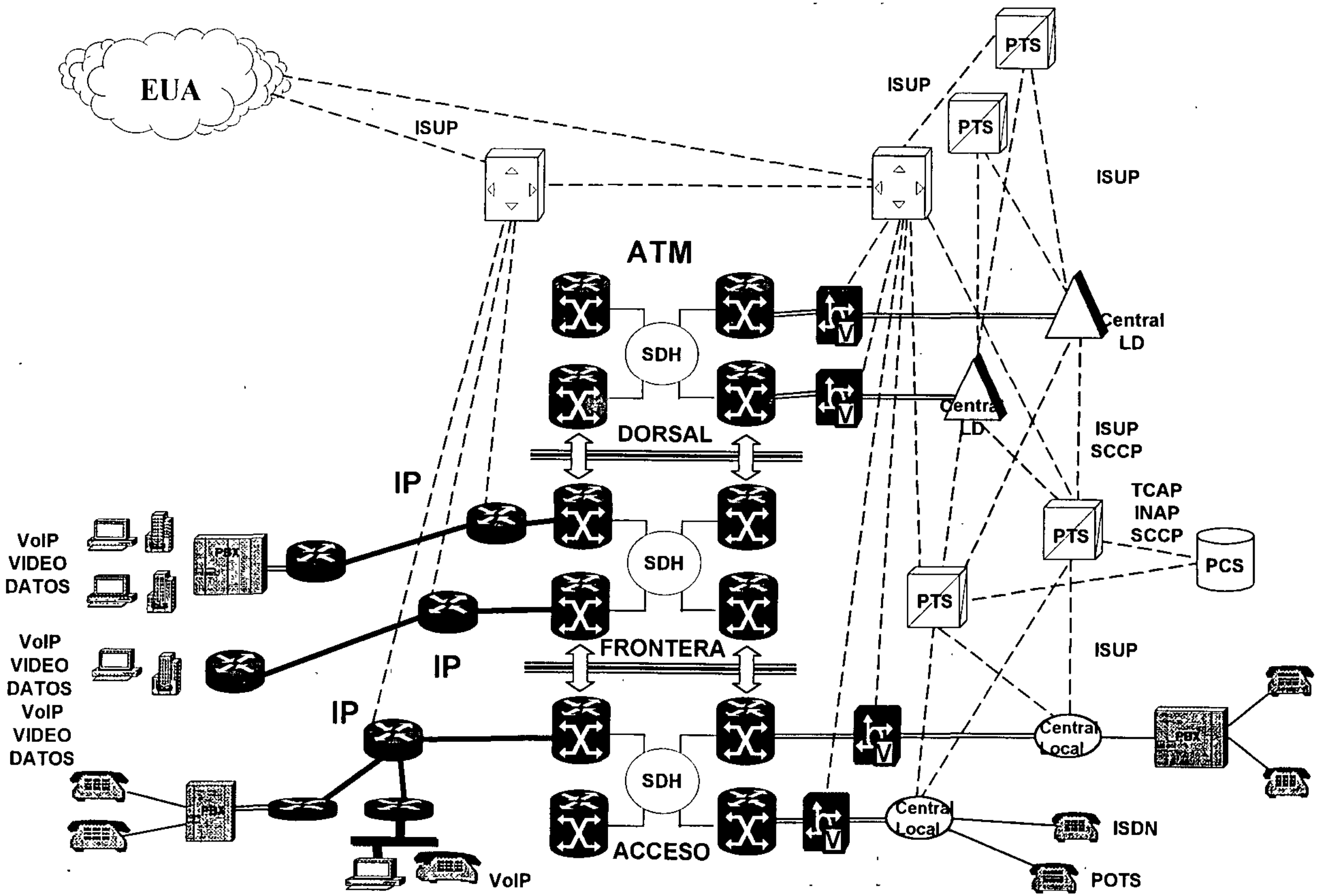
ALCANCE DE LA NGN

- ESTA BASADA EN WDM Y SDH PARA EL TRANSPORTE .
- APROVECHA LAS CARACTERISTICAS MAS IMPORTANTES DE CADA UNA DE LAS TECNOLOGIAS ATM E IP.
- EN EL ACCESO CONSIDERA LAS TECNOLOGIAS XDSL ,WLL, IP Y ATM
-

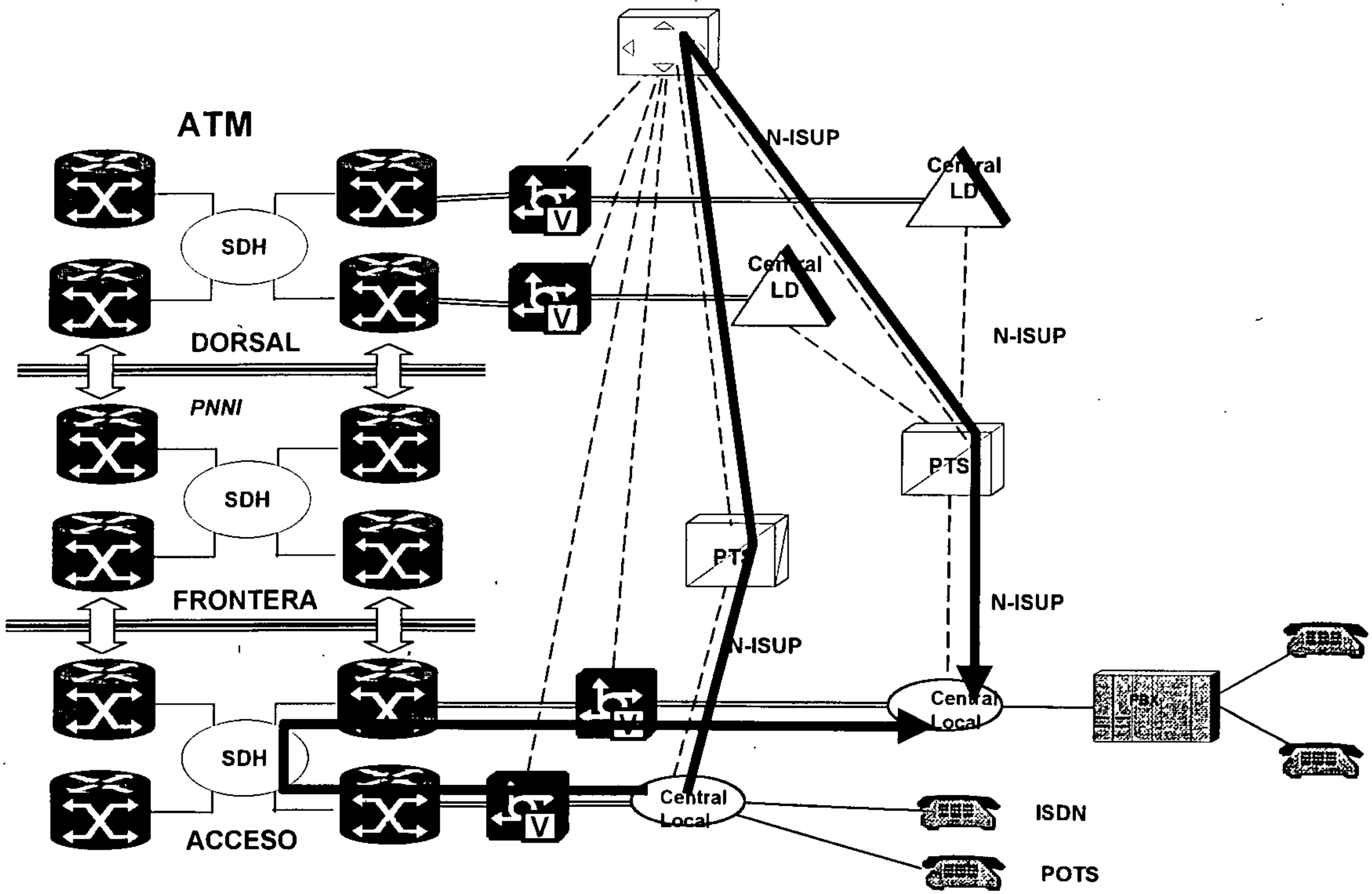


OSS's
Billing
NGN OSSs:
Delivery, NCON, PAWS
NGN Call Agent
NGN Announcement Server
NGN Accounting Gateway
SCP
VPN,
Prepaid Calling Card
C7
STP
Data & Voice
VPNs
CEM
PBX
Leased Line
Frame Relay
ATM
ATM
xDSL for High
Speed Internet
Access
DSLAM
IP
PSTN
PSTN
Calling Card
Services
International &
Domestic VoIP

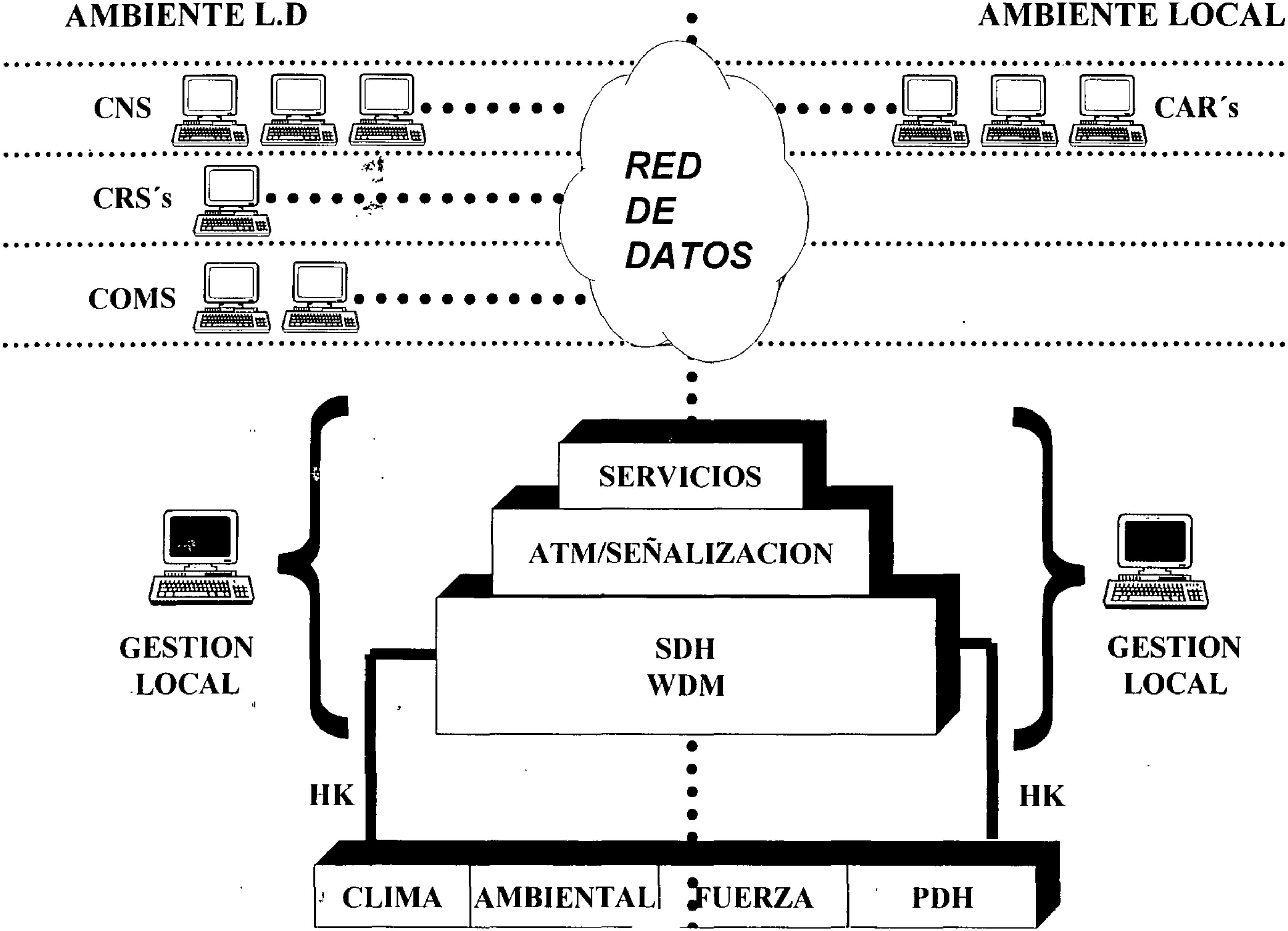
ESCENARIOS DE SEÑALIZACION PARA EL SISTEMA DE S.C.C. No. 7



EJEMPLOS DE PROCESOS DE LLAMADA ENTRE ATM Y EL SISTEMA DE S.C.C. No. 7



EJEMPLOS DE SUPERVISION Y GESTION



SUPERVISION Y GESTION

SERVICIOS

- FACTURACION.
- GESTION DE RED MULTISERVICIO.
- INTERACCION CON OTROS OPERADORES DE SERVICIOS.
- INTERACCION ENTRE SERVICIOS.

ATM/SEÑALIZACION

- INTERACCION CON LA RED DE SERVICIOS EN USO, DESEMPEÑO Y DISPONIBILIDAD.
- SOPORTE DE SERVICIOS.
- APROVISIONAMIENTO Y MODIFICACION DE FACILIDADES.
- CONTROL DE VISTA DE RED.

SDH
WDM

- GESTION DE RED FOTONICA.
- GESTION DE CADA UNO DE LOS ELEMENTOS DE RED PDH, FUERZA Y CLIMA.

FUENTE: RECOMENDACION M.3010 /UIT

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REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

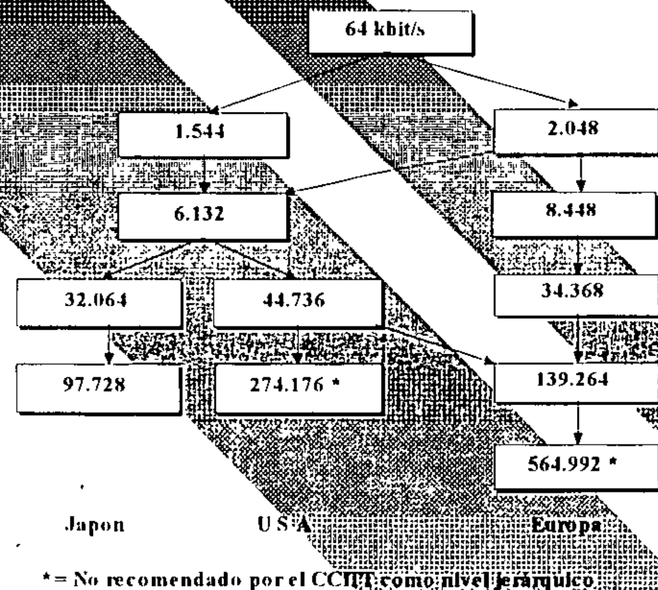
PRINCIPIOS Y ELEMENTOS DE RED SDH PARA UNA INFRAESTRUCTURA DE TELECOMUNICACIONES

EXPOSITOR: ING. RODOLFO BUENDÍA
PALACIO DE MINERÍA
JUNIO DE 1999

PRINCIPIOS Y ELEMENTOS DE RED SDH PARA LA INFRAESTRUCTURA DE TELECOMUNICACIONES

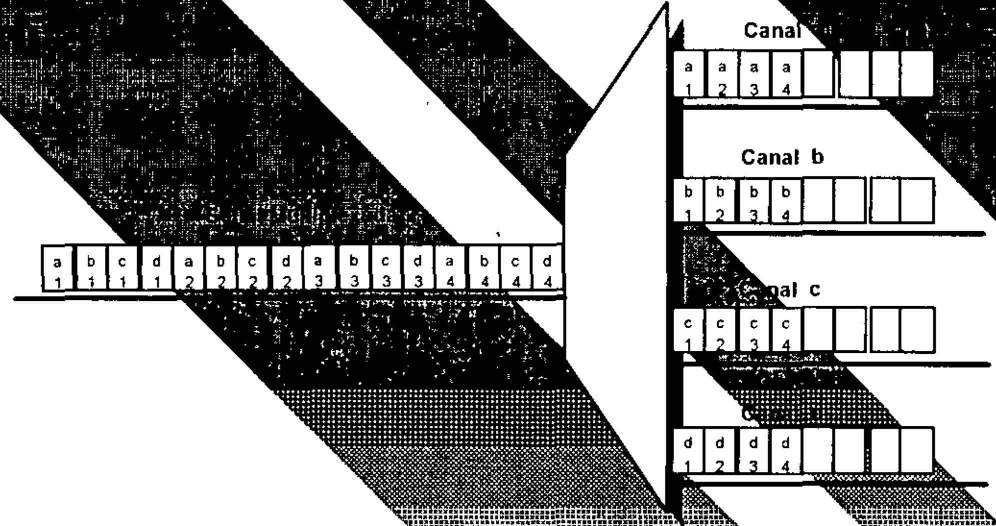
Ing. Rodolfo Buendía
NORTEL Ingeniería de Transmisión
Tel. (5) 480-8269
Fax. (5) 480-8292
rodolfo_buendia@nt.com

Velocidades de transmisión para PDH

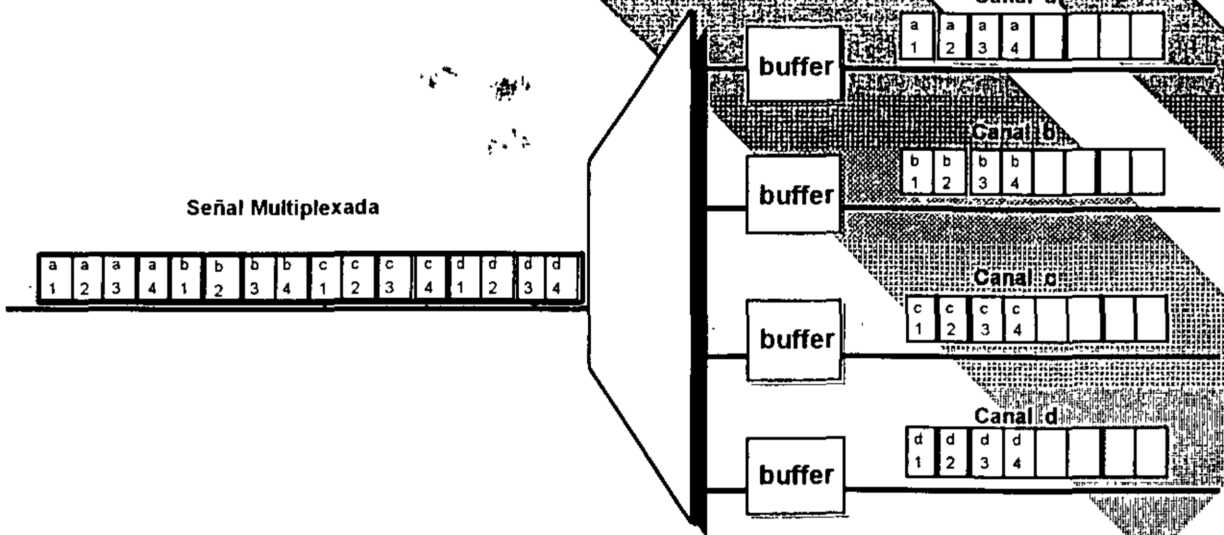


Velocidades de transmisión para SDH

SONET / USA		CCITT / EUROPA
SDS-1	51.84 Mbit/s	
SDS-3	155.52 Mbit/s	STM-1
SDS-9	466.56 Mbit/s	
SDS-12	622.08 Mbit/s	STM-4
SDS-18	933.12 Mbit/s	
SDS-24	1244.16 Mbit/s	
SDS-36	1866.24 Mbit/s	
SDS-48	2488.32 Mbit/s	STM-16



Multiplexaje
SDH

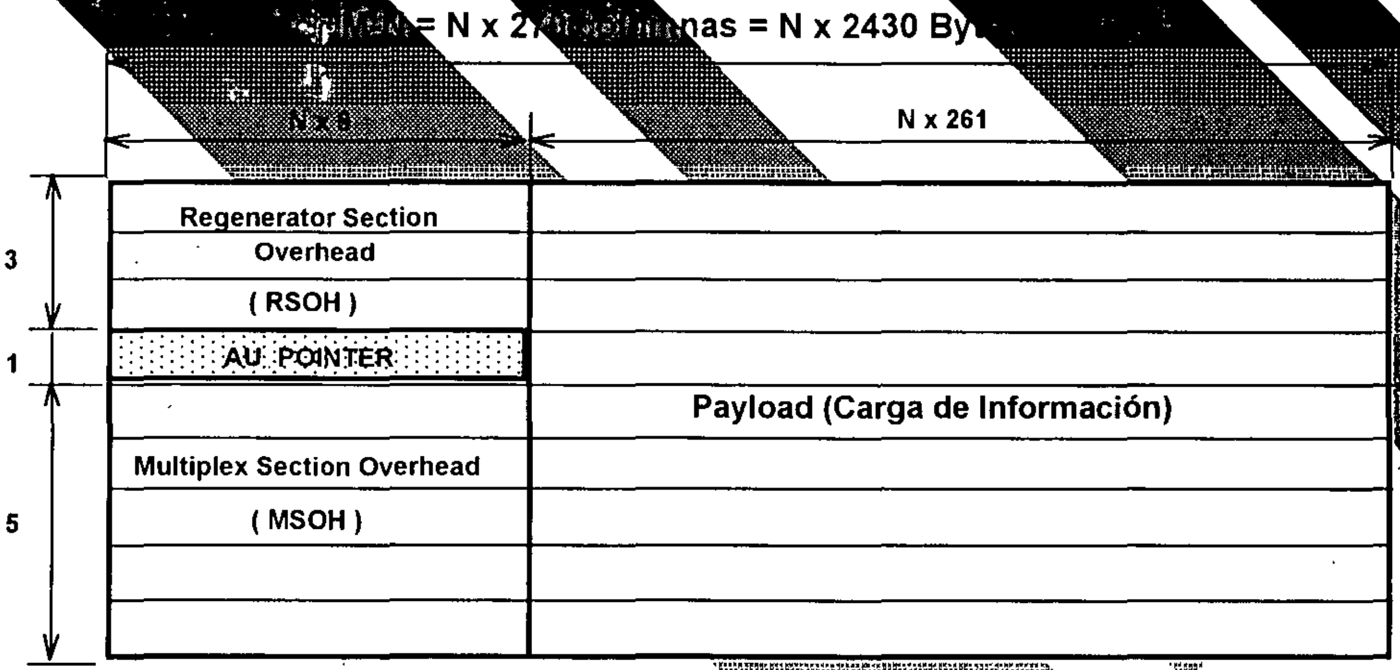


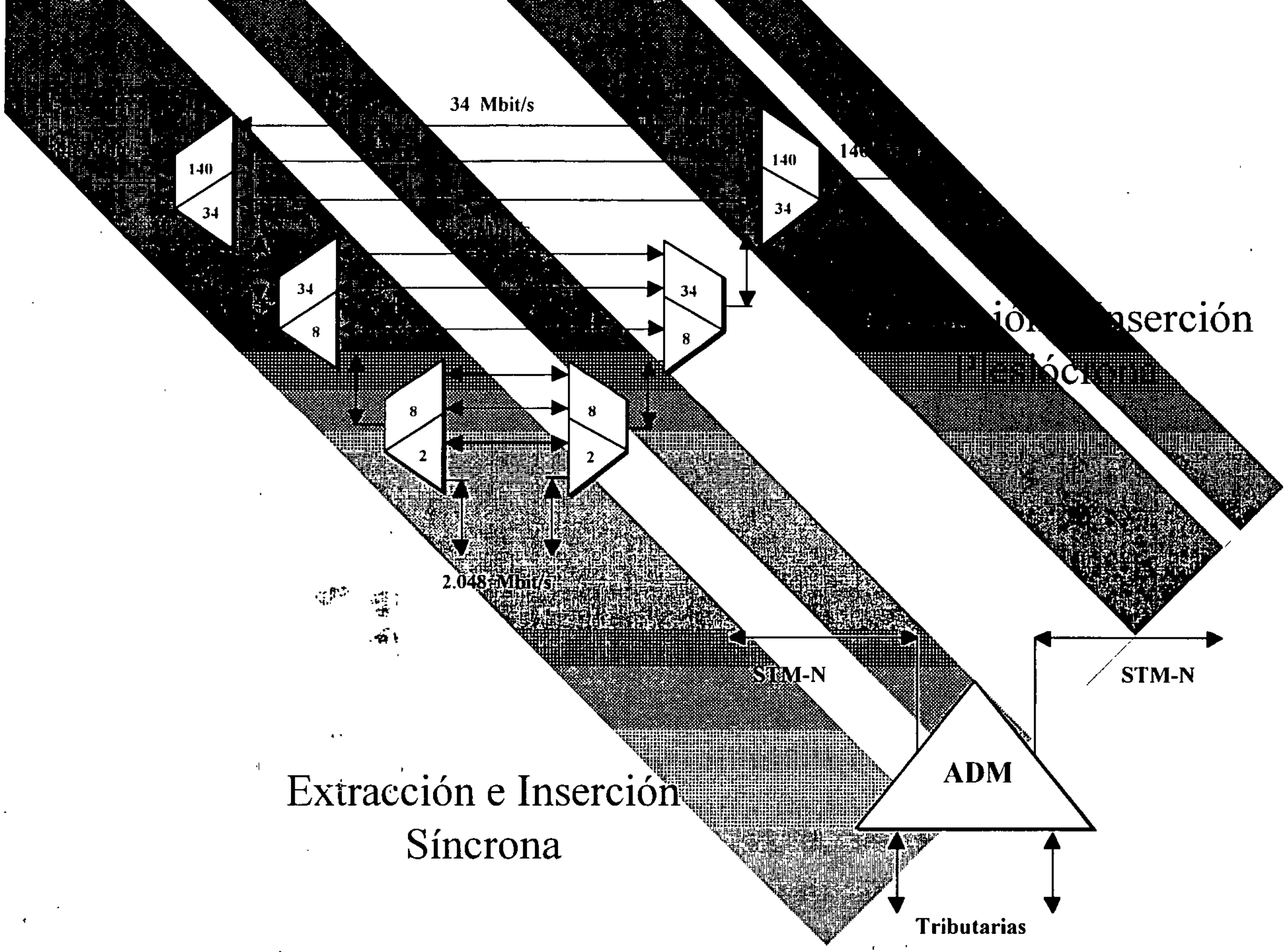
Multiplexaje
SDH

Adaptación de las Velocidades de Transmisión

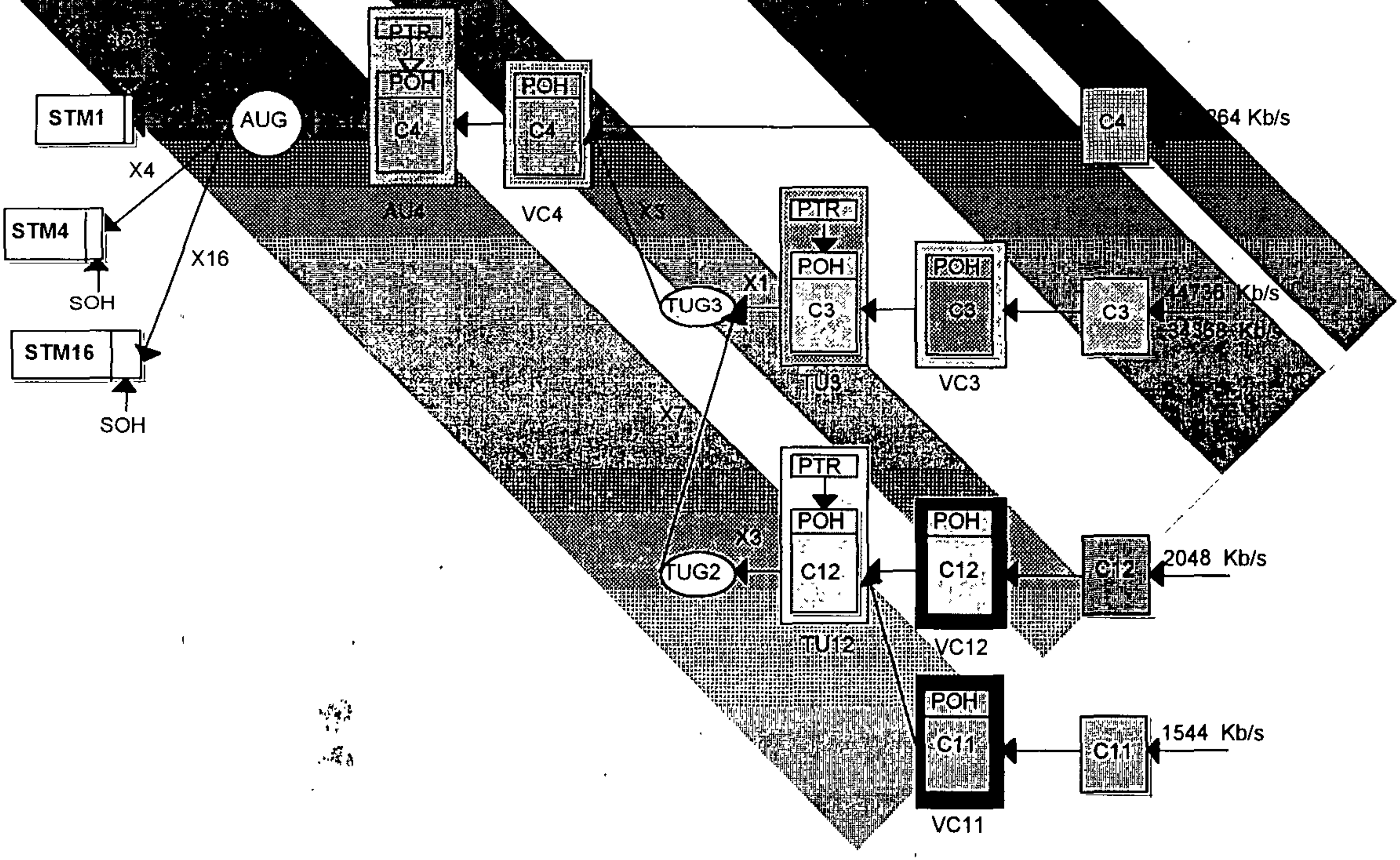
- En PDH las velocidades de las señales tributarias tienen que ser igualadas mediante bits de relleno antes de ser multiplexadas.
- En SDH no se necesita ningún tipo de igualación de velocidades de señales tributarias al momento de ser insertadas.

Structura... Trama

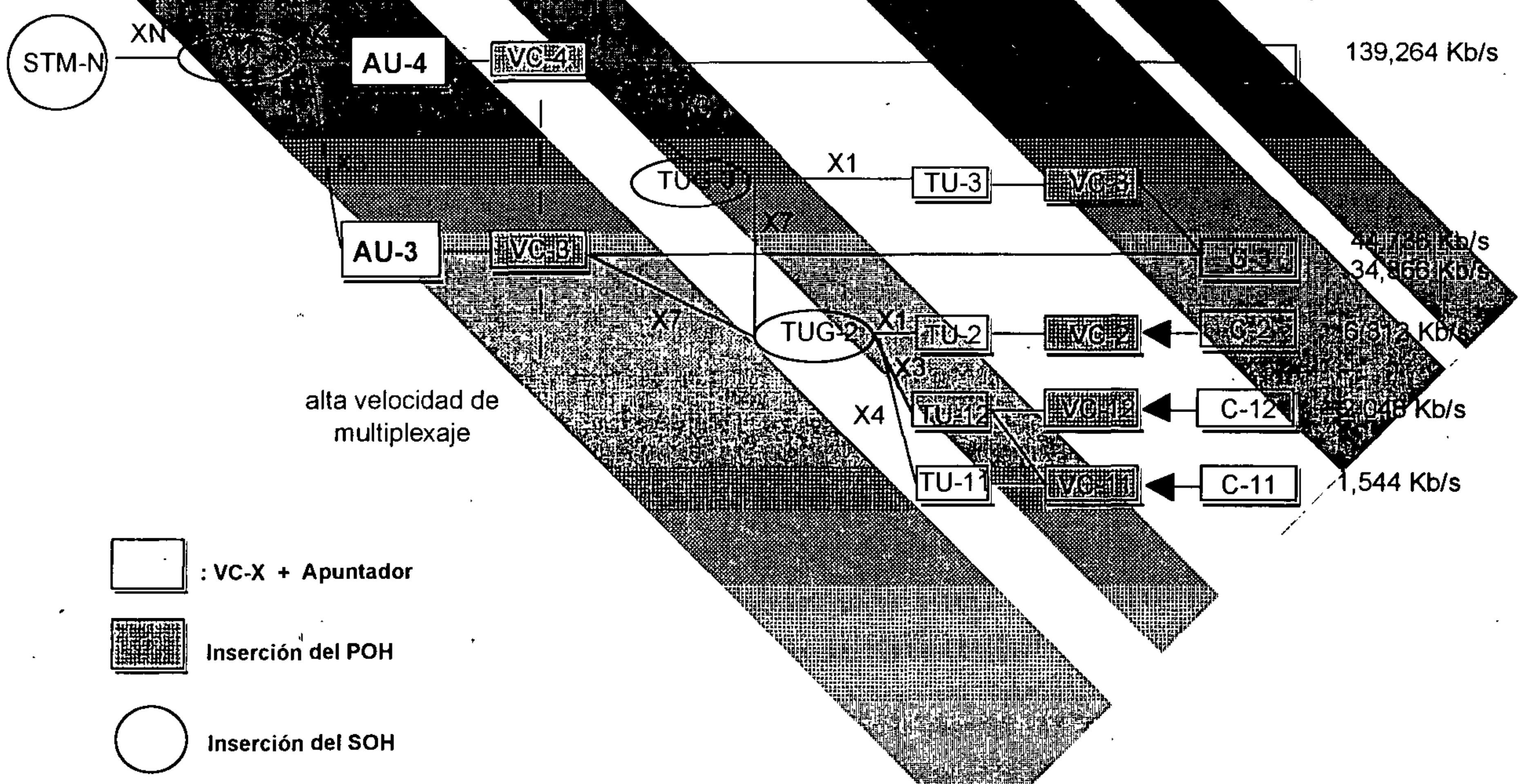


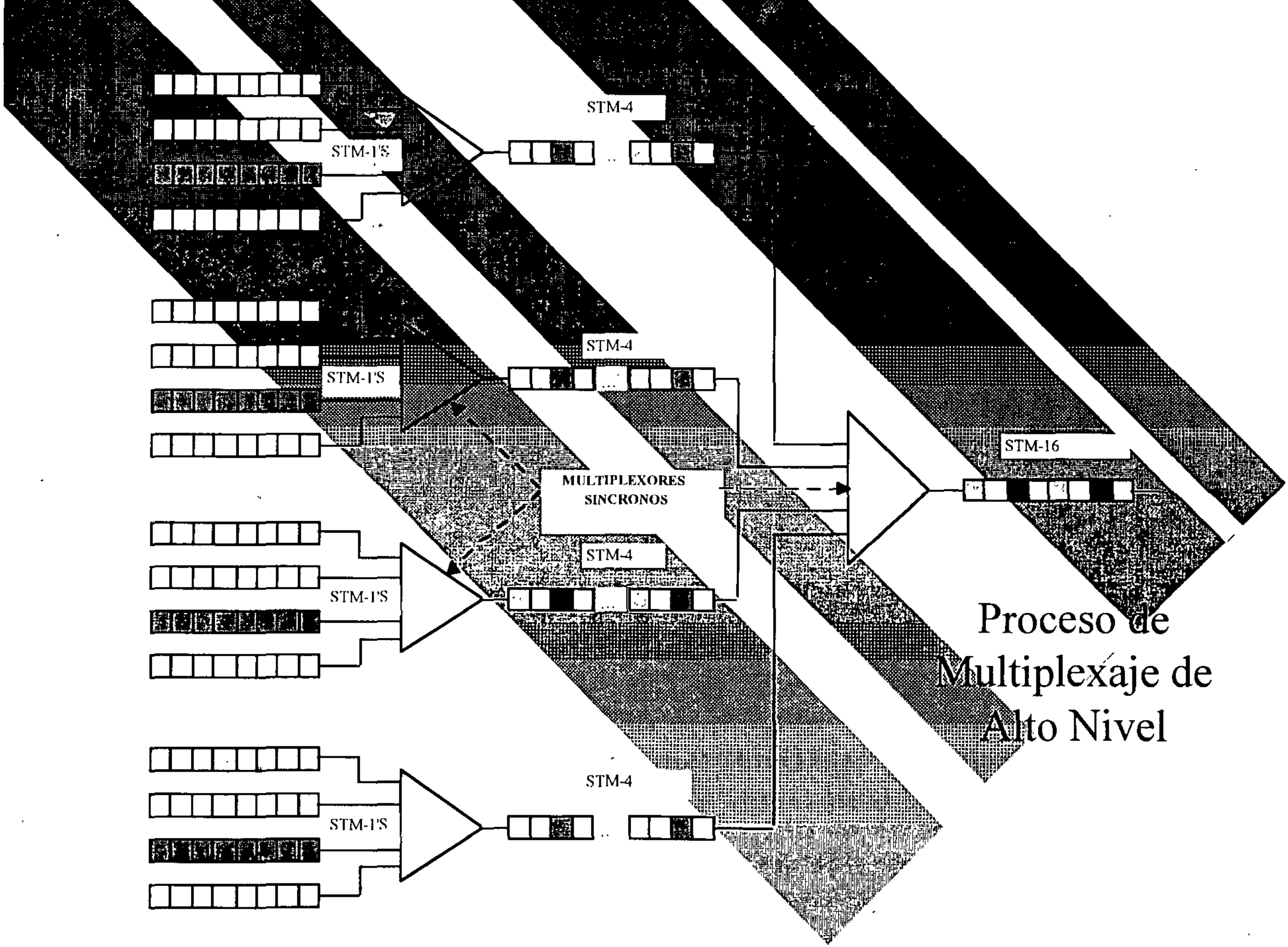

34 Mbit/s
140
34
140
34
34
8
34
8
8
2
8
2
2.048 Mbit/s
STM-N
STM-N
ADM
Tributarias
Extracción e Inserción
Síncrona

Elementos de Multiplexaje en SDH

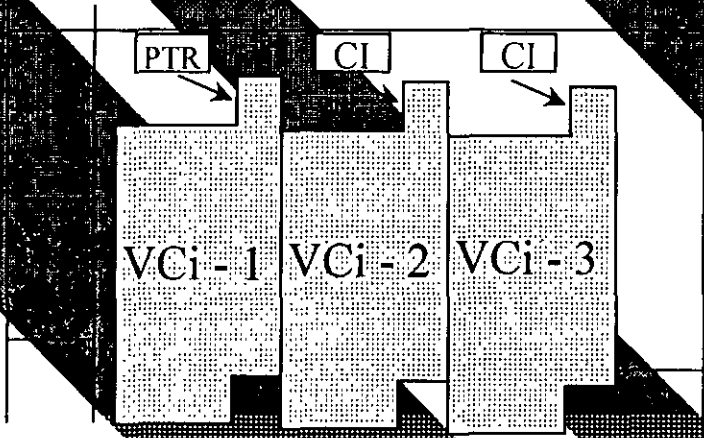


Estructura de multiplexaje





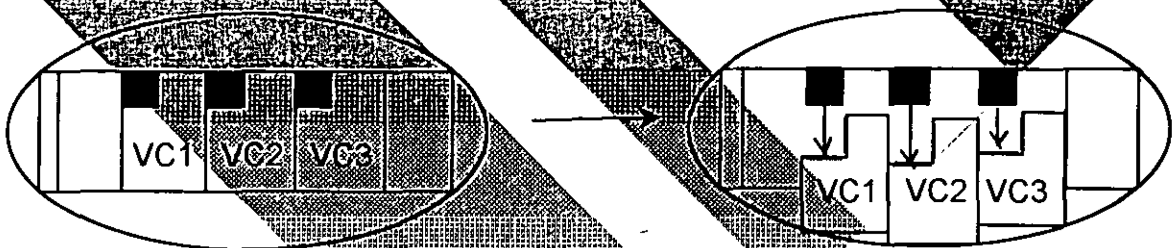
Proceso de Multiplexaje de Alto Nivel



CI = Indiacion de concatenación

[illegible]eso de
[illegible]nca[illegible]ción
[illegible]ntig[illegible]

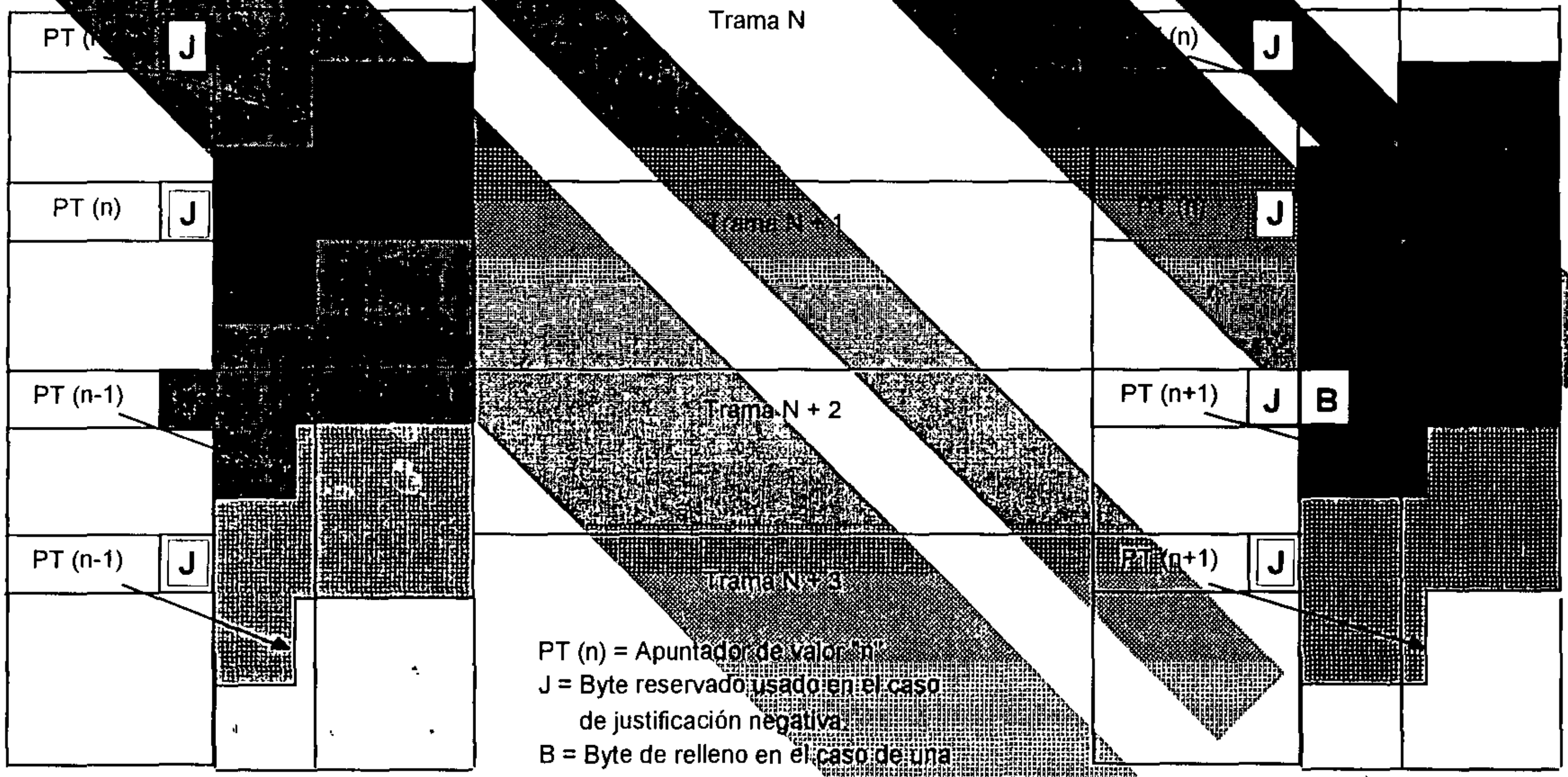
Proceso de
Concatenación
Virtual



Transmisión: El valor de los apuntadores es idéntico para los VCs.

Receptor: Utilización de los apuntadores para poner los VC en fase.

Utilización del Apuntador

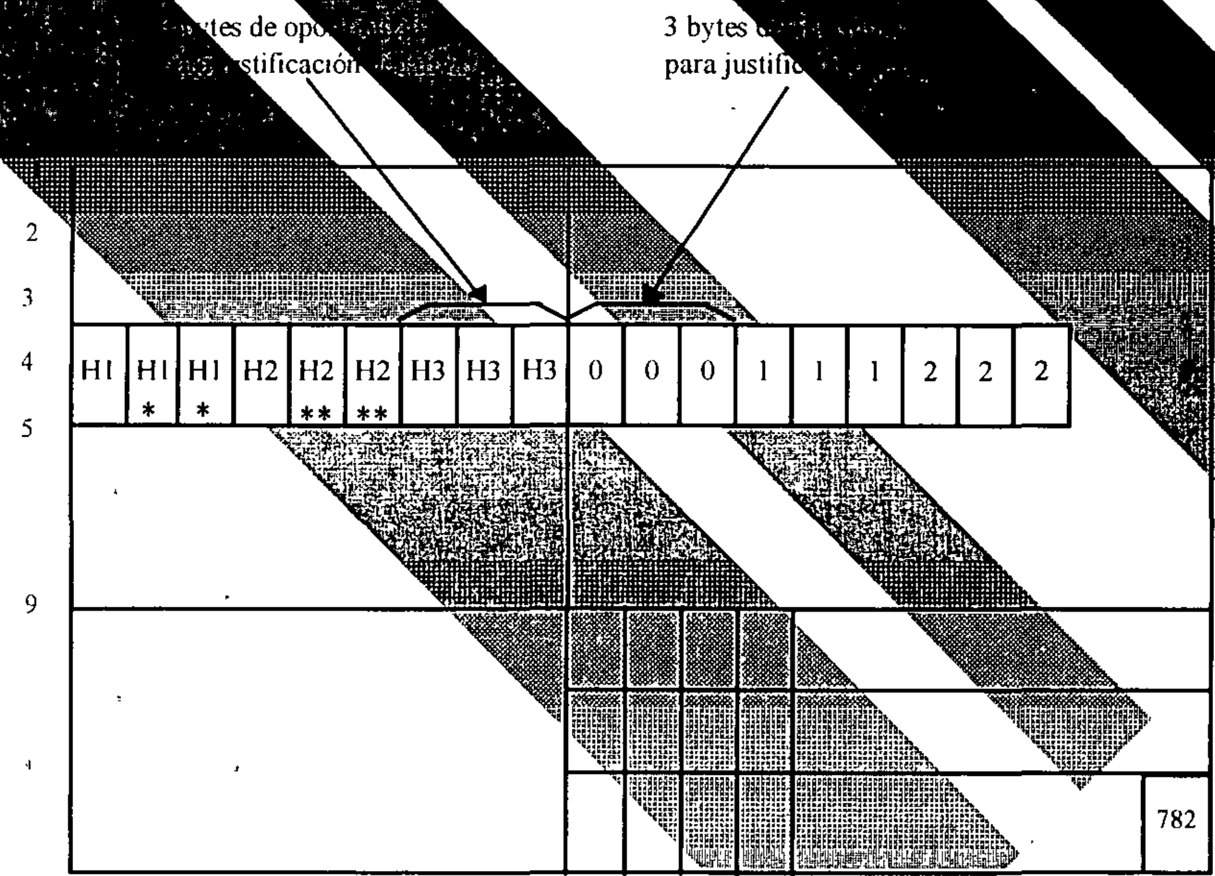


PT (n) = Apuntador de valor "n"
J = Byte reservado usado en el caso de justificación negativa.
B = Byte de relleno en el caso de una justificación positiva.

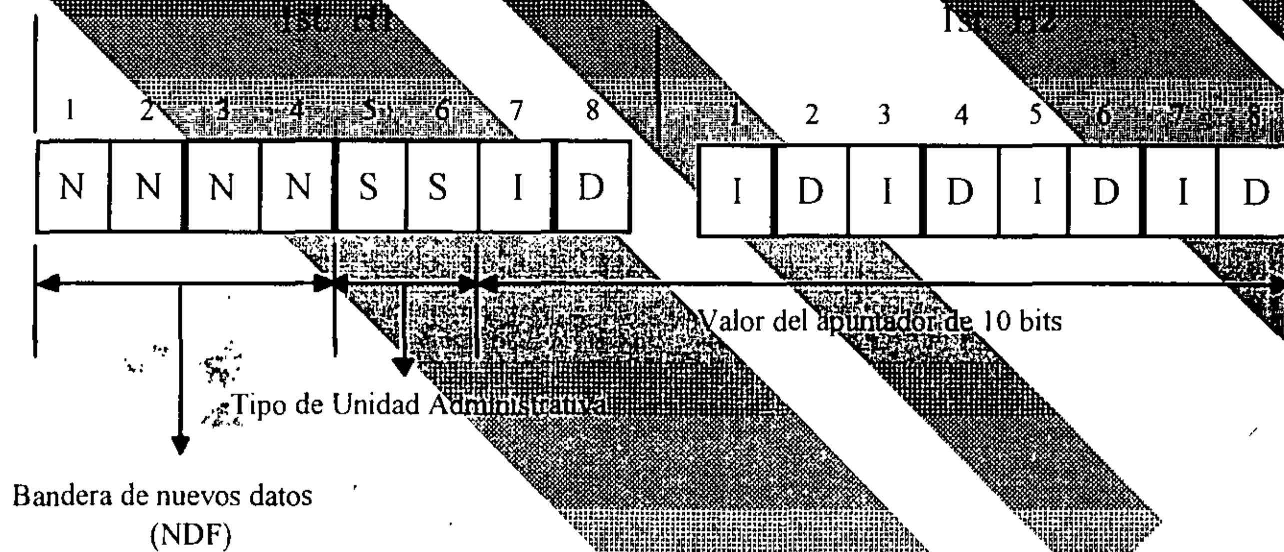
a) Justificación Negativa

b) Justificación Positiva

Bytes de Justificación



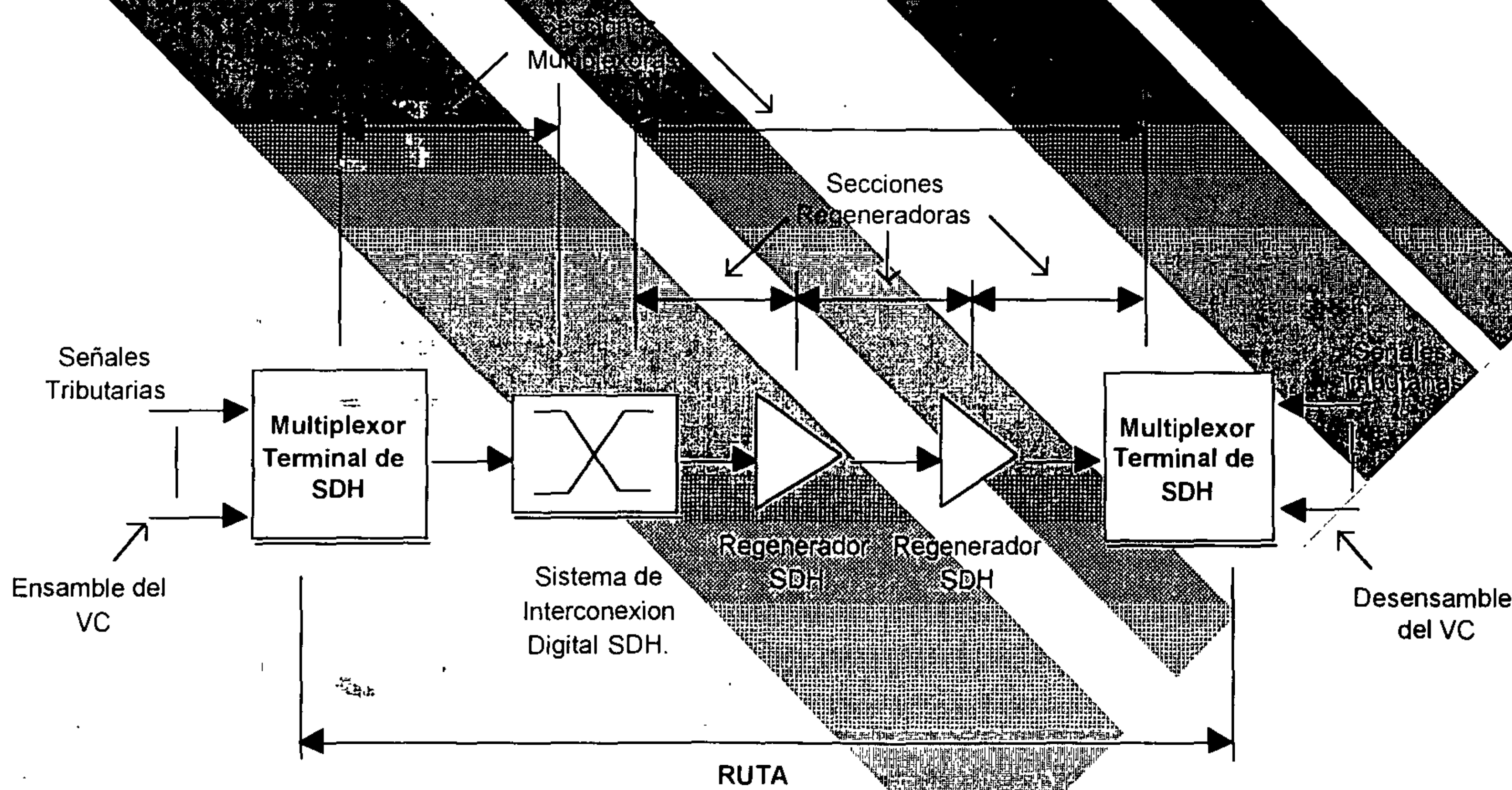
Valor del Indicador del Apuntador



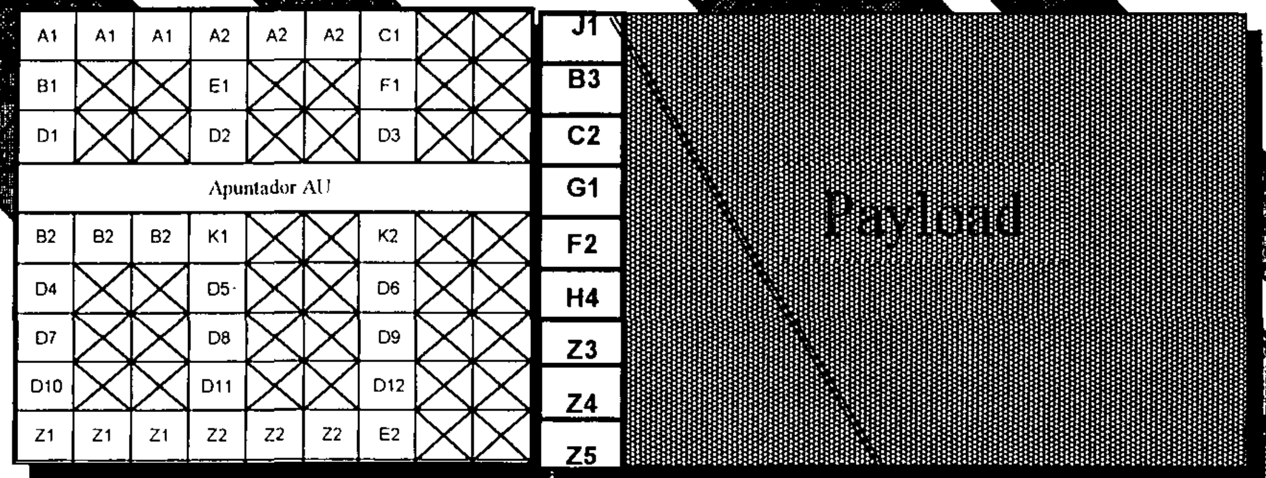
Encabezado de Sección

[illegible]	[illegible]	A1	[illegible]	A2	A2	C1		
[illegible]			E1			F1		
D1			D2			D3		
Apuntador AU								
B2	B2	B2	K1			K2		
D4			D5			D6		
D7			D8			D9		
D10			D11			D12		
Z1	Z1	Z1	Z2	Z2	Z2	E2		

Sección de Regeneración y Multiplexaje

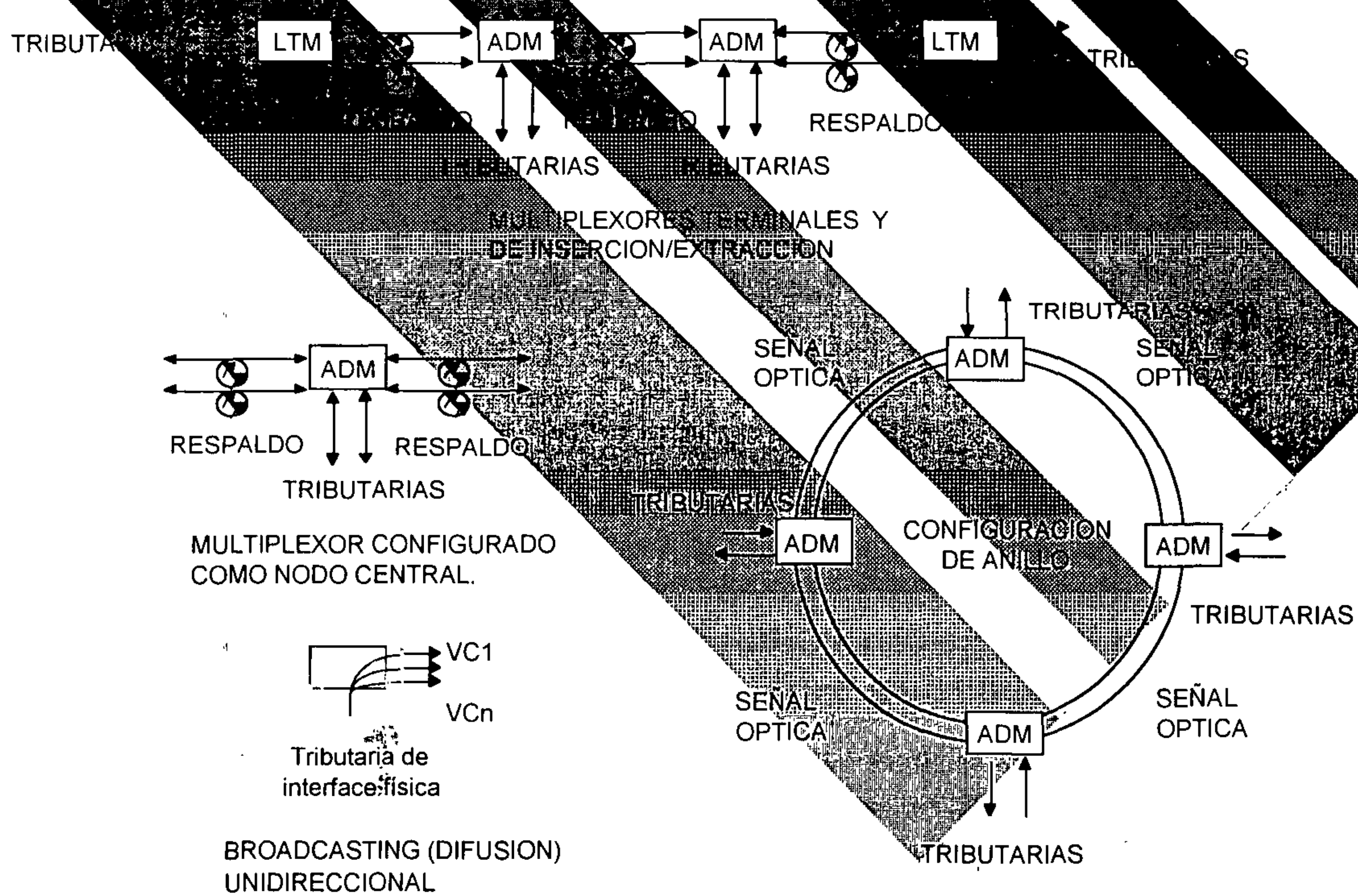


Localización de los bytes en el POH

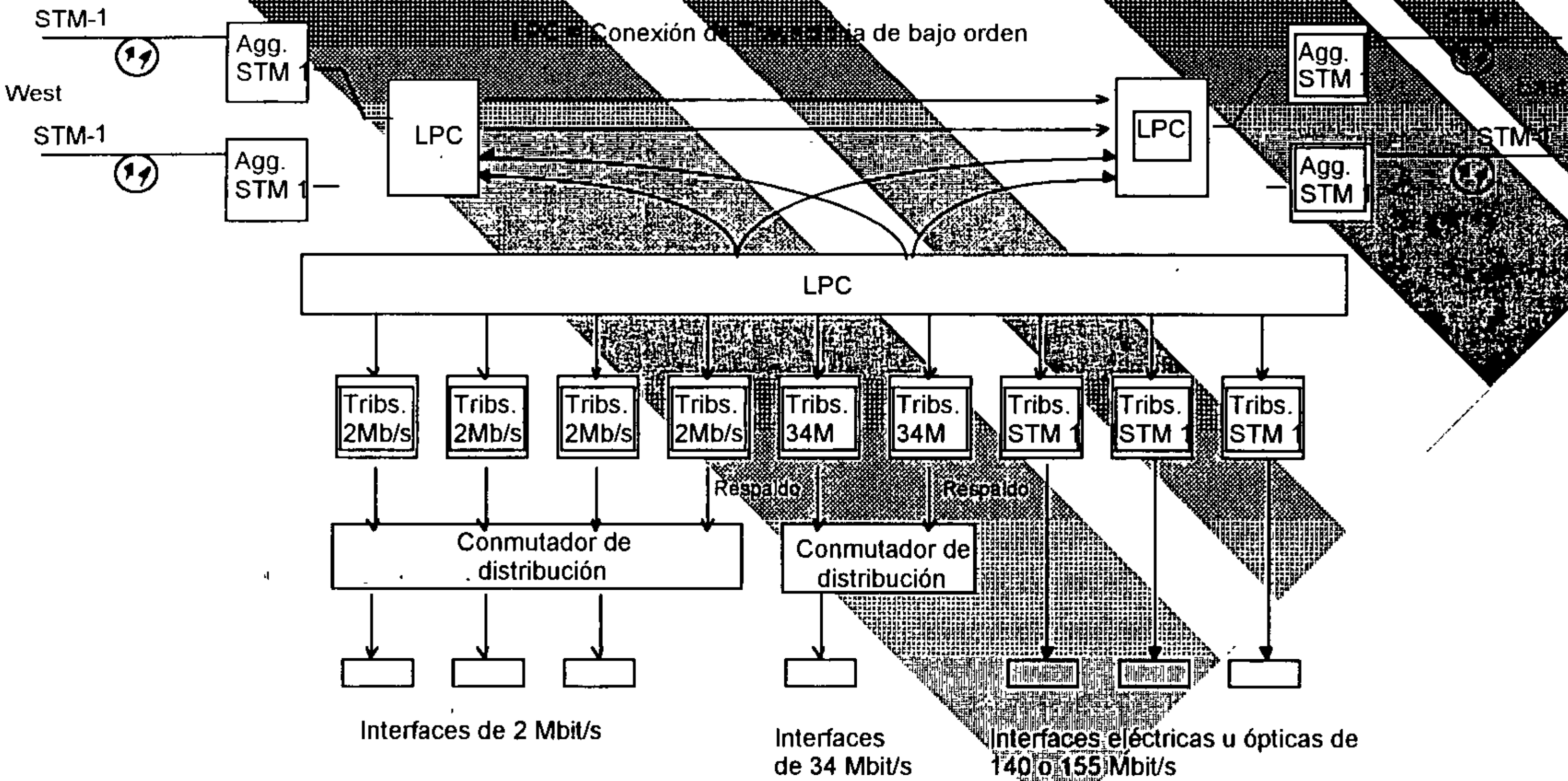


J1	Trazado de Ruta
B3	BIP-8
C2	Etiqueta de Señal
G1	Estado de Ruta
F2	Canal Usuario
H4	Multitrama
Z3	Crecimiento
Z4	Crecimiento
Z5	Crecimiento

Configuraciones y Funcionalidad de Equipos ADM



[illegible]gra[illegible]a bloques[illegible]un[illegible]uipo ADM



Multiplexor Terminal de línea

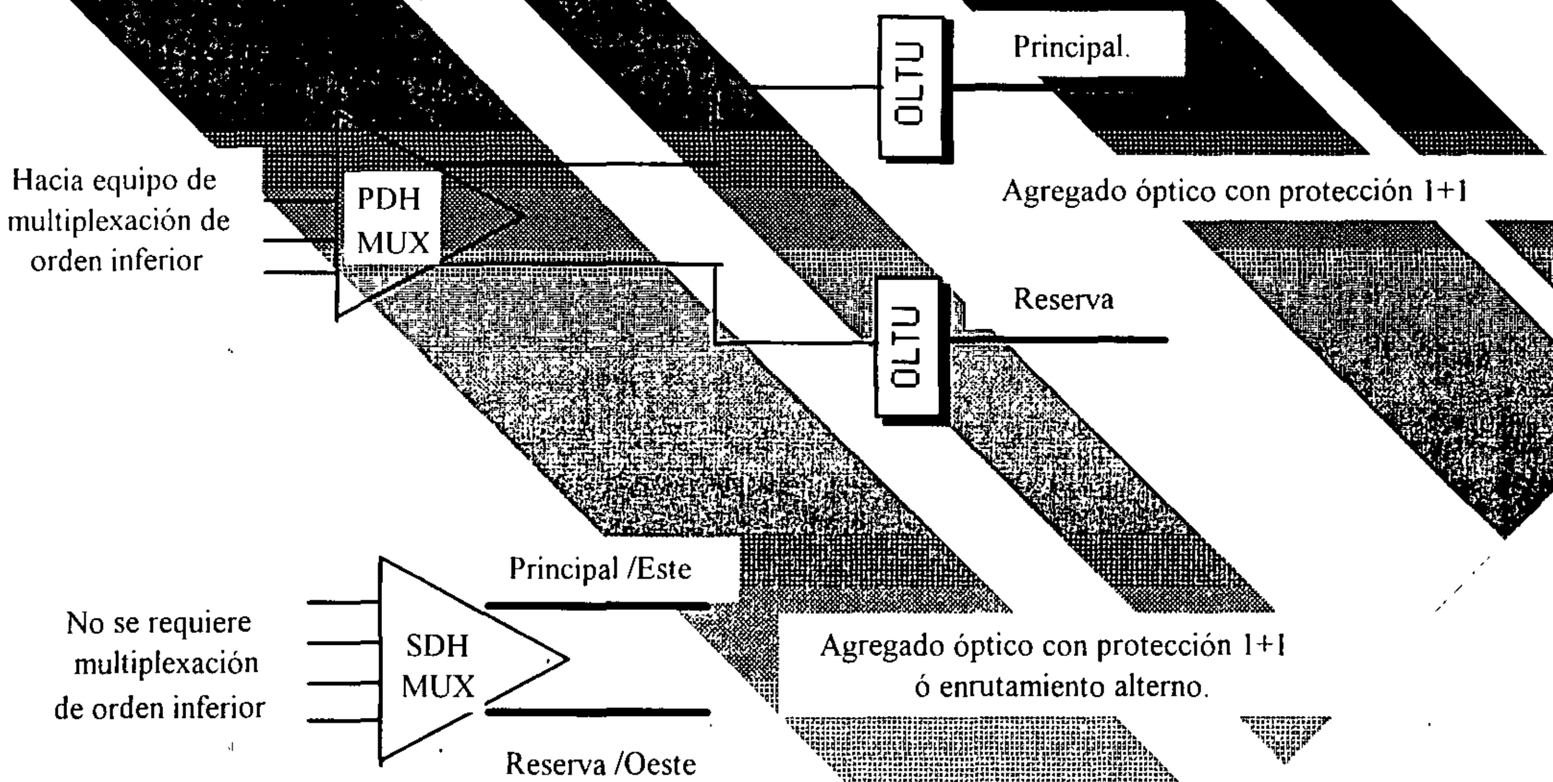


Diagrama a Blo[illegible] [illegible]cionales de un [illegible]

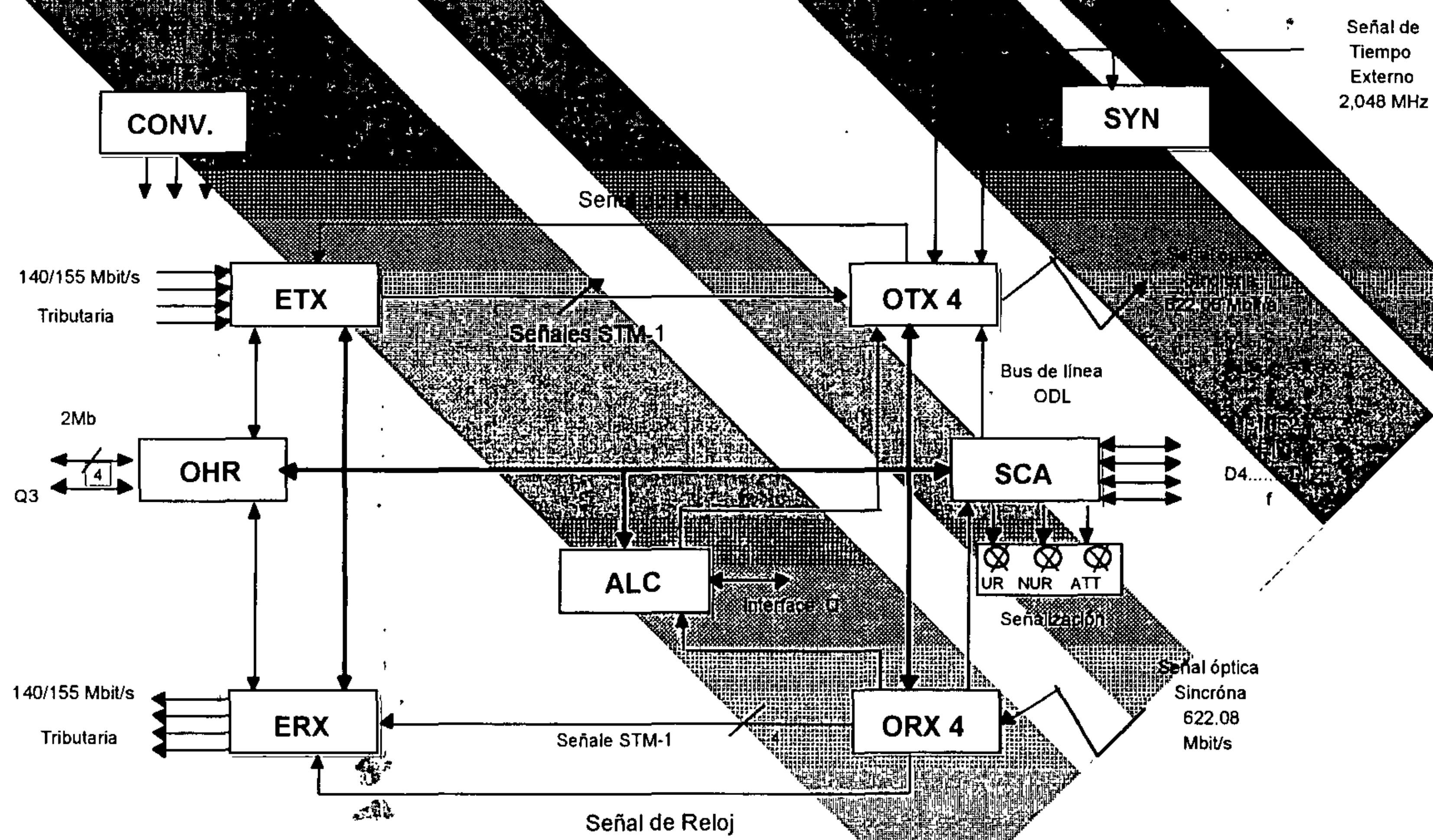


Diagrama Funcional de un Equipo de Cross-conexión

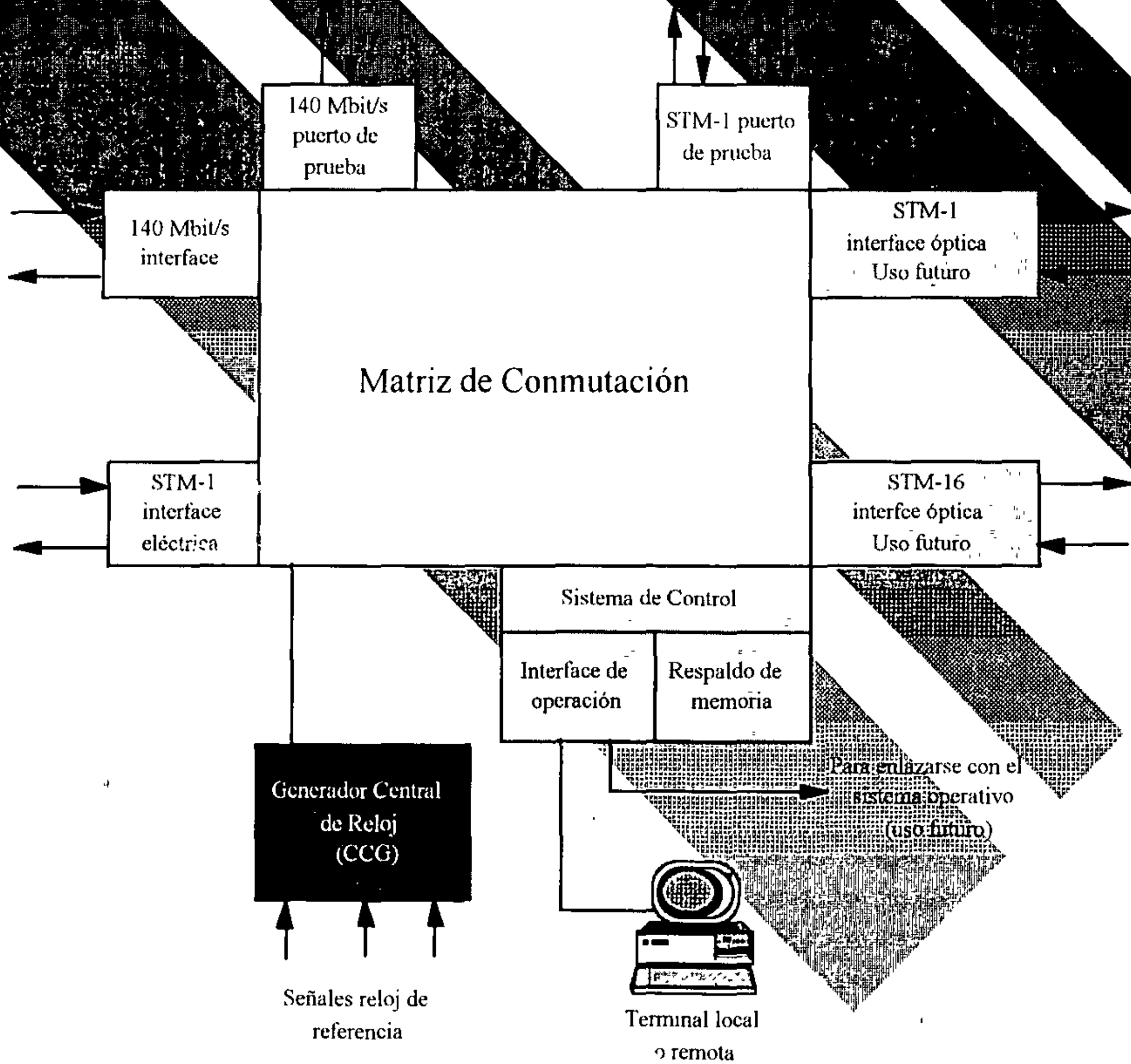
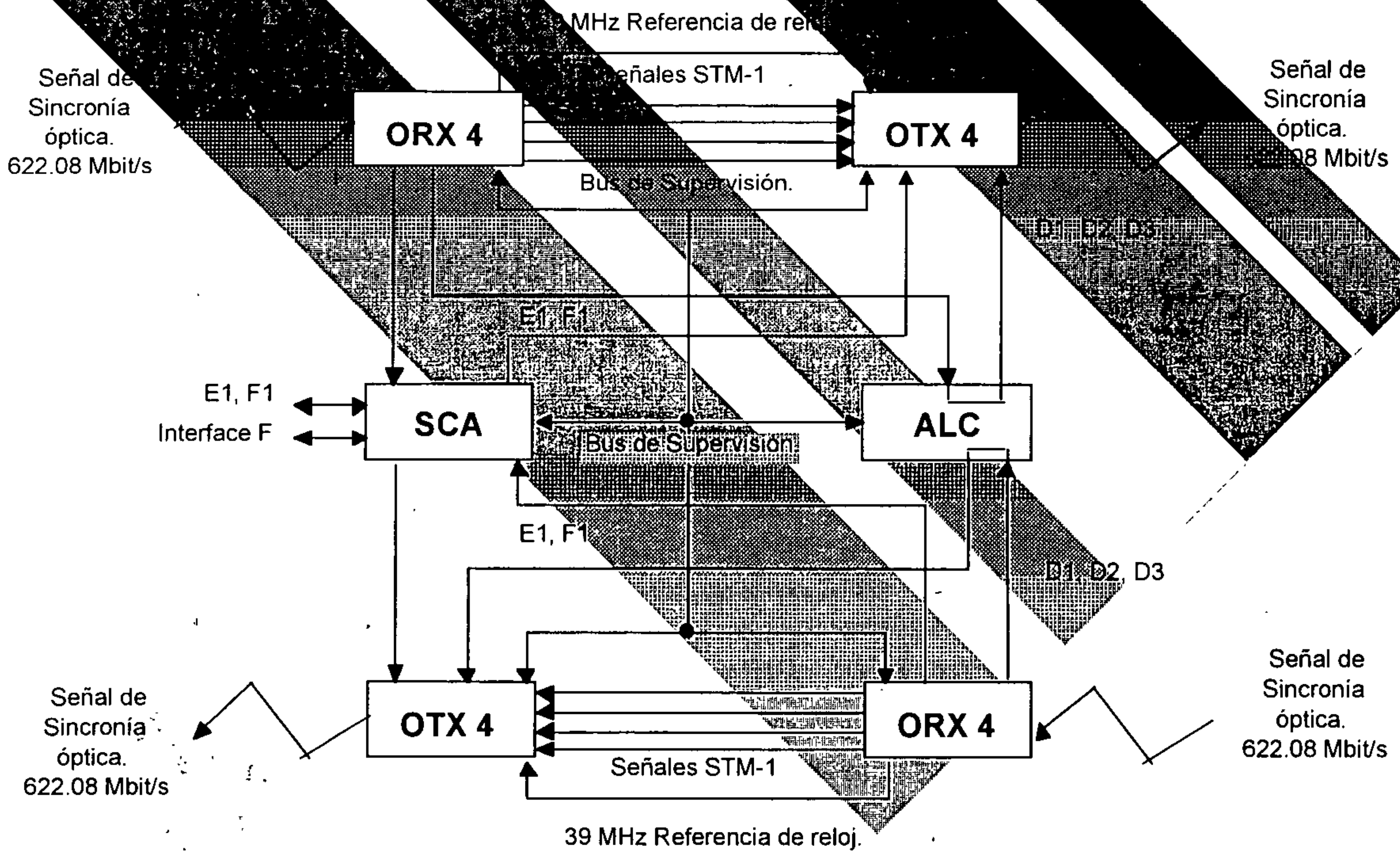
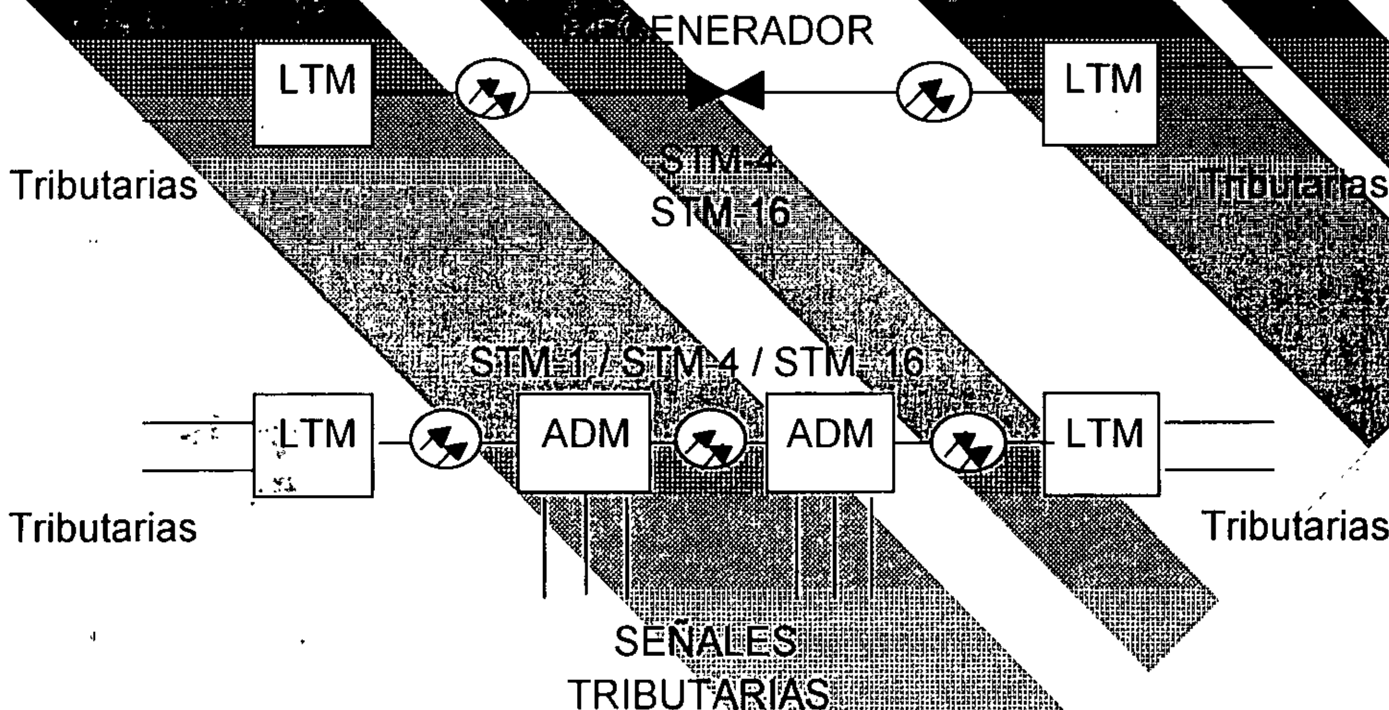


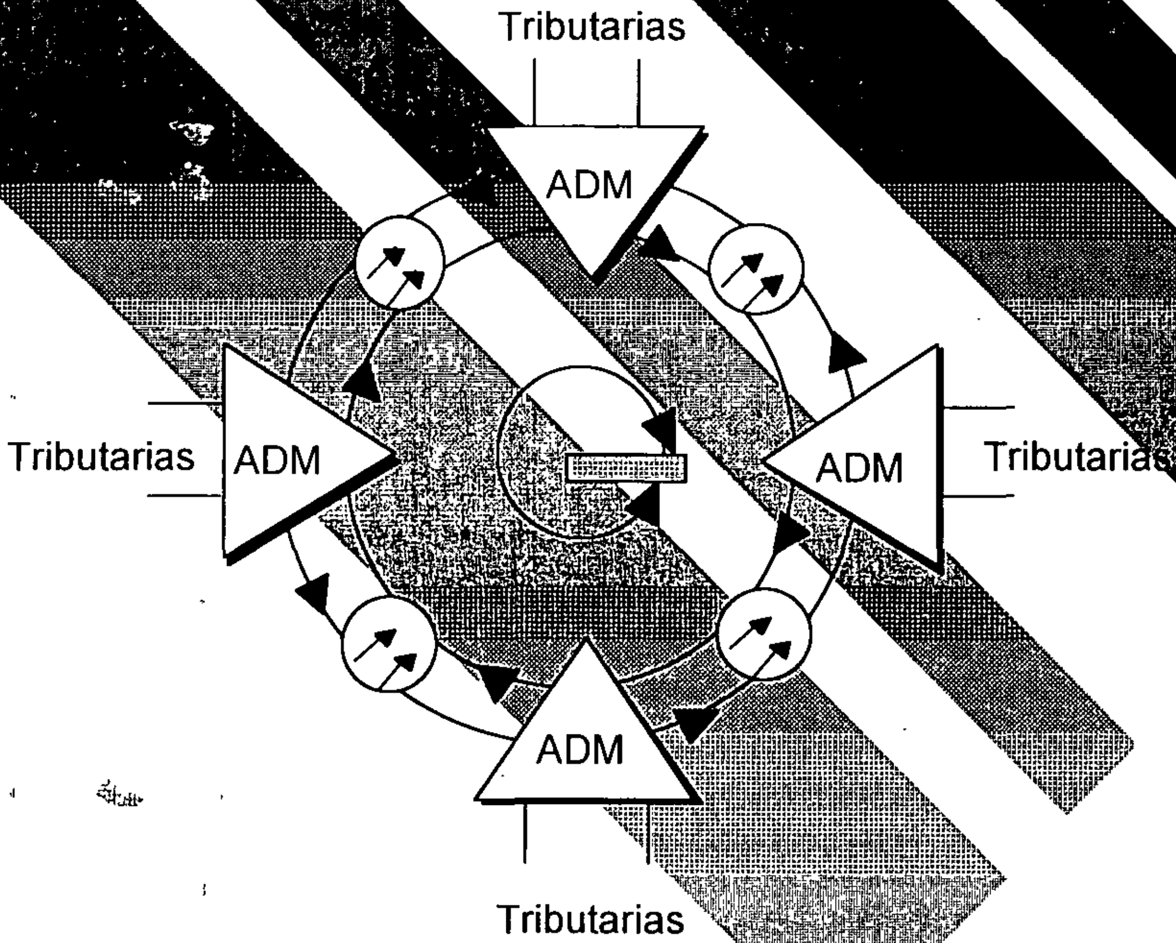
Diagrama a bloques del Regenerador



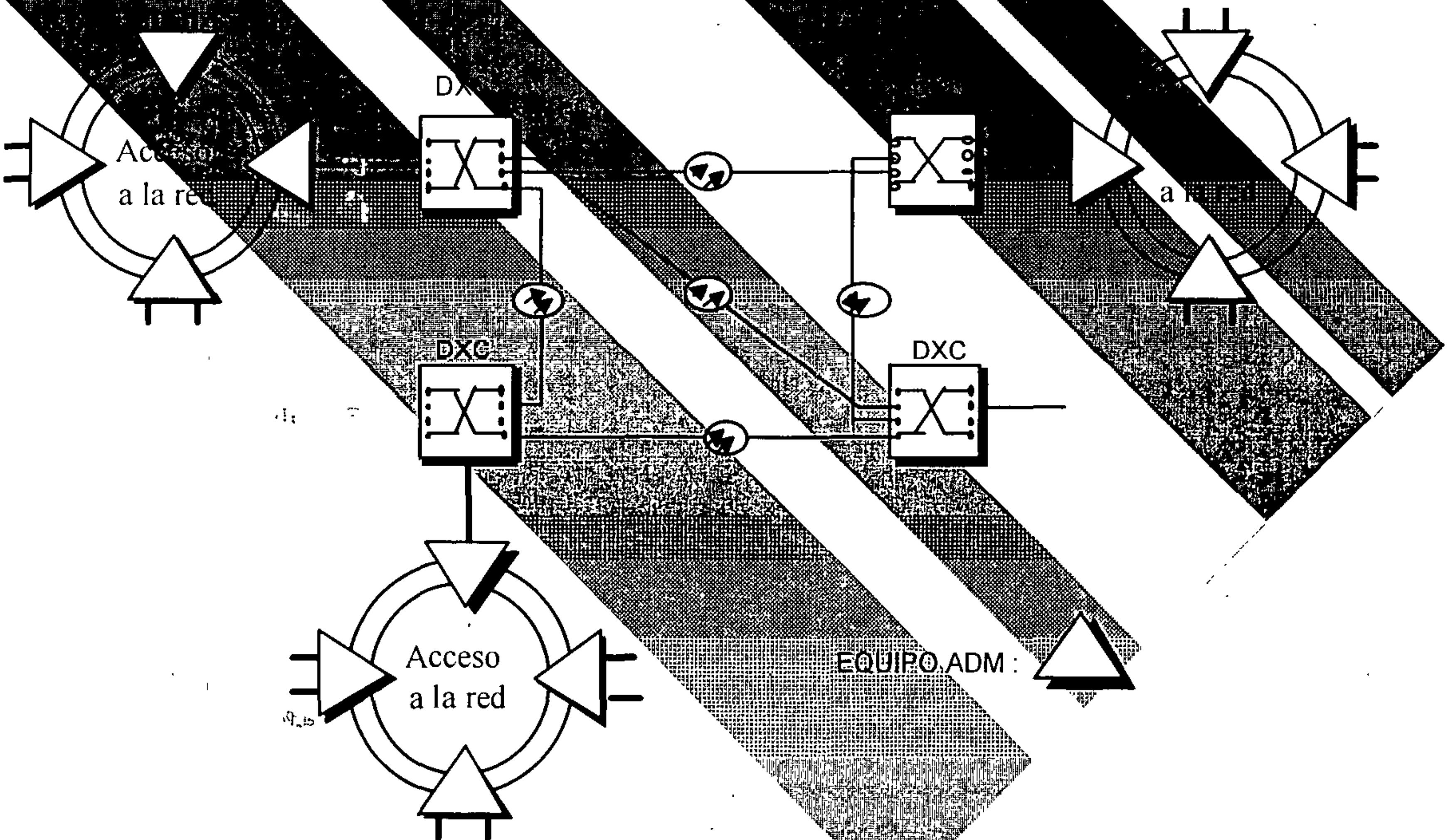
Configuración Punto a Punto o Bus con Regeneradores o Nodos Intermedios



Topología de anillo



Mal… y Anillo… …figuración defin…

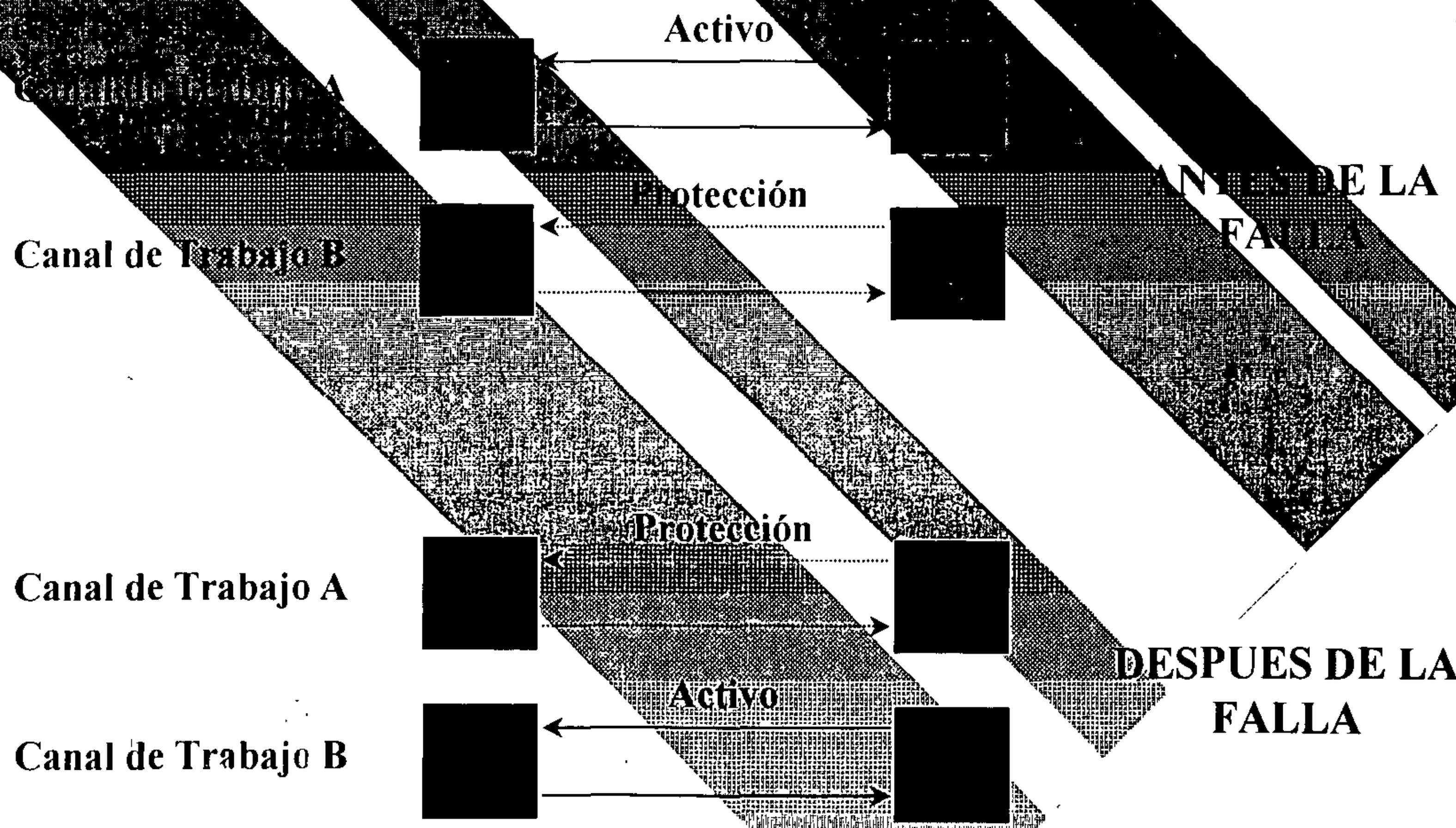


Tipos de Protección y Aplicaciones

Falla	Protección		
	Redundancia	Ejemplo	[illegible]
Componente	Tarjeta	EPS: Conmutación de protección de equipo	EPS N [illegible] EPS 1 + 1
	Tarjeta y cable	APS Conmutación de Protección Automática (Cables en el mismo ducto)	APS N [illegible] APS 1 + 1 APS N : 1 APS 1 : 1
Enlace Causa: - excavación - sabotaje	Ruta	Protección de cable con diferentes rutas.	APS 1 + 1; 1 : 1 con 2 rutas - Anillo - Malla
Nodo Causa: - incendio - falla de energía	Estación	Protección de nodo	- Anillo - Malla

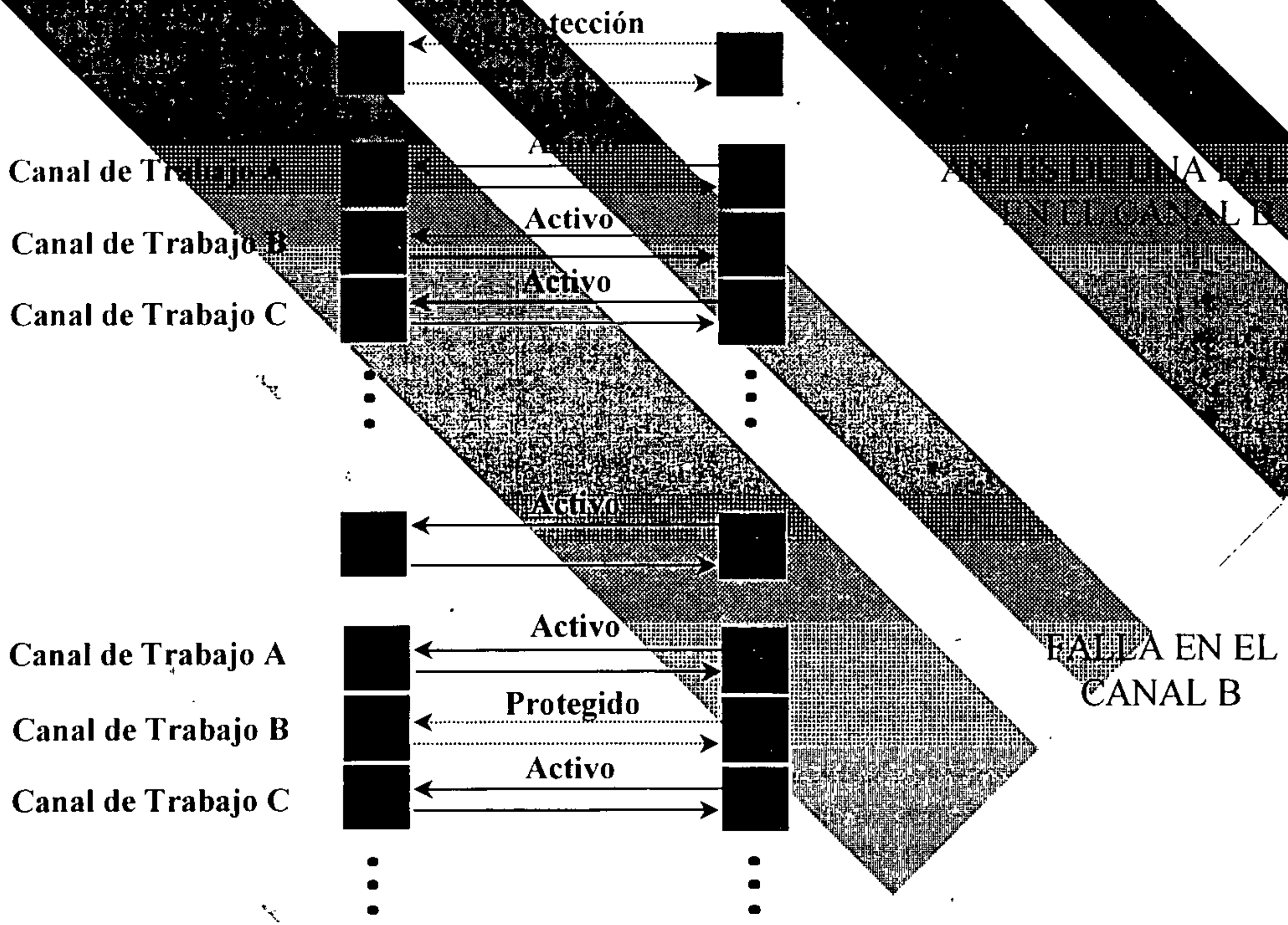
Esquemas de Conmutación

Protección 1+1 (No reversivo)

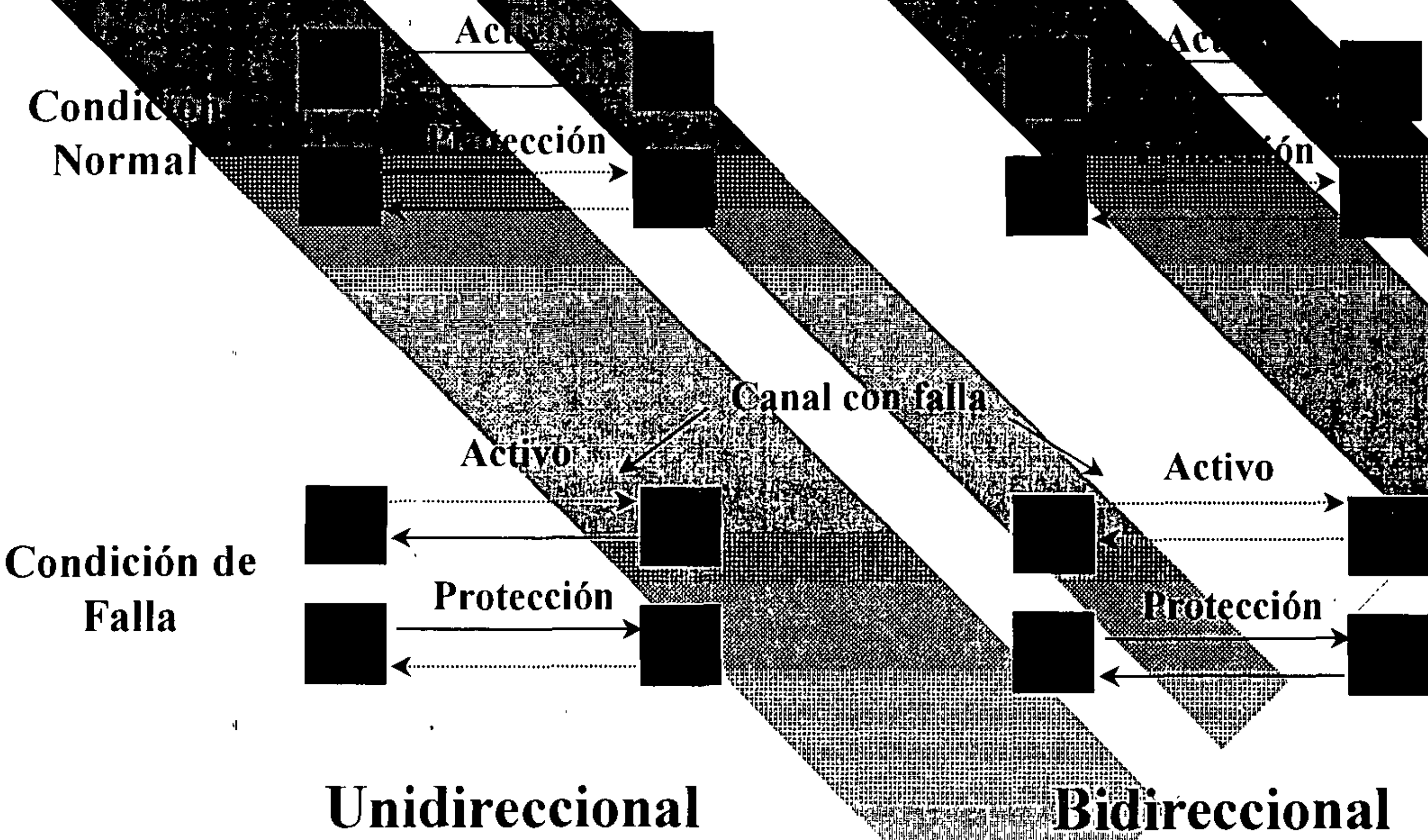


Esquemas de Conmutación

Protección 1:N



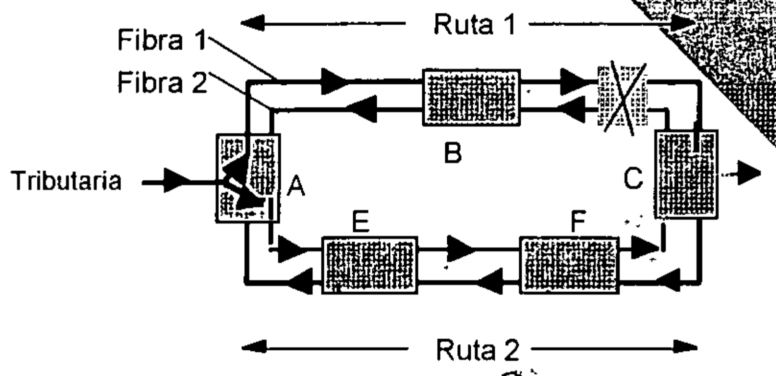
Modos de Conmutación



Anillo de Protección Dedicada

Restauración del anillo unidireccional

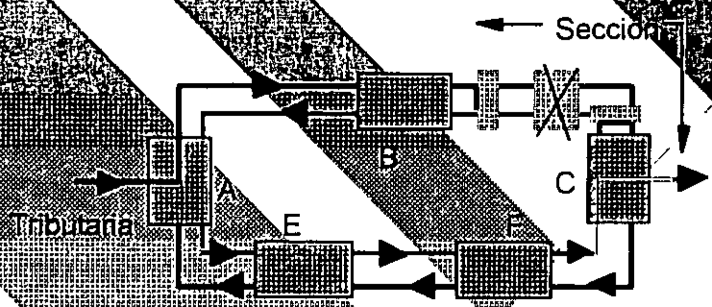
Protección de Ruta



Anillo de Protección Compartida

Restauración del anillo bidireccional

Protección de Sección



Red de Sincronización

PRS: Fuente de Referencia Primaria - Maestro

SRS: Fuente de Referencia Secundaria - Esclavo

Ventajas de una Red SDH

- Simplificación de la red
- Capacidad de supervivencia
- Control mediante software
- Ancho de banda
- Normalización

LMDS

Broadband Wireless Access

&

Fixed Wireless Access

Carlos Bueso
Wireless Business Development

NORTEL NETWORKS

Agenda

LMDS (BWA) & FWA

- What is it?
- Why is it important?
- Where does it fit?
- How does it work?
- Who will purchase it?
- Future Evolution
- Summary

NORTEL NETWORKS

NORTEL NETWORKS

What is it?

BWA: What is it?

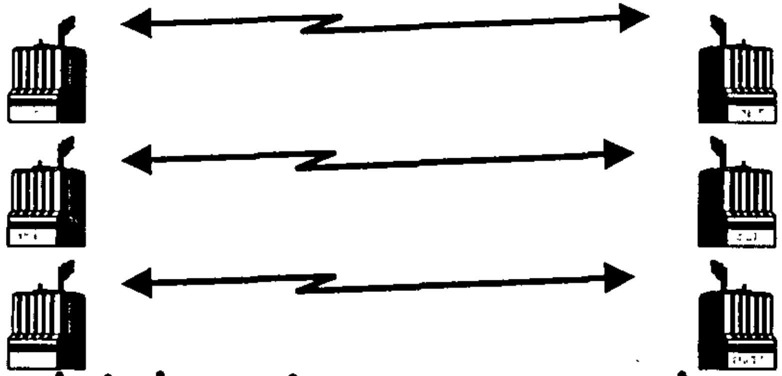
LOCAL
MULTIPOINT
DISTRIBUTION
SERVICE

A broadband wireless (wireless fiber) point to multipoint communication system for digital two way data, internet, video and voice services.

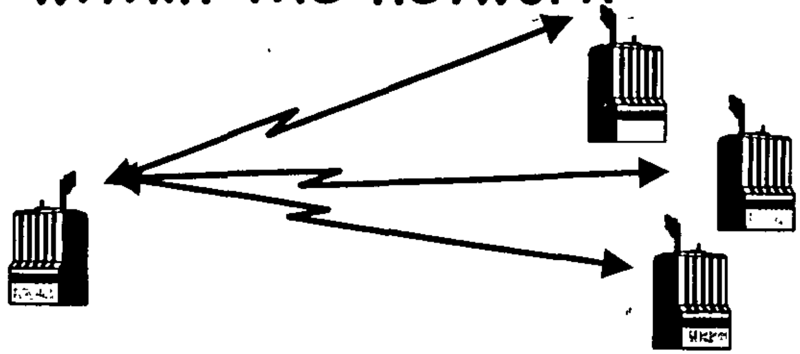
NORTEL NETWORKS

LMDS: What is it ?

Not new technology (Microwave Transport), rather a new implementation of existing technology



Multiplexing multiple microwave paths through a single infrastructure path fosters network and cost economies within the network



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LMDS: What is it?

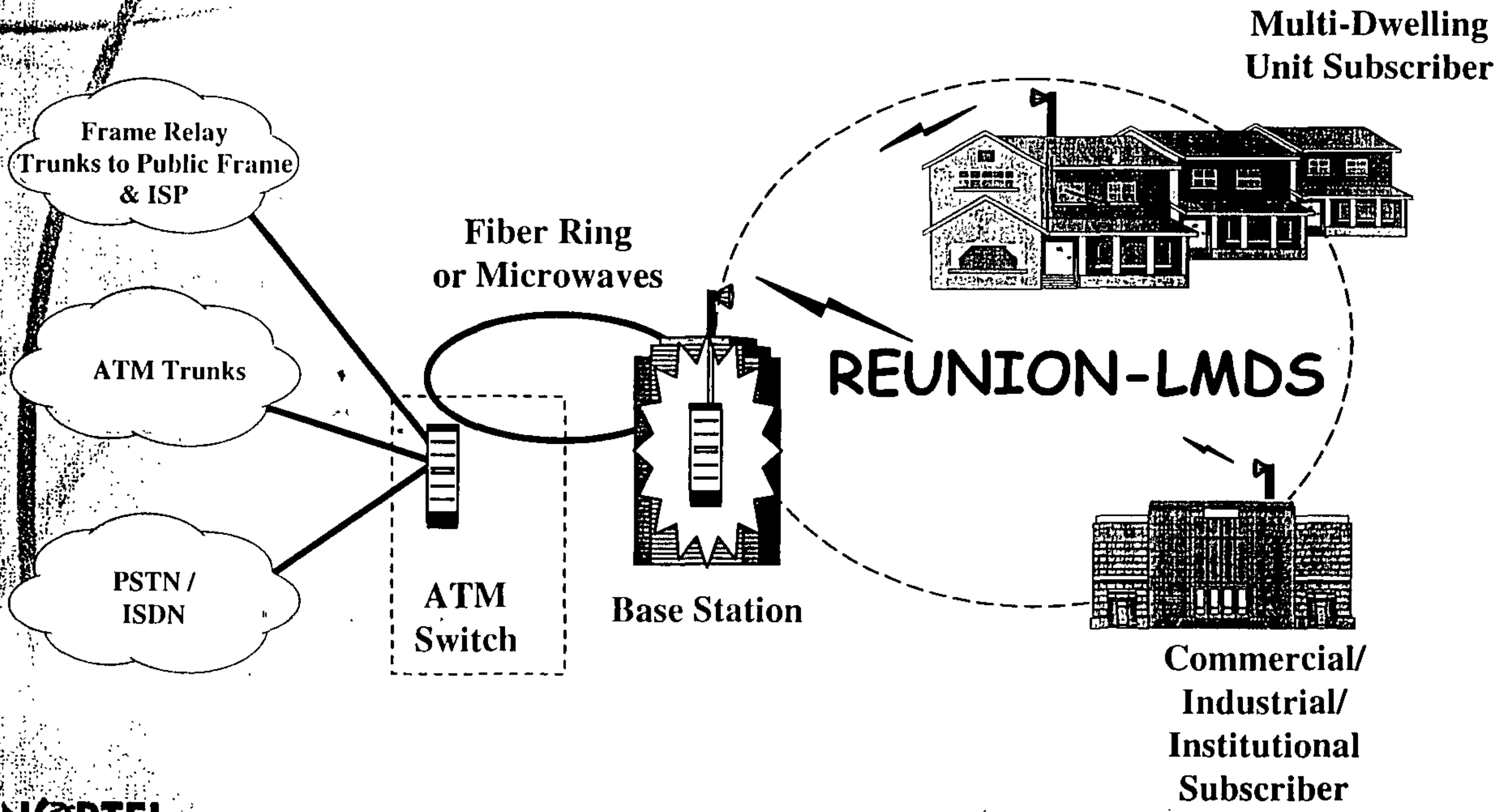
A fixed Wireless technology capable of delivering a multitude of current or future services

- Voice, Fax, HS Data
- Internet & Intranet Access
- Leased lines E1, nxE1
- ATM / FR Networks
- Video Conferencing
- Wireless LAN

NORTEL NETWORKS

LMDS: What is it?

High capacity Pt to Multi-pt Microwave systems in the 2.5 GHz - 40 GHz range for local distribution of services



LMDS/BWA: What is it?

- Uses spectrum from 2.5 GHz to 40 GHz
- Point to multipoint radio access
- Demand driven deployment
- Virtual ATM air interface supports voice, data and video services
- Bandwidth is shared among users in a cell

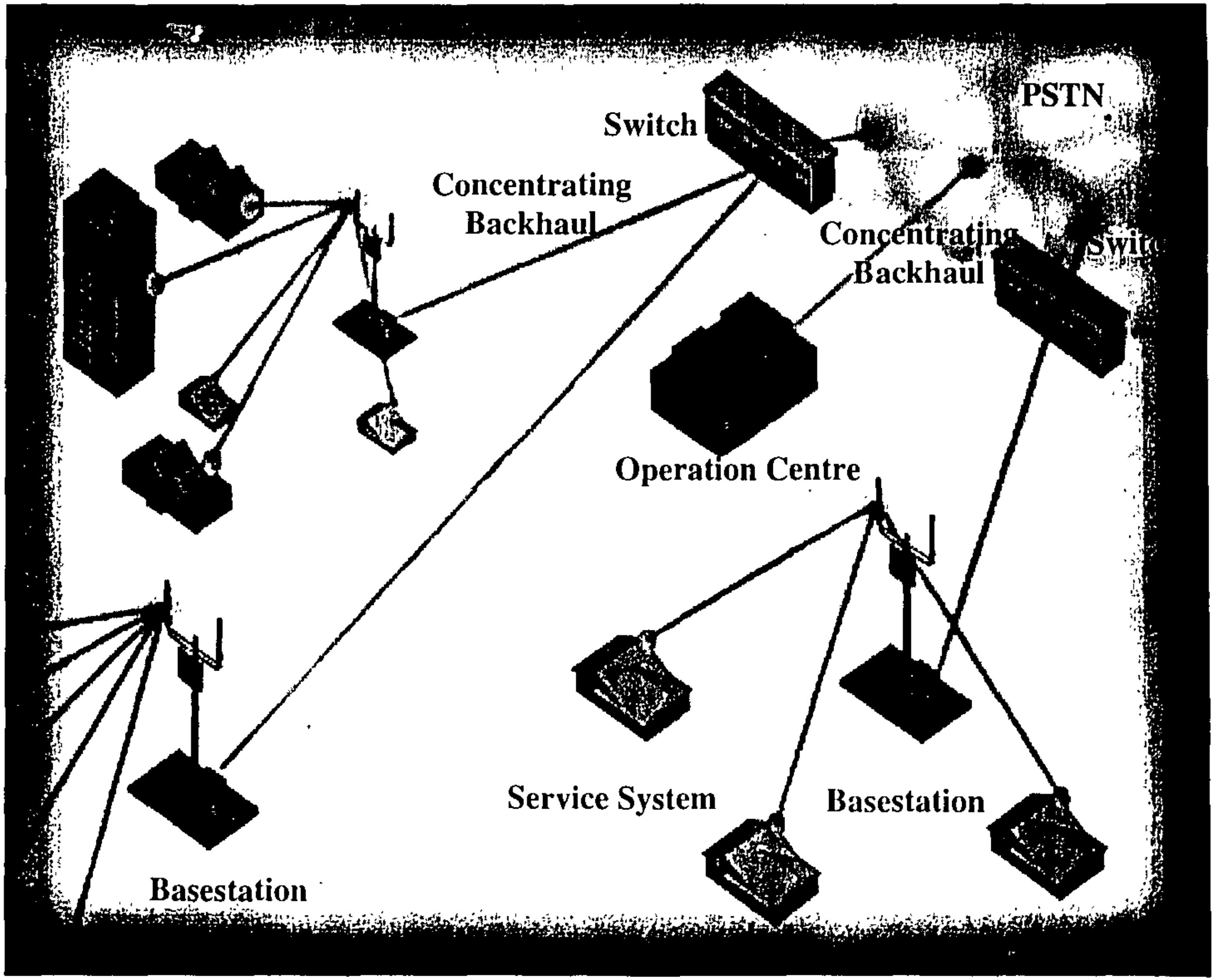
NORTEL NETWORKS

FWA: What is it?

FIXED
WIRELESS
ACCESS

A NARROW BAND wireless (wireless copper) point to multi-point communication system for digital two way data, fax, internet and voice services.

FWA:What is it ?



NORTEL NETWORKS

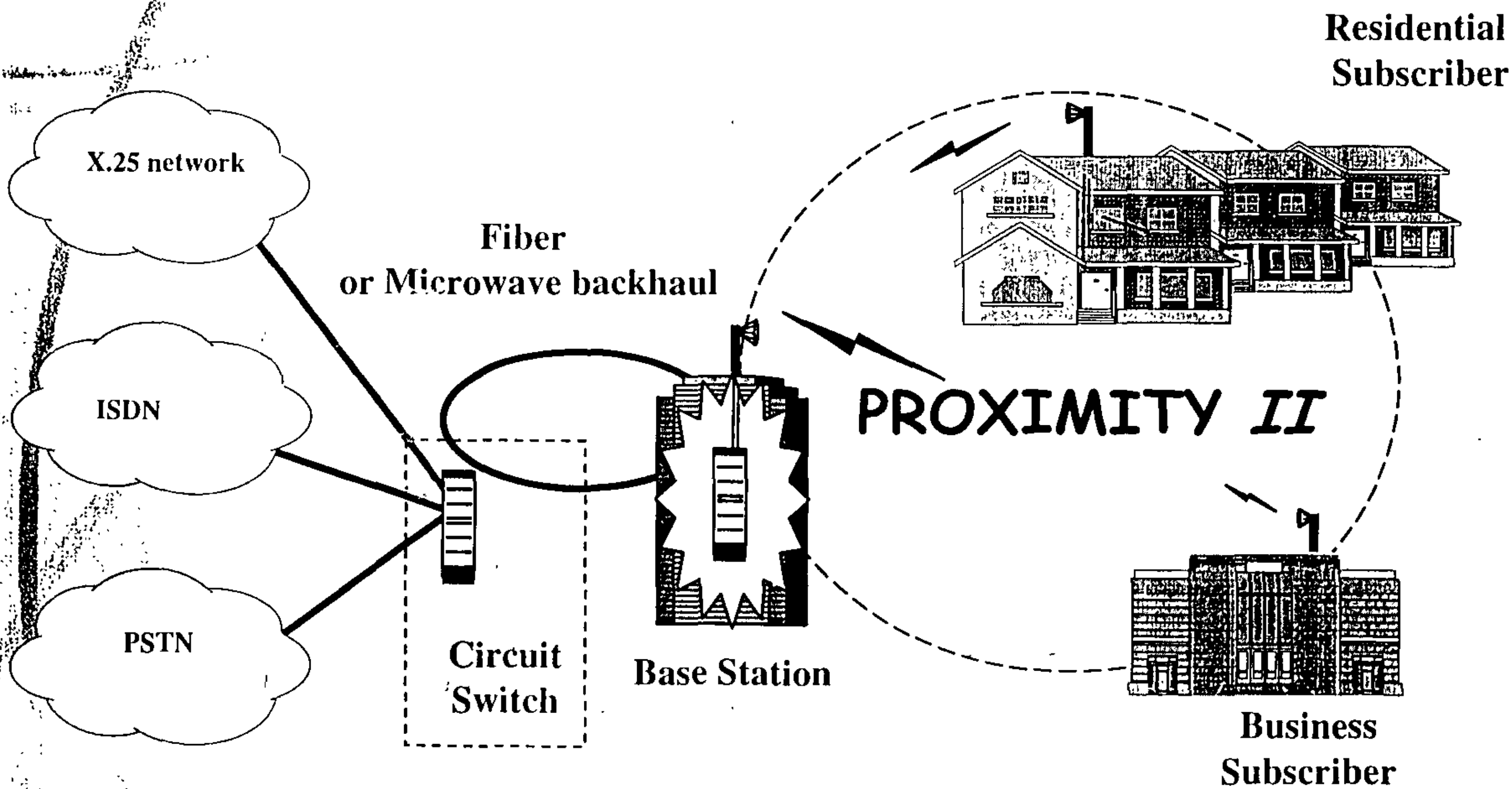
FWA: What is it?

A technology capable of delivering high quality telecommunications services

- Voice
- Fax
- Voice Band Data
- Internet & Intranet Access at up to 128kbps
- ISDN; 1B+D, 2B+D
- 'Always on' e-mail

FWA: What is it?

High capacity microwave access systems in the 1.9-3.5 GHz band for local access



NORTEL NETWORKS

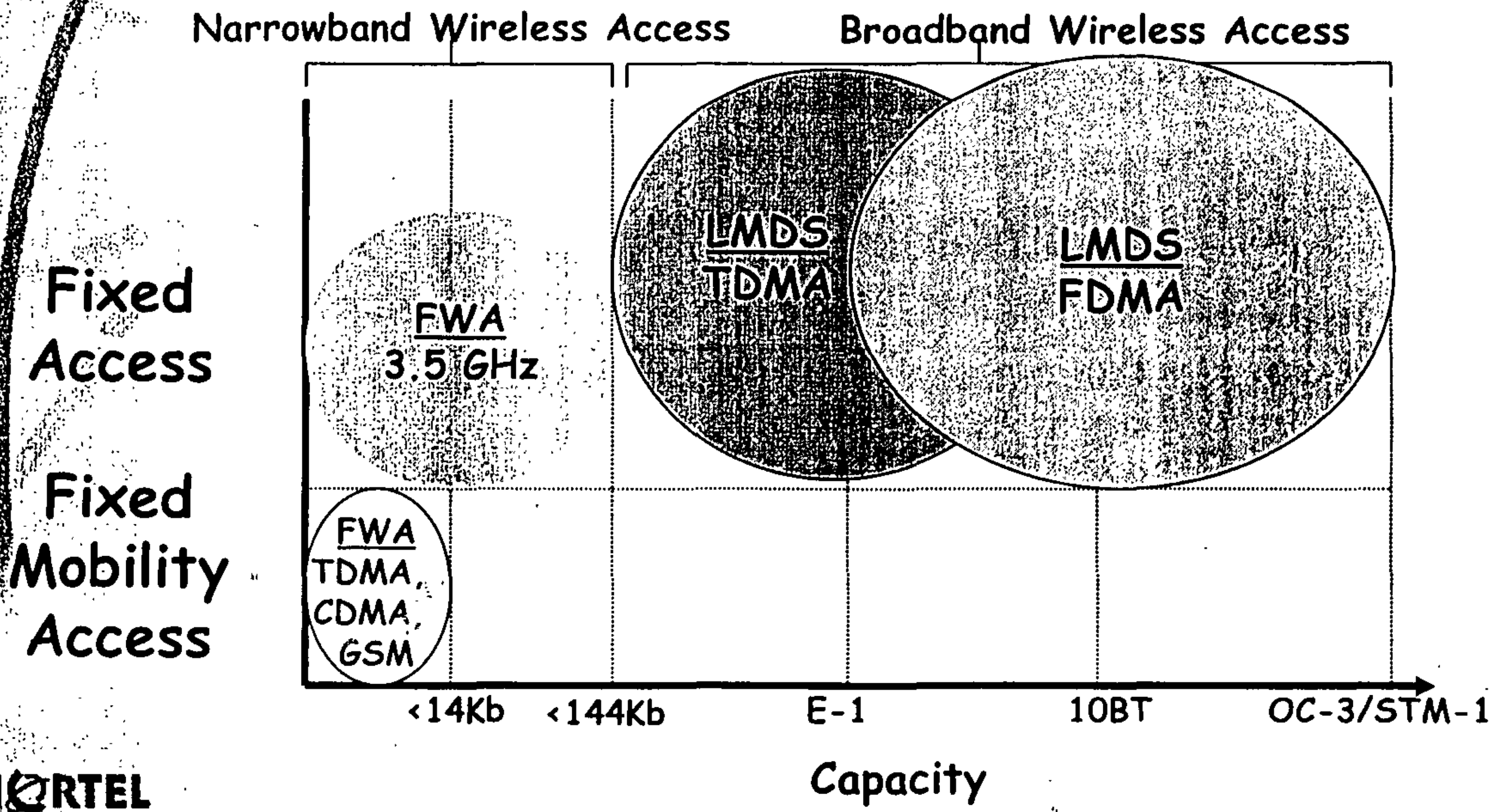
FWA: What is it?

- Uses spectrum 1.9 GHz to 3.5 GHz
- Point to multipoint radio access
- Demand driven deployment
- Virtual ISDN air interface supports voice, data and video services
- Bandwidth is shared among users in a cell

Why is it important?

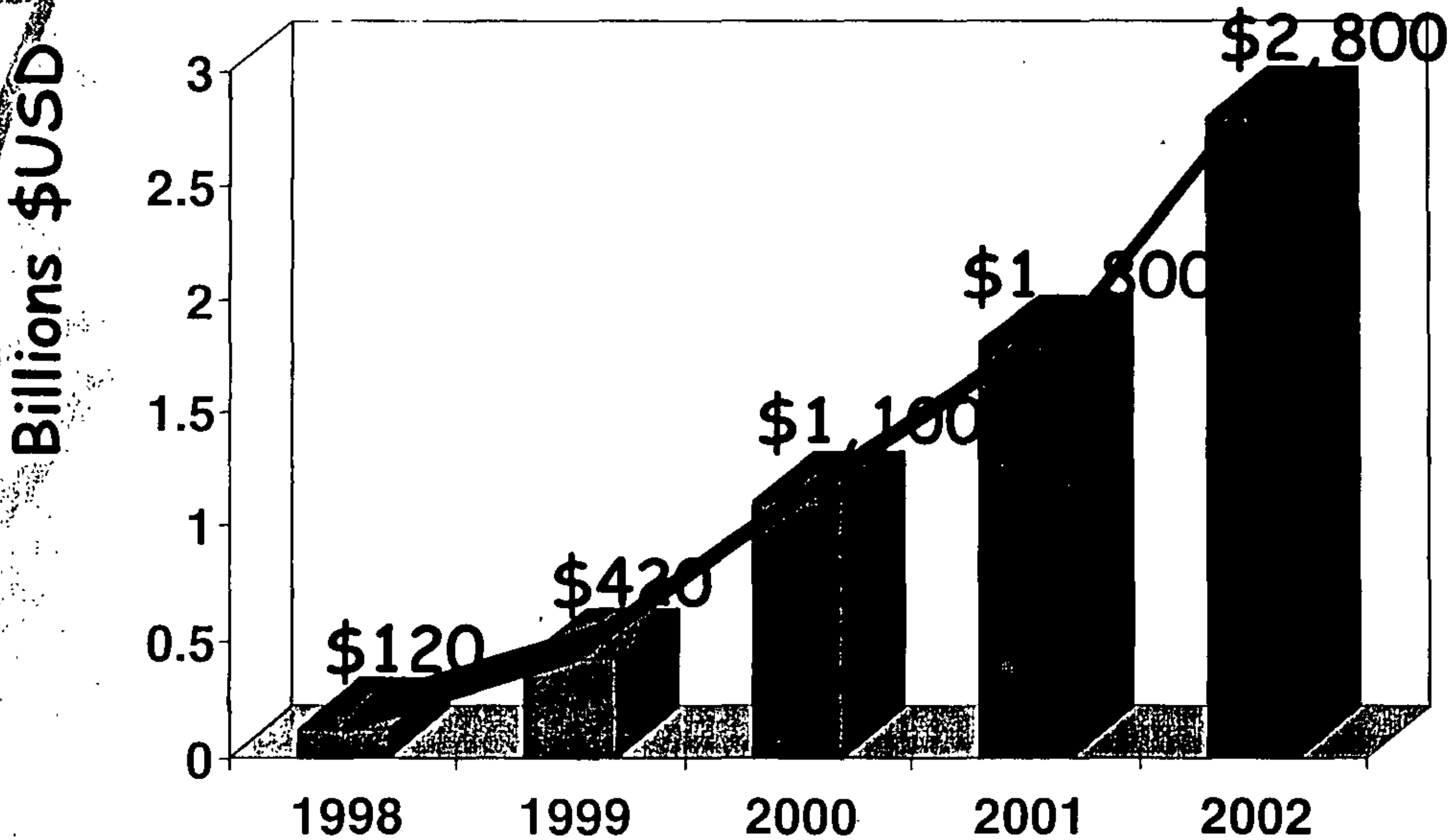
BWA/FWA: Why is it important?

Complements existing portfolio...



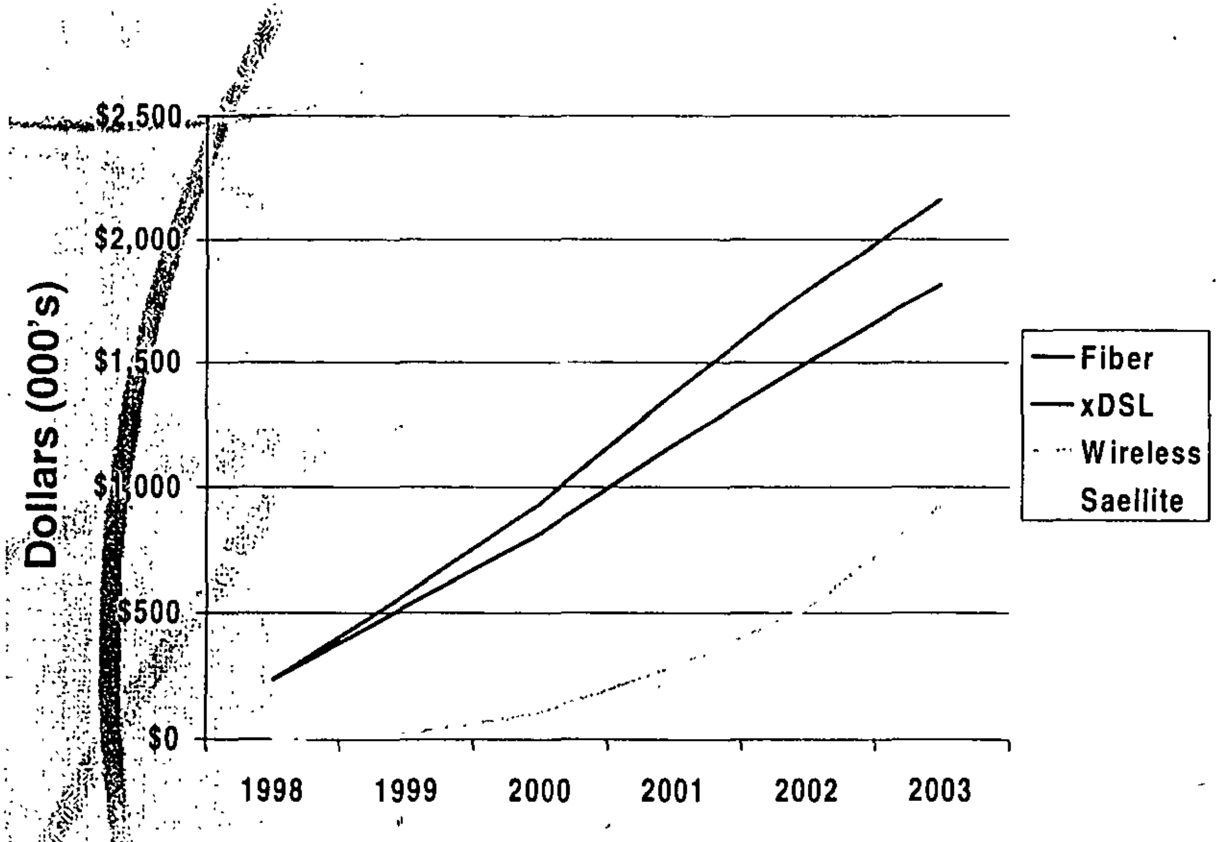
NORTEL NETWORKS

Global BWA Market Potential



NORTEL NETWORKS

Mexico BWA Market



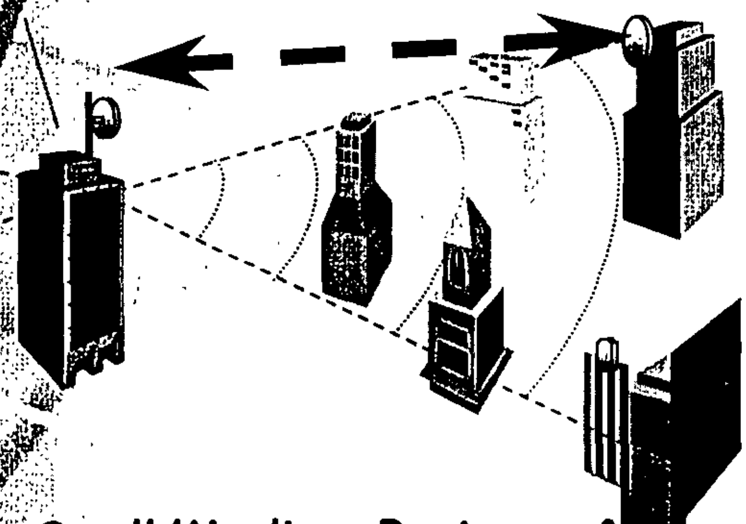
- BWA market of US$1000M in 2003
- Total market of US$5,000M in 2003
- Wireless BWA market growing at 189% annually
- Wire broadband access market growing at 58% annually

*The Strategis Group, June 1998

NORTEL NETWORKS

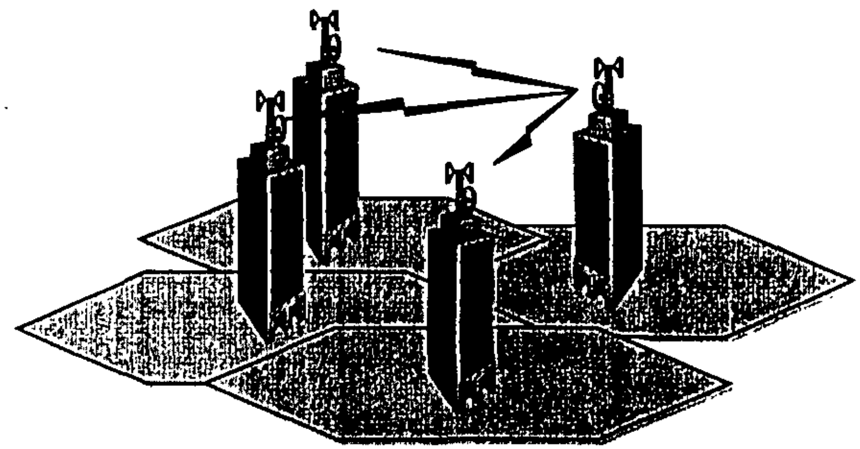
BWA: Where does it fit?

BWA Applications



Small/Medium Business &
Urban Residential Access

- Telephony & FAX
- High Speed Data
- Internet & Intranet Access
- Leased T1/E1, T3/E3
- Video conferencing
- Wireless LAN



FWA/Cellular/PCS microcell infrastructure

- Replace conventional point to point with point to multipoint infrastructure for cost and network economies

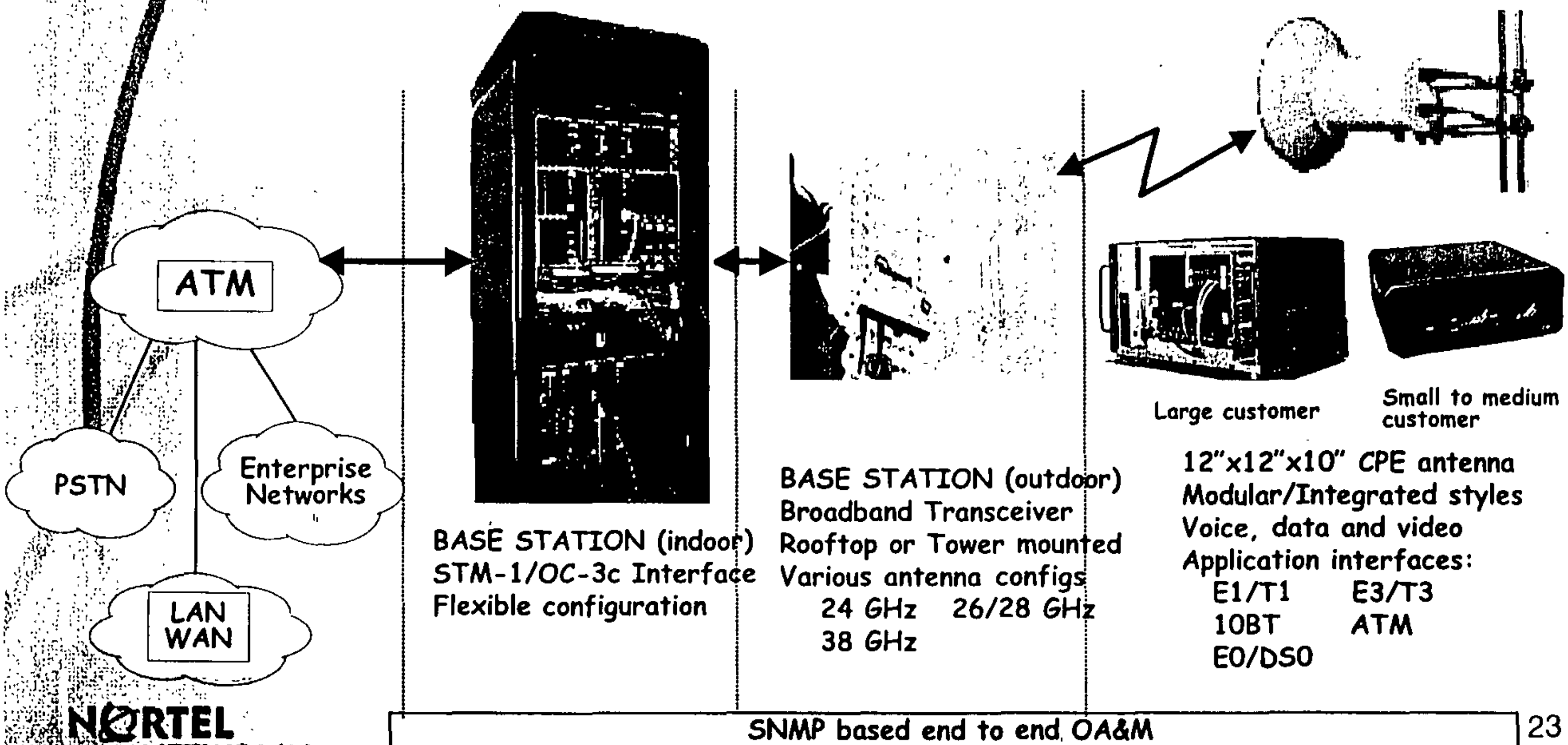
NORTEL NETWORKS

How does it work?

LMDS: How does it work?

Reunion Product

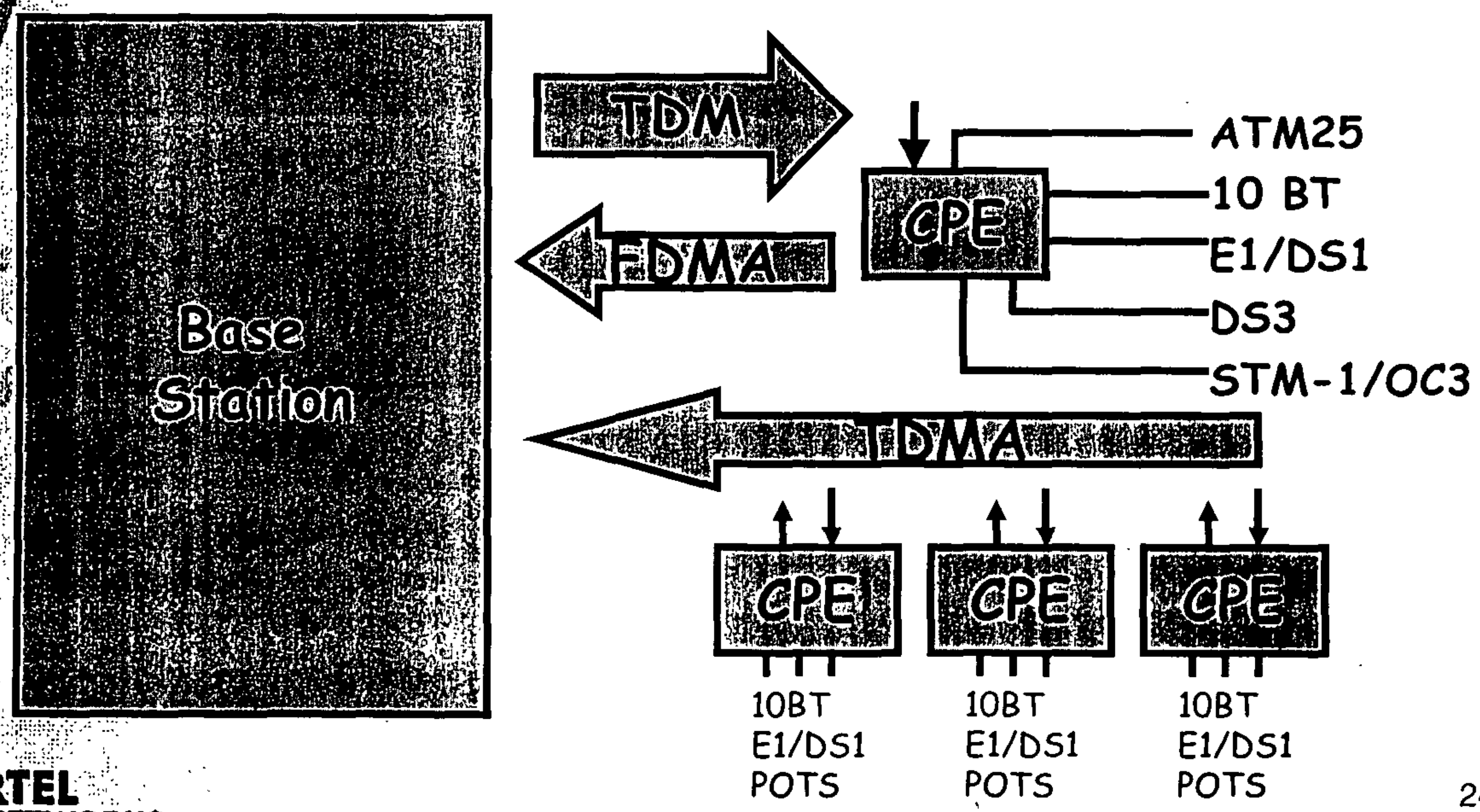
Point to multipoint technology allows high speed, high capacity two way multimedia services to be delivered directly to the local access market by wirelessly bypassing the local exchange carrier



NORTEL NETWORKS

LMDS: How does it work?

Access Methodologies



NORTEL NETWORKS

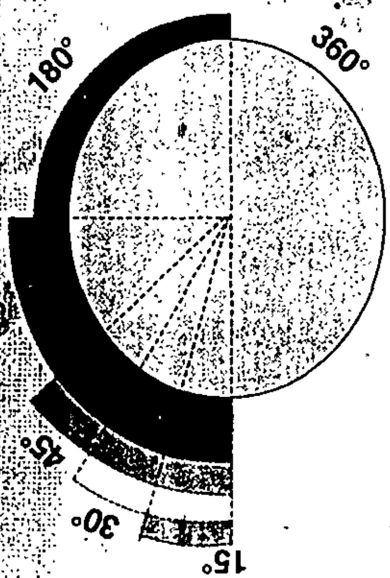
LMDS: How does it work?

- **FDMA is best suited for:**
  - for large bandwidth requirements per CPE
  - performance for high capacity traffic per CPE (aggregated)
- TDMA is best suited for:
  - for low minimum bandwidth requirements per CPE
  - burst like data patterns
- The cross over between TDMA and FDMA is 1.5 Mb/s to 2.5 Mb/s depending on traffic characteristics

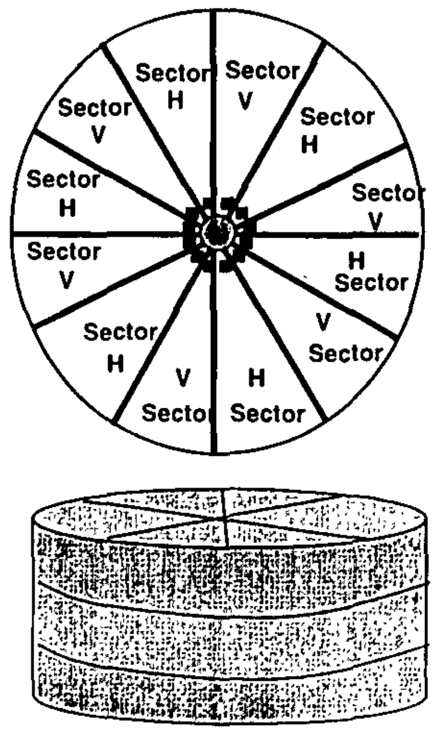
NORTEL NETWORKS

Link Budget / System Design

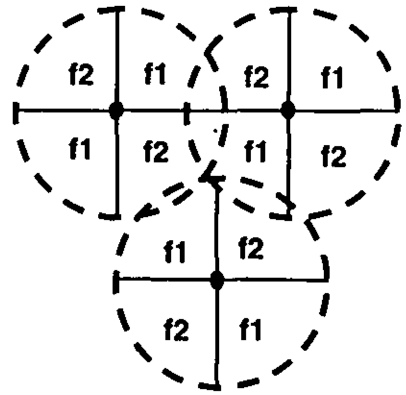
Effects on Coverage
<u>with Antenna / Sector Patterns</u>



Cell with 12 x sectors (30 °)
<u>Increase Capacity & Coverage</u>



3 Cell Pattern
<u>Extend Coverage</u>



NORTEL NETWORKS

LMDS: *System Design Considerations*

- Frequency band and bandwidth available
- Rain region
- Overall reliability % objective
- Target areas/location of subscribers/rollout plans
- Type of services
  - Data
  - Internet
  - Video Conferencing
  - Voice
- Number of Base Stations & Sectors
- Modulation/antenna types
- Network interconnection
- Current infrastructure

Reunion
Current Spectrum Capacity

Band [GHz]	Spectrum [MHz]	# E1	Radius [Km]	90^0 area [Km^2]	Total #E1 per Area
10.5	60	11.7	16	201	11.7
25-29	500	97.5	3	7.1	2,778.0
37-40	200	39.1	2.2	3.8	2,066.1

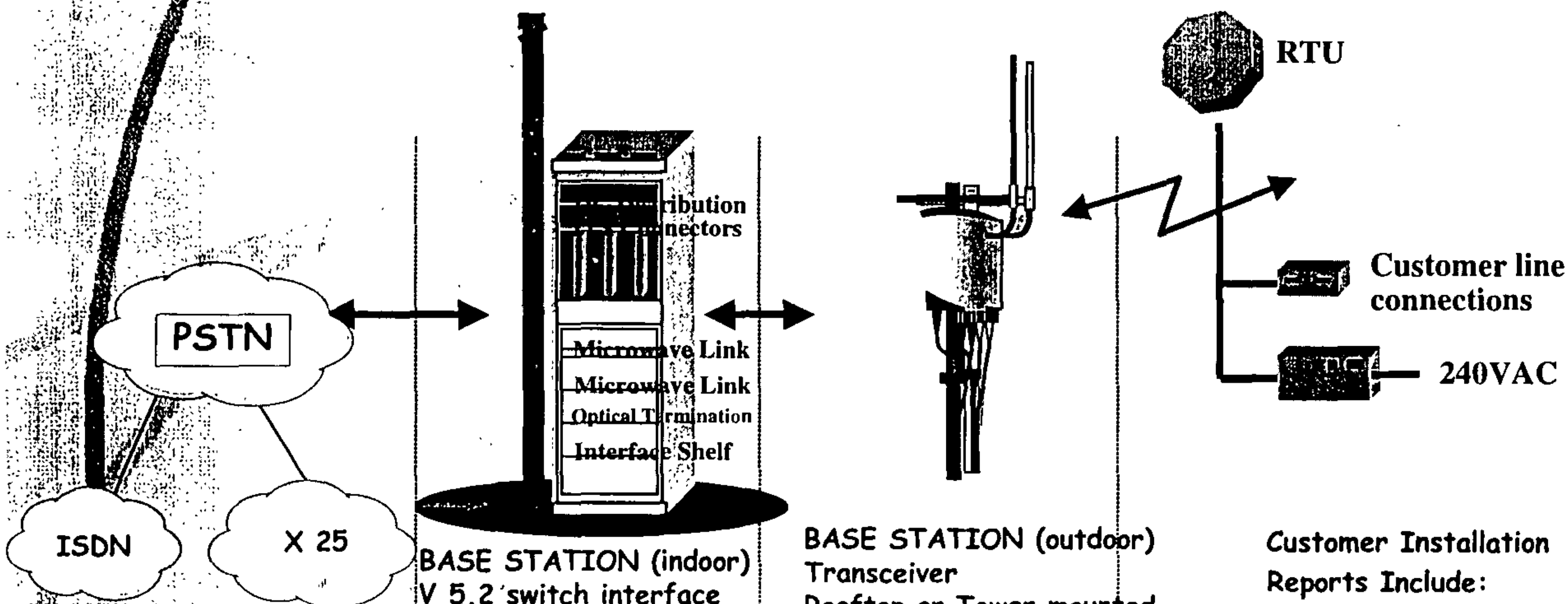
Assumptions

- All traffic is symmetrical
- Reuse factor of 1.5
- 20% overhead required for ATM and FEC
- Rain region "K"
- 99.99% availability
- QPSK modulation [1.5 bits/Hz]
- 90 degree antenna
- 3 carriers
- 10 MHz per carrier
- Terrain Type "A"
- CNR = 12 dB
- Vertical polarity

FWA: How does it work?

Proximity *II* Product

Point to multipoint technology allows high quality two way telephony services to be delivered directly to the local access market by wirelessly bypassing the local exchange carrier



BASE STATION (indoor)
V 5.2 switch interface
Scaleable configuration
External cabinet option
No base station controller
redundant architecture

BASE STATION (outdoor)
Transceiver
Rooftop or Tower mounted
Various antenna configs
3.5 GHz, 50-100 MHz duplex

Customer Installation

Reports Include:

- Bit Error Rate
- Rx signal level
- back-up battery condition
- 240VAC failure

TMN based end to end OA&M

NORTEL NETWORKS

FWA: How does it work?

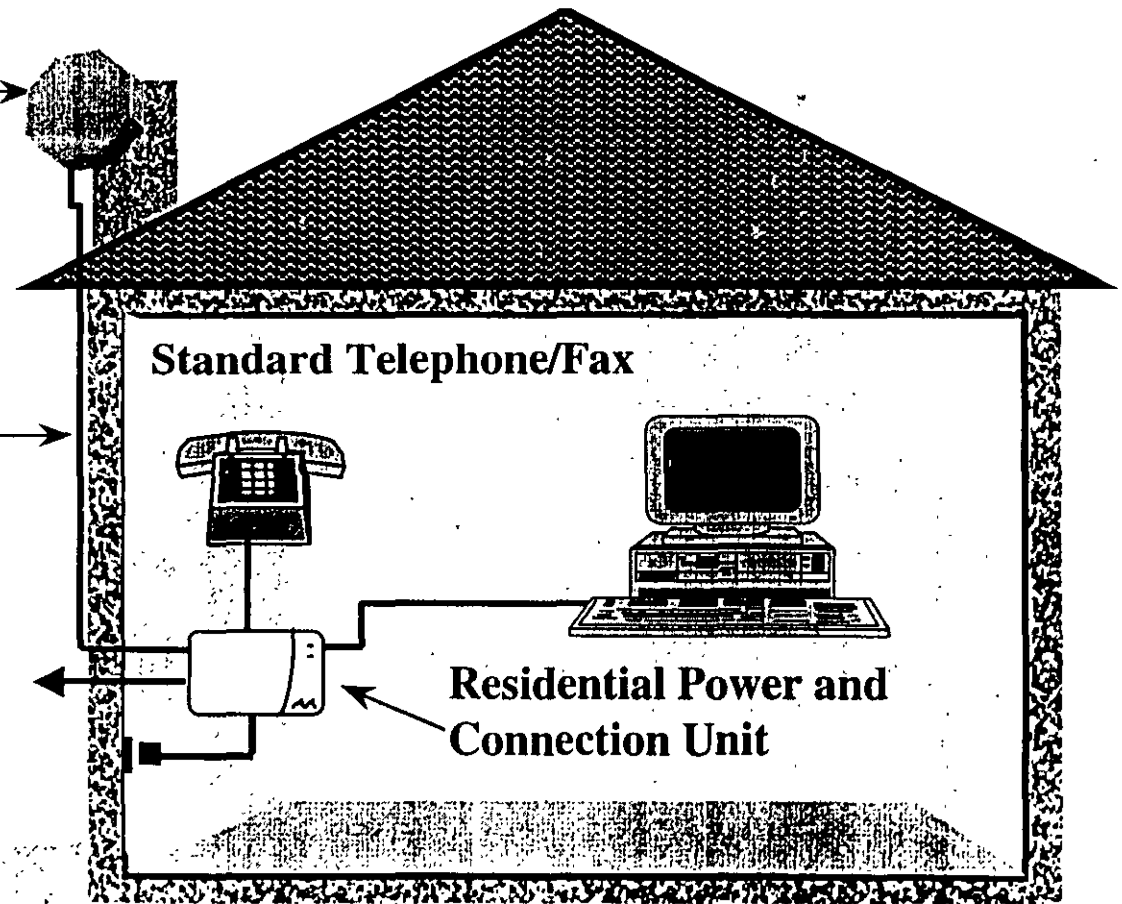
USING RPCU

Residential Transceiver Unit

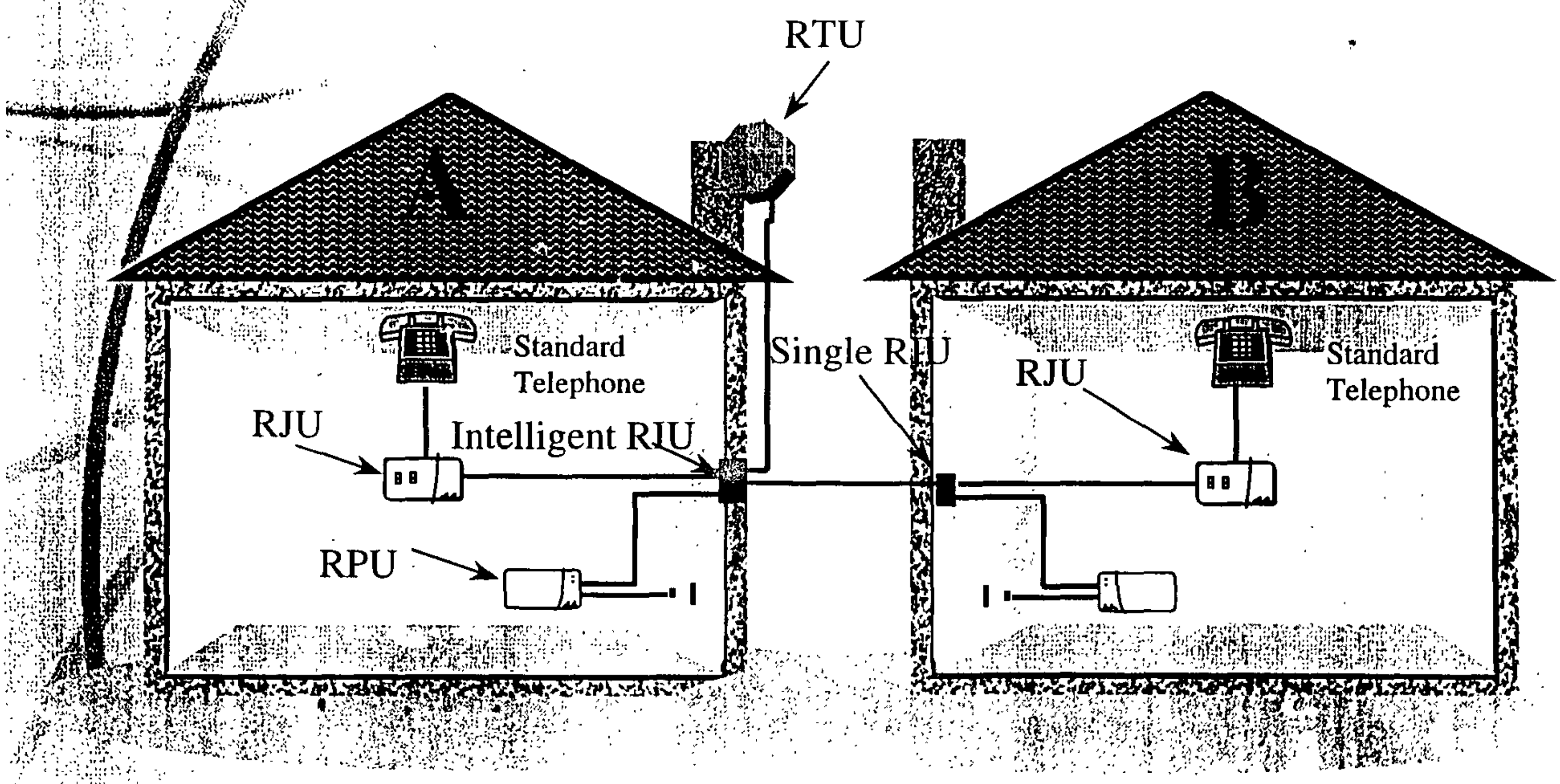
Residential Drop Cable

Optional external Power system
or Solar Power

Supports 2 Lines
Each with a REN of 4



FWA: How does it work?

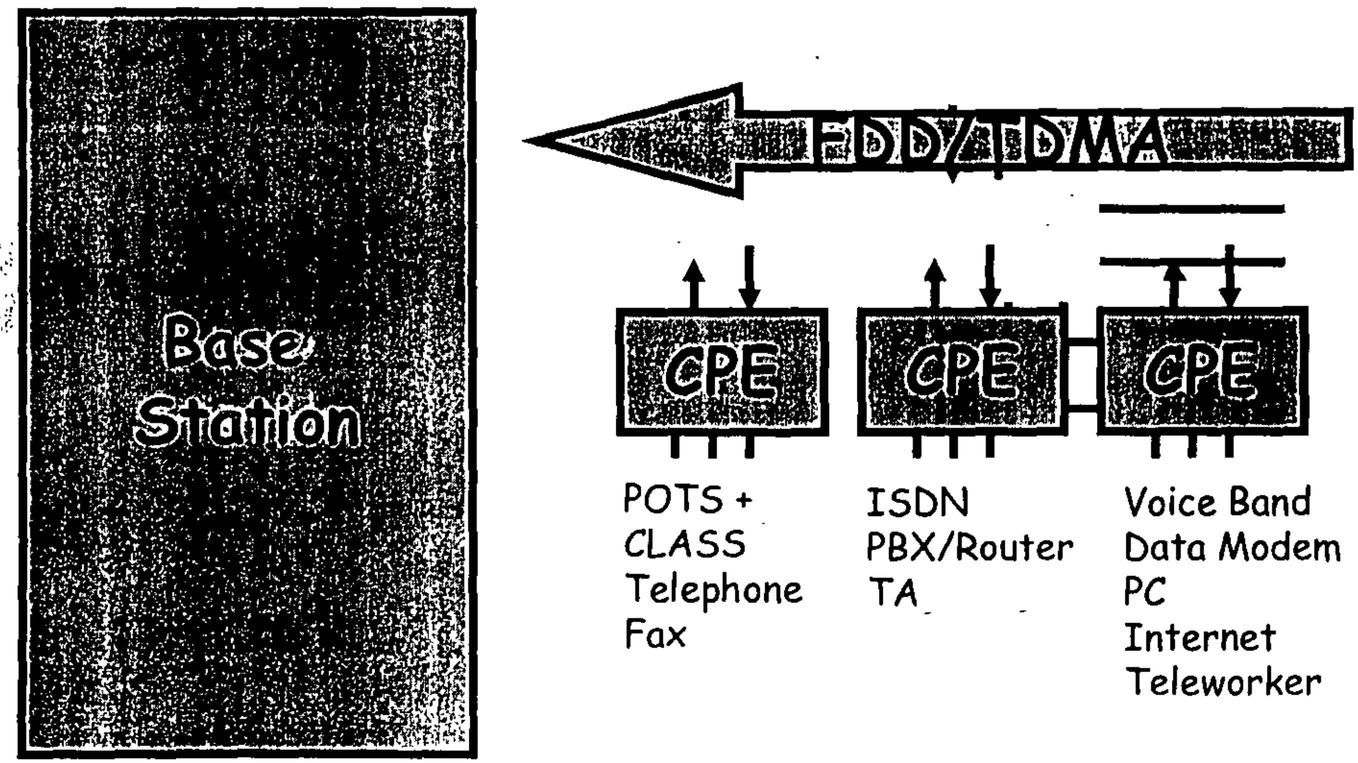


INTELLIGENT RIU SHARES POWER EQUALLY. ALSO SENSES AND COMPENSATES FOR EITHER HOUSE TURNING OFF POWER

NORTEL NETWORKS

FWA: How does it work?

Access Methodology



FWA: System Design Considerations

- Frequency band and bandwidth available
- Target areas/location of subscribers/rollout plans
- Type of services
  - Data
  - Internet
  - Voice
- Number of Base Stations & Sectors
- Antenna types-omni v sectored
- Network interconnection-switch interface
- Current infrastructure

FWA: Who will buy it?

- Rural operators for up to 40km range
- Alternate operators for niche market applications
- Internet Service Providers to bypass the local loop
- Incumbent operators for second line deployment

NORTEL NETWORKS

Technology comparison

Technology	Frequency	Band-Spectrum	Range	Data Rates
BWA LMDS	24-38 GHz	60~1000 MHz	<6 km	Up to 155 mbps (per sector)
WLL DECT	1.9 GHz	20 MHz	<3 km	28.8 kbps
WLL CDMA	1.9 GHz	40 MHz	<60 km	14.4 kbps
WLL TDMA*	3.5GHz	34 MHz	<40 km	128 kbps

* Nortel Networks Proximity II

NORTEL NETWORKS

BWA/FWA: Who will buy it?

Regulatory Environment

GHz Frequency Band

	1.9	5	10.5	15	23	28	38	40
Argentina	L		P			L	L	L
Brazil	L		P	L	L	P	L	
Chile						P		
Colombia			P			P	L	
Mexico	L		L	L	L	P		
Venezuela	L				L	L		

L = Licensed for potential broadband service
P = Planned to be licensed

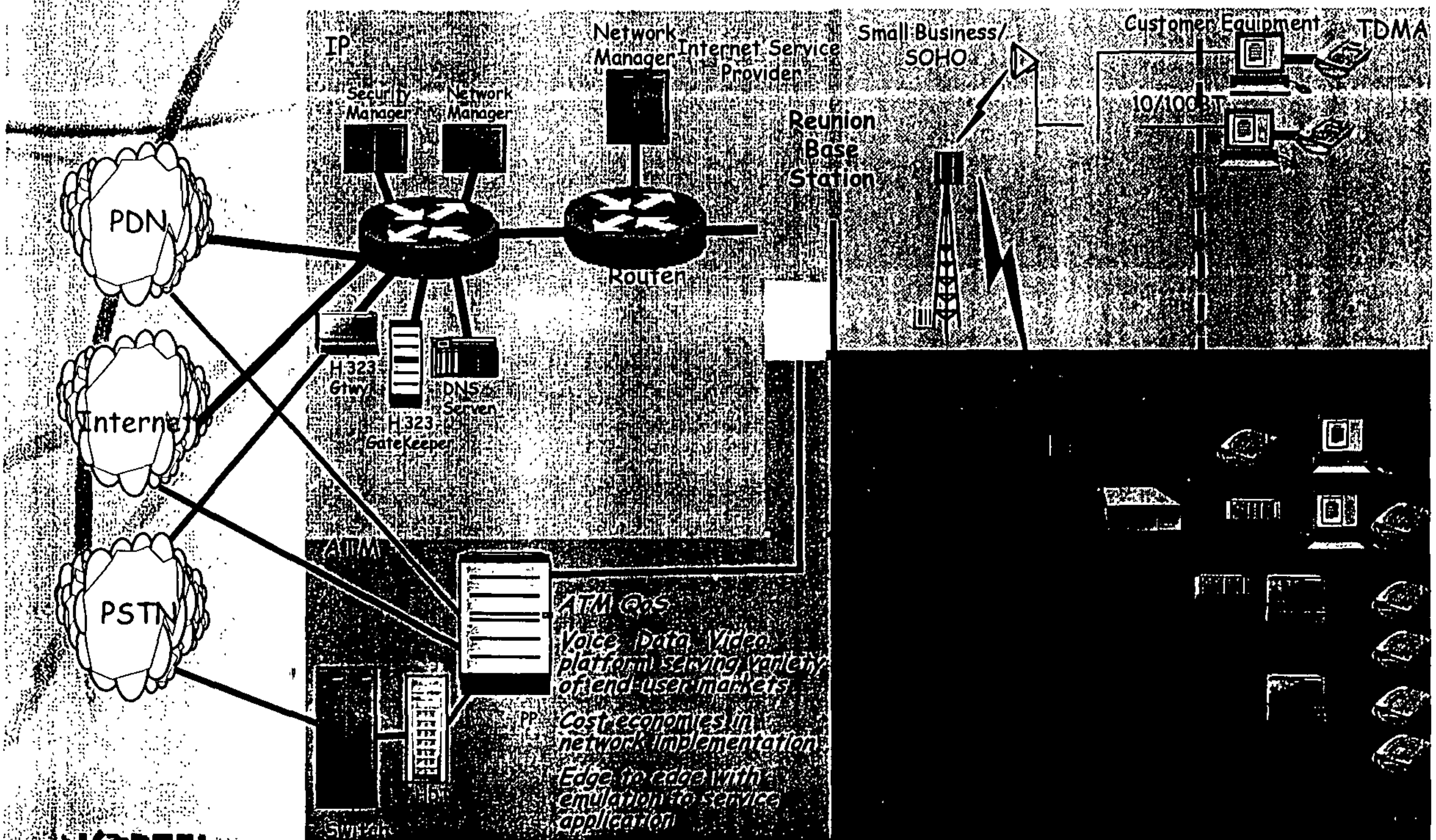
Source: The Strategis Group, 1998

NORTEL NETWORKS

NORTEL NETWORKS

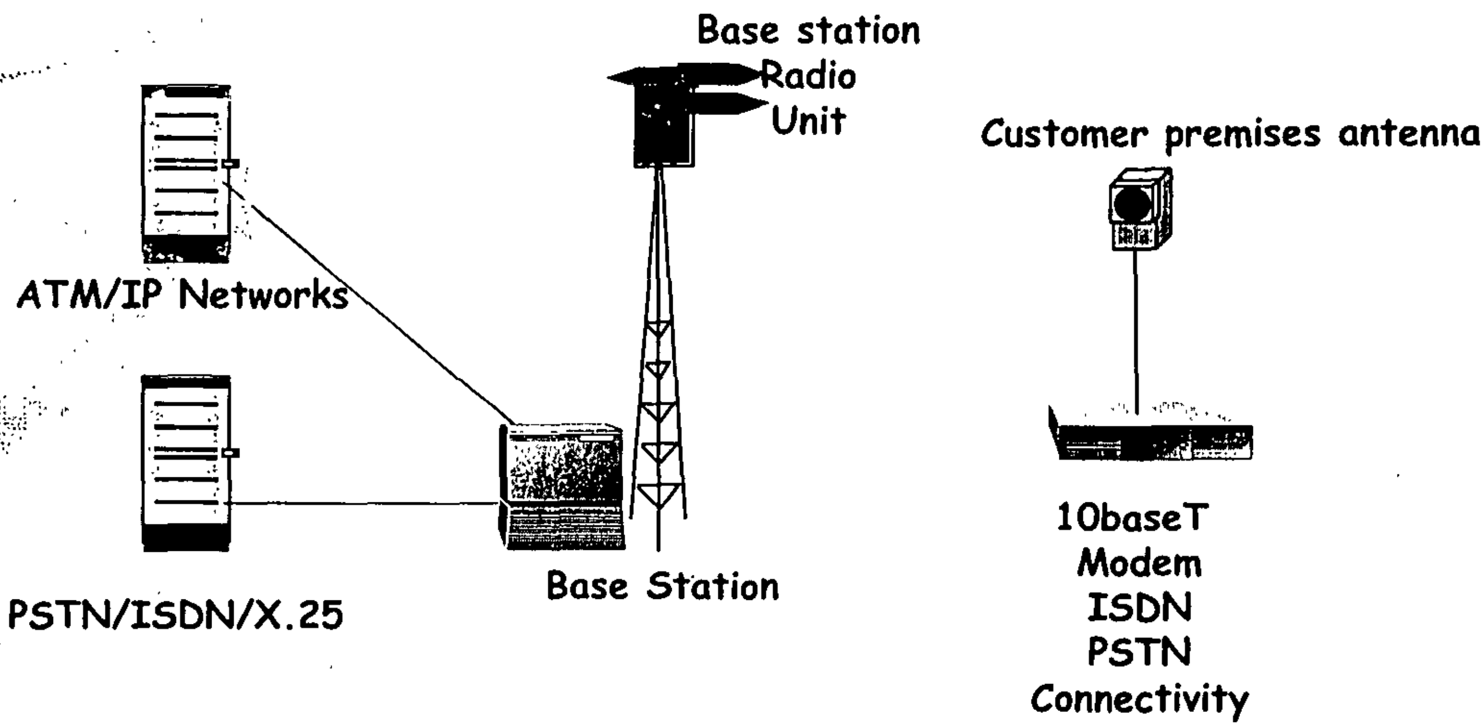
Future Evolution

LMDS: Quad 4 Architecture



NORTEL NETWORKS

FWA Evolution



Any network, any service, any operator

Summary

LMDS: Competitive Positioning

- Total Network Solution!
- State of the art RF technology
- Flexible to support any service
  - High Speed Data (variable bit rate including E1 circuit emulation), webtone/IP
  - Voice (POTS, ISDN and voiceband data)
  - Broadcast Video
- Use of radio propagation planning tools for high frequency systems
- Support network connectivity for ATM, PSTN, IP, and private data networks
- Fast deployment, cost effective alternative to high-speed wired access

FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

<u>SISTEMAS DE TELEFONÍA MÓVIL</u>

EXPOSITOR: ING. JORGE GONZÁLEZ Y GONZÁLEZ
PALACIO DE MINERÍA
JUNIO DE 1999

Palacio de Minería Calle de Tacuba 5 Primer piso Deleg. Cuauhtemoc 06000 Mexico, D.F. tel.: 521-40-20 Apdo. Postal M-2285

HISTORIA

1899	PRIMERA TRANSMISION DE RADIO GUILLERMO MARCONI.
1940'S	LAS POLICIAS DE LAS GRANDES CIUDADES EN LOS ESTADOS UNIDOS UTILIZAN SISTEMAS RADIOTELEFONICOS PRIVADOS.
1970'S	EN LA CIUDAD DE NUEVA YORK CON SOLO 12 CANALES DE RADIO SE DA SERVICIO A 545 USUARIOS.
1980'S	IMTS SISTEMAS DE 100 A 200 CANALES PARA SOPORTAR 3000 A 5000 USUARIOS.
1984	CELULAR NACE EL PRIMER SISTEMA CELULAR SE LANZA COMERCIALMENTE EN CHICAGO. • RE-USO DE FRECUENCIA • BAJAS POTENCIAS

RE-USO DE FRECUENCIAS

- **PARA MINIMIZAR INTERFERENCIA**

  AISLAMIENTO DE UN SITIO CELULAR CON OTRO QUE UTILICE EL MISMO GRUPO DE FRECUENCIAS.

- **DISTANCIA ENTRE SITIOS CELULARES**

D/R

D = DISTANCIA ENTRE CELULARES USANDO LA MISMA FRECUENCIA.

R = RADIO DE LA CELULA.

UN VALOR TIPICO DE D/R PARA LOS SISTEMAS CELULARES ANALOGICOS ES DE 4.6

VGR D = 4.6

R = 1

ESTO SIGNIFICARIA QUE UN CANAL USADO EN UNA CELULA CON UN RADIO DE 1 Km. SE PODRIA REUTILIZAR A 4.6 Kms. DE DISTANCIA.

LOS SISTEMAS DIGITALES SON MUCHO MAS EXIGENTES.

EXISTEN DOS TIPOS DE SISTEMAS CELULARES:

- ANALOGICO ➡ MODULACION FM
- DIGITAL ➡ MODULACION "/4 PSK

1990' TENDENCIAS MUNDIALES A DIGITALIZAR LOS SISTEMAS.

"DIGITALIZACION DE LA INTERFAZ DE AIRE"

OPERACION EN MODO DUAL

"DUAL MODE"

SISTEMAS DIGITALES :

MAYOR CAPACIDAD

PRIVACIA

CALIDAD DE AUDIO (VOCODER)

S N S

TECNOLOGIAS DIGITALES : TDMA = DAMPS = IS-136

METODOS DE ACCESO : CDMA = IS-95

GSM

CANALES DE CONTROL

CANALES DE VOZ o DE TRAFICO.

VOCODERS

DIGITALIZACION DE LA VOZ ➡ 64 KBPS

PARA TRANSMITIR VIA RADIO 64KBPS SE REQUERIRIA UN CANAL DE 64 KHz.

EL CANAL DISPONIBLE DE LAS DIFERENTES TECNOLOGIAS CELULARES ES DE 25 ó 30 KHz.

∴ SE REQUIERE COMPRESION - CODIFICACION

PARA	**TDMA**	**IS-136**	**COMPRESION 8 : 1**
PARA	**CDMA**	**IS-95**	**COMPRESION VARIABLE 8 : 1 A 64 : 1**
PARA	**CSM**		**COMPRESION 5 : 1**

EL PROCESO DE COMPRESION - DESCOMPRESION SE LLEVA ALREDEDOR DE 50 A 100 MSEG. PROVOCANDO ECO.

∴ USO DE CANCELADORES DE ECO EN MSC.

ESTANDARES PARA RADIOTELEFONIA CELULAR:

AMPS - ADVANCE MOBILE PHONE SYSTEM (EIA-553)

FCC EN 1974 ASIGNA 40 MHz PARA ESTE SERVICIO

FCC EN 1984 ASIGNA 10 MHz ADICIONALES

UPLINK	**824 A 849 MHz.**
DOWNLINK	**869 A 894 MHz.**

ESTO DA UNA CAPACIDAD DE 832 CANALES DUPLEX DE 30KHz.

BANDA	**A**	**25 MHz.**	**416 CANALES**
BANDA	**B**	**25 MHz.**	**416 CANALES**

TELEFONOS CLASE 1 :	**3 WATT = 5 dBw**	**=**	**34 dBm**
CLASE 2 :	**1.6 WATT = 2 dBw**		
CLASE 3 :	**0.6 WATT = -2 dBw**	**=**	**28 dBm**

LA POTENCIA DE TRANSMISION DEL MOVIL SE AJUSTA EN PASOS DE 4 dB.

LA POTENCIA MINIMA ES DE 6 Mw = -22 dBw = 8 dBm

<u>TACS</u> - TOTAL ACCESS COMMUNICATION SYSTEM

- INSTALADO EN 1985 EN EL REINO UNIDO.
- FRECUENCIAS COMUNMENTE ASIGNADAS A TACS

UPLINK	890 MHz A 915 MHz.
DOWNLINK	935 MHz A 960 MHz.

25 MHz	FULL DUPLEX
45 MHz	SEPARACION UP/DOWN LINKS

BW = 25 KHz POR CANAL.

TELEFONOS CLASE 1 :	10 WATTS
CLASE 2 :	3 WATTS
CLASE 3 :	1.2 WATTS
CLASE 4 :	0.6 WATTS

LA POTENCIA DE TRANSMISION DEL MOVIL TAMBIEN SE AJUSTA EN PASOS DE 4 dB.

LA POTENCIA MINIMA TAMBIEN ES DE 6 mW.

J TACS	-	TACS MODIFICADO PARA JAPON.
N TACS	-	DE BANDA ANGOSTA BW = 12,5 KHz.

NMT - NORDIC MOBILE TELEPHONE

- NMT 450 INTRODUCIDO EN 1981
- NMT 900 INTRODUCIDO EN 1986

- DESARROLLADO POR LOS PAISES NORDICOS:SUECIA
  DINAMARCA
  NORUEGA
  FINLANDIA

NMT - 450 180 CANALES HASTA 50w
BW = 25 KHz.

ESPARCIMIENTO DUPLEX = 10MHz

NMT - 900 1000 CANALES HASTA 25W
BW = 25 KHz

TELEFONOS	NMT - 450	15 W 1 . 5 W 0 . 15 W	MOVILES
		1 . 0 W 0 . 1 W	PORTATILES
	NMT-900	6 W 1 W 0 . 1 W	MOVILES
		1 . 0 W 0 . 1 W	PORTATILES

NAMPS - NARROWBAND AMPS

- **INTRODUCIDO EN 1991 POR MOTOROLA**

  BW = 10 KHz

- **ESTANDAR IS-88 UNA EVOLUCION DE EAI-553**

JMCS - JAPANESE MOBILE CELLULAR SYSTEM (MCS)

- **INTRODUCIDO EN JAPON EN 1979**

  OPERA EN LA BANDA DE 800 MHz.

  BW = 25 KHz

CNET

- **SISTEMA USADO EN ALEMANIA, PORTUGAL Y SUDAFRICA.**
- **INTRODUCIDO "TEMPORALMENTE" EN ALEMANIA EN 1985.**
- **OPERA EN LA BANDA DE 450 MHz.**

  461.3 A 465.74 MHz.

  451.3 A 455.74 MHz.

  BW = 20 KHz.

  SEPARACION UP/DOWN : 10MHz.

GSM - GLOBAL SYSTEM FOR MOBILE COMMUNICATION

- CREADO PARA PROPORCIONAR UN ESTANDAR UNICO EN EUROPA.
- UTILIZA TECNOLOGIA TDMA.
- PRIMER SISTEMA EN OPERACION EN 1991.
- SISTEMA QUE HA EVOLUCIONADO PARA USO EN BANDAS :

  900 MHz

  1800 MHz PCN EN EUROPA

  1900 MHz PCS EN AMERICA

- ACTUALMENTE EXISTEN ALREDEDOR DE 150 PAISES USANDO GSM.
- SISTEMA SOLO DIGITAL NO COMPATIBLE CON TECNOLOGIAS ANALOGICAS.
- BW = 200 KHz TDMA - 8 FULL RATE
- TDMA - 16 HALF RATE (VOCODER EN DESARROLLO).
- GSM BIT RATE = 270 KBPS.
- GSM- 900 45 MHz SEPARACION UP/DOWN
- GSM - 1900 80 MHz SEPARACION UP/DOWN

<u>CDMA</u> - IS 95   CODE DIVBISION MULTIPLE ACCESS

- **DESARROLLADO POR QUALCOMM.**
- **1995 PRIMER SISTEMA EN OPERACION EN HONG KONG.**
- **BW = 1.23 MHz.**
- **MULTIPLES CONVERSACIONES SIMULTANEAS EN LA MISMA FRECUENCIA DIFERENCIADAS POR CODIGOS.**
- **COMPATIBLE CON AMPS ANALOGICO (EIA - 553)**

<u>IMT - 2000</u> EL FUTURO

- **INFOCOM   VOZ**
  DATOS ALTA VELOCIDAD
  VIDEO
  MULTIMEDIA

- **INFOCOM   ES LA CONVERGENCIA DE :**
  -TELECOMUNICACIONES INALAMBRICAS.
  - INTERNET.

SISTEMA DE SEÑALIZACION POR CANAL COMUN No. 7

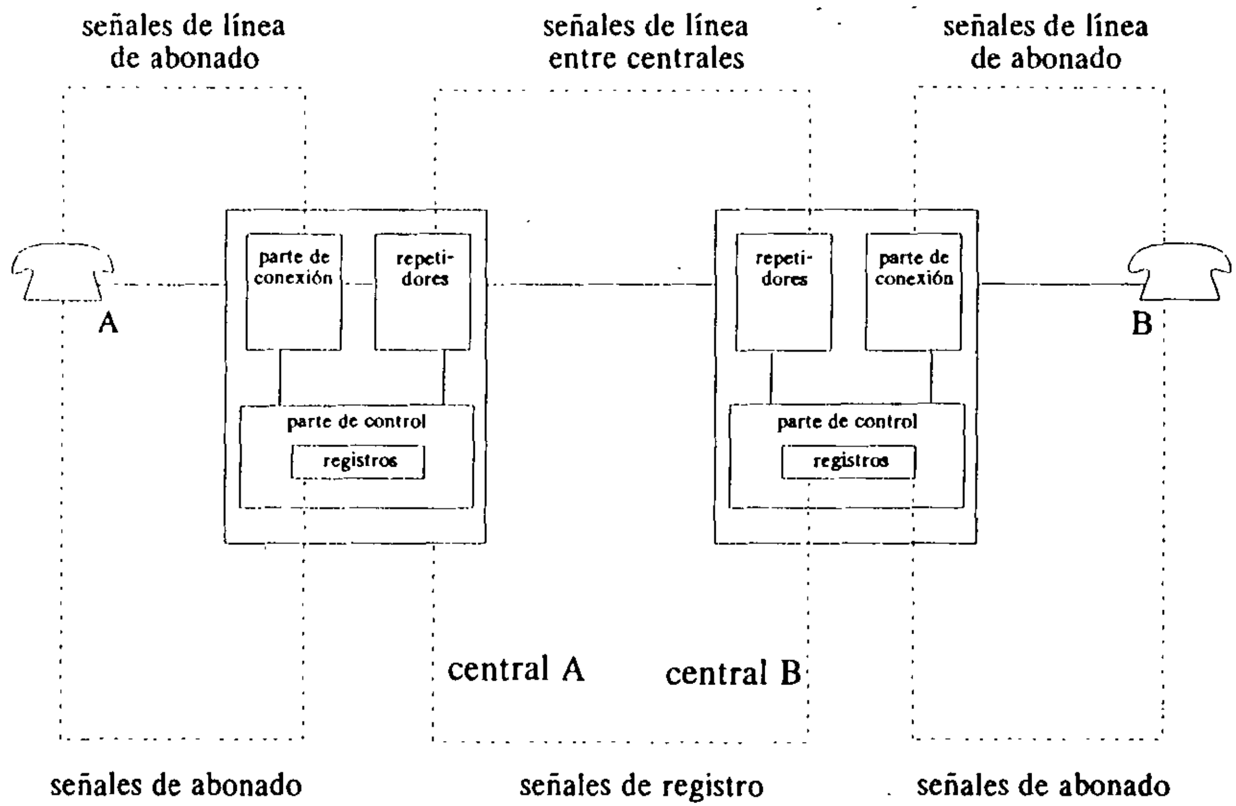
M.C. MARTIN LARA BARRON

Señalización.- Es el intercambio de información entre equipos que forman la planta telefónica, a través de señales que permiten establecer y controlar las comunicaciones telefónicas.

Funciones Básicas:

* Supervisión
* Selección
* Operación

Señalización Actual en la Red Nacional



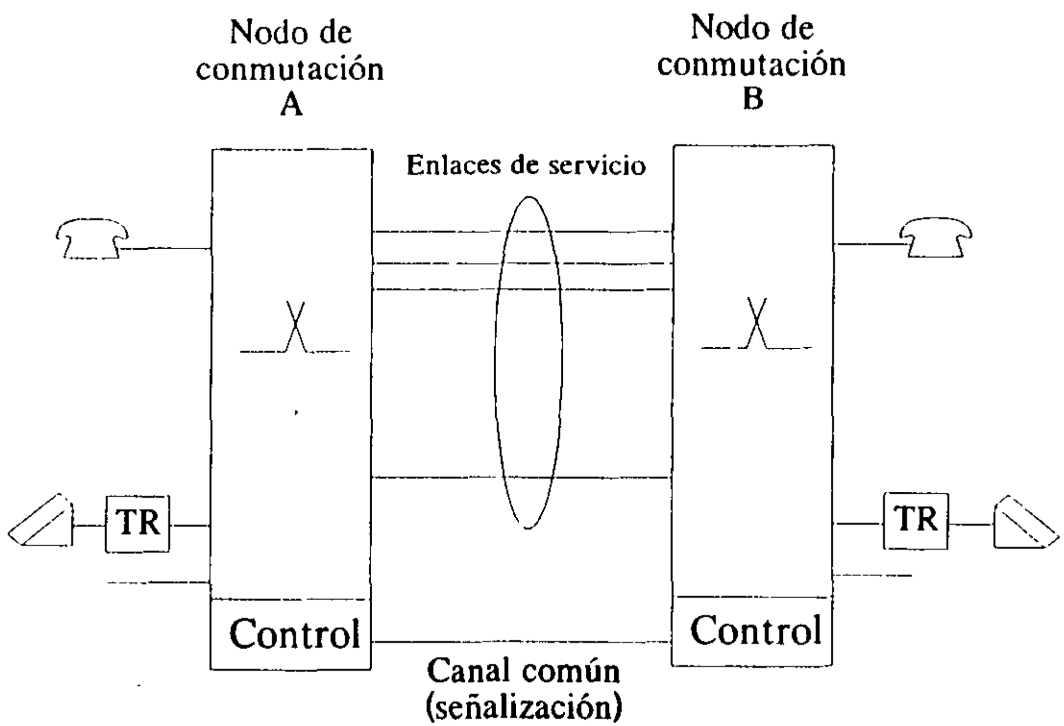
señales de línea → sistema No. 3
señales de registro → sistema R2

Características:

- utiliza la red de voz para señalizar el enlace en cuestión
- numero limitado de señales
- aplicación únicamente para telefonía
- tiempo de transferencia de señalización del orden de segundos
- no puede emplearse en circuitos vía satélite
- manejo de señales de línea y de registro

Señalización por Canal Común

Un sólo canal, común para un número de enlaces de voz, transfiere la información de señalización en paquetes que se identifican mediante etiquetas.



Con la evolución de la tecnología electrónica y la introducción de centrales de control por programa almacenado digitales, se presenta la necesidad de optimizar la función de señalización en la red telefónica digital.

Es por esto que se ha desarrollado el sistema de

señalización por canal común CCITT No. 7

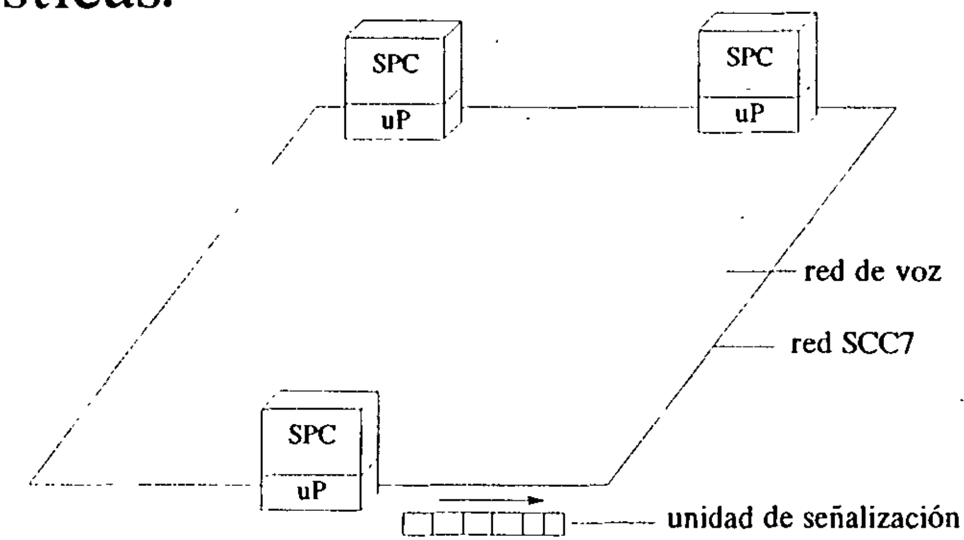
Sistemas de Señalización Internacionales Normalizados por el CCITT

sistema	año normalizado	aplicación	tipo de senalización
1	1934	manual internacional	trayectoria de voz
2	1938	automático dos hilos	trayectoria de voz
3	1954	automático y semiautomático intracontinental	trayectoria de voz
4	1954	automático y semiautomático intracontinental	trayectoria de voz
5	1964	automático y semiautomático intercontinental	trayectoria de voz
6	1968	automático y semiautomático intercontinental	canal común
6'	1976	automático intercontinental	canal común
7	1980	automático internacional	canal común
R1	1968	automático y semiautomático regional	trayectoria de voz
R2	1968	automático y semiautomático regional	trayectoria de voz

Señalización por Canal Común CCITT No. 7

- Desarrollado para operar en un sistema totalmente digital de 64 Kbps.
- Aplicación general normalizada internacionalmente tanto para redes nacionales como internacionales
- Adecuado para uso en enlaces punto a punto tanto terrestres como vía satélite
- Operación bajo el principio de conmutación de paquetes.

Características:



- Utiliza una red separada
- Capacidad ilimitada en el servicio de señales
- Puede manejar cualquier servicio de telecomunicaciones
- Tiempo de transferencia de señalización del orden de milisegundos
- Transparente al medio de transmisión
- Manejo de un solo tipo de señales

SCC7

- Su estructura funcional permite una gran flexibilidad y modularidad para diversas aplicaciones dentro de un concepto de sistema.

  * parte de transferencia de mensajes
  * parte de usuario
  * parte de control de la conexión de señalización
  * parte de aplicación de las capacidades de transacción

- Desarrollado en base a una arquitectura de niveles

  * Nivel 1: Funciones del enlace de datos de señalización
  * Nivel 2: Funciones de enlace de señalización
  * Nivel 3: Funciones de la red de señalización
  * Nivel 4:
    - Parte de usuario
    - Parte de control de la conexión de señalización
    - Parte de aplicación de las capacidades de transacción

Nivel 1. Funciones del enlace de datos de señalización: define las características físicas, eléctricas y funcionales del enlace de señalización y los medios para accesar al mismo.

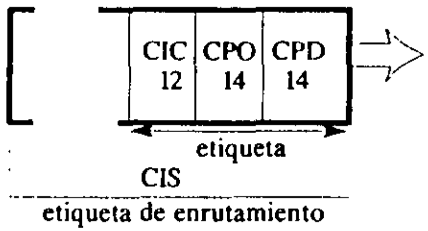
Nivel 2. Funciones del enlace de señalización: define las funciones y procedimientos para la transferencia de los mensajes de señalización generados por los niveles jerárquicos superiores, a través de un determinado enlace de señalización.

+ control de errores
+ supervisión del enlace
+ generación de tres tipos de mensajes de señalización

Nivel 3

Funciones de la red de señalización:
define las funciones y procedimientos
para la transferencia de los mensajes de
señalización entre puntos de señalización
y los aspectos relativos a tal transferecia.

* tratamiento de los mensajes de señalización
  + discriminación
  + distribución
  + enrutamiento



* gestión de la red de señalización
  + gestión del tráfico
  + gestión de la ruta
  + gestión del enlace

Nivel 4. Parte de usuario: define las funciones y procedimientos que son particulares a un determinado tipo de usuario.

* usuarios con funciones de control de comunicaciones telefónicas y datos

  + PUT telefonía
  + PUD datos
  + PUSI RDSI

* usuarios con funciones de transferencia de información para fines de gestión y mantenimiento

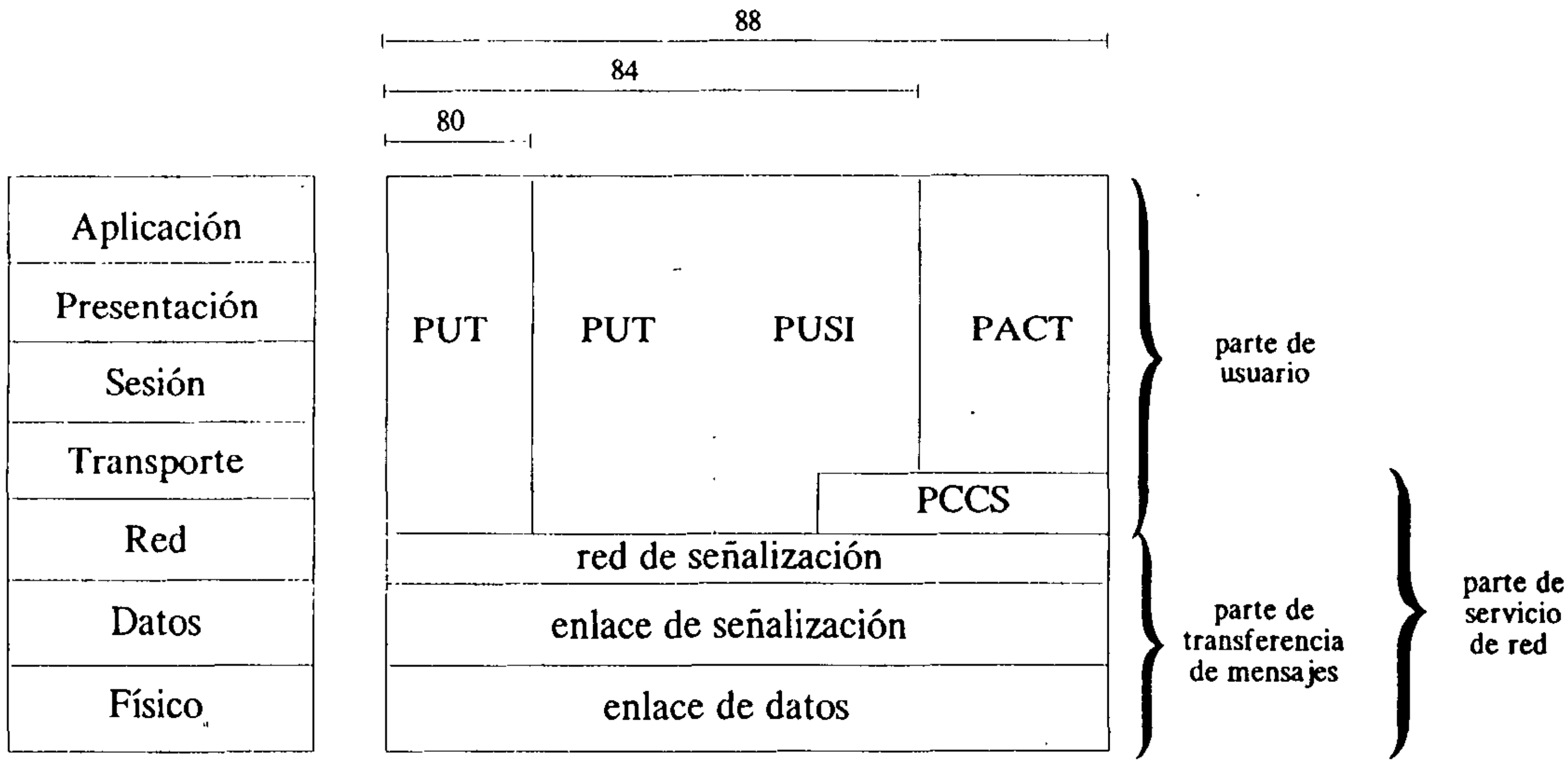
  + POM operación y mantenimiento
  + PUCR control remoto
  + PUFC facturación

Estructura Funcional del SCC7

División func. básica	Niveles	p. señalización origen	p. señalización destino
parte de usuario	- usuario de telefonía - usuario de datos - otro tipo	nivel 4	nivel 4
parte de transferencia de mensajes	- funciones de la red de señalización	nivel 3	nivel 3
	- funciones del enlace de señalización	nivel 2	nivel 2
	- funciones del enlace de datos de señalización	nivel 1	nivel 1

—— enlace físico   · · · enlace virtual

Evolución CCITT No. 7



Recomendaciones Q.7XX, Libro Azul

* Q.701 - Q.704, Q.706 - Q.707	parte de transferencia de mensajes
* Q.721 - Q.725	parte de usuario de telefonía
* Q.730	servicios suplementarios
* Q.741	parte de usuario de datos (≈X.61)
* Q.761 - Q.764, Q.766	parte de usuario de RDSI
* Q.711 - Q.714, Q.716	parte de control de la conexión de señalización
* Q.771 - Q.775	parte de aplicación de las capacidades de transacción

Existen otras diez recomendaciones que describen aspectos tales como estructura de red, numeración y pruebas, pero que no forman parte de las interfaces de señalización.

El uso de SCC7 traerá consigo:

* Aumento de la eficiencia de la red telefónica, ya que esta no se emplea para el establecimiento de las llamadas.

* Reducción potencial en la inversión de equipo al desarrollar una red más sencilla.

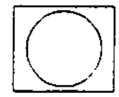
* Disminución de gastos para la gestión de la red.

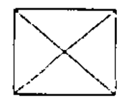
* Creación de la infraestructura necesaria para evolucionar hacia una red digital de servicios integrados [RDSI].

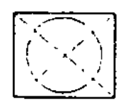
RED SCC7 - Es necesario establecer la arquitectura de la red para especificar las funciones a desempeñar por esta y sus componentes

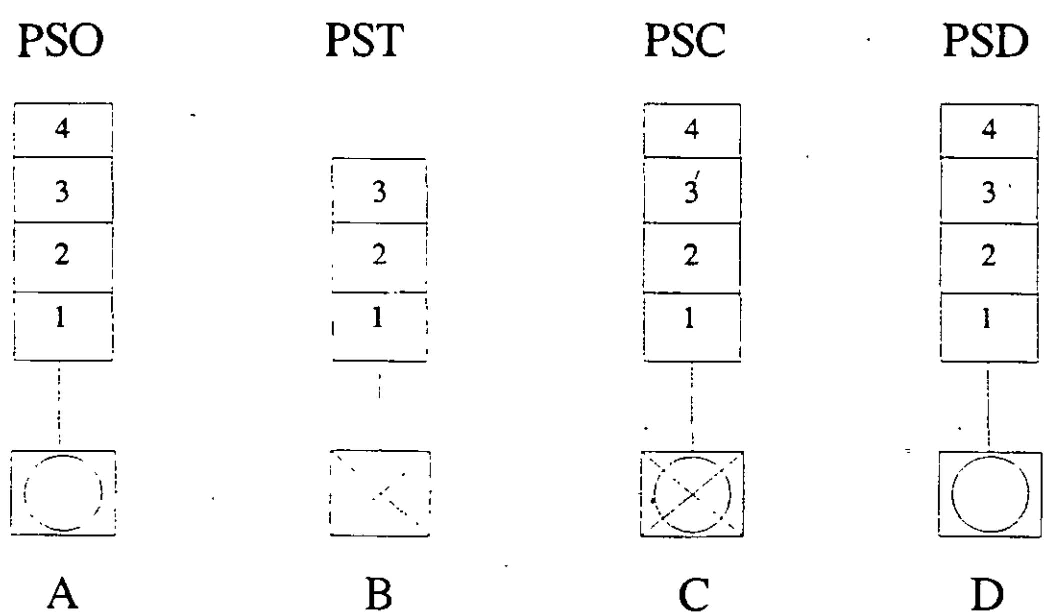
- Confiabilidad
- Accesibilidad
- Niveles jerárquicos
- Posibilidades de reconfiguración
- Tiempos de transferencia

La planeación de la red de señalización debe considerar la arquitectura de la red y las características funcionales de los equipos terminales, como un solo sistema, ya que están directamente relacionados.

Nomenclatura	Símbolo
PSX PSO = PSX de origen PSD = PSX de destino	

Nomenclatura	Símbolo
PST	

Nomenclatura	Símbolo
PSC	



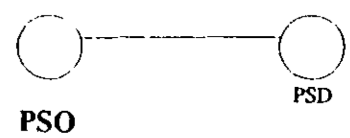
Puntos de señalización

Red SCC7

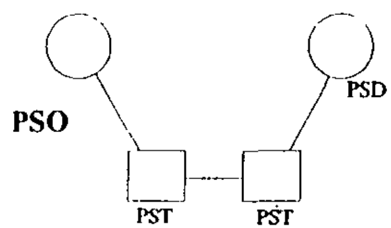
Es una red de telecomunicación que da servicio al sistema de señalización por canal común, compuesta por un número de nodos (puntos de señalización) de conmutación y proceso, interconectados por enlaces de transmisión (enlaces de señalización)

Modos de señalización:

Modo Asociado

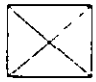
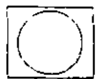


Modo Cuasiasociado

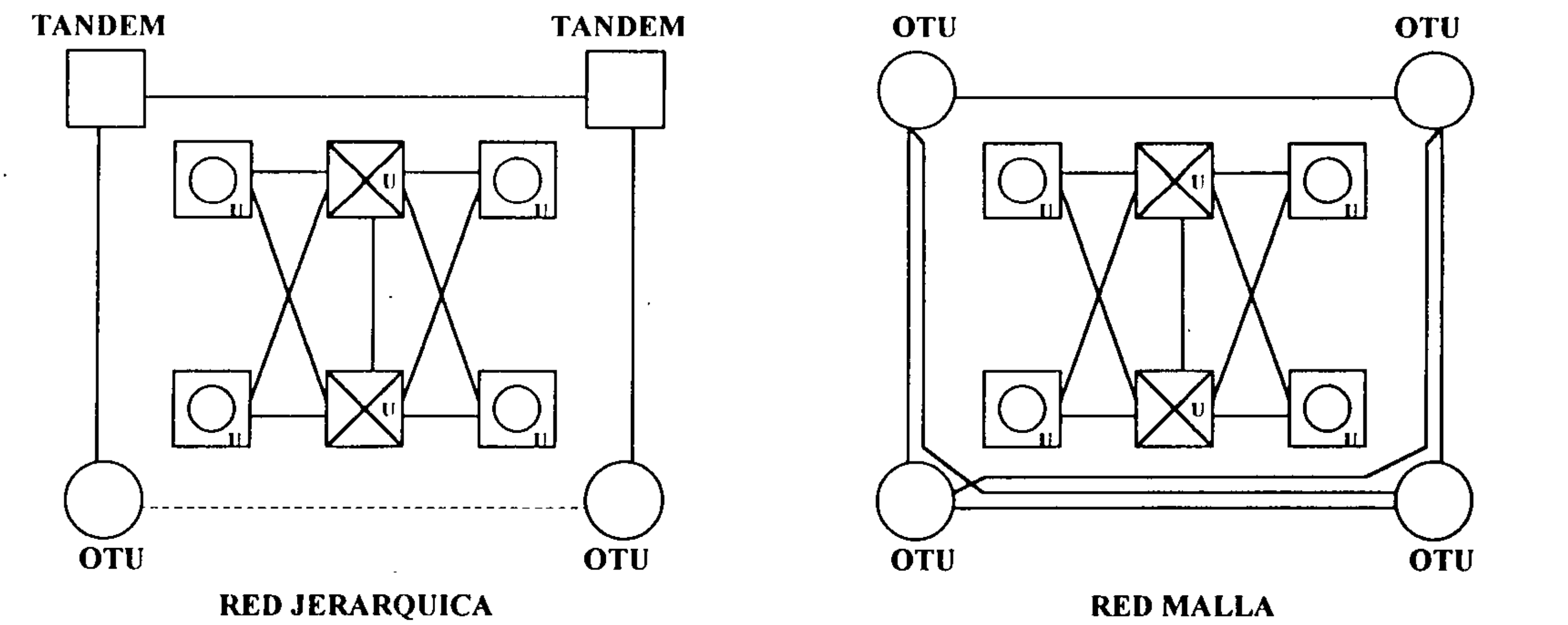


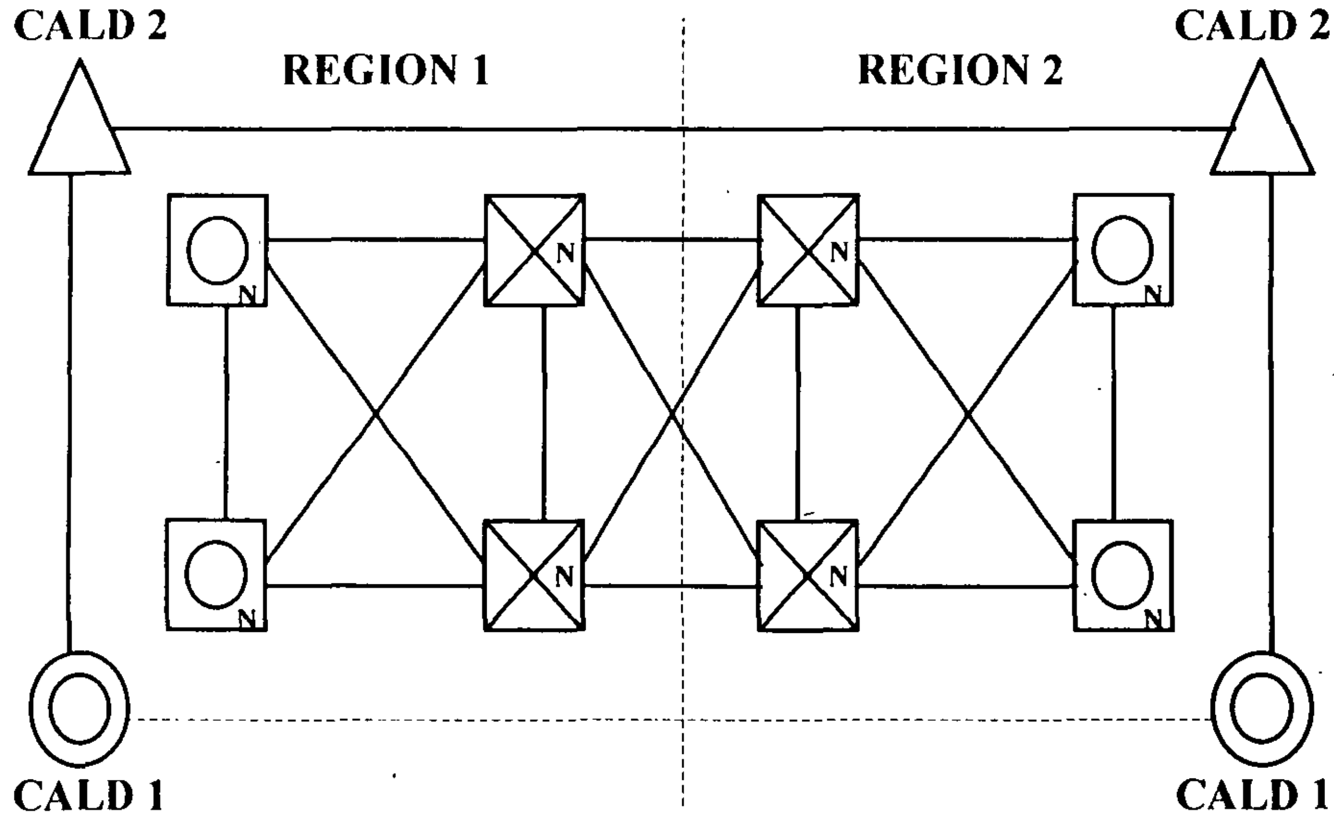
Relación de señalización

—— Red SCC7

Símbolo	Nomenclatura	Descripción
	PST	punto de señalización de transferencia
	VS	via de señalización
	PSX	punto de señalización terminal

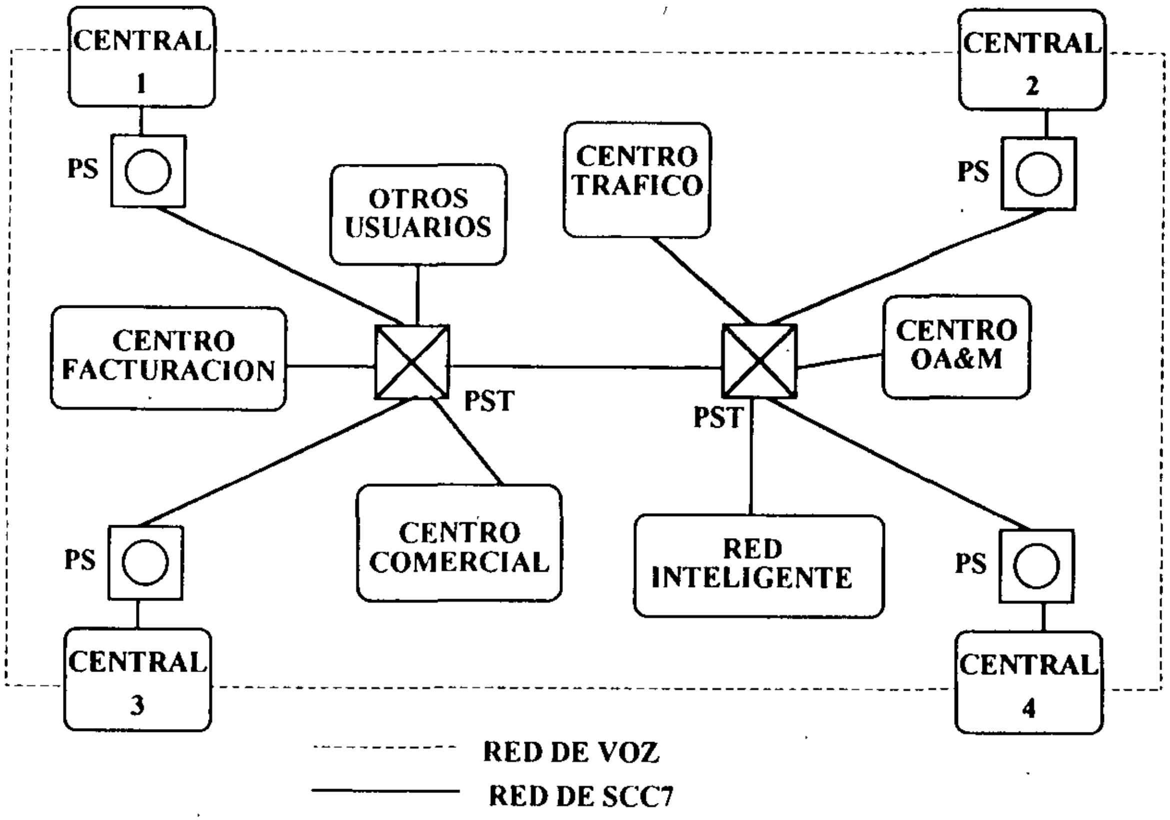
Tipo de Error	Tasa de Error Máxima	Comentarios
Pérdida de mensajes	10^{-7}	Como consecuencia de una falla en la PTM, no se deberá perder más de un mensaje de cada 10^{7} mensajes.
Secuencia incorrecta de mensajes	10^{-10}	Para el modo cuasi-asociado y como consecuencia de una falla en la PTM no se deberá entregar más de un mensaje fuera de secuencia de cada 10^{10} mensajes. Se considera tambien la duplicación de mensajes.
Errores no detectados	10^{-10}	Como consecuencia de una falla en la PTM, no se deberá entregar más de un mensaje con información errónea de cada 10^{10} mensajes.
Indisponibilidad de un conjunto de rutas de señalización.	10 min/año	Como consecuencia de una falla en los PS's y/o VS's que constituyen el conjunto de rutas de señalización.


TANDEM
TANDEM
OTU
OTU
RED JERARQUICA
OTU
OTU
OTU
OTU
RED MALLA



RED DE LARGA DISTANCIA NACIONAL

FACILIDADES DE LA RED SCC7



USM

BANDERA 01111110	BITS DE CONTROL DE ERRORES BCE	CAMPO DE INFORMACION DE SEÑALIZACION CIS	OCTETO DE INFORMACION DE SERVICIO OIS		INDICADOR DE LONGITUD IL	BID	NUMERO SECUENCIAL DIRECTO NSD	BII	NUMERO SECUENCIAL INVERSO NSI	BANDERA 01111110
8	16	n X 8	8	2	6	1	7	1	7	8

USE

BANDERA 01111110	BITS DE CONTROL DE ERRORES BCE	CAMPO DE ESTADO CE		INDICADOR DE LONGITUD IL	BID	NUMERO SECUENCIAL DIRECTO NSD	BII	NUMERO SECUENCIAL INVERSO NSI	BANDERA 01111110
8	16	n X 8	2	6	1	7	1	7	8

USR

BANDERA 01111110	BITS DE CONTROL DE ERRORES BCE		INDICADOR DE LONGITUD IL	BID	NUMERO SECUENCIAL DIRECTO NSD	BII	NUMERO SECUENCIAL INVERSO NSI	BANDERA 01111110
8	16	2	6	1	7	1	7	8

Octeto de información de servicio (OIS)

Campo de Subservicio		Indicador de Servicio
indicador de red	reserva	
D C	B A	D C B A

D C	Asignación
0 0	Red internacional / mundial
0 1	reserva internacional / mundial
1 0	red nacional
1 1	reserva nacional

D C B A	Asignación de PU
0 0 0 0	mensajes de gestión de red SCC7
0 0 0 1	mensajes de prueba y mantenimiento de la red SCC7
0 0 1 0	reserva
0 0 1 1	parte de control de la conexión de señalización (PCCS)
0 1 0 0	parte de usuario de telefonía (PUT)
0 1 0 1	parte de usuario de la RDSI (PUSI)
0 1 1 0	parte de usuario de datos (PUD) [llamadas y circuitos]
0 1 1 1	parte de usuario de datos (PUDF) [registro y cancelación de facilidades]
1 0 0 0 a 1 1 1 1	reserva

PARTE DE USUARIO TELEFONICO (TUP)

La parte de usuario de telefonía define las funciones de señalización telefónicas necesarias mediante la utilización del sistema de señalización No. 7, para el control de llamadas de servicios de telecomunicaciones tales como telefonía y transmisión de datos por conmutación de circuitos.

Se ha especificado con el propósito de que tenga las mismas características de señalización telefónica que otros sistemas utilizados en la RTPC, de modo que pueda existir interfuncionamiento entre ellos.

La especificación de la parte de usuario de telefonía, define las señales y tipos de mensajes que serán utilizados para el establecimiento de enlaces nacionales e internacionales con el fin de tener las mismas características de señalización. Sin embargo se permite a las administraciones una capacidad de reserva para aplicaciones propietarias.

El intercambio de información entre partes de usuario se lleva a cabo mediante unidades de señalización de mensajes (USM), mismas que contienen formatos y códigos específicos para cada parte de usuario. Las USM referentes a partes de usuario telefónico contienen información de servicio, señalización telefónica e información de administración de la red de señalización.

Las USM son grupos de bits que constituyen por si mismas entidades transferibles en forma separada y que se utilizan para transportar información.

El conjunto de mensajes, los parámetros y los procedimientos especificados para el protocolo de la TUP están basados en las Recomendaciones Q.721 a Q.725 del CCITT.

CIS

BAN 01111110	BCE	CAMPO DE MENSAJES Y SEÑALES	CODIGO DE ENCABEZAMIENTO		ETIQUETA 'A'			OIS								
			E1	E0	CIC	CPO	CPD	CS	IS		IL	BID	NSD	BII	NSI	BAN 01111110
8	16	n X 8	4	4	12	14	14	4	4	2	6	1	7	1	7	8

Formato general de la USM para TUP

CIS

BAN 01111110	BCE	CAMPO DE MENSAJES Y SEÑALES	TIPO DE MENSAJE	ETIQUETA 'B'					OIS								
					CIC	SCS	CPO	CPD	CS	IS		IL	BID	NSD	BII	NSI	BAN 01111110
8	16	n X 8	8	4	12	4	14	14	4	4	2	6	1	7	1	7	8

PARTE FACULTATIVA FIJA/VARIABLE	PARTE OBLIGATORIA LONGITUD VARIABLE	PARTE OBLIGATORIA DE LONGITUD FIJA

Formato general de la USM para ISUP

PARTE DE USUARIO DE LA RDSI (ISUP)

La parte de usuario RDSI (ISUP) es el protocolo del sistema de señalización No. 7 que proporciona las funciones de señalización necesarias para el servicio portador básico, así como para servicios suplementarios para aplicaciones vocales y no vocales en una red digital de servicios integrados.

La parte de usuario RDSI es también apropiada para su uso en redes telefónicas especializadas y redes de datos con conmutación de circuitos, así como en redes analógicas y mixtas analógicas/digitales. En especial, la parte de usuario RDSI satisface los requisitos definidos para el manejo del tráfico de datos con conmutación de circuitos y telefónico automático y semiautomático internacional mundial.

Además, la parte de usuario RDSI se presta para las aplicaciones nacionales. La mayor parte de los procedimientos de señalización, elementos de señalización y tipos de mensaje especificados para uso internacional son también necesarios en las aplicaciones nacionales típicas como lo son; el servicio CENTREX de Cobertura Amplia (CCA) y la Red Inteligente (RI).

La parte de usuario RDSI utiliza los servicios proporcionados por la parte transferencia de mensajes (MTP) para la transferencia de información entre partes de usuario RDSI.

Los requisitos de numeración para la RDSI siguen el plan de numeración internacional definido para la RDSI para proporcionar un servicio básico con conmutación de circuitos entre terminales RDSI o entre éstos y los terminales que se conectan a la red telefónica nacional o internacional existente.

El conjunto de mensajes, los parámetros y los procedimientos especificados para el protocolo de la ISUP están basados en las Recomendaciones Q.761 a Q.764 y Q.767 del CCITT.

Formato del CIIPS

Código de zona / red		
Identificación de región	Identificación de zona / red	Identificación de PS
N M L	KJIHGFED	C B A

Código de zona / red			
Identificación de región		Identificación de zona / red	
decimal	binario	decimal	binario
3	011	068	01000100

PSXI	Identificación de PS	
	decimal	binario
Tulancingo	0	000
México	1	001
Monterrey	2	010
Cd. Juarez	3	011
Nogales	4	100
Tijuana	5	101

PUT - Mensajes

		E1 / EO	0000	0001	0010	0011	0100	0101	0110	0111	1000	1001	1010	1011	1100	1101	1110	1111
		0000	reserva para uso nacional															
MDA	→	0001		MID	MIA	MSD	SDU											
MEL	⇐	0010		MIE		COM	FCO											
MPE	⇐	0011		MPG														
MEC	⇐	0100		MDC														
MEI	←	0101		CEC	CGC	CRN	SDI	SLI	ABO	NNA	LFS	TIE	SAP	TDN	PRM			
MSL	⇔	0110	SRS	RCT	RST	COL	FIN	RRE	INT	SLA								
MSC	⇔	0111		LGU	BLO	ARB	DBL	ARD	PPC	RCI								
MSG	⇔	1000		BGM	ABM	DGM	ADM	BGE	ABE	DGE	ADE	MRG	ARG	BGL	ABL	DGL	ADL	
		1001			reserva para uso internacional													
GRC	⇐	1010		CCA														
		1011																
MND	⇒	1100					OFR	CAN	REL									
MNA	⇐	1101					FAN											
MNP	⇐	1110					TEB											
		1111	reserva para uso nacional															

	MENSAJE	CODIGO
IAM	Mensaje Inicial de Dirección	00000001
SAM	Dirección (o número) subsiguiente	00000010
INR	Petición de información	00000011
INF	Información	00000100
COT	Continuidad	00000101
SGM	Segmentación	00111000
UPA	Parte de usuario disponible	00110101
UPT	Prueba de parte de usuario	00110100
CRG	Información de tasación	00110001
ACM	Dirección completa	00000110
CON	Conexión	00000111
CPG	Progresión de la llamada	00101100
ANM	Respuesta	00001001
FOT	Transferencia hacia adelante (Intervención)	00001000
REL	Liberación	00001100
IDR	Petición de identificación	00110110
IRS	Respuesta de identificación	00110111
DRS	Liberación diferida	00100111
RLC	Liberación completada	00010000
CCR	Petición de prueba de continuidad	00010001
RSC	Reinicialización de circuito	00010010
LPA	Acuse de establecimiento de bucle	00101000
BLO	Bloqueo	00010011
UBL	Desbloqueo	00010100
UCIC	Código de identificación de circuito no equipado	00101110
BLA	Acuse de bloqueo	00010101
UBA	Acuse de desbloqueo	00010110
OLM	Sobrecarga	00110000
SUS	Suspensión	00001101
RES	Reanudación	00001110
CFN	Confusión	00101111
CGB	Bloqueo de grupo de circuitos	00011000
CGU	Desbloqueo de grupo de circuitos	00011001
CGBA	Acuse de bloqueo de grupo de circuitos	00011010
CGUA	Acuse de desbloqueo de grupo de circuitos	00011011
GRS	Reinicialización de grupo de circuitos	00010111
GRA	Acuse de reinicialización de grupo de circuitos	00101001
CQM	Indagación sobre grupo de circuitos	00101010
CQR	Respuesta a indagación sobre grupo de circuitos	00101011
CMR	Petición de modificación de llamada	00011100
CMC	Modificación de llamada completada	00011101
CMRJ	Rechazo de modificación de llamada	00011110
FAA	Facilidad aceptada	00100000
FAR	Petición de facilidad	00011111
FAC	Facilidad	00110011
FRJ	Rechazo de facilidad	00100001
NRM	Gestión de recurso de red	00110010
PAM	Paso de largo	00101000
USR	Información de usuario a usuario	00101101
OFR	Oferta	11111100
RLL	Rellamada	11111110
CAN	Cancelación de oferta	11111101
FAN	Falsa contestación	11111111

CODIFICACION DE MENSAJES DE LA PUSI

PARTE DE CONTROL DE LA CONEXION DE SEÑALIZACION (SCCP).

La Parte de Control de Conexión de Señalización (SCCP) proporciona funciones adicionales a las de la Parte de Transferencia de Mensajes (MTP) con objeto de prestar servicios de red sin conexión y servicios de red con conexión, para transferir información de señalización relacionada con el circuito y no relacionada con el circuito de los usuarios de la SCCP, tales como; la Parte de Usuario de la Red Digital de Servicios Integrados (ISUP), TCAP, gestión de la SCCP, etc., e información de otros tipos entre las centrales y centros especializados en la red de telecomunicaciones vía una red del sistema de señalización por canal común No. 7 (SCC7).

La combinación de la MTP y la SCCP se denomina Parte de Servicio de Red y debe reunir los requisitos de red definidos por el modelo de la International Standards Organization (ISO).

El protocolo de SCCP debe utilizarse entre dos sistemas que proporcionan el servicio de Parte de Servicio de Red a las capas superiores. El intercambio de información entre los usuarios de la PCCS permite:

a) El establecimiento de conexiones de señalización lógicas;
b) la liberación de las conexiones de señalización lógicas;
c) transferencia de datos con conexiones de señalización lógicas y sin ellas.

La especificación de la SCCP se encuentra en las recomendaciones Q.711, Q.712, Q.713, Q.714 y Q.716 del CCITT.

PARTE DE APLICACION DE CAPACIDADES DE TRANSACCION (TCAP).

Las capacidades de transacción (TC) proporcionan funciones y protocolos a gran número de aplicaciones distribuídas entre centrales y centros especializados en las redes de telecomunicación.

La TCAP forma parte de la capa 7 de acuerdo al modelo de referencia de la International Standards Organization (ISO).

La finalidad general de las Capacidades de Transacción es proporcionar medios de transferencia de información entre nodos, así como suministrar servicios genéricos a las aplicaciones, aunque manteniendo su independencia con respecto a ellas.

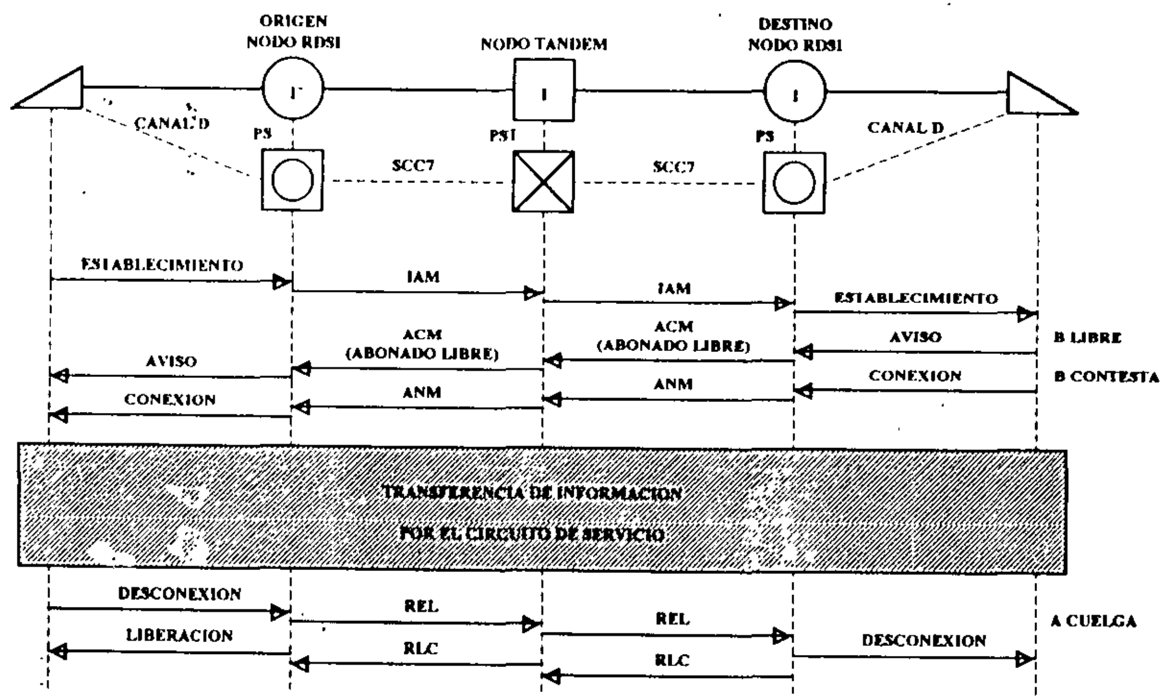
Las TC de la red con sistema de señalización por canal común No. 7,deben poder ser utilizdas entre:

- centrales;
- una central y un centro de servicio;
- centros de servicio.

La TCAP utilizará el servicio de red del sistema de señalización por canal común No.7, por lo que utilizará las opciones de direccionamiento soportadas por la Parte de control de la conexion de señalizacion (SCCP).

La especificación de la TCAP se encuentra en las recomendaciones Q.771, Q.772, Q.773, Q.774 y Q.775 del CCITT.

LLAMADA ORDINARIA CON SEÑALIZACION DE PUSI PARA LA RDSI



PLAN TECNICO FUNDAMENTAL DE SEÑALIZACION.

Objetivo.

El presente plan tiene como objetivo establecer las bases para el adecuado uso y administración de los recursos nacionales asociados a la señalización entre redes públicas de telecomunicaciones. Con el fin de lograr la eficiente interconexión e interoperabilidad de dichas redes en beneficio de los usuarios y Operadores de Telecomunicaciones de México. Los criterios rectores de este plan son la asignación eficiente, justa, equitativa y no discriminatoria de los recursos disponibles.

I. Definición de términos.

Operador :

Persona física o moral que cuenta con un título de concesión o permiso que le autoriza a prestar servicios de telecomunicaciones.

Operador de larga distancia:

Persona física o moral que cuenta con un título de concesión o permiso que le autoriza a prestar el servicio de larga distancia.

Señalización por canal común 7 (SCC-7):

Norma Internacional de señalización que utiliza una red separada de transporte de señales.

Protocolo de Parte de Usuario para Servicios Integrados - México (PAUSI-MX):

Protocolo internacional de Parte de Usuario para Servicios Integrados adaptado a las características técnicas locales.

Punto de Señalización (PS):

Punto a través del cual se tiene acceso a una red de señalización.

Punto de transferencia de señalización (PTS):

Punto inteligente de transferencia dentro de una red de señalización.

II. Administración del Plan Técnico Fundamental de Señalización.

Para la debida administración del presente Plan, la Secretaría de Comunicaciones y Transportes tendrá, entre otras, las siguientes facultades:

- Interpretar el presente Plan para efectos administrativos.
- Asignar los códigos de puntos de señalización Nacionales e Internacionales a los Operadores.
- Representar a México ante la UIT en materia de señalización.
- Solicitar códigos de zona de señalización e identificación de red para los códigos de puntos de señalización Internacional ante la UIT.
- Supervisar y controlar los recursos del Plan.
- Establecer y poner a disposición de los operadores información referente a la señalización

III. Códigos de puntos de señalización.

La red mundial de señalización está estructurada en dos niveles funcionales, lo que permite que los planes de asignación de códigos para puntos de señalización Nacionales pueden ser independientes unos de otros.

1. Estructura de los códigos de puntos de señalización nacionales (CPSN)

Los CPSN tendrán una estructura de 14 bits basada en la recomendación Q.704 de la UIT.

2. Estructura a la que deberán sujetarse los CPSN.

Para satisfacer las necesidades de los operadores entrantes, la estructura de los CPSN será variable en función del tamaño de las redes, utilizando un número "n" de bits para la identificación del operador de la red, y los remanentes "14-n" bits para ser administrados y asignados independientemente al interior de cada red.

Para lograr un uso y administración eficientes de los CPSN, la Secretaría asignará a los operadores códigos de 14 bits, conformados de acuerdo con cualquiera de las siguientes 3 estructuras:

a) Una estructura con 3 bits para la identificación del operador y 11 bits para la asignación interna de bloques de 2048 CPSN. El código 000 se mantendrá como reserva.

0	0	1											

3 bits
Código de
Identificación
de Operador

11 bits que definen 2048
CPSN

b) Una estructura con 7 bits para la identificación del operador y 7 bits para la asignación interna de bloques de 128 CPSN La Secretaría asignará esta estructura a los operadores de redes que, por su compleja topología, así lo requieran.

7 bits
Código de identificación de operador

7 bits que definen 128
CPSN

c) Una estructura con 11 bits para la identificación del operador y 3 bits para la asignación interna de bloques de 8 CPSN La secretaría asignará esta estructura a redes que, por la simplicidad de su topología, así lo requieran o bien a conjuntos de PS sin funcionalidad de PTS

11 bits
Código de identificación de operador

3 bits que
definen 8 CPSN

3. Estructura de los códigos de puntos de señalización Internacionales (CPSI).

La estructura de los CPSI está descrita en la Recomendación UIT-T Q.708 y se compone de tres elementos:

- Un identificador de región de 3 bits,
- un identificador de red de 8 bits y
- un identificador de punto de señalización de 3 bits.

Los dos primeros elementos conforman el código de zona de señalización/identificación de red (CZRS) Y SON Administrados por la U.I.T. La estructura de estos códigos se indica a continuación:

N	M	L	K	J	I	H	G	F	E	D	C	B	A

Identificador de región | Identificador de red | Identificador de Punto de Señalización

A la fecha México tiene asignados 4 CRZS: 3-068 ,3-069, 3-070 Y 3-071, correspondientes a 32 códigos de punto de señalización Internacional.

IV. Asignación de códigos de punto de señalización.

Todas las redes de señalización por canal común, dentro del territorio mexicano, deberán contar al menos, con un código de identificación de operador de red de señalización.

V. Protocolos de señalización.

El Protocolo PAUSI-MX será el protocolo que deberán usar las redes públicas de telecomunicaciones para su interconexión.

Los operadores deberán informar a la Secretaría y a los demás operadores respecto de cualquier nueva funcionalidad en sus protocolos que pretendan implantar en su red.

El sistema de señalización de las redes públicas de telecomunicaciones deberá estar preparado para permitir la introducción de protocolos especializados, tales como PACT y la PCCS, a fin de que puedan prestar nuevos servicios de telecomunicaciones en el país.

VI. Intercambio de información en la interconexión de redes.

Además de la información necesaria para establecer y liberar la llamada, la información mínima que deberá intercambiarse en tiempo real para la interconexión de redes será la siguiente:

- Número de A con formato de número nacional.
- La categoría de A conteniendo, al menos, la información que indique si se trata de una llamada de teléfono público, abonado normal o de operadora.
- El Número de B en formato nacional (red de destino) o en formato nacional o internacional (tránsito).
- Estado de B incluyendo al menos Número de A la información que permita determinar si la llamada ha sido contestada o si la línea se encuentra libre, ocupada o congestionada.
- Adicionalmente, la llamada deberá acompañarse de la información relativa al tipo de servicio, tipo de selección de red y la necesaria para su tarifación, de conformidad con las indicaciones señaladas por el usuario a través del procedimiento de marcación empleado.
- La información adicional para tasación de la llamada como el número de cargo (cuando sea diferente del número de A o códigos o prefijos de servicios cuando B paga).
- La información para el establecimiento de la llamada se enviará en bloque completo en un mensaje MID y no en forma traslapada.

FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

<u>EVOLUCIÓN DE LAS REDES DE TELECOMUNICACIONES</u>

PALACIO DE MINERÍA
JUNIO DE 1999

ALCATEL

Evolución de las Redes de Telecomunicaciones

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Contenido

- ▼ Evolución y Tendencia del Servicio
- ▼ Modelos de Red
- ▼ Tendencia General
- ▼ Visión de Evolución Bellcore
- ▼ Modelo de Evolución de Alcatel
- ▼ Conclusiones

Evolución y Tendencia del Servicio

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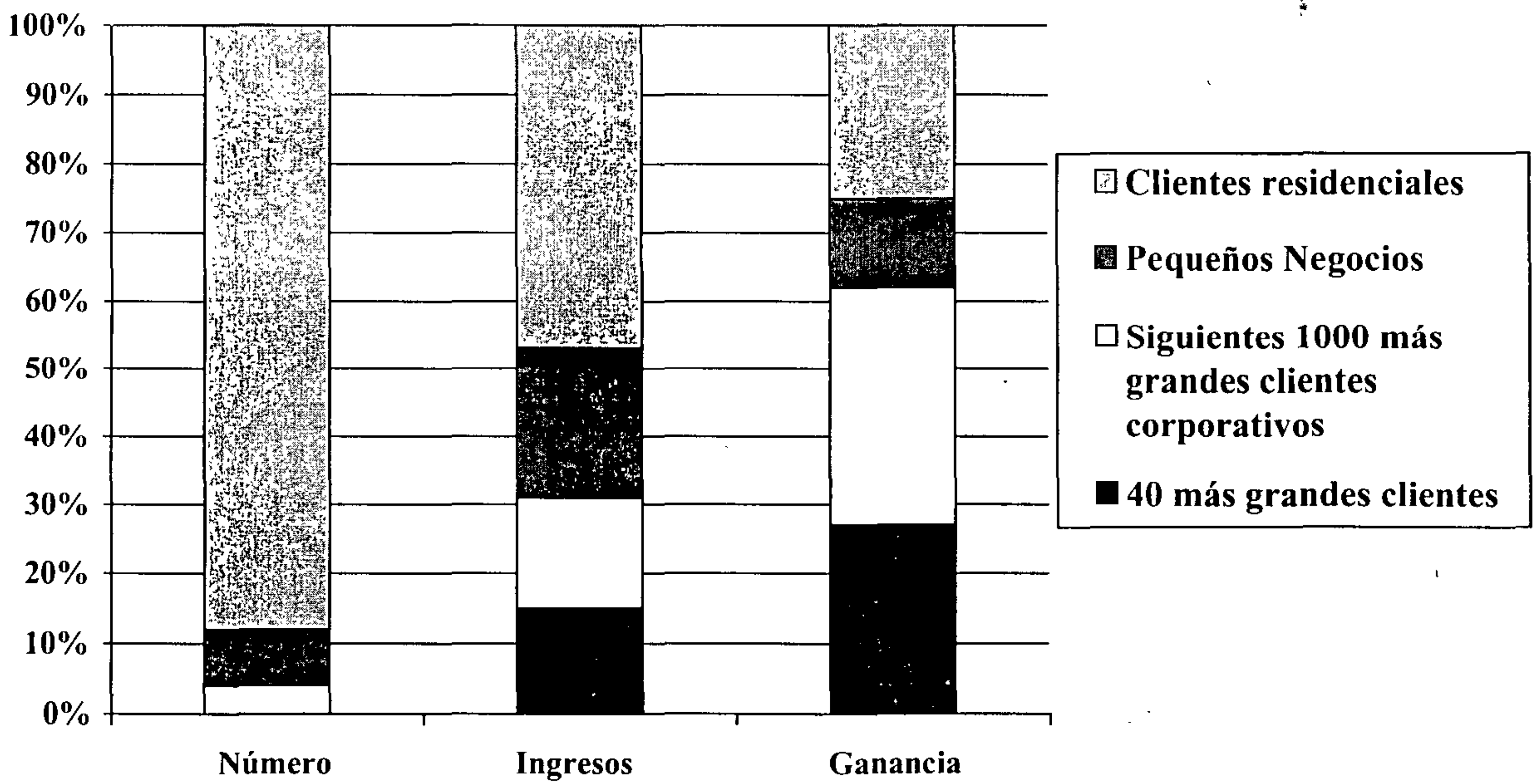
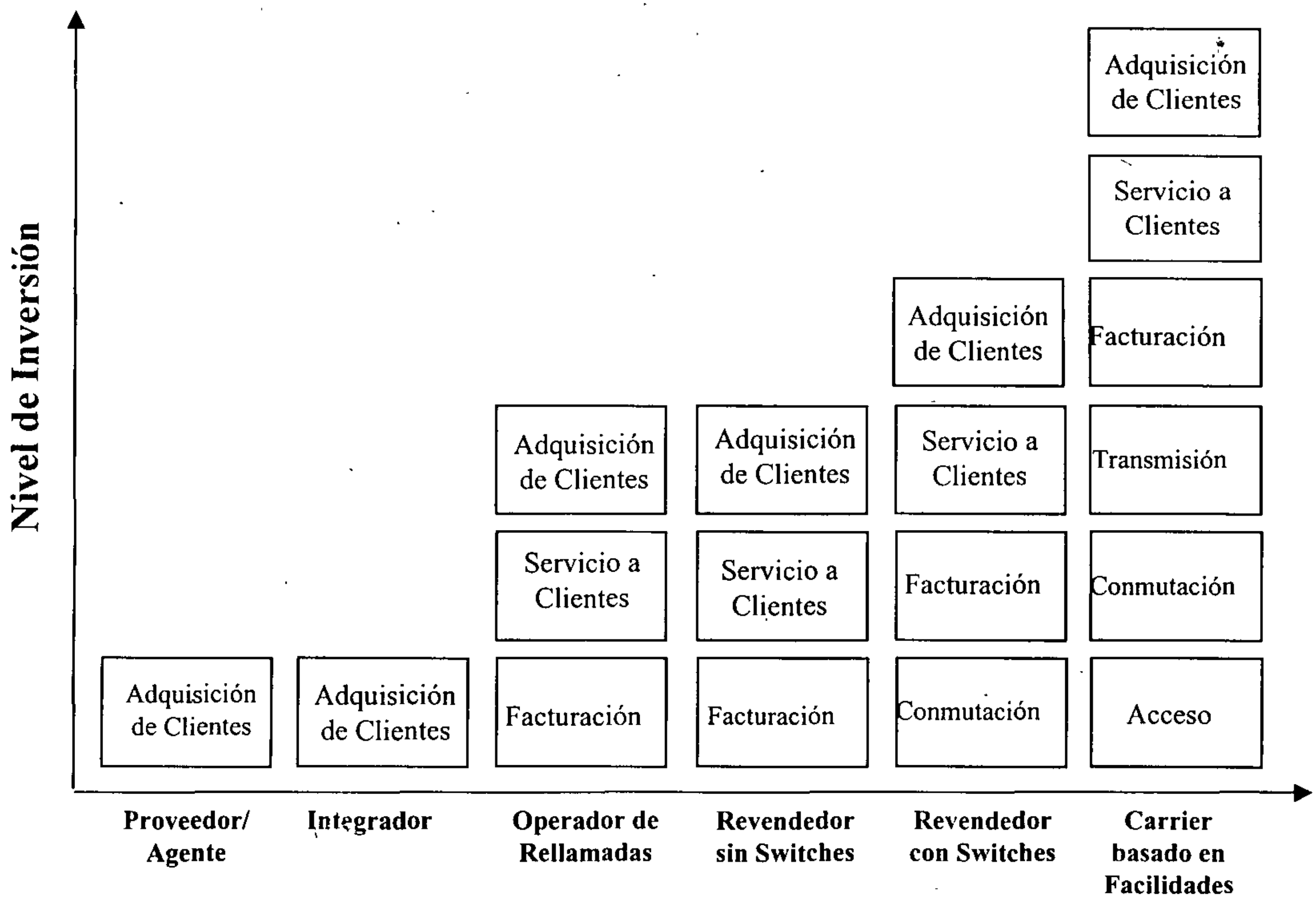
Distribución de Rentabilidad



▼ Los grandes clientes corporativos dominan la rentabilidad

Nuevas Estrategias de Ataque para la Entrada



Dinámicas Competitivas
Ataque Basado en Facilidades

Sitios

	Corporativo	Empresas medianas y pequeñas	Proveedores Individuales	Residencial
Doméstico	**Teleworkers** PC y línea dedicada. Requiere E-mail / Acceso FTP. Normalmente, trabajores de oficina o ejecutivos.	**Teleworkers** Línea dedicada para usuarios con carga de tráfico.	**Oficina en Casa** A menudo, la única oficina, con miembros de la familia como empleados. Pueden tener línea dedicada, fax	Usualmente solo una línea dedicada. Penetración del CATV aún baja.
Pequeño	**Oficinas Pequeñas** Pueden ser conectadas a redes corporativas como extensiones externas.	**Oficinas Pequeñas** Pueden ser conectadas a un host como extensiones externas del Sistema Llave.	A menudo solo una línea dedicada. Usuarios de fax con alto tráfico. Primeros adoptadores de servicios móviles.	
Medio	**Sucursales** A menudo con Sistema Llave y LAN propios. Necesitan conexiones de voz y datos a la red corporativa.	**Oficina Central** Normalmente más de 6 líneas dedicadas, más el Sistema Llave. En ocasiones una línea dedicada para el fax.		
Grande	**Grandes Sitios** PABX, ACDs, mainframe hosts, servidores, etc.			

Usuarios

- **Más altos niveles de tráfico e ingresos**
- **Más bajo costo para dar servicio**
- **Corto tiempo para implementación: manera más rápida para atraer ingresos a caudales**
- **Por lo tanto, blanco primario para nuevos operadores**

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Dinámicas Competitivas
Otras Oportunidades de Mercado

Sitios \ Usuarios	Corporativo	Empresas medianas y pequeñas	Proveedores Individuales	Residencial
Doméstico	**Teleworkers** PC y línea dedicada. Requiere E-mail / Acceso FTP. Normalmente, trabajores de oficina o ejecutivos.	**Teleworkers** Línea dedicada para usuarios con carga de tráfico.	**Oficina en Casa** A menudo, la única oficina, con miembros de la familia como empleados. Pueden tener línea dedicada, fax.	Usualmente solo una línea dedicada. Penetración del CATV aun baja.
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Grande	**Grandes Sitios** PABX, ACDs, mainframe hosts, servidores, etc.			

La Zona de Guerra!

Servicios SME:
Centrex
Correo de Voz
E-Mail
Internet
Correo de Fax

Servicios SOHO:
CLASS
Correo de Voz
E-Mail/ Internet/
Correo de Fax
Fijo/Móvil

Res + Servicios :
CLASS
Correo de Voz
E-Mail/Internet
2da Línea
Fijo/Móvil

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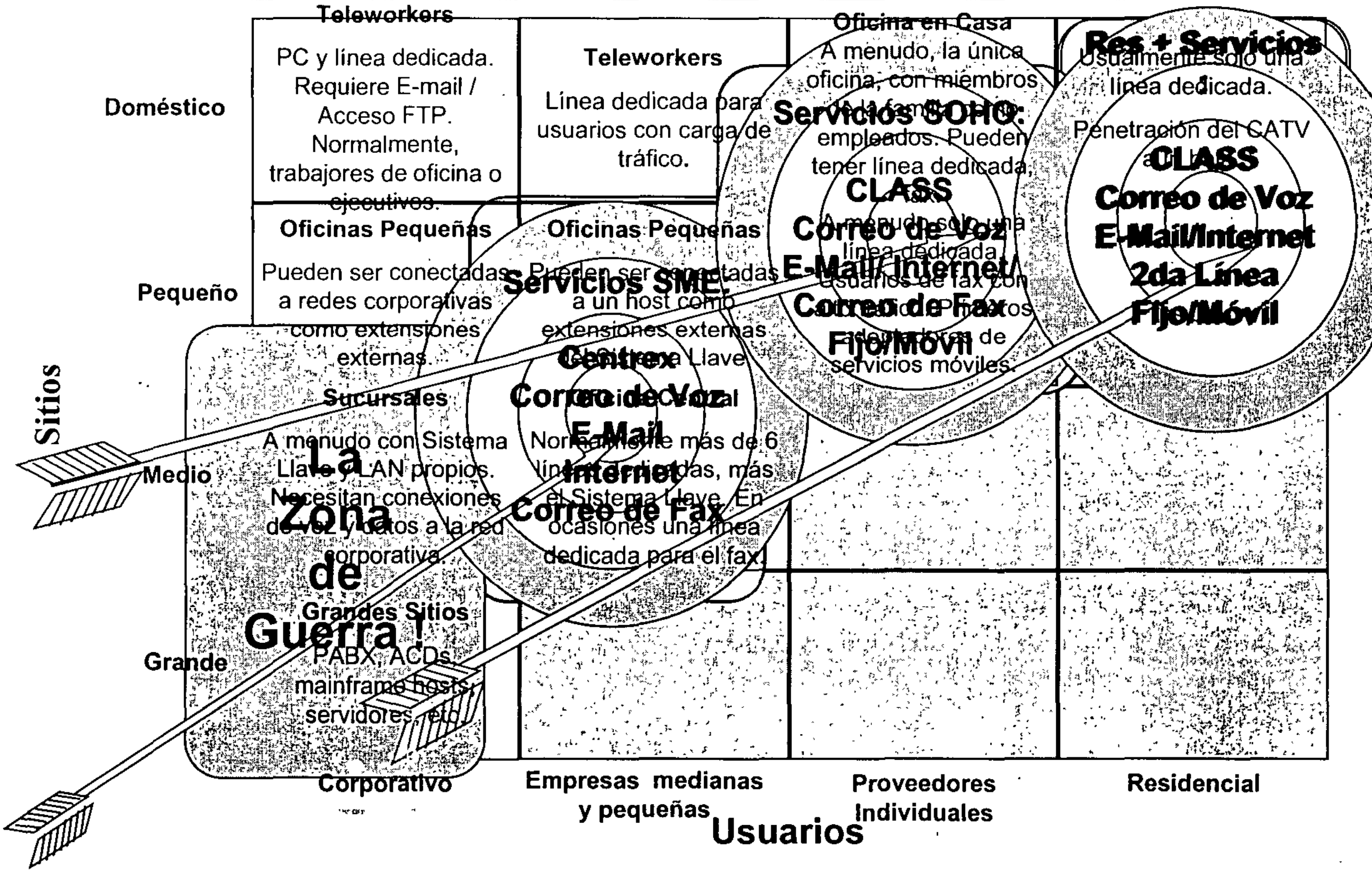
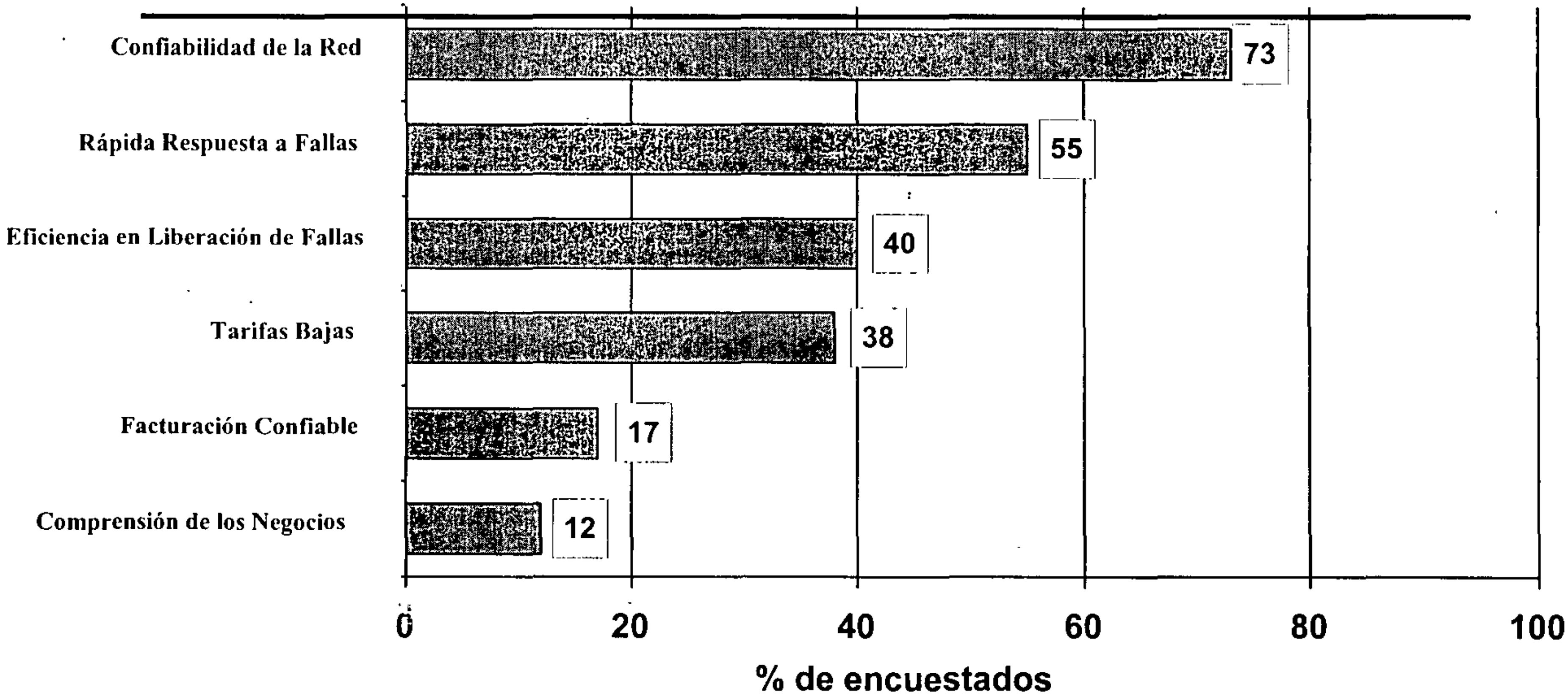
Dinámicas Competitivas
Ataque Basado en No-Facilidades



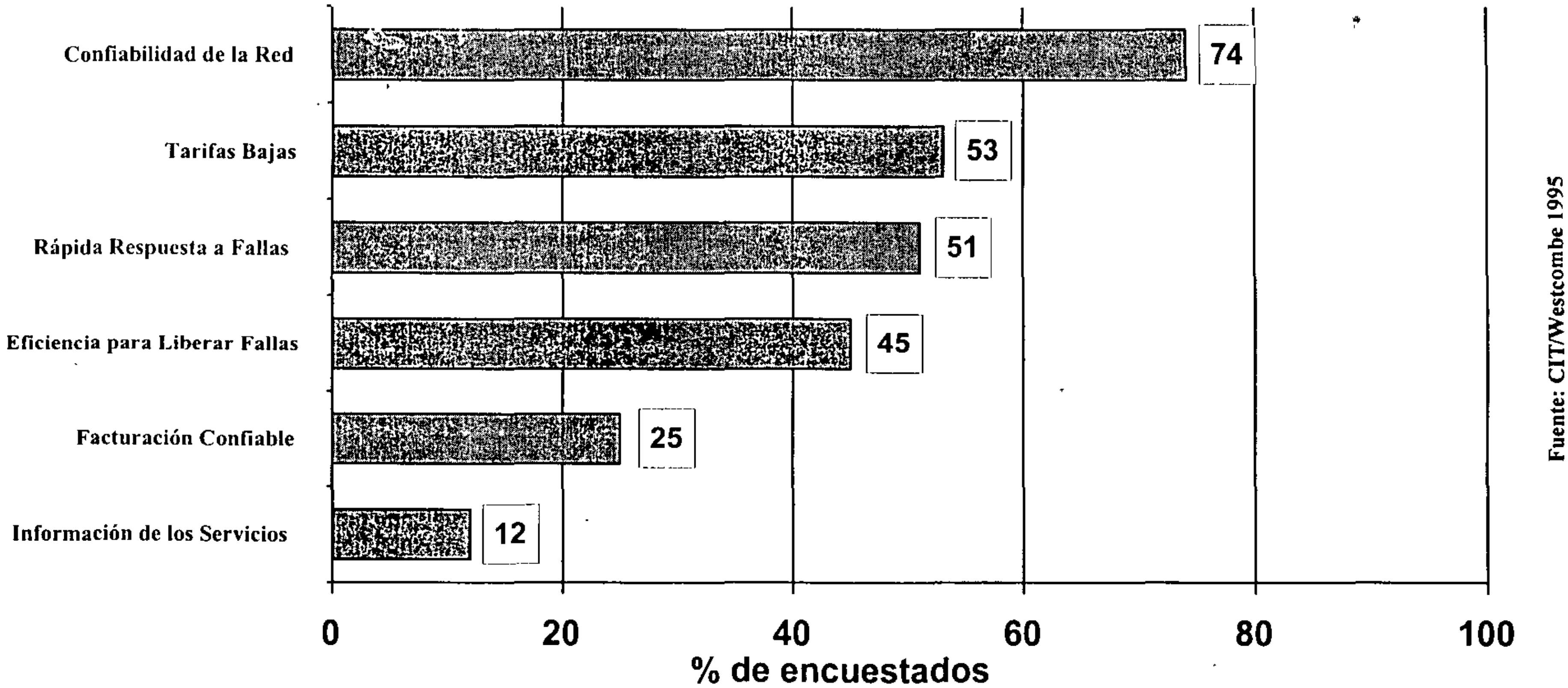
Prioridades del Usuario Corporativo



- Porcentaje de las más importantes prioridades de los usuarios.
- Los ítems no presentados (cobertura, rango de servicios, flexibilidad de facturación, etc.) fueron importantes para menos del 10% de los usuarios
- Muestra de 270 encuestados en compañías con más de 500 empleados

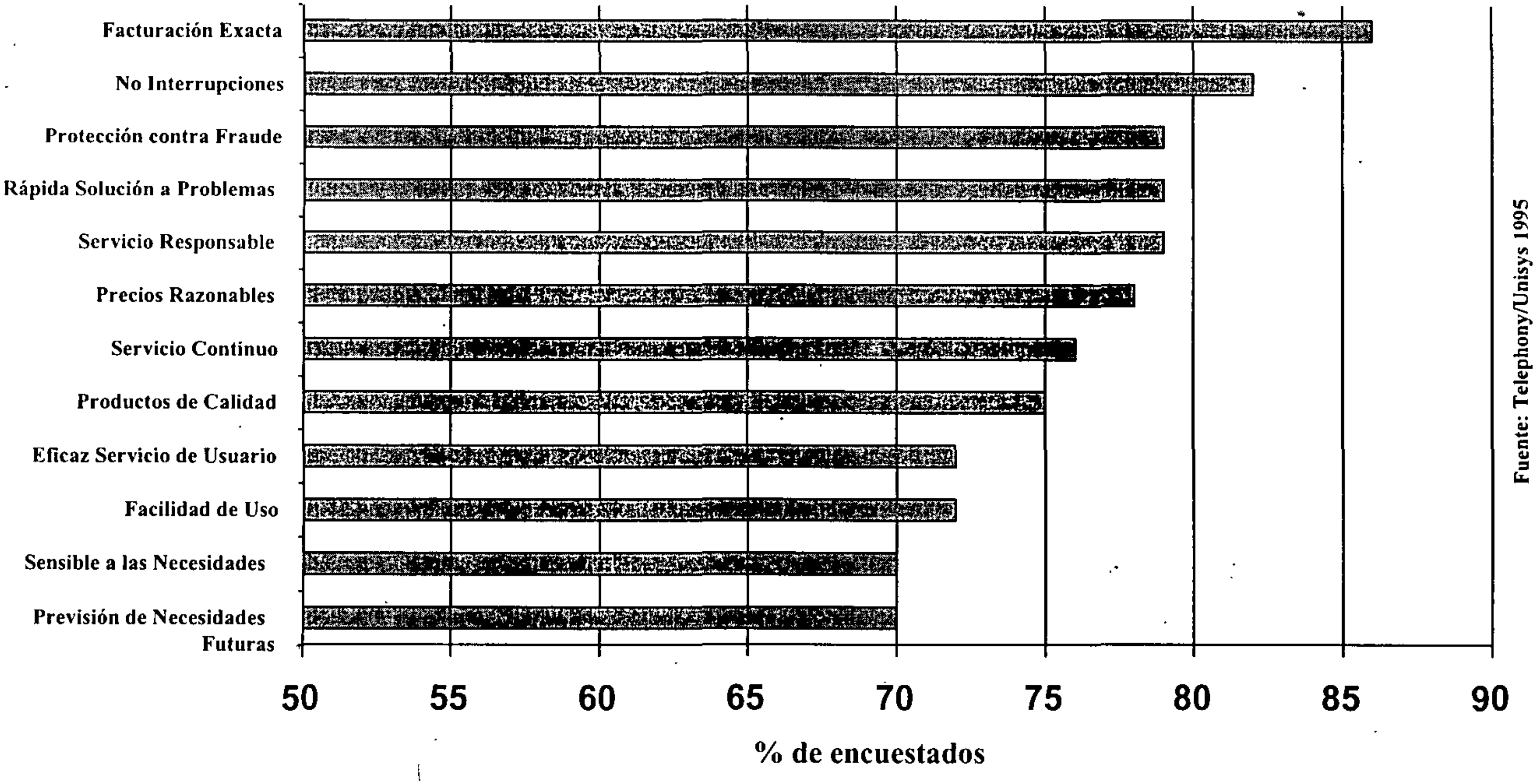
Fuente: CIT/Westcombe 1995

Prioridades de Empresas Medianas y Pequeñas



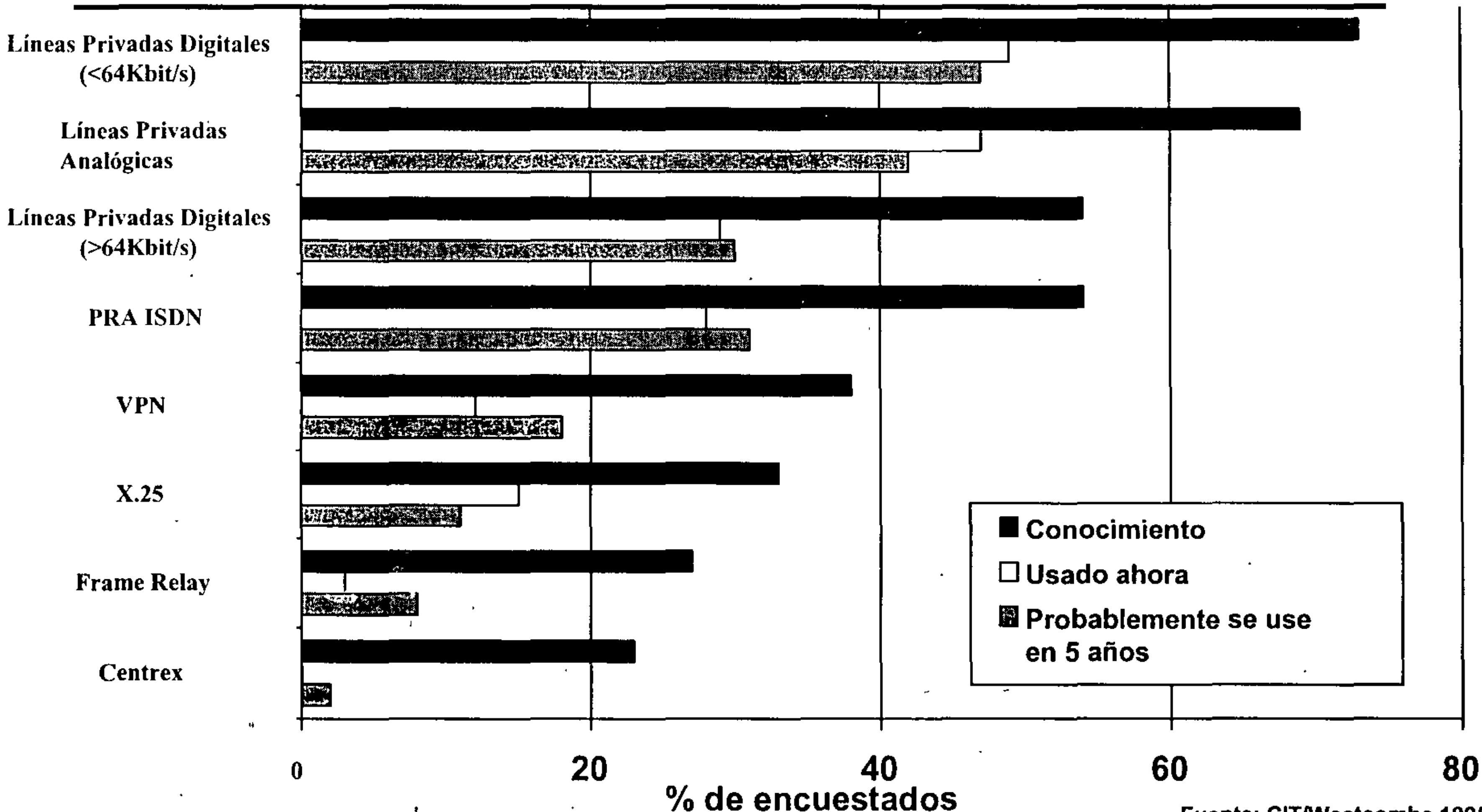
- Porcentaje de las tres más importantes prioridades de los encuestados
- Otros ítems (cobertura, rango de servicios, flexibilidad en la facturación, etc.) fueron importantes para menos del 10% de los encuestados
- Muestra de 370 encuestados en compañías de 1 a 10 empleados, con más de una línea

Prioridades para Usuarios Residenciales



- Porcentaje de encuestados calificando el ítem como "muy importante"
- 12 Mayores prioridades de servicio para una muestra de 250 usuarios residenciales en E.U.

Usuarios Corporativos: Niveles de Conocimiento y Penetración de Servicios



Fuente: CIT/Westcombe 1995

- Los servicios no-PSTN (SMDS, Datos Inalábrico, etc.) abajo del 20% de conocimiento
- Muestra de 276 encuestados de compañías del Reino Unido con más de 500 empleados

Internet e Intranet

Servicio	Ingresos (£m)	Ingresos (%)
Residencial : Internet de Banda Angosta	254	9
Residencial : Internet de Banda Ancha	45	2
Residencial : VoD Interactivo	221	8
Residencial : Difusión de TV	448	16
Total Residencial	*969*	*35*
Negocios: Internet /Intranet B. Angosta	1290	46
Negocios: Internet/Intranet B. Ancha	553	19
Total de Negocios	*1843*	*65*
Total	**2812**	**100**

- **UK, Año 2000**
- **Suposiciones "optimistas" de Banda Ancha**
- **Fuente: Analysis, Junio 1996**

- **Negocios e Internet/Intranet dominan**
- **Servicios de Banda Angosta aún claves**
- **Grandes corporaciones con comunicaciones de mensajes y datos manejan el mercado**

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Diferentes patrones de actividad y presupuestos...

Corporativo

- En casa o en una pequeña oficina
- Por lo general una o dos personas unicamente
- Trabajo de tiempo parcial y tiempo completo
- Costos con cargo directo o indirecto a la corporación.

Una ventana a las empresas de comunicaciones y los sistemas de información

Profesional

- Trabajador independiente
- Usualmente en casa o en una pequeña oficina
- Control de costos eficiente, requerimientos clave
- Paga sus propios costos.

Sistema de administración de negocios y herramientas de productividad

Oficina en Casa

- Familias con cierto conocimiento técnico
- tareas como :procesador de palabra, e-mail, mercadeo en red
- Usos competitivos :juegos y para trabajo en casa
- Pago del presupuesto familiar

Mejoramiento del nivel de vida a través del entretenimiento, la educación y la productividad

... Así como, los críterios de compra ...

Profesional

- Productividad
- confiabilidad/resistencia en el trabajo pesado
- Soporte de calidad
- Herramienta de trabajo y control de costos

Corporativo

- Rendimiento
- Confiabilidad
- Gran interes en la Integración a los servicios corporativos
- Fácil de manejar y administrar

Oficina en el Hogar

- Respaldo de marcas reconocidas
- Oferta segmentada en paquetes
- Fácil de adquirir
- Fácil de usar

← Interesa el costo de uso

Interesa el precio inicial →

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... La instalación y el soporte del servicio son la clave

- La atención a un gran número de pequeños sitios/usuarios presentan retos específicos
- El Servicio completo debe incluir :
  - suministro de servicio relacionado CPE (Tarjetas PC, NTEs,dispositivos de acceso remoto, etc.)
  - suministro de servicio-relacionado con software (clientes, APIs, seguridad,etc.)
  - instalación y activación del sistema
- El Soporte debe incluir
  - reemplazo de hardware y reparación
  - monitoreo del funcionamiento en línea y diagnósticos técnicos de soporte tipo hot-line
  - mantenimiento y soporte en sitio
  - El Servicio tiene que estar disponible en una amplia área geográfica, con una coordinación trans-nacional para mejores resultados

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Experiencia en los E.U. (1)

- La penetración de CLASS ha alcanzado el 26%
  - Dominada por suscripciones de Llamada en Espera
- CLI ha alcanzado una penetración de 12%, y se espera que crezca explosivamente en 1996:
  - CLI en California será lanzado en Abril: esto representa el 30% de todas las llamadas nacionales
  - CLI será nacional en Julio, en cuanto las IXC's acojan el mandato FCC
- Todos los participantes mayores están desplegando ahora ADSI como un mejor MMI para facilidades CLASS y de Correo de Voz
  - 7 RBOC's mas GTE y otras independientes grandes
  - Bell Canada
- Base terminal ADSI
  - 1995 : 290,000
  - 1996 : 1 millón (estimación de NorTel)
- Primer adopción fuertemente relacionada al uso telefónico, más que a factores socioeconómicos o demográficos

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Experiencia en los E.U. (2)

Facilidad	Ingresos Residenc. ($m)	Ingresos por Negocios ($m)	Total ($m)
Llamada en Espera	1687	92	1779
Identidad del Llamante	587	52	639
Redireccionamiento de Llamada	250	143	393
Llamada Tripartita	312	40	352
Rellamada Automática	233	18	251
Marcación Rápida	205	28	233
Bloqueo Selectivo de Llamada	103	13	116
Completación de Llamada en Ocupado	76	12	88
Timbrado Distintivo	45	8	53
Redireccionmaiento Selectivo de Llamada	33	3	36
Total	3531	409	3940

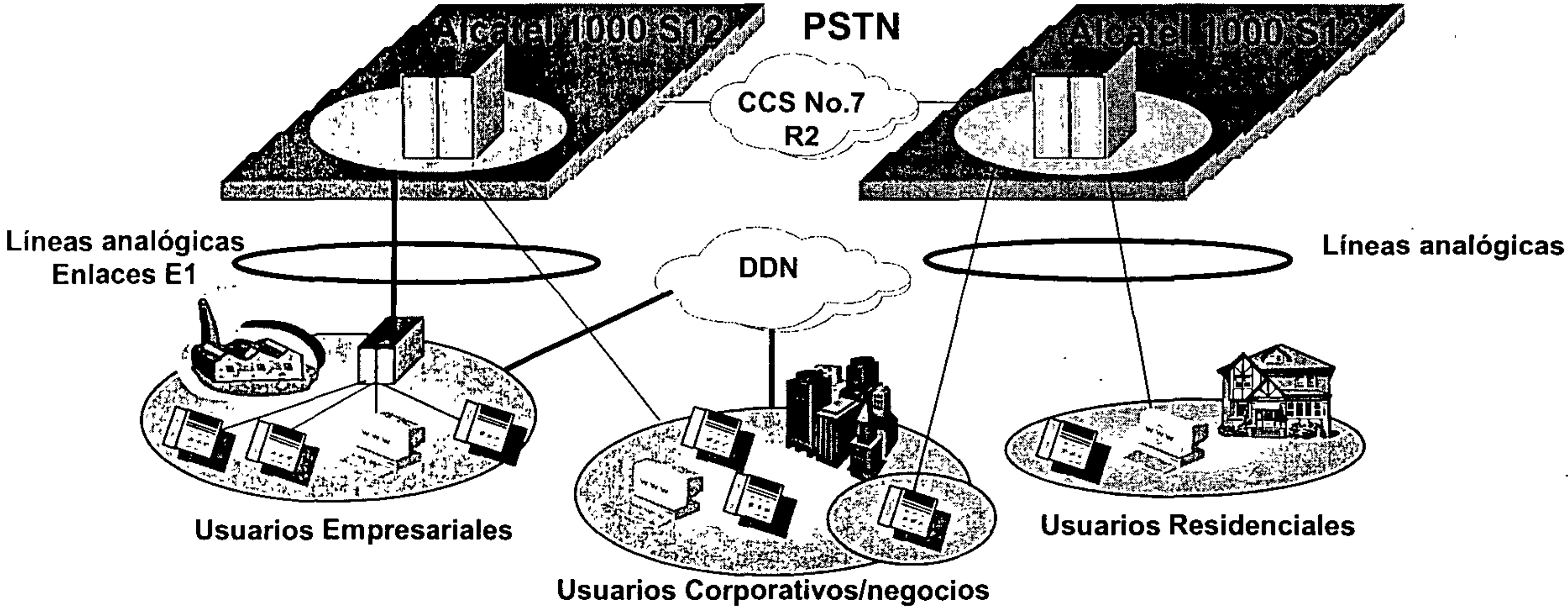
- Ingresos en 1995 para las 7 RBOC's y GTE solamente
- Estos son ingresos por suscripción solamente. Los pagos por activación y los ingresos relativos al CPE son adicionales
- La falta de servicio medido local quita importancia a las oportunidades de terminación de llamada

Modelos de Red

Modelos de Red (1)
Elementos Clave de Tecnología

Servidores	Oficina Central	Acceso	Conectividad de Usuario
Correo de Voz y Correo de Fax	Fábrica de Banda Angosta	Ganancia de Par	CPE CLI y CLASS
SCP/SMP y SCE de Red Inteligente	FacilidadesC LASS	Inalámbrico	Teléfono de Pantalla ADSI
SV ADSI	Puente de Conferencia	ISDN	Pagers
Gateway a Internet	Facilidades ISDN	Fibra SDH/PDH	PCs, Modems y Software
Atención y Facturación al Usuario	Inter-working ISDN		Puentes y Ruteadores

Modelo de Red
Servicios y Ventajas Actuales



▼ Servicios:
- Conmutación de Voz o de Datos (vía modem)
- Acceso a Internet a 56 Kbps (vía modem)
- Identificación de llamada, Espera, Tripartita
- Transferencia, Despertador
- Servicios de datos utilizando enlaces dedicados
- Redes privadas virtuales de datos (VPN)

▼ Ventajas de la Red
- Red 100% digitalizada
- Servicios conmutados y dedicados
- Servicios de circuito y de paquete
- Manejo de Señalización No.7, SDH
- Planta de cobre instalada
- Gestión centralizada

Tendencia General

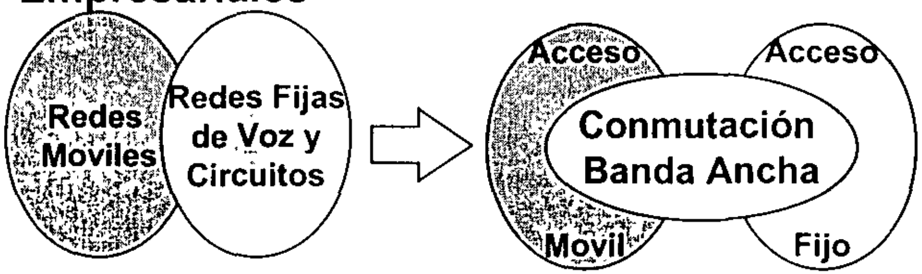
¿Que está sucediendo en el mercado?

- Fuerte crecimiento del tráfico de datos, lo que significa que las arquitecturas de red serán más manejadas por el tráfico de datos.

  - Backbone común para voz y datos
  - La funcionalidad de voz en redes de datos soportada por Call servers
  - Parte del tráfico de voz se tramitará en forma de paquete.

Estatus actual de las redes

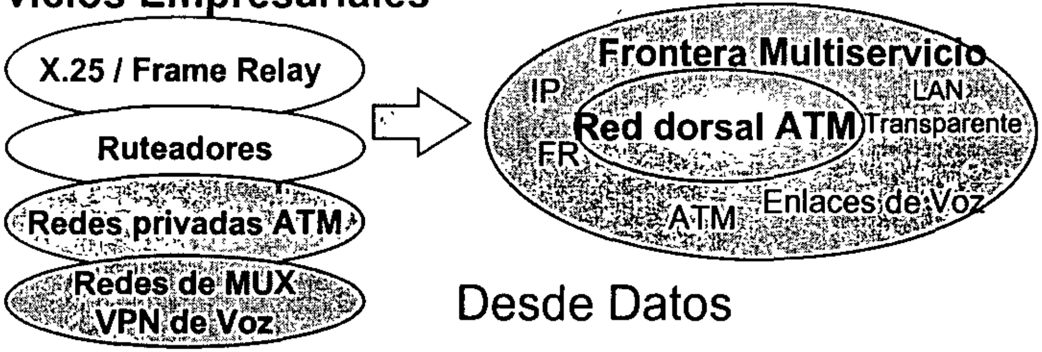
Servicios Residenciales y Empresariales



Desde Voz

- A partir de la RTPC
- Aprovechar capacidad de migración y planta instalada
- Enfasis en integración, presencia y flexibilidad

Servicios Empresariales



Desde Datos

- A partir de Datos
- Enfasis en usuarios de datos
- Esfuerzo en presencia y flexibilidad
- Esfuerzo en integración de Voz y ciruitos dedicados

La importancia de la Visión Estratégica de la Red y la Demanda del Mercado

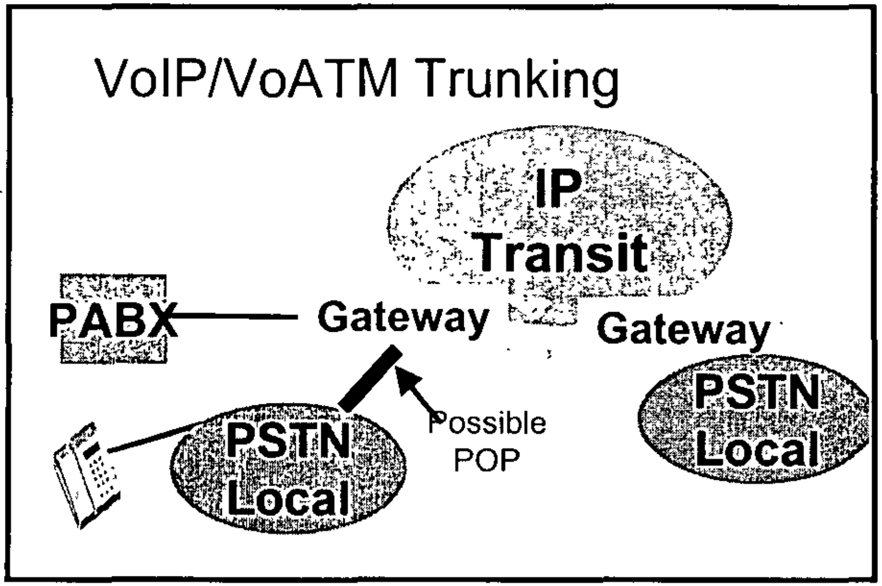
ALCATEL

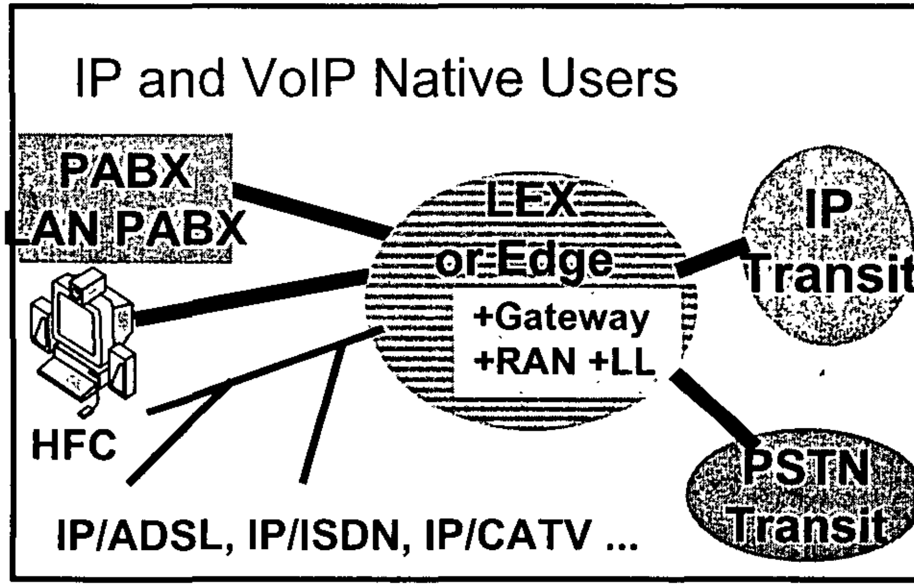
Un backbone común

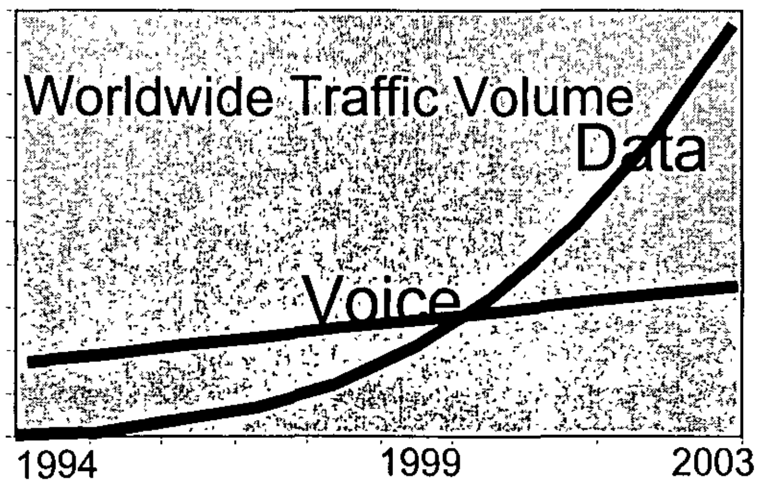
- Los operadores están considerando poner tráfico de voz sobre un backbone de datos (voz sobre ATM o voz sobre IP), en vez de invertir en una renovación de los switches de tráfico de voz.

- Los operadores están considerando utilizar la infraestructura central de otro operador.

Manejadores de mercado con mayores cambios







¿Que está cambiando? El transporte

- El medio de transporte puede tomar diferentes formas dependiendo de la red en particular y el tipo de tráfico dominante.

- El tráfico IP será transportado sobre WDM, SDH/Sonet, ATM, MPLS,...

ALCATEL

¿Que está sucediendo? El backbone

- Algunos operadores mantendrán separados los backbones de voz y datos por un tiempo considerable. Nuevos switches de tránsito con interfaces SDH son mucho más densos y de menor costo.

- Otros operadores transportarán voz sobre redes de datos e introducirán interfaces para voz sobre ATM o voz sobre IP.

ALCATEL

¿Que está sucediendo? Aplicaciones

- Los servicios de voz son en este momento mucho más ricos en facilidades que los servicios de datos. Este también es el caso para el PABX comparado con la LAN.

- Los servicios de voz son aún la principal fuente de ingresos para los operadores, quienes no tienen interés en disminuir sus ingresos por voz moviéndose en la dirección de sólo transporte de bits.

ALCATEL

¿Qué está sucediendo? Aplicaciones

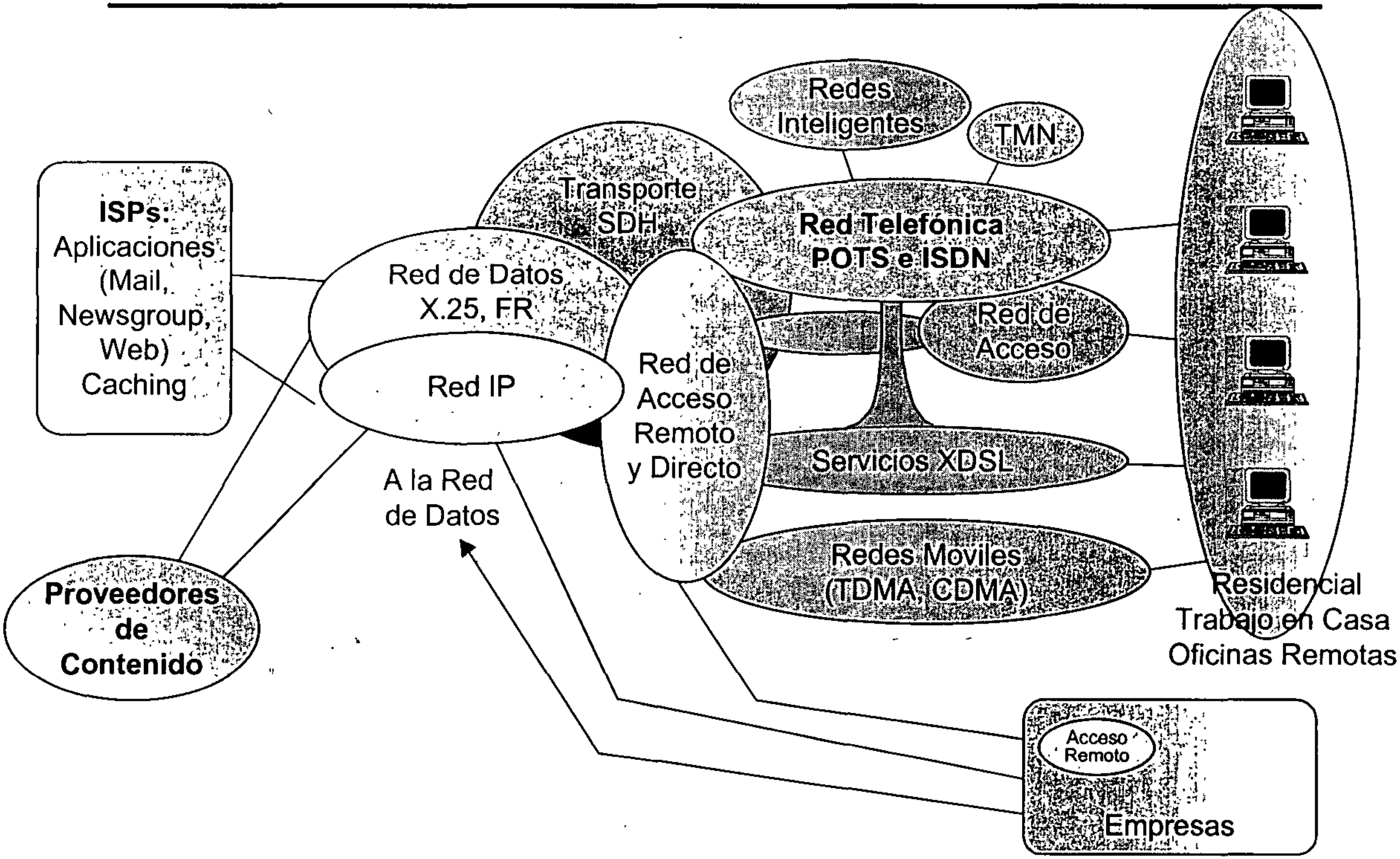
- Los servicios de voz requieren manejo de llamada y supervisión en tiempo real. Lo mismo es requerido para interactividad multimedia de persona a persona.

- Los operadores requieren de rápida introducción de nuevos servicios, con el fin de competir en funcionalidad más que en precio. Esto se logra a través de la red Inteligente y a través del creciente uso de servidores de llamada.

- El cambio a un ambiente multimedia requiere la misma riqueza en facilidades que los servicios de datos.

ALCATEL

Apreciación global de la arquitectura de la red e IP



ALCATEL

¿Que está sucediendo? AO&M

- La mayoría de los operadores tienen al día de hoy una multitud de redes de datos y dentro de ellas diferentes elementos de red, todos ellos con diferentes sistemas de administración. Los costos de operación son enormes.

- La calidad del servicio y la integración de redes están llegando a ser cada vez más importantes.

ALCATEL

Visión de Evolución Bellcore

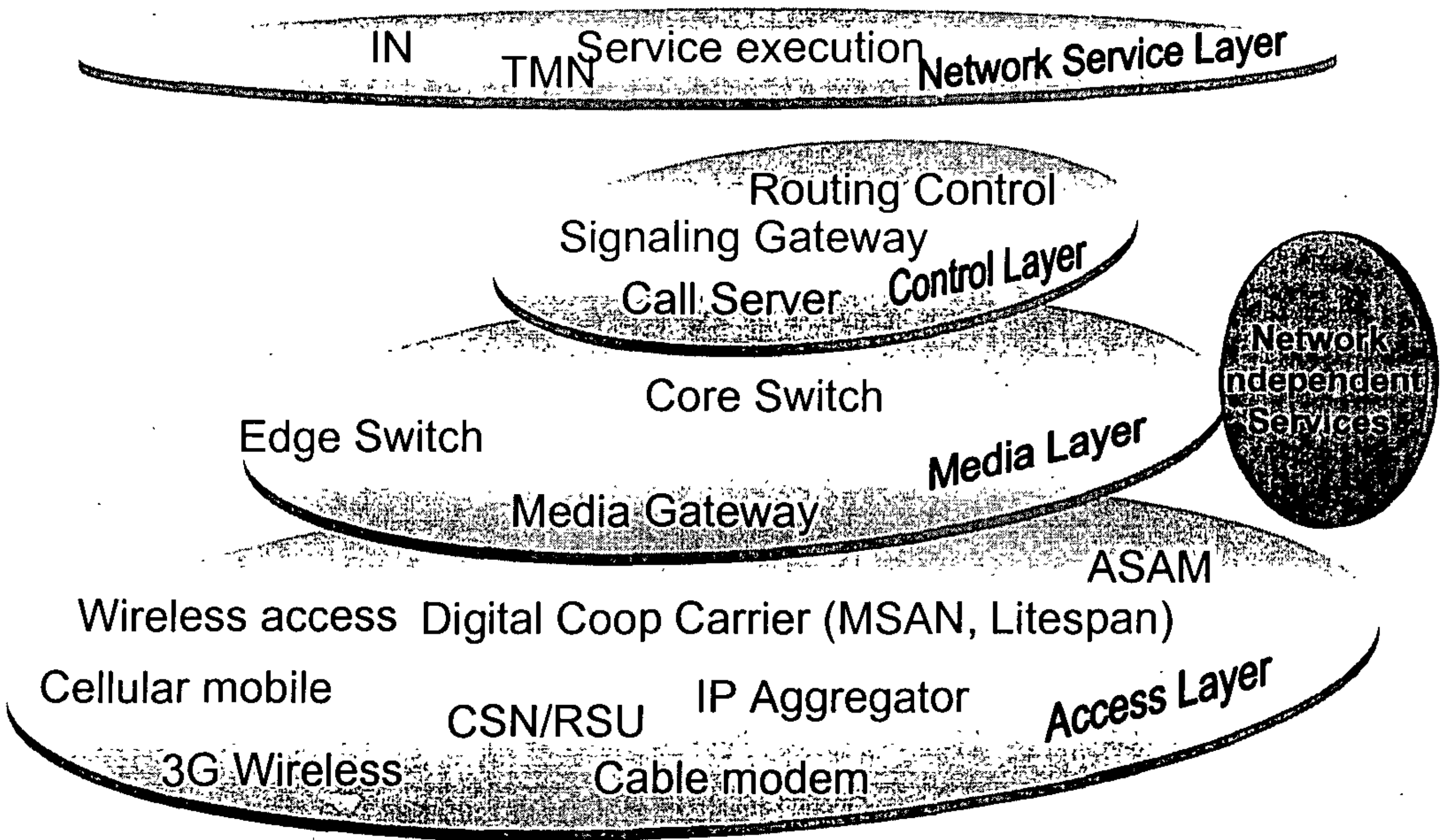
ALCATEL

¿Que está sucediendo?

- Las compañías de redes IP (Cisco,...) y consultoras (Bellcore,...) están promoviendo nuevas arquitecturas de red.
- El mensaje es que éstas arquitecturas resultarán en menores costos y aumento en la flexibilidad.
- Los operadores están disminuyendo la inversión en extensión o reemplazo de switches de voz.

ALCATEL

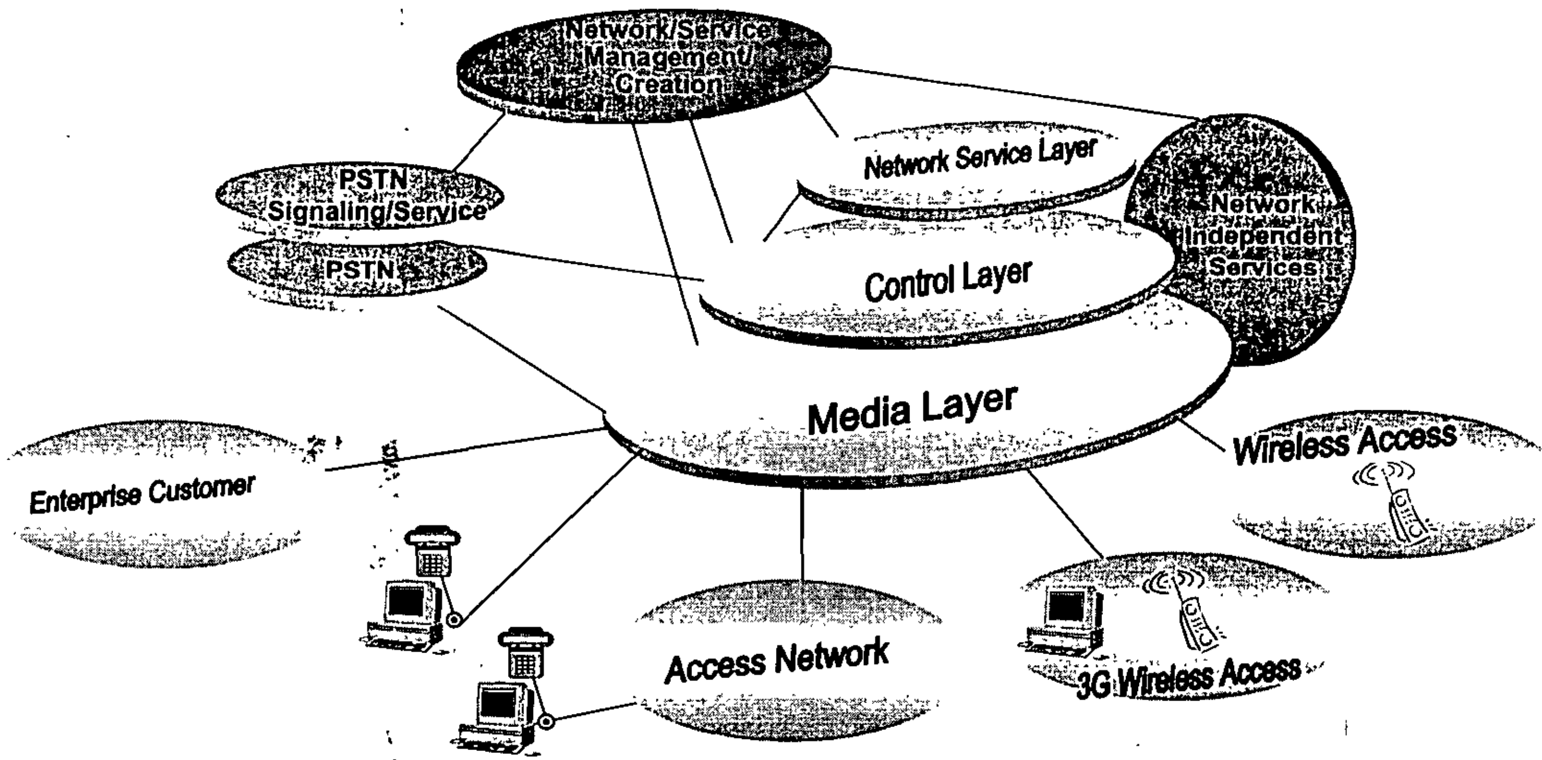
Visión de Bellcore: capas funcionales en la evolución de la Red



ALCATEL

Modelo de Evolución Alcatel

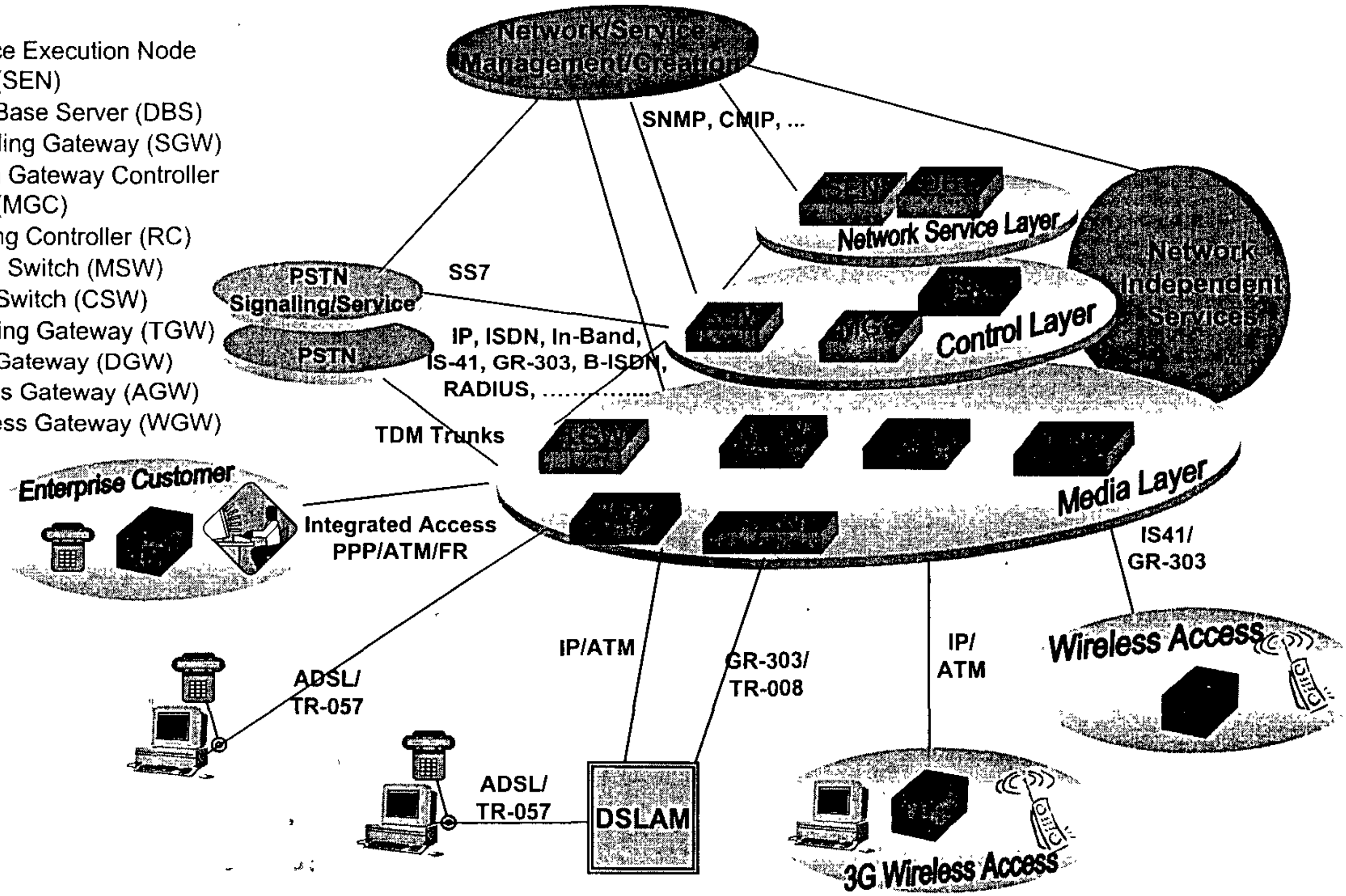
El modelo de Red del Futuro



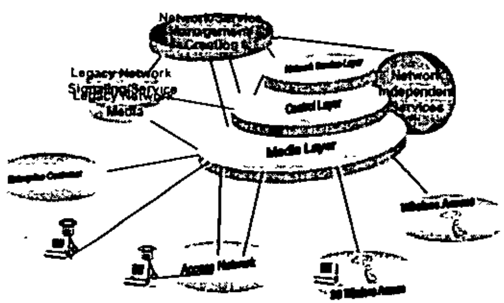
- Principios
  - Aislar control, servicios, transporte
  - Interoperabilidad con la red telefónica existente
  - Múltiples tecnologías de datos (ATM and IP)
  - Basado en estándares

El modelo de Red del Futuro
Componentes

- Service Execution Node (SEN)
- Data Base Server (DBS)
- Signaling Gateway (SGW)
- Media Gateway Controller (MGC)
- Routing Controller (RC)
- Media Switch (MSW)
- Core Switch (CSW)
- Trunking Gateway (TGW)
- Data Gateway (DGW)
- Access Gateway (AGW)
- Wireless Gateway (WGW)



Vista del Modelo de red del Futuro de Alcatel



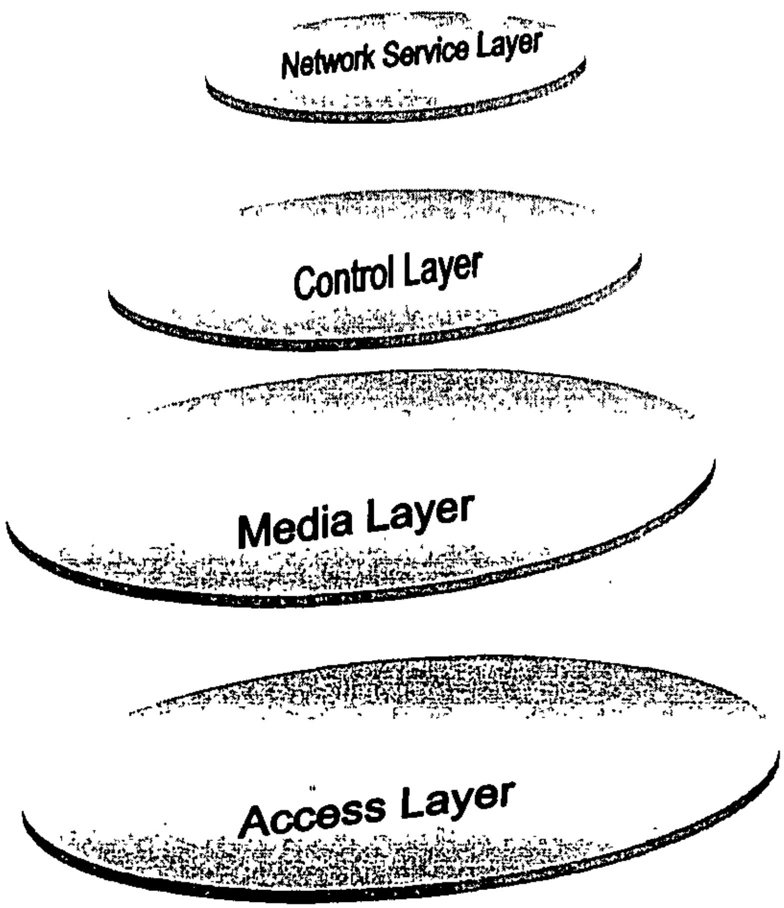
- Alcatel soporta totalmente este modelo.
- La pregunta importante es:
  - ¿Cómo puede éste modelo ser implementado en realidad?

¿Significativa reutilización de lo ya instalado y servicios y productos disponibles?

¿Iniciando desde "scratch"?

- Alcatel puede soportar todos los niveles de este modelo, basándose en:
  - Reutilización de productos y servicios existentes
  - Adición de nuevas tecnologías y servicios
  - Manteniendo la calidad de clase de carrier

Soluciones Alcatel para los Diferentes Niveles de red



Alcatel 1400 - intelligent network
Alcatel 145x - service creation environment
Alcatel 13xx - TMN
Alcatel 1135 SMC - service management center

Alcatel 1000 CS - call server

Alcatel 1000 S12 - multi-service switch
Alcatel 1000 BBX - core router
Alcatel 1100 LSS - switch routers (Xylan)
Alcatel Powerrail - routing switch (Packet Engines)
Alcatel 1131 RAN/DANA

Alcatel 1540 MSAN/GA
Alcatel 1570 BB - hybrid fiber coax
Alcatel 9800 - fixed wireless access
Alcatel 9700 - satellite access

... and many more

Criterios para la Evolución a la Red del Futuro

- Servicios sin Unión (Seamless)
  - Independientes del medio de transporte (cobre, fibra, radio, ...)
  - Independientes de la localización (en casa, en el trabajo, en el camino)
    - ➡ El mismo portafolio de servicios de voz
    - ➡ Innovativa convergencia de servicios

Ejemplo de una consecuencia:

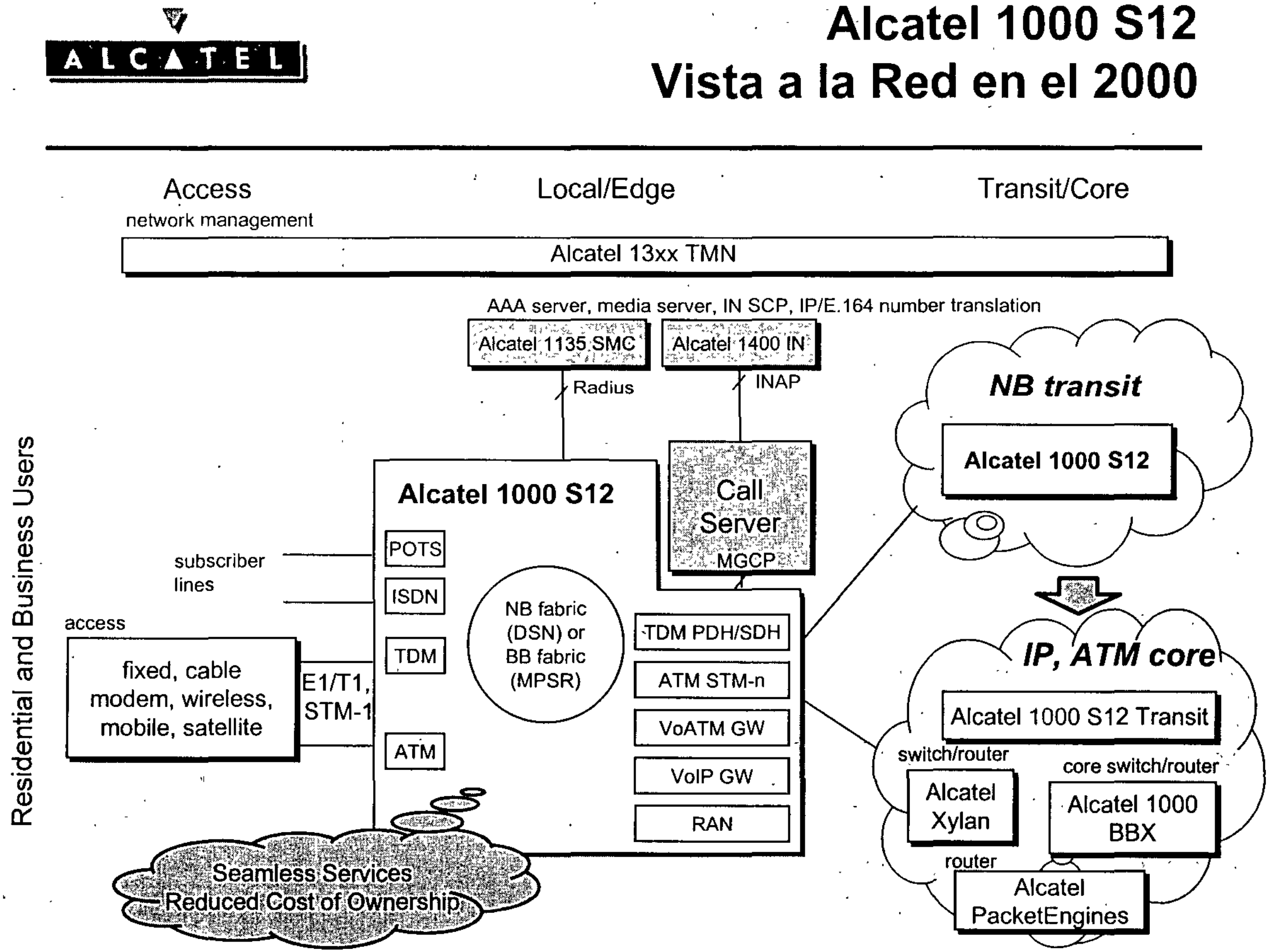
Los servidores de voz para redes IP requerirán similares funciones/servicios como los implementados en nuestros días en los sistemas de conmutación de voz.

- Los usuarios de datos requerirán servicios similares a los conocidos ahora en las redes de voz (p.ej., servicios de red inteligente)

Criterios para la Evolución a la Red del Futuro

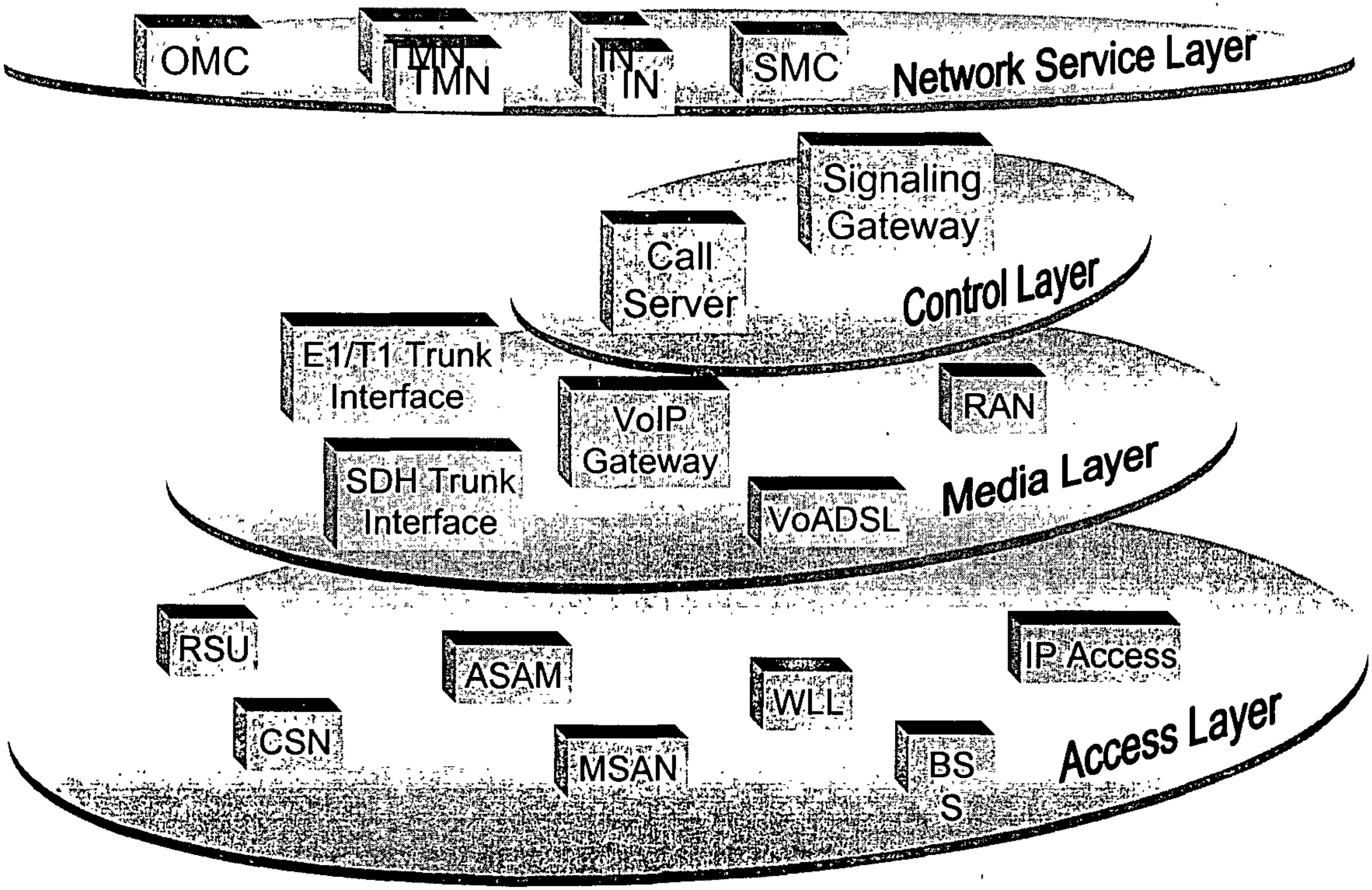
- Bajo costo de mantenimiento (COO):
  - Funciones integradas (en particular: gateway)
  - Interfaces estándar
  - Red general y administración de servicio

Alcatel 1000 S12
Vista a la Red en el 2000

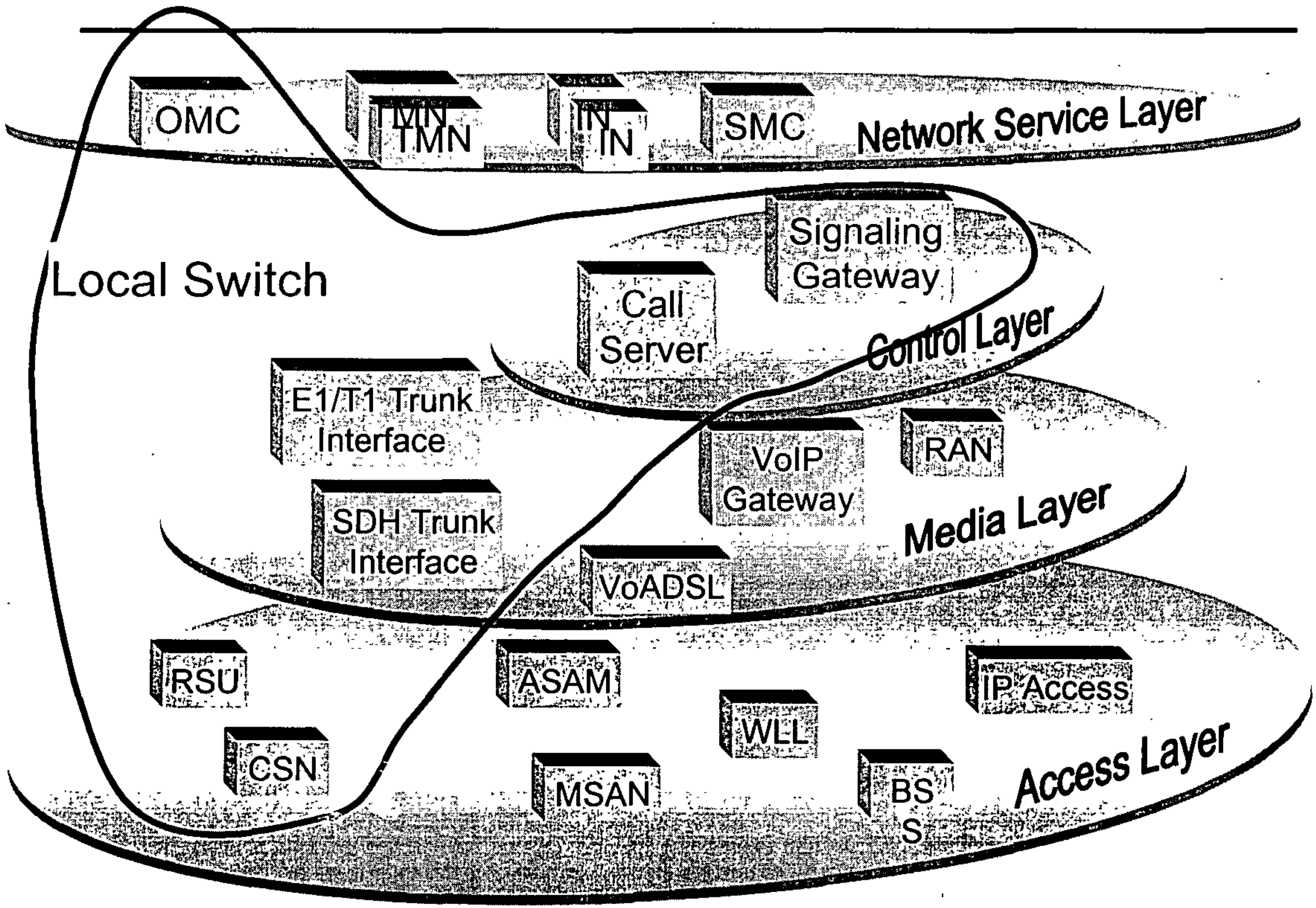


ALCATEL

La Red Explotada

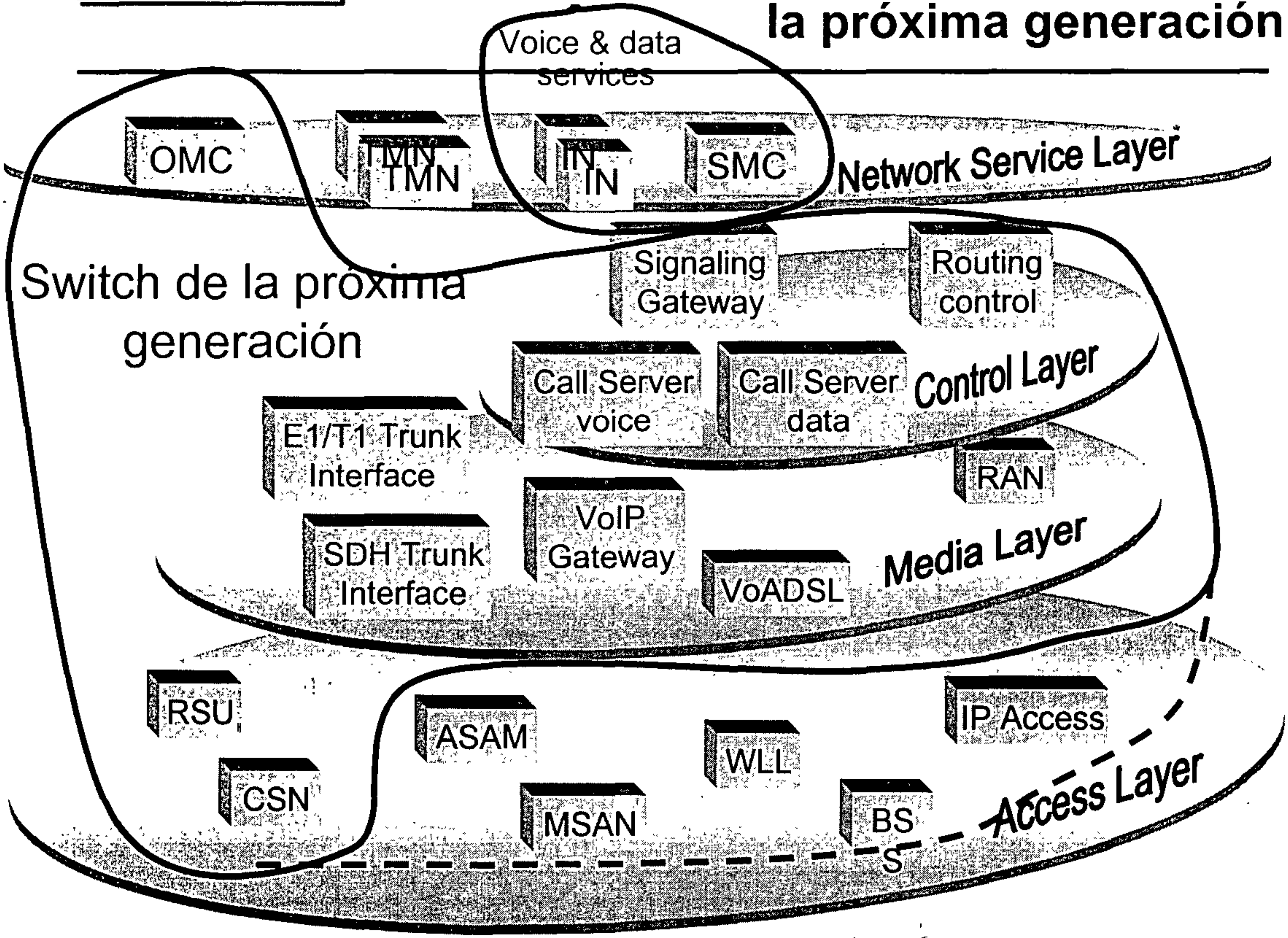


Red de Conmutación de Hoy



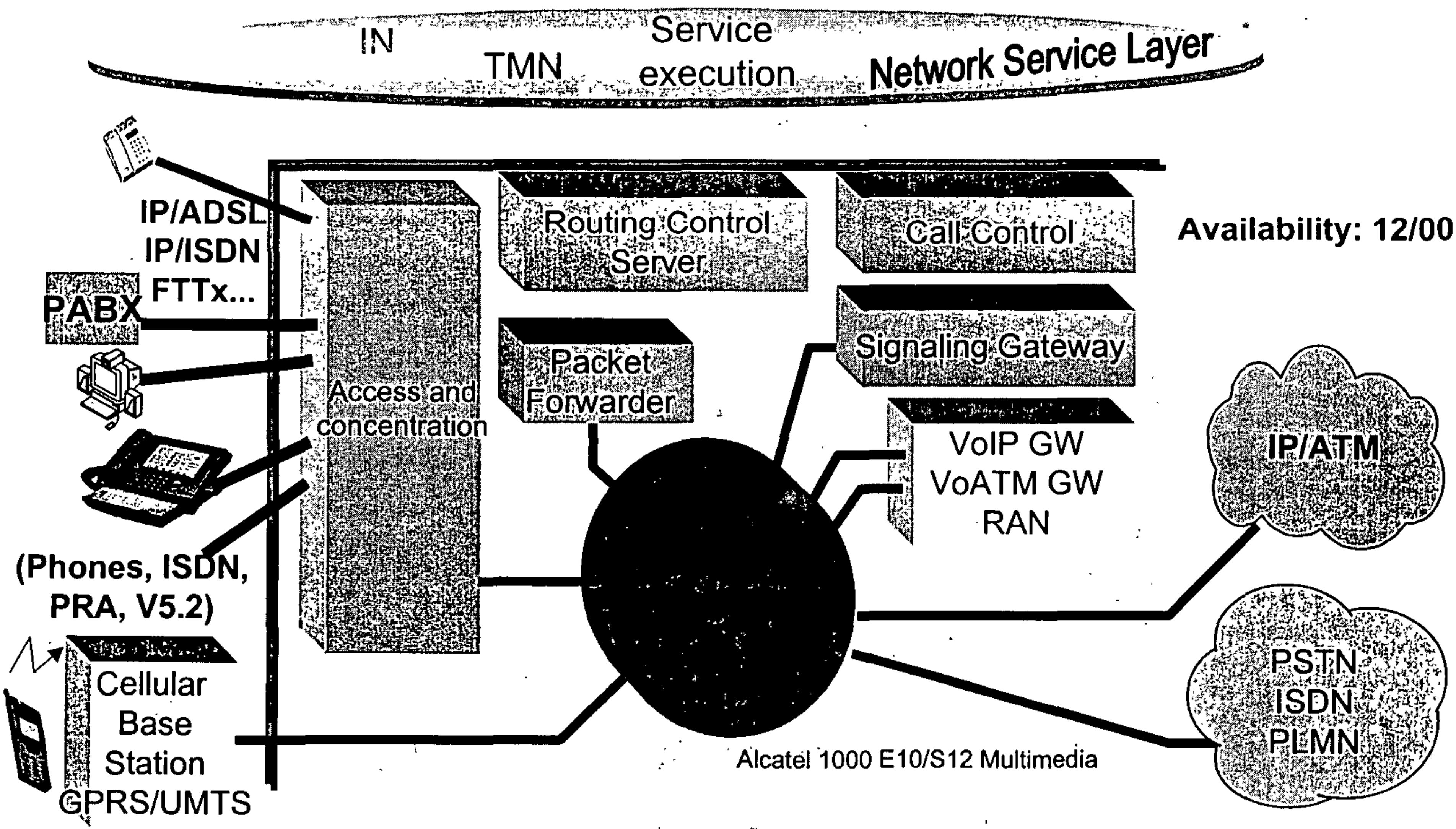
ALCATEL

Integración alrededor del Switch de la próxima generación



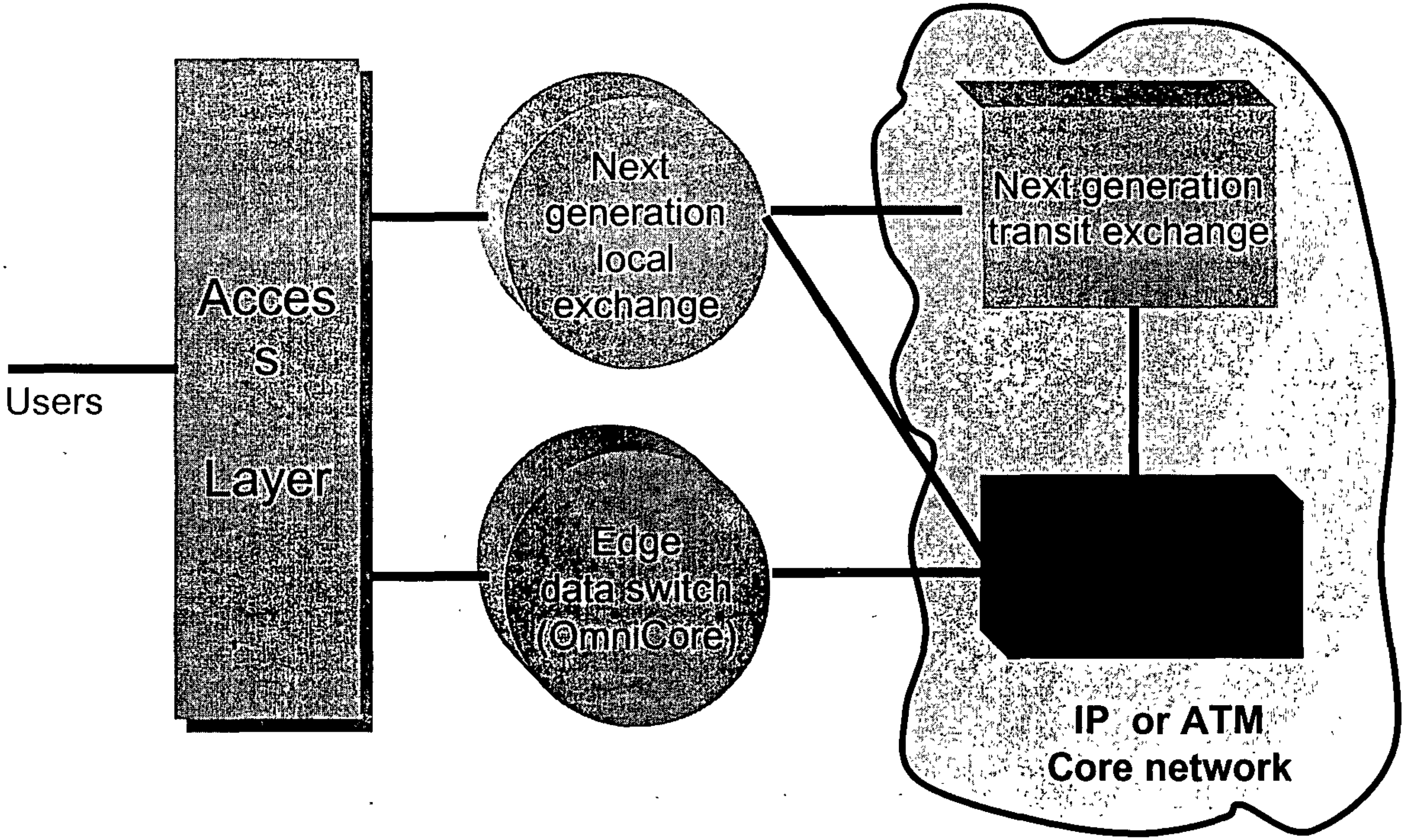
ALCATEL

Alcatel 1000 S12 Multimedia

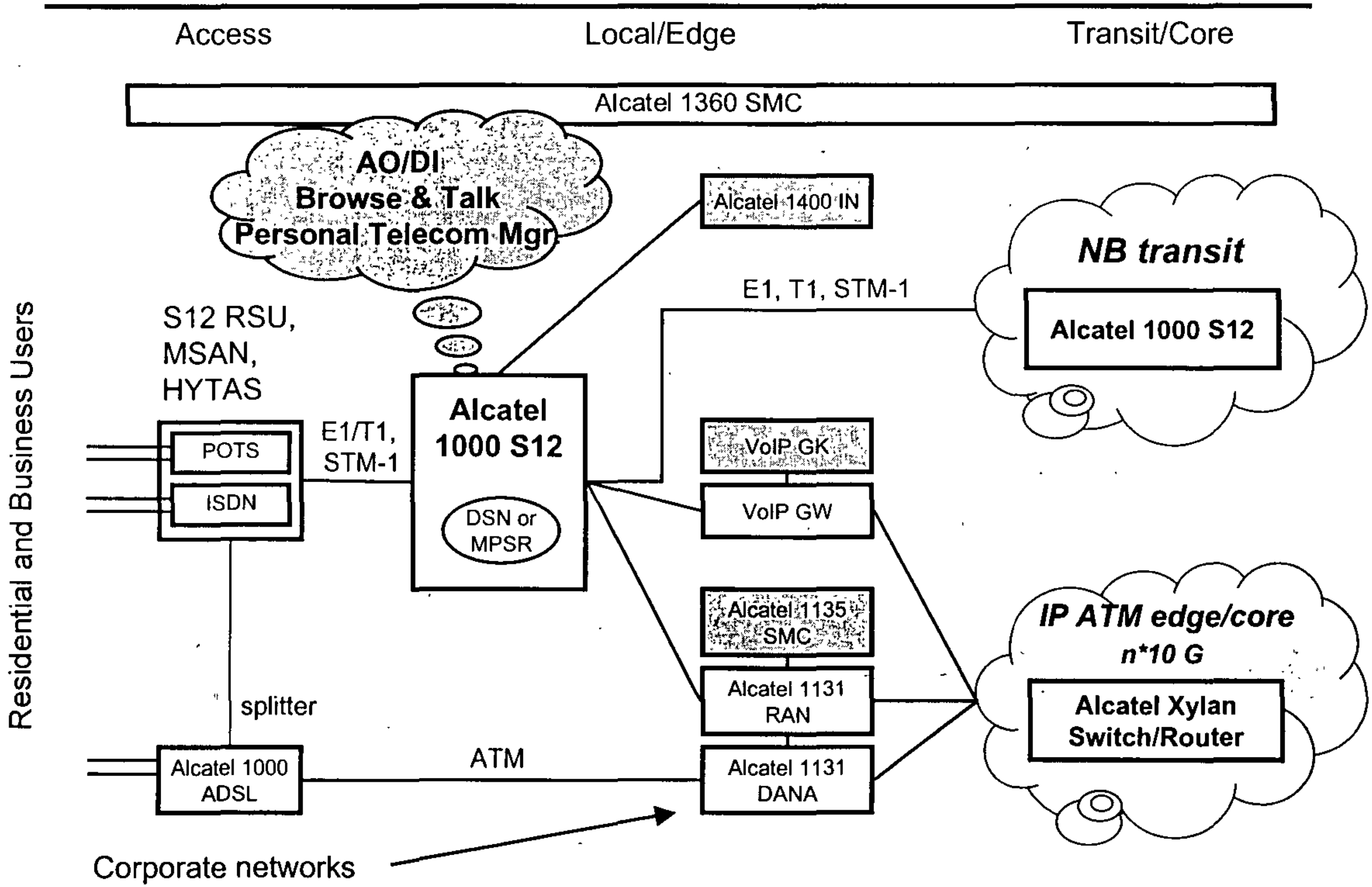


ALCATEL

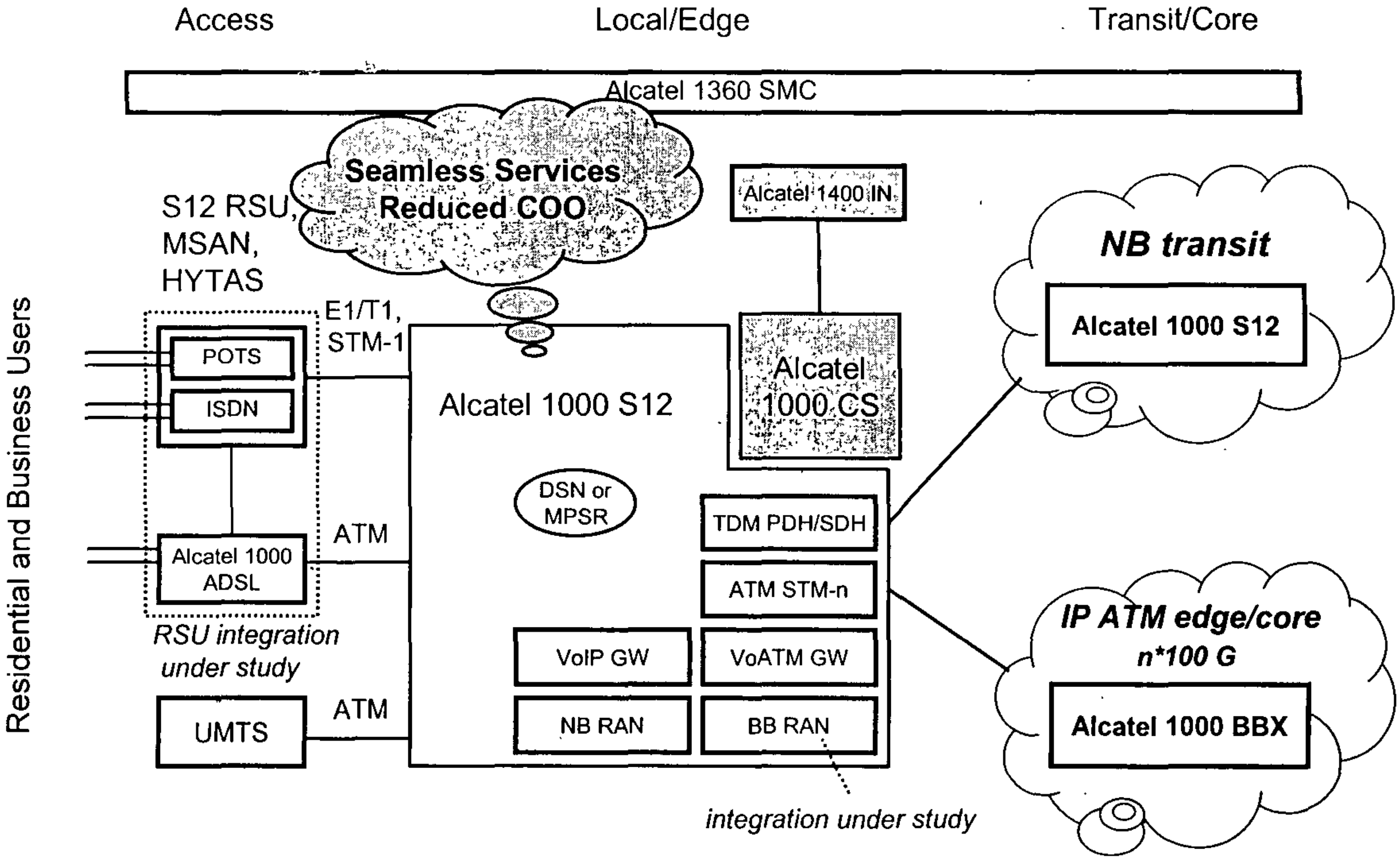
Red de Datos y Backbone



Vista de la Red 1999 para Usuarios Alcatel 1000 S12



Vista de la Red 2000 para Alcatel 1000 S12



Conclusiones

SERVICIOS

RESIDENCIALES	COMERCIALES
VIDEOTELEFONO TELEFONO DE ALCANCIA CON TARJETA DE CREDITO/DEBITO NUMERO UNIVERSAL NUMERO PERSONAL COBRO REVERTIDO CORREO ELECTRONICO	VIDEOTEX VIDEOCONFERENCIA DATOS FACSIMIL CORREO DE VOZ LOCALIZACION VEHICULAR

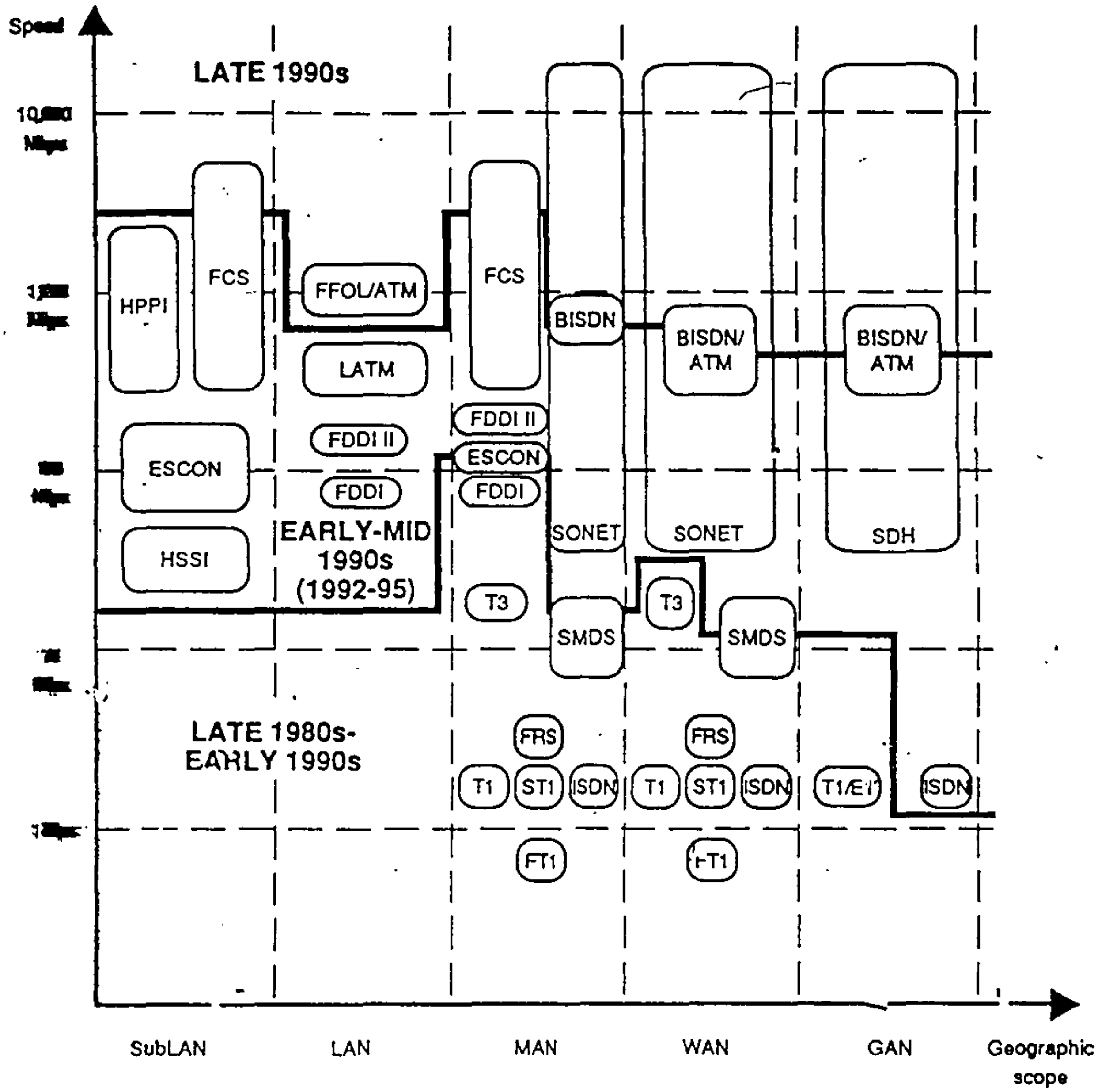
EQUIPOS Y SISTEMAS

DACS, CONMUTADORES DE CIRCUITOS, CONMUTADORES DE PAQUETES, NODOS INTELIGENTES, NODOS ESPECIALIZADOS DE TELECOMUNICACIONES, REDES INTELIGENTES, RED SUPERPUESTA, RED SATELITAL, PUNTOS DE ACCESO VIDEOTEX, PUNTOS DE CONTROL DE SERVICIOS, ADMINISTRADORES DE RED, SEÑALIZACION POR CANAL COMUN

TECNOLOGIAS CLAVE

MICROELECTRONICA	OPTOELECTRONICA	PROC. DE INFORMACION	TRANSMISION RADIADA
COMPONENTES EN Si, GaAs TECNOLOGIA BICMOS TECNOLOGIAS DE MONTAJE SMT CIRCUITOS ASICS, PLDS MMICS	CONMUTACION OPTICA AMPLIFICADORES OPTICOS FOTODETECTORES Y FOTOEMISORES REGENERADORES MODULACION DIRECTA Y COHERENTE	PROCESAMTO. DIGITAL DE SEÑALES PROTOCOLOS ALMACENAMTO. DE INFORMACION COMPRESION/MULTIPLEXAJE DATOS CODIFICACION ARQUIT. DE PROCESADORES LENGUAJE DE COMPUTO SISTEMAS OPERATIVOS	RADIO DIGITAL SATELITE RADIO CELULAR OPTIMIZACION DE BANDAS DE FRECUENCIAS

INTERRELACION TECNOLOGICAS CLAVE-EQUIPOS Y SISTEMAS-SERVICIOS



ISDN: Integrated Services Digital Network
BISDN: Broadband ISDN
SONET: Synchronous Optical Network
FCS: Fiber Channel Standard
HPPI: High-Performance Parallel Interface
FDDI: Fiber Distributed Data Interface
SDH: Synchronous Digital Hierarchy
LATM: Local ATM
FT1: Fractional T1
ST1: Switched T1
ESCON: Enterprise Systems Connection
SMDS: Switched Multi-Megabit Data Service
FFOL: FDDI Follow-On LAN
FRS: Frame Relay Service
HSSI: High-Speed Serial Interface

Fig Broadband LAN/MAN digital technologies of the 1990s.

SERVICIOS PORTADORES

1) MODO CIRCUITO

- 64 kbps no restringido
- 64 kbps para transmisión de voz
- 64 kbps para audio a 3.1 KHz
- Uso alternado para voz / 64 kbps no restringido
- 384 kbps no restringido
- 1536 kbps no restringido
- 1920 kbps no restringido

2) MODO PAQUETE

- Llamada virtual y Circuito virtual permanente
- Sin conexión
- Señalización de usuario

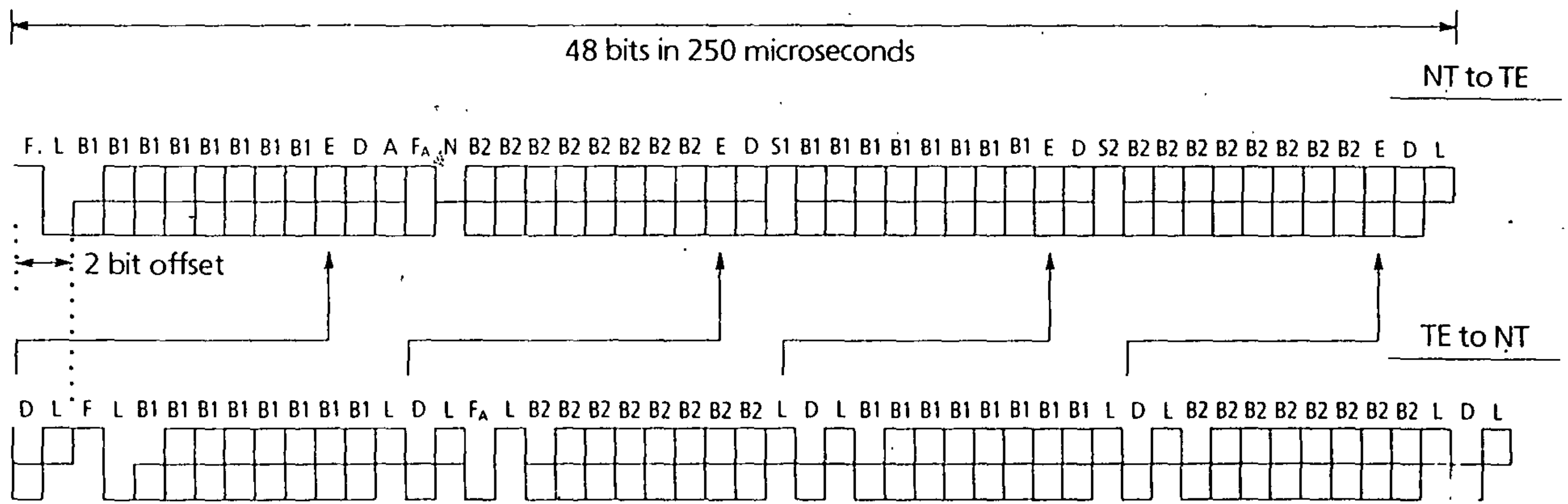
SERVICIOS SUPLEMENTARIOS

- Marcación Directa Entrante (DDI)
- Presentación del número del usuario que llama (CLIP)
- No presentación del número del usuario que llama (CLIR)
- Presentación del número del usuario llamado (COLP)
- No presentación del número del usuario llamado (COLR)
- Identificación de llamadas maliciosas (MCI)
- Subdireccionamiento (SUB)
- Transferencia de llamadas (CT)
- Desvio de llamada en caso de número ocupado (CFB)
- Desvio de llamada en caso de no contesta (CFNR)
- Desvio incondicional de llamadas (CFU)
- Desvio de llamadas (CD)
- Busqueda de línea (LH)
- Llamada en espera (CW)
- Retención de llamada (HOLD)
- Terminación de llamadas a números ocupados (CCBS)

SERVICIOS SUPLEMENTARIOS (CONT)

- Conferencia (CONF)
- Conferencia tripartita (3. TY)
- Grupo Cerrado de Usuarios (CUG)
- Plan de numeración privado (PNP)
- Llamada con tarjeta de crédito (cred)
- Envio de información de tarificación (AOC)
- Llamadas por cobrar (REV)
- Señalización Usuario a Usuario (UUS)

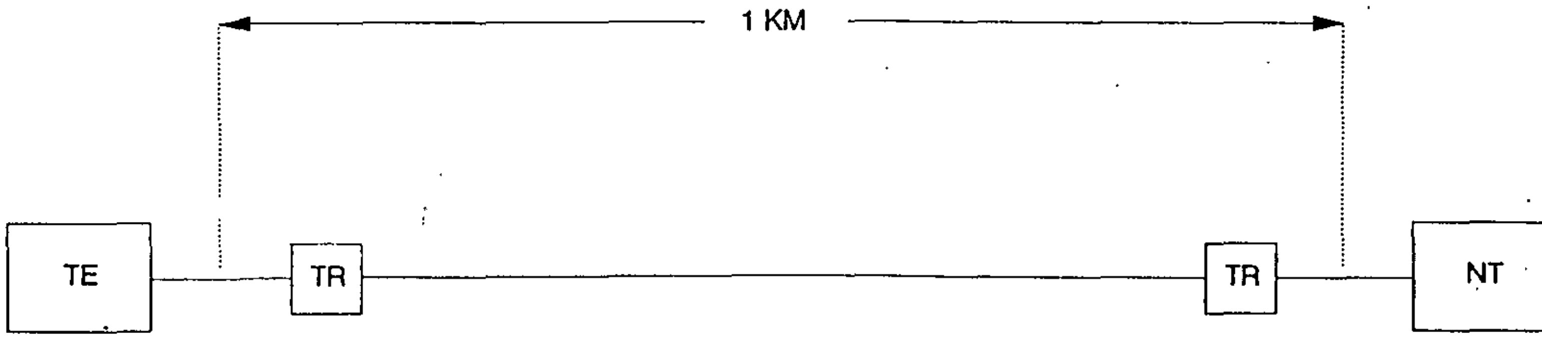
Layer 1 Frame Structure



F = Framing bit	F_A = Auxiliary bit (=0)	B2 = Bit within B channel 2
L = DC balancing bit	N = Bit set to binary value $= \bar{F}_A$	A = Bit used for activation
D = D channel bit	B1 = Bit within B channel 1	S1, S2 = Reserved for future standardisation
E = D echo-channel bit		

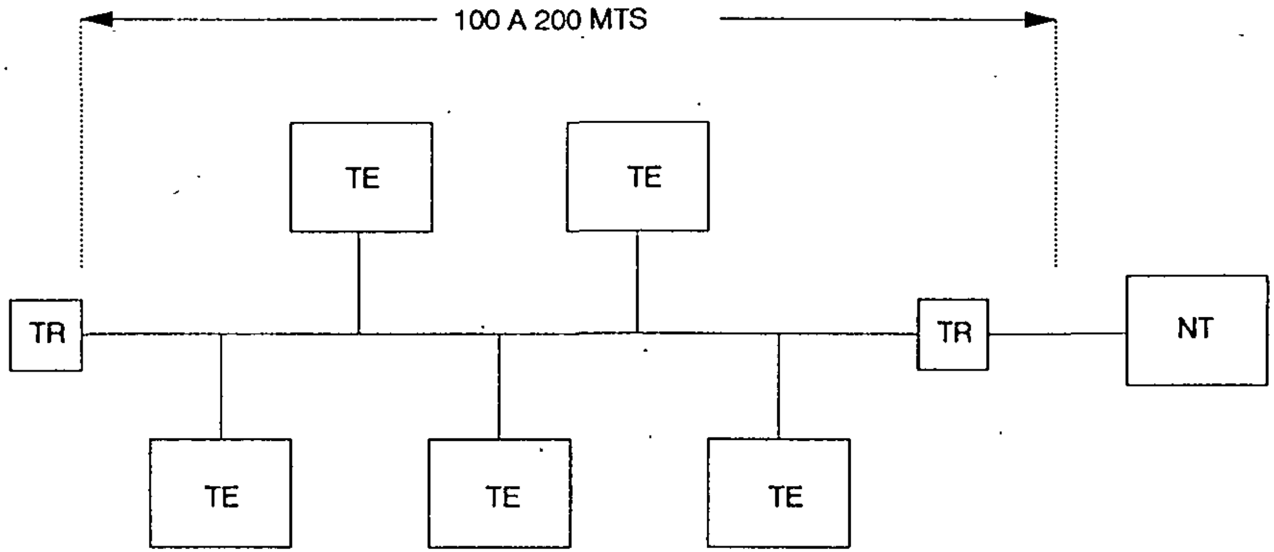
Fig. 5.3 Frame structure at the basic access.

CONFIGURACIÓN PUNTO A PUNTO



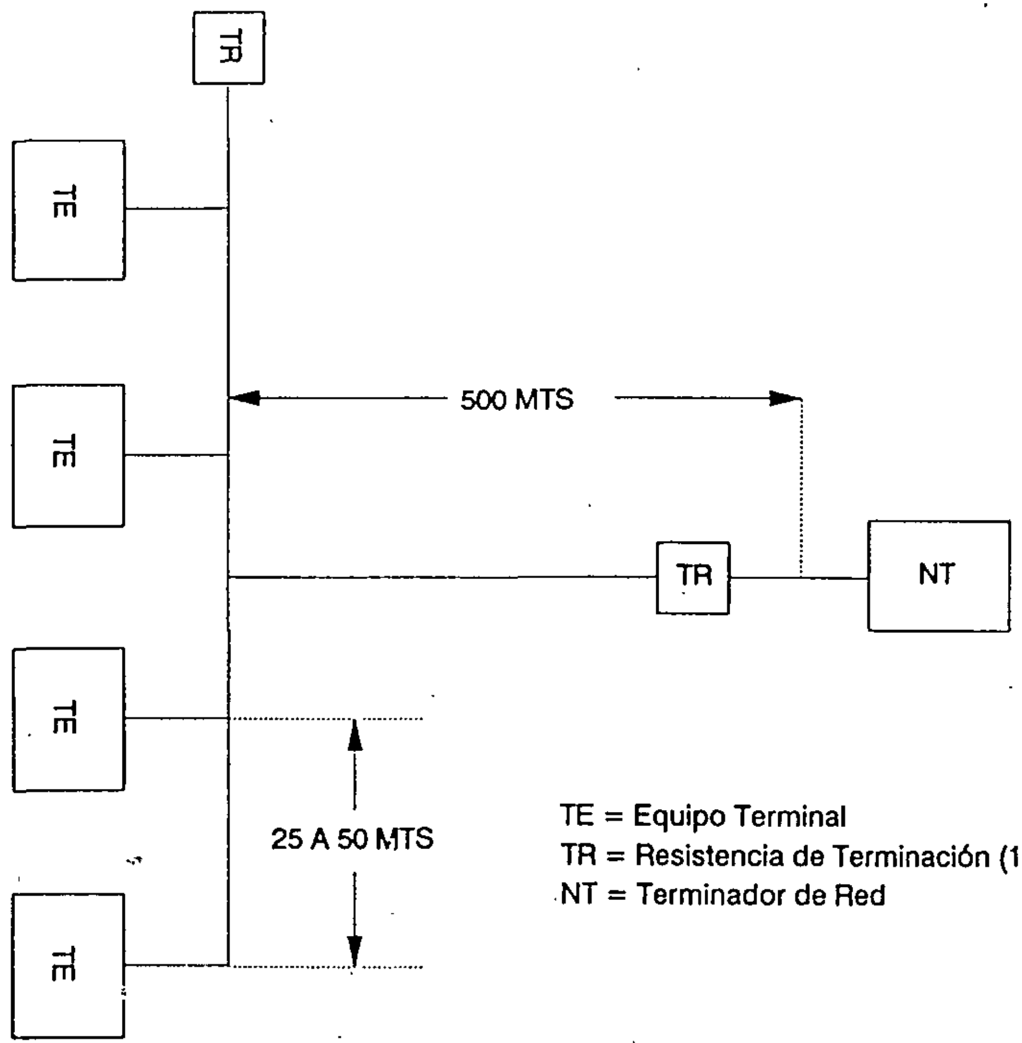
TE = Equipo Terminal
TR = Resistencia de Terminación (100 Ohms + 5%)
NT = Terminador de Red

CONFIGURACIÓN BUS PASIVO CORTO

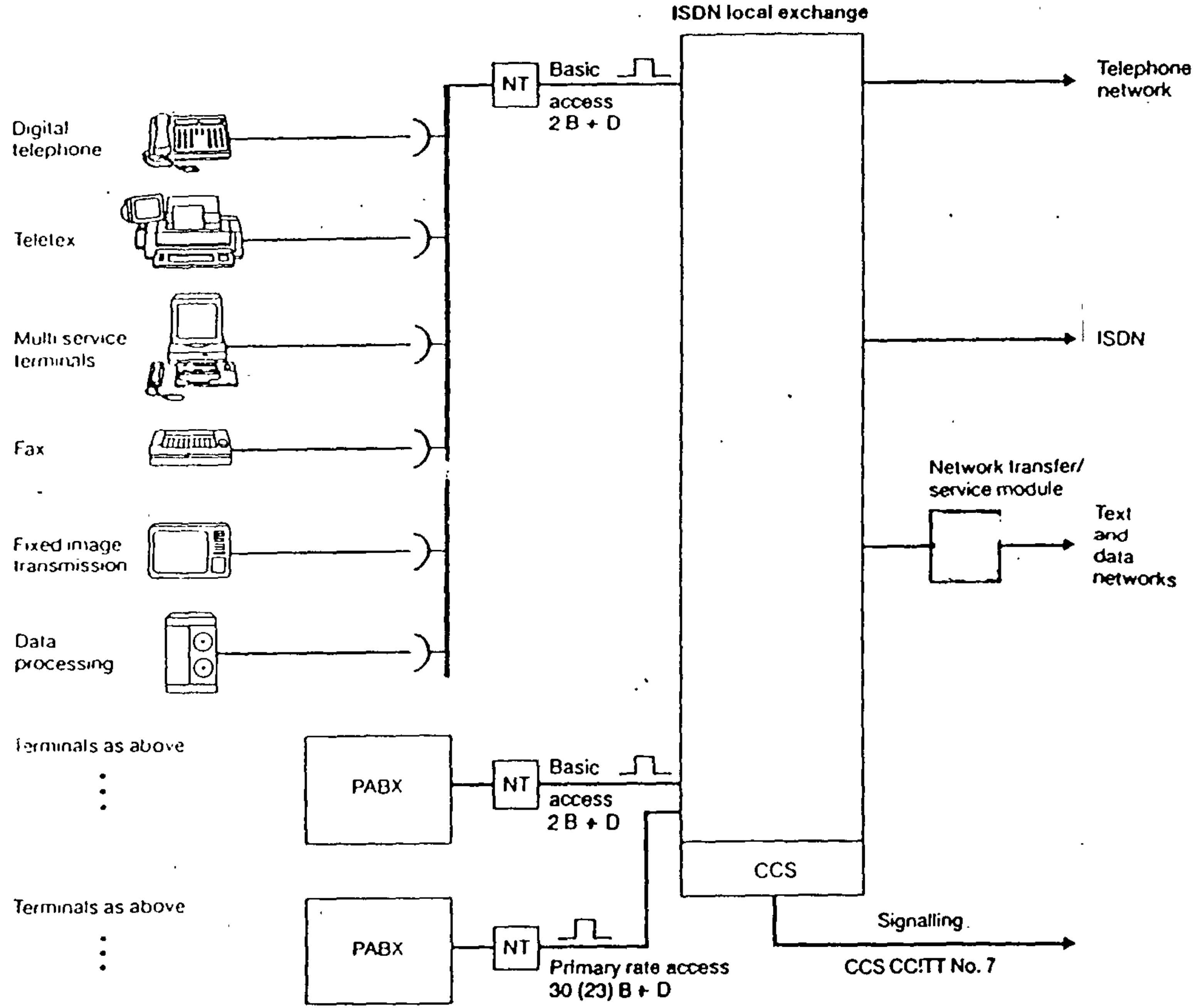


TE = Equipo Terminal
TR = Resistencia de Terminación (100 Ohms + 5%)
NT = Terminador de Red

CONFIGURACIÓN BUS PASIVO EXTENDIDO



TE = Equipo Terminal
TR = Resistencia de Terminación (100 Ohms + 5%)
NT = Terminador de Red



ISDN local exchange
NT
Basic
access
2 B + D
Digital
telephone
Teletex
Multi service
terminals
Fax
Fixed image
transmission
Data
processing
Terminals as above
PABX
NT
Basic
access
2 B + D
Terminals as above
PABX
NT
Primary rate access
30 (23) B + D
CCS
Telephone
network
ISDN
Network transfer/
service module
Text
and
data
networks
Signalling
CCS CCITT No. 7

RDSI EN EUROPA: ACTUALIZACION

PAIS	PRUEBAS	COMERCIALIZACION	CENTRAL
AUSTRIA	OPERACION PILOTO DE ACCESO BASICO Y PRIMARIO PARA ESTE AÑO	A PRINCIPIOS DE 1990	EWSD; DMS MODIFICADO
BELGICA	LAS PRUEBAS SE INICIARON EN 1985-86, PRUEBA PILOTO DE ACCESO BASICO Y Y PRIMARIO PARA ESTE AÑO	1991 - 1992	S-12; EWSD
DINAMARCA	LA PRUEBA PILOTO SE INICIO EN 1987-88	A PRINCIPIO DE 1990; LAS TARIFAS YA HAN SIDO ANUNCIADAS	S-12; AXE
FINLANDIA	LAS PRUEBAS DE HELSINKI TELEPHONO CO. SE INICIARON EN 1986 ACCESO BASICO EN 87	COBERTURA DEL AREA METROPOLITANA PARA HELSINKI PARA MEDIADOS DE 1990 EL SERVICIO PILOTO PTT SE ESTA IMPLEMENTADO DE 1988-89 PARA SEGUIR CON SERVICIO COMERCIAL	DX-200;S-12;EWSD
HOLANDA	PRUEBA PILOTO DE ACCESO BASICO	LAS TARIFAS NO SE HAN DADO, PERO SE ESPERA QUE SEAN SIMILARES A LAS DE ALEMANIA	AXE; 5ESS/PRXD
NORUEGA	LA PRUEBA DE LA RED INTEGRADA EMPESO EN 1984; ACCESO PRIMARIO RDSI PUESTO A A PRUEBA A FINALES DE 1988	A FINALES DE 1989	S-12;AXE
ESPAÑA	PRUEBA PRE-RDSI EMPEZO EN 1980 Y SE OFRESIERON SERVICIOS CASI RDSI A USUARIOS COMERCIALES CON IBERCOM; PRUEBA DE ACCESO BASICO Y PRIMARIO	DESPUES DE 1989	AXE; 5EBB; S-12
SUECIA	LAS PRUEBAS DE CAMPO EMPEZARON EN 1986-1987; ACCESO BASICO Y PRIMARIO PILOTO EN 1988-89	1990	AXE
SUIZA	PRUEBA PILOTO DE ACCESO BASICO 1988-89	1991	EWSD; AXE
REINO UNIDO	PRUEBA PILOTO DE ACCESO DIGITAL INTEGRADO EN 1985; MULTILINEA IDA EN BLOQUES DE 30 X 64 KBPS DE 30x64 Kbps A FINALES DE OTOÑO; MERCURY LANZO EL SERVICIO DASS-2 BASADO EN CASI-RDSI A FINALES DEL AÑO PASADO	ACCESO BASICO EN 1989	SISTEMA X
FRANCIA	PRUEBA RENAN A MEDIADOS DE 1980	EL PRIMER SERVICIO COMERCIAL RDSI DE ACCESO BASICO A NIVEL MUNDIAL EMPEZO EN BRITANIA A FINALES DE 1987; EL SERVICIO FUE AMPLIADO AL AÑO SIGUIENTE; ACCESO ENTRE RDSI Y LA RED DE PAQUETES EN OPERACION TRANSPAC ACCESO PRIMARIO PLANEADO EN OTOÑO; DISPONIBILIDAD NACIONAL PROGRAMADA PARA FINALES DE 1990. LAS TARIFAS YA SE DIERON A CONOCER	E-10B; E-10MT; AXE ESTARA EN UN FUTURO
ALEMANIA	PRUEBA DE ACCESO BASICO DURANTA 1984-85; SERVICIO PILOTO EN 1987	SE PROGRAMARON 100 CENTRALES PARA MEJORAR UN SERVICIO EN RDSI EN 1989; COVERTURA NACIONAL PARA 1993; EL OBJETIVO A LARGO PLAZO ES INTRODUCIR CONMUTACION POR PAQUETES EN LA RDSI: LAS TARIFAS YA SE PAQUETES EN LA RDSI: LAS TARIFAS YA DIERON A CONOCER	EWSD; S-12
ITALIA	EMPEZANDO LOS SERVICIOS DE ACCESO BASICO	1990-91; LAS TARIFAS YA SE DIERON A CONOCER	PROTEO/GTD-5;

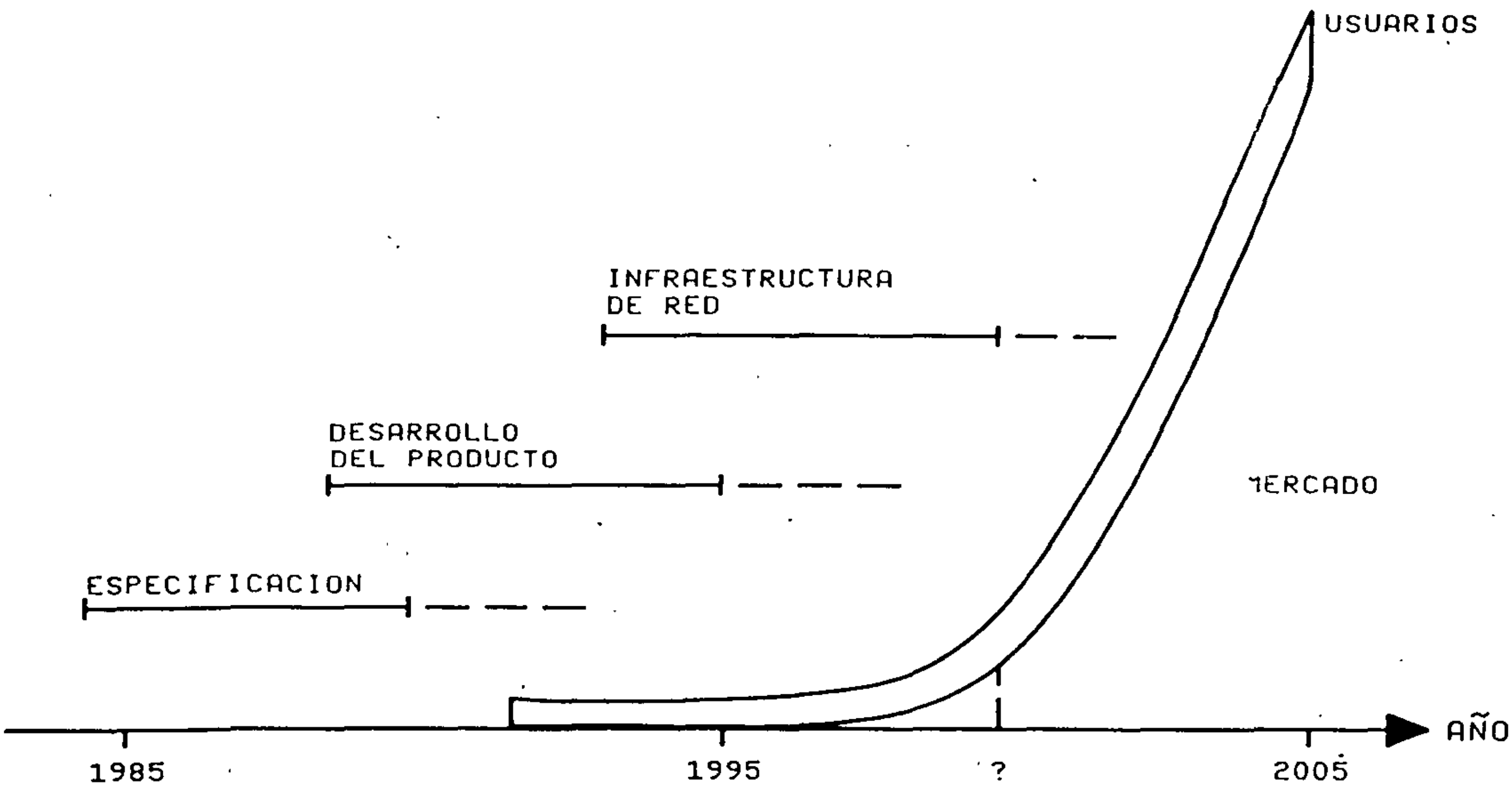


FIG. 4.4.- PROGRAMA DE INTRODUCCION DE RDSI-BANCH POR RACE (INVESTIGACION Y DESARROLLO EN TECNOLOGIAS AVANZADAS PARA COMUNICACIONES EN EUROPA).

AB9A35

RDSI EN EUROPA: ACTUALIZACION

PAIS	PRUEBAS	COMERCIALIZACION	CENTRAL
AUSTRIA	OPERACION PILOTO DE ACCESO BASICO Y PRIMARIO PARA ESTE AÑO	A PRINCIPIOS DE 1990	EWSD; DMS MODIFICADO
BELGICA	LAS PRUEBAS SE INICIARON EN 1985-86, PRUEBA PILOTO DE ACCESO BASICO Y Y PRIMARIO PARA ESTE AÑO	1991 - 1992	S-12; EWSD
DINAMARCA	LA PRUEBA PILOTO SE INICIO EN 1987-88	A PRINCIPIO DE 1990; LAS TARIFAS YA HAN SIDO ANUNCIADAS	S-12; AXE
FINLANDIA	LAS PRUEBAS DE HELSINKI TELEPHONO CO. SE INICIARON EN 1986 ACCESO BASICO EN 87	COBERTURA DEL AREA METROPOLITANA PARA HELSINKI PARA MEDIADOS DE 1990 EL SERVICIO PILOTO PTT SE ESTA IMPLEMENTADO DE 1988-89 PARA SEGUIR CON SERVICIO COMERCIAL	DX-200;S-12;EWSD
HOLANDA	PRUEBA PILOTO DE ACCESO BASICO	LAS TARIFAS NO SE HAN DADO, PERO SE ESPERA QUE SEAN SIMILARES A LAS DE ALEMANIA	AXE; 5ESS/PRXD
NORUEGA	LA PRUEBA DE LA RED INTEGRADA EMPESO EN 1984; ACCESO PRIMARIO RDSI PUESTO A A PRUEBA A FINALES DE 1988	A FINALES DE 1989	S-12;AXE
ESPAÑA	PRUEBA PRE-RDSI EMPEZO EN 1980 Y SE OFRESIERON SERVICIOS CASI RDSI A USUARIOS COMERCIALES CON IBERCOM; PRUEBA DE ACCESO BASICO Y PRIMARIO	DESPUES DE 1989	AXE; 5ESS; S-12
SUECIA	LAS PRUEBAS DE CAMPO EMPEZARON EN 1986-1987; ACCESO BASICO Y PRIMARIO PILOTO EN 1988-89	1990	AXE
SUIZA	PRUEBA PILOTO DE ACCESO BASICO 1988-89	1991	EWSD; AXE
REINO UNIDO	PRUEBA PILOTO DE ACCESO DIGITAL INTEGRADO EN 1985; MULTILINEA IDA EN BLOQUES DE 30 X 64 KBPS DE 30x64 Kbps A FINALES DE OTOÑO; MERCURY LANZO EL SERVICIO DASS-2 BASADO EN CASI-RDSI A FINALES DEL AÑO PASADO	ACCESO BASICO EN 1989	SISTEMA X
FRANCIA	PRUEBA RENAN A MEDIADOS DE 1980	EL PRIMER SERVICIO COMERCIAL RDSI DE ACCESO BASICO A NIVEL MUNDIAL EMPEZO EN BRITANIA A FINALES DE 1987; EL SERVICIO FUE AMPLIADO AL AÑO SIGUIENTE; ACCESO ENTRE RDSI Y LA RED DE PAQUETES EN OPERACION TRANSPAC ACCESO PRIMARIO PLANEADO EN OTOÑO; DISPONIBILIDAD NACIONAL PROGRAMADA PARA FINALES DE 1990. LAS TARIFAS YA SE DIERON A CONOCER	E-10B; E-10MT; AXE ESTARA EN UN FUTURO
ALEMANIA	PRUEBA DE ACCESO BASICO DURANTA 1984-85; SERVICIO PILOTO EN 1987	SE PROGRAMARON 100 CENTRALES PARA MEJORAR UN SERVICIO EN RDSI EN 1989; COVERTURA NACIONAL PARA 1993; EL OBJETIVO A LARGO PLAZO ES INTRODUCIR CONMUTACION POR PAQUETES EN LA RDSI: LAS TARIFAS YA SE PAQUETES EN LA RDSI: LAS TARIFAS YA DIERON A CONOCER	EWSD; S-12
ITALIA	EMPEZANDO LOS SERVICIOS DE ACCESO BASICO	1990-91; LAS TARIFAS YA SE DIERON A CONOCER	PROTEO/GTD-5;

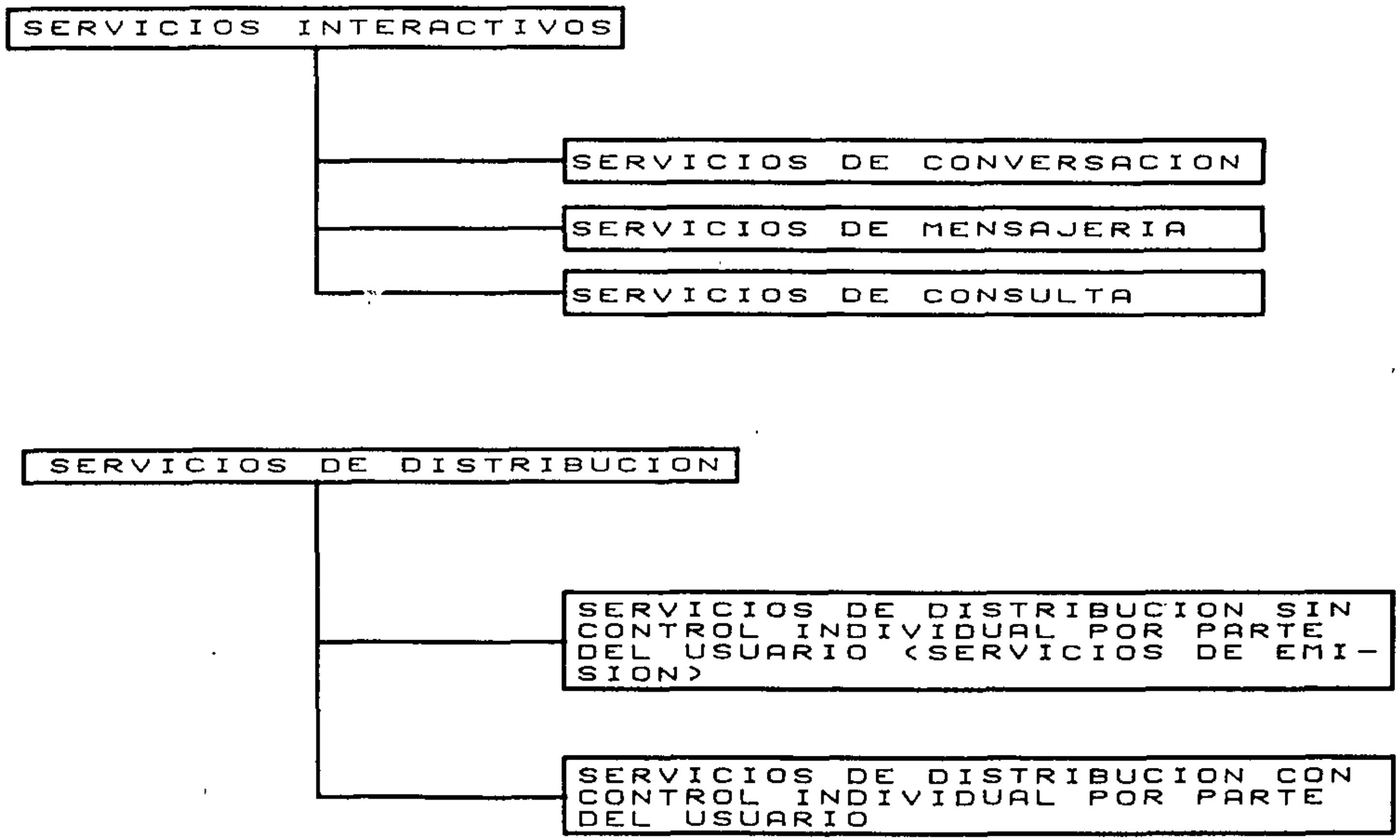


FIG. 2 CLASES DE SERVICIOS RDSI-BANCH [BISDN].

AB9A34

CLASES DE SERVICIOS	TIPO DE INFORMACION	EJEMPLOS DE SERVICIOS DE RDSI.	APLICACIONES	ALGUNOS ATRIBUTOS POSIBLES	CATEGORIA PROPUESTA	
					TELESERVICIOS	SERVICIOS PORTADORES
SERVICIOS DE DISTRIBUCION SIN CONTROL INDIVIDUAL POR PARTE DEL USUARIO.	VIDEO	SERV. DE DISTRIB. DE TV DE CALIDAD (PAL,NTSC,SECAM)	DISTRIBUCION DE PROGRAMAS DE TV	- DEMANDA (SELECCION)/ PERMANEN. - DIFUSION - BIDIREC. ASIMET./UNIDIRECIONAL	X	
		SERV. DE DISTRIB. DE TV DE CALIDAD MEJORADA. -DISTRIB. DE TV DE ALTA DEFINICION HDTV -TV DE ALTA CALIDAD HQTV	DISTRIBUCION DE PROGRAMAS DE TV	- DEMANDA (SELECCION)/ PERMANEN. - DIFUSION - BIDIREC. ASIMET./UNIDIRECIONAL	X	
		TV DE PAGA (PAGO POR VER,PAGO POR CANAL)	DISTRIBUCION DE PROGRAMAS DE TV	- DEMANDA (SELECCION)/ PERMANEN. - DIFUSION - BIDIREC. ASIMET./UNIDIRECIONAL	X	
	TEXTOS, GRAFICAS, IMAGENES FIJAS	SERVICIO DE DISTRIBUCION DE DOCUMENTOS.	PERIODICO ELECTRONICO, EDICION ELECTRONICA.	- DEMANDA (SELECCION)/PERMANENTE - DIFUSION/MULTIPUNTO - BIDIR. ASIM./UNIDIRECCIONAL	X	
	DATOS	SERV. DE DIST. DE INF. DIGITAL A ALTA VELOC. SIN RESTRICCIONES.	DISTRIBUCION DE DATOS SIN RESTRICCIONES	- PERMANENTE - DIFUSION - UNIDIRECCIONAL		X
	PELICULAS	SERV. DE DISTRIB. DE INF. DE VIDEO.	DISTRIBUCION DE SEÑALES DE VIDEO/AUDIO	- PERMANENTE - DIFUSION - UNIDIRECCIONAL		X
SERVICIOS DE DISTRIBUCION CON CONTROL INDIVIDUAL POR PARTE DEL USUARIO.	TEXTOS, GRAFICAS, SONIDO, IMAGENES FIJAS	DIFUSION DE VIDEOTEX DE CANAL COMPLETO.	- CAPACITACION Y EDUCACION A DISTANCIA - PUBLICIDAD - CONSULTA DE NOTICIAS - TELESOFTWARE	- PERMANENTE - DIFUSION - UNIDIRECCIONAL	X	

TABLA 4B.3: POSIBLES SERVICIOS EN LA RDSI DE BANDA ANCHA (CONTINUACION)

CLASES DE SERVICIOS	TIPO DE INFORMACION	EJEMPLOS DE SERVÍCIOS DE RDSI.	APLICACIONES	ALGUNOS ATRIBUTOS POSIBLES	CATEGORÍA TELESERVICIOS	PROPUESTA SERVICIOS PORTADORES
COVERSACION	DATOS	TELEACCION A ALTA VELOCIDAD.	- CONTROL DE TIEMPO REAL - TELEMETRIA - ALARMAS	------------------------------	X	X
	DOCUMENTOS	TELEFAX A ALTA VELOCIDAD.	TRANSFERENCIA DE TEXTOS, IMAGENES, DIBUJOS, ETC. DE USUARIO A USUARIO	- DEMANDA - PUNTO A PUNTO/MULTIPUNTO - BIDIR. SIMET./BIDIR. ASIMET.	X	
		SERVICIO DE COMUNICACION DE DOCUMENTOS	TRANSFERENCIA DE DOCUMENTOS VARIADOS DE USUARIO A USUARIO	- DEMANDA - PUNTO A PUNTO/MULTIPUNTO - BIDIR. SIMET./BIDIR. ASIMET.	X	
MENSAJERIA	IMAGENES: EN MOVIMIENTO (VIDEO) Y SONIDO.	SERVICIO DE CORREO DE VIDEO.	SERVICIO DE BUZON ELECTRONICO PARA LA TRANSFERENCIA DE IMAGENES EN MOVIMIENTO ACOMPAÑADAS DE SONIDO.	- DEMANDA - PUNTO A PUNTO/MULTIPUNTO - BIDIR. ASIM./UNID.(EN ESTUDIO)	X	
	DOCUMENTOS	SERVICIOS DE CORREO DE DOCUMENTOS	SERVICIO DE BUZON ELECTRONICO PARA DOCUMENTOS VARIADOS.	- DEMANDA - PUNTO A PUNTO/MULTIPUNTO - BIDIR. ASIM./UNID.(EN ESTUDIO)	X	
CONSULTA	TEXTOS, GRAFICAS, DATOS, SONIDO, IMAGENES FIJAS, IMAGENES EN MOVIMIENTO.	VIDEOTEX DE BANDA ANCHA.	- VIDEOTEX INCLUYENDO IMAGENES EN MOVIMIENTO - EDUCACION Y CAPACITACION A DISTANCIA - TELESOFTWARE - PUBLICIDAD - TELEVENTAS - CONSULTAS DE NOTICIAS	- DEMANDA - PUNTO A PUNTO - BIDIRECCIONAL ASIMETRICA	X	
		SERVICIO DE CONSULTA EN VIDEO	- PROPOSITO DE ENTRETENIMIENTO - EDUCACION Y CAPACITACION A DISTANCIA	- DEMANDA / RESERVADA - PUNTO A PUNTO / MULTIPUNTO - BIDIRECCIONAL ASIMETRICA	X	
		SERVICIO DE CONSULTA DE IMAGEN DE ALTA RESOLUCION	- PROPOSITO DE ENTRETENIMIENTO - EDUCACION Y CAPACITACION A DISTANCIA	- DEMANDA / RESERVADA - PUNTO A PUNTO - BIDIRECCIONAL ASIMETRICA	X	
		SERVICIO DE CONSULTA DE DOCUMENTOS	CONSULTA DE "DOCUMENTOS VARIADOS" DE CENTROS DE INFORMACION, ARCHIVOS, ETC.	- DEMANDA - PUNTO A PUNTO - BIDIRECCIONAL ASIMETRICA	X	

TABLA 4B.2: POSIBLES SERVICIOS EN LA RDSI DE BANDA ANCHA (CONTINUACION)

CLASES DE SERVICIOS	TIPO DE INFORMACION	EJEMPLOS DE SERVICIOS DE RDSI.	APLICACIONES	ALGUNOS ATRIBUTOS POSIBLES	CATEGORIA PROPUESTA	
					TELESERVICIOS	SERVICIOS PORTADORES
CONVERSACION	IMAGENES EN MOVIMIENTO (VIDEO) Y SONIDO	VIDEOTELEFONIA, VIDEOCONFERENCIA DE PUNTO A PUNTO.	COMUNICACION PARA LA TRANSFERENCIA DE VOZ (SONIDO), IMAGENES EN MOVIMIENTO, EXPLORACION DE IMAGENES FIJAS Y DE DOCUMENTOS ENTRE 2 LOCALIDADES (PERSONA A PERSONA, PERSONA A GRUPO, GRUPO A GRUPO)	- DEMANDA/RESERVA/PERMANENTE. - PUNTO A PUNTO. - BIDIRECCIONAL SIMETRICO/BIDIRECCIONAL ASIMETRICO. - (LA TRANSFERENCIA DE INFORMACION ESTA EN ESTUDIO).	X	
		VIDEOCONFERENCIA-MULTIPUNTO.	COMUNICACION MULTIPUNTO PARA LA TRANSFERENCIA DE VOZ (SONIDO), IMAGENES EN MOVIMIENTO Y EXPLORACION EN VIDEO DE IMAGENES FIJAS Y DOCUMENTOS ENTRE MAS DE 2 LOCALIDADES (PERSONA A PERSONA, PERSONA A GRUPO, GRUPO A GRUPO)	- DEMANDA/RESERVA/PERMANENTE. - MULTIPUNTO. - BIDIRECCIONAL SIMETRICO/BIDIRECCIONAL ASIMETRICO.	X	
		VIDEOVIGILANCIA	- VIGILANCIA EN EDIFICIOS. - MONITOREO DE TRANSITO.	- DEMANDA/RESERVA/PERMANENTE. - PUNTO A PUNTO/MULTIPUNTO. - BIDIRECCIONAL ASIMETRICO/UNIDIRECCIONAL.	X	
		INFORMACION DE VIDEO/AUDIO, SERVICIO DE TRANSMISION.	- TRANSFERENCIA DE SEÑAL DE T.V. - DIALOGO VIDEO/AUDIO.	- DEMANDA/RESERVA/PERMANENTE. - PUNTO A PUNTO/MULTIPUNTO. BIDIRECCIONAL SIMETRICO/BIDIRECCIONAL ASIMETRICO.		X
	DATOS	SERVICIO DE TRANSMISION DE INFORMACION DIGITAL A ALTA VELOCIDAD SIN RESTRICCIONES.	- TRANSFERENCIA DE DATOS A ALTA VELOCIDAD +INTERCONEXION DE LAN'S +INTERCONEXION DE COMPUTADORA A COMPUTADORA - TRANSFERENCIA DE VIDEO Y OTROS TIPOS DE INFORMACION. - TRANSFERENCIA DE IMAGEN FIJA.	- DEMANDA/RESERVA/PERMANENTE - PUNTO A PUNTO/MULTIPUNTO - BIDIRECCIONAL SIMETRICA/BIDIRECCIONAL ASIMETRICA.		X
		SERVICIO DE TRANSFERENCIA DE ARCHIVOS DE ALTO VOLUMEN.	- TRANSFERENCIA DE ARCHIVOS DE DATOS	- DEMANDA - PUNTO A PUNTO/MULTIPUNTO - BIDIRECCIONAL SIMETRICA/BIDIRECCIONAL ASIMETRICA.	X	

TABLA 4B.1: POSIBLES SERVICIOS EN LA RDSI DE BANDA ANCHA (BISDN).

FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

VoP: STANDARS REMAIN ELUSIVE

EXPOSITOR: M. en C. ARTURO ELIE ALALUF OLIVARES
PALACIO DE MINERÍA
JUNIO DE 1999

may 1999

VoP: Standards Remain Elusive

Microsoft's NetMeeting popularized the ITU's H.323 specification for VoIP, but an alternate standard, MGCP, is now giving H.323 a run for its money

William E Witowsky

It was only a few years ago that voice over packet (VoP) telephony consisted of shareware downloaded onto multimedia PCs, enabling half-duplex conversations over the Internet Users visited a chat server and calls suffered from poor quality of service (QoS) due to the PC platform itself; excessive delay, variable delay (jitter) and network congestion also caused lost packets. PC-to-PC VoP calls quickly evolved into POTS phone-to-phone VoP calls using PC-based gateways containing PSTN interface cards. These early VoP telephone gateways also suffered from poor voice QoS

Since then, VoP QoS has improved tremendously due to improvements in VoP gateway devices (which have migrated to embedded systems platforms), the availability of higher bandwidth network pipes, and improvements to algorithms and protocols in routers and switches that reduce latency and lost multimedia packets.

Most of the initial commercial adaptation of VoP telephony has not occurred over the Internet, but has taken place over private IP networks (i.e., corporate enterprises and long-distance service providers), thereby avoiding many of the QoS problems associated with the Internet.

Businesses with remotely located branch offices avoid access charges and settlement fees by adding VoP gateways to their existing corporate intranets. Installation of VoP can easily pay for itself within six months at companies with multiple international sites. For these companies, interoperability with other organizations and public VoP gateways is not an issue. Their focus is on selecting a reputable VoP gateway vendor to provide good QoS voice and a turnkey solution for lowering the cost of long-distance phone calls

Similarly, new long-distance providers use VoP technology to offer inexpensive international phone services between major cities These service providers select VoP gateways from a particular vendor and deploy them at all sites. Unlike the enterprise deployments, these service providers must develop significant management and administrative systems to handle billing, routing, authorization and network management for their customers.

The tremendous success of VoP technology has driven the need for interoperability Networks must communicate with each other to provide seamless, PSTN-like interconnectivity; corporations and service providers have to be able to mix and match equipment from different vendors.

The Frame Relay Forum (FRF) was one of the earlier standards bodies to regulate voice and fax over frame relay networks The Voice Over Frame Relay Implementation Agreement (FRF 11) addresses peer-to-peer communications between two VoP gateways This includes encoding of messages, negotiation of standard voice codecs, the support of Group 3 fax relay, the support of silence suppression and relay of dial digits and channel associated signaling bits To date, there is only a standard for transferring voice and fax over permanent virtual circuits (PVCs) Proprietary signaling schemes have been used for supporting voice and fax over switched virtual circuits (SVCs)

Standards support for voice and fax over ATM networks was initially provided by the ATM Forum's Voice and Telephony Over ATM (VToA) Technical Working Group in the form of circuit emulation services that enable voice and fax to be transported transparently as constant bit rate (CBR) 64 kbps data. Today, there is a VToA standard under final ballot titled "AAL2 Trunking Using AAL2 for Narrowband Services," which provides voice compression, silence removal and fax demodulation. Interworking VToA with voice over frame relay and voice over IP (VoIP) is slated as future work.

By far, the strongest interest and activity, as well as controversy, involves standards for supporting VoIP networks. As the dominant protocol standard, most deployments using VoP technology are based on transporting packet voice and fax over TCP/IP networks. Most of the original standards activities for VoIP are defined by the ITU-T Recommendation H.323, "Packet-based Multimedia Communications Systems."

Aside from being an ITU-T recommendation, the acceptance of H.323 by gateway manufacturers was strongly fueled by Microsoft's decision to incorporate H.323 into its NetMeeting product, which is distributed for free. Thus, gateway manufacturers were compelled to demonstrate interoperability with NetMeeting at some basic level of operation even though NetMeeting is typically used for PC-to-PC communications.

Demonstrated interoperability between different vendors supporting H.323 is a fairly recent phenomenon. Some of the common interoperability problems are:

• Version incompatibility. Vendors have implemented different versions of the H.323 standard.

• Complexity. There are many options and supplementary services. As with any relatively new standard, ambiguities of the specification can lead to different interpretations.

• Gateway to GateKeeper communications. GateKeepers perform address translation, access control and gateway locator services. The syntax is well defined but the semantics for using the syntax is not. GateKeeper to GateKeeper communications are still undefined by the standards. This will be defined in Annex G of H.225, a standard that describes the signaling protocols used by H.323-compliant devices.

ITU-T Recommendation T.38 defines the protocol to be used between gateways for supporting Group 3 fax over IP networks T.38 supports two underlying network protocols: UDP/IP for reliable networks and TCP/IP for unreliable networks H.323 Annex D, which was recently approved as a standard, describes the protocol for switching from packet voice mode to packet fax mode upon detection of fax signals from the gateways

Gateway manufacturers have proposed VoIP standards such as Level 3 Communications' IP Device Control (IPDC) and Bellcore and Cisco's Simple Gateway Control Protocol (SGCP) as alternatives to H.323 to the Internet Engineering Task Force (IETF). These two proposals were subsequently merged into a common Media Gateway Control Protocol (MGCP) that has gained widespread support from industry vendors which have joined to form the Multiservice Switching Forum (MSF) in promotion of MGCP. The cable modem industry has been an early adopter of MGCP since the PacketCable initiative sponsored by CableLabs selected MGCP as the call signaling protocol for Data Over Cable Service Interface Specification (DOCSIS)-compliant cable modems

The main reason that many vendors and service providers are adopting MGCP instead of H 323 is its overall architectural approach MGCP assumes a call control architecture where the call control intelligence is outside the gateway itself and is handled by external call control elements referred to as call agents. This centralized call agent approach marries well with the existing PSTN architecture approach where signaling for SS7 calls are handled centrally. This allows scalability for millions of clients

The IETF has also produced specifications for other types of multimedia applications These include the

Session Initiation Protocol (SIP), the Session Announcement Protocol (SAP), the Real-Time Streaming Protocol (RTSP) and the Session Description Protocol (SDP). SIP in particular has been proposed as an alternative to H.323, but, to date, has not received widespread support.

The maturity of VoP technology, the development of standards and growing market demand are all positive trends that will facilitate widespread adoption of interoperability standards. In 1999, there will be more announcements and demonstrations of vendor gateway interoperability for both H.323 and MGCP. Support for H.323 remains strong, particularly in Europe H.323 standards and implementations are much more mature than MGCP. H.323 interoperability activities include the ETSI Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) initiative as well as the VoIP working group overseen by the International Multimedia Teleconferencing Consortium (ITMC). Several vendors have recently announced the iNow! initiative for interoperability between gateways and GateKeepers. On the MGCP front, the CableLabs PacketCable initiative is driving interoperability of MGCP clients and call agents Vendors are also building carrier-class VoIP gateways and call agents that allow MGCP to interface to the SS7 world

Finally, for there to be a new world order based on VoP telephony, standards must be robust and scale on a global basis. Important standards issues that will take a little longer to sort out include billing and settlement interoperability, security and overall network scalability whereby callers can truly reach any destination in the world.

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RSNo 305

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CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

IP & ATM IN THE WAN INFRAESTRUCTURE

EXPOSITOR: M. en C. ARTURO ELIE ALALUF OLIVARES
PALACIO DE MINERÍA
JUNIO DE 1999

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june 1999

IP & ATM in the WAN Infrastructure

Public Networks
Visionaries proclaim that IP is the convergence protocol. But what builds networks and revenues in the interim before universal IP convergence?

Steve Willis and Chris Baldwin

IP: The Internet runs on it; private data services are migrating to it; and new voice-over-IP (VoIP) developments auger the transition of voice to a common IP infrastructure. But this IP vision is long-term. The size and scope of the wide area network (WAN) preclude overnight transformations.

New service providers are deploying extensive fiber optic transmission plants and promoting both wholesale and retail services, shepherding existing traffic and new bandwidth-intensive services onto their new networks. But how are existing revenue-generating services and new IP-based data services integrated onto a common infrastructure cost-effectively?

ATM is promoted as the network convergence layer integrating legacy services and allowing fine-tuned allocation of network capacity to different traffic types without the inefficiencies associated with traditional overlay network architectures. ATM's quality-of-service (QoS) tools enable sophisticated quality control beyond simple bandwidth measures. ATM, however, is not without its own issues such as the bandwidth "ATM cell tax," which is unacceptable to IP-only customers. If ATM's chief attraction is its ability to allow the convergence of legacy traffic types, is it merely a transitional technology deserving only passing deployment and investment?

In truth, the differences between ATM and IP are overstated: IP datagrams and ATM cells are both packets, and IP and ATM service platforms both operate by packet switching. Integrating IP and ATM is no longer technical alchemy; Moore's Law and the ever-evolving art of system design now allow service providers to have their cake and eat it, too: ATM and IP simultaneously with service-time configuration between the two. Even QoS is now common to IP and ATM and operates on very similar principles: traffic classification and rate enforcement on switch ingress with class-based queuing on switch egress.

Service providers are actively building a new packet division multiplexed (PDM) infrastructure where both IP and ATM platforms are deployed, as both are required. The PDM infrastructure represents a radical rethinking of the traditional layered WAN architecture. Gigabit-speed IP and ATM service platforms are now being deployed in transport and service protocol roles. The previous strict delineation between transport and service systems evaporates, along with significant operational and capital expenses associated with layered architectures. PDM infrastructure is used in the "last five miles" of the WAN infrastructure that lies between the new core transport network, composed of wavelength division multiplexing (WDM) devices, and the customer premise. IP and ATM are both required in this crucial area of the network to integrate legacy traffic and new IP data traffic on a common packet-switched infrastructure.

Wholesale Provider Issues

Wholesale service providers typically deploy fiber along railroad or oil/gas pipeline rights of way and provide transport services to local and regional providers. Historically wholesale providers have offered pure bandwidth services composed of high capacity leased lines. The traditional SONET/SDH technology, however, allows only a coarse partitioning of transport capacity in often inconvenient increments leading to transport bandwidth waste. Bandwidth-only offering is an increasingly low-margin business, and considerable expertise is required of the local and regional providers to deliver usable services on this raw capacity.

Wholesale providers, therefore, need to climb the value chain and offer higher-function services on finer-tuned bandwidth increments. In many cases, ATM is an ideal technology for wholesale providers. ATM virtual circuits (VCs) can be provisioned with incremental bandwidth control apportioning bandwidth resources efficiently, and VC provisioning is performed end-to-end across a network with simple "point and click" operations. When wholesale providers, however, try to offer wholesale ATM-based services to ISPs, they usually encounter a strenuous objection to the ATM cell tax. ATM operates by segmenting data packets into 53-byte cells, each with a 5-byte header, separate from the IP header. Each cell now wastes 10 percent of the link's bandwidth, and if the traffic pattern is composed of small packets that do not fit neatly into one cell, additional bandwidth is wasted due to the repetitive transport of partially filled cells, bringing the total cell tax as high as 30 percent in some cases. ISPs want a wholesale service based on the transmission of IP packets directly on SONET/SDH, or packet on SONET (PoS) services

Wholesale providers quickly discover they need to offer both ATM- and IP packet-based services ATM for wholesale multiservice bandwidth; PoS for ISPs and new IP-oriented services. Unfortunately, wholesale providers can rarely predict which types of service offering are needed where, and Murphy's Law ensures that the wrong equipment is always deployed in excess capacity in the wrong location. The wholesale provider needs IP and ATM, with service-time configuration between the two

Retail Provider Issues

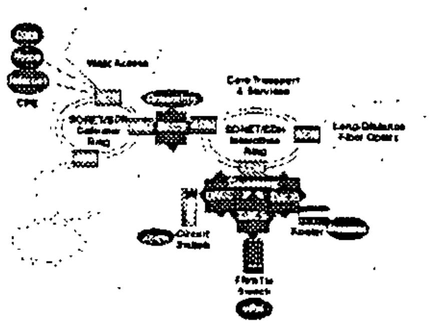
Retail providers encounter a different set of issues. Their sales forces enter multiple buildings in an office park selling services. At the first building they may find a 20-year-old company, which has a little bit of every known WAN protocol, voice, SNA, frame relay and Internet access, all on separate leased lines. In this case, a retail provider wins business with a consolidated offering that combines all of these diverse technologies onto a common infrastructure for a lower price without using separate overlay networks. For such a customer, ATM is an ideal offering. An ATM service multiplexer on the customer premise adapts the diverse technologies into different ATM VCs to be switched by the network. The customer does not have to re-engineer any applications or networks, and both customer and provider are well served with an ATM infrastructure. An IP-only infrastructure would not win this customer's business because a wholesale rewrite of all the applications to IP is not cost justifiable, and an IP-only solution solves too few of the customer's overall WAN requirements.

The next stop on the sales representative's journey may, however, find a 3-year-old company that only uses IP and is adamant about getting as much bandwidth efficiency out of its service as possible It opposes any ATM-based service offering due to the familiar ATM cell-tax issue. This company demands pure IP PoS services with migration to QoS support and even wants to consolidate its voice traffic onto this common IP infrastructure The ATM-based provider leaves empty-handed, but the IP PoS provider leaves with the order

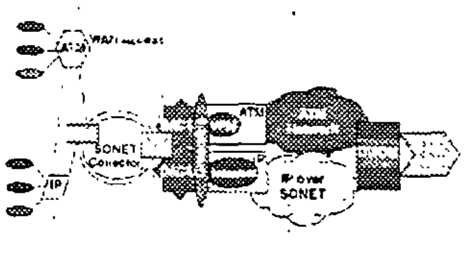
Service providers with both ATM and PoS services with service-time configuration between the two are in a better position to meet the diverse needs of the customer base

Overlay Networks

Traditional WAN construction uses a time division multiplexing (TDM) transport layer based on SONET/SDH technologies to groom digital bitstreams from the customer premise to the service platform Each service becomes an overlay network carried over a TDM transport layer As illustrated in **Figure 1**, this architecture is based on a rigorous differentiation between transport and service platforms Collector rings aggregate traffic from office parks and metropolitan locations, and traffic is groomed with digital cross connects (DACS) onto interoffice rings, which in turn require additional DACS to groom traffic onto service-specific platforms. The grooming of traffic between rings and onto service platforms is a bandwidth-inefficient operation with considerable capital and operations expense.



Today's bandwidth-intensive and cost-effective WAN architecture consists of directly connecting gigabit-speed IP and ATM platforms to WDM transport equipment and building the last-five-mile infrastructure with PDM technology, instead of the familiar group of SONET collector and interoffice rings with DACS traffic grooming **(see Figure 2)**. ATM technology is an excellent legacy traffic integrating technology because it has a comprehensive set of adaptation layers (AAL1 through AAL5), which convert existing traffic streams into an ATM cell format in a standards-based, multivendor, interoperable environment. IP routing delivers native-data performance for new applications without the ATM cell tax. IXC Communications, Frontier Globalcenter, Sprint and Williams Communications Group, among others, are now deploying this IP and ATM WAN architecture.



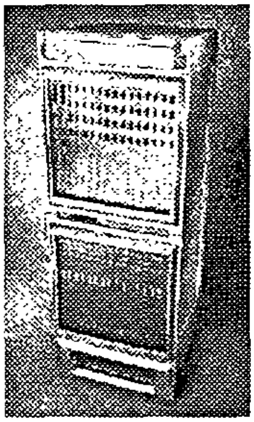
Today, neither IP nor ATM alone is a perfect solution, although each has its strengths and each is required. To win new revenue quickly, service providers should not attempt to force a change in customers' legacy systems, but should offer to bridge their legacy systems and the emerging IP technologies. Today's emerging PDM architecture represents a dramatic rethinking of the role of ATM and IP protocols: ATM and IP now represent an integration of service-protocol platforms and transport technology, instead of the traditional practice of constructing separate, parallel service networks with separate switching platforms fed by a common TDM-based transport infrastructure. ATM and IP deliver both the transport capacity and the underlying service. New IP telephony developments suggest that even voice traffic may eventually migrate to the new PDM infrastructure.

Steve Willis is the cofounder and chief technology officer of Argon Networks. Before Argon, Willis cofounded Wellfleet Communications/Bay Networks (now Nortel Networks). Willis was the coarchitect of Wellfleet's multiprocessor, multiprotocol bridge-router. Christopher Baldwin, Argon's vice president of marketing, once served as product marketing director at Cascade Communications (acquired by Ascend and now Lucent). He was responsible for the launch of the Cascade 500 ATM switch.

RSNo 343

Pluris -- 20000 Terabit Network Router (TNR) Product of the Month

June 1999



Is it too early to think about terabit speeds for the network core? Well, a little But the timing's just right to establish a market lead Most of the big carriers and IXCs are carefully evaluating boxes from the big three terabit switch router vendors--Avici, Nexabit Networks and Pluris--as they become available, and release dates are being bumped up accordingly. Many analysts agree that deployment will likely come next year

Most analysts are behind Pluris, too Although Avici may get to market first, Pluris' architecture and scalability are impressive The 20000 Terabit Network Router (TNR) supports up to 184 terabits of aggregate switching capacity while being managed as a single logical system, which leads the market The system can also scale effectively From 90-Gbps switching and 10-Gbps line capacity, carriers can move up to 184 Tbps of nonblocking switching capacity and 19.2-Tbps line capacity with OC-12, OC-48 and OC-192 interfaces, another industry first Interconnection is accomplished by fiber and intershelf links at the switching fabric level instead of between line cards, as is traditionally done. This means valuable port space isn't consumed by interconnecting line cards, and operational and equipment costs are reduced by as much as 60 percent, according to the vendor.

"The demand for Internet bandwidth makes it increasingly difficult for service providers to scale their networks by interconnecting increasing numbers of today's routers with throughput capacities peaking at 20 Gbps to 30 Gbps," said Joe Skorupa, director of switching and routing at RHK "Building a large router by clustering smaller devices forces the majority of each router's capacity to be dedicated to communicating with other routers in the cluster Additionally, the task of configuring and managing this ever-growing and increasingly complex network quickly degenerates into an exercise in futility For these reasons, vendors [such as Pluris] that can deliver platforms that scale from tens of gigabits to terabits and from dozens of ports to thousands while in service will have a significant advantage."

The fabric design provides multiple high-speed fiber-optic paths for evenly distributed ATM or IP traffic. The traffic is shaped without multiple Layer 3 lookups or forwarding, but rather with MPLS and a single Layer 3 lookup at the ingress point: intrashelf forwarding is done through internal fast switching Line-speed quality of service (QoS) is also a selling point, and the TNR delivers this with programmable QoS, class of service (CoS), weighted fair queuing and buffer management, as well as per flow SLAs, IP multicast, IP tunneling and IP bonding for aggregation of multiple high-speed dense wavelength division multiplexing (DWDM) channels into fat IP pipes Finally, the TNR boasts in-service upgrade capacity, redundancy, hot swappable components, full IP routing protocol support and in-band or out-of-band management.

The Pluris 20000 Terabit Network Router will be available in Q4 99. Pricing has not yet been announced. For more information, visit the company's Web site at *www.pluris.com*.

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<u>WHO HAS THE WINNNG STRATEGY</u>

EXPOSITOR: M. en C. ARTURO ELIE ALALUF OLIVARES
PALACIO DE MINERÍA
JUNIO DE 1999

april 1999

Americas

Who Has the Winnng Strategy?

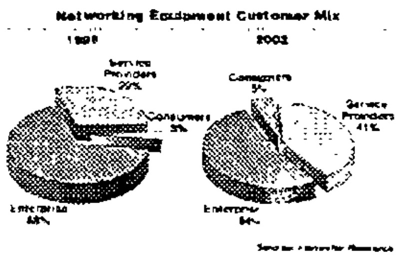
The Battleground. The U.S telecom market.
The Combatants: The Big 6 network vendors
The Prize: Leadership in next-generation networks

Susan O'Keefe and Sam Masud, senior editors

As revenues from the traditional voice infrastructure stagnate and traffic on packet networks soars, the Big 6 equipment vendors--Alcatel, Cisco, Ericsson, Lucent, Nortel Networks and Siemens--are faced with a challenge and an incredible opportunity Each vendor subscribes to the notion of all-purpose networks, but a battle is brewing over the migration path to those converged networks. The company or companies with the winning formula could emerge from the fray in the next few years with more than $100 billion in telecom equipment revenues.

Despite predictions that worldwide demand for digital central office switching systems will fall this year, no one expects a wholesale change to packet networking In fact, analysts predict that in five years, the network will be substantially the same as it is today because of the investment in installed infrastructure. But service providers are expected to increase spending by at least 30 percent over the next few years and those dollars will cement the direction of the industry **(See Figure 1)** "We are not going to see a revolution in the next five years, but we will know in two to three years what that revolution is going to look like, what technologies are going to win and which vendors are going to win," said Tom Nolle, president of CIMI Corp. "All the vendors know it and know that if they want to be players in the rest of the world in terms of this 21st century network, they have to be players in the United States first."

Nolle called the U.S. market the Petri dish in which the 21st century network will grow. Rick Malone, principal analyst with Vertical Systems, agreed. "The United States is not one or two years ahead of the rest of the world, it's four to five years ahead," Malone said "The reason the Big 3 in North America are excited is the same reason the three European players are making aggressive moves into the United States. Demands on the network are causing the need for infrastructure to migrate to higher speeds, provide additional applications and combine voice onto this packet architecture"



The early moves have been made by the North American players Nortel bought Bay Networks for its IP expertise; Lucent purchased Ascend Communications for its installed base of ATM switches, and Cisco, the only Big 6 member with a pure data networking background, has made a number of small acquisitions More recently, Alcatel made back-to-back acquisitions of Xylan and Assured Access, and Siemens made the long-predicted purchase of Argon Networks (a hybrid IP router/ATM switch start-up that has yet to ship a product) and the surprise buy of Castle Networks, a start-up with a product currently in beta testing that helps bridge the circuit-switched and packet-switched worlds Siemens followed up the Castle acquisition with the proposed purchase of IP edge switch maker Redstone Communications Although a deal with 3Com was also rumored, it did not come to fruition by the time Siemens announced its new data networking venture, Unisphere Solutions, in March Ericsson, the leader in mobile communications, thus far has been low key in terms of acquisitions in the United States, although last year it acquired Advanced Computer Communications (ACC), a remote access concentrator maker, and also formed a data networking group last fall. Analysts predict Ericsson will get more aggressive in the coming months.

More buyouts could come fast and furious, creating a rich opportunity for start-ups. Two hot areas every vendor is watching are high-speed routing and services mediation products. Many of the vendors have minority investments in start-ups that provide gigabit/terabit routers such as Juniper, Pluris, Avici, Nexabit and Netcore, and could seek to buy them outright Service mediation products from vendors such as Castle Networks, Salix, TransMedia and Sonus are the first generation of so-called agnostic switches and are receiving praise from industry analysts. "These are next-generation Class 4 and Class 5 switches that are really going to have a big play, and the big equipment vendors know that," said Frank Dzubeck, president of Communications Network Architects (CNA). Malone agreed: "A lot of these boxes are going to be put into carrier networks, especially the ones that can scale, because we're talking about a lot of circuits."

At a minimum, acquisitions help fill out vendors' product portfolios, although increasingly the big carriers prefer a multivendor environment. So how important is the notion of a one-stop shop? According to vendors, it is crucial because service providers are looking to suppliers not only for products but for systems integration as well. "If your customers are looking to you to do the integration for them, you have a lot more control and ability to do that if you own most of the major components rather than relying on OEM agreements," said Mike Day, Alcatel USA's director of strategic network planning "It makes it a lot easier to guarantee service quality and network evolution " Kevin Oye, Lucent's vice president of strategic and business development for data networking, agreed "The one thing we don't want is to be viewed by our customers as simply box deliverers," he said

According to Dzubeck, even though the product lines of the big vendors are becoming richer and richer, carriers are hesitant to give too much responsibility to one supplier. Still, vendors are getting bonus points for having their own product sets rather than sourcing them from partners. "There are tactical products to provide a tick on an RFP, and those are the products that vendors can OEM. But a vendor has to own the strategic products and that's why we're seeing so much merger activity," Dzubeck said. Siemens is a classic example of a vendor that needed to gain core competencies through acquisitions. It partnered with Newbridge and 3Com for ATM backbone and edge switches and IP service level management; invested in Juniper Networks for backbone routers; and resells 3Com's remote access concentrators. Those arrangements are still in place, but the company gained strength in backbone routing and services mediation through its acquisitions of Argon and Castle.

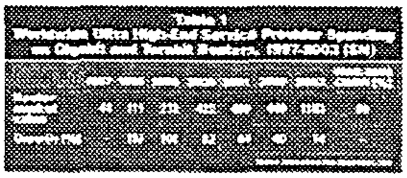
"Dense wavelength division multiplexing (DWDM), ATM and IP over ATM are going to be the issues that drive these competitive players," Nolle said. "It's really their positions in those spaces and their strategies for the satisfaction of the voice requirements in this new-generation infrastructure that are going to decide the victories."

Router Wars

Though none would dispute Cisco's leadership in the router market, there are some analysts who question whether the Cisco 12000 is a true carrier-class product. "Cisco's strength is its customer relationships with the ISPs and its understanding of what needs to be done to build large IP networks, but the 12000 is not a carrier product in terms of being fault-tolerant and NEBS compliant," said Joseph Skorupa, director of routing and switching at analyst group RHK. Cisco, analysts said, is rewriting the Cisco IOS from the ground up to make the software modular and portable; it is expected to release the beta version in the second half of this year.

IDC, which separates high-end routers into gigabit and terabit products, does classify the Cisco 12000 in the carrier-class gigabit group along with the Ascend GRF, Lucent PacketStar, Juniper M40, Netcore Everest, Torrent IP 9000 and Nortel Versalar 15000, and gives Cisco praise for its early lead by selling approximately 1000 of the 12000 routers. "It's Cisco's market to lose," said Lee Doyle, vice president of data communications research at IDC. Skorupa sees it differently, giving Juniper the nod as the superior performer. "We know that Juniper has over a dozen paying customers including three operational networks with volume deployment. There are three or four start-ups chasing Juniper, but they are at least six months behind." Cable & Wireless USA became a Juniper customer last month when it agreed to deploy 16 routers in its Internet backbone. Five equipment vendors (Ericsson, Lucent, Nortel and, jointly, Siemens/Newbridge) and UUnet have invested in Juniper and have product distribution rights as well as the right to integrate Juniper's technology into their product lines and services.

The terabit router group is a more rarified club consisting of Avici (in which Nortel has a stake), Nexabit and Pluris. IDC predicts that in 2003 vendors will ship about 6500 gigabit routers representing more than $900 million in revenues; by contrast, only 600 terabit routers will be shipped that same year and will generate about $192 million in revenues **(see Table 1)**.



ATM Thrives

Whether the router or the ATM switch will be at the core of the network is an on-going religious debate within the industry. "Cisco is trying to make that case because they need to, that's their whole line," said Lucent's Oye. But Junaid Islam, Cisco's group manager for service provider marketing, said that there is room for both IP and ATM. "People have gotten into thinking one network, but there is nothing wrong in having both an IP and an ATM network if that makes sense. We're focusing on the people who are spending new dollars, the people spending on IP networks." CNA's Dzubeck agreed. "It's the newbies [not the mainline facilities-based carriers] who want IP. That's the Qwests, Level 3s and ISPs. They want IP not ATM."

Despite the focus on IP, Dittberner Associates noted in a recent report that installation of ATM switches worldwide is occurring at a rate that far exceeds the early growth in digital CO equipment. But the firm also warned that the "role of ATM as the dominant switching technology in public networks over the next decade is not assured." Of the 35,000 ATM systems installed globally, Dittberner gave Cisco the lead, followed by Nortel and Newbridge **(see Table 2)**. Although analysts said that Cisco has done much to revamp its ATM technology since it first sold its ATM switches to AT&T, they give the edge to Ascend's ATM GX550 as a true core switch As a result of its pending $20-billion acquisition of Ascend, Lucent has bought its way into being the ATM player, with both the GX550 core switch and CBX500 multiservice edge switch in its stable, Skorupa said Nolle agreed: "Lucent now has an incumbency in the ATM networks of almost every one of the major players. That's worth $20 billion because without it, Lucent would have had to build a product from scratch and fight to establish it in a marketplace where Cisco is already established because of its deal with AT&T. With the acquisition of Ascend, Lucent has more ATM switches in U S. service providers than Cisco does." Lucent's Oye said the Ascend purchase (scheduled to close in May) goes beyond that: "Yes, Ascend has great products today, but what I'm excited about is what we can do with those same people for the next generation of products "

Table 2
Installed Base of ATM Systems
(System Count as Reported by Suppliers)

[illegible]	[illegible]	[illegible]
[illegible]	[illegible]	[illegible]
[illegible]	[illegible]	[illegible]
[illegible]	[illegible]	[illegible]
[illegible]	[illegible]	[illegible]
[illegible]	[illegible]	[illegible]
[illegible]	[illegible]	[illegible]
Others	[illegible]	[illegible]
Total Installed Base	[illegible]	100%

[illegible]

As for the other players, Nortel is fleshing out its Passport line of ATM switches with the Passport 15000 core switch that is expected to ship this quarter Alcatel and Siemens have a presence in ATM switching, but Alcatel seems to be backing off in terms of pushing its 1100 HSS ATM switch Its acquisition of Xylan has more impact in the enterprise market rather than being a strong move in the service provider space. Siemens' MainStreetXpress line has been developed through a partnership with Newbridge, and one of the products it has contributed to the mix, the 36190, has "not been successful in North America or anywhere else in the world," Skorupa said Indeed, last month Siemens/Newbridge announced a five-year deal with Global One, and the initial product in play will be the 36170, which comes from Newbridge.

"Siemens seems like it has been downplaying that relationship with Newbridge a lot these days With the Argon acquisition, they're buying some equipment of their own, so it will be interesting to watch what will happen there," said IDC's Esmeralda Silva. Malone anticipated the possible acquisition of companies such as Newbridge or Fore Systems (which is in fourth place with a 9.2-percent share of the global ATM market), a view shared by Silva "Newbridge would make a decent acquisition for anyone looking to get into that space. They are especially strong in the ATM edge switch area," she noted.

Although the U.S is Ericsson's second-largest market after China, the vendor has failed to penetrate this market with its ATM equipment It did. however, recently win a contract to upgrade Swiss Telecom from a circuit-switched network to an ATM network With last year's launch of the AXD 301, an ATM switch that scales from 10 Gbps to 160 Gbps, Ericsson is hoping to crack the U.S. market. "U S operators have not deployed these high-capacity switches, but we've got two carriers considering deployment," said Gary Pinkham, Ericsson's vice president of business development for data networking solutions

SONET or DWDM?

Despite the emergence of DWDM, a study last month by Communications Industry Researcher cautions that accounts of SONET's demise are premature. As evidence, CIR cites Cisco's participation in the $53-million funding of Cerent, which in February announced a SONET/SDH transport system for aggregating voice, data and video services over SONET running up to 10 Gbps. Lucent's acquisition of Sybarus Technologies, a Canadian SONET/SDH maker, is proof that the "smart money" is still flowing to SONET, CIR said Incumbent local exchange carriers (ILECs), which account for more than 70 percent of the SONET market in the U.S., will continue to buy SONET equipment over the next decade, according to CIR. Market projections from RHK note that, although the SONET market will rise from $4.5 billion last year to slightly over $5 billion in 2002, 10-Gbps SONET equipment will assume an increasingly large share of the overall market

Although strong in SONET, Nortel, like Lucent, seems to be focused more on DWDM. Siemens, which formed an optical networking group about 18 months ago to focus on the long-haul market, has almost no presence in SONET despite being the No 2 player in SDH technology worldwide "Because of our SDH capability, we have technology strength in SONET, but no marketing strength," acknowledged Mike McLaughlin, the group's vice president and general manager. "When we looked at entering the market, we saw a lot of difficulty going up against the embedded SONET suppliers We believe that SONET will be in the networks for years to come, but the majority of growth will move to optical interfaces in gigabit routers and ATM switches So we decided to enter the long-haul DWDM market because the technology is still evolving so rapidly " Siemens flagship product is a 32-channel system operating at 10 Gbps per channel

Cisco does not have a play in DWDM, a market that RHK predicts will grow 70 percent in North America from $1.9 billion last year to $3 2 billion by 2002 But it might make its own move shortly, although it has a cooperative marketing relationship with Ciena Systems, which claims to have an 80-percent share of the global business for 16-channel systems "We'll substantially increase our competence in DWDM in the next few months," said Cisco's Islam, hinting that this could mean a partnership, minority investment or some other alliance. RHK expects systems with 16 or more channels to be the fastest-growing market segment Nearly all vendors are either working on or have announced 40-Gbps single-laser optical transmission systems, but don't expect to see any commercial trials before the end of the year. Lucent, for example, recently announced MCI WorldCom-hosted early lab trials of its 40-Gbps TDM-based WaveStar 40G Express and will do commercial testing in Q4 99.

Frost & Sullivan estimated that Lucent has the largest share--30 percent--of the DWDM market in the United States. "We've put 80 wavelengths on a fiber and we'll go higher," said Rich Gitlin, Lucent's CTO for data networking systems. "We're reducing the risks for service providers, and we're giving them the ability to deliver more and more bandwidth." Alcatel also claims competency in DWDM, and according to RHK is No 3 in North America in integrated SONET and DWDM systems **(see Table 3)**.

Table 3
1998 DWDM Market Leaders by Dollar Value Shipped

[illegible] Leaders	Integrated SONET & WDM Systems	Open DWDM Systems
1. Nortel	1. Nortel	1. Ciena
2. Lucent	2. Lucent	2. Lucent
3. Ciena	3. Alcatel	3. NEC

Source: RHK

Although Ericsson has a complete family of SONET and DWDM systems (in December it was selected by Spain's Telefonica S.A. for an expandable 16-to-32-channel system), it has chosen to focus on the metropolitan DWDM market in the U S Last month, Ericsson announced a DWDM system that provides the efficiency of a protected ring and is designed to operate up to 500 km, a distance sufficient to serve several small cities. "With the metro ring, you could do Gigabit Ethernet over DWDM on one channel, packet over SONET on another and perhaps a leased line on another channel," said Roselyne Genin, vice president of optical networks in Ericsson's Network Operators Group. Leveraging its acquisition of Cambrian Systems last year, Nortel also has introduced a metro DWDM system, called OPTera Metro, that can be used in a point-to-point or a survivable ring configuration. Bell Canada plans to trial the system. Cisco, building on its IP expertise, has added the dynamic packet transport line card to the Cisco 12000 router The card combines SONET restoration principles with LAN capabilities such as packet prioritization for service providers to support the delivery of voice and video services over IP and virtual private networks.

Given the acquisition climate, two new companies in the optical networking space, Lightera Networks and Sycamore Networks, are also attracting the attention of industry watchers. In fact, Ciena acquired Lightera in mid-March. Both specialize in optical switches that make it easier to provision high-speed private lines such as OC-48 connections. According to Dzubeck, a company such as Sycamore is an example of how fast advances in technology can come. "Sycamore has gone from product inception to shipment in nine months, when in the optical world you're talking about product development taking multiple years," Dzubeck said Nor are service providers reluctant to buy products from start-ups, as evidenced by Williams Network's $24-million pact for Sycamore products Executives at these new companies are often on their second or third start-up and have well-established contacts among service providers

The Slow Road to Convergence

In the current IP-versus-ATM debate, Skorupa believes that too much is being made of whether IP is a suitable protocol for voice traffic. "I prefer to call it voice over non-circuit switched networks. The first place we'll see the deployment of packet voice is in trunk networks--that is, replacing the tandem switches with packet switches whether they are IP or ATM. With these switches you've got what's called the tyranny of DS0 because everything has to be converted back to DS0 everywhere in the network That's where companies like Sonus, Salix, TransMedia and Castle are focused With both voice over IP and voice over ATM there is too much discussion about voice bits riding for free. But telephony is not about bit transport, it's about services like call processing, billing and so on, and that's where Nortel and Lucent have great strength "

All of the vendors recognize that the market demands not boxes, but new solutions, while allowing carriers to preserve their collective trillion-dollar investment in the existing infrastructure With its recent announcement of the Succession

Network, which SBC, AT&T and France Telecom are testing, Nortel offers carriers a way to move telephony services to the multivendor ATM network that they are using today to provide data services, while leaving the door open to migrate the service to an IP network.

Skorupa praised the introduction of the Succession Network, saying this kind of decoupling of solutions demonstrates that companies such as Nortel and Lucent understand that, in the future, they may have to do business with customers that opt for another vendor's hardware "Nortel has a good story to tell because three or four years ago they took their software and rewrote it from the ground up to make it more modular and portable," Skorupa said.

Despite the pace of change, Skorupa said that service providers should not feel they have to make a choice between IP and ATM as an all-purpose platform "Convergence will happen in multiple stages Some people will build pure ATM networks; others are saying that if they can collapse six networks to two that's better than going to one because now they don't have to force fit everything," Skorupa said "So they're building a purely IP network optimized for IP and lighting one color [channel] on the [DWDM] network, and they're building an ATM network for everything else and lighting up another color. We don't believe this myth of convergence that says it has to be a single technology."

That means the European players still have a chance to get in the game in the United States, Silva said "Siemens, Ericsson and Alcatel maybe behind right now in terms of where Cisco, Lucent and Nortel are, but true convergence hasn't even begun. So it will be a race to see who gets products out and who signs up the accounts." Lucent's Oye agreed that there is a lot of change ahead "I could draw a map of different product segments that are supported today, but I think all of that is going to get thrown up in the air What protocols are going to be riding over the network of the future and the services that are delivered will see an awful lot of evolution I don't think there's a lot of people saying that any of the protocols that are around today will absolutely be the protocol five years from now"

Susan O'Keefe and Sam Masud are senior editors at Telecommunications.

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DLS AND THE ACCESS RACE

EXPOSITOR: M. en C. ARTURO ELIE ALALUF OLIVARES
PALACIO DE MINERÍA
JUNIO DE 1999

may 1999

International

DSL and the Access Race

Much is at stake in the emerging battle for fast network access provision in the US Unless the regional Bell operating companies (RBOCs) shake off their old ways of doing business and grab market share before their cable rivals finish upgrading their own infrastructure, digital subscriber line (DSL) technology could be another well-documented failure The question industry commentators are now asking is will the RBOCs turn DSL into another ISDN?

Patrick Flanagan

By rights, DSL should be manna from heaven for the RBOCs. DSL is a premium value-added service, there is little immediate competition, consumer and business demand is there, and the technology is available and proven. However, at the end of 1998, there were still only around 39,000 DSL subscribers, compared to 700,000 cable modem users. This is despite the fact that for every one line configured for cable modem services, there are 20 telephone lines installed that can support asymmetric DSL (ADSL), according to the Multimedia Telecommunications Association's 1999 Market Review and Forecast

It is hard to ignore the sense of deja vu that DSL conjures up when compared to ISDN (integrated services digital network). Look at these similarities:

• Confusing specifications DSL varies greatly in bandwidth availability, whether configured as ADSL, ADSL Lite, RADSL (rate adaptive), or IDSL (integrated), while ISDN comes as basic rate interface and primary rate interface;

• Slow rollout: When DSL will be widely available is anybody's guess, but deployment is simpler than ISDN because ADSL runs on overlay networks; therefore, the RBOCs do not have to integrate or upgrade existing switches or deploy combined signalling systems;

• Complex end user installation: It takes two technicians to install and fine tune DSL (one for the circuit provisioning and equipment, another for the modem). Installing ISDN took a number of visits before it was at least somewhat plug and play;

• Non-competitive pricing: The low-end pricing for plain vanilla DSL (192/256 kbps) is US$ 40 per month, a level set by the cable modem industry with no bearing on actual costs. To reach 1.5 Mbps downstream, the cost is US$ 60 to 100 per month. A primary reason for ISDN's US market flop was pricing, which originally was well over US$ 100 per month plus usage fees over a few hours. Today it runs at prices of between US$ 30 to 60 per month;

• Poor marketing: Although the public and small businesses seem to understand broadband technology better now, DSL must still be deployed, priced and packaged better than ISDN was. Looking at ISDN line deployment in 1997 and 1998, when demand grew considerably for higher speed Internet access, shows that few were buying. Total users went from just under one million to less than 1.4 million.

Limited Chances

The RBOCs have a limited window of opportunity. Most view cable modems as the primary threat, but the bidirectional hybrid fibre coax (HFC) infrastructure required to deploy them passes only 12 million homes worldwide, mostly in the US. Conversely, the RBOCs have ADSL-capable copper in virtually every office, home and school, and the number of DSL-passed customers in the US at the end of 1998 was more than 19 million, according to analysts, TeleChoice.

To duplicate this ubiquity, the cable companies need to spend billions. For example, upgrading a coax system from unidirectional to bidirectional requires new amplifiers that cost about US$ 25 per passed home. Upgrading from bidirectional coax to hybrid fibre coax can cost as much as US$ 200 per passed home. "There's room for DSL to pull ahead of cable modems, but it better happen by the end of 1999 or cable will take a commanding lead," said Claudia Bacco, senior DSL analyst for TeleChoice.

According to the Yankee Group, DSL consumer pricing cannot exceed US$ 40 a month, including ISP charges. This is the figure at which cable modems are currently priced with their apparently faster downstream data rates of 1 5 Mbps (however, users are on a 'party' line that degrades service as their neighbours join in) "Forty dollars is an aggressive level of pricing, and this means that the cost of deployment, including the CPE device cost, must come down to US$ 100 to 150," said Jim Wahl, a Yankee Group analyst. The hardware costs will reach this level by the second half of 1999.

The competitive local exchange carriers (CLECs) have a good head start, according to Wahl This is due to several factors. There are no T1 revenues to cannibalise, there are few legacy systems; and their infrastructure is designed for data delivery The largest advantage may well turn out to be the ability to 'cherry pick' or sign up those customers who want DSL and are willing to pay for it now. Until now, the primary CLEC emphasis has been on replacement T1 services, such as SDSL, HDSL and IDSL, in geographic areas with high densities of technologically advanced consumers and businesses. Covad and NorthPoint are two examples of CLECs putting this approach into action.

One strategy the RBOCs could employ to counter CLEC DSL cherry picking is to dramatically reduce the price of leased line services. The Yankee Group estimates that these cash cows have gross margins of 40 to 60 per cent "A 30 to 40 per cent price reduction would dramatically increase their attractiveness as an alternative to ADSL for small and medium businesses." Wahl said.

The Growing Cable Modem Menace

In metropolitan markets, DSL and cable modems will compete hotly for large numbers of customers which are anticipated to have low churn rates. The Yankee Group predicts that pricing for cable modems and RBOC or CLEC DSL services will be evenly matched. The deciding factor may well be aggressiveness, an area where the RBOCs have a poor to non-existent track record in the ISDN context. They are also running behind competitors. At the end of 1998, cable modems had a "significant, yet surmountable lead," said Wahl. There is momentum on the cable side as well, with the Yankee Group predicting that in 2002 there will be 4 3 million cable modems in use compared to 2.7 million DSL customers.

One often overlooked aspect of cable modems is their importance to the interexchange carriers (IXCs), which are loath to collaborate in any way with the RBOCs. The best capitalised of the cable modem providers is @Home, with a US$ 5 82 billion bankroll. AT&T owns 40 per cent of @Home. Microsoft's US$ 1 billion investment in Comcast is well known, and both it and Compaq are investors in Road Runner, @Home's biggest competitor

The Wholesale-Only Option

One uniform aspect of DSL among the RBOCs is that they all have a wholesale operation It is therefore no surprise that the most aggressive DSL vendors are little-known outfits such as Red Dog and City Access. Their primary customers are early adopters, who could just as easily buy from the incumbent local exchange carrier (ILEC) if it was making the same promotional effort Wahl believes the wholesale avenue is the most comfortable one for the RBOCs "To protect their legacy T1 leased lines, the RBOCs primarily want to target residential customers and they don't know how to do this Wholesalers do," he said

Significant wholesale revenues will come from IXCs offering DSL as part of a bundled group of consumer services In particular, Sprint's ION network will rely on ILECs or CLECs to provide local loop transport, which can produce considerable revenue with no advertising or promotional overhead This strategy has already resulted in agreements by Sprint with SBC, GTE, BellSouth and Ameritech for last mile ION connectivity, with trials due to begin in late 1999.

Handicapping the Players
While none of the RBOCs will reveal exactly how many DSL subscribers they have, one player is clearly in the lead. US West has deployed 80 per cent or more of the DSL lines currently in use -- as many as 30,000 according to some estimates, with 85 per cent going to residential customers. "They are rolling out DSL on a large scale, including a very high bit rate DSL that can more than compete with cable throughput," said TeleChoice's Bacco.

In the middle of the field are SBC and GTE. SBC's rollout is largely in California, where guaranteed bandwidth to the hub is also provided. Such quality of service (QoS) guarantees are rare for all forms of DSL, and Bacco noted that corporate users will be interested in the guaranteed rate service for remote access to LANs GTE currently offers DSL in more states than any other provider --16 -- and is believed to have the second highest number of lines in service. Ameritech has done little with DSL, but once its merger with SBC is completed it could become more aggressive. "SBC is somewhat aggressive in its DSL pricing at about US$ 50 a month and this will help Ameritech," said Wahl

At the back of the pack are Bell Atlantic and BellSouth There could be a new-found aggressiveness on the part of Bell Atlantic as a result of its alliance with America Online (AOL) "This means they don't have to worry about marketing," commented Wahl. It is hard to assess exactly what the impact of the Bell Atlantic/GTE merger will be, particularly since the companies are deploying DSL using products from two separate vendors "There's no natural fit, but deployment will continue because they can't delay," said Bacco. BellSouth has essentially removed itself from direct DSL involvement by establishing BellSouth com as an ISP to handle all DSL marketing. "This strategy gives BellSouth an opportunity to be aggressive in the residential market and stay one step removed," Wahl said

Avoiding Another ISDN
Industry consultants have numerous ideas on how the RBOCs can avoid another ISDN-scenario with

DSL:

- Make it cheaper: The objective is to grab market share. Profits will be there down the road for the provider with the largest DSL installed base. DSL will have a low churn rate because of the high up-front costs Had there been a large installed base of ISDN customers, the RBOCs would now have a lock on the DSL market;
- Let others do the marketing: AOL could save the RBOCs from themselves by providing the marketing and pricing needed to get the consumer DSL market on board There is a strong motivation for AOL to go in this direction. Cable modem users have to go with a cable-owned ISP such as @Home or RoadRunner. Every DSL subscriber AOL gets means one less customer lost to a competitor,

- Make DSL plug and play: Only US West is encouraging consumers to install their own DSL modems. If they do, it represents a saving of US$ 195.50 The real market for DSL will be in new computers that are shipped DSL-ready. For this to happen, buyers must be able to complete the DSL connection without a visit from a technician;
- Kill ISDN: This is the first step towards simplifying the number of broadband solutions on the market Free upgrades for existing ISDN customers as part of DSL rollout, as Bell Atlantic is doing, will keep the legacy broadband customer base loyal;
- Offer a simplified DSL portfolio: Killing ISDN is the first stage in creating a DSL product line with one set of pricing that cuts across the traditional consumer/business divide There is a blurring between consumer and business customers that makes two-tier pricing obsolete Small businesses in particular will respond to product offerings that are easily understood and fairly priced.

However, all this is a tall order for the RBOCs. If the DSL status quo remains, the cable industry will dominate the broadband Internet access market by as early as mid-2000 This is the first step toward providing an RBOC-less telecom services bundle. DSL is a strategic technology by which the RBOCs can protect their local services franchise. But they must act now.

Patrick Flanagan is a contributing editor with Telecommunications Americas edition.

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CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

NEX GENERATION TELCOS THRASH IT OUT

EXPOSITOR: M. en C. ARTURO ELIE ALALUF OLIVARES
PALACIO DE MINERÍA
JUNIO DE 1999

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Americas

Next-Generation Telcos Thrash It Out

There is a perception that it's not hard to be a next-gen telco, but the reality is more complicated. In fact, there's little agreement about what the essence of the next-generation telecom provider really is.

Lenore V Tracey

Marketing terminology that's meant to excite often does little to enlighten On the NextGen telco page at *www.pulver.com*, there are almost 70 next-generation telcos listed (as of Feb. 15, 1999), and they are ready to provide a variety of services across the globe. A linked page suggests how a company can become a next-gen telco But it's not so easy, say both the providers and the vendors Gordon VanderBrug, executive vice president of Massachusetts-based VIP Calling, suggests that "there is a perception that it's not that hard to be a next-gen telco All you do is buy a couple of gateways, plug them into the Internet and start passing minutes" But the reality is more complicated, and there is little agreement as to what the essence of the next-generation telecom provider really is.

The term next-gen telco is even confusing To some, a next-gen telco provides next-gen services. To others it means offering new or conventional services using advanced technologies New market entrants--CLECs, cable companies, utilities, ISPs, and others--often consider themselves next-gen telcos. To the incumbent telcos, next-gen companies are those that will ultimately define a new way of operating and doing business with consumers and companies across the globe: the one-stop shopping model.

Pulver com focuses on providers that offer voice services using voice over Internet Protocol (VoIP), once the province of hobbyists, but now, with the advent of new standards and platforms, a burgeoning market capturing much attention. The site's NextGen telco list includes start-ups, Internet telephony service providers (ITSPs), ISPs and incumbents.

Service providers such as Qwest Communications have positioned themselves as next-gen telcos, adding to their Internet service offerings products such as VoIP and voice over frame relay Qwest plans to complete a high capacity, IP-based fiber optic network by mid-1999. The company's stated mission is "to enable customers to seamlessly exchange multimedia content--images, data, and voice--as easily as traditional telephone networks enabled voice communication " Q.talk and Q.biz are communications services offered to the consumer and business markets, respectively. Q talk provides consumers with U.S. domestic long-distance service, paging, Internet access and prepaid calling cards Q biz offers switched inbound and outbound services, domestic and international calling, toll-free services, calling cards, directory and operator assistance, audio conferencing and broadcast fax

Targeting a different market, VIP Calling is in the wholesale business, carrying toll-quality minutes over a managed IP network--a carrier's carrier VIP's current competition comes from traditional, incumbent carriers of international minutes Without disclosing names, VanderBrug included more than half of the top 15 international service providers in his company's customer base He says VIP competes by using the best equipment, getting good IP connections (using major IP carrier backbones) and rigorously monitoring the network

The company has coined the term "assured quality routing" (AQR) to refer to all of the monitoring processes that allow it to deliver toll quality VanderBrug is well aware of the stakes in the wholesale market: "If we do not provide quality to our customers, they simply take us out of the routing tables."

New market entrants such as CLECs are also dubbing themselves next-gen telcos. For example, Lightship Telecom, a Manchester, N.H.-based CLEC, will enter the market this spring with a suite of services--some conventional, some advanced Rather than taking a technology focus, this company represents those new market entrants taking advantage of deregulation and bundling service sets targeted at specific markets. Lightship was launched in early 1998 and is currently certified to operate in New Hampshire, Maine and Vermont. According to Lightship COO Jeff Koester, the company will sell local, long-distance and Internet services to businesses (six to eight lines and above). It will initially provide local service by reselling Bell Atlantic minutes. Long-distance will be provided through a combination of the company's own facilities and agreements with established long-distance carriers to terminate minutes internationally and in domestic areas where Lightship does not have trunks.

Lightship's offerings will include a variety of other standard products such as pre-paid calling cards and conference calling. According to supplier Octave Communications' President and CEO Rob Scott, one of Lightship's key markets is the insurance industry Octave develops advanced conferencing products, and to complicate matters, calls itself a next-gen supplier. Lightship will use Octave products to terminate audio-conferencing minutes, responding to the insurance industry's high use of this business tool

Lightship is not next-gen in its transport technology, voice will initially be circuit-switched However, the bundling of local, long-distance and other services combined with differentiation in billing and customer service practices is where this new market entrant plans to distinguish itself.

Critics argue that many of these companies are still only offering conventional services or approaching the telco market in conventional ways. Hilary Mine, executive vice president at Probe Research Inc., offered several insights. Many companies claiming to be next-gen telcos look to her "an awful lot like the regular old thing."

"Qwest has circuit switches just like everybody else. They have traditional TDM in their network and they have acquired other companies that also have traditional networks as well as acquiring ISPs. They look pretty much like any other carrier. They've got traditional switching; they've got ATM switching; they've got overlay networks. And if the definition of a next-gen telco was that they were going to indeed define themselves in terms of new technology, then certainly Qwest has not done that." However, she added that Qwest has paid a tremendous amount of attention to network management. Its business proposition is that its network is cheaper to run and operate than any incumbent older network. And Mine believes Qwest will maintain a cost advantage.

In comparison, Level 3 Communications is sticking to the IP concept. Mine said Level 3 is "not really interested in voice. They will carry voice because they think voice rides free on their network." But Level 3 is interested in applications and has been driving the VoIP vendors to come up with the products it needs to stick to its IP vision.

Probe has a study underway to evaluate whether there is a true advantage to new networks. The multiclient study includes domestic and international carriers and major industry vendors. Probe is developing scenarios to determine the true cost of delivering services over an IP network versus a circuit-switched network.

A somewhat different approach that might be more worthy of the next-gen descriptor belongs to Enron Communications. A wholly-owned subsidiary of Enron Corp., the energy company, Enron Communications is building an IP-over-DWDM network and delivering services by partnering with ISPs and other providers throughout the United States. Although Enron's press releases state that the company "is committed to bringing next-generation services and applications to the desktop market," Jim Crowder, vice president of strategic development, doesn't consider his company a next-generation telco "Frankly, putting us in a category is not the easiest thing in the world to do," Crowder said, adding that he sees next-gen telcos as "incrementally different from the existing telco model--not radical, but incremental. We think Enron Communications is a radical departure from that traditional telco model."

Enron developed its business model by studying the needs of the enterprise and end customer--not by examining how to use existing assets and capital equipment. "Enterprises are looking for end-to-end solutions. They're frustrated by a lack of accountability and a lack of performance and a lack of guarantee and execution on end-to-end connectivity for applications. They have to go through an incredible labyrinth of infrastructure and companies to try to get various locations up and running, and then when they've done so, they are paying exorbitant rates for nailed-up circuits."

Responding to this need has resulted in the development of both a different architecture and business model. Crowder does not think that Enron's fiber build out is what makes it different. He called it a "me-too" strategy in terms of deploying fiber around North America What is different is the company's decision to deploy an IP network with no SONET, ATM or frame relay. Enron has consciously walked away from a lot of opportunities to keep its focus--building an architecture that responds to the pent-up demand for distributed applications

Looks Like a Duck, Walks Like a Duck

To the incumbents, the next-gen telco designation seems to have a much broader meaning "The next-gen company has a full service offering that is available in a simple, direct manner to the customer and provides an advanced set of capabilities and functionalities off the deployed technical base," said Ted Schell, senior vice president of strategic planning and corporate development at Sprint. "To do that you've got to have the advanced technology, the advanced networks, the operating systems, the provisioning systems, and the capability to integrate it in the back office to get the economies and the efficiencies you need to be the low-cost player. You have to make it very simple and very easy for the customer to use," he added.

Schell observed that customers are looking for features such as high quality voice, but don't care whether it is packet-switched or circuit-switched. Many people have defined the next-gen telco in terms of what companies such as Qwest are doing. "All those companies have done is placed a network bet on building a backbone infrastructure--a very high- speed infrastructure--and have a very, very limited product set in front of the customer today ...They may expand that product set, but as they do they will have to invest in operating systems. customer service support, billing systems, R&D, product development and so forth," Schell said

Corporate Culture

The incumbents and the new entrants recognize that in addition to new technologies and services, something else is very different about the companies aiming for next-generation telco status attitudes and organizational attributes. In January 1999, the head count at VIP Calling was in the 40s, up from about 12 one year earlier The company is keen on recruiting the "right" employees VanderBrug summarized the company's hiring strategy, saying that one crucial characteristic of the people it recruits is flexibility. The company seeks "people who understand how things are done either in the data world or in the telco world, but are not married to it." It looks for people who see voice becoming just an application on an IP network and who "know that sometimes you want to do things the way you have done them in the past and s: m·· mes there's a new way"

But this self-styled next-gen telco is not alone in articulating the need for the right employees to do the job and carefully crafting its organization Sprint cited similar attitudes Calling Sprint an "all-gen telco." Schell argued that with 62,000 employees, Sprint considers itself small and nimble Compared to some of its competitors, it may be But Schell knows that agility is more a matter of mind than size. He expressed Sprint's understanding of the next-gen mentality by saying, "It has to do with layers within the bureaucracy or how flat your organization is. It has to do with your intent and the skill with which you communicate to your people. It has to do with how compelling the vision is that you articulate "

Some of the new market entrants understand the absolute requirement that an organization execute its vision, and they feel their companies have an advantage in this regard by virtue of their smaller size. But that does not always prove to be the case

If you ask the suppliers that work with these service providers, they too have recognized the attitudes and attributes that differentiate the "MO" of the next-gen telco. "Are they really different?" asked Olwyn Walter, assistant vice president of the switching group at Newbridge Networks. "The short answer is yes. And more and more we're seeing incumbent service providers trying to model themselves after some of these alternates so that they can compete more effectively within and outside of their regions."

Unlike small and nimble Sprint, with 62,000 employees, AT&T is undergoing a corporate culture metamorphosis with its 108.000 workers Bob Annunziata, president of AT&T business services, joined the company as part of the merger of AT&T and Teleport Communications Group (TCG), where he was chairman, president and CEO. Annunziata launched Teleport in 1984 and grew the company from five employees and no revenue to 5000 employees and over $1.3 billion in revenue. Now he has returned to AT&T, where he worked for 17 years before TCG, bringing his start-up experience to benefit Grandma Bell Annunziata is working toward transforming AT&T's culture to reflect the need for responsiveness, advanced technologies and global reach.

AT&T doesn't take for granted any of the start-ups vying for telecom business either Instead, the company recognizes them as potential opportunities to compensate for its limitations by bringing in new technologies or expertise To get the transformation job done, AT&T has brought in new leadership in Mike Armstrong, acquired a facilities-based business company in TCG, bought a facilities-based consumer company in Tele-Communications Inc.

(TCI); began a global venture with BT, bought the IBM global services network for IP; and expanded its wireless footprint with Vanguard Cellular. In early February, AT&T announced a joint venture with Time Warner Inc. that will allow the company to enter the local-service market via cable in 33 states. And AT&T is not through yet.

How is Annunziata changing the culture at AT&T? "You get out and say here's where we're going, now together we have to do it. You talk to the people; you listen to them; and you give them the tools to help them do their jobs better." Although it is just in the beginning of the process, Annunziata said the company is already getting good results. AT&T is not simply protecting its territory, but growing the business.

Differences in market strategy and decision-making may also be characteristic of the next-gen telco "Typically, the new generation service providers will be very targeted and focused in their start-up mode of operation They will begin with a specific target market, and they will focus on being best in class at one or two types of service. They will also start out competing on price," said Newbridge's Walter.

Walter also characterized the new market entrants as quick on their feet. They don't have to worry about a legacy infrastructure or the problem of continuing to generate revenue from an installed base of equipment Walter added that operational support systems (OSSs), such as network management and billing systems, that can number in the 40 to 100 range for the incumbents, pose an additional burden, whereas new-generation service providers are starting from scratch. With technology that is available today, new entrants are in a very strong position to offer services at very competitive prices

AT&T's Annunziata acknowledged the claim that legacy systems will hold organizations like his back. But he added that no one should expect AT&T to sit back and not change. "The new organization won't allow that to happen, the new requirements from our customer base won't allow that to happen, and the new economics won't allow that to happen."

Mine predicts that for incumbents with existing equipment, it will still be a while before it becomes cheaper to deploy new equipment. However, she commented, "When I say that Qwest has a cost advantage, I think they have a cost advantage in the sense that their network is probably cheaper to operate and maintain. But AT&T may still have an overall cost structure advantage because all of their equipment and real estate is paid off."

Working the Market

Suppliers focus on helping incumbent service providers retain market share and cut costs. With new providers, the focus is on capturing and growing market share. Newbridge often finds these customers looking to it for help in identifying market opportunities that the equipment they are buying may make possible

That requires telco suppliers to bring on new expertise. In fact, Walter herself came to Newbridge with a 12-year history with service providers operating inside and outside of regulations. Others like her were also brought on. Many suppliers are doing the same thing, building up a telco provider skill set in-house or lining up consultants to provide these skills as needed. This business model seems to be forging closer relationships between suppliers and service providers

For the incumbents, similar trends are emerging, but for different reasons The combination of dwindling deep pockets and regulatory changes has brought about some departures from business as usual, Walter has observed Business units at the incumbents need to demonstrate their viability. "Cross-subsidization from one service to the next is generally not permitted in the competitive environment in which we're operating, particularly in North America," said Walter "So we are getting similar requests from RBOCs and from large international service providers--to help them with some of the front end business activities or help them prove-in services or business opportunities"

Without having to accommodate legacy systems, the next-gens can focus on telco equipment that supports the convergence of voice and data, leaving the company's options open and preparing for change right out of the gate. Newbridge and other suppliers with multiservice platforms are playing successfully to these companies. While the incumbents will be buying this equipment also, their focus is more transitional. Meanwhile, the new entrants are where the market is hot. And they appear to make decisions about equipment replacement very differently than the incumbents.

In the fall of 1998, VIP Calling moved its entire network over to Cisco equipment VanderBrug called the Cisco product line "the strongest in the marketplace now in terms of its reliability and robustness and in terms of the quality of the voice it [delivers]." VIP Calling actually replaced all of its equipment--not something that the traditional carriers, with their heavy investment in the legacy network, are inclined to do

Newbridge's Walter said that the only time she has ever seen a service provider replace equipment is when a vendor has gone in and actually bought out the old network Except for the step-by-step replacement of PSTN equipment from 50 years ago, service providers typically will keep equipment in their networks "It's a sunk cost, and if it's not broken, they won't do anything to change it They will figure out how to make investments going forward," said Walter

Octave's Scott offered additional insight into the market for multiservice platforms Octave calls its product a next-generation platform and although the first released ap lic ion is for audio-conferencing, it plans more releases to add different functionality throughout the year According to Scott, next-gen telcos are looking for ways to add applications to existing platforms without having to undergo forklift replacements of their equipment Savvy incumbents will follow suit.

Would the Real Next-Gen Providers Please Stand Up?

Which companies will be the real next-generation telcos, surviving the changes wrought by deregulation, competition, and the onset of advanced communications technologies? The applications and the market share may be the drivers of success, not simply the use of new technologies to transport voice or converge voice and data Some of the so-called next-generation telcos are really just new market entrants, trying to succeed with new platforms, new products and new approaches in the deregulated market. More effective organizations, integrated support systems and exemplary customer care are key factors in determining which companies will succeed

Mine speculates that the next-gen telcos might actually be the old-gen telcos. Referring to AT&T Jens, (cited in the Feb. 1, 1999 issue of The Pulver Report as the number one Internet telephony service provider in terms of international IP telephony minutes), Mine commented: "AT&T is the number one IP telephony service provider today, in terms of just generating minutes on its network. They are doing more minutes of IP telephony than any other carrier in the world What does that tell us?"

We'll only know in retrospect, when the dust has settled and the remaining companies and business models are still standing. There seems to be no question, however, that those that are left will have many different faces--large companies with broad market penetration, extensive product offerings and excellent OSSs, and small to mid-size companies with niche expertise or offerings in advanced technologies Strategic partnerships, mergers and acquisitions will continue at breakneck speed while companies vie to offer consumers and businesses new and better services, one-stop shopping and top-notch customer care, deploying the most advanced technologies to do so

Lenore V. Tracey is a consulting editor with Telecommunications.

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NEXTGEN TELCOS

NextGen Telcos & ITSPs

Last update. March 15, 1999

- AT&T Jens
- Glocalnet - Glocalnet is a next generation telecommunications company. The Company is mixing traditional telecom and new data network technologies to produce flexible and cost-efficient services. Currently, Glocalnet is deploying the largest pan-European VoIP network. Today Glocalnet has eight Points-of-Presence in Europe: Sweden, the Netherlands, Great Britain, Germany, Belgium, Italy, France and Switzerland.
- DeltaThree - Delta Three currently manages a network of 30 POPs worldwide, making it the world's largest international VOIP carrier. The company offers PC-to-Phone and Phone-to-Phone services with billing information available real-time through the Online Interactive Center.
- Telia Light - Telia Light AB is a fully owned subsidiary of Telia AB, Scandinavia's leading provider of telecom services Telia Light offers Telia branded VoIP services in Scandinavia and through partners in the rest of the world.
- Telnor Nextel
- Tele 2
- TeleMatrix - TeleMatrix is the largest VoIP network of its kind in Asia.
- VIPCalling - One of the largest and fastest growing networks of NextGen Telcos with 6 DS-3's and dozens of gateways. Carrying commercial wholesale traffic since 1997 Expanding through partnerships worldwide
- AlphaNet Telecom
- IncomTel TG - Gateway in the countries - Canada, Russia (cities- Moscow. St. Petersburg. Ekaterinburg), Ukraine (cities - Kiev. Kharkov), Byelorussia (Minsk). Provide pc-to-phone. phone-to-phone services for individual and corporate customer.
- GXC Com
- Net2Phone - With hundreds of gateways installed worldwide and millions of IP minutes routed monthly, Net2Phone's robust IP telephony network is growing through internal buildout as well as through its extensive reseller program
- Glocalnet
- POPTEL - We are one of the first ITSPs in Germany

offering high quality phone-to-phone Internet telephony. Our network includes nodes in Germany, Spain, China and the United States.

- Tokis - IP and PSTN Phone, Fax, Pager Gateways unifying Phone, Pager, Fax, Email,and Web based communication. Service and Software available. Nodes in U.S. and abroad include broadcasting, forwarding messages and voice and fax mailboxes.
- Net2phone Europe - Altern Telecom ITM is a gateway provider with an european office in France.
- Free World Dialup
- WIN
- INTERLINE - INTERLINE's principal focus is that of a Service Provider to the VoIP service industry. We use our own technology to deliver leading edge Voice over IP services - so we are in complete control of our destiny. Our Membership includes names like Mitsubishi, Hyundai, Hong Kong Telecom Canada, Concentric Networks, amongst others. These key industry players chose Interline over all others, as, quite simply, only Interline could provide a complete service solution. Backed up by a comprehensive in-house R&D program, Interline ensures that they, as service providers, stay at the leading edge of the market.
- Knowledge by Design - Knowledge by Design is a premier Global ITSP providing service and gateway solutions for enterprise to home user. With Pc to Phone as well as Phone to Phone capabilities,with full H.323 compatibility. OzTel covers over 50 countries with clients such as Hewlett Packard, Nortel, Telstra Australia, IBM and TNT.
- NetCall
- Pacific Telekey Network - PTN is an institutionally - funded "Nextgen Telco" focused on the North American and Asian markets. We are aggressively building a state-of-the-art-managed bandwidth Internet Protocol ("IP") communications network to provide and market telephony services to wholesalers, resellers and consumers at significant savings By combining our IP network with traditional telephony services, PTN is able to provide high quality, high value global communications services to our customers. We are also seeking partners worldwide
- CyberCall -
- Webcall GmbH - Webcall is specialized in Telehousing other ITSP's VoIP and FoIP gateways. We offer Telehousing services throughout EMEA (Europe, the Middle East and Africa). For ITSPs and NextGen Telcos using either Lucent, Cisco, Vocaltec or Array Gateways, we also provide local Termination services in the EMEA region
- ISPtel.com - IP Telephony Network deploying 50 nodes internationally. Global expansion through partnership alliances worldwide.
- Televideo Global Inc - Televideo provides high quality telecommunications services to carriers, corporations, and service retailers by utilizing leading-edge data telephony technologies across a managed global data

network.

- Net Communications Inc. - Progressive NextGen Telco that provides integrated voice, fax, data and multi-media services over its Global Multi Media Network, with POP's in the United States, Europe, South America, Africa and China.
- Planet Telecommunications, Inc.
- .comfax. The Internet Fax Company The leading Internet fax provider and only independent NetCentric-enabled network. Users can eliminate fax machines, fax servers, and phone lines. Faxes forwarded to email. Broadcast faxes are a snap. RSA security. Real-time billing & tracking.
- Cybertel Communications - Cybertel is one of the first ITSP in Australia, providing phone-to-phone Internet telephony service and is a member of ITXC WWeXchange Program.
- ADVANCE - We are one of the first ITSPs in Ukraine. We wait to terminate your calls to Ukraine.
- VCN - Global ITSP from Malaysia
- Net2Phone China - Exclusive Net2Phone Agent in China
- LATIC
- FLAT99 - FLAT99 is long distance calling at 99 cents per call. Talk as long as you want!
- RESCHA AG / EuroCall
- Inter-Tel
- NKO
- EconoFax - Global Internet Fax Services
- Franklin Telecom
- Networks Telephony
- Exicom - Internet Telephony Service provider mainly serving Lebanon and Syria through Pc-toPhone and Phone-to-Phone
- EscortTelecom - I would like to inform you that we are Vocaltec NextGen Partner Telco in Turkey
- Access Power - - Access Power's network currently provides complete coverage in the U.S. for PC to Phone calling Phone to Phone service is available from selected markets to anywhere in the U S. and Canada. Additionally, the company provides Carrier services and is developing markets Internationally through joint ventures
- CalTech International Telecom
- Net2Phone (S'pore) - Net2Phone (S'pore) is the exclusive representative for Net2Phone services in Singapore, Malaysia, China and the Asean region. Also offering PC-to-Phone
- Halidon
- SouthNet TeleComm Services SouthNet TeleComm Services, Inc. has VoIP Gateways in New York City, Atlanta, GA., Washington D.C. and Savannah, GA. In August, 1998 we will be adding 65 new U.S cities to our VoIP network.
- Internet Telephony Service Provider
- Rhinocerus Global Communications (RGC)- Provides

phone-to-phone and pc-to-phone services in Russia (Moscow, St. Petersburg. Sakhalin Island, Surgut, Tumen, Omsk etc.) and Germany (The whole country)

- Bcmfax.net is an international internet faxing network.
- Mexico city based Internet faxing solutions
- Tario - Tario.Net covers the following locations: USA; China; Cyprus; Georgia - Tbilisi; Russia - Moscow, S.-Peterburg, Novosibirsk, Irkutsk, Krasnoyarsk, Khabarovsk, Vladivostok, Samara, Togliatti, Ufa, Barnaul, Perm, Rostov.
- CT-T - Canadian NextGen telco. Has installed a demonstration VoIP link between Canada and the US. Listed on the Vancouver Exchange CUO or CUO.V
- IPTalk - ITSP to provide the service from America to China
- ISPhone - is a consortium of regional ISPs which offer voice and fax services.
- Iscom, Inc. - Communications for the next generation...
- USA GLOBAL LINK - USA Global Link is in the process of putting up 1000 internet telephony switches connecting them to its current global internetwork. It should have over 175 switches installed by Jan.1st of 1998.
- LD Exchange.com - LD Exchange.com, Inc. is a facilities based carrier providing wholesale worldwide terminations to Carriers and IP Telephony Providers.
- WebTelecom - The winners of ITXC and Pulver.com contest We provide Internet Phone Numbers that support calls from any H323 standard supportive software. Allowing web site owners to talk with surfers on-line, show them products and provide immediate technical support
- ViaNEXUS - New startup ITSP now offering PC-to-Phone service in the Eastern Mass./Boston area

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- Exchange Carrier in Latin America- Cti Brasil function as exchange carrier based in Brazil with service qeyed to Mercosul Latin America and twenty countries worldwide using Vocaltec Technology - facilitating communications worlwide.
- ITXC Corp - ITXC Corp operates the largest network of IP telephony service providers The network currently covers over 120 IP PoPs in 36 countries. ITXC sells minutes to carriers and resellers and pays gateway operators around the world to terminate the calls
- ITXC Ltda - ITXC Ltda offers wholesale internet telephony customers discounted calls to anywhere in the world ITXC Ltda is routing customer calls not to every phone in Brazil, but also has the ability to route to every phone in Mercosur and in the world.
- TransNexus
- AT&T Global Clearinghouse - AT&T Global Clearinghouse provides a complete, centrally managed solution that allows Internet Service Providers (ISPs)

and Telecommunications Authorities to easily establish and operate phone-to-phone IP Telephony services to more than 200 countries worldwide.

- Arbinet
- GRIC - GRIC Communications, Inc is a leading provider of integrated Internet Protocol based telecommunications networks and solutions to Internet Service Providers and telephone companies worldwide.
- VIP Calling
- DeltaThree
- Planet Telecommunications, Inc.

Bandwidth / Minutes Brokering

- Arbinet
- Band-X - The Bandwidth Exchange
- Min-X - The Minutes Exchange
- RateXchange - The Real Time Bandwidth eXchange
- SpotRates

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VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

DLS AND THE ACCESS RACE

EXPOSITOR: M. en C. ARTURO ELIE ALALUF OLIVARES
PALACIO DE MINERÍA
JUNIO DE 1999

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International

DSL and the Access Race

Much is at stake in the emerging battle for fast network access provision in the US Unless the regional Bell operating companies (RBOCs) shake off their old ways of doing business and grab market share before their cable rivals finish upgrading their own infrastructure, digital subscriber line (DSL) technology could be another well-documented failure. The question industry commentators are now asking is will the RBOCs turn DSL into another ISDN?

Patrick Flanagan

By rights, DSL should be manna from heaven for the RBOCs. DSL is a premium value-added service, there is little immediate competition, consumer and business demand is there, and the technology is available and proven However, at the end of 1998, there were still only around 39,000 DSL subscribers, compared to 700,000 cable modem users This is despite the fact that for every one line configured for cable modem services, there are 20 telephone lines installed that can support asymmetric DSL (ADSL), according to the Multimedia Telecommunications Association's 1999 Market Review and Forecast

It is hard to ignore the sense of deja vu that DSL conjures up when compared to ISDN (integrated services digital network) Look at these similarities:

• Confusing specifications DSL varies greatly in bandwidth availability, whether configured as ADSL, ADSL Lite, RADSL (rate adaptive), or IDSL (integrated), while ISDN comes as basic rate interface and primary rate interface;

• Slow rollout: When DSL will be widely available is anybody's guess, but deployment is simpler than ISDN because ADSL runs on overlay networks; therefore, the RBOCs do not have to integrate or upgrade existing switches or deploy combined signalling systems;

• Complex end user installation: It takes two technicians to install and fine tune DSL (one for the circuit provisioning and equipment, another for the modem). Installing ISDN took a number of visits before it was at least somewhat plug and play,

• Non-competitive pricing: The low-end pricing for plain vanilla DSL (192/256 kbps) is US$ 40 per month, a level set by the cable modem industry with no bearing on actual costs. To reach 1.5 Mbps downstream, the cost is US$ 60 to 100 per month A primary reason for ISDN's US market flop was pricing, which originally was well over US$ 100 per month plus usage fees over a few hours. Today it runs at prices of between US$ 30 to 60 per month;

• Poor marketing: Although the public and small businesses seem to understand broadband technology better now, DSL must still be deployed, priced and packaged better than ISDN was. Looking at ISDN line deployment in 1997 and 1998, when demand grew considerably for higher speed Internet access, shows that few were buying. Total users went from just under one million to less than 1 4 million.

Li ited Chances

The RBOCs have a limited window of opportunity. Mo view cable modems as the primary threat, but the bidirectional hybrid fibre coax (HFC) infrastructure required to deploy them passes only 12 million homes worldwide, mostly in the US. Conversely, the RBOCs have ADSL-capable copper in virtually every office, home and school, and the number of DSL-passed customers in the US at the end of 1998 was more than 19 million, according to analysts, TeleChoice

To duplicate this ubiquity, the cable companies need to spend billions For example, upgrading a coax system from unidirectional to bidirectional requires new amplifiers that cost about US$ 25 per passed home. Upgrading from bidirectional coax to hybrid fibre coax can cost as much as US$ 200 per passed home. "There's room for DSL to pull ahead of cable modems, but it better happen by the end of 1999 or cable will take a commanding lead," said Claudia Bacco, senior DSL analyst for TeleChoice.

According to the Yankee Group, DSL consumer pricing cannot exceed US$ 40 a month, including ISP charges. This is the figure at which cable modems are currently priced with their apparently faster downstream data rates of 1.5 Mbps (however, users are on a 'party' line that degrades service as their neighbours join in). "Forty dollars is an aggressive level of pricing, and this means that the cost of deployment, including the CPE device cost, must come down to US$ 100 to 150," said Jim Wahl, a Yankee Group analyst. The hardware costs will reach this level by the second half of 1999

The competitive local exchange carriers (CLECs) have a good head start, according to Wahl. This is due to several factors There are no T1 revenues to cannibalise, there are few legacy systems; and their infrastructure is designed for data delivery. The largest advantage may well turn out to be the ability to 'cherry pick' or sign up those customers who want DSL and are willing to pay for it now Until now, the primary CLEC emphasis has been on replacement T1 services, such as SDSL, HDSL and IDSL, in geographic areas with high densities of technologically advanced consumers and businesses Covad and NorthPoint are two examples of CLECs putting this approach into action.

One strategy the RBOCs could employ to counter CLEC DSL cherry picking is to dramatically reduce the price of leased line services. The Yankee Group estimates that these cash cows have gross margins of 40 to 60 per cent "A 30 to 40 per cent price reduction would dramatically increase their attractiveness as an alternative to ADSL for small and medium businesses," Wahl said

The Growing Cable Modem Menace

In metropolitan markets, DSL and cable modems will compete hotly for large numbers of customers which are anticipated to have low churn rates. The Yankee Group predicts that pricing for cable modems and RBOC or CLEC DSL services will be evenly matched. The deciding factor may well be aggressiveness. an area where the RBOCs have a poor to non-existent track record in the ISDN context. They are also running behind competitors. At the end of 1998, cable modems had a "significant, yet surmountable lead," said Wahl. There is momentum on the cable side as well, with the Yankee Group predicting that in 2002 there will be 4.3 million cable modems in use compared to 2.7 million DSL customers

One often overlooked aspect of cable modems is their importance to the interexchange carriers (IXCs). which are loath to collaborate in any way with the RBOCs. The best capitalised of the cable modem providers is @Home. with a US$ 5 82 billion bankroll AT&T owns 40 per cent of @Home Microsoft's US$ 1 billion investment in Comcast is well known. and both it and Compaq are investors in Road Runner. @Home's biggest competitor.

The Wholesale-Only Option

One uniform aspect of DSL among the RBOCs is that they all have a wholesale operation. It is therefore no surprise that the most aggressive DSL vendors are little-known outfits such as Red Dog and City Access Their primary customers are early adopters. who could just as easily buy from the incumbent local exchange carrier (ILEC) if it was making the same promotional effort Wahl believes the wholesale avenue is the most comfortable one for the RBOCs "To protect their legacy T1 leased lines. the RBOCs primarily want to target residential customers and they don't know how to do this Wholesalers do." he said

Significant wholesale revenues will come from IXCs offering DSL as part of a bundled group of consumer services In particular. Sprint's ION network will rely on ILECs or CLECs to provide local loop transport. which can produce considerable revenue with no advertising or promotional overhead This strategy has already resulted in agreements by Sprint with SBC. GTE. BellSouth and Ameritech for last mile ION connectivity, with trials due to begin in late 1999.

Handicapping the Players

While none of the RBOCs will reveal exactly how many DSL subscribers they have, one player is clearly in the lead. US West has deployed 80 per cent or more of the DSL lines currently in use -- as many as 30,000 according to some estimates, with 85 per cent going to residential customers. "They are rolling out DSL on a large scale, including a very high bit rate DSL that can more than compete with cable throughput," said TeleChoice's Bacco.

In the middle of the field are SBC and GTE. SBC's rollout is largely in California, where guaranteed bandwidth to the hub is also provided. Such quality of service (QoS) guarantees are rare for all forms of DSL, and Bacco noted that corporate users will be interested in the guaranteed rate service for remote access to LANs GTE currently offers DSL in more states than any other provider --16 -- and is believed to have the second highest number of lines in service. Ameritech has done little with DSL, but once its merger with SBC is completed it could become more aggressive "SBC is somewhat aggressive in its DSL pricing at about US$ 50 a month and this will help Ameritech," said Wahl.

At the back of the pack are Bell Atlantic and BellSouth. There could be a new-found aggressiveness on the part of Bell Atlantic as a result of its alliance with America Online (AOL) "This means they don't have to worry about marketing," commented Wahl. It is hard to assess exactly what the impact of the Bell Atlantic/GTE merger will be, particularly since the companies are deploying DSL using products from two separate vendors "There's no natural fit, but deployment will continue because they can't delay," said Bacco BellSouth has essentially removed itself from direct DSL involvement by establishing BellSouth com as an ISP to handle all DSL marketing "This strategy gives BellSouth an opportunity to be aggressive in the residential market and stay one step removed," Wahl said

Avoiding Another ISDN

Industry consultants have numerous ideas on how the RBOCs can avoid another ISDN-scenario with

DSL:

- Make it cheaper The objective is to grab market share. Profits will be there down the road for the provider with the largest DSL installed base. DSL will have a low churn rate because of the high up-front costs. Had there been a large installed base of ISDN customers, the RBOCs would now have a lock on the DSL market,
- Let others do the marketing: AOL could save the RBOCs from themselves by providing the marketing and pricing needed to get the consumer DSL market on board There is a strong motivation for AOL to go in this direction Cable modem users have to go with a cable-owned ISP such as @Home or RoadRunner Every DSL subscriber AOL gets means one less customer lost to a competitor.

- Make DSL plug and play Only US West is encouraging consumers to install their own DSL modems. If they do, it represents a saving of US$ 195.50. The real market for DSL will be in new computers that are shipped DSL-ready For this to happen, buyers must be able to complete the DSL connection without a visit from a technician;
- Kill ISDN: This is the first step towards simplifying the number of broadband solutions on the market Free upgrades for existing ISDN customers as part of DSL rollout, as Bell Atlantic is doing, will keep the legacy broadband customer base loyal;
- Offer a simplified DSL portfolio Killing ISDN is the first stage in creating a DSL product line with one set of pricing that cuts across the traditional consumer/business divide There is a blurring between consumer and business customers that makes two-tier pricing obsolete Small businesses in particular will respond to product offerings that are easily understood and fairly priced

However, all this is a tall order for the RBOCs. If the DSL status quo remains, the cable industry will dominate the broadband Internet access market by as early as mid-2000 This is the first step toward providing an RBOC-less telecom services bundle. DSL is a strategic technology by which the RBOCs can protect their local services franchise. But they must act now

Patrick Flanagan is a contributing editor with Telecommunications Americas edition.

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SURVIVAL OF THE CLECS IN THE NEXT - GENERATION PUBLIC NETWORK

EXPOSITOR: M. en C. ARTURO ELIE ALALUF OLIVARES
PALACIO DE MINERÍA
JUNIO DE 1999

april 1999

Survival of the CLECs in the Next-Generation Public Network

CLECs have a unique opportunity as data and voice networks converge. If they cannot rapidly deploy competitive networks coupled with solid business models, the competition could overwhelm them

Kurt Bauer

The demands of tomorrow's marketplace are driving today's development of the next-generation public network Because of the exponential growth of data traffic that is flooding traditional voice networks, the next-generation public network is in demand and those who can deliver it in conjunction with unique service offerings will be successful Booming technology advances, deregulation and the financial resources available in today's strong economy are allowing new market players to enter the fray.

Competitive local exchange carriers (CLECs) have a particularly unique window of opportunity as the next-generation public network is built, but immediate actions must be taken to ensure their longevity. Deregulation is intended to offer equal opportunity for all, allowing new competitors to thrive Many CLECs have taken advantage of deregulation, quickly implementing technologically-advanced, data-centric networks and are grabbing their pieces of the telecom pie by bundling a wide range of competitive services However, if they cannot rapidly deploy competitive networks coupled with solid business models, competition from other CLECs and incumbent carriers will overwhelm them

The New Paradigm

The telecom industry in the United States is not the first to experience widespread deregulation of the type facilitated by the Telecom Act of 1996. U.S airlines, banks, interstate trucking firms, long-distance telecom and natural gas pipeline industries are a few of the formerly-sedate industries that have recently been enveloped by a new wave of competition Nor is telecom the first industry to experience new competition at the same time a technology revolution is taking place.

If the new telecom paradigm follows the pattern of other deregulatory efforts, a series of trends can be expected to emerge. Those trends include.

- Industry boundaries crumbling and reforming as new types of companies and former non-competitors rapidly emerge in the market;
- Customer segmentation and loyalty growing in strategic importance;
- Elaborate new packaging, pricing schemes, geographic scale and product scope emerging as key differentiators,
- Unit prices falling, on average, and service quality dropping as costs are squeezed while companies struggle to maintain profitability at lower revenue levels; and
- Customer usage of all telecom services increasing dramatically.

Though pertinent to all service providers, many of these characteristics can be easily corre ited with CLECs. How they handle the adjustments today will affect their ability to thrive in the next-generation public network For example, collapsing boundaries are allowing carriers to attack local, wireless and Internet access business; cable television companies are entering the local phone business, and long-distance resellers are attacking international long-distance markets using Internet Protocol (IP) technology. The ability to offer a range of services is aiding similar growth and expansion of the CLEC industry

The CLEC Offering

Many CLECs are horizontally bundling a wide range of services and selling them to a restricted customer base, emphasizing sales channels rather than ownership of all the underlying network assets. Some data-centric CLECs, on the other hand, eschew such bundling in favor of a single-minded strategy. Some provide digital subscriber line (DSL) technology, while others offer local calling services on the theory that larger customers want to mix and match services from numerous providers. The new pattern can be viewed as segmentation and a game of well-defined niches. The emergence of wholesale and retail strategies, consumer and business focus, and the whole gamut of bundled services approaches are examples. Such institutional rearrangements seem to be a staple of the deregulation process.

The blurred lines between service providers are also creating business opportunities for CLECs with other service providers. For example, offering virtual points of presence (POP) managed port services to ISPs could be the next major data service revenue opportunity for CLECs. ISPs get the subscriber access ports and Internet backbone connections they need wholesale from the CLEC, which provides one or several virtual POPs for the ISP. The CLEC provides the ISP with a turnkey solution that includes the managed ports. Internet backbone connections and network management The ISP's traditional networking infrastructure is consolidated into a single IP connection (between the CLEC and the ISP) and the CLEC then charges a monthly network service fee based on the ISP's dial-up access port and Internet backbone connection requirements. Without deregulation, these types of services might not be possible, and much of the market that has allowed a CLEC boom would not exist.

The bundled services approach that is allowing CLECs to compete also demands a technology base to support many types of services Fortunately, telecom deregulation is taking place at a time of booming technology advancements Packet technology, including IP, ATM and frame relay, allows a wide range of traffic and therefore a wide range of services. The support of access switching capabilities and network management enhancements has helped round out the strong opportunity for CLECs.

To support the exploding growth of data traffic, as well as the existing need for voice support, vendors are continually upgrading their offerings with multiservice support and increased network management offerings. Everybody knows that data traffic will soon eclipse voice traffic in volume and legacy voice networks will not be able to support the traffic volumes coming their way. Much as long-distance deregulation thrived with the boom of fiber-optic technology, telecom will receive a boost from the ongoing upgrading of multiservice networks.

Today's strong economy, coupled with boundless opportunity, allows CLECs to build the technologically-advanced networks they need to compete. The economy has allowed new service providers to obtain the funding they need to build competitive networks. This actually gives CLECs an opportunity that incumbent service providers don't have By starting fresh, CLECs can better justify development of the data-centric next-generation public network from the bottom up Many established service providers are relying on legacy voice networks and have more difficulty justifying expenses necessary to replace or integrate their existing network infrastructure with new offerings Because CLECs are starting from scratch with the wealth of funding available for new business, they can start with today's best technology, giving them an advantage over existing competitors in addressing tomorrow's needs

The Challenge

Despite the advantages of deregulation, technology advances and a strong economy to support them, CLECs need to fight to stay alive in the next-generation public network Because of increased competition, service differentiators are essential to attracting customers away from other service providers. Most providers recognize the need to offer a range of voice and data services in order to compete as a full service provider Providing innovative new offerings such as voice over IP, quality of service and service guarantees can be used to garner attention as well as market share. Being able to control operating costs with state of the art management tools further ensures success in the competitive network services climate.

One fear that many CLECs face is consolidation and absorption of small players by larger companies. It is fairly easy to see that larger service providers, such as ISPs, could easily snatch up CLECs to avoid the fees involved in working with them as business partners. More likely, CLECs would acquire the smaller ISPs. CLECs have an interest in populating their switches and also in capitalizing on reciprocal compensation from the ILECs (as long as the provisions exist).

The bottom line is that a CLEC's success hinges on the ability to quickly grab its share of the market by delivering differentiated services over an all-digital network. In part, that is because all-digital networks are less expensive to build and operate, more flexible and a better platform for introducing high-value new services Integrated networks are expensive, so new service revenue growth is essential No single service provides an adequate return for an integrated, all-digital network Only multiple services and multiple revenue streams can do that--and tomorrow's market is already demanding those services Disproportionate rewards will accrue for those CLECs who move quickly to occupy positions in the data-centric new world n

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TEMA

LATIN AMERICAN TELECOM CHANGES: LEARNIG FROM MÉXICO AND BRAZIL

EXPOSITOR: M. en C. ARTURO ELIE ALALUF OLIVARES
PALACIO DE MINERÍA
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Americas

Latin American Telecom Changes: Learning from Mexico and Brazil

Telecom developments in these two countries could have a profound effect on the rest of the world.

Norman C. Lerner

This time last year there were so many telecom activities and opportunities in Mexico and Latin America that it was difficult to keep track of them all! In the past year, many expected events have come to pass with exciting results. Brazil privatized Telebras and awarded its B-band cellular licenses; Mexico completed its wireless PCS auctions; Chile approved the CTC acquisition and consolidation of long-distance carrier VTR, Argentina decided to eliminate the Telefonica and Telecom Argentina monopolies in 1999, Peru and Telefonica agreed to end the national monopoly a year early and authorized several long-distance competitors, Venezuela elected a new president, hoping to reverse its economic slide, and Ecuador and Paraguay are again talking about privatizing their national carriers

Despite the good news, however, Brazil tripped over its economic shoelaces and nearly took its friends and neighbors with it. The damage appears modest--for now. Perhaps it is because world economies have experienced similar financial crises in Asia, Russia and Mexico (in 1995) and survived Nevertheless, the Brazilian currency devaluation in January 1999 has provided, so far, only a sobering realization that economic/financial and political risks in developing countries are real and significant.

The economic forecasts for 1999 that were developed before Brazil's devaluation had already reflected a general decline in regional growth Now, however, it is worse, with some forecasts predicting a decline of up to 6 percent in Brazil's economy. The simple reason for pessimism is that Brazil is the eighth-largest economy in the world and accounts for 40 percent of Latin America's economic output, which makes it the problem of Argentina, Venezuela, Chile and Mexico as well As 20 percent of U S exports go to Latin America, these problems are U.S. problems, too.

It is within this fragile economic framework that the telecom evolution (technological, regulatory and competitive) is still occurring successfully, and regional viability of telecom growth seems reasonably secure. So far, it appears that the growth momentum will be only somewhat slowed. In this regard, it is instructive to focus on two countries, Brazil and Mexico, to identify the trends and issues that can help the Latin American countries ensure continued growth

These countries can be viewed as representing opposite ends of the development spectrum the frenetic regulatory and investment activity in Brazil contrasts the more mature development of competition and licensing in Mexico Economic issues notwithstanding, these two countries represent the focus of technological, regulatory and market development from which other countries in the region--from the few still working toward privatization to those already intensely competitive--can learn what to emulate and what to avoid

The Brazilian Challenge

In addition to its financial problems, Brazil still leads the way in addressing issues related to technology, regulation and competition. It was just last year that Telebras was privatized for $19 billion, establishing three regional wirelines (Tele Norte Leste, Tele Centro Sul and Telesp), plus the long-distance international Embratel and nine cellular A-band spinoffs. Almost simultaneously, the B-band cellular licenses were awarded, and plans were formalized for auctioning three licenses to mirror the three wireline Baby Bras, plus Embratel. The consolidation of ownership, operations, regulation and competitive relationships is pervasive and continuing.

Two of the four mirror licenses were awarded in January to the only bidders: The Bell Canada International (BCI) consortium (16.2 percent Qualcomm interest) promised almost $50 million, payable over two years, to acquire the license and spectrum for the 20-year right to compete in the northeast region against Tele Norte Leste; and a National Grid consortium (U.K. 50 percent; Sprint 25 percent) paying about $45 million to compete against MCI-Embratel, which was privatized earlier for $2.4 billion. NG-Sprint claims that it will build a new network covering 38 cities in one year and "most" Brazilian customers within three years. More recently, sale of the remaining two mirror licenses has been rescheduled for April 23, due to requests for more time by newly interested bidders Qualcomm do Brasil, Bahtel Engenheria and Splice do Brasil.

Each mirror company will be allowed to compete in the specifically defined Baby Bras' area in a controlled duopoly through the year 2000. Unlike the Telebras spin-offs, the mirror companies will be unrestricted with wireless coverage areas and tariffs (BCI is planning mostly fixed wireless access), although they will have to meet certain minimum service targets. However, success of the mirror companies will depend on the ability of firms to compete with organizations that acquired the Telebras companies Telefonica, Telecom Italia, Telecom Portugal and MCI WorldCom, the latter already rumored to be developing a pan American alliance with Telefonica and Telecom Portugal to consolidate their positions The mirror companies must address both A- and B-band cellular operators, future PCS and LMDS licensing, and the satellite entrants that are almost on-line. This is happening even while the incumbent wireline companies are expanding their current 10.8 percent teledensity about 16 percent per year for the next three years

(see Table 1). These examples can only highlight the breadth and depth of the momentum in Brazil, amplified by even more dynamic wireless activities. Brazil is forecast to have 23 million cellular subscribers by 2007, accounting for more than 42 percent of the almost 72 million projected for the region as a whole. **Table 2** shows the most recent number of cellular phone subscribers throughout the region. As expected, Brazil still has the highest number, fueled mainly by the waiting list developed over the last few years.

Table 1
Country Teledensity

Country	1997	1998
Argentina	19.3%	20.4%
Bolivia	5.3%	6.0%
Brazil	8.0%	10.8%
Chile	17.2%	18.7%
Colombia	13.0%	15.4%
Ecuador	6.9%	7.7%
Mexico	9.6%	9.8%
Peru	6.4%	6.9%
Uruguay	25.3%	27.8%
Venezuela	11.0%	12.6%

Source: ITU

PCS and wireless local loop/fixed wireless access (WLL/FWA) activities help account for the continued growth In PCS, Chile is currently in first place, beginning operations last year with Chilesat's 1 9-GHz CDMA system. Mexico and Paraguay have just awarded licenses, and authorizations to provide service have just been issued in Mexico Brazil, Argentina, Colombia and Venezuela plan to allow market entry before the year 2000.

In the near term, cellular radio facilities and services, particularly in Brazil and Mexico, are being expanded rapidly, to the benefit of equipment manufacturers and potential subscribers In Brazil, for example, the big news continues to be BellSouth, leader of the BCP Telecomunicacoes consortium in the city of Sao Paulo, which pai more than $2 5 billion for its B-band cellular license. Since becoming operational last May, it already has more than 1 million Nortel TDMA subscribers and expects to have over 1 2 million subscribers in less than one year of operation

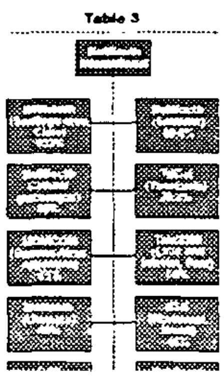
Table 2
Mexico and South America
Cellular Radio Subscribers
Fourth Quarter [illegible]

Argentina	1,732,200
Bolivia	190,700
Brazil	4,438,105
Chile	600,800
Colombia	1,890,000
Ecuador	107,250
Mexico	2,279,600
Paraguay	113,500
Peru	600,804
Uruguay	111,000
Venezuela	1,300,800

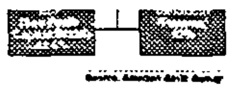
Source: U.S. Department of Commerce/[illegible]

Telesp Celular, the Telebras spin-off competing with BCP, is busy adding capacity with its new owners, Portugal Telecom and Telefonica; the latter also owns the Sao Paulo wireline carrier. Telesp Celular expects to add 800,000 subscribers this month, raising its total base to 1.7 million Previously using IS-136 TDMA, it has now awarded an IS-95 CDMA contract to Lucent Technologies. At the same time, Qualcomm do Brasil has a $35 million contract to provide 800-MHz dual mode CDMA digital handsets, making Telesp the first Brazilian carrier to use the digital/analog phone

Shortly after the Sao Paulo start-up, BellSouth also launched its TDMA service in Recife, a city of about 2.6 million people in northeastern Brazil, where only 5 percent of the residents have telephones and less than 2 percent have cellular phones Both licenses provide potential coverage for more than 44 million people BellSouth is also showing its confidence in Brazil by recently increasing its ownership in both systems and its potential purchase of 25 percent ($850 million) of Tele Norte Leste, the Telebras spinoff that covers an area approximately two-thirds the size of the United States **(see Table 3)**



However, continued high subscriber growth rates in the region must be assessed within the framework of increasing competition and changes in each country's economic framework For example, BellSouth's exceptional growth has virtually eliminated the backlog that fed subscriber growth. In addition,



subscriber growth. In addition, Telesp is organizing and pricing its services to be more competitive. At the same time, Brazil's financial difficulties will likely force BellSouth to reduce its current $265 sign-up fee and $200 phone. The combination of these factors will likely reduce subscriber growth, operator revenues and facilities expansion, and increase financial pressures, at least in the near term.

Globo, the Brazilian conglomerate, is an early example of such pressure. It sold its share of two cellular companies: Tele Celular Sul and Tele Nordeste Celular, to Telecom Italia for $113 million to raise cash for other investments, and it sold its share in Tele Centro Sul, the wireline carrier, back to Telecom Italia because revenues were declining Other established operators such as Lightel are seeking partners and ways to refinance their expansion plans in the wake of the devaluation At the same time, Telefonica, Portugal Telecom and Iberdrola are prepaying the government $3.6 billion for their Telebras purchases now instead of in the year 2000, demonstrating a strong vote of confidence in Brazil's stable future.

Under the best of circumstances, the Brazilian government would be strained to meet the financial and regulatory requirements that continuous telecom expansion dictates. The current financial crisis makes the implementation even more tenuous Nevertheless, investor groups appear to be only modestly deterred, refinancing existing commitments as well as attempting to assist the government directly.

Momentum in Mexico

While Brazil has maintained the limelight for the past year with its marketplace dynamics, Mexico has been demonstrating the more methodical competitive, regulatory and technological integration activities that follow privatization. The focus for these changes and expectations is on the entrenched giant Tel□fonos de Mexico (Telmex), privatized in 1990, and COFETEL, the federal telecom commission charged with regulating Telmex and the telecom sector.

Since long-distance competition with Telmex was initiated in 1997, AT&T and MCI WorldCom, the two largest of the 10 competitors, have invested more than $2 billion in Avantel and Alestra, respectively, and have successfully acquired about 25 percent of the Mexican long-distance market This year, competitors' network expansion will require Telmex to interconnect with them in a total of 150 cities, most relying on available Telmex facilities rather than independent construction. Despite Telmex's aggressive behavior toward its competitors, long-distance competition is continuing. To the extent that COFETEL moves away from any incumbent protectionism, Telmex will continue to face strong competition for long-distance services.

One example is the excessively high domestic and international interconnection fees Telmex charges its competitors. Since the inception of competition, Telmex's international call connection rate charges have been subjected to a 58-percent surcharge on all incoming calls to support local services and network expansion In January, COFETEL announced elimination of that surcharge and reduced the rate to 2.6 cents per minute (the same as the Telmex domestic interconnect rate for fixed and mobile), but only until Dec 31, 2000 This rate is still being contested as too high and not cost-based, as compared to those in the United States and many other jurisdictions.

Apparently Telmex has studied the obstructive activities AT&T itself employed after divestiture in the United States and, ironically, Telmex is now operating successfully within the U S In a bold move, Telmex has successfully teamed with Sprint (TSC) to secure FCC authorization to enter the domestic U S market, despite opposition from AT&T and MCI WorldCom Telmex is already consolidating its position in the southwestern United States with service in Tucson, Ariz , and is scheduling four more cities immediately There are about 50 million U.S residents with Hispanic connections and the U.S -Mexico market is worth about $2 billion annually; Telmex wants to get 5 percent to 10 percent of that market in less than a year

Table 4
Mexican Cellular and PCS Forecasts

Mobile service market	1997	1998	1999	2000	2001
Subscribers (000)	1750	[illegible]	[illegible]	[illegible]	7000
Growth (%)	47.4	92.6	[illegible]	[illegible]	15.7
Annual revenue ($ millions)	842	1320	[illegible]	[illegible]	3042
Growth (%)	38.2	56.9	[illegible]	[illegible]	17.4

In another successful overseas activity, Telmex acquired an operating contract with Luca, the consortium that won control of Telgua, Guatemala's main long-distance and international carrier. It now has an option of buying up to 49 percent of Telgua, with a contract providing revenue and net profit sharing Telmex also has purchased controlling interest in U.S -based Topp Telecom, a developer of prepaid wireless products, for $57.5 million These international extensions likely reflect the confidence engendered by its relatively secure revenue sources and protected competitive strength at home

Locally, COFETEL has been very slow to introduce competition in local phone service, even though the cost to consumers increased more than 63 percent in 1997 Telmex's local dominance, however, will be challenged by new wireless networks, both mobile and fixed COFETEL claims that local service competition will begin in Mexico this year with the adoption of Federal Telecom Law Article 63 Currently, however, the lack of specific local service rules coupled with Telmex's local dominance have precluded competition in these markets that are, theoretically, open to competition.

One set of forecasts for combined cellular and PCS subscribers is shown in **Table 4**. The PCS license winners, which paid $432.4 million for the concessions, are shown in **Table 5** At the end of last year, all except two groups received COFETEL authorization to offer service Unefon and Midicel are delayed until their license payments are made. Additionally, all the carriers must negotiate separate interconnect agreements with Telmex Nevertheless, the most recent estimates are for cellular/PCS growth at 90 percent for the year, achieving a 7-percent penetration At the same time, Telmex invested $400 million in 1998 to expand its long-distance and local networks to achieve a 15-percent fixed-line density by 2000 The objective is for a total teledensity of 25 percent in five years.

Fixed wireless access is also expanding. Telcel, for example, with over 1.8 million cellular subscribers, is deploying rural public fixed wireless terminals in the interior regions of Mexico to serve 8.5 million people who currently lack basic telecom services. Additionally, Iusacell, the largest independent cellular provider, is committed to a $100-million network expansion and is planning to award a $60-million contract to Lucent for the construction of CDMA service this year in the golden triangle area of Mexico City, Guadalajara and Monterey. At the same time, Iusacell must compete more intensively with Telmex's Telcel, prospective PCS entrants such as Qualcomm's nationwide CDMA systems, and Sistemas Profesionales de Comunicacion's (SPC) Nortel system, the latter to be built over the next five years for $590 million

Within this framework, cellular carriers seek to maintain subscriber growth by advocating COFETEL approval of calling party pays. This has been adopted in nearly all of Central and South America, except for Mexico and Brazil, resulting in substantial subscriber increases. However, its implementation is being delayed at COFETEL because of objections from Telmex Telmex claims that, since cellular subscribers are increasing substantially each year, the additional stimulation that calling party pays would bring to the cellular market is unnecessary Furthermore, Telmex claims, that it will be necessary to charge its 10 million fixed line customers about 23 cents a minute, approximately 15 times more than they now pay for originating calls to mobiles, to cover the cost of implementation Nevertheless, COFETEL appears likely to approve the plan, but probably at about 18 cents a minute[4]

Table 5
Mexican PCS License Winners

Company	Bid Total (in millions)	Coverage
[illegible]	[illegible]	National
Pegaso	[illegible]	National
Telcel (Telmex)	[illegible]	National
[illegible]	377	Four regions
Iusacell	[illegible]	Two regions
Grupo [illegible]	52	One region

Source: [illegible] Communications Research Center

Current subscriber growth rates, manufacturers' contracts and the battle between CDMA and

TDMA are only results reflecting prior activities in wireless telecom. These statistics, however, suggest the next round of major issues, particularly regulation, pricing and other competitive factors. Mexico, with its COFETEL-Telmex competition issues, is an example of the conflicts that occur with the saturation process Brazil and others will soon encounter.

Ahead: Continued Growth

The current industry dynamics and uncertainties make it necessary to identify several important trends and anticipate issues that can foster regional telecom development in Mexico and South America. The current attempts at austerity in Brazil, if successful, will likely stabilize the currency and the overall economy. However, austerity may slow growth Since Brazil represents almost 50 percent of all economic activity in Latin America, a Brazilian recession will lead to further economic deceleration in the other regional countries, particularly Argentina, Brazil's most important Mercosur trading partner.

One country that could potentially avoid the financial problems and reduction in telecom growth for the immediate term is Mexico. In November, Mexico surpassed Japan as the United States' second-largest trading partner after Canada Although economic growth is expected to be lower, at 3 percent for the year, it is quite good compared to its southern neighbors

The momentum for telecom growth in Brazil--in terms of number of subscribers, scope of contracts awarded and magnitude of private investment--appears to be sufficient to continue albeit at a somewhat slower rate, if the worst case is only stable, lower economic growth The investors and operators are continuing their programs and even assisting the government directly, based on this assumption. Other countries in the region are proportionally in the same situation

Despite the claims by various suppliers (Ericsson claims 50 percent of all B-band cellular in Brazil), many manufacturers are now sharing in the wireline and wireless growth in Mexico, Brazil and other countries. It should be expected, however, that certain technologies will soon show their preeminence (CDMA vs. TDMA) as will some preferred manufacturers (Lucent with Telefonica).

Technology applications, as well as their accessibility, pricing and regulation, are the next set of issues to be addressed for continued growth in the region, regardless of the near term economic problems. History and the examples of Mexico and Brazil demonstrate the importance of strong regulatory management while competitive markets are emerging. Fostering innovative technology and achieving competitive market benefits depend on the extent to which regulatory authorities in privatizing/liberalizing countries are motivated and able to control their incumbent carriers.

Current statistics do not reflect the issues that have yet to affect the marketplace. These include, for example, the extent to which cellular, PCS and WLL/FWA operators will increase their data transmission capabilities to encourage Internet usage-based growth Similarly, although very high bandwidth satellites such as Teledisic are not yet available in Latin America, other suppliers such as Iridium, Globalstar, Satmex and PanAmSat are arriving rapidly with diverse capabilities. The manner and extent to which these systems will be regulated and are competitive and/or complementary to terrestrial facilities and services--both wireless and wireline--are yet to be determined

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FACULTAD DE INGENIERIA U.N.A.M.
DIVISION DE EDUCACION CONTINUA

CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

THE 10 HOTTEST TECHNOLOGIES

EXPOSITOR: M. en C. ARTURO ELIE ALALUF OLIVARES
PALACIO DE MINERÍA
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The 10 Hottest Technologies

No flash in the pan-- Here are 10 technologies with real sizzle

Every month, Telecommunications editors scan the horizon for important trends, emerging technologies and exciting ventures that will change the nature of telecom services. But only once each year do we really try to enumerate the leading technologies we believe will quickly enter the telecom mainstream and possibly precipitate a sea change in communications infrastructure. This year was tough. So much is happening that the contenders vied closely to make the top 10

As usual, we spoke with industry pundits to gain a wide variety of perspectives This year we also asked them to expound on their choices for 1999's hottest technologies--which often differed from our own top 10. John Morency of Renaissance Worldwide, Larry Swasey from Allied Business Intelligence Inc., NetReference's William Flanagan and Brett Azuma of Dataquest are among those who shared their top picks for emerging hot technologies

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Experts Tell Us
What's Hot

10 Hottest Technologies 1998

1 Multilayer Ultrafast Routing/Switching

2 Erbium Doped Fiber Amplifiers

3. Packetized Voice
4. LMDS for Broadband Interactive Services
5. Wireless Local Loop
6. Internet VPNs
7. Web-Enabled Call Centers
8. eXtensible Markup Language (XML)
9. Telecommunications Management Network (TMN)
10. Java-Based Network Management

How We Decide What's Hot and What's Not

For all who participated in the selection process, we set down these guidelines for qualifying as a hot technology. Hot means capable of entering the mainstream of telecom systems/operations within one to two years, while now having sufficient development dollars and industry support to become economically viable. Telecom is defined broadly to encompass cable, interactive media and other emerging forms of

The top 10 this year seem to reflect the explosion in wireless communications, the convergence of voice and data networks, and the focus on speed and access. Here, in no particular order, are *Telecommunications*' picks of the 10 hottest technologies for 1999.

1. Service Mediation
2. Metro DWDM
3. OC-768 SONET
4. HDSL2
5. Unified Messaging
6. Policy-based Management
7. MPLS
8. Smart Antennas
9. Optical Switching
10. Wireless Location

Service Mediation: The New PSTN

Informed opinion holds that the PSTN isn't going away anytime soon. Converged voice/data networks still represent telecom Xanadu for many carriers that can't compete cost-effectively with parallel networks. Instead, the leading switch vendors, start-ups and veterans alike, have looked to bridge the voice and data worlds through SS7 gateways and protocols such as the Media Gateway Control Protocol (MGCP). The goal is to enable enhanced voice/data services for business customers, but given the abundance of disparate network infrastructures and languages out there, creating a universal language for network convergence is a complex problem. Several start-ups, including TransMedia Communications, Salix Technologies, Sonus Networks and Castle Networks, have announced switches that handle both voice and data traffic and provide technology-agnostic switching or, more commonly, service mediation. Telcos are interested in this technology for two reasons. Service mediation helps enrich voice services and allows service providers to easily and quickly create and add new customized billable voice features for both revenue and differentiation. The Yankee communications, as well as the standard inclusion of voice, data and video transmission. A technology is not a product, but a product can execute a technology. To qualify, a new technology also has to have the potential to make an impact on existing systems, operations or procedures

Group estimates this market to be worth more than $1.5 billion by 2002.

Tom Nolle, principal at CIMI Corp., offered an historical perspective: "In today's PSTN, we built a service network, meaning that the service features of telephony are created by network elements that also handle the call path. Future networks will not be built that way; we need service-independent transport. Furthermore, future networks will probably be based at least initially on diverse technologies as service providers build out their infrastructures to suit their primary service targets. As a result, we will have networks that speak a language of connection (e g., IP sessions, ATM Q.2931) different from the service language of the user (e g , analog DTMF, H 323). The thing that translates between service languages, meaning the signaling protocols, to create a service between diverse users across diverse infrastructure is what is referred to as mediation."

These switches, though differing in target applications and functionality, act as intermediaries between networks and users for receiving service signals and creating connections They read and interpret user service requests and deliver those same services to the user through whatever feature implementation is convenient. TransMedia refers to this functionality as "media awareness" Salix prefers the term "class-independent switching," since the network features are equally available at all points instead of separated according to a network hierarchy as with Class 4 and Class 5 switches. With this architecture, any feature can be delivered to any service point, even if moving between networks with different service signaling languages Because these platforms in effect "unbundle" voice switching from Class 4 and/or Class 5 equipment, service providers can quickly expand into regional markets for far less than the price of deploying new voice switches.

Products using service mediation technology are largely in beta trials now, primarily with CLECs, ILECs, IXCs and next-gen telcos such as Qwest, Williams and Level 3 "As we deploy our new network, we require scalable media gateways that combine voice over IP, circuit-grooming and packet-switching capabilities, and also support the emerging MGCP," said Isaac Elliott, senior director of the voice network at Level 3 Communications, a Salix customer. "This technology will allow us to provide our customers high-quality, carrier-class telephony services over our IP-based network at attractive price points."

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Metro DWDM: Close to Home

In order to squeeze more bandwidth out of their fiber networks, long-haul carriers are deploying dense wavelength division multiplexers (DWDM) to build backbones that might have dozens of channels riding on a single strand of fiber, with each channel operating at multigigabit speeds. Now the DWDM equipment vendors are moving their technology away from the wide area network (WAN) and positioning it in the metropolitan area network (MAN) The target customers for metro DWDM technology from companies such as Nortel, Ericsson, Sycamore, Ciena and Osicom are the ILECs and CLECs, cable companies, wireless network operators and even enterprise customers with dark fiber

Installing metro DWDM does not necessarily require replacing existing SONET networks because the two can live side by side with metro DWDM actually carrying SONET traffic Instead, the idea is to better utilize the raw bandwidth offered by DWDM technology by making the multiple wavelengths supplied by metro DWDM the platform for delivering services, whether these are SONET, IP or ATM Metro DWDM does this because it supports a variety of protocols in their native formats Thus metro DWDM systems not only can accept traffic from devices supporting SONET interfaces such as ATM switches and routers, but also accommodate other protocols such as Gigabit Ethernet (GigE), FDDI and ESCON Mike Guess, vice president of engineering for IXC Communications, said his personal view is that "Metro DWDM is the only place where you'll find widespread implementation of non-SONET signals on fiber, because in long-haul networks it is pretty expensive to tie up a wavelength with a Gigabit Ethernet signal."

Bell Canada's network and technology Vice President Bao Le, who is conducting lab tests of Nortel's equipment (known as OPTera), believes there is a powerful argument for deployment of metro DWDM in ILECs' and CLECs' interoffice rings. "As our customer demands grow, we can't just dig up sidewalks and parks to put down more fiber. So what do I do when my OC-48 or OC-192 SONET bandwidth runs out?" Le asked. The answer, he said, is to use existing spare fiber for metro DWDM or turn down a SONET network and use that fiber for metro DWDM, since the latter, unlike SONET, provides a growth path by allowing channels to be added as needed. Either in a point-to-point or ring configuration, metro DWDM provides protected wavelengths. For example, a metro DWDM system might support 16 protected wavelengths or 32 unprotected wavelengths. Some systems may offer protection switching on a wavelength-by-wavelength basis. The benefit of this is that payloads that have their own protection scheme, like SONET, can ride on unprotected wavelengths while other traffic, such as GigE, can be put on protected channels. Metro DWDM systems from different vendors will vary in the number of milliseconds they take to restore connections, and there will be differences in ring size--ranging from under 100 kilometers to hundreds of kilometers--supported by a particular metro DWDM system. "[DWDM] metropolitan area networks--the links between carrier switching centers, ISPs and corporate networks--will require investment from the carrier to better support the influx of data traffic entering the public network," said Mat Steinberg, RHK's director of optical networking

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OC-768 SONET: Fatter Is Better

Just when network engineers thought it was safe to rest at OC-192 capacity, along comes OC-768. True, the next step in bandwidth capacity isn't here today--analyst estimates predict 40-Gbps links won't be a reality until at least 2001. But according to Dana Cooperson, senior analyst at RHK, the last six months have been encouraging, and 40-Gbps transmission looks a bit closer than expected based on recent announcements by Alcatel and Lucent. "Though RHK had forecast 2002 as the earliest we would see OC-768 deployed, this prediction was based on the technical and cost challenges related to making OC-768 viable, as well as the difficulties foreseen in transporting that data rate over existing cable plant," she said. But if recent lab testing by carriers (MCI in the case of Lucent and Deutsche Telecom in the case of Alcatel) goes forward with continued success, we may see 40-Gbps sooner. "The recently announced lab tests have reported transmission distances in the 100-km range over standard fiber While the results are limited to 40-Gbps transmission as opposed to OC-768 SONET (e g., limited or no SONET overhead, non-SONET modulation, no ring-type protection), they are encouraging and show healthy R&D progress," Cooperson said.

With carriers starting to fully incorporate OC-192 (10 Gbps) into their networks, is there really a market for OC-768? Can routers even fill the pipe quickly enough, given the advances in DWDM bandwidth creation? Vendors such as Nexabit Networks and Ciena are confident the answer is yes Nexabit's NX6400 router is scheduled to support OC-768 interfaces by Q4 99; Ciena's CoreDirector platform may be OC-768-ready by Q3. According to Rick Dodd, director of product management for Ciena's core switching division, "Carriers want the flexible expansion potential to go from OC-48 to OC-192 to OC-768 in various combinations or in one single increment as their bandwidth growth continues to skyrocket For all practical purposes, carriers are already consuming 16-channel OC-48 DWDM in large numbers As part of the natural evolution of progressive bandwidth expansion, consumption and management, it is prudent to plan for a multiwavelength switch fabric that can cost-effectively support OC-768 in the near future"

OC-768 will certainly take in OC-48 and OC-192 tributaries, but there is some disagreement about where it will be deployed. Communications Industry Researchers Inc 's recent report on SONET cites the technology's current 100-km distance limitations as reason enough to limit use to metropolitan applications Cooperson is reasonably confident of breakthroughs in optical amplifier and regenerator distance performance that would increase performance to 200 km or more, respectively. "IXCs will feel they need to deploy 40 Gbps to achieve maximum throughput and use of their fiber facilities," she said. "Vendors are trying to get them there on conventional single mode fiber and at traditional amplifier and repeater spacings in order to make it attractive to them." According to Cooperson, Level 3 has set itself up to change out its fiber plant if necessary to implement new technologies. "Greenfield carriers will likely spring up to take advantage of new fibers to implement the new transmission scheme. As usual, it will be the incumbents who have the toughest time and who will have to test their cable plant and investigate polarization mode dispersion mitigation techniques to take advantage of OC-768," she said.

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HDSL2: Something Old. Something New

ADC Telecommunications calls it "a new twist on an old pair" The technology behind high bit rate digital subscriber line version 2 (HDSL2) isn't brand new; indeed, it is similar to the original HDSL HDSL2, however. enables T1 communications to be delivered over a single pair of copper wires rather than the conventional two-pair with the same robustness. reach and spectral capability According to John Griffin, general manager of ADC's Loop Division, "HDSL2 is the most significant development for T1 and E1 delivery in years." With T1 rates growing at about 30 percent a year, the ability to double capacity on a pair is attractive to both incumbent and competitive service providers

A spiffed up modulation scheme developed to deliver service is one of the differences HDSL2 has to offer: overlapped pulse amplitude modulation with interlocking spectra (OPTIS), an advanced PAM 2B1Q line code "With a technology like asymmetric DSL (ADSL), interference is a key concern," said Claude Romans, director of forward loop access for RHK. "Within some binder groups you can have crosstalk between signals and cause interference, so there is a spectral compatibility issue This isn't the case with HDSL2 There's high tolerance for the services being offered on adjacent pairs." According to Dataquest, HDSL2 has mixed crosstalk performance at less than 5 decibels, making it a fit for deployment in worst-case environments.

Another difference is standards: HDSL is mostly proprietary technology based on broad definitions developed by Bellcore and The American National Standards Institute (ANSI). The HDSL2 standard is estimated to be 90-percent complete and vendors are already announcing interoperability plans between their HDSL2, line and network termination units. Commercial availability could happen as soon as Q3 99 ADC and PairGain recently announced successful interoperability testing, and the two companies, along with Adtran, have released prestandards versions to trial customers

A third benefit of HDSL2 is reach. While HDSL and HDSL2 have approximately the same reach (12,000 feet before signal regeneration is required), two-pair HDSL2 could increase that reach to 16,500 feet, depending on copper conditions

ILECs are mostly interested in the technology to provide solutions in high-demand areas that are suffering from copper exhaust. The CLECs, however, see dollar signs: The cost savings could be tremendous, considering that CLECs on average get half of their revenues from T1 services. According to industry estimates, CLECs will control as much as 40 percent of all T1 business by 2002 Using one pair instead of two also frees up a CLEC's reliance on an ILEC to provide timely access to the copper loop The cost savings won't be much of an initial factor for ILECs because of their installed base of HDSL equipment HDSL2 is a direct competitor to HDSL, but the extent that one replaces the other is dependent on price, Romans said. About half of the T1s delivered last year were over copper, the other half over fiber. "Last year about 600,000 HDSL modems were delivered to customers, so the potential is there for HDSL2 to really take off," he concluded

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Unified Messaging: Any Way, Any Time, Anywhere:

The idea is simple enough: People should be able to access their voice mail, e-mail and fax messages through a single source and respond to them the same way regardless of whether they are using their computers in a hotel room or calling from payphones in the airport. But unified messaging has been slow to take off because of interoperability issues, the effort required to migrate systems and the lack of standards. Analysts predict, however, that the technology is nearly ready for prime time. While most of the current deployment has been in CPE for the enterprise, a number of service providers, including GTE, AT&T, BT, Bell Canada, Concord Technologies, JFAX and Telia Mobile of Sweden, have become much more aggressive in pushing unified messaging services.

Unified messaging systems combine a number of technologies previously used for single medium messaging (e.g., message servers), but leverage new technologies for multimedia conversion. According to research firm Ovum, text-to-speech engines enable e-mail headers and text to be received over a standard telephone; interactive voice response provides for complex commands to be given quickly and efficiently from a telephone, and voice over IP technology allows voice messages to be transmitted to a PC over the Internet. "The technology side is really exciting right now because vendors are all coming at this from different angles," said Blair Pleasant, a consultant with the Pelorus Group. "Cisco, for example, is putting unified messaging capabilities on its routers. It's going to be available in a lot of different ways." Other vendors with strong solutions include Comverse, Amteva and Lucent's Octel Messaging Division, which teams with Sun Microsystems.

There are still hurdles to overcome. There are hundreds of thousands of legacy voice mail systems in place and because the benefits of unified messaging, such as worker productivity, are difficult to quantify, many users and businesses may have a difficult time justifying the expense. Because countries outside North America have been slow to roll out basic voice mail systems, unified messaging may take off internationally much more quickly, Pleasant said.

Current solutions haven't solved the problem of interoperability with existing voice mail, fax and e-mail systems in a multivendor environment, but Pleasant said a few vendors are almost there Still, Ovum predicts that "the introduction of unified messaging services will be as significant as the introduction of direct dialing" and forecasts that there will be 14 million active mailboxes worldwide by 2002, growing to 170 million by 2006. The Pelorus Group predicts that revenues from unified messaging will grow to $2 3 billion in 2002, up from $28.6 million in 1997. Pleasant said many service providers and enterprise users are holding back on deploying unified messaging services because of Y2K concerns. "That's being very short sighted," she added "If you have to upgrade your voice mail system or do a patch, you may as well upgrade to a unified messaging system"

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Policy-Based Management: Being in Control
More intelligent networks sound great, but how do you get there? More and more service provider and enterprise network managers are eyeing policy-based management as a solution to their QoS woes. Sam Alunni, president of Sterling Research, considers the advent of policy-based management solutions as a defining moment in the telecom industry. "How do you implement QoS?" Alunni asked. "You can hire an army of network administrators or automate the whole process by turning it over to a policy-management capability" Policy management is more than traffic shaping It allows network administrators to map business policies to network policies that control network behavior, allowing for features such as QoS, security, accounting and billing "It gives you the capability to set and establish policies on a time-of-day and day-of-week level so that different classes of users get the end-to-end network resources they need," Alunni said

Policy-based management will first have a strong play in the Fortune 1000 companies, but service providers looking to offer features such as voice over IP, videoconferencing and VPNs are also showing strong interest in the technology Cisco, 3Com, Nortel, Lucent, Intel, Fore Systems and Hewlett-Packard are all looking to get the technology to customers quickly, but most of the solutions are proprietary. Start-up Ukiah Software, however, recently entered the market with a solution that can implement networking with policy enforcement in a multivendor environment, according to Michael Howard, principal at Infonetics. Analysts agree that policy management will factor significantly in service provider and enterprise networks within two years.

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MPLS: High Octane IP

When was the last time a traffic engineering protocol generated as much excitement as Multi-Protocol Label Switching (MPLS)? "MPLS is the single most important issue in the data networking space today," said Tom Nolle, principal at CIMI Corp. "It provides a customer-acceptable framework for delivering IP services with the security and QoS assurance of frame relay. AT&T kind of validated this point in January when it introduced its 'IP-enabled frame relay' service, which is really an MPLS/ATM VPN service."

Other analysts point to MPLS's ability to simplify network complexity so that overall network costs are reduced by up to 50 percent. In addition to AT&T, major carriers such as MCI WorldCom and UUnet have begun testing the protocol. So far, so good. MCI just launched an OC-48 link between Los Angeles and San Francisco for the National Science Foundation's vBNS research network, using MPLS as its primary traffic engineering mechanism. If successful, similar implementations will be deployed in the carrier's commercial network. Some Fortune 500 companies are also early adopters. General Motors decided to update its data systems, charging MPLS with the traffic engineering chores that will migrate legacy traffic to IP, while maintaining the reliability and scalability of older protocols.

The value of MPLS lies in its ability to create connection-mode properties within a connectionless network such as IP. The idea is to map IP services to connection-oriented ATM or frame relay infrastructure by separating the routing and forwarding plane and creating a label-switching plane to assign a 32-bit label that specifies a packet's path through the network. Packets are labeled before they are forwarded to their next hops. At subsequent hops, no further analysis of the packet's header is required. Instead, the label is used as an index into a table, which specifies the next hop and a new label. MPLS dictates that once a packet is assigned to a particular forwarding path, no further header analysis is done by subsequent routers; all forwarding is driven by the labels instead of analyzing each packet at every router hop. MPLS effectively removes the need for dedicated connections, while assuring their reliability; it will be particularly effective for dictating paths based on network or traffic characteristics such as congestion or QoS concerns, policy-based management, customized routing and ingress router decisions MPLS can be used in several modes to engineer traffic QoS control will certainly be at the top of the list, but it can also be used to create an explicit end-to-end route in the network that acts like a PVC, enabling frame-like IP VPNs.

Industry support, analyst backing and market opportunity all point to a bright future for MPLS as an indispensable network protocol, one which may greatly impact the Internet by reducing network congestion and providing better end-to-end service "With MPLS you can build a public IP data network that isn't the Internet and that will serve as the foundation for those company VPN applications that require more than best-effort service and better stability than the Internet provides." Nolle said "We believe that virtually all intracompany traffic in public IP services will be served by MPLS-based VPNs and not by the Internet By 2010, VPNs based at least in part on MPLS will account for about three-quarters of all the public IP service revenue worldwide."

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Smart Antennas: Baseline IQ

The move from analog to digital cellular phones improved voice transmission quality and allowed network operators to offer users more features The challenge now, as spelled out by the ITU, is to develop third-generation (3G) wireless systems with sufficient capacity (up to 384 kbps for mobile applications and up to 2 Mbps for stationary applications) that would let subscribers make phone calls, surf the Web, exchange e-mail and conduct video conferences simultaneously. Thus, a need for smart antennas. The move to 3G will require smart antennas more so than in today's networks," said Sandeep Chandel, Nortel Networks product manager for RF capacity and performance.

Just as the popularity of the Internet is forcing ISPs to increase backbone capacity, the burgeoning number of wireless phone users is straining the capacity of cell sites. Since interference is the key threat that limits the capacity of a cell, the simple, albeit expensive, solution is to add more base stations. In other words, if one base station serves an area within a 4,000-foot radius, then by splitting that cell into smaller cells through the addition of more base stations, an operator could reduce the area served by the base station to a 1000-foot radius. Market research firm Allied Business Intelligence estimates that wireless subscribers worldwide will number 667 million by the end of 2003. According to ABI analyst Larry Swasey, operators will deploy 2.5 million base stations by year-end 2003 to keep up with the growing number of subscribers. This is one reason why Texas Instruments, a leading provider of silicon for handsets, recently started focusing on the DSP market for base stations.

Conventional base stations waste energy because only a small amount of the signal reaches the intended recipient. Also, when a base station listens for signals, it not only receives the desired signal but also interference from other signals. Smart antennas, on the other hand, are able to listen to a particular subscriber and deliver energy to that subscriber more efficiently. Martin Cooper, chairman of smart antenna maker ArrayComm and putatively the father of the cell phone, uses this analogy to explain the concept behind smart antennas. Even in a crowded room a person is able to filter out irrelevant conversations and pick up on the voice of a particular individual. "Similarly, with smart antennas you try to listen only to people you want to listen to, and you talk back to them," Cooper said "Five years from now any new base station coming into the market will use smart antennas."

Companies such as ArrayComm, Metawave and Andrew Corp. hope to either license their technologies to the established infrastructure vendors or sell their products to network operators. For instance, Metawave has developed a smart antenna, SpotLight 2000, that works with both analog and digital systems. It is preparing for a trial of its antenna in China Telecom's GSM networks. Metawave claims its antennas result in a 50-percent boost in capacity in a CDMA system and 100-percent capacity boost in an analog system Meanwhile, Ericsson has underway a research and evaluation project of its Wideband CDMA (WCDMA) system with two GSM network operators in Germany. In contrast to ETSI, which has specified the CDMA air interface for 3G systems, ANSI will permit any of the three (CDMA, TDMA and GSM) interfaces.

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Optical Switching: Adding Brains to Brawn

Although SONET was a major breakthrough for delivering bandwidth in long-haul networks, it only provided a single channel over fiber. Today, DWDM is able to provide dozens of channels over a single fiber, each channel running at a rate of 2.5 Gbps or higher. However, for interexchange carriers building DWDM networks, these are basically "dumb" point-to-point pipes. While switching technology is firmly ensconced in other networking schemes, it has yet to make an appearance in the domain of optical networks "No one today is doing wavelength-to-wavelength conversion, and if they are doing switching, it is with an electrical stage," according to Mike Guess, IXC Communications' vice president of engineering What a few leading-edge equipment makers are promising to deliver soon are boxes that take in and spew out optical signals, bringing a level of intelligence and granularity to DWDM channels Inside the boxes, however, the switching involves a conversion from optical to electric and back to optical.

A new breed of bandwidth management technology is emerging, targeted at long-haul carriers' points-of-presence. The technology allows for provisioning services across a network built with multiple DWDM terminals and automatically restoring those services in the case of a fiber break Switches based on this technology, from startups Monterey Networks and Lightera Networks (recently acquired by Ciena), also enable carriers to build a flatter network that eliminates the need for such devices as ATM switches, SONET add/drop multiplexers and digital cross-connects. For example, to build an optical Internet, service providers would only need Monterey's Wavelength Routers, high-speed IP routers and DWDM terminals By contrast, Lightera's switch makes it easier to provision services--from OC-1 to wavelengths--across an optical backbone "As networks continue to grow, Lightera is addressing the combined need for traffic grooming, high-speed protection and restoration, and different levels of survivability," said Joe Skorupa, RHK's director for switching and routing Mark E. Allen, manager of technology development for Williams Network, described the technology as promising: "We expect to be deploying these kinds of products in our network."

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Wireless Location Technology: Where in the World?

Sometimes vendors push technology, other times service providers push vendors to push technology In the case of wireless location, regulation may be forcing technology into action. Wireless location services are designed to determine the physical location of a cellular phone user within a certain radius through technology embedded in either a handset or cellular base station. The idea, originally conceived to meet the FCC's Enhanced-911 (E-911) mandate, is shaping up to have some serious marketing potential for wireless carriers as well This technology became famous in the mainstream media recently, because it was not available. For example, a woman lost in a South Dakota snowstorm for more than a day has become a poster child in the lobby for wireless location because her location was not able to be pinpointed after she called 911.

Phase 1 of the FCC's mandate, which was supposed to be completed last year, required that when a public safety answering point (PSAP) in charge of directing emergency services receives a 911 call, it must be provided with a callback number and general calling location. Most carriers outsourced the routing of information to the PSAP to companies such as SCC Communications and Xypoint, which have mapped out the locations of PSAPs and base stations. Reports on the success of Phase 1 are spotty, and service providers point to legacy PSAP systems as part of the holdup Phase 2, mandated for completion by October 2001, requires a caller's location to be pinpointed within a radius of 400 feet 67 percent of the time. Many vendors are in trials with service providers such as US West Wireless, GTE. Bell Atlantic Mobile and Western Wireless and are claiming much greater accuracy than regulations require

There are two types of solutions on the market from a number of vendors Network-based solutions employ an old military method called triangulation, which uses time difference of arrival (TDOA) to determine where a caller is in relation to a base station. Handset-based solutions incorporate global positioning system (GPS) chips "Operators sign up with a service and the position of a caller is given either to the operator. who sends it on to the PSAP, or it goes directly to the PSAP," said Larry Swasey, senior analyst with Allied Business Intelligence "This measurement is much more accurate. but it requires that a GPS chip be added to the phone." SnapTrack. a San Jose. Calif-based company, estimates that installing a GPS chip will cost \$5 to \$10 per phone. with the cost declining with time and scale. Upgrading base stations could cost upward of \$25.000 each

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The big decision for service providers is whether to implement a bare-bones approach to just meet the FCC requirements or install wireless intelligent network (WIN) capabilities A WIN standard should be resolved by the end of 1999 The WIN option is costly for service providers to implement. but it allows capabilities beyond E-911 to be available throughout the network, instead of on a switch-by-switch basis. The Strategis Group estimates that within 10 years the location-based services market in the United States could reach $8.4 billion for services such as location-sensitive billing, public safety, fleet vehicle monitoring, traffic reporting and a variety of concierge services, such as restaurant and hotel locators.

--Senior Editors Susan O'Keefe and Sam Masud; Assistant Editor Doug Allen.

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CURSOS ABIERTOS

VIII CURSO INTERNACIONAL EN TELECOMUNICACIONES

MÓDULO IV:

REDES DIGITALES: "ACTUALIDAD Y PERSPECTIVA"

TEMA

IDENTIFIERS FOR ORGANIZATIONS FOR TELECOMUNICATIONS ADDRESSING (IOTA)

EXPOSITOR: M. en C. ARTURO ELIE ALALUF OLIVARES
PALACIO DE MINERÍA
JUNIO DE 1999

Identifiers For Organizations For Telecommunications Addressing (IOTA)

(and for other purposes)
Using the ISO/IEC 6523 ICD System

For more information or to make an application please complete the form below

CONTENTS

Background

There are several methods of creating telecommunications addresses, and identifiers for OSI information objects. One of the methods uses an International Code Designator (ICD) issued by an International Registration Authority (RA) in accordance with ISO 6523

The ICD (as defined by the standard, ISO 6523) 'indicates the particular organization coding system, used for the purpose of unique identification of an organization when data relevant to the organization is interchanged'. It is important to note that the ICD does not directly identify an organization and the RA must reject applications for this purpose. Those requiring an ICD to identify a scheme which they operate must apply through a sponsoring authority, such as their National Standards body, who are required by the Standard to check the eligibility of the application The RA is not permitted to process applications sent to them direct by applicants.

More than one hundred and twenty ICDs have already been allocated Many organizations requiring an identifier using the ICD system will be featured in one or more of the organization identification schemes already registered. The organization identifiers used in any registered scheme can be used in conjunction with its ICD to create telecommunications addresses, and identifiers for OSI information objects

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Introduction to IOTA

Despite the number of organization schemes already registered many organizations find difficulty in obtaining an identifier appropriate for their use. Even if they can establish the existence of an organization identification scheme, that includes their organization among the membership, the identifiers allocated may be inconveniently long or include characters that are invalid for their purpose. It is also possible that some schemes may not guarantee absolute stability and permanence of identifiers allocated. They may therefore be subject to change without the knowledge or agreement of the organizations identified.

The IOTA organization identification scheme has been introduced to provide a readily available source of ICD format identifiers for organizations, anywhere in the world, that wish to use an ICD based scheme for any purpose The term 'organization' includes individuals requiring identifiers. The need for an organization to own more than one identifier is thought to be unlikely, but if an application does impose physical constraints on the number of sub identifiers that can be appended, one or more additional organization identifiers may be requested.

The introduction of the scheme was particularly motivated by a demand for ATM addresses Representatives of the ATM Forum were therefore consulted to ensure that the scheme is suitable for that purpose but it is believed to be almost universally applicable

The IOTA scheme has been assigned an ICD by the International Registration Authority in accordance with ISO 6523. The Issuing Organization responsible for the administration of the IOTA scheme, is the Business Development Group of DISC, British Standards Institution (BSI). It has considerable experience in the management of similar schemes because it acts as the UK Name Registration Authority. The BSI is the National Standards Body of the UK. It is also the International Registration Authority for a number of International Standards, including ISO/IEC 6523 but its role with the IOTA scheme is quite independent of these other activities. BSI is an organization incorporated by Royal Charter and its involvement as the issuing organization for the IOTA scheme gives assurance of a stable and responsible administration.

Price

The issuing organization charges a single administration fee of £150 (+ VAT) for the allocation of an organization identifier, for use in conjunction with the ICD A register is maintained by BSI-DISC of all registered organizations

Application of IOTA Organization Identifiers

The prime purpose of the IOTA scheme is the provision of organization identifiers for use by those organizations to create unique addresses and identifiers for their own purposes

The IOTA scheme organization identifiers are not intended for use as a coded representation of the organization name and there is no provision for their promulgation Nevertheless, there is no reason why this should not be so employed, but it would be necessary for those wishing to use them in this way to maintain their own internal directory.

The fundamental IOTA requirement is a value that can be relied upon to be unique in the context in which it is used. The meaning to be associated with the value is determined by the users and is almost unlimited in nature. Examples may include persons, objects such as items of equipment, blocks of computer program code, geographical locations, and attachment points on electrical circuits.

Systems for the generation of large numbers of identifiers usually rely on delegated authority. An authority at each level is responsible for ensuring that every identifier allocated at that level is unique. The whole structure is depicted as an inverted tree with the root segment, which is always a single value, at the top. Various terminology is used but a level, depicted as a branch in the tree, is often known as an arc.

A unique identifier constructed in this manner therefore consists of a number of discrete components. Each component is a single value taken from an arc in the structure and all components must be present unless any of the values in the higher arcs can be implied

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Format

With the ICD format, the value of the ICD is one arc, and the organization identifier the value in the arc immediately below it

The IOTA identifier format consists of a fixed length four digit ICD, value 0124, and a six digit fixed length organization identifier, (which may include non significant zeros) Once an organization identifier has been allocated it will not be amended or reallocated.

ISO 6523 -1984 specifies the ICD as having a fixed length of four digits. The publication of a new version is imminent and this specifies it of being variable length of up to four digits, (which provides the option to omit the non-significant preceding zero).

The updated version of ISO 6523 is also likely to extend the Standard to cover the identification of organization parts, but this does not seem relevant for the purposes of the creation of identifiers and addresses If the owner of an organization identifier wishes to allocate sub identifiers to parts of the organization, or to anything else, they can already do so

The format of the organization identifiers is not covered by international standards. The issuing organization for IOTA intends the organization identifiers to be used as fixed length six digit values However, this does not preclude the omission of non-significant preceding zeros for applications not dependent on character comparison, or where the representation of non-significant zeroes is impracticable

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Additional information

As in the case of most other organization identification schemes the issuing organization

(BSI-DISC) reserves absolute discretion in the operation of the scheme. This includes the right to decline an application although it is not anticipated that the need will arise.

Although, there is no provision for the promulgation of identifiers issued, BSI-DISC does reserve the discretion to answer queries on the ownership of identifiers unless confidentiality is specially requested at the time of application. Experience indicates that freedom to disclose information is usually beneficial to participating organizations. In some large global organizations the easiest way to find out whether the organization has already been allocated an identifier may be to ask the issuing organization.

EXAMPLES

For the purposes of the following examples the ICD is assumed to be 0124 and the Organization identifier to be 001234

In the first example there is a slight disparity in the terminology because ISO/IEC 8824 uses the term "Organisation Code" for what is called 'Organization Identifier' in ISO/IEC 6523.

Forming an OSI Object Identifier in accordance with ISO/IEC 8824-1: 1995

The ISO assignment of object identifier component values is specified in annex B of the Standard. The relevant clauses and values are: -

Clause	Identifier	Value
B1	ISO	1
B4	Identified Organization	3
B7	ICD	0124
B7	Organization Code	001234
B7	Further arcs	As defined by the identified organization but so that values are unique

Forming an ATM telecommunications address in accordance with ISO/IEC 8348:1996 and the ATM Forum Recommendation

The organisation identifier is used to construct an ICD formatted ATM End System Address (AESA) as illustrated in the following figure The IOTA organisation ID is coded into the first of three octets of the HO-DSP portion of an AESA

Octet	Description	Content	Value
1	Authority and Format Identifier (AFI)	Constant designating ICD ATM Format	47
2-3	Initial Domain Identifier (IDI)	ICD	0124
4-6	Organization ID	Organization Identifier	001234
7-13	Organizationally Assigned	As determined by the owner of the Organization Identifier	See below

14-19	End System Identifier	See below	-do-
20	Selector	See below	-do-

The AFI value, 47, indicates an ICD NSAP using binary coding in the domain specific part.

Note that leading zeroes in the four digit IDI (0124) and the six character organisation identifier (001234) are significant This is due to the use of a binary coded Domain Specific Parts in AESAs and the use of a pattern matching in ATM Forum PNNI routing

The remaining seven bytes of the Domain Specific part (DSP) are administered by the identified organisation as it sees fit. It may, for example, allocate part of the space to other organizations such as customers. It is advisable to define a hierarchical structure for the DSP that reflects the topology of the organisation's current or planned network

The ESI typically contains the Media Access Control (MAC) address associated with the end system hardware, while the Selector byte is reserved for possible use within the end system.

The organisation identifier can also be used to construct a group address See ATM Forum UNI Signalling 4.0 and PNNI version 1.0 for more information on this and for other information on AESAs

All ATM Forum specifications, including the above, are freely available at http://www.atmforum.com.

To make a request for an Identifier within the IOTA Organization Scheme or for more information please leave your details below

Would you like an application form? Yes ◉ No ○
Please select **YES** if you require an application
Select **NO** for information only.

Name:

Job Title

Address:

Company / department name

Size of Company (*No of employees*)

Building name and/or number

Street

Town

County / region

Postcode

Country:
Tel number:
FAX number:
email address:
Comments:

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