# Telecommunications Systems and Standards

Latonia Gist

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Email: info@wtbooks.com

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# Chapter 1

# Call Accounting

A **Call Accounting** System is a telecommunications software or hardware application that captures, records, and costs telephone usage events. Internationally call accounting systems may be referred to as call logging systems. Call accounting systems detect outbound and inbound calls, call ring outs, call routings, abandoned calls, and other activities.

## *Common Applications of Call Accounting*

Service Billing and Provisioning

Call accounting systems may provide packaging, pricing, provisioning, billing, and posting or presentment of telephone services for purposes of revenue generation. Professional services firms utilize call accounting software for account code or client based billing of their phone usage. The hospitality industry uses call accounting to resell phone services to visiting guests and groups. These call accounting systems often provide accessible application-specific rating and provisioning capabilities found generally on carrier-level operational support systems (OSS) and business support systems (BSS).

Departmental and Employee Chargeback

The original purpose of call accounting systems was within corporate entities for purposes of cost allocations within the enterprise. Enterprises use call accounting to allocate costs back to divisions, departments, and even individual employees. Such systems may also provide data directly to corporate accounting and human resource systems.

Cost and Revenue Optimization

Call accounting software can reconcile multiple telecom carrier's billing reports by integrating telecom invoices, wireless billing, long distance charges and calling cards into a single platform, allowing your business to use the convergent expense software to provide management reports, analytic reports, alerts and robust presentations

Staff Productivity

Call accounting systems provide visibility into the calling patterns and activity of employees and can be used to minimize productivity losses through non-business calling activity. They can also be used to evaluate the effectiveness of revenue-generating staff and sales processes, and manage the responsiveness of customer service staff.

Network Optimization

Companies also use call accounting systems to determine whether their voice and data networks are being utilized efficiently, in a cost effective manner, or to capacity. Call accounting applications are used to monitor network activity and bandwidth, identify over- and under-used trunks to optimize trunking configurations, pinpoint the root cause of circuit outages, monitor call routing effectiveness, queue times, abandoned calls and other information, and report usage trends and statistics. Call accounting can help companies more efficiently allocate telecom resources and make better planning decisions affecting a telecommunications network.

Security and Compliance

Call accounting applications enable IT departments to shield companies from a variety of internal and external security threats by monitoring for network attacks, intrusion attempts and telecom activity that exceeds acceptable or established thresholds.

The Sarbanes-Oxley Act established new or enhanced standards for financial reporting by public companies and accounting firms, obliging them to examine and revise their internal governance, accounting and reporting controls. Examples of those “internal controls” include procedures for securing, protecting and ensuring the availability of critical infrastructure (like telecommunications systems), monitoring those systems for performance issues, and preventing or quickly detecting unauthorized attempts to acquire company records or assets.

Call accounting applications help facilitate regulatory compliance by providing tools to aggregate, monitor and analyze telecommunications data in order to correctly track and report expenses. They can be used to enhance corporate accounting practices by documenting fraud-related costs to substantiate telephony-related disputes with carriers. And they can automate the monitoring of internal communications to improve visibility into sales and financial processes.

## *Forms*

Call accounting capabilities can be acquired in several forms.

Stand-alone hardware device

Call accounting can be provided from a hardware device. Such solutions, especially ones with proprietary designed hardware buffers, have generally been static and with limited features, and are intended to support a single telephone switch. However some hardware solutions use a barebone PC or mini PC with software written to work within an operating system. These PC-based devices can support a wider range of PABXs depending on the software module written and is usually done by replacing or adding PABX specific software modules.

Stand-alone server software

Call accounting can be provided from a stand-alone server software product. Such solutions have more robust features and are intended to support a single or small number of phone switches and users. However these solutions also have a higher total cost of ownership by requiring management of one or several self-standing systems.

Centralized enterprise software

Call accounting can be provided from a hosted multi-property enterprise server solution. Such solutions provide more robust features and can support large numbers of distributed phone switches and users, all without the complexity of many distributing systems scattered across the enterprise. Enterprise call accounting systems centralize management and monitoring of call accounting across an enterprise. Such systems over an enterprise WAN, VPN, or over the public Internet, and eliminate distributed standalone distributed call accounting systems and as a result provide a lower cost of ownership and a consistent distribution of features and functions across an enterprise. Care should be taken to insure that such enterprise solutions operate from a single real-time truly multi-property data bases versus multiple duplicated/replicated copies of the software itself which mitigates the value of centralization. Hosted enterprise solutions are appropriate for anybody supporting more than one telephone PBX/PABX or anyone who seeks one central call accounting solution regardless of multiple possible PBX/PABX vendors. Given rapid changes in voice and more generally VOIP, centrally hosted solutions also make it easier to continuously adapt to changing requirements by providing one place for managing system change. Centralized call accounting solutions are easily justified if you compare it the total life cycle cost of ownership for the distributed technology.

Software as a Service (SaaS)

Call accounting software functionality can be accessed as an online software service. Software services are typically hosted by the software vendor themselves. SaaS providers often include optional levels of systems monitoring, management, and analysis services atop the call accounting software itself. SaaS solutions (also sometimes called managed service solutions or application service provider (ASP) solutions) generally provide greater operational simplicity while actually providing more software capability and greater return on investment. Like the hosted enterprise call accounting solutions, managed service providers *eliminate* distributed call accounting systems and with them the related management and attention required to keep multiple distributed systems operating, refreshed, and current. Like enterprise call accounting software above, SaaS is easily justified if you compare it to the total lifecycle cost of ownership for the distributed technology.

Web services

The more recent approach to call accounting when using VOIP-enabled phone portals and devices is to embed call accounting information into telephony devices as part of a service-oriented architecture (SOA). This is not to be confused with embedding an entire call accounting software package into a voice solution but instead as embedding integration to a centrally hosted call accounting web service with your overall telephone solution. Web services is generally a very inexpensive way to get customized access to sophisticated call accounting features and eliminate the need to manage technology to get that capability.

## *Trends In Call Accounting*

Integrated Call Accounting and Management Services

Call accounting systems are increasingly augmented with telemanagement/telemanager services. Call accounting data often requires expert analysis from which to leverage its value. Meanwhile, the telecommunications world continues to evolve rapidly making retention of up-to-date core competency more difficult. Call accounting system owners therefore often engage telecommunications management (often called telemanager or telemanagement) services to operate and optimize their call accounting technology. Such subscription services can transcend any one specific call accounting technology and can actually unify reporting and management across many technologies. When engaging telecommunications assistance take care to assess the telemanager's tool set as this is a primary driver of the cost effective value they can create for you.

Integrating voice and broadband internet accounting into one accounting platform

With the convergence of voice and data the leading call accounting systems are also providing billing, provisioning, and accounting for broadband internet services. Solutions for such systems may be called "communications accounting systems". Like call accounting systems they may be distributed or centralized systems.

Centrally Hosted Call Accounting

Voice is increasingly managed at an enterprise level - above any one PBX or communications server. Also many PBX technologies can manage multiple points of presence across an enterprise. As a result centrally hosted enterprise call accounting systems are displacing distributed standalone premise-based call accounting systems. Application service providers (ASP's) can also provide call accounting as an internet based service, usually bundled with related telecommunications services.

### *How Call Accounting Systems Work*

Generally, call accounting systems collect data from a key system, a PBX, iPBX, or Voice over IP (VOIP) gateway generated by service activity on all or selected phone extensions or devices. The system attaches costs and possibly revenues to that activity. Call accounting systems in the United States and its territories must cost and surcharge phone activity using metered rates structured round the North American Numbering Plan (NANP). More sophisticated call accounting systems will actually provision services on the PBX's and communications servers. Traditional PBX's send calling activity information out of a serial port or via a proprietary TCP/IP network service. The call accounting system has a capturing module or a capturing hardware device that is then able to store the data and feed data to and from the rating engine. More recent iPBX's provide access to information by retaining it in online data bases for extraction by external systems. The voice related data collected usually includes calling party, date, time, duration, destination party and authorization or account code. This data is sometimes called Call Detail Recording (CDR) or Station Message Detail Recording (SMDR).

### *Call Accounting In Hospitality*

Hotels make more sophisticated use of call accounting systems than many corporate entities. First, hotels require realtime processing from their call accounting systems. Also, while corporate call accounting systems largely provide departmental chargeback, call accounting systems in hospitality provide more sophisticated chargeback and markup algorithms for revenue based resale of phone services to targeted visitors, staff, partners, and guests. Also, the hospitality industry frequently leverages centralized enterprise call accounting and fully managed call accounting services as hoteliers often lack on-property staff that can operate on-premise systems and seek simplicity and bottom line cost savings. TSPS was an early method of providing such service.

Traditionally, hotel chains and management companies have suggested that properties keep their call accounting systems up-to-date and accurate. They have done this for four main reasons: (1) to recover the cost of long-distance calls, (2) to properly allocate, account for, and charge customers for their phone usage, (3) to generate revenue through the resale of phone calls, and (4) to track phone calls made to and from their property for marketing, planning and other purposes. However, given the low cost of telecommunications capacity available to the hoteliers today, the low phone usage rates in hotels, and the limited qualified staff available within individual hotel properties, such activity is increasingly problematic. Call accounting is therefore increasingly bundled within more comprehensive telemanager services provided to the hotel by third parties or by the hotel's corporate staff. Such services manage the more complete financial aspect of telecommunications usage and facilities.

## Chapter 2

# Convergence (Telecommunications) and Public Switched Telephone Network

# Convergence (telecommunications)

In the field of telecommunications, **telecommunications convergence** or **network convergence** or simply **convergence** are broad terms used to describe emerging technologies, and network architecture designs used to migrate voice and data networks into a single network. Specifically this involves the converging of previously distinct media such as voice telephony and internet into common interfaces on single devices. It is a concept dating to AT&T in 1928, but has evolved in the 21st century to dominate the market positioning of telecoms operators. Telecommunication convergence is a disruptive technology.

Communication media including data communications, telecommunications and broadcast originally developed as business operations providing distinct services. Broadcasting, voice telephony and on-line computer services evolved on different platforms: TV and radio sets, telephones and computer and were managed by different business support systems. Different media were each regulated differently by different regulators. Telecom Media Convergence is about merging these multiple industries.

### *History*

The historical roots of convergence can be traced back to the emergence of mobile telephony and the Internet, although the term properly applies only from the point in marketing history when fixed and mobile telephony began to be offered by operators as joined products.

- Fixed and mobile operators were, for most of the 1990s, **independent companies**. Even when the same organization marketed both products, these were sold and serviced independently.

## *Benefits*

Enterprises who transition to a converged network often do so to realize savings in IT operational costs, and to integrate their business IT systems. Moving voice and data services to a common application layer, on a shared data network allows enterprises to gain access to data, once maintained in a segmented, silo architecture. For example, converged network call-center data and statistics, may now be integrated directly into an Enterprise Resource Planning system.

Many service providers now offer turn-key converged network solutions tailored to the small to medium enterprise market. Such offerings require far less capital expenditure than a comparable segmented, legacy voice & data solution. This recently availability of service, low capital expenditure, and relative easy transition has driven many enterprises to adapt converged network solutions.

## *Technology implications*

Convergent solutions include both fixed-line and mobile technologies. Recent examples of new, convergent services include:

- Using the Internet for voice telephony
- Video On Demand
- Fixed-Mobile Convergence
- Mobile-to-Mobile Convergence
- Location Based Services
- Integrated Products and Bundles

Convergent technologies can integrate the fixed-line with mobile to deliver convergent solutions. Convergent technologies include:

- IP Multimedia Subsystem (IMS)
- Session Initiation Protocol (SIP)
- IPTV
- VOIP
- Voice Call Continuity
- Digital Video Broadcasting - Handheld
- Video On Demand technologies

### Single view of customer

Convergent solutions integrate with marketing and customer relationship management in order to target bundled offerings at specific user groups. The integration of multiple services as convergent solutions requires the network operator to examine the interests and the value adding opportunities of extending base products. As convergent solutions involve multiple billing and provisioning systems so too must convergent solutions provide unified and enhanced customer support

### Usability and Quality of Service

The maturity of convergent solutions beyond early adoption models depends upon the usability and quality of service provided by convergent solutions. PacketCable Multimedia is an example of an application-independent Quality of Service architecture for real-time IP based services.

### Policy based security and authorization

Both fixed-line and mobile operators use policy servers as part of a policy-based network that provides authorization services and control of network systems. As part of a convergent single view of customer network policy, servers must control access to content and services delivered by multiple providers.

## *Convergence regulation*

Convergence has also raised several debates about classification of certain telecommunications services. As the lines between data transmission, audio cast and voice transmission are eroded, regulators are faced with the task of how best to classify the converging segments of the telecommunication sector.

# Public switched telephone network

The **public switched telephone network** (**PSTN**) is the network of the world's public circuit-switched telephone networks. It consists of telephone lines, fiberoptic cables, microwave transmission links, cellular networks, communications satellites, and undersea telephone cables all inter-connected by switching centers which allows any telephone in the world to communicate with any other. Originally a network of fixed-line analog telephone systems, the PSTN is now almost entirely digital in its core and includes mobile as well as fixed telephones.

The technical operation of the PSTN utilises standards created by the ITU-T. These standards allow different networks in different countries to interconnect seamlessly. There is also a single global address space for telephone numbers based on the E.163 and E.164 standards. The combination of the interconnected networks and the single numbering plan make it possible for any phone in the world to dial any other phone.

## *History*

The first telephones had no network but were in private use, wired together in pairs. Users who wanted to talk to different people had as many telephones as necessary for the purpose. A user who wished to speak whistled into the transmitter until the other party heard.

Soon, however, a bell was added for signalling, and then a switchhook, and telephones took advantage of the exchange principle already employed in telegraph networks. Each telephone was wired to a local telephone exchange, and the exchanges were wired together with trunks. Networks were connected together in a hierarchical manner until they spanned cities, countries, continents and oceans. This was the beginning of the PSTN, though the term was unknown for many decades.

Automation introduced pulse dialing between the phone and the exchange, and then among exchanges, followed by more sophisticated address signaling including multi-frequency, culminating in the SS7 network that connected most exchanges by the end of the 20th century.

The growth of the PSTN meant that traffic engineering techniques needed to be deployed to deliver quality of service (QoS) guarantees for the users. The work of A.K. Erlang established the mathematical foundations of methods required to determine the capacity requirements and configuration of equipment and the number of personnel required to deliver a specific level of service.

In the 1970s the telecommunications industry began implementing packet switched network data services using the X.25 protocol transported over much of the end-to-end equipment as was already in use in the PSTN.

In the 1980s the industry began planning for digital services assuming they would follow much the same pattern as voice services, and conceived a vision of end-to-end circuit switched services, known as the Broadband Integrated Services Digital Network (B-ISDN). The B-ISDN vision has been overtaken by the disruptive technology of the Internet.

Today, only the oldest parts of the telephone network still use analog technology for anything other than the last mile loop to the end user, and in recent years digital services have been increasingly rolled out to end users using services such as DSL, ISDN, FTTx and cable modem systems.

There are a number of large private telephone networks which are not linked to the PSTN, usually for military purposes. There are also private networks run by large companies which are linked to the PSTN only through limited gateways, like a large private branch exchange (PBX).

### *PSTN operators*

The task of building the networks and selling services to customers fell to the network operators. The first company to be incorporated to provide PSTN services was the Bell Telephone Company in the United States.

In some countries however, the job providing telephone networks fell to government as the investment required was very large and the provision of telephone service was increasingly becoming an essential public utility. For example, the General Post Office in the United Kingdom brought together a number of private companies to form a single nationalised company.

In recent decades however, these state monopolies were broken up or sold off through privatization.

## *Regulation of the PSTN*

In most countries, the central government has a regulator dedicated to monitoring the provision of PSTN services in that country. Their tasks may be for example to ensure that end customers are not over-charged for services where monopolies may exist. They may also regulate the prices charged between the operators to carry each others traffic.

## *Technology in the PSTN*

### Network topology

The PSTN network architecture had to evolve over the years to support increasing numbers of subscribers, calls, connections to other countries, direct dialling and so on. The model developed by the US and Canada was adopted by other nations, with adaptations for local markets.

The original concept was that the telephone exchanges are arranged into hierarchies, so that if a call cannot be handled in a local cluster, it is passed to one higher up for onward routing. This reduced the number of connecting trunks required between operators over long distances and also kept local traffic separate.

However, in modern networks the cost of transmission and equipment is lower and, although hierarchies still exist, they are much flatter, with perhaps only two layers.

### Digital channels

As described above, most automated telephone exchanges now use digital switching rather than mechanical or analog switching. The trunks connecting the exchanges are also digital, called circuits or channels. However analog two-wire circuits are still used to connect the last mile from the exchange to the telephone in the home (also called the local loop). To carry a typical phone call from a calling party to a called party, the analog audio signal is digitized at an 8 kHz sample rate using 8-bit pulse code modulation (PCM). The call is then transmitted from one end to another via telephone exchanges. The call is switched using a call set up protocol (usually ISUP) between the telephone exchanges under an overall routing strategy.

The call is carried over the PSTN using a 64 kbit/s channel, originally designed by Bell Labs. The name given to this channel is Digital Signal 0 (DS0). The DS0 circuit is the basic granularity of circuit switching in a telephone exchange. A DS0 is also known as a timeslot because DS0s are aggregated in time-division multiplexing (TDM) equipment to form higher capacity communication links.

A Digital Signal 1 (DS1) circuit carries 24 DS0s on a North American or Japanese T-carrier (T1) line, or 32 DS0s (30 for calls plus two for framing and signaling) on an E-carrier (E1) line used in most other countries. In modern networks, the multiplexing function is moved as close to the end user as possible, usually into cabinets at the roadside in residential areas, or into large business premises.

These aggregated circuits are conveyed from the initial multiplexer to the exchange over a set of equipment collectively known as the access network. The access network and inter-exchange transport use synchronous optical transmission, for example, SONET and Synchronous Digital Hierarchy (SDH) technologies, although some parts still use the older PDH technology.

Within the access network, there are a number of reference points defined. Most of these are of interest mainly to ISDN but one – the V reference point – is of more general interest. This is the reference point between a primary multiplexer and an exchange. The protocols at this reference point were standardized in ETSI areas as the V5 interface.

# Chapter 3

# Operations Support System

**Operations Support Systems** (also called **Operational Support Systems** or **OSS**) are computer systems used by telecommunications service providers. The term OSS most frequently describes "network systems" dealing with the telecom network itself, supporting processes such as maintaining network inventory, provisioning services, configuring network components, and managing faults. The complementary term Business Support Systems or **BSS** is a newer term and typically refers to "business systems" dealing with customers, supporting processes such as taking orders, processing bills, and collecting payments. The two systems together are often abbreviated **BSS/OSS** or simply **B/OSS**.

Different subdivisions of the BSS/OSS systems are made, depending on whether they follow the TeleManagement Forum's diagrams and terminology, industry research institutions or BSS/OSS vendors own view. Nevertheless in general, an OSS covers at least the application areas:

- Network Management Systems
- Service Delivery
- Service Fulfillment, including the Network Inventory, Activation and Provisioning
- Service Assurance
- Customer Care
- Billing

## *History and development of OSS*

Before about 1970, many OSS activities were performed by manual administrative processes. However, it became obvious that much of this activity could be replaced by computers. In the next 5 years or so, the telephone companies created a number of computer systems (or software applications) which automated much of this activity. This was one of the driving factors for the development of the Unix operating system and the C programming language. The Bell System purchased their own product line of PDP 11 computers from Digital Equipment Corporation for a variety of OSS applications. OSS systems used in the Bell System include AMATPS, CSOBS, EADAS, RMAS, SCCS, SES, TIRKS, and many more. OSS systems from this era are described in the Bell System Technical Journal, Bell Labs Record, and Telcordia Technologies (formerly Bellcore) Special Report SR-2275, Telcordia Notes on the Networks.

Many OSS systems were initially not linked to each other and often required manual intervention. For example, consider the case where a customer wants to order a new telephone service. The ordering system would take the customer's details and details of their order, but would not be able to configure the telephone exchange directly - this would be done by a switch management system. Details of the new service would need to be transferred from the order handling system to the switch management system - and this would normally be done by a technician re-keying the details from one screen into another - a process often referred to as "swivel chair integration". This was clearly another source of inefficiency, so the focus for the next few years was on creating automated interfaces between the OSS applications - OSS integration. Cheap and simple OSS integration remains a major goal of most telecom companies.

## *A brief history of OSS architecture*

A lot of the work on OSS has been centered on defining its architecture. Put simply, there are four key elements of OSS:

- Processes
  - the sequence of events
- Data
  - the information that is acted upon
- Applications
  - the components that implement processes to manage data
- Technology
  - how we implement the applications

During the 1990s, new OSS architecture definitions were done by the ITU-T in its TMN model. This established a 4-layer model of TMN applicable within an OSS:

- Business Management Level (BML)
- Service Management Level (SML)
- Network Management Level (NML)
- Element Management Level (EML)

(Note: a fifth level is mentioned at times being the elements themselves, though the standards speak of only four levels) This was a basis for later work. Network management was further defined by the ISO using the FCAPS model - Fault, Configuration, Accounting, Performance and Security. This basis was adopted by the ITU-T TMN standards as the Functional model for the technology base of the TMN standards M.3000 - M.3599 series. Although the FCAPS model was originally conceived and is applicable for an IT enterprise network, it was adopted for use in the public networks run by telecommunication service providers adhering to ITU-T TMN standards.

A big issue of network and service management is the ability to manage and control the network elements of the access and core networks. Historically many efforts have been spent in standardization fora (ITU-T, 3GPP) in order to define standard protocol for network management, but with no success and practical results. On the other hand IETF SNMP protocol (Simple Network Management Protocol) has become the de-facto standard for internet and telco management, at the EML-NML communication level.

From 2000 and beyond, with the growth of the new broadband and VoIP services, also the management of the home networks is entering the scope of OSS and network management. DSL Forum TR-069 specification has defined the CPE WAN Management Protocol (CWMP), suitable for managing home networks devices and terminals at the EML-NML interface.

## *TM Forum (formerly the TeleManagement Forum)*

TM Forum is an international membership organization of communications service providers and suppliers to the communications industry. While OSS is generally dominated by proprietary and custom technologies, TM Forum is regarded as the most authoritative source for standards and frameworks in OSS. TM Forum has been active in proving a framework and discussion forum for advancements in OSS and BSS.

The newest developments in OSS architecture are the results of the TM Forum's New Generation Operations Systems and Software (NGOSS) program, which was established in 2000. This established a set of principles that OSS integration should adopt, along with a set of models that provide standardized approaches.

### NGOSS models

- An information model (the Shared Information/Data model, or SID) - now more commonly referred to as the Information Framework,
- A process model (the enhanced Telecom Operation Map, or eTOM) - now more commonly known as the Business Process Framework,
- An application model (the Telecom Applications Map) - now known as the Application Framework, an architecture (the Technology Neutral Architecture) and a lifecycle model.

### NGOSS architectural standards

The TM Forum describes NGOSS as an architecture that is:

- "loosely coupled"
- distributed
- component based

The components interact through a common communications vehicle (using an information exchange infrastructure; e.g., EAI, Web Services, EJB).

The behavior can be controlled through the use of process management and/or policy management to orchestrate the functionality provided by the services offered by the components.

## History

The early focus of the TM Forum's NGOSS work was on building reference models to support a business stakeholder view on process, information and application interaction. Running in parallel were activities that supported an implementation stakeholder view on interface specifications to provide access to OSS capability (primarily MTNM). The MTNM work evolved into a set of Web Services providing Multi-Technology Operations System Interfaces MTOSI. Most recently, the OSS through Java initiative (OSS/J) joined the TMF to provide NGOSS-based BSS/OSS APIs.

## Future work

Considerable work remains, primarily in building out the system stakeholder reference models, which are needed to support a business process driven and SOA styled approach to using model driven development for specifying the additional implementation stakeholder interface specs (for SOA Web Services, EJB and EAI). These are required to meet the demands of Service Providers operating using the IMS architectural framework and NGN communications networks.

## Chapter 4

# NetExpert

| NetExpert | |
|---|---|
| **Developer(s)** | Objective Systems Integrators |
| **Stable release** | 6.2.2 / July, 2009 |
| **Preview release** | 6.3 / 2010 first half |
| **Written in** | Java, C++, |
| **Operating system** | UNIX; Linux, Windows clients |

| NetExpert Neon | |
|---|---|
| **Developer(s)** | Objective Systems Integrators |
| **Stable release** | 9.1 / 2010 |
| **Preview release** | 10.0 / TBD |
| **Written in** | Java, C++, |
| **Operating system** | UNIX; Linux, Windows clients |

***NetExpert*** and ***NetExpert Neon*** are used by Communications Service Providers (fixed, broadband, and wireless companies) to manage their networks, services, products, and customers. Also used in Utilities, Enterprise and Financial Institutions to manage mission-critical networks and services.

## Operational Support Systems

The telecommunications industry refers to NetExpert and products similar to it as operational support systems (OSS) because they support and maintain the integrity of Network Operation Centers, Service Operation Centers, and Customer Care organizations. Typically Network Operation Centers require extensive software systems to collect, correlate, and automate events and exceptions from network resources as well as service impacting resources.

Service Operation Centers have been created recently by CSP's to focus on key services which are consumer facing. Service Impact and Quality dashboards are also leveraged by Customer Care organizations.

OSS for these large service providers need to be mission-critical and run 7x24.

### Software Developer

Objective Systems Integrators develops and markets NetExpert and NetExpert Neon which are enabling technologies for solutions for the Communications Service Provider networks, services, and corporate customers..

### Trademark

NetExpert is a registered trademark throughout the world. "Charting the future of Service Management" is also a registered trademark brand.

### Customer Assurance

Objective Systems Integrators develops NetExpert Neon. A platform for Service Quality and Impact Management across next generation services such as IPTV, SMS, Blackberry, Wireless Broadband, Carrier, and other Service and Customer Health needs. NetExpert Neon is also the foundation for Customer Insight, Experience Analysis, and SLA Management.

## *Overview*

### NetExpert

NetExpert monitors and controls networks and service impacting resources using object-oriented and expert systems technologies.

### NetExpert Neon

NetExpert Neon enabling technologies is focused on services and customers:

- Service Health - Service Quality and Impact management leveraging industry standard portal technology for a holistic view of critical customer services.
- Customer Health - Customer SLA, Experience Analysis, and Insight of corporations, groups within corporations, and individuals.

## History

- NetExpert was initially developed as a framework for fault management in 1989. The framework started as an offshoot of combining expert systems technology with network management.
- The framework was also adapted to activation and performance collection, and many companies worldwide now use it for those purposes.
- In the mid-1990s OSI developed over 120 adapters for NetExpert fault, activation, and performance collection for a host of network element manufacturers.
- In the second half of the 1990s, OSI started to package applications to help reduce the time it took to develop adapters. These became known as the eXEL series and included FMeXEL, CMeXEL, and PMeXEL. The eXELs were based on the NetExpert rule language and provided application logic out-of-the-box. This helped accelerate the time it took to deploy applications.
- A policy framework called Distributed Management Policies (DMP) introduced after the year 2000 added automation to the existing NetExpert framework. DMP policies enable deployment without rule writing.
- New applications based on DMP are now available to provide service assurance and service management for many telecommunications industry domains and services.
- May, 2009 - OSI launches NetExpert Neon. "Neon" represents the new architecture for Service Quality Management solutions. Also, NetExpert is once again spelled using the uppercase "N" and "E" versus NETeXPERT. "NetExpert" is the original spelling of the product dating back to the 1990s.

## Background

NetExpert is considered an OSS, used in managing wireline and wireless networks and services.

It consists of a series of coordinated modules that fall into three general groups:

- gateways,
- object persistence and behavior servers,
- user/operator workstation/Web interfaces.

NetExpert is a scalable and distributable architecture that supports flexible configuration while maintaining individual component independence. Its application packages address many areas of communication services management, including fault, performance, reporting, activation, IP services, and others. These can be further tailored to individual customer environments and management requirements.

This framework consists of a set of integrated software modules and graphical user interface (GUI) development tools to enable the creation and deployment of complex management solutions. The object-oriented architecture of the NetExpert framework provides the building blocks to implement operations support and management systems using high-level tools rather than low-level programming languages.

The NetExpert framework is founded on open systems and object-oriented methodology. NetExpert supports different standards, transmission protocols, and equipment data models. NetExpert is based on the Telecommunications Management Network architecture created by the Telecommunications Standardization Sector of the International Telecommunications Union. It supports the development and deployment of applications for the main TMN management areas—fault, configuration, accounting, performance, and security—and the implementation of layered management architectures. In addition, the NetExpert framework employs expert rules and policies that replace complex programming languages and enable network analysts to model desired system behaviors by using GUI-based rule editors.

## *Functional areas and applications*

NetExpert operates in these related management functional areas and applications as defined by ITIL, eTOM, FCAPS, and other sources.

- Service level management
- Service management
    - Exception management
    - Quality of service
    - Key performance indicators (KPI) are essential parts of a measurable objective, which is made up of a direction, KPIs, benchmark, target, and time frame.
- Performance management
    - Network performance
    - Network performance management
- Fault management
    - Event correlation is a technique for making sense of a large number of events and pinpointing the few events that are really important in that mass of information. Event correlation usually takes place inside the management platform.
    - Root cause analysis (RCA) is rooted in the practice of failure analysis as employed in operations.
- Network management
- System management
- Configuration management
- Common Information Model (CIM) is an open standard that defines how managed elements in an IT environment are represented as a common set of objects and relationships between them. This is intended to allow consistent management of these managed elements, independent of their manufacturer or provider.

## *Solution domains*

NetExpert solutions have addressed these domains (legacy, IP, and next generation networks, resources, and services):

- SMS
- SONET/SDH
- 2G wireless
- 3G wireless
- Digital subscriber line (DSL)
- IPTV
- IP Multimedia Subsystem
- WiMAX

## *NetExpert system integration tools*

NetExpert gateways and APIs support resource management, data collection, monitoring, and control. NetExpert communication protocols and APIs:

### Application layer

- Simple Network Management Protocol (SNMP)
- Common management information protocol (CMIP)
- Common Object Request Broker Architecture (Corba)
- Extensible Markup Language (XML)
- OSS/J
- TELNET
- FTP
- HTTP
- Web Services
- X.25
- SQL
- Transaction Language 1 (TL1)
- FIFO
- Secure Shell, Shell (computing)

### Transport layer

- TCP/IP

### Network layer

- IPv4, IPv6

### *NetExpert vendor-specific adapters*

NetExpert has managed operational support systems, element management systems, and network elements from the following companies:

- ADVA Optical Networking
- Agilent Technologies
- Alcatel-Lucent
- Aldiscon
- BREW
- Centigram
- Cisco Systems
- CMG
- Compaq
- Comverse Technology
- Cramer
- CRBT
- Critical Path, Inc.
- DMC
- DPS
- DSC
- ECI
- Enavis
- Ericsson
- Extreme Networks
- FibCom
- Fujitsu
- Hughes
- IBM
- Ineoquest
- Jatayu
- Logica
- Marconi
- Motorola
- Nabla
- Navis
- NEC
- Nokia
- Nortel
- OpenWave
- PSA
- Remedy
- Siemens
- Spatial
- SS8
- Tejas

- Tekelec
- Telcordia
- Tellabs
- Tivoli
- TruePosition
- UTStarcom
- ZTE

### *Framework*

#### Advantages

- Open architecture allows service providers to access the managed information, events, KPI's, and object information for additional use. NetExpert leverages Oracle for its information base.
- Flexibility, through gateways, object models, rule language, policies, and graphical clients, enables NetExpert solutions to be extended by customers and system integrators.
- Adaptibility permits interoperability with various network elements, management systems, and other Operational Support Systems through various protocols and through bi-directional collection, command, and control.

#### Challenges

- In the past the pure NetExpert framework has been leveraged as a toolset which required rule-based customization.
- Predeveloped policy applications helped with time to market and usability.
- The latest NetExpert Neon solution offers more out-of-the-box features, such as service model templates and wizards for increased usability, reducing the challenges of rule-based customization.

# Chapter 5

# Managed Facilities-Based Voice Network

A **Managed Facilities-based Voice Network**, or MFVN, is a physical network owned and operated by a voice service provider that delivers traditional telephone service via a loop start analog telephone interface. MFVNs are interconnected with the Public Switched Telephone Network (PSTN) and provide dialtone to end users. Historically, this was provided by equipment at Bell company central offices, however today's MFVNs can include a combination of access network (last mile network of copper, coaxial cable, or fiber optics), customer premises equipment (CPE), network switches and routers, network management systems, voice call servers, and gateways to the larger PSTN.

MFVN providers include cable operators and telephone companies, but do not include Internet based providers such as Vonage, Magic Jack, and others that use the public internet to carry calls.

## *Definition*

MFVN providers:

(a) Manage and maintain their network to ensure end-to-end service quality and reliability from the service subscriber location to the PSTN or other MFVN peer network, (b) Provide a service that is functionally equivalent to traditional analog phone service with respect to dialing, dial plan, call completion, carriage of voice signals and protocols, and loop voltage treatment, (c) Provide real-time transmission of voice signals that carry FAX, data, Point of Sale (POS), burglar alarm and fire alarm system formats unchanged, (d) Provide both professional installation and subscriber information on home wiring practices for residential installations if self-installation is available, which preserves primary line seizure for alarm system interconnection, and (e) Have major and minor disaster recovery plans to address both individual customer outages and widespread events such as tornados, ice storms or other occurrences of a catastrophic nature, which include specific network power restoration procedures equivalent to those of traditional landline telephone services.

## *History*

The term MFVN was introduced in 2007 by various telephony user organizations and stakeholders who rely on telephone service to provide security and life safety services. The concern of these organizations and stakeholders was the reliability of new telephone technology and services. This new technology was based on packet voice technology, or the Voice over Internet Protocol, which was not well understood. These organizations and stakeholders increasingly realized that they could no longer simply assume that phone service would be reliable enough, because it was increasingly being delivered in various ways, even by traditional providers. Clear performance requirements were needed to define when a phone line was suitable for security and life safety services.

This issue was not new, as analog copper based networks had been transitioning to digital telephony technology for 25 years (via fiber buildout by telephone companies), and to IP technology methods for the last 10 years (via broadband buildout by telco, cable, and competitive local exchange carriers). What was new was that copper based analog phone service was not even an option anymore in many areas, as it was being completely replaced by digital and IP based phone service.

Starting in the early part of the 2000s, IP based voice services began being offered by non-traditional providers such as cable television service providers and Internet voice service providers. The demand for these services grew due to competitive pricing and value added services not offered by the traditional telephone providers. The use of these non-traditional telephone methods for security and life safety communications was not well understood, so use was discouraged and in some cases not allowed by local authorities. There was no distinction between voice services provided over the "best-effort" Internet and voice services provided over managed facilities. It became clear that only managed facilities based providers could assure reliability end to end. Only facilities based providers could monitor and maintain the expected quality of service (call quality, operation during power failure, wiring procedures that guaranteed pre-emption of existing calls for emergency calls, and local disaster recovery capabilities).

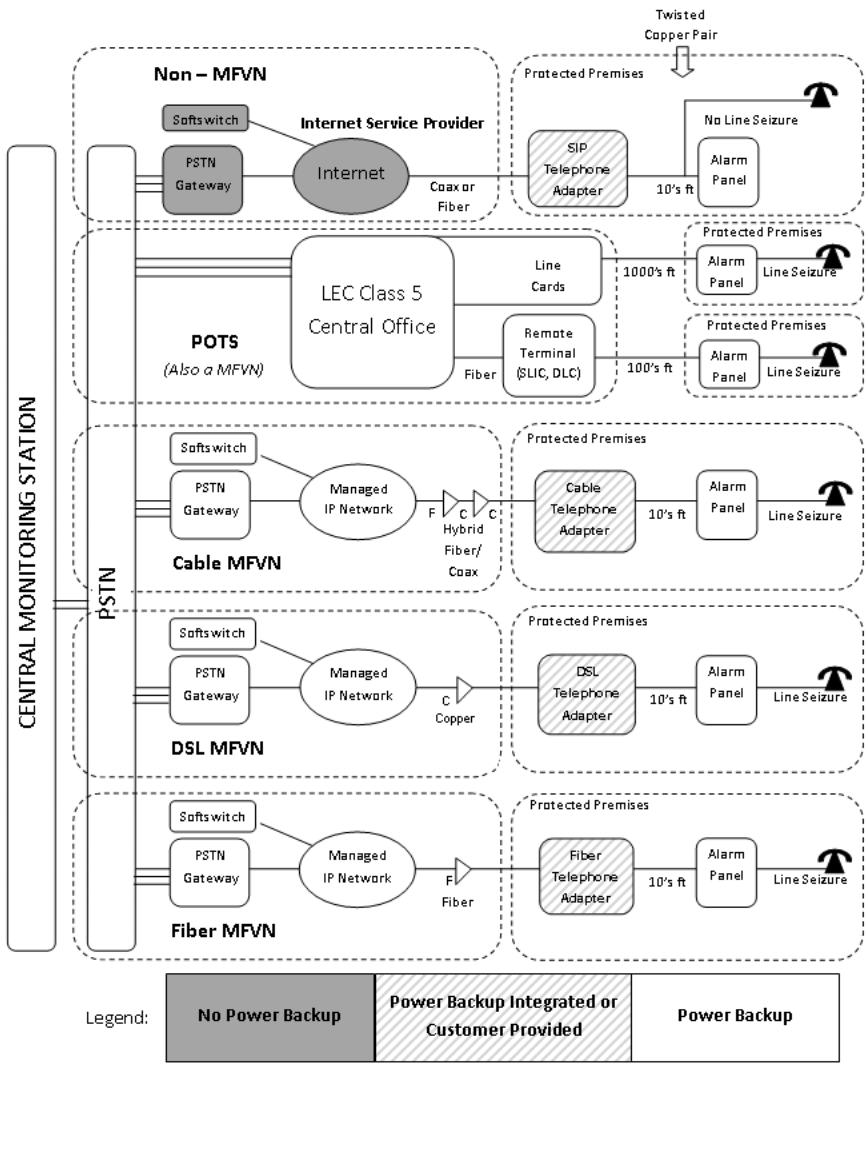
In 2007, the concept of the Managed Facilities-based Voice Network was introduced by non-traditional telephone providers as a way to think about the PSTN as a collection of managed networks, rather than as a single, monolithic entity.

The National Fire Protection Association incorporated this concept into the latest fire code, NFPA 72 2010, which is now the basis for determining whether a given phone line is an acceptable method for fire alarm signaling transmission from a protected premises to a supervising central monitoring station. Local authorities, such as fire inspectors, now no longer need to make these determinations on a individual case basis.

States have begun to recognize and accept the use of MFVN. In Florida, it has been adopted by statute, whereby all quaified MFVNs are now allowed for fire alarm monitoring.

- Florida Division of State Fire Marshal, Florida Administrative Code (F.A.C.), Monitoring section of the FIRE SAFETY STANDARDS FOR FIRE ALARM SYSTEMS Rule Number 69A-48.008, ID 7860509

The following diagram is a high level view of the different types of MFVNs vs non-MFVNs compared side-by-side. They are the non-MFVN Internet VoIP, Plain Old Telephone Service MFVN (POTS), MFVN Cable, MFVN DSL, and MFVN Fiber.



# Chapter 6

# Telex

The **telex** network is a switched network of teleprinters similar to a telephone network, for the purposes of sending text based messages. This network provided the first common media for international record communications using standard signalling techniques and operating criteria as specified by the International Telecommunications Union.

## *Development*

A Siemens T100 telex machine


A late-model British Telecom "Puma" telex machine of the 1980s

By 1935, message routing was the last great barrier to full automation. Large telegraphy providers began to develop systems that used telephone-like rotary dialing to connect teletypes. These machines were called "telex" (**TEL**eprinter **EX**change). Telex machines first performed rotary-telephone-style pulse dialing for circuit switching, and then sent data by Baudot code. This "type A" telex routing functionally automated message routing.

The first wide-coverage telex network was implemented in Germany during the 1930s as a network used to communicate within the government.

At the rate of 45.45 (±0.5%) baud — considered speedy at the time — up to 25 telex channels could share a single long-distance telephone channel by using *voice frequency telegraphy multiplexing*, making telex the least expensive method of reliable long-distance communication.

Canada-wide automatic teleprinter exchange service was introduced by the CPR Telegraph Company and CN Telegraph in July 1957 (the two companies, operated by rivals Canadian National Railway and Canadian Pacific Railway, would join to form CNCP Telecommunications in 1967). This service supplemented the existing international telex service that was put in place in November 1956. Canadian telex customers could connect with nineteen European countries in addition to eighteen Latin American, African, and trans-Pacific countries. The major exchanges were located in Montreal (01), Toronto (02), and Winnipeg (03).

In 1958, Western Union started to build a telex network in the United States. This telex network started as a satellite exchange located in New York City and expanded to a nationwide network. Western Union chose Siemens & Halske AG, now Siemens AG, and ITT to supply the exchange equipment, provisioned the exchange trunks via the Western Union national microwave system and leased the exchange to customer site facilities from the local telephone company. Teleprinter equipment was originally provided by Siemens & Halske AG and later by Teletype Corporation. Initial direct international telex service was offered by Western Union, via W.U. International, in the summer of 1960 with limited service to London and Paris.

In 1962, the major exchanges were located in New York City (1), Chicago (2), San Francisco (3), Kansas City (4) and Atlanta (5). The telex network expanded by adding the final parent exchanges cities of Los Angeles (6), Dallas (7), Philadelphia (8) and Boston (9) starting in 1966.

The telex numbering plan, usually a six-digit number in the United States, was based on the major exchange where the customer's telex machine terminated. For example, all telex customers that terminated in the New York City exchange were assigned a telex number that started with a first digit "1". Further, all Chicago based customers had telex numbers that started with a first digit of "2". This numbering plan was maintained by Western Union as the telex exchanges proliferated to smaller cities in the United States. The Western Union Telex network was built on three levels of exchanges. The highest level was made up of the nine exchange cities previously mentioned. Each of these cities had the dual capability of terminating both telex customer lines and setting up trunk connections to multiple distant telex exchanges. The second level of exchanges, located in large cities such as Buffalo, Cleveland, Miami, Newark, Pittsburgh and Seattle, were similar to the highest level of exchanges in capability of terminating telex customer lines and setting up trunk connections. However, these second level exchanges had a smaller customer line capacity and only had trunk circuits to regional cities. The third level of exchanges, located in small to medium sized cities, could terminate telex customer lines and had a single trunk group running to its parent exchange.

Loop signaling was offered in two different configurations for Western Union Telex in the United States. The first option, sometimes called local or loop service, provided a 60 milliampere loop circuit from the exchange to the customer teleprinter. The second option, sometimes called long distance or polar was used when a 60 milliampere connection could not be achieved, provided a ground return polar circuit using 35 milliamperes on separate send and receive wires. By the 1970s, and under pressure from the Bell operating companies wanting to modernize their cable plant and lower the adjacent circuit noise that these telex circuits sometimes caused, Western Union migrated customers to a third option called F1F2. This F1F2 option replaced the DC voltage of the local and long distance options with modems at the exchange and subscriber ends of the telex circuit.

Western Union offered connections from telex to the AT&T Teletypewriter eXchange (TWX) system in May 1966 via its New York Information Services Computer Center. These connections were limited to those TWX machines that were equipped with automatic answerback capability per CCITT standard.

In 1970, Cuba and Pakistan were still running 45.5 baud type A telex. Telex is still widely used in some developing countries' bureaucracies, probably because of its reliability and low cost. The UN asserted at one time that more political entities were reliably available by telex than by any other single method.

Around 1960[?], some nations began to use the "figures" Baudot codes to perform "Type B" telex routing.

Telex grew around the world very rapidly. Long before automatic telephony was available, most countries, even in central Africa and Asia, had at least a few high-frequency (shortwave) telex links. Often these radio links were first established by government postal and telegraph services (PTTs). The most common radio standard, CCITT R.44 had error-corrected retransmitting time-division multiplexing of radio channels. Most impoverished PTTs operated their telex-on-radio (TOR) channels non-stop, to get the maximum value from them.

The cost of TOR equipment has continued to fall. Although initially specialised equipment was required, many amateur radio operators now operate TOR (also known as RTTY) with special software and inexpensive hardware to adapt computer sound cards to short-wave radios.

Modern "cablegrams" or "telegrams" actually operate over dedicated telex networks, using TOR whenever required.

## *Operation and applications*

Telex messages are routed by addressing them to a telex address, e.g., "14910 ERIC S", where 14910 is the subscriber number, ERIC is an abbreviation for the subscriber's name (in this case Telefonaktiebolaget L.M. Ericsson in Sweden) and S is the country code.

Solutions also exist for the automatic routing of messages to different telex terminals within a subscriber organization, by using different terminal identities, e.g., "+T148".

A major advantage of telex is that the receipt of the message by the recipient could be confirmed with a high degree of certainty by the "answerback". At the beginning of the message, the sender would transmit a WRU (Who aRe yoU) code, and the recipient machine would automatically initiate a response which was usually encoded in a rotating drum with pegs, much like a music box. The position of the pegs sent an unambiguous identifying code to the sender, so the sender could verify connection to the correct recipient. The WRU code would also be sent at the end of the message, so a correct response would confirm that the connection had remained unbroken during the message transmission. This gave telex a major advantage over less verifiable forms of communications such as telephone and fax.

The usual method of operation was that the message would be prepared off-line, using paper tape. All common telex machines incorporated a 5-hole paper-tape punch and reader. Once the paper tape had been prepared, the message could be transmitted in minimum time. Telex billing was always by connected duration, so minimizing the connected time saved money. However, it was also possible to connect in "real time", where the sender and the recipient could both type on the keyboard and these characters would be immediately printed on the distant machine.

Telex could also be used as a rudimentary but functional carrier of information from one IT system to another, in effect a primitive forerunner of Electronic Data Interchange. The sending IT system would create an output (e.g., an inventory list) on paper tape using a mutually agreed format. The tape would be sent by telex and collected on a corresponding paper tape by the receiver and this tape could then be read into the receiving IT system.

One use of telex circuits, in use until the widescale adoption of x.400 and Internet email, was to facilitate a message handling system, allowing local email systems to exchange messages with other email and telex systems via a central routing operation, or switch. One of the largest such switches was operated by Royal Dutch Shell as recently as 1994, permitting the exchange of messages between a number of IBM Officevision, Digital Equipment Corporation All-In-One and Microsoft Mail systems. In addition to permitting email to be sent to telex addresses, formal coding conventions adopted in the composition of telex messages enabled automatic routing of telexes to email recipients.

## *Teletypewriter eXchange*

The Teletypewriter eXchange (TWX) was developed by the Bell System in the United States and originally ran at 45.45 baud or 60 words per minute, using five level Baudot code. Bell later developed a second generation of TWX called "four row" that ran at 110 baud, using eight level ASCII code. The Bell System offered both "3-row" Baudot and "4-row" ASCII TWX service up to the late 1970s.

TWX used the public switched telephone network. In addition to having separate Area Codes (510, 610, 710 and 810) for the TWX service, the TWX lines were also set up with a special Class of Service to prevent connections to and from POTS to TWX and vice versa.

The code/speed conversion between "3-row" Baudot and "4-row" ASCII TWX service was accomplished using a special Bell "10A/B board" via a live operator. A TWX customer would place a call to the 10A/B board operator for Baudot – ASCII calls, ASCII – Baudot calls and also TWX Conference calls. The code / speed conversion was done by a Western Electric unit that provided this capability. There were multiple code / speed conversion units at each operator position.

Western Union purchased the TWX system from AT&T in January 1969. The TWX system and the special area codes (510, 610, 710 and 810) continued right up to 1981 when Western Union completed the conversion to the Western Union Telex II system. Any remaining "3-row" Baudot customers were converted to Western Union Telex service during the period 1979 to 1981.

The modem for this service was the Bell 101 dataset, which is the direct ancestor of the Bell 103 modem that launched computer time-sharing. The 101 was revolutionary, because it ran on ordinary unconditioned telephone subscriber lines, allowing the Bell System to run TWX along with POTS on a single public switched telephone network.

### *International Record Carriers*

Bell's original consent agreement limited it to international dial telephony. The Western Union Telegraph Company had given up its international telegraphic operation in a 1939 bid to monopolize U.S. telegraphy by taking over ITT's PTT business. The result was a de-emphasis on telex in the U.S. and a "cat's cradle" of international telex and telegraphy companies. The Federal Communications Commission referred to these companies as "International Record Carriers" (IRCs).

- Western Union Telegraph Company developed a subsidiary named Western Union Cable System. This company later was renamed as Western Union International (WUI) when it was spun off by Western Union as an independent company. WUI was purchased by MCI Communications (MCI) in 1983 and operated as a subsidiary of MCI International.
- ITT's "World Communications" division (later known as ITT World Communications) was amalgamated from many smaller companies: "Federal Telegraph", "All American Cables and Radio", "Globe Wireless", and the common carrier division of Mackay Marine. ITT World Communications was purchased by Western Union in 1987.
- RCA Communications (later known as RCA Global Communications) had specialized in global radiotelegraphic connections. In 1986 it was purchased by MCI International.

- Before World War I, the Tropical Radiotelegraph Company (later known as Tropical Radio Telecommunications, or TRT) put radio telegraphs on ships for its owner, the United Fruit Company (UFC), to enable them to deliver bananas to the best-paying markets. Communications expanded to UFC's plantations, and were eventually provided to local governments. TRT eventually became the national carrier for many small Central American nations.
- The French Telegraph Cable Company (later known as FTC Communications, or just FTCC), which was owned by French investors, had always been in the U.S. It laid undersea cable from the U.S. to France. It was formed by Monsieur Puyer-Quartier. International telegrams routed via FTCC were routed using the telegraphic routing ID "PQ", which are the initials of the founder of the company.
- Firestone Rubber developed its own IRC, the *"Trans-Liberia Radiotelegraph Company"*. It operated shortwave from Akron, Ohio to the rubber plantations in Liberia. TL is still based in Akron.

Bell Telex users had to select which IRC to use, and then append the necessary routing digits. The IRCs converted between TWX and Western Union Telegraph Co. standards.

### *Demise*

Telex is still in operation, but has been mostly superseded by fax, email, and SWIFT.

## Chapter 7

# Digital Signal 1

**Digital signal 1** (**DS1**, also known as **T1**, sometimes "DS-1") is a T-carrier signaling scheme devised by Bell Labs. DS1 is a widely used standard in telecommunications in North America and Japan to transmit voice and data between devices. **E1** is used in place of **T1** outside of North America, Japan, and South Korea. Technically, DS1 is the logical bit pattern used over a physical T1 line; however, the terms "DS1" and "T1" are often used interchangeably.

### *Bandwidth*

A DS1 circuit is made up of twenty-four 8-bit channels (also known as timeslots or DS0s), each channel being a 64 kbit/s DS0 multiplexed carrier circuit. A DS1 is also a full-duplex circuit, which means the circuit transmits and receives 1.544 Mbit/s concurrently. A total of 1.536 Mbit/s of bandwidth is achieved by sampling each of the twenty-four 8-bit DS0s 8000 times per second. This sampling is referred to as 8-kHz sampling. An additional 8 kbit/s of overhead is obtained from the placement of one framing bit, for a total of 1.544 Mbit/s, calculated as follows:

$$\left(8\,\frac{\text{bits}}{\text{channel}}\times 24\,\frac{\text{channels}}{\text{frame}}+1\,\frac{\text{framing bit}}{\text{frame}}\right)\times 8000\,\frac{\text{frames}}{\text{second}}$$
$$=1544000\,\frac{\text{bits}}{\text{second}}$$
$$=1.544\,\frac{\text{Mbit}}{\text{second}}.$$

### *DS1 frame synchronization*

Frame synchronization is necessary to identify the timeslots within each 24-channel frame. Synchronization takes place by allocating a framing, or 193rd, bit. This results in 8 kbit/s of framing data, for each DS1. Because this 8-kbit/s channel is used by the transmitting equipment as overhead, only 1.536 Mbit/s is actually passed on to the user. Two types of framing schemes are Super Frame (SF) and Extended Super Frame (ESF).

A Super Frame consists of twelve consecutive 193-bit frames, whereas an Extended Super Frame consists of twenty-four consecutive 193-bit frames of data. Due to the unique bit sequences exchanged, the framing schemes are not compatible with each other. These two types of framing (SF, and ESF) use their 8 kbit/s framing channel in different ways.

## *Connectivity and alarms*

**Connectivity** refers to the ability of the digital carrier to carry customer data from either end to the other. In some cases, the connectivity may be lost in one direction and maintained in the other. In all cases, the terminal equipment, i.e., the equipment that marks the endpoints of the DS1, defines the connection by the quality of the received framing pattern.

### Alarms

Alarms are normally produced by the receiving terminal equipment when the framing is compromised. There are three defined alarm indication signal states, identified by a legacy color scheme: red, yellow and blue.

**Red alarm** indicates the alarming equipment is unable to recover the framing reliably. Corruption or loss of the signal will produce "red alarm." Connectivity has been lost toward the alarming equipment. There is no knowledge of connectivity toward the far end.

**Yellow alarm** indicates reception from the far end of a data or framing pattern that reports the far end is in "red alarm." Red alarm and yellow alarm states cannot exist simultaneously on a single piece of equipment because the "yellow alarm" pattern must be received within a framed signal. For ESF framed signals, all bits of the Data Link channel within the framing are set to data "0"; the customer data is undisturbed. For D4 framed signals, the pattern sent to indicate to the far end that inbound framing has been lost is a coercion of the framed data so that bit 2 of each timeslot is set to data "0" for three consecutive frames. Although this works well for voice circuits, the data pattern can occur frequently when carrying digital data and will produce transient "yellow alarm" states, making ESF a better alternative for data circuits.

**Blue alarm** indicates a disruption in the communication path between the terminal equipment. Communication devices, such as repeaters and multiplexers must see and produce line activity at the DS1 rate. If no signal is received that fills those requirements, the communications device produces a series of pulses on its output side to maintain the required activity. Those pulses represent data "1" in all data and all framing time slots. This signal maintains communication integrity while providing no framing to the terminal equipment. The receiving equipment displays a "red alarm" and sends the signal for "yellow alarm" to the far end because it has no framing, but at maintenance interfaces the equipment will report "AIS" or Alarm Indication Signal. AIS is also called "all ones" because of the data and framing pattern.

These alarm states are also lumped under the term Carrier Group Alarm (CGA). The meaning of CGA is that connectivity on the digital carrier has failed. The result of the CGA condition varies depending on the equipment function. Voice equipment typically coerces the robbed bits for signaling to a state that will result in the far end properly handling the condition, while applying an often different state to the customer equipment connected to the alarmed equipment. Simultaneously, the customer data is often coerced to a 0x7F pattern, signifying a zero-voltage condition on voice equipment. Data equipment usually passes whatever data may be present, if any, leaving it to the customer equipment to deal with the condition.

## *Inband T1 versus T1 PRI*

Additionally, for voice T1s there are two main types: so-called "plain" or Inband T1s and PRI (Primary Rate Interface). While both carry voice telephone calls in similar fashion, PRIs are commonly used in call centers and provide not only the 23 actual usable telephone lines (Known as "B" channels for bearer) but also a 24th line (Known as the "D" channel for Data) that carries signaling information. This special "D" channel carries: Caller ID (CID) and Automatic Number Identification (ANI) data, required channel type (usually a B, or Bearer channel), call handle, DNIS info, requested channel number and a request for response.

Inband T1s are also capable of carrying CID and ANI information if they are configured by the carrier to do so but PRIs handle this more efficiently. While an Inband T1 seemingly has a slight advantage due to 24 lines being available to make calls (as opposed to a PRI that has 23), each channel in an Inband T1 must perform its own setup and tear-down of each call. A PRI uses the 24th channel as a data channel to perform all the overhead operations of the other 23 channels (including CID and ANI). Although an inband T1 has 24 channels, the 23 channel PRI can set up more calls faster due to the dedicated 24th signalling channel (D Channel).

## *Origin of name*

The name T1 came from the carrier letter assigned by AT&T to the technology. Essentially, the "T" is a part number that was assigned by AT&T. Just as there is the generally known L-carrier and N-carrier systems, T-carrier was next letter available and T1 is the first level in the hierarchy. DS-1 meant "Digital Service - Level 1", and had to do with the × service to be sent (originally 24 digitized voice channels over the T1). The terms T1 and DS1 have become synonymous and include a plethora of different services from voice to data to clear-channel pipes. The line speed is always consistent at 1.544 Mbit/s, but the payload can vary greatly.

## *Alternative technologies*

**Dark Fiber**: Dark fiber refers to unused fibers, available for use. Dark fiber has been, and still is, available for sale on the wholesale market for both metro and wide area links, but it may not be available in all markets or city pairs.

Dark fiber capacity is typically used by network operators to build SONET and dense wavelength division multiplexing (DWDM) networks, usually involving meshes of self-healing rings. Now, it is also used by end-user enterprises to expand Ethernet local area networks, especially since the adoption of IEEE standards for Gigabit Ethernet and 10 gigabit Ethernet over single-mode fiber. Running Ethernet networks between geographically separated buildings is a practice known as "WAN elimination".

### *Semiconductor*

The T1/E1 protocol is implemented as a "line interface unit" in silicon. The semiconductor chip contains a decoder/encoder, loop backs, jitter attenuators, receivers, and drivers. Additionally, there are usually multiple interfaces and they are labeled as dual, quad, octal, etc., depending upon the number.

The transceiver chip's primary purpose it to retrieve information from the "line", i.e., the conductive line that transversed distance, by receiving the pulses and converting the signal which has been subjected to noise, jitter, and other interference, to a clean digital pulse on the other interface of the chip.

### *Examples*

The global telephone network (also known as the Public Switched Telephone Network or PSTN).

# Chapter 8

# 8b/10b Encoding

In telecommunications, **8b/10b** is a line code that maps 8-bit symbols to 10-bit symbols to achieve DC-balance and bounded disparity, and yet provide enough state changes to allow reasonable clock recovery. This means that the difference between the count of 1s and 0s in a string of **at least** 20 bits is no more than 2, and that there are not more than five 1s or 0s in a row. This helps to reduce the demand for the lower bandwidth limit of the channel necessary to transfer the signal.

The code was described in 1983 by Al Widmer and Peter Franaszek in the IBM Journal of Research and Development. IBM was issued a patent for the scheme the following year. IBM's patent notwithstanding, the method, implementation and goals are very similar to Group Code Recording (GCR), used by IBM in its 3400 Series 6250 cpi 9-track tape drives introduced with its System/370 in 1970, by Apple in the floppy disk controller of its Apple II series introduced in 1978, and by Commodore in the floppy disk controller of the Commodore 2040 introduced in 1979.

8b/10b encoding is normally done entirely in link layer hardware. 8b/10b encoding is an 8-bit clean and transparent encoding, so upper layers of the software stack should be unaware that this encoding is being used.

## *How it works*

As the scheme name suggests, 8 bits of data are transmitted as a 10-bit entity called a *symbol*, or *character*. The low 5 bit of data are encoded into a 6-bit group (the 5b/6b portion) and the top 3 bits are encoded into a 4-bit group (the 3b/4b portion). These code groups are concatenated together to form the 10-bit symbol that is transmitted on the wire. The *data symbols* are often referred to as D.x.y where x ranges over 0–31 and y over 0–7. Standards using the 8b/10b encoding also define up to 12 *special symbols* (or *control characters*) that can be sent in place of a *data symbol*. They are often used to indicate start-of-frame, end-of-frame, link idle, skip and similar link-level conditions. At least one of them (i.e. a "comma" symbol) needs to be used to define the alignment of the 10 bit symbols. They are referred to as K.x.y and have different encodings from any of the D.x.y symbols.

Because 8b/10b encoding uses 10-bit symbols to encode 8-bit words, some of the possible 1024 (10 bit, $2^{10}$) codes can be excluded to grant a run-length limit of 5 consecutive equal bits and grant that the difference of the count of 0s and 1s is no more than 2. Some of the 256 possible 8-bit words can be encoded in two different ways. Using these alternative encodings, the scheme is able to affect long-term DC-balance in the serial data stream. This permits the data stream to be transmitted through a channel with a high-pass characteristic, for example Ethernet's transformer-coupled unshielded twisted pair or optical receivers using automatic gain control.

## Encoding tables

Note that in the following tables, for each input byte, with *A* being the least significant bit, and *H* the most significant. The output gains two extra bits, *i* and *j*. The bits are sent low to high: a, b, c, d, e, i, f, g, h, and j; i.e., the 5b/6b code followed by the 3b/4b code. This ensures the uniqueness of the special bit sequence in the comma codes.

The residual effect on the stream to the number of zero and one bits transmitted is maintained as the *running disparity* (*RD*) and the effect of slew is balanced by the choice of encoding for following symbols.

Each 6- or 4-bit code word has either equal numbers of 0s and 1s (a disparity of 0), or comes in a pair of forms, one with two more 1s than 0s (four 1s and two 0s, or three 1s and one 0, respectively) and one with two less. When a 6- or 4-bit code is used that has a non-zero disparity (count of 1s minus count of 0s; i.e., −2 or +2), the choice of positive or negative disparity encodings must be the one that toggles the running disparity. I.e., the non zero disparity codes alternate.

## Running disparity

8b/10b coding is DC-free, meaning that the long-term ratio of 1s and 0s transmitted is exactly 50%. To achieve this, the difference between the number of 1s transmitted and the number of 0s transmitted is always limited to ±2, and at the end of each symbol, it is either +1 or −1. This difference is known as the *running disparity* (RD).

This scheme only needs two states for running disparity of +1 and −1. It starts at −1.

For each 5b/6b and 3b/4b code with an unequal number of 1s and 0s, there are two bit patterns that can be used to transmit it: one with two more 1 bit and one with all bit inverted and thus two more 0s. Depending on the current running disparity of the signal, the encoding engine selects which of the two possible 6- or 4-bit sequences to send for the given data. (Obviously, if the 6- or 4-bit code has equal numbers of 1s and 0s, there is no choice to make, as the disparity would be unchanged.)

Rules for running disparity

| Previous RD | Disparity of code word | Disparity chosen | Next RD |
|---|---|---|---|
| −1 | 0 | 0 | −1 |
| −1 | ±2 | +2 | +1 |
| +1 | 0 | 0 | +1 |
| +1 | ±2 | −2 | −1 |

## 5b/6b

5b/6b code

| Input | EDCBA | RD = −1 abcdei | RD = +1 abcdei | Input | EDCBA | RD = −1 abcdei | RD = +1 abcdei |
|---|---|---|---|---|---|---|---|
| D.00 | 00000 | 100111 | 011000 | D.16 | 10000 | 011011 | 100100 |
| D.01 | 00001 | 011101 | 100010 | D.17 | 10001 | 100011 | |
| D.02 | 00010 | 101101 | 010010 | D.18 | 10010 | 010011 | |
| D.03 | 00011 | 110001 | | D.19 | 10011 | 110010 | |
| D.04 | 00100 | 110101 | 001010 | D.20 | 10100 | 001011 | |
| D.05 | 00101 | 101001 | | D.21 | 10101 | 101010 | |
| D.06 | 00110 | 011001 | | D.22 | 10110 | 011010 | |
| D.07 | 00111 | 111000 | 000111 | D.23 † | 10111 | 111010 | 000101 |
| D.08 | 01000 | 111001 | 000110 | D.24 | 11000 | 110011 | 001100 |
| D.09 | 01001 | 100101 | | D.25 | 11001 | 100110 | |
| D.10 | 01010 | 010101 | | D.26 | 11010 | 010110 | |
| D.11 | 01011 | 110100 | | D.27 † | 11011 | 110110 | 001001 |
| D.12 | 01100 | 001101 | | D.28 | 11100 | 001110 | |
| D.13 | 01101 | 101100 | | D.29 † | 11101 | 101110 | 010001 |
| D.14 | 01110 | 011100 | | D.30 † | 11110 | 011110 | 100001 |
| D.15 | 01111 | 010111 | 101000 | D.31 | 11111 | 101011 | 010100 |
| | | | | K.28 | 11100 | 001111 | 110000 |

† Same code is used for K.x.7

## 3b/4b

3b/4b code

| Input | HGF | RD = −1 fghj | RD = +1 fghj | Input | HGF | RD = −1 fghj | RD = +1 fghj |
|---|---|---|---|---|---|---|---|
| D.x.0 | 000 | 1011 | 0100 | K.x.0 | 000 | 1011 | 0100 |
| D.x.1 | 001 | 1001 | | K.x.1 ‡ | 001 | 0110 | 1001 |

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| D.x.2 | 010 | 0101 | | K.x.2 ‡ | 010 | 1010 | 0101 |
| D.x.3 | 011 | 1100 | 0011 | K.x.3 | 011 | 1100 | 0011 |
| D.x.4 | 100 | 1101 | 0010 | K.x.4 | 100 | 1101 | 0010 |
| D.x.5 | 101 | 1010 | | K.x.5 ‡ | 101 | 0101 | 1010 |
| D.x.6 | 110 | 0110 | | K.x.6 ‡ | 110 | 1001 | 0110 |
| D.x.P7 † | 111 | 1110 | 0001 | | | | |
| D.x.A7 † | 111 | 0111 | 1000 | K.x.7 † ‡ | 111 | 0111 | 1000 |

† For D.x.7, either the Primary (D.x.P7), or the Alternate (D.x.A7) encoding must be selected in order to avoid a run of five consecutive 0s or 1s when combined with the preceding 5b/6b code. Sequences of five identical bits are used in comma codes for synchronization issues. D.x.A7 is only used for x=17, x=18, and x=20 when RD=−1 and for x=11, x=13, and x=14 when RD=+1. With x=23, x=27, x=29, and x=30, the same code forms the control codes K.x.7. Any other x.A7 code can't be used as it would result in chances for misaligned comma sequences.

‡ The alternate encoding for the K.x.y codes with disparity 0 allow for K.28.1, K.28.5, and K.28.7 to be "comma" codes that contain a bit sequence that can't be found elsewhere in the data stream.

## Control symbols

The control symbols within 8b/10b are 10b symbols that are valid sequences of bits (no more than six 1s or 0s) but do not have a corresponding 8b data byte. They are used for low-level control functions. For instance, in Fibre Channel, K28.5 is used at the beginning of four-byte sequences (called "Ordered Sets") that perform functions such as Loop Arbitration, Fill Words, Link Resets, etc.

Resulting from the 5b/6b and 3b/4b tables the following 12 control symbols are allowed to be sent:

Control symbols

| | Input | | RD = −1 | RD = +1 |
|---|---|---|---|---|
| | **DEC** | **HGF EDCBA** | **abcdei fghj** | **abcdei fghj** |
| K.28.0 | 28 | 000 11100 | 001111 0100 | 110000 1011 |
| K.28.1 † | 60 | 001 11100 | 001111 1001 | 110000 0110 |
| K.28.2 | 92 | 010 11100 | 001111 0101 | 110000 1010 |
| K.28.3 | 124 | 011 11100 | 001111 0011 | 110000 1100 |
| K.28.4 | 156 | 100 11100 | 001111 0010 | 110000 1101 |
| K.28.5 † | 188 | 101 11100 | 001111 1010 | 110000 0101 |
| K.28.6 | 220 | 110 11100 | 001111 0110 | 110000 1001 |
| K.28.7 ‡ | 252 | 111 11100 | 001111 1000 | 110000 0111 |

| | | | |
|---|---|---|---|
| K.23.7 | 247 | 111 10111 | 111010 1000 000101 0111 |
| K.27.7 | 251 | 111 11011 | 110110 1000 001001 0111 |
| K.29.7 | 253 | 111 11101 | 101110 1000 010001 0111 |
| K.30.7 | 254 | 111 11110 | 011110 1000 100001 0111 |

† Within the control symbols, K.28.1, K.28.5, and K.28.7 are "comma symbols". Comma symbols are used for synchronization (finding the alignment of the 8b/10b codes within a bit-stream). If K.28.7 is not used, the unique comma sequences 0011111 or 1100000 cannot be found at any bit position within any combination of normal codes.

‡ If K.28.7 is allowed in the actual coding, a more complex definition of the synchronization pattern than suggested by † needs to be used, as a combination of K.28.7 with several other codes forms a false misaligned comma symbol overlapping the two codes. A sequence of multiple K.28.7 codes is not allowable in any case, as this would result in undetectable misaligned comma symbols.

K.28.7 is the only comma symbol that cannot be the result of a single bit error in the data stream.

### Example encoding of D31.1

D31.1 for both running disparity cases

| Input | RD = −1 | RD = +1 |
|---|---|---|
| **HGFEDCBA** | **abcdei fghj** | **abcdei fghj** |
| 00111111 | 101011 1001 | 010100 1001 |

## *Technologies that use 8b/10b*

After the above mentioned IBM patent expired, the scheme became even more popular and is the default DC-free line code for new standards.

Among the areas in which 8b/10b encoding finds application are

- PCI Express prior to v3.0
- IEEE 1394b
- Serial ATA
- SAS
- Fibre Channel
- SSA
- Gigabit Ethernet (except for the twisted pair based 1000Base-T)
- InfiniBand
- XAUI
- Serial RapidIO
- DVB Asynchronous Serial Interface (ASI)
- DisplayPort Main Link

- DVI and HDMI Video Island (Transition Minimized Differential Signaling)
- HyperTransport
- Common Public Radio Interface (CPRI)
- USB 3.0.

## Digital audio

The encoding has found a heavy use in digital audio applications which use this modulation scheme:

- Digital Audio Tape
- Digital Compact Cassette (DCC).

A differing but related scheme is used for audio CDs and CD-ROMs:

- Compact Disc Eight-to-Fourteen Modulation

## Exceptions

For 10 Gigabit Ethernet's 10GBASE-R Physical Medium Dependent (PMD) interfaces, 64b/66b encoding is used. This scheme is considerably different in design to 8b/10b encoding, but was created with similar considerations of DC balance, maximum run length, transition density and electromagnetic emission minimization.

Note that 8b/10b is the encoding scheme, not a specific code. While many applications do use the same code, there exist some incompatible implementations; for example, Transition Minimized Differential Signaling, which also expands 8 bits to 10 bits, but it uses a completely different method to do so.

# Chapter 9

# APCO-16 and Carrier Grade Linux

## APCO-16

**Project 16** or **APCO Project 16** was a standard development effort started in the 1970s by the Association of Public-Safety Communications Officials-International (APCO), a trade association of mostly police and fire service providers. The program was funded by the Law Enforcement Assistance Administration (LEAA), a part of the US Department of Justice.

### *Details*

In telecommunications, **APCO Project 16** is a standard describing the characteristics and capabilities of public safety trunked radio systems such as:

- channel access times
- automated priority recognition
- data systems interface
- individuality of system users
- command and control flexibility
- system growth capability
- frequency use
- reliability

With the Federal Communications Commission's pending release of the first 800 MHz band licenses, the LEAA funded a project to develop required capabilities and standards needed in trunked public safety two-way radio systems. The report defined proposed methods for frequency reuse, coordination, and interference reduction. The standards also gained acceptance in businesses such as Specialized Mobile Radio, utility communications systems, and refineries.

The study concluded that those frequencies would be suitable for Public Safety mobile radio uses. The study recognized that certain technical problems like "picket fencing", foliage interference and abrupt signal fall out posed some minor problems, also addressed concerns about health effects from 800 MHz transmitters but did not reveal definitive findings.. The availability of the significant additional spectrum and the long term possibility of the eventual collocation of nearly all Public Safety communications into one segment of the spectrum far outweighed these problems.

While the program succeeded in creating basic performance standards and feature sets, it failed to create a signaling standard. The result: three companies built APCO Project 16 compliant systems but radios from each manufacturer were incompatible with one another. In California, for example, University of California, Riverside bought a Motorola system and the County of Riverside purchased a General Electric. In order to communicate, some patch or other custom-built link would have to be installed. Intercommunication was possible but not seamless.

A by product of the work on Project 16 was the recognition that the problems of interagency cooperation inherent in the then standard allocations of separate frequencies to separate functions and agencies in the Public Safety Service might be solved by the use of digital addressing, trunking techniques.

## Project 16A

A follow-on project titled APCO Project 16A was funded by a second LEAA grant. It addressed a proposal to open the 800 MHz band. This program defined technical details such as "channel access time," "system growth capability," and "reliability." Project 16A identified the organizational advantages that would accrue from assigning individual unit addresses and adding an additional "group" address element.

The group element would permit routine unit communications privacy among members of a group while permitting the intercommunications between groups as controlled by a central "group assignment" controller. The extent of the scope of such intra unit coordination would be limited only by the design of the management structure involved and limitations of the addressing and control mechanism technology.

## Project 16B

A third LEAA grant funded Project 16B, "Draft System Implementation Plans for Participating Communities," which addressed how such a system might be implemented in four US cities. This project was funded as a study by the LEAA but actual hardware funding was never authorized.

# Carrier Grade Linux

**Carrier Grade Linux** (**CGL**) is a set of specifications which detail standards of availability, scalability, manageability, and service response characteristics which must be met in order for Linux to be considered "carrier grade" (i.e. ready for use within the telecommunications industry). The term is particularly applicable as telecom converges technically with data networks and commercial off-the-shelf commoditized components such as blade servers.

Carrier-grade is a term for public network telecommunications products that require up to 5 nines or 6 nines (or 99.999 to 99.9999 percent) availability, which translates to downtime per year of 30 seconds (6 nines) to 5 minutes (5 nines). The term "5 nines" is usually associated with carrier-class servers, while "6 nines" is usually associated with carrier-class switches.

## *CGL Project and Goals*

The primary motivation behind the CGL effort is to present an open architecture alternative to the closed, proprietary software on proprietary hardware systems that are currently used in telecommunication systems. These proprietary systems are monolithic (hardware, software and applications integrated very tightly) and operate well as a unit. However, they are hard to maintain and scale as telecommunications companies have to utilize the services of the vendor for even relatively minor enhancements to the system.

CGL seeks to progressively reduce and / or eliminate this dependence on proprietary systems and provide a path for easy deployment and scalability by utilizing cheap COTS systems to assemble a telecommunications system.

The CGL effort was started by the Open Source Development Lab (CGL Working Group). The specification is now in the combined Linux Foundation. The latest specification release is CGL 4.0. Several CGL-registered Linux distributions exist, notably Debian, MontaVista, NexusWare and Wind River.

## *Applications and Services*

The OSDL CGLWG defines three main types of applications that carrier-grade Linux will support — gateways, signaling servers, and management.

- **Gateway applications** provide bridging services between different technologies or administrative domains. Gateway applications are characterized by supporting a large number of connections in real-time over a large number of interfaces, with the requirement of not losing any frames or packets. An example of a gateway application is a media gateway, which converts conventional voice circuits using TDM to IP packets for transmission over an IP-switched network.

- **Signaling server applications**, which include SS7 products, handle control services for calls, such as routing, session control, and status. Signaling server applications are characterized by sub-millisecond real-time requirements and large numbers of simultaneous connections (10,000 or more). An example signaling server application would include control processing for a rack of line cards.

- **Management applications** handle traditional service and billing operations, as well as network management. Management applications are characterized by a much less stringent requirement for real-time, as well as by additional database and communication-oriented requirements. A typical management application might handle visitor and home location registers for mobile access, and authorization for customer access to billable services.

## Chapter 10

# Global Dialing Scheme and HVLAN

## Global Dialing Scheme

The **Global Dialing Scheme** (GDS) is numbering plan for H.323 audio-visual communication networks (often used for videoconferencing). Based on the numerology provided by the United Nations International Telecommunications Union, GDS numerology resembles the international telephone system numbering plan, with some exceptions.

The Global Dialing Scheme uses a hierarchy of gatekeepers to route call set-up information nationally and internationally. National gatekeepers have knowledge of all zones within a country, World gatekeepers have knowledge of all National gatekeepers.

Each basic number consists of four parts: <IAC><CC><OP><EN>.

### *GDS Around the World*

GDS is used heavily in many of the european countries. Educonf as part of GÉANT maintains a connectivity table and interactive map which displays which countries current run and maintain active GDS gatekeepers. Many countries also have a series of test number that can be dialed for both technical testing and GDS connectivity verification.

### *GDS in North America*

The North American root gatekeepers serve the United States and its territories, Canada, Bermuda, and many Caribbean nations, including Anguilla, Antigua & Barbuda, Bahamas, Barbados, British Virgin Islands, Cayman Islands, Dominica, Dominican Republic, Grenada, Jamaica, Montserrat, St. Kitts and Nevis, St. Lucia, St. Vincent and the Grenadines, Trinidad and Tobago, and Turks & Caicos. Their purpose is to resolve h.323 numbers at the '001' prefix level under the Global Dialing Scheme (GDS) plan.

Terminology in this document follows the <IAC><CC><OP><EN> format of general GDS documentation. The '001' above refers to the IAC of 00 and the CC of 1 for North America.

This North American node of the Global Dialing Scheme utilizes an enhanced version of the North American Numbering Plan (NANP) to distribute addresses. The address space is divided into two parts: North American E.164 Space and North American Super Space. North American E.164 Space correlates to existing telephone number assignments and is well-suited for IP telephony applications. North American Super Space utilizes unused NANP address space starting with 0 or 1 to create an address space that is separate from existing telephone numbering addresses. This North American Super Space is well suited to video over IP or other all-IP applications that desire to be distinct from telephony applications and NANP regulations.

GDS users in North America may request addresses in either or both spaces, if needed.

**North American E.164 Space**

Addresses allocated from this range will be based upon the ITU-T e.164 telephone number assigned to the current subscriber of a range of telephone numbers, rather than to the service provider carrying those numbers. For example, if a university held +1.919.226.6100 through +1.919.226.6199, then that university would be eligible for the GDS prefix 00191922661. That university could assign the remaining two digits to endpoints 00-99. Aside from maintaining direct inward dial (DID) capability for endpoints, there is no reason to limit endpoint numbering to two digits. For example, the university might use five digit endpoint numbers for a total address space of 001919226610000 through 001919226619999, yielding 10,000 usable addresses. Organizations that do not have a DID range may use this extension technique to map their entire address space onto a single 10-digit telephone number. Implementors of voice over IP applications may wish to adhere more strictly to the NANP numbering convention.

**North America Super Space**

The organizational prefixes <OP> of addresses in North American Super Space (NASS) start with a 0 or 1 immediately following the country code (1). These digits are not assigned under the NANP, being used rather for special indications as described in 1947 by AT&T and Bell Laboratories.

NASS addresses are of the form:

001PX9<EN>

Where P is a 0 or 1. X is a variable length string of digits consisting of any digit between 0 and 8. 9 is used as a delimiter. <EN> is a variable length user-defined number consisting of any digits 0-9. Thus, NASS addresses are variable in length.

Some examples of fully qualified GDS NASS addresses; all address below contain the GDS IAC (00) and CC (1) with endpoint numbers indicated as <EN>:

0010 Reserved

00119<EN> (OP = 19)

001109<EN> (OP = 109)

001119<EN> (OP = 119)

001129<EN> (OP = 129)

001139<EN> (OP = 139)

001189<EN> (OP = 189)

00110123456789<EN> (OP = 10123456789)

# HVLAN

**Hierarchical VLAN (HVLAN)** is a proposed Ethernet standard that extends the use of enterprise Ethernet VLAN (802.1Q) to carrier networks. A number of developments have emerged in recent years to help bring Ethernet, a flexible and cost-efficient packet transport technology, to carrier networks. These developments include Q-in-Q (802.1ad), PBB (802.1ah), PBT (Provider Backbone Transport), and PBB-TE, which bring a set of features to traditional Ethernet to make it "carrier-grade", adding to it high-availability, OA&M, and more.

While attempting to retain the core features that made Ethernet attractive in the first place, these technologies do not address other inefficiencies that could limit their use in the long term. This is especially true when considering the expected significant growth of multipoint network applications – IPTV, Private LANs, gaming, and others. The delivery of such services is better supported by PBB and associated protocols than alternatives such as MPLS, yet could hit scalability issues should services evolve as predicted.

HVLAN introduces the concept of hierarchical addressing schemes into the VLAN tag to provide both enterprise and carrier transport networks the characteristics they need in the long run.

## *Evolution of carrier-grade Ethernet standards*

### Ethernet

Ethernet is a connectionless technology. It does not have a routing mechanism and its address scheme is based on 48-bit MAC addresses. However, its flat address scheme results in a potential explosion of forwarding database entries and an uncontrolled flooding of broadcast messages throughout the network. In order to overcome Ethernet's scalability issues, a partitioning scheme, named VLAN, was introduced.

### VLAN (802.1Q)

**A virtual LAN**, commonly known as VLAN, is a method of creating independent logical Ethernet networks within a physical network. Several VLANs can co-exist within such a network. This helps in reducing the broadcast domain and aids in network administration by separating logical segments of a LAN (like company departments) that should not exchange data using a LAN (they still can exchange data by routing).

VLANs are configured through software rather than hardware, which makes them extremely flexible. Frames having a VLAN tag carry an explicit identification of the VLAN to which they belong. The value of the VLAN Identification (VID) in the tag header signifies the particular VLAN the frame belongs to. The main problem with VLAN is its limited VID space (4096). While this space may suffice for enterprise applications, it is much too small for carrier networks, which must support many customers and services.

### Q-in-Q (802.1ad)

A number of solutions have been proposed to increase VLAN's scalability. A first proposal, called Q-in-Q, also known as Provider Bridge, VLAN stacking or tag stacking, allows service providers to **insert an additional VLAN tag** (referred to as provider VLAN) in the Ethernet frame in order to identify the service, resulting in a unique 24-bit length label. While this solution enables one, in theory, to identify up to 16 million services (4094 * 4094), in reality, one provider VLAN is dedicated to one customer, and therefore the number of supported customers is still limited to 4094.

Q-in-Q also introduces a scalability issue within the core of the carrier network, where every core switch needs to learn and maintain forwarding entries for every customer MAC address.

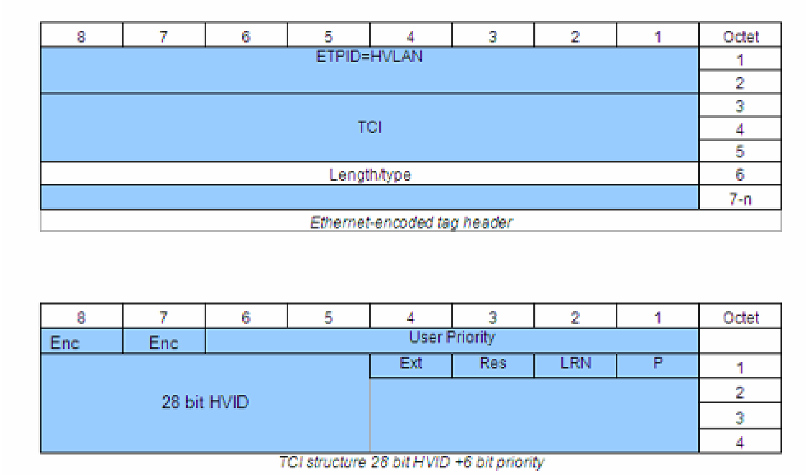
### Mac-in-Mac (802.1ah)

PBB, PBT, and PBB-TE use an alternative proposed solution, known as MAC-in-MAC, described in the proposed IEEE 802.1ah Provider Backbone Bridges standard, which encapsulates Ethernet frames with a Service Provider MAC header. MAC-in-MAC technology overcomes the inherent scalability limitations of VLAN and Q-in-Q networks that make them impractical for use in larger networks by enabling up to 4000 times as many service instances as supported by traditional VLAN and Q-in-Q networks.

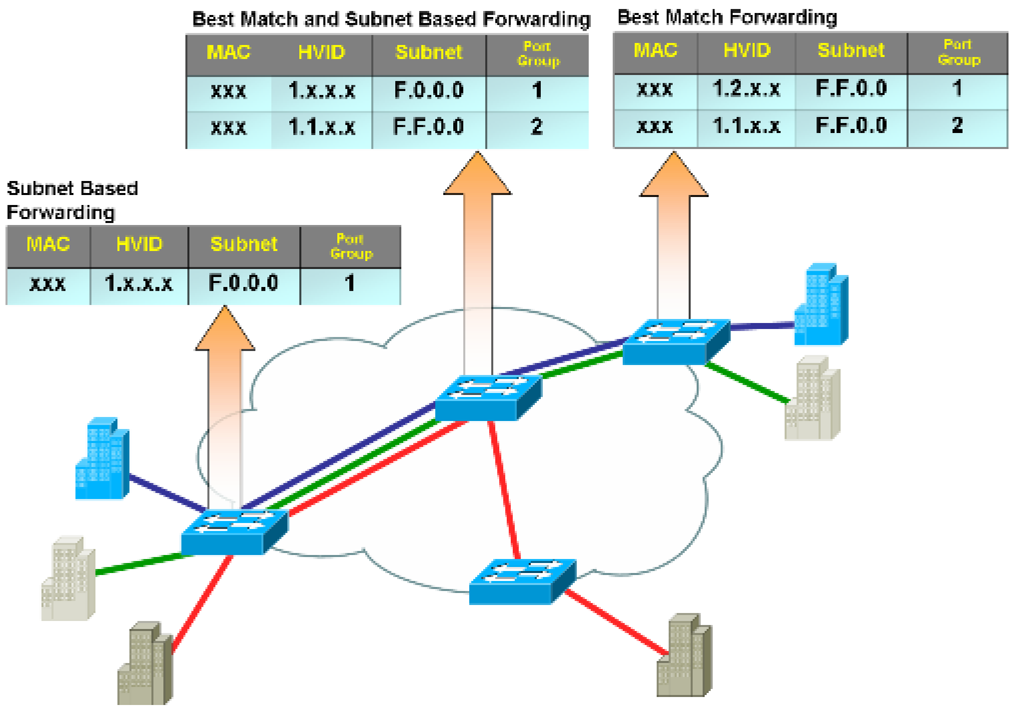
In PBB and PBT switches at the edge of the carrier network **encapsulate customer traffic within an 802.1ah frame**. The carrier network core is only responsible for transporting frames from an edge device to another, alleviating Q-in-Q's issue of forwarding table scalability. The same feature – assigning a MAC address per edge device, not per service – creates a scalability issue for multipoint services. Multipoint services require full mesh connectivity between edge devices, a very inefficient method as all frames are duplicated at the root nodes, rather than at the optimal point as in VLAN connectivity. Moreover the need to create forwarding entries for each unicast connection within the full mesh (as opposed to a single VLAN forwarding tree in the case of VLAN connectivity) will quickly become unsustainable as multipoint services become predominant in the near future.

Furthermore, the addition of a MAC header augments the frame size by about 128 bits, a significant overhead given the small size (64 byte) of real-time application (e.g. voice and video) packets.

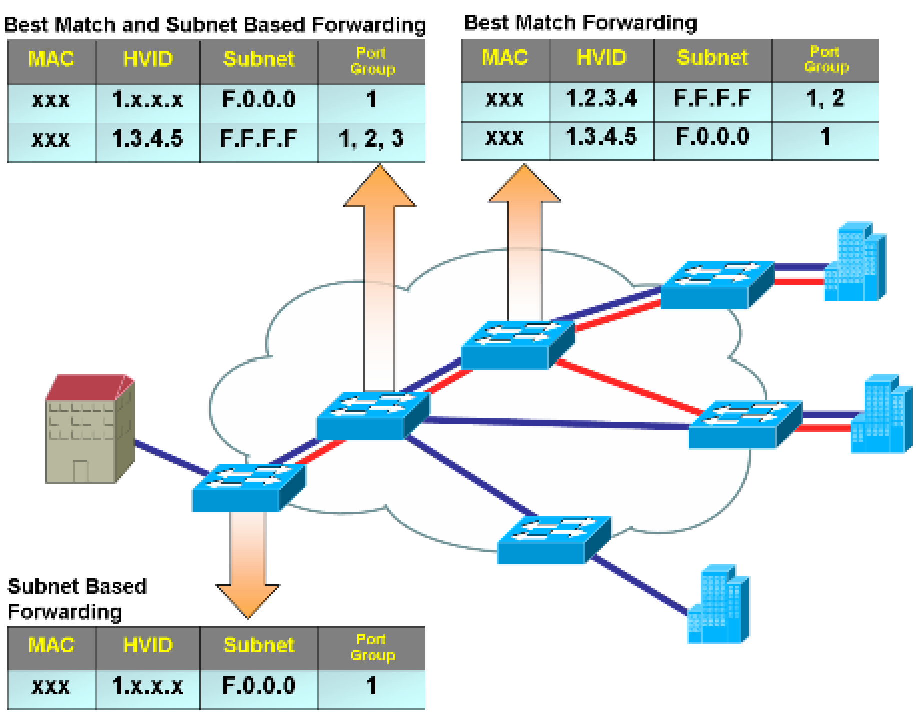
## *HVLAN*




HVLAN Frame Format




Point-to-Point Scenario




Point-to-Multipoint Scenario

Accordingly, there is a long felt need for leveraging the forwarding efficiency of VLAN networking, while at the same time solving its addressing space scalability issues described previously. Increasing the VLAN tag size would mean bigger forwarding table, longer forwarding table entries, and a modification of current mass-market Ethernet chips, requirements that are not vital to the enterprise world.

HVLAN introduces **hierarchy into the VLAN tag**, in a way somewhat similar to classless subnets in the Internet Protocol with Classless Inter-Domain Routing (CIDR). Consequently forwarding at each node uses a “best match” approach that substantially reduces the number of forwarding entries in core switches. Additionally HVLAN removes the need for encapsulation in many cases, reducing the overall transport overhead. The proposed HVLAN frame format is as follows:

A full description of the HVLAN header can be found in , the most important field being the HVID. When traversing the carrier’s Ethernet network, the HVLAN frames can be forwarded using HVID only, MAC address only or a combination of both. There is an explicit bit in an HVLAN frame that prevents the carrier’s core switches from learning the HVLAN frames' MAC address when unnecessary. To understand HVLAN operation, consider a scenario (see diagram) which illustrates the provision of 3 point-to-point services (blue, green, and red) over an HVLAN network. The diagram shows all forwarding table entries needed to transport the 3 services. Only forwarding entries for one direction (left-to-right) are displayed, similar entries implement the other direction.

**Point-to-point services** are provisioned using a unique HVID per service. Planning HVIDs wisely enables summarization (as shown at the leftmost edge device) and reduces the number of forwarding entries to a strict minimum; the network now scales to support millions of point-to-point services with minimum packet overhead (it can be noted that no encapsulation was used, frames were forwarded using HVID only).

A further example (see diagram) shows HVLAN operation in the case of **point-to-multipoint services** (e.g. IPTV). The diagram shows all forwarding table entries needed to transport the 2 multipoint services (red and blue) from a server (left) to 3 clients (right).

As with point-to-point services, point-to-multipoint services are provisioned using a unique HVID per service. Encapsulation is not required and frames can be forwarded using HVID only. Summarization of HVIDs reduces the size of forwarding tables and creates scalability. Millions of point-to-multipoint services can be provided. The case of multipoint-to-multipoint is handled by HVLAN using encapsulation and provider MAC addresses. A full description of HVLAN multipoint-to-multipoint operation is provided in .

## Conclusion

Hierarchical VLAN is a proposed extension to VLAN which, like PBB and PBT, turns cost-efficient Ethernet into a flexible, carrier-grade transport technology. Unlike other technologies HVLAN uses the mature VLAN functionality to support all connectivity schemes - point-to-point, point-to-multipoint, and multipoint-to-multipoint. To achieve this it uses a hierarchical VLAN allocation technique to allow summarization and reduce the number of forwarding table entries within the carrier network switches.

HVLAN is compatible with VLAN-related standards and is currently being discussed at ITU-T and IEEE with the aim of getting it standardized.

# Chapter 11

# MTOSI and Parlay Group

## MTOSI

In telecommunications, **Multi-Technology Operations System Interface** (MTOSI) is a standard for implementing interfaces between OSSs. Service providers (carriers) use multiple Operational Support Systems (OSS) to manage complex networks. Since the various parts of the network must interact, so must the OSSs. It is standardized by the Telemanagement Forum (TM Forum). The TMF NGOSS provides a set of reference models that aid in analyzing and designing next generation BSS and OSS solutions that may utilize the MTOSI interface specifications.

A visual representation of the TMF MTOSI offered SOA Service Candidates is shown in the figure.

The MTOSI specifications are produced by the TM Forum mTOP program.

MTOSI standard is a unified open interface that can be used among multiple types of management systems to provide network and service management. MTOSI standard covers all communication technologies (from layer 1, e.g., SONET/SDH, through higher layer technologies such as VoIP).

MTOSI facilitates application-to-application inter-working, reduces time to deployment, and lowers the cost of ownership of systems.

Figure 1

Current management and support system implementations employ diverse middleware technologies, a reality that is not likely to change in the immediate future. To be widely adopted, MTOSI cannot mandate specific middleware technologies for its implementation. Therefore the MTOSI interfaces are sufficiently abstract to be middleware neutral, yet rigorous enough that vendors can map them quickly to their middleware of choice. The CCV is the common middleware required to implement MTOSI.

CCV is a middleware abstraction that allows MTOSI interfaces to be bound to different middleware technologies as needed. By exploiting the expressive power of Web Services Description Language, MTOSI interfaces are composed of logical and physical definitions.

MTOSI standard offers a number of unique business advantages (1-4) as well as advantages applicable to any well-designed and well-supported interface standard (5-8):

1. MTOSI provides a standard interface between different systems for fulfillment and assurance functionality. In effect different instances of the same interface are reused at different reference points. Benefit: Knowledge can be re-used in the design of systems
2. MTOSI uses XML (eXtended Mark-Up Language) based messaging. Benefit: XML technology is widely accepted and used technology.
3. MTOSI provides rules for versioning and for vendor extensions to the XML messages. Benefit: When MTOSI is deployed, the server and consumer application ends of the interfaces can be upgraded independently. Also, when several vendors' equipment is deployed, the proprietary extensions are managed in a consistent manner.
4. MTOSI uses standard communication patterns to support business activities that can be implemented by a range of IT platforms and transport protocols. Benefit: The underlying platform can be changed the without propagating the change to the applications.
5. MTOSI allows service providers to implement management and support systems quickly. For example, without MTOSI, each of the four EMS providers in Figure 1 would need to define and agree upon a common interface (on a pair-wise basis), build the interface and then do interoperability testing. Benefit: MTOSI lowers the time and costs needed to integrate management and support system software from different suppliers.
6. MTOSI is designed to support service provider requirements for an open systems environment. Benefit: This allows service providers to more easily deploy management and support systems from multiple vendors and to replace existing ones. This increase in choice creates a more competitive environment for service providers, allowing them to choose products that best fit their functional and financial needs.
7. MTOSI encourages system integrators to pre-integrate products that are MTOSI-compliant. Benefit: This results in lower up-front costs and faster deployment for service providers.
8. MTOSI helps carriers to avoid wholesale replacements of legacy systems and instead allows them to introduce and integrate point applications that can address new solutions and services. Benefit: Allows a service provider to preserve its investment in legacy systems while still addressing the need to manage new technologies and services.

# Parlay Group

'***Parlay/OSA'*** was an open API for the telephone network. It was developed by The Parlay Group, which worked closely with ETSI and 3GPP, which all co-publish it. Within 3GPP, Parlay is part of Open Services Architecture.

## *Overview*

The **Parlay** Group was a technical industry consortium (founded 1998, ended around 2007) that specifies APIs for the telephone network. These APIs enable the creation of services by organizations both inside and outside of the traditional carrier environment. In fact, it is hoped that services can be created by IT developers, rather than telephony experts.

**Important Parlay APIs include**: call control, conferencing, user interaction (audio and text messaging, SMS/MMS), and billing. The APIs are specified in the CORBA Interface definition language and WSDL. The use of CORBA enables remote access between the Parlay gateway and the application code. A set of Java mappings allow the APIs to be invoked locally as well. A major goal of the APIs is to be independent of the underlying telephony network technology (e.g. CDMA, GSM, landline SS7).

### Parlay X

In 2003 the Parlay Group released a new set of web services called Parlay X. These are a much simpler set of APIs intended to be used by a larger community of developers. The Parlay X web services include Third Party Call Control (3PCC), location and simple payment. The Parlay X specifications complement the more powerful but more complex Parlay APIs. Parlay X implementations are now (September 2004) in commercial service from BT and Sprint.

Parlay work historically stems from the TINA effort. Parlay is somewhat related to JAIN, and is currently (early 2003) completely *unrelated* to the Service Creation Community.

## *Parlay Technology*

The objective of Parlay/OSA is to provide an API that is independent of the underlying networking technology and of the programming technology used to create new services. As a result the Parlay/OSA APIs are specified in UML. There are then a set of realizations, for specific programming environments:

- CORBA/IDL
- Java
- Web services specified by WSDL

## Parlay Framework

The role of the Parlay/OSA Framework was to provide a way for the network to authenticate applications using the Parlay/OSA API. The Framework also allows applications to discover the capabilities of the network, and provides management functions for handling fault and overload situations.

This is to ensure to a telecom network operator that any applications using the Parlay API cannot affect the security or integrity of the network.

## Implementing Parlay

The Parlay/OSA specifications define an API, they do not say how the API is to be implemented.

The typical Parlay/OSA implementation adds a new network element - the Parlay/OSA Gateway, which implements the Framework. It may implement the individual service APIs, or may interact with other network elements such as switches to provide individual service capabilities such as call control or location. Some vendors treat the Parlay/OSA Gateway as a stand-alone network element (e.g., the Ericsson NRG, jNetX OSA/Parlay GW, AePONA Network Gateway (formerly known as Causeway), HERIT Parlay/Parlay X Gateway), others include this function in an IN Service Control Point (e.g., the Telcordia OSP).

## Parlay and Web Services

The Parlay X APIs define a set of simple telecom-related Web services. Parlay X Version 1, published in May 2003, defines web services for:

- Third Party Call, Network Initiated Third Party Call, Send SMS, Receive SMS, Send Message, Receive Message, Amount Charging, Volume Charging, User Status and Terminal Location

Parlay X Version 2.1, published in June 2006, defines web services for:

- Third Party Call, Call Notification, Short Messaging, Multimedia Messaging, Payment, Account Management, Terminal Status, Terminal Location, Call Handling, Audio Call, Multimedia Conference, Address List Management, and Presence.

The current draft specifications for Parlay X 3.0 as of September 2007 defines web services for:

- Third Party Call, Call Notification, Short messaging, Multimedia Messaging, Payment, Account Management, Terminal Status, Terminal Location, Call Handling, Audio Call, Multimedia Conference, Address List Management, Presence, Message Broadcast, Geocoding, Application Driven QoS, Device Capabilities and Configuration, Multimedia Streaming control, Multimedia Multicast Session Management.

# Chapter 12

# T-Carrier

Two Network Interface Units. On the left with a single card, the right with two

In telecommunications, **T-carrier**, sometimes abbreviated as *T-CXR*, is the generic designator for any of several digitally multiplexed telecommunications carrier systems originally developed by Bell Labs and used in North America, Japan, and South Korea.

The basic unit of the T-carrier system is the DS0, which has a transmission rate of 64 kbit/s, and is commonly used for one voice circuit.

The **E-carrier** system, where 'E' stands for European, is incompatible with the T-carrier (though cross compliant cards exist) and is used in most locations outside of North America, Japan, and Korea. It typically uses the **E1** line rate and the E3 line rate. The E2 line rate is less commonly used.

### *T1*

Existing frequency-division multiplexing carrier systems worked well for connections between distant cities, but required expensive modulators, demodulators and filters for every voice channel. For connections within metropolitan areas, Bell Labs in the late 1950s sought cheaper terminal equipment. Pulse-code modulation allowed sharing a coder and decoder among several voice trunks, so this method was chosen for the T1 system introduced into local use in 1961. In later decades, the cost of digital electronics declined to the point that an individual codec per voice channel became commonplace, but by then the other advantages of digital transmission had become entrenched.

The most common legacy of this system is the line rate speeds. "**T1**" now means any data circuit that runs at the original 1.544 Mbit/s line rate. Originally the T1 format carried 24 pulse-code modulated, time-division multiplexed speech signals each encoded in 64 kbit/s streams, leaving 8 kbit/s of framing information which facilitates the synchronization and demultiplexing at the receiver. T2 and T3 circuit channels carry multiple T1 channels multiplexed, resulting in transmission rates of 6.312 and 44.736 Mbit/s, respectively.

Supposedly, the 1.544 Mbit/s rate was chosen because tests done by AT&T Long Lines in Chicago were conducted underground. To accommodate loading coils, cable vault manholes were physically 2000 meter (6,600 ft) apart, and so the optimum bit rate was chosen empirically — the capacity was increased until the failure rate was unacceptable, then reduced to leave a margin. Companding allowed acceptable audio performance with only seven bits per PCM sample in this original T1/D1 system. The later D3 and D4 channel banks had an extended frame format, allowing eight bits per sample, reduced to seven every sixth sample or frame when one bit was "robbed" for signaling the state of the channel. The standard does not allow an all zero sample which would produce a long string of binary zeros and cause the repeaters to lose bit sync. However, when carrying data (Switched 56) there could be long strings of zeroes, so one bit per sample is set to "1" (jam bit 7) leaving 7 bits x 8,000 frames per second for data.

A more detailed understanding of how the rate of 1.544 Mbit/s was derived is as follows. (This explanation glosses over T1 voice communications, and deals mainly with the numbers involved.) Given that the telephone system nominal voiceband (including guardband) is 4,000 Hz, the required digital sampling rate is 8,000 Hz. Since each T1 frame contains 1 byte of voice data for each of the 24 channels, that system needs then 8,000 frames per second to maintain those 24 simultaneous voice channels. Because each frame of a T1 is 193 bits in length (24 channels X 8 bits per channel + 1 framing bit = 193 bits), 8,000 frames per second is multiplied by 193 bits to yield a transfer rate of 1.544 Mbit/s (8,000 X 193 = 1,544,000).

Initially, T1 used **Alternate Mark Inversion** (AMI) to reduce frequency bandwidth and eliminate the DC component of the signal. Later **B8ZS** became common practice. For AMI, each mark pulse had the opposite polarity of the previous one and each space was at a level of zero, resulting in a three level signal which however only carried binary data. Similar British 23 channel systems at 1.536 Mbaud in the 1970s were equipped with ternary signal repeaters, in anticipation of using a 3B2T or 4B3T code to increase the number of voice channels in future, but in the 1980s the systems were merely replaced with European standard ones. American T-carriers could only work in AMI or B8ZS mode.

The AMI or B8ZS signal allowed a simple error rate measurement. The D bank in the central office could detect a bit with the wrong polarity, or "bipolarity violation" and sound an alarm. Later systems could count the number of violations and reframes and otherwise measure signal quality and allow a more sophisticated alarm indication signal system.

### Historical note on the 193-bit T1 frame

The decision to use a 193-bit frame was made in 1958. To allow for the identification of information bits within a frame, two alternatives were considered. Assign (a) just one extra bit, or (b) additional 8 bits per frame. The 8-bit choice is cleaner, resulting in a 200-bit frame, 25 8-bit **channels**, of which 24 are traffic and 1 8-bit channel available for operations, administration, and maintenance (**OA&M**). AT&T chose the single bit per frame not to reduce the required bit rate (1.544 vs 1.6 Mbit/s), but because AT&T Marketing worried that "if 8 bits were chosen for OA&M function, someone would then try to sell this as a voice channel and you wind up with nothing."

Soon after commercial success of T1 in 1962, the T1 engineering team realized the mistake of having only one bit to serve the increasing demand for **housekeeping** functions. They petitioned AT&T management to change to 8-bit framing. This was flatly turned down because it would make installed systems obsolete.

Having this hindsight, some ten years later, **CEPT** chose 8 bits for framing the European **E1**.

## *Higher T*

In the late 1960s and early 1970s Bell Labs developed higher rate systems. T-1C with a more sophisticated modulation scheme carried 3 Mbit/s, on those balanced pair cables that could support it. T-2 carried 6.312 Mbit/s, requiring a special low-capacitance cable with foam insulation. This was standard for Picturephone. T-4 and T-5 used coaxial cables, similar to the old L-carriers used by AT&T Long Lines. TD microwave radio relay systems were also fitted with high rate modems to allow them to carry a DS1 signal in a portion of their FM spectrum that had too poor quality for voice service. Later they carried DS3 and DS4 signals. During the 80's companies such as RLH Industries, Inc. developed T1 over optical fiber. The industry soon developed and evolved with multiplexed T1 transmission schemes.

## *Digital signal crossconnect*

DS1 signals are interconnected typically at Central Office locations at a common metallic cross-connect point known as a DSX-1. A DS1 signal at a DSX-1 is measured typically at 6 Volts Peak-to-peak (0dBdsx signal level at 772 kHz Nyquist) at plus or minus 1.2 volts to permit easy interconnection of DS1 equipment NCI Code=04DS9/ /). When a DS1 is transported over metallic outside plant cable, the signal travels over conditioned cable pairs known as a T1 span. A T1 span can have up to -130 Volts of DC power superimposed on the associated four wire cable pairs to line or "Span" power line repeaters, and T1 NIU's (T1 Smartjacks). T1 span repeaters are typically engineered up to 6,000 feet (1,800 m) apart, depending on cable gauge, and at no more than 36 dB of loss before requiring a repeated span. There can be no cable bridge taps across any pairs.

T1 copper spans are being replaced by optical transport systems, but if a copper (Metallic) span is used, the T1 is typically carried over an HDSL encoded copper line. Four wire HDSL does not require as many repeaters as conventional T1 spans. Newer two wire HDSL (HDSL-2) equipment transports a full 1.544 Mbit/s T1 over a single copper wire pair up to approximately twelve thousand (12,000) feet (3.5 km), if all 24 gauge cable is used. HDSL-2 does not employ repeaters as does conventional four wire HDSL, or newer HDSL-4 systems.

One advantage of HDSL is its ability to operate with a limited number of bridge taps, with no tap being closer than 500 feet (150 m) from any HDSL transceiver. Both two or four wire HDSL equipment transmits and receives over the same cable wire pair, as compared to conventional T1 service that utilizes individual cable pairs for transmit or receive.

DS3 signals are rare except within buildings, where they are used for interconnections and as an intermediate step before being muxed onto a SONET circuit. This is because a T3 circuit can only go about 600 feet (180m) between repeaters. A customer who orders a DS3 usually receives a SONET circuit run into the building and a multiplexer mounted in a utility box. The DS3 is delivered in its familiar form, two coax cables (1 for send and 1 for receive) with BNC connectors on the ends.

Reference: ANSI T1.403//ANSI T1.231//ANSI T1.404//ANSI T1.510.

## *Bit robbing*

Twelve DS1 frames make up a single T1 Superframe (T1 SF). Each T1 Superframe is composed of two signaling frames. All T1 DS0 channels that employ in-band signaling will have its eighth bit over written, or "robbed" from the full 64 kbit/s DS0 payload, by either a logical ZERO or ONE bit to signify a circuit signaling state or condition. Hence robbed bit signaling will restrict a DS0 channel to a rate of only 56 kbit/s during two of the twelve DS1 frames that make up a T1 SF framed circuit. T1 SF framed circuits yield two independent signaling channels (A&B) T1 ESF framed circuits four signaling frames in a twenty four frame extended frame format that yield four independent signaling channels (A, B,C,& D).

NOTE: 56 kbit/s DS0 channels are associated with digital data service (DDS) services typically do not utilize the eighth bit of the DS0 as voice circuits that employ A&B out of band signaling. One exception is Switched 56 kbit/s DDS. In DDS, bit eight is used to identify DTE request to send (RTS) condition. With Switched 56 DDS, bit eight is pulsed (alternately set to logical ZERO and ONE) to transmit two state dial pulse signaling information between a SW56 DDS CSU/DSU, and a digital end office switch.

The incident use of robbed-bit signaling in North America has decreased significantly as a result of Signaling System Seven (SS7) on inter-office dial trunks. With SS7, the full 64 kbit/s DS0 channel is available for use on a connection, and allows 64 kbit/s, and 128 kbit/s ISDN data calls to exist over a switched trunk network connection if the supporting T1 carrier entity is optioned B8ZS (Clear Channel Capable).

REFERENCES: ANSI T1.403//The Book On ESF, Verilink Corporation, 1986//D4 Digital Channel Bank Family, Bell System Technical Journal, Nov 1982

## *Carrier pricing*

Carriers price DS1 lines in many different ways. However, most boil down to two simple components; local loop (the cost the local incumbent charges to transport the signal from the end user's central office, otherwise known as a CO, to the point of presence, otherwise known as a POP, of the carrier) and the port (the cost to access the telephone network or the Internet through the carrier's network). Typically, the port price is based upon access speed and yearly commitment level while the loop is based on geography. The farther the CO and POP, the more the loop costs.

The loop price has several components built into it, including the mileage calculation (performed in V/H coordinates, not standard GPS coordinates) and the telco piece. Each local Bell operating company - namely Verizon, AT&T, and Qwest - charge T-carriers different price per mile rates. Therefore, the price calculation has two distance steps: geomapping and the determination of local price arrangements.

While most carriers utilize a geographic pricing model as described above, some Competitive Local Exchange Carriers (CLECs), such as Covad and Integra Telecom, offer national pricing. Under this DS1 pricing model, a provider charges the same price in every geography it services. National pricing is an outgrowth of increased competition in the T-carrier market space and the commoditization of T-carrier products. Providers that have adopted a national pricing strategy may experience widely varying margins as their suppliers, the Bell operating companies (e.g., Verizon, AT&T and Qwest), maintain geographic pricing models, albeit at wholesale prices.

For voice DS1 lines, the calculation is mostly the same, except that the port (required for Internet access) is replaced by LDU (otherwise known as Long Distance Usage). Once the price of the loop is determined, only voice-related charges are added to the total. In short, the total price = loop + LDU x minutes used.

## *Notes*

*Note 1:* The designators for T-carrier in the North American Digital multiplex hierarchy correspond to the designators for the digital signal (DS) level hierarchy.

*Note 2:* T-carrier systems were originally designed to transmit digitized voice signals. Current applications also include digital data transmission.

*Note 3:* Historically, if an "F" precedes the "T", optical fiber cables are utilized at the same rates.

*Note 4:* The North American and Japanese hierarchies are based on multiplexing 24 voice-frequency channels and multiples thereof, whereas the European hierarchy is based on multiplexing 32 voice-frequency channels and multiples thereof. *See table below.*

*Note 5:* Will be directed to this table by certain Network+ books. *See table below.*

| T-carrier and E-Carrier Systems | North American | Japanese | European (CEPT) |
|---|---|---|---|
| **Level zero (Channel data rate)** | 64 kbit/s (DS0) | 64 kbit/s | 64 kbit/s |
| **First level** | 1.544 Mbit/s (DS1) (24 user channels) (T1) | 1.544 Mbit/s (24 user channels) | 2.048 Mbit/s (32 user channels) (E1) |
| **(Intermediate level, US. hierarchy only)** | 3.152 Mbit/s (DS1C) (48 Ch.) | – | – |
| **Second level** | 6.312 Mbit/s (DS2) (96 Ch.) | 6.312 Mbit/s (96 Ch.), or 7.786 Mbit/s (120 Ch.) | 8.448 Mbit/s (128 Ch.) (E2) |
| **Third level** | 44.736 Mbit/s (DS3) (672 Ch.) (T3) | 32.064 Mbit/s (480 Ch.) | 34.368 Mbit/s (512 Ch.) (E3) |
| **Fourth level** | 274.176 Mbit/s (DS4) (4032 Ch.) | 97.728 Mbit/s (1440 Ch.) | 139.264 Mbit/s (2048 Ch.) (E4) |
| **Fifth level** | 400.352 Mbit/s (DS5) (5760 Ch.) | 565.148 Mbit/s (8192 Ch.) | 565.148 Mbit/s (8192 Ch.) (E5) |

*Note 1:* The DS designations are used in connection with the North American hierarchy only. Strictly speaking, a DS1 is the data carried on a T1 circuit, and likewise for a DS3 and a T3, but in practice the terms are used interchangeably.

*Note 2:* There are other data rates in use, e.g., military systems that operate at six and eight times the DS1 rate. At least one manufacturer has a commercial system that operates at 90 Mbit/s, twice the DS3 rate. New systems, which take advantage of the high data rates offered by optical communications links, are also deployed or are under development. Higher data rates are now often achieved by using synchronous optical networking (SONET) or synchronous digital hierarchy (SDH).

*Note 3:* A DS3 is delivered native on a copper trunk. DS3 may be converted to an optical fiber run when needing longer distances between termination points. When a DS3 is delivered over fiber it is still an analog type trunk connection at the termination points. When delivering data over an OC3 or greater SONET is used. A DS3 transported over SONET is encapsulated in a STS-1 SONET channel. An OC-3 SONET link contains 3 STS-1s, and therefore may carry 3 DS3s. Likewise, OC-12, OC-48, and OC-192 may carry 12, 48, and 192 DS3s respectively.

# Chapter 13

# TIA/EIA-568

**TIA/EIA-568** is a set of three telecommunications standards from the Telecommunications Industry Association, a 1988 offshoot of the EIA. The standards address commercial building cabling for telecom products and services. The three standards are formally titled ANSI/TIA/EIA-568-B.1-2001, -B.2-2001, and -B.3-2001.

The TIA/EIA-568-B standards were first published in 2001. They supersede the TIA/EIA-568-A standards set, which are now obsolete. They themselves have now been superseded by TIA/EIA-568-C.

Perhaps the best known features of TIA/EIA-568-B.1-2001 are the pin/pair assignments for eight-conductor 100-ohm balanced twisted pair cabling. These assignments are named T568A and T568B, and are frequently referred to (erroneously) as TIA/EIA-568A and TIA/EIA-568B.

## *History*

TIA/EIA-568-B was developed through the efforts of more than 60 contributing organizations including manufacturers, end-users, and consultants. Work on the standard began with the Electronic Industries Alliance (EIA), a standards organization, to define standards for telecommunications cabling systems. EIA agreed to develop a set of standards, and formed the TR-42 committee, with nine subcommittees to perform the work.

The first revision of the standard, TIA/EIA-568-A.1-1991 was released in 1991, and was updated in 1995. The demands placed upon commercial wiring systems increased dramatically over this period due to the adoption of personal computers and data communication networks and advances in those technologies. The development of high-performance twisted pair cabling and the popularization of fiber optic cables also drove significant change in the standards, which were eventually superseded by the current TIA/EIA-568-B set.

### *Goals*

TIA/EIA-568-B attempts to define structured cabling standards that will enable the design and implementation of structured cabling systems for commercial buildings, and between buildings in campus environments. The bulk of the standards define cabling types, distances, connectors, cable system architectures, cable termination standards and performance characteristics, cable installation requirements and methods of testing installed cable. The main standard, TIA/EIA-568-B.1 defines general requirements, while -568-B.2 focuses on components of balanced twisted-pair cable systems and -568-B.3 addresses components of fiber optic cable systems.

The intent of these standards is to provide recommended practices for the design and installation of cabling systems that will support a wide variety of existing and future services. Developers hope the standards will provide a lifespan for commercial cabling systems in excess of ten years. This effort has been largely successful, as evidenced by the definition of category 5 cabling in 1991, a cabling standard that (mostly) satisfied cabling requirements for 1000BASE-T, released in 1999. Thus, the standardization process can reasonably be said to have provided at least a nine-year lifespan for premises cabling, and arguably a longer one.

All these documents accompany related standards that define commercial pathways and spaces (569-A), residential cabling (570-A), administration standards (606), grounding and bonding (607) and outside plant cabling (758).

### *Structured cable system topologies*

TIA/EIA-568-B defines a hierarchical cable system architecture, in which a main cross-connect (MCC) is connected via a star topology across backbone cabling to intermediate cross-connects (ICC) and horizontal cross-connects (HCC). Telecommunications design traditions utilized a similar topology, and many people refer to cross-connects by their older, nonstandard names: "distribution frames" (with the various hierarchies called MDFs, IDFs and wiring closets). Backbone cabling is also used to interconnect entrance facilities (such as telco demarcation points) to the main cross-connect. Maximum allowable backbone fibre distances vary between 300m and 3000m, depending upon the cable type and use.

Horizontal cross-connects provide a point for the consolidation of all horizontal cabling, which extends in a star topology to individual work areas such as cubicles and offices. Under TIA/EIA-568-B, maximum allowable horizontal cable distance is 90m of installed cabling, whether fibre or twisted-pair, with 100m of maximum total length including patch cords. No patch cord should be longer than 5m. Optional consolidation points are allowable in horizontal cables, often appropriate for open-plan office layouts where consolidation points or media converters may connect cables to several desks or via partitions.

At the work area, equipment is connected by patch cords to horizontal cabling terminated at jackpoints.

TIA/EIA-568-B also defines characteristics and cabling requirements for entrance facilities, equipment rooms and telecommunications room.

### *T568A and T568B termination*

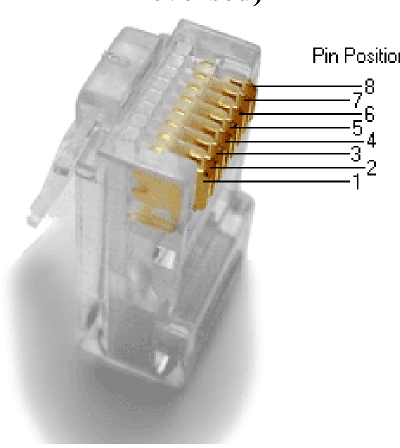
Perhaps the widest known and most discussed feature of TIA/EIA-568-B.1-2001 is the definition of pin/pair assignments for eight-conductor 100-ohm balanced twisted-pair cabling, such as Category 3, Category 5 and Category 6 unshielded twisted-pair (UTP) cables. These assignments are named T568A and T568B and they define the pinout, or order of connections, for wires in 8P8C (often incorrectly referred to as RJ45) eight-pin modular connector plugs and sockets. Although these definitions consume only one of the 468 pages in the standards documents, a disproportionate amount of attention is paid to them. This is because cables that are terminated with differing standards on each end will not function normally.

TIA/EIA-568-B specifies that horizontal cables should be terminated using the T568A pin/pair assignments, "or, optionally, per [T568B] if necessary to accommodate certain 8-pin cabling systems." Despite this instruction, many organizations continue to implement T568B for various reasons, chiefly associated with tradition (T568B is equivalent to AT&T 258A). The United States National Communication Systems Federal Telecommunications Recommendations do not recognize T568B.

The primary color of pair one is blue, pair two is orange, pair three is green and pair four is brown. Each pair consists of one conductor of solid color, and a second conductor which is white with a stripe of the same color. The specific assignments of pairs to connector pins varies between the T568A and T568B standards.

### Wiring

| Pin | T568A Pair | T568B Pair | Wire | T568A Color | T568B Color | Pins on plug face (socket is reversed) |
|---|---|---|---|---|---|---|
| 1 | 3 | 2 | tip | white/green stripe | white/orange stripe | |
| 2 | 3 | 2 | ring | green solid | orange solid | |
| 3 | 2 | 3 | tip | white/orange stripe | white/green stripe | |
| 4 | 1 | 1 | ring | blue solid | blue solid | |



| | | | | | |
|---|---|---|---|---|---|
| 5 | 1 | 1 | tip | white/blue stripe | white/blue stripe |
| 6 | 2 | 3 | ring | orange solid | green solid |
| 7 | 4 | 4 | tip | white/brown stripe | white/brown stripe |
| 8 | 4 | 4 | ring | brown solid | brown solid |

Note that the only difference between T568A and T568B is that pairs 2 and 3 (orange and green) are swapped. Both configurations wire the pins "straight through", i.e., pins 1 through 8 on one end are connected to pins 1 through 8 on the other end. Also, the same sets of pins are paired in both configurations: pins 1 and 2 form a pair, as do 3 and 6, 4 and 5, and 7 and 8. One can use cables wired according to either configuration in the same installation without significant problem; problems involving crosstalk can occur (which is normally minimized by correctly twisting a pair together), but are usually insignificant in all but the most stringent specifications such as Category 6 cable. The primary thing one has to be careful of is not to accidentally wire the ends of the **same cable** according to **different configurations** (unless one intends to create an Ethernet crossover cable).

### Use for T1 connectivity

In T1 service, the pairs 1 and 3 (T568A) are used, and the USOC-8 jack is wired as per spec RJ-48C. The Telco termination jack is often wired to spec RJ-48X, which provides for a Transmit-to-Receive loopback when the plug is withdrawn.

Vendor cables are often wired with Tip and Ring reversed—i.e. pins 1 and 2 reversed, or pins 4 and 5 reversed. This has no effect on the signal quality of the T1 signal, which is fully differential, and uses the Alternate Mark Inversion (AMI) signaling scheme.

### Backwards compatibility

Because pair 1 connects to the center pins (4 and 5) of the 8P8C connector in both T568A and T568B, both standards are compatible with the first line of RJ11, RJ14, RJ25, and RJ61 connectors that all have the first pair in the center pins of these connectors.

If the second line of an RJ14, RJ25 or RJ61 plug is used, it connects to pair 2 (orange/white) of jacks wired to T568A but to pair 3 (green/white) in jacks wired to T568B. This makes T568B potentially confusing in telephone applications.

Because of different pin pairings, the RJ25 and RJ61 plugs cannot pick up lines 3 or 4 from either T568A or T568B without splitting pairs. This would most likely result in unacceptable levels of hum, crosstalk and noise.

## Theory

The original idea in wiring modular connectors, which you see exemplified in the registered jacks, was that the first pair would go in the center positions, the next pair on the next outermost ones, and so on. Also, signal shielding would be optimized by alternating the "live" and "earthy" pins of each pair. As you can see, the TIA/EIA-568-B terminations vary a little bit from this concept. That's because on the 8 position connector, this results in a pinout in which the outermost pair are too far apart to meet the electrical requirements of high-speed LAN protocols.

## Chapter 14

# High-Level Data Link Control

**High-Level Data Link Control** (**HDLC**) is a bit-oriented synchronous data link layer protocol developed by the International Organization for Standardization (ISO). The original ISO standards for HDLC are:

- ISO 3309 — Frame Structure
- ISO 4335 — Elements of Procedure
- ISO 6159 — Unbalanced Classes of Procedure
- ISO 6256 — Balanced Classes of Procedure

The current standard for HDLC is ISO 13239, which replaces all of those standards.

HDLC provides both connection-oriented and connectionless service.

HDLC can be used for point to multipoint connections, but is now used almost exclusively to connect one device to another, using what is known as Asynchronous Balanced Mode (ABM). The original master-slave modes Normal Response Mode (NRM) and Asynchronous Response Mode (ARM) are rarely used.

### *History*

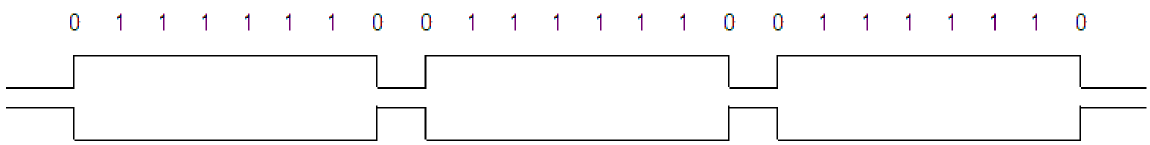
HDLC is based on IBM's SDLC protocol, which is the layer 2 protocol for IBM's Systems Network Architecture (SNA). It was extended and standardized by the ITU as LAP, while ANSI named their essentially identical version **ADCCP**.

Derivatives have since appeared in innumerable standards. It was adopted into the X.25 protocol stack as LAPB, into the V.42 protocol as LAPM, into the Frame Relay protocol stack as LAPF and into the ISDN protocol stack as LAPD. It was the inspiration for the IEEE 802.2 LLC protocol, and it is the basis for the framing mechanism used with the PPP on synchronous lines, as used by many servers to connect to a WAN, most commonly the Internet. A mildly different version is also used as the control channel for E-carrier (E1) and SONET multichannel telephone lines. Some vendors, such as Cisco, implemented protocols such as Cisco HDLC that used the low-level HDLC framing techniques but added a protocol field to the standard HDLC header. More importantly, HDLC is the default encapsulation for serial interfaces on Cisco routers. It has also been used on Tellabs DXX for destination of Trunk.

## *Framing*

HDLC frames can be transmitted over synchronous or asynchronous links. Those links have no mechanism to mark the beginning or end of a frame, so the beginning and end of each frame has to be identified. This is done by using a frame delimiter, or *flag*, which is a unique sequence of bits that is guaranteed not to be seen inside a frame. This sequence is '01111110', or, in hexadecimal notation, 0x7E. Each frame begins and ends with a frame delimiter. A frame delimiter at the end of a frame may also mark the start of the next frame. A sequence of 7 or more consecutive 1-bits within a frame will cause the frame to be aborted.

When no frames are being transmitted on a simplex or full-duplex synchronous link, a frame delimiter is continuously transmitted on the link. Using the standard NRZI encoding from bits to line levels (0 bit = transition, 1 bit = no transition), this generates one of two continuous waveforms, depending on the initial state:




This is used by modems to train and synchronize their clocks via phase-locked loops. Some protocols allow the 0-bit at the end of a frame delimiter to be shared with the start of the next frame delimiter, i.e. '011111101111110'.

For half-duplex or multi-drop communication, where several transmitters share a line, a receiver on the line will see continuous idling 1-bits in the inter-frame period when no transmitter is active.

Since the flag sequence could appear in user data, such sequences must be modified during transmission to keep the receiver from detecting a false frame delimiter. The receiver must also detect when this has occurred so that the original data stream can be restored before it is passed to higher layer protocols. This can be done using bit stuffing, in which a "0" is added before the occurrence of every "11111" in the data. The receiver, when detects these "11111" in the data, removes a "0" added by the transmitter.

### Synchronous framing

On synchronous links, this is done with bit stuffing. Any time that 5 consecutive 1-bits appear in the transmitted data, the data is paused and a 0-bit is transmitted. This ensures that no more than 5 consecutive 1-bits will be sent. The receiving device knows this is being done, and after seeing 5 1-bits in a row, a following 0-bit is stripped out of the received data. If the following bit is a 1-bit, the receiver has found a flag.

This also (assuming NRZI with transition for 0 encoding of the output) provides a minimum of one transition per 6 bit times during transmission of data, and one transition per 7 bit times during transmission of flag, so the receiver can stay in sync with the transmitter. Note however, that for this purpose encodings such as 8b/10b encoding are better suited.

HDLC transmits bytes of data with the least significant bit first (little-endian order).

### Asynchronous framing

When using asynchronous serial communication such as standard RS-232 serial ports, bits are sent in groups of 8, and bit-stuffing is inconvenient. Instead they use "control-octet transparency", also called "byte stuffing" or "octet stuffing". The frame boundary octet is 01111110, (7E in hexadecimal notation). A "control escape octet", has the bit sequence '01111101', (7D hexadecimal). If either of these two octets appears in the transmitted data, an escape octet is sent, followed by the original data octet with bit 5 inverted. For example, the data sequence "01111110" (7E hex) would be transmitted as "01111101 01011110" ("7D 5E" hex). Other reserved octet values (such as XON or XOFF) can be escaped in the same way if necessary.

## *Structure*

The contents of an HDLC frame are shown in the following table:

| Flag | Address | Control | Information | FCS | Flag |
|---|---|---|---|---|---|
| 8 bits | 8 or more bits | 8 or 16 bits | Variable length, 0 or more bits | 16 or 32 bits | 8 bits |

Note that the end flag of one frame may be (but does not have to be) the beginning (start) flag of the next frame.

Data is usually sent in multiples of 8 bits, but only some variants require this; others theoretically permit data alignments on other than 8-bit boundaries.

The frame check sequence (FCS) is a 16-bit CRC-CCITT or a 32-bit CRC-32 computed over the Address, Control, and Information fields. It provides a means by which the receiver can detect errors that may have been induced during the transmission of the frame, such as lost bits, flipped bits, and extraneous bits. However, given that the algorithms used to calculate the FCS are such that the probability of certain types of transmission errors going undetected increases with the length of the data being checked for errors, the FCS can implicitly limit the practical size of the frame.

If the receiver's calculation of the FCS does not match that of the sender's, indicating that the frame contains errors, the receiver can either send a negative acknowledge packet to the sender, or send nothing. After either receiving a negative acknowledge packet or timing out waiting for a positive acknowledge packet, the sender can retransmit the failed frame.

The FCS was implemented because many early communication links had a relatively high bit error rate, and the FCS could readily be computed by simple, fast circuitry or software. More effective forward error correction schemes are now widely used by other protocols.

### *Types of Stations (Computers), and Data Transfer Modes*

Synchronous Data Link Control (SDLC) was originally designed to connect one computer with multiple peripherals. The original "normal response mode" is a master-slave mode where the computer (or **primary terminal**) gives each peripheral (**secondary terminal**) permission to speak in turn. Because all communication is either to or from the primary terminal, frames include only one address, that of the secondary terminal; the primary terminal is not assigned an address. There is also a strong distinction between **commands** sent by the primary to a secondary, and **responses** sent by a secondary to the primary. Commands and responses are in fact indistinguishable; the only difference is the direction in which they are transmitted.

Normal response mode allows operation over half-duplex communication links, as long as the primary is aware that it may not transmit when it has given permission to a secondary.

Asynchronous response mode is an HDLC addition for use over full-duplex links. While retaining the primary/secondary distinction, it allows the secondary to transmit at any time.

Asynchronous balanced mode added the concept of a **combined terminal** which can act as both a primary and a secondary. There are some subtleties about this mode of operation; while many features of the protocol do not care whether they are in a command or response frame, some do, and the address field of a received frame must be examined to determine whether it contains a command (the address received is ours) or a response (the address received is that of the other terminal).

Some HDLC variants extend the address field to include both source and destination addresses, or an explicit command/response bit.

### *HDLC Operations, and Frame Types*

There are three fundamental types of HDLC frames.

- Information frames, or **I-frames**, transport user data from the network layer. In addition they can also include flow and error control information piggybacked on data.
- Supervisory Frames, or **S-frames**, are used for flow and error control whenever piggybacking is impossible or inappropriate, such as when a station does not have data to send. S-frames **do not** have information fields.
- Unnumbered frames, or **U-frames**, are used for various miscellaneous purposes, including link management. Some U-frames contain an information field, depending on the type.

The general format of the control field is:

HDLC control fields

| 7 6 5 | 4 | 3 2 1 | 0 | |
|---|---|---|---|---|
| N(R) Receive sequence no. | P/F | N(S) Send sequence no. | 0 | **I-frame** |
| N(R) Receive sequence no. | P/F | type 0 | 1 | **S-frame** |
| type | P/F | type 1 | 1 | **U-frame** |

There are also extended (2-byte) forms of I and S frames. Again, the least significant bit (rightmost in this table) is sent first.

Extended HDLC control fields

| 15 14 13 12 11 10 9 | 8 | 7 6 5 4 3 2 1 | 0 | |
|---|---|---|---|---|
| N(R) Receive sequence no. | P/F | N(S) Send sequence no. | 0 | **Extended I-frame** |
| N(R) Receive sequence no. | P/F | 0 0 0 0 type 0 | 1 | **Extended S-frame** |

## The P/F bit

Poll/Final is a single bit with two names. It is called Poll when set by the primary station to obtain a response from a secondary station, and Final when set by the secondary station to indicate a response or the end of transmission. In all other cases, the bit is clear.

The bit is used as a token that is passed back and forth between the stations. Only one token should exist at a time. The secondary only sends a Final when it has received a Poll from the primary. The primary only sends a Poll when it has received a Final back from the secondary, or after a timeout indicating that the bit has been lost.

- In NRM, possession of the poll token also grants the addressed secondary permission to transmit. The secondary sets the F-bit in its last response frame to give up permission to transmit. (It is equivalent to the word "Over" in radio voice procedure.)
- In ARM and ABM, the P bit forces a response. In these modes, the secondary need not wait for a poll to transmit, so need not wait to respond with a final bit.
- If no response is received to a P bit in a reasonable period of time, the primary station times out and sends P again.
- The P/F bit is at the heart of the basic **checkpoint retransmission** scheme that is required to implement HDLC; all other variants (such as the REJ S-frame) are optional and only serve to increase efficiency. Whenever a station receives a P/F bit, it may assume that any frames that it sent before it last transmitted the P/F bit and not yet acknowledged will never arrive, and so should be retransmitted.

When operating as a combined station, it is important to maintain the distinction between P and F bits, because there may be two checkpoint cycles operating simultaneously. A P bit arriving in a command from the remote station is not in response to our P bit; only an F bit arriving in a response is.

## N(R), the receive sequence number

Both I and S frames contain a receive sequence number N(R). N(R) provides a positive acknowledgement for the receipt of I-frames from the other side of the link. Its value is always the first frame *not* received; it acknowledges that all frames with N(S) values up to N(R)-1 (modulo 8 or modulo 128) have been received and indicates the N(S) of the next frame it expects to receive.

N(R) operates the same way whether it is part of a command or response. A combined station only has one sequence number space.

## N(S), the sequence number of the sent frame

This is incremented for successive I-frames, modulo 8 or modulo 128. Depending on the number of bits in the sequence number, up to 7 or 127 I-frames may be awaiting acknowledgment at any time.

## I-Frames (user data)

Information frames, or **I-frames**, transport user data from the network layer. In addition they also include flow and error control information piggybacked on data. The sub-fields in the control field define these functions.

The least significant bit (first transmitted) defines the frame type. 0 means an I-frame. Except for the interpretation of the P/F field, there is no difference between a command I frame and a response I frame; when P/F is 0, the two forms are exactly equivalent.

## S-Frames (control)

Supervisory Frames, or **S-frames**, are used for flow and error control whenever piggybacking is impossible or inappropriate, such as when a station does not have data to send. S-frames **do not** have information fields.

The S-frame control field includes a leading "10" indicating that it is an S-frame. This is followed by a 2-bit type, a poll/final bit, and a sequence number. If 7-bit sequence numbers are used, there is also a 4-bit padding field.

The first 2 bits mean it is an S-frame. All S frames include a P/F bit and a receive sequence number as described above. Except for the interpretation of the P/F field, there is no difference between a command S frame and a response S frame; when P/F is 0, the two forms are exactly equivalent.

The 2-bit type field encodes the type of S frame.

### Receive Ready (RR)

- Indicate that the sender is ready to receive more data (cancels the effect of a previous RNR).
- Send this packet if you need to send a packet but have no I frame to send.
- A primary station can send this with the P-bit set to solicit data from a secondary station.
- A secondary terminal can use this with the F-bit set to respond to a poll if it has no data to send.

### Receive Not Ready (RNR)

- Acknowledge some packets and request no more be sent until further notice.
- Can be used like RR with P bit set to solicit the status of a secondary station.
- Can be used like RR with F bit set to respond to a poll if the station is busy.

### Reject (REJ)

- Requests immediate retransmission starting with N(R).
- Sent in response to an observed sequence number gap. After seeing I1/I2/I3/I5, send REJ4.
- Optional to generate; a working implementation can use only RR.

### Selective Reject (SREJ)

- Requests retransmission of only the frame N(r).
- Not supported by all HDLC variants.
- Optional to generate; a working implementation can use only RR, or only RR and REJ.

## U-Frames

Unnumbered frames, or **U-frames**, are used for link management, and can also be used to transfer user data. They exchange session management and control information between connected devices, and some U-frames contain an information field, used for system management information or user data. The first 2 bits (11) mean it is a U-frame. The 5 type bits (2 before P/F bit and 3 bit after P/F bit) can create 32 different types of U-frame

- Mode settings (SNRM, SNRME, SARM, SARME, SABM, SABME, UA, DM, RIM, SIM, RD, DISC)
- Information Transfer (UP, UI)
- Recovery (FRMR, RSET)
    - Invalid Control Field
    - Data Field Too Long

- Data field not allowed with received Frame Type
    - Invalid Receive Count
- Miscellaneous (XID, TEST)

## *Link Configurations*

Link configurations can be categorized as being either:

- *Unbalanced*, which consists of one primary terminal, and one or more secondary terminals.
- *Balanced*, which consists of two peer terminals.

The three link configurations are:

- *Normal Response Mode* (NRM) is an unbalanced configuration in which only the primary terminal may initiate data transfer. The secondary terminal transmits data only in response to commands from the primary terminal. The primary terminal polls the secondary terminal(s) to determine whether they have data to transmit, and then selects one to transmit.
- *Asynchronous Response Mode* (ARM) is an unbalanced configuration in which secondary terminals may transmit without permission from the primary terminal. However, the primary terminal still retains responsibility for line initialization, error recovery, and logical disconnect.
- *Asynchronous Balanced Mode* (ABM) is a balanced configuration in which either station may initiate the transmission.

An additional link configuration is *Disconnected mode*. This is the mode that a secondary station is in before it is initialized by the primary, or when it is explicitly disconnected. In this mode, the secondary responds to almost every frame other than a mode set command with a "Disconnected mode" response. The purpose of this mode is to allow the primary to reliably detect a secondary being powered off or otherwise reset..

## *HDLC Command and response repertoire*

- Commands (BALA, I, RR, RNR, (SNRM or SARM or SABM) DISC
- Responses (I, RR, RNR, UA, DM, FRMR)

### Basic Operations

- Initialization can be requested by either side. When the six-mode set-command is issued. This command:
    - Signals the other side that initialization is requested
    - Specifies the mode, NRM, ABM, ARM
    - Specifies whether 3 or 7 bit sequence numbers are in use.

The HDLC module on the other end transmits (UA) frame when the request is accepted. And if the request is rejected it sends (DM) disconnect mode frame.

## Functional Extensions (Options)

- For Switched Circuits
    - Commands: ADD - XID
    - Responses: ADD - XID, RD
- For 2-way Simultaneous commands & responses are ADD - REJ
- For Single Frame Retransmission commands & responses: ADD - SREJ
- For Information Commands & Responses: ADD - Ul
- For Initialization
    - Commands: ADD - SIM
    - Responses: ADD - RIM
- For Group Polling
    - Commands: ADD - UP
- Extended Addressing
- Delete Response I Frames
- Delete Command I Frames
- Extended Numbering
- For Mode Reset (ABM only) Commands are: ADD - RSET
- Data Link Test Commands & Responses are: ADD - TEST
- Request Disconnect. Responses are ADD - RD
- 32-bit FCS

### *HDLC Command/Response Repertoire*

| Type Of Frame | Name | Command/ Response | Description | Info | C-Field Format 7 6 5 4 3 2 1 0 |
|---|---|---|---|---|---|
| Information(I) | | C/R | User exchange data | | N(R) P/F N(S) 0 |
| Supervisory (S) | Receive Ready (RR) | C/R | Positive Acknowledgement | Ready to receive I-frame N(R) | N(R) P/F 0 0 0 1 |
| | Receive Not Ready (RNR) | C/R | Positive Acknowledgement | Not ready to receive | N(R) P/F 0 1 0 1 |
| | Reject (REJ) | C/R | Negative Acknowledgement | Retransmit starting with N(R) | N(R) P/F 1 0 0 1 |
| | Selective Reject (SREJ) | C/R | Negative Acknowledgement | Retransmit only N(R) | N(R) P/F 1 1 0 1 |

## Unnumbered Frames

Unnumbered frames are identified by the low two bits being 1. With the P/F flag, that leaves 5 bits as a frame type. Even though fewer than 32 values are in use, some types have different meanings depending on the direction they are sent: as a request or as a response. The relationship between the **DISC** (disconnect) command and the **RD** (request disconnect) response seems clear enough, but the reason for making **SARM** command numerically equal to the **DM** response is obscure.

| Name | Command/ Response | Description | Info | C-Field Format<br>**7 6 5 4 3 2 1 0** |
|---|---|---|---|---|
| Set normal response **SNRM** | C | Set mode | Use 3 bit sequence number | 1 0 0 P 0 0 1 1 |
| Set normal response extended mode **SNRME** | C | Set mode; extended | Use 7 bit sequence number | 1 1 0 P 1 1 1 1 |
| Set asynchronous response **SARM** | C | Set mode | Use 3 bit sequence number | 0 0 0 P 1 1 1 1 |
| Set asynchronous response extended mode **SARME** | C | Set mode; extended | Use 7 bit sequence number | 0 1 0 P 1 1 1 1 |
| Set asynchronous balanced mode **SABM** | C | Set mode | Use 3 bit sequence number | 0 0 1 P 1 1 1 1 |
| Set asynchronous balanced extended mode **SABME** | C | Set mode; extended | Use 7 bit sequence number | 0 1 1 P 1 1 1 1 |
| Set initialization mode **SIM** | C | Initialize link control function in the addressed station | | 0 0 0 P 0 1 1 1 |
| Disconnect **DISC** | C | Terminate logical link connection | Future I and S frames return **DM** | 0 1 0 P 0 0 1 1 |
| Unnumbered Acknowledgment **UA** | R | Acknowledge acceptance of one of the set-mode commands. | | 0 1 1 F 0 0 1 1 |
| Disconnect Mode **DM** | R | Responder in Disconnect Mode | mode set required | 0 0 0 F 1 1 1 1 |
| Request Disconnect **RD** | R | Solicitation for **DISC** Command | | 0 1 0 F 0 0 1 1 |
| Request Initialization | R | Initialization | Request for | 0 0 0 F 0 1 1 1 |

| | | | | |
|---|---|---|---|---|
| Mode **RIM** | | needed | **SIM** command | |
| Unnumbered Information **UI** | C/R | Unacknowledged data | has a payload | 0 0 0 P/F 0 0 1 1 |
| Unnumbered Poll **UP** | C | Used to solicit control information | | 0 0 1 P 0 0 1 1 |
| Reset **RSET** | C | Used for recovery | Resets N(R) but not N(S) | 1 0 0 P 1 1 1 1 |
| Exchange Identification **XID** | C/R | Used to Request/Report capabilities | | 1 0 1 P/F 1 1 1 1 |
| Test **TEST** | C/R | Exchange identical information fields for testing | | 1 1 1 P/F 0 0 1 1 |
| Frame Reject **FRMR** | R | Report receipt of unacceptable frame | | 1 0 0 F 0 1 1 1 |
| Nonreserved 0 **NR0** | C/R | Not standardized | For application use | 0 0 0 P/F 1 0 1 1 |
| Nonreserved 1 **NR1** | C/R | Not standardized | For application use | 1 0 0 P/F 1 0 1 1 |
| Nonreserved 2 **NR2** | C/R | Not standardized | For application use | 0 1 0 P/F 1 0 1 1 |
| Nonreserved 3 **NR3** | C/R | Not standardized | For application use | 1 1 0 P/F 1 0 1 1 |
| Configure for test **CFGR** | C/R | Not part of HDLC | Was part of SDLC | 1 1 0 P/F 0 1 1 1 |
| Beacon **BCN** | R | Not part of HDLC | Was part of SDLC | 1 1 1 F 1 1 1 1 |

The UI, XID and TEST frames contain a payload, and can be used as both commands and responses.

- A UI frame contains user information, but unlike an I frame it is not acknowledged or retransmitted if lost.
- The XID frame is used to exchange terminal capabilities. IBM Systems Network Architecture defined one format, but the variant defined in ISO 8885 is more commonly used. A primary advertisies its capabilities with an XID command, and a secondary returns an XID response.
- The TEST frame is simply a ping command for debugging purposes. The payload of the TEST command is returned in the TEST response.

The FRMR frame contains a payload describing the unacceptable frame. The first 1 or 2 bytes are a copy of the rejected control field, the next 1 or 2 contain the current send and receive sequence numbers, and the following 4 or 5 bits indicate the reason for the rejection.

# Chapter 15

# E-carrier and Parlay X

## E-carrier

In digital telecommunications, where a single physical wire pair can be used to carry many simultaneous voice conversations by time-division multiplexing, worldwide standards have been created and deployed. The European Conference of Postal and Telecommunications Administrations (CEPT) originally standardized the **E-carrier** system, which revised and improved the earlier American T-carrier technology, and this has now been adopted by the International Telecommunication Union Telecommunication Standardization Sector (ITU-T). This is now widely used in almost all countries outside the USA, Canada and Japan.

The E-carrier standards form part of the Plesiochronous Digital Hierarchy (PDH) where groups of E1 circuits may be bundled onto higher capacity E3 links between telephone exchanges or countries. This allows a network operator to provide a private end-to-end E1 circuit between customers in different countries that share single high capacity links in between.

In practice, only E1 and E3 versions are used. Physically E1 is transmitted as 32 timeslots and E3 512 timeslots, but one is used for framing and typically one allocated for signalling call setup and tear down. Unlike Internet data services, E-carrier systems permanently allocate capacity for a voice call for its entire duration. This ensures high call quality because the transmission arrives with the same short delay (latency) and capacity at all times.

E1 circuits are very common in most telephone exchanges and are used to connect to medium and large companies, to remote exchanges and in many cases between exchanges. E3 lines are used between exchanges, operators and/or countries, and have a transmission speed of 34.368 Mbit/s.

### *E1*

An E1 link operates over two separate sets of wires, usually twisted pair cable. A nominal 3 volt peak signal is encoded with pulses using a method that avoids long periods without polarity changes. The line data rate is 2.048 Mbit/s (full duplex, i.e. 2.048 Mbit/s downstream and 2.048 Mbit/s upstream) which is split into 32 timeslots, each being allocated 8 bits in turn. Thus each timeslot sends and receives an 8-bit PCM sample, usually encoded according to A-law algorithm, 8000 times per second (8 x 8000 x 32 = 2,048,000). This is ideal for voice telephone calls where the voice is sampled into an 8 bit number at that data rate and reconstructed at the other end. The timeslots are numbered from 0 to 31.

One timeslot (TS0) is reserved for framing purposes, and alternately transmits a fixed pattern. This allows the receiver to lock onto the start of each frame and match up each channel in turn. The standards allow for a full Cyclic Redundancy Check to be performed across all bits transmitted in each frame, to detect if the circuit is losing bits (information), but this is not always used.

One timeslot (TS16) is often reserved for signalling purposes, to control call setup and teardown according to one of several standard telecommunications protocols. This includes Channel Associated Signaling (CAS) where a set of bits is used to replicate opening and closing the circuit (as if picking up the telephone receiver and pulsing digits on a rotary phone), or using tone signalling which is passed through on the voice circuits themselves. More recent systems used Common Channel Signaling (CCS) such as ISDN or Signalling System 7 (SS7) which send short encoded messages with more information about the call including caller ID, type of transmission required etc. ISDN is often used between the local telephone exchange and business premises, whilst SS7 is almost exclusively used between exchanges and operators. In theory, a single SS7 signaling timeslot can control up to 4096 circuits per signalling channel using a 12-bit Channel Identification Code (CIC), thus allowing slightly more efficient use of the overall transmission bandwidth because additional E1 links would use all 31 voice channels. ANSI uses a larger 14-bit CIC and so can accommodate up to 16,384 circuits. In most environments, multiple signalling channels would be used to provide redundancy in case of faults or outages.

Unlike the earlier T-carrier systems developed in North America, all 8 bits of each sample are available for each call. This allows the E1 systems to be used equally well for circuit switch data calls, without risking the loss of any information.

While the original CEPT standard G.703 specifies several options for the physical transmission, almost exclusively HDB3 format is used.

**Definition**

**Link** An unidirectional channel residing in one timeslot of a E1 or T1 Line, carrying 64 kbit/s (64'000 bit/s) raw digital data.

**Line** An unidirectional E1 or T1 physical connection.

**Trunk** A bidirectional E1 or T1 physical connection.

### *Hierarchy levels*

The PDH based on the E0 signal rate is designed so that each higher level can multiplex a set of lower level signals. Framed E1 is designed to carry 30 E0 data channels + 1 signalling channel, all other levels are designed to carry 4 signals from the level below. Because of the necessity for overhead bits, and justification bits to account for rate differences between sections of the network, each subsequent level has a capacity greater than would be expected from simply multiplying the lower level signal rate (so for example E2 is 8.448 Mbit/s and not 8.192 Mbit/s as one might expect when multiplying the E1 rate by 4).

Note, because bit interleaving is used, it is very difficult to demultiplex low level tributaries directly, requiring equipment to individually demultiplex every single level down to the one that is required.

| Signal | Rate |
|---|---|
| E0 | 64 kbit/s |
| E1 | 2.048 Mbit/s |
| E2 | 8.448 Mbit/s |
| E3 | 34.368 Mbit/s |
| E4 | 139.264 Mbit/s |
| E5 | 564.992 Mbit/s |

# Parlay X

**Parlay X** was a set of standard Web service APIs for the telephone network (fixed and mobile.). It's defunct and now replaced by OneAPI, which is the current valid standard from the GSM association for Telecom third party API.

It enables software developers to use the capabilities of an underlying network. The APIs are deliberately high level abstractions and designed to be simple to use. An application developer can, for example, invoke a single Web Service request to get the location of a mobile device or initiate a telephone call.

The Parlay X Web services are defined jointly by ETSI, the Parlay Group, and the Third Generation Partnership Program (3GPP). OMA has done the maintenance of the specifications for 3GPP release 8.

The APIs are defined using Web Service technology: interfaces are defined using WSDL 1.1 and conform with Web Services Interoperability (WS-I Basic Profile).

The APIs are published as a set of specifications.

| Parlay X 4.0 Specification | Available functionality |
|---|---|
| Part 1: "Common" | Definitions re-used across multiple Parlay X specifications |
| Part 2: "Third Party Call" | Creating and managing a call initiated by an application |
| Part 3: "Call Notification" | Handling calls initiated by a subscriber in the network. One variant (i.e. application interface) allows application to "direct" the handling of the call and the other simply provides a notification. |
| Part 4: "Short Messaging" | Receive and send SMS (including delivery receipts) |
| Part 5: "Multimedia Messaging" | Receive and send Multimedia Messages |
| Part 6: "Payment" | Payment reservations, pre-paid payments, and post-paid payments |
| Part 7: "Account Management" | Account querying, direct recharging and recharging through vouchers |
| Part 8: "Terminal Status" | Get the status (i.e. reachable, unreachable or busy) of a terminal |
| Part 9: "Terminal Location" | Getting location information about a terminal |
| Part 10: "Call Handling" | Specify how calls are to be handled for a specific number. There is no 'per-call interaction' with the application unlike in the Call Notification API. |
| Part 11: "Audio Call" | Provide multimedia message delivery and the dynamic management of the media involved for the call participants |
| Part 12: "Multimedia Conference" | Create a multimedia conference and the dynamic management of the participants involved |
| Part 13: "Address List Management" | Manage groups (aliases) of subscribers |
| Part 14: "Presence" | Presence information to be obtained about or registered for users used e.g. by Instance Messaging clients |
| Part 15: “Message Broadcast” | Send messages to all the fixed or mobile terminals in a specified geographical area |

| | |
|---|---|
| Part 16: “Geocoding” | Get the location address of a subscriber e.g. country, state, district, city, street, house number, additional information, and zip/postal code |
| Part 17: “Application-driven Quality of Service (QoS)” | Dynamically change the quality of service (e.g. bandwidth) available on end user network connection |
| Part 18: "Device Capabilities and Configuration" | Get information about device capabilities and push device configuration to a device |
| Part 19: “Multimedia Streaming Control” | Control streaming of multimedia to a subscriber e.g. to transfer stream between a user's terminals |
| Part 20: “Multimedia Multicast Session Management | Control a multicast session, its members and multimedia stream, and obtain channel presence information |
| Part 21: "Content management" | The content management web service enables uploading content into the network (or a third party content provider) and consuming content from the network (or a third party content provider). |
| Part 22: "Policy" | The Policy Web Service is defined to offer provisioning and evaluation functions for policies. |

In general Parlay X provides an abstraction of functionality exposed by the more complex, but functionally richer Parlay APIs. ETSI provide a set of (informative not normative) Parlay X to Parlay mapping documents.

Parlay X services have been rolled out by a number of telecom operators, including BT, Korea Telecom, T-Com, Mobilekom and Sprint.