Triple-Play Service Deployment

A Comprehensive Guide to Test, Measurement, and Service Assurance
Triple-Play Service Deployment

A Comprehensive Guide to
Test, Measurement, and Service Assurance
# Table of Contents

## Chapter 1: The Vision—Reliable Triple-Play Service Delivery
- The Competition .................................................. 2
- The Strategy ....................................................... 2
- The Building Blocks .............................................. 4
- The Ground Rules .................................................. 6
- The Basics—Testing Voice, Video, and Data ..................... 7
- Triple Play Requires Triple Testing ............................. 13

## Chapter 2: Triple-Play Service Delivery over Ethernet Networks ............................. 15
- Ethernet Networks: Carrier-grade requirements ............... 16
- Understanding the Inherent Benefits of Ethernet Technology .................................................. 18
  - Class of Service (CoS) Concept ................................ 21
  - CoS Prioritization Schemes ..................................... 22
- Triple-play Services within Ethernet Aggregation Networks .................................................. 25
  - Video .................................................................. 26
  - Voice over IP .......................................................... 28
  - Data .................................................................. 29
- Tools for Turning Up and Maintaining Carrier-Grade Networks .................................................. 30
  - Portable solutions .................................................. 30
  - Field Test Applications ............................................ 33
- Service Assurance Solution .......................................... 35
- The Total Solution .................................................... 36

## Chapter 3: Deploying and Troubleshooting Fiber Networks ........................................... 37
- Understanding FTTx networks and associated optical testing .................................................. 38
  - FTTx Architectures .................................................. 38
  - FTTP Infrastructure and Technology ......................... 41
  - Point-to-Multipoint Networks ................................. 42
  - PON Signals and Network Structure ......................... 43
- Implementing a PON test strategy ................................. 48
  - Test Strategy for characterizing connectorized passive optical networks ............................. 48
  - Stages 1 and 2: PON Feeder and distribution cable construction .................................................. 49
  - Stage 3: PON Frame Installation and Connection ........... 52
  - Stage 4: PON feeder cable acceptance test .................... 55
  - Stage 5: PON distribution cable acceptance test .......... 57
  - Stage 6: PON Equipment Installation and Testing ........ 59
  - Photonic Layer Testing of OLT and ONT Signals ........ 61
  - Stage 7: PON troubleshooting ................................... 62
  - Spliced PON testing variation versus connectorized PON .................................................. 65
  - Point-to-Point FTTH testing variation versus PON testing .................................................. 67
Preface

From headend through home, the technologies associated with delivering high bandwidth services to meet end users’ quality expectations evolve continuously. Compression techniques for video services are migrating from MPEG-2 to MPEG-4 and beyond. The Metro/Core network is transitioning from legacy SONET/SDH to next generation packet technology designed to transport voice, video, and data over an Application-Aware IP backbone. All of this while the Access Network, including the “last mile,” as we know it today is being extended throughout the consumer’s home or business with myriad in-home or premises distribution technologies. Reliable delivery of triple-play services all of the way to the consumer’s television or PC, however, remains key for service provider success.

JDSU offers you this guide as a starting point to understand best practices for test, measurement, and quality assurance for technologies and services that are indeed constantly changing. It represents the first in a series of resources that will assist the service provider in gaining an understanding of the challenges and associated solutions available to develop test and service assurance plans for triple-play networks. Future publications will address advances in technology (MPEG-4 and H.264 details, for example) and services (Interactive TV) and associated test and measurement requirements.

We invite you to contact JDSU at tripleplay@jdsu.com or call your local sales representative for further information about the concepts, technologies, and solutions presented in The Guide.
The JDSU *Triple-Play Service Deployment Guide* is the result of company-wide collaboration.

**Authors**
Assaji Aluwihare  
Jon Beckman  
Robert Flask  
Eli Kerch  
Jerome Laferriere  
Mirna Mekic  
Jim Nerschook  
Nisha Parbhakar  
Thad Ward  
John Williams

**Contributors**
Melissa Ashraf  
Sascha Chwalek  
Scott Haerr  
Tim Miller  
Kevin Oliver  
Todd Rapposelli  
Dave Rerko  
Andrew Sachs  
Paul Snead  
Ron Vogel  
Kevin Williams  
Tim Yount

**Editor**
Laurie Rerko

**Graphic Design and Production**
Jennifer Pitts  
Robert Taylor
Foreword

Broadband is booming and bringing us the triple play of voice, video, and data. Broadband subscribers are expected to grow from 227 million in 2006 to 417 million worldwide in 2010 and a rapidly growing portion of these are triple-play subscribers. The widespread rollout of broadband access networks to residential subscribers and the simultaneous emergence of IP and Ethernet as the access technologies of choice are revolutionizing the delivery of voice and data services to subscribers—and also enabling the delivery of video services, including IPTV, video on demand, video telephony, and online gaming. In an effort to increase average revenue per user (ARPU) to compensate for decreasing fixed line voice revenue, and to reduce churn among a selective subscriber base with plenty of options, service providers are betting on the triple play to stay ahead of their competition.

Around the world, the competitive lines separating network operators have disappeared. For many years, the cable company and the home’s coaxial input were associated strictly with television services, and today coax can serve as the interface to voice and data services as well; where once the twisted pair was associated with only POTS, today it can also be the delivery mechanism for high speed data and video services.

Network operators are redefining and realigning themselves to be the one-stop shop for all things digital for residential and enterprise subscribers. Triple-play services are not merely a means of increasing top-line revenue, but a means of self-preservation by controlling all services to the home. Digital media and its convergence and interaction are the primary drivers, not just voice, data, and video each as standalone services. Services such as a caller ID screen popping up on the subscriber’s TV set, automatic pausing of live television when a phone call is answered, Internet video services, and presence management services are a-la-carte features of the main triple-play applications that could easily add incremental revenue from every subscriber. Because of the proliferation of these integrated services, investments are under way around the world in new access technologies and platforms that enable service providers to deliver more bandwidth, more control, and more applications.
In North America, cable operators have been adding VoIP or digital voice services and have won over 3 million RBOC customers in the process. With Cox and Comcast alone controlling nearly 30 million video subscribers, the revenue potential of adding voice services to even a fraction of these subscribers is too attractive to pass up. In response, both AT&T and Verizon are building out advanced broadband networks to support the triple play. AT&T’s Project Lightspeed, which is in the early stages of deployment, uses both copper based DSL and fiber based PON, and Verizon’s FiOS network, based on PON, are delivering triple-play services incorporating advanced IPTV.

In Europe, competitive providers, such as FastWeb in Italy, have achieved notable success in securing triple-play subscribers. Incumbent providers around the region are scrambling to respond with their own offerings. At the end of 2006, FastWeb (which was acquired by Swisscom) had approximately 300,000 triple-play subscribers, representing 35% of their subscriber base. Using a combination of PON, fiber Ethernet, and DSL, FastWeb has been able to secure traditional Telecom Italia customers throughout major cities with high population densities. In France, Iliad’s Free division reported that 1.3 million subscribers have signed up for its triple-play service and that nearly 650,000 subscribers use their high speed data and VoIP bundle. Iliad’s success has forced incumbent France Telecom to expedite the rollout of its own triple play service, and already the carrier has secured nearly 850,000 subscribers for its Orange TV IPTV service.

In Hong Kong, PCCW has rolled out the largest IPTV network of any telco in the world. Spurred by competition from Wharf Cable, the incumbent video provider, PCCW leveraged its substantial voice and high speed data subscriber base and quickly built out its NOW Broadband TV service, which gives subscribers a la carte channel options. At the end of August 2007, PCCW reported over 850,000 NOW Broadband TV subscribers. But more interestingly, adding video services has stopped the hemorrhaging of subscriber lines from a peak of 38,000 lost lines per month in 2003 to a net addition of nearly 40,000 lines in early 2007. In Japan, NTT East and West are using PON and DSL technologies to deliver triple-play services.
But the launch of triple-play services requires that traditionally “best-effort” data networks evolve into highly scalable and resilient infrastructures to ensure a high quality experience (QoE). The introduction of broadcast and video on demand services creates a situation in which bandwidth and CoS requirements to each subscriber can swing wildly, while QoS must remain constant. For example, ensuring that a high definition video stream doesn’t eat up bandwidth required for an ongoing voice call is critical.

The inherent difficulty of deploying combined voice, data, and video services to subscribers, the relative novelty of network platforms, and the fact that standards for these platforms and technologies are still being hammered out, all make ensuring interoperability among platforms a primary technical challenge for triple-play service providers.

On the voice side, ensuring interoperability among softswitches, media gateways, application servers, and signaling gateways, as well as provisioning voice features, such as Call Waiting, Call Forwarding, and Caller ID, along with subscriber CPE, remains a challenge, even though protocols such as SIP and H.248 have evolved and matured. On the video side, still a relatively new technology for some service providers, ensuring interoperability among multiple headend devices and/or the headend devices of their content partners plus IGMP-capable routers, DSLAMs, VOD, middleware, and conditional access servers and software poses a challenge just within the video delivery network itself.

The most formidable challenges reside in the subscriber’s home, including ensuring that in-home wiring has the capability to support multiple services, and ensuring that the IP set top box (STB) works with a residential gateway or broadband router and can ping the middleware servers for OS and EPG upgrades. Subscribers fully expect that if there are problems with the in-home wiring, the operator will send a technician to resolve that problem at little to no cost to them. But if the operator has the means to diagnose and troubleshoot the issue without rolling a truck, they will see a tremendous decline in the total cost of ownership of their fledgling triple-play networks.
Finally, securing all these services to the satisfaction of the video content owners and the subscribers themselves, presents an additional layer of complexity to the deployment of triple-play networks. With all these considerations and potential roadblocks before them, network operators are increasingly working with companies experienced in full lifecycle testing to ensure their new triple-play services work to consumers’ expectations from the start. From lab and interoperability testing to ongoing QoE testing and measurement to diagnose and troubleshoot technical issues before subscribers are aware of them, network operators need to have a full range of diagnostic tools at their disposal to ensure service uptime, lower operational costs, and eliminate any reason for a subscriber to take their business elsewhere.

In addition, network operators need to have an accurate picture of their entire triple-play service network, from headend to home. Ongoing diagnostics can’t stop at the DSLAM or OLT any longer. They must include a detailed snapshot of the subscriber’s in-home wiring, residential gateway, and IP set top box to get an accurate picture of any packet loss, jitter, or latency issues affecting QoE.

Triple play is being considered or delivered by MSOs, telcos, incumbents, competitive providers, satellite operators, and media/content players. It is at once a threat, a competitive advantage, a revenue generator, a customer retention vehicle, an exciting new consumer service, and the source of new technology options and a bevy of technical and operational challenges (all of which have solutions in fact or on the drawing board). This book is designed to examine these issues and technologies in more detail.

Michael Howard
Infonetics
In today's communications service delivery market, there's more pressure than ever before to maximize revenue. Adding service offerings, decreasing the cost of doing business, and fine-tuning network efficiency are goals that top every provider's business plan. Triple play—the bundled delivery of voice, video, and data services—is the grand panacea. Properly planned and executed, triple play can indeed yield satisfied customers and a robust bottom line. Yet, a comprehensive view of the landscape is critical before embarking on triple-play implementation.

**The Competition**

Alternative service providers now are offering voice and data to their subscribers in addition to broadcast video. And, they are rolling out new video-on-demand services. Wireless carriers are capturing traditional voice service business while offering new wireless broadband data services—including video—to their own subscribers. With the convergence of the multitude of service offerings in the marketplace, it becomes ever more critical for incumbent providers to ensure that existing services meet the growing expectations of customers, both new and loyal.

**The Strategy**

The market wants bandwidth, and to survive, providers must employ innovation that delivers—economically and flawlessly. Network migration is key and fiber to the X (FTTx), where X can represent N for node, C for curb, H for Home, or P for premises, is the answer. FTTx networks are an essential part of the competitive strategy. Advances in the technologies including passive optical networking (PON), digital subscriber line (DSL), and video compression techniques, are bringing real solutions for customer base retention and growth within reach.
There’s no denying that the demands of deploying these new networks are great. And many more challenges await to compound the circumstance. The triple-play services riding on the network must perform perfectly—all of the time. They demand service that is always available and always reliable. Customers simply will not wait for providers to get it right. There are too many competitors waiting to deliver the same service packages at lower rates with tall promises of better quality.

To make delivery of triple-play services over an FTTx infrastructure a successful reality, providers must incorporate an application-aware test and service assurance strategy along with the traditional line-up of test processes and procedures. The advent and proliferation of triple-play’s signature IP-based applications demands new test and measurement equipment, as well as service assurance systems with highly integrated capabilities and multi-port access.

Further, these next generation solutions must enable field technicians and operational support personnel to test the quality of service (QoS) of each IP-based application as well as allowing testing of physical layer performance. Both the network’s physical transport mechanisms and packet flow mechanisms have to meet stringent parameters so that the network can deliver proper application-specific QoS levels.
The Building Blocks

The rapid adoption and evolutionary nature of triple-play services over FTTx is resulting in a myriad of scenarios for delivering the voice, video, and data bundle to the customer (see Figure 1.1). Triple-play services can be delivered via multiple core and access technologies. With respect to the network core, the majority of triple-play services will be transported via Ethernet carried over new or existing Fiber based network using, WDM and SONET/SDH infrastructures. In the Access network, services will be provided to the end user via a derivative of an FTTx network architecture.

Figure 1.1 The Building Blocks of the triple-play services delivery infrastructure
FTTx encompasses the entire set of new network architectures that are enabling providers to bring broadband services to the customer. In FTTx, optical fiber may or may not extend all the way to the premises as it does in FTTP (P for premises) or FTTH (H for home) architectures. An xDSL technology, such as ADSL2+ or VDSL, may be used in the last mile from the fiber node (FTTN) or curb (FTTC) to the customer premises equipment (CPE). Typically the termination at the home is a residential gateway (RG) incorporating the terminating access technology unit such as a VDSL modem or FTTH ONT.

In all cases, a critical, often complex, combination of in-home wiring or connection system and distribution components exists. For example, Category 5 wire may run from the RG to a home router, PC, or set top box (STB). Wireless technology may be used as an in-house network, or Home PNA (twisted pair LAN technology) or Home Plug (power Line LAN technology) can be present. Even coaxial cable (instead of twisted copper cable) may be used to carry a Home Phone Network Alliance (PNA) technology. MoCA (Multi-media over Coax Alliance) technology which is specifically designed to use the coax cabling in a home for in-house data and video connections is yet another alternative. In fact, in most FTTx networks, varying combinations of fiber, copper, coax, and/or wireless will carry the signals bearing triple-play applications to integrated home networking devices, TV sets, and adapters for VoIP and voice applications.

The complexity of the in-home wiring alone underscores the unprecedented importance of testing all physical media utilized in triple-play service delivery over an FTTx infrastructure. Historically, providers could simply test the copper, focusing only on narrowband voice. Not so in FTTx networks. Since xDSL, with its use of a much wider frequency spectrum, places more stringent demands on the copper plant a new, copper characterization test strategy is now necessary to ensure that the copper plant can deliver proper service performance.
**The Ground Rules**

*Triple Play.* Once a term used only in telecom trade communications, it is now a household phrase. Market leading providers have pushed demand creation into overdrive, and for this next generation service concept, the future is now. What these pioneering providers have learned rapidly is that this future brings with it a new set of rules of engagement. The essence is do more with less, and quickly, because the competition is bearing down.

In the rush to deploy, however, corners must not be cut. Take every precaution: test the components; test the infrastructure. Experience the service—before the customer does. Multi-layer testing including network performance and the application QoS must be carried out during the entire triple-play lifecycle.

An effective multi-layer test strategy includes multiple testing methodologies and a range of equipment to be used at key points throughout the network. Field technicians must be able to efficiently turn up new service, perform end-to-end service verification, and support an efficient trouble resolution processes. When the network is deployed and services are in use, service assurance testing must be on-going with continuous monitoring of specified performance indicators.
The Basics—Testing Voice, Video, and Data

Triple-play service turn-up is rolled out in four stages (see Figure 1.2). Each lifecycle stage brings with it unique, service-specific challenges.

**IP Voice**

To turn up and provision an enterprise voice over IP (VoIP) application, providers first must determine the enterprise network’s suitability for VoIP’s delay-sensitive traffic. Routers must implement class-of-service (CoS) mechanisms and networks must be revised or re-engineered to accommodate additional voice traffic. Load planning may impact necessary network equipment and capabilities, especially wide area network (WAN) interface bandwidth requirements. Additionally, providers may need to negotiate new service level agreements (SLAs) to ensure reliable performance at key hand-off points in the network.

In the installation phase, integrated access devices (IADs), voice gateways, routers, and phones are installed and tested. Physical layer testing of the WAN links is performed. Necessary connectivity and network elements then can be provisioned, and new cabling may be installed to the new end points (IP phones or terminal adapters).
Prior to turn-up, technicians focus on VoIP QoS testing and placement of IP phones. Provisioning errors and voice quality issues are identified and resolved before service provisioning is completed. Test records gathered during this phase of the process should be captured and recorded for future use. If problems with service do occur later, these records can expedite the trouble resolution process.

During turn-up, technicians must verify connectivity to signaling gateways, service provisioning, and call quality. Call quality must be verified by placing both on-network and off-network (to PSTN) test calls. Critical test call parameters include packet delay, packet loss, and jitter. However, the mean opinion score (MOS), a measurement that includes these parameters, is the most critical single quality metric. MOS is widely used in SLAs to measure overall VoIP service quality.

During service delivery, a variety of issues can cause poor VoIP service quality. A comprehensive service assurance plan is critical to ensure continued customer satisfaction. Issues that can degrade VoIP service include customer premises equipment (CPE)-related trouble caused by handsets, processor/DSP performance, and microphone/earpiece acoustics. Network echo canceller performance also can affect service. Network hand-offs between the packet network and the TDM network, typically managed by a voice gateway switch, are critical test points in all networks.

**IP Video**

For broadcast video service, what happens in the headend (HE) propagates through to the home. Quality in/quality out truly epitomizes IP video service delivery. This is why content source inputs must be monitored closely and analyzed thoroughly—prior to transcoding, grooming, and entry into the distribution network. HE quality control strategies must be established.
Bandwidth (BW) requirements vary based on several factors. These include the number of simultaneous programs to be delivered to the subscriber, the compression scheme used such as MPEG-4 AVC, and the type of programming such as standard definition or high definition TV (HDTV). These configuration items directly impact the overall network design. Video CoS mechanisms, such as VLAN tag segregation, load planning, and resulting bandwidth, must be determined. Digital subscriber line access multiplexer (DSLAM) equipment must be upgraded to support Internet Group Management Protocol (IGMP) snooping and IP multicast functionality. Last mile access topology must be established to support required IP video bandwidths. (Typically, the delivery of three simultaneous broadcast video channels is required.)

After these items have been addressed, necessary network elements and connectivity can be provisioned. Network elements that need to be installed and tested include DSLAM ports, optical network terminals (ONTs), optical network units (ONUs), or DSL modems, residential gateways, in-home wiring distribution technology components, and STB. Re-qualification of access links also may be completed.
When turning up and provisioning IP video service, provisioning errors and video quality issues are identified. Case-by-base troubleshooting must be performed before service installation can be completed. Ideally, test records in this phase of the process should be captured to provide a baseline reference for future service assurance activities. Such recorded data will improve mean-time-to-repair (MTTR) and result in a positive experience for the customer, ultimately reducing churn.

To sustain a quality installation process, field technicians must have the skills and instruments to verify performance in three key areas: adequate xDSL performance, video service provisioning, and video QoS metrics. When established parameters are verified, service can be turned-up. The test equipment must be able to emulate the customer’s STB, obtain video program flows, and validate the video QoS metric thresholds established by the service provider for each parameter.

The technician’s test tools must analyze all critical parameters that affect video flows. For example, if the program clock reference (PCR) jitter is high, the decoder cannot properly decode the video payload. IP packet loss and jitter are critical. Trouble with IGMP latency impacts the time required to change broadcast video channels and, therefore, is an important “customer experience” component. The number of lost packets in the video transport stream, as measured by the continuity error indicator, are important, too. Pass/fail thresholds should be set in testing devices for each of these parameters, promoting consistency in operational practices and helping improve service assurance processes.

To successfully troubleshoot IP video problems such as video pixelization or frame freezes, field test equipment must enable technicians to isolate specific video flow fault conditions. For example, lost packets, or excessive packet jitter, may cause both
conditions. If no errors are detected at the physical layer of the access link under test, it is then likely that packet loss is taking place upstream of the DSLAM. PCR jitter also could be the problem source. Excessive PCR jitter may be specific to a channel, indicating a problem at the headend, with local advertisement insertions, or with source material transcoding operations. Testing two video flows simultaneously is important in determining whether the excessive PCR jitter is seen on both flows. If it is present in both, then the source is likely to be network-induced jitter.

Packet loss impacts vary. For instance, loss of a single packet carrying B-frame data in an MPEG-2 video signal typically affects one or two frames of the video flow. Loss of a single packet carrying I-frame data propagates across all frames until the next I-frame. With a group of picture (GOP) size setting of 14 or 15 (a typical size), the error would affect nearly half a second of video with a frame rate of 30. Another key metric is the video transport stream packet “error indicator” count. If the encoder implements the use of this option, it will indicate a problem with the source program material seen by the video encoder in the headend. If a count is present, packets have been corrupted and errors typically will be discernable on the television. This is significant in that it indicates a content problem. Additionally, this metric can be analyzed even if the payload is encrypted.

Key performance indicators such as these should be monitored on a continuous basis to ensure the network is delivering desired service quality. Packet flow monitoring at key test points is required to keep IP video applications running at QoS levels that will deliver the quality of experience (QoE) customers have come to expect from competitive video service providers. The QoE must be better than, or at the very least, equivalent to their previous experience.
**IP Data**

To provide subscribers with IP Internet data service, ISP accounts must be established for each customer, and traffic planning must be modified to accommodate additional data flow. Broadband remote access servers will be impacted and control/routing and bandwidth planning must be completed. In addition to establishing connectivity to ISPs and provisioning necessary network elements for increased data flow and CoS treatment, DSLAM ports may be reconfigured for dual latency path support in this mixed application environment.

To complete the installation process, field technicians must verify DSL physical layer performance (where DSL access is used), in-home connectivity from PC to wireless access module (WAM) access point (typically a residential gateway), ISP connectivity, and data service throughput. This is carried out via a test tool with Web browser and FTP throughput test capabilities. Using selectable test file sizes and both up-load and down-load testing, FTP throughput tests establish performance of the link that more closely models actual use cases than a simple download test. An HTTP test using a Web browser is needed to ensure that end users' ISP access/connectivity is working properly.

As IP Internet data service is turned up, provisioning accuracy must be validated. Again, records of any troubleshooting or issues should be captured and kept for future reference.
Triple Play Requires Triple Testing

Turning up triple-play service and assuring its delivery is a multifaceted and on-going process whose tasks must be addressed comprehensively and with exacting attention to detail. Testing must be performed at multiple layers, including physical layer, data link layer, network layer, and application layer. Simple physical layer testing in the access network does not reveal all of the potential QoS issues that can impact packet-based triple-play applications.

Figure 1.3  Triple-play service delivery protocol stack example
Clearly, service quality is affected by end-to-end packet flow. Service providers deploying FTTx networks to deliver IP-based services need a complete *application-aware* test and service assurance strategy—one which outlines the way to properly install, provision, and maintain reliable networks that carry consistently high quality triple-play services. Sustained business performance depends on it.

These new testing requirements may seem overwhelming and costly, with endless details that field technicians and their equipment must address. The essential economy is a strategy that simplifies, automates, and expedites processes and procedures in which errors can lead to increased operational expenses (OPEX). Advanced centralized service assurance solutions, coupled with next generation field tools, are needed to bring triple-play services on line. These solutions are helping to ensure that the significant investments service providers are making will control OPEX and yield churn-resistant customers.

The following chapters will guide the reader through many of the challenges associated with successfully deploying triple-play networks. This guide will assist in identifying key performance indicators (KPIs) to consider during network design. It offers advice for expediting new service turn-up; and, it presents comprehensive service assurance planning information on sectionalizing anomalies and isolating faults for quick trouble resolution. Use this guide as the map to a destination where profitable networks deliver outstanding QoS to meet—or exceed—customers’ QoE expectations.
Triple-Play Service Delivery over Ethernet Networks

Chapter 2
Ethernet Networks: Carrier-grade requirements

Ethernet is widely accepted as a cost-effective, resilient, and scalable switching and transport technology. It facilitates the availability of affordable bandwidth on demand and highly secure private network communication. Yet despite the lucrative and strategic advantages it brings, many providers are reluctant to embark on full implementation to bring carrier-grade Ethernet offerings, such as triple-play services, to market. This is due, in part, to the inherently greater complexity of maintaining carrier-class quality-of-the services (e.g., 99.999% reliability/less than 6 minutes of downtime per year), at the node and network levels, as these offerings bring about a number of technical and operational challenges. These challenges include:

- Ethernet aggregation networks support services that have different characteristics and place different burdens on the network due to CoS treatments
- Customer’s quality of experience (QoE) and associated QoS metrics are drastically different than those depicted for the traditional voice services
- Deployment of these services takes place over a multitude of new and existing networks, topologies and transport technologies
- Providers need to account for associated changes in responsibilities for support personnel implementing these services
In a triple-play offering, Ethernet, as an aggregation network over which converged voice, video, and data travel, must deliver carrier-grade performance. This is particularly important as it relates to transport network characteristics, such as packet loss and jitter, while maintaining the following attributes:

- Rapid scalability, resiliency, and redundancy
- Five-nines of reliability (99.999%)
- Ability to meet key network SLAs (Connectivity, Throughput, Packet Loss, Packet Jitter, and Delay)
- Cost-effective deployment of high bandwidth services
- Convergence of business, residential, and wireless services

Even with these significant challenges, demand for Ethernet aggregation networks is increasing. This, coupled with the providers’ rapid adoption and evolution of triple-play services and service delivery mechanisms, is mitigating the implementation of new provisioning and service assurance strategies to address increasingly complex networks and customer expectations. Providers understand they can ill-afford to have customer satisfaction issues resulting from poor video service (with quality problems such as pixelization), dropped VoIP packets (affecting voice quality), slow Internet connections, or any other number of problems that may be caused by the Ethernet aggregation networks. Thus, the traditional quality assurance methodologies, test tools, and processes are insufficient for maintaining and understanding true end-to-end QoS for triple-play services. Providers are seeking solutions that will allow them to efficiently install, troubleshoot, maintain, and analyze services, without sacrificing the quality of experience.
Understanding the Inherent Benefits of Ethernet Technology

The importance of Ethernet networks in triple-play offerings stems from its evolution from the local area network (LAN) environment, to the metropolitan area network (MAN), and wide area network (WAN) environments. Ethernet is a technology of choice for many providers dealing with building convergent networks required to carry next generation services. With the evolution of voice, video, and data services, the traditional public switched telephone network (PSTN), along with traditional ATM deployments for virtual private network (VPN)/data offerings have proven to be inadequate in transporting these services. Therefore, the need to deploy cost-effective networks with scalable characteristics that can both provide a path for future growth and utilize economies of scale is pivotal.

As a result, the industry has come together to develop standards and design new network elements, services, and service attributes that will enable Ethernet to become a carrier workhorse for building ubiquitous converged networks to carry triple-play services.
Ethernet, as a baseline technology, allows providers to build next generation networks by taking advantage of the following important benefits:

- It is a standardized, IP-friendly technology, which brings economies of scale to network deployments as it leverage hundreds of millions of Ethernet ports deployed worldwide.
- It allows providers and carriers to scale their LAN backbones and build MAN and WAN solutions with the simplicity of Ethernet technology, thereby supporting all these topologies in a seamless network.
- It provides scalable bandwidth from 10 Mbps to 10 Gbps, allowing for ample growth in triple-play services.
- Ethernet is simplifying the Metro area, as shown in Figure 2.1.

![Figure 2.1 Comparison of traditional vs. Ethernet network structures](image-url)
Ultimately, Ethernet as an aggregation network allows providers and carriers to meet the growing demands of end-customers by offering broader service coverage and choices. Further, it ensures capital and operating expense reductions in the networks—all of this by meeting the bandwidth gap that existed in the Metro network with traditional leased lines. As Figure 2.2 outlines, Ethernet as a transport technology allows providers to maintain the competitive edge, while offering services without sacrificing the bandwidth requirements. Furthermore, Ethernet technology being standardized and IP-friendly allows for compatibility between the Enterprise, Metro, and Core network topologies, without the need for expensive transport conversion deployments.

Figure 2.2  Ethernet in the Metro network addresses the bandwidth gap that existed with traditional leased line services.
Class of Service (CoS) Concept

With the evolution of Ethernet as a baseline transport for converged networks and the need to provide sufficient bandwidth for triple-play services, QoS and resilience become the focus of well-built carrier-grade Ethernet networks. These networks must be scalable, cost-effective, and have the ability to prioritize and process traffic based on its importance to the end user. To accomplish this effectively, the industry has come together to design class of service (CoS) mappings for various services. These often consist of tagging a certain type of traffic and assigning a priority to it which is used by network elements in making routing decisions.

It is important to note that CoS handing and routing decisions have immense effects on the end-user experience, and thus considerations for tagging, prioritization and routing in the Ethernet aggregation network must be well understood and analyzed.

In order to provide successful CoS mechanisms in the network several key metrics need to be verified per service: packet loss, packet jitter, and delay. Each service may have different sensitivity to these metrics. For example, the effect of packet loss on voice (VoIP) service may not be as detrimental as the same metric for video services. Conversely, deteriorating delay measurement may not be as service-impacting for video as for voice service.

Figure 2.3  Example of CoS tagging structure for triple-play service delivery
CoS Prioritization Schemes
A variety of mechanism and tunneling technologies exist today that allow providers to choose the network implementations that will best meet their needs to reliably carry triple-play services while maintaining CoS for those services. These can be grouped into two emerging trends:

– Native Ethernet protocol extensions that are considerations of the IEEE standards body–VLAN technique (often referred to as 802.1Q/p), Q-in-Q technique (often referred to as VLAN Stacking or 802.1ad), and MAC-in-MAC technique

– Encapsulations by multi protocol label switch (MPLS) networks, which also include the Layer 2 (VPLS) and Layer 3 versions

These schemes insert additional tags/fields in the customer Ethernet frames at the ingress nodes (crossing the edge node into the Metro domain) and strip them off at the egress nodes (crossing the edge node out of the Metro domain) before the frames are handed over to the appropriate customer traffic segments. Issues such as backward compatibility, comparative performance, and complexity are key elements that influence the choice of one scheme over the others.

Some networks deploy a combination of these techniques. For example, portions of the core network may utilize MPLS-based CoS methodologies (where MPLS sub-50 ms recovery times are leveraged to improve the recovery times if a link failure occurs in this part of the network), which are then converted into a VPLS mesh of elements (with the focus on improving routing and handling MAC table explosions). Meanwhile in the access network, VLANs are utilized for their simplicity and ability to prioritize traffic in proper granularity. This type of network topology provides channel-changing response times that are comparable to traditional networks.
Chapter 2: Triple-Play Service Delivery over Ethernet Networks

Customer A VLANs
VLAN 20, 30

Customer B VLANs
VLAN 30, 40

Overlapping VLAN IDs between customers

VLAN Tagging/Stacking (Q-in-Q)
Packet can enter network as
- Untagged
- With VLAN tag – Single or stacked (Q-in-Q)
Provider will add a tag to the packet at the Edge device.
Rate limiting policies take place at the Provider Edge router

Figure 2.4  VLAN Tagging and Stacking

VLAN Tagging
VLAN tag (per IEEE 802.1Q) adds 4 bytes to the Ethernet frame, including the Tag ID and a Priority (802.1p). The maximum size of the modified frame is 1522 bytes.

VLAN Stacking
Double VLAN tag (IEEE 802.1ad) adds 8 bytes to the Ethernet frame, including the Service Provider tag and a Customer tag. This allows for ‘stacking’ of several VLAN tags for tunneling applications. Maximum frame size is 1526 bytes.
VPLS Frame
Virtual Private LAN Service (VPLS) offers Layer 2 MPLS-based VPN service with multipoint connectivity. VPLS Frame consists of an Ethernet frame with MPLS tags inserted for proper information tunneling across the network. Customer frame information is stored in the Data portion of the VPLS frame. VPLS technology is an IETF standard, currently in Internet Draft status.

MPLS Frame
Multi-Protocol Label Switching (MPLS) utilizes label switching/tunneling in conjunction with network layer routing in applications such as MPLS-based VPNs. MPLS can be implemented over Layer 2 technologies, which may include Frame Relay, ATM, and Ethernet. It is defined in numerous IETF standards.

Figure 2.5 VPLS and MPLS encapsulations
Triple-play Services within Ethernet Aggregation Networks

Each component of the triple-play—voice, video, and data—poses specific requirements on the Ethernet aggregation network. In each case, user quality of experience can be negatively affected by poor transport network metrics. Understanding the service requirements is vital to the successful service deployment.

The following diagram illustrates how objective metrics may be mapped to subjective QoE issues for services by organizing the metrics into four quality categories.

- Content Quality: the actual video payload
- Video Stream Quality: the stream packet flows
- Transport Quality: the Ethernet/IP packet flows
- Transactional Quality: the interaction between the user and the service

Figure 2.6 QoS and QoE mapping for services
The stability, scalability, and functionality of the Ethernet network may have tremendous effect on the actual transport quality of video service. In addition, it is important to understand that the quality of the transport carrying these services contributes to the overall QoE for the subscriber. Therefore, the ability to correlate errors that take place in the Ethernet (aggregation) network to the access network will enable providers to more quickly and efficiently troubleshoot problems, provide solutions, and thereby keep customers satisfied.

In this aggregation network, packet loss and packet jitter are key metrics that have impact on the pixelization, tiling, frame freeze, and the overall QoE. Moreover, video stream characteristics such as the program clock reference (PCR) jitter measurement, depending on where measured, can be affected by overall network packet jitter, which again may surface in the Ethernet aggregation portion of the network.
In order to depict the effect of packet loss on video service, it is important to note that with protocol encapsulations, a typical IP packet carries seven video packets. A single lost packet event will create some kind of visible impairment. But, for example, the subscriber’s subjective response to minor pixelization events, caused by packet loss, varies with the pattern, distribution, or repeatability of these events.

Therefore, in an Ethernet network it may be important to provide transport quality metrics which can be correlated to the actual experience of the users. RFC3357, for example, is an Internet Engineering Task Force document that defines two metrics: loss distance and loss period. These metrics capture loss patterns experienced by packet streams in the aggregation network. As IP experiences certain bursty effects that can impact the user’s experience, the loss patterns or loss distribution may become a key parameter that determines the performance of video for the end customer. RFC3357 states that for the same packet loss rate, two different loss distributions could produce widely different perceptions of performance.

Mapping this kind of analysis to perceived QoE may also be accomplished by a video mean opinion score (MOS) algorithm, a concept which is not standardized today; however, several standards bodies, as well as some RFCs, are presenting visions of how this may be accomplished in the future. If the content is not encoded in the aggregation network, and therefore can be analyzed, proprietary algorithms do exist that allow users to perform analysis of the content and obtain the quality metrics. Some of the better known metrics are the TVQ algorithm (by Telchemy, also know as VMOS), and the VQS algorithm.
For video quality measurements, additional important metrics may include the measurement of the PCR jitter in the Ethernet aggregation network, as these may be caused by the content quality problems (at the super head end) or overall network packet jitter. The source of the problem can be differentiated by evaluating more than one channel/program at a time. If excessive PCR jitter is present at more than one channel, the overall network jitter is most likely at fault. If excessive PCR jitter is present on only one channel, then a video source problem (at a super headend) is typically the cause.

**Voice over IP**

(Note: VoIP deployment and testing is described in detail in Chapter 8.)

Successful voice over IP (VoIP) service delivery over a convergent network can be facilitated by an understanding of the service characteristics and quality metrics that are affected by events in the Ethernet aggregation network. Furthermore, it is important to state that Ethernet and IP layers are critical to the deployment and troubleshooting of VoIP service, and have significant impact on the overall QoE parameters, including dropped calls and call quality. Items that affect the overall transport quality of VoIP service are IP packet delay, loss, jitter, and out of sequence (OoS) packets. While these are described in more detail in Chapter 8, their impact is briefly summarized below.

- Packet delays can have varying effects on voice quality, so it is important to measure the delay at the time of service installation to provide a benchmark for verification against potential problems.

- Packet loss can occur for a variety of reasons inside a network. Periodic losses in excess of 5 to 10 percent of all voice packets transmitted will degrade voice quality significantly.
– Packet jitter will make speech choppy and difficult to understand. For high-quality voice, the average inter-arrival packet time at the receiver should be nearly equal to the inter-packet gaps at the transmitter and the standard deviation should be low.

– Out of sequence packets have a similar effect to that of lost packets because voice codecs often discard them. It is also important to note that lost packets and OoS packets are always measured on the local link and can help in the segmentation of the overall problem.

**Data**
(Note: Data service deployment and testing is described in detail in Chapter 9.)

Data service (depending on the architecture) may arrive through the Ethernet network from a different point, such as Internet network, Internet point of presence (POP), etc. As a result, in the Ethernet aggregation network, it is important to verify that data service arrives without errors; is prioritized correctly (has the correct level of CoS mapping provisioned); and, does not negatively impact higher priority traffic such as video and voice. The focus of data testing in the Ethernet network is at the IP layer, where it is important to verify that IP traffic is free of errors, that correct data bandwidth is received, and that data traffic is correctly prioritized with respect to other triple-play services.

RFC2544 is often used for data testing to verify the throughput, loss, delay, and burstability of this service. Test methodologies may consist of verifying physical layer characteristics, and then executing the RFC2544 testing for Layer 2 or 3 data service verification.
Tools for Turning Up and Maintaining Carrier-Grade Networks

Combining a range of instruments, systems, and software into an integrated test suite is the recommended, proactive approach for turning up and maintaining carrier-grade networks. An integrated test suite offers problem-solving capabilities that can generate consistent and repeatable processes for turning up new services in the field. Not only does it facilitate problem detection and expedite fault isolation for the maintenance of service integrity, but an integrated test suite also greatly reduces operations costs. The requirements outlined below are key in ensuring that the Ethernet aggregation network will facilitate successful, economical delivery of triple-play services.

Portable solutions
QoS verification begins during service installation and commissioning. Providers must certify, from the moment service is turned up, that the network meets the parameters agreed upon in the service level agreement (SLA). Test sets that can run the full range of tests save both time and money. Not only can the provider dispatch a single technician to perform the task, but the technician will not require in-depth training to operate the test set. Additionally, when a single test set can perform tests from the physical layer through the data and network layers, the solution brings even more cost savings because technicians to not need to carry multiple instruments to perform their jobs.
Verifying the services in this way assures the customer that the link meets the SLA requirements and, of equal, importance, creates a baseline of performance against which the network can be subsequently measured. Accurate testing and the correct recording of results can thereby supply both a reference for a potential maintenance issues and a method of reducing dispatches to rectify incorrect installations.

Ethernet services may be installed and verified by performing an automated RFC2544 test, or by performing a sequence of steps as described below, which may be described as Methods and Procedures for service installation.

**Connectivity**

Prior to testing throughput in a network, the provisioned service path must be verified. This is accomplished via a specific VLAN circuit setup, or by performing a ping test in a routed network. If connectivity problems with the far end exist, it may be necessary to verify the route between the source and destination host by running a Traceroute application. Traceroute is used as a tool to facilitate an understanding of where problems in the network may occur, and it is useful for gaining a sense of a particular route behavior in the network itself.

**Throughput**

In order for a new service hand-off to take place, proof must be provided that the circuit can handle the service it will carry. During service installation it is therefore required to measure throughput and verify that the network can deliver the bandwidth allocated to the end customer. On a live network, congestion may occur and prevent the subscribed service from running properly. To prove the service level agreement, line-rate traffic must be generated to stress the circuit. Moreover, an additional purpose of this test is to verify that no data is lost or corrupted as it travels the network.
**Frame/Packet Loss**
A frame loss test determines the utilization rate at which no frame loss is present on the customer’s circuit. The test is performed by generating traffic at the assigned service line-rate in order to verify that this bandwidth allocation does not cause problems in the network path. The test takes into consideration network buffering issues and verifies that traffic is not lost due to potential congestion issues in the service path.

**Packet Jitter**
To determine the average and peak packet jitter, testing may be performed in an end-to-end or loopback configuration. The test is performed by generating traffic at the assigned service line-rate in order to assess whether any packet jitter is caused by jitter buffers in the service path. Excessive jitter will likely cause service deterioration and affect the QoE.

**Round-trip Delay (latency)**
To determine the average round-trip delay of the network, latency testing is performed using a circuit terminated in a loop-back. During this test, traffic is generated at the service line-rate, and the measurements are performed continuously on the received traffic. Determining circuit latency at the time of installation is an important step because the data captured may be needed as a benchmark for subsequent network performance.
**Field Test Applications**

Test applications for portable solutions can be divided into two main categories: Installation and Troubleshooting. The characteristics and capabilities outlined below are critical.

**Installation**
- Ability to verify physical layer (fiber characterization, signal code violations, if applicable)
- Ability to connect to a proper port and establish connectivity to that port; support IP connectivity (DHCP, Static addressing, ARP, etc.)
- Ability to receive valid video stream and verify transport quality (packet loss, packet jitter, delay)
- Ability to verify transport layer (packet loss/jitter) for VoIP and data applications
- Ability to transmit and emulate multiple service streams to another test set with various priority and characteristics:
  - Video stream with highest CoS priority (EF value)
  - VoIP stream with lesser CoS priority (AF value), with proper frame size
  - Data stream with best effort CoS priority (BE value)
- Measure throughput, packet loss, packet jitter, and delay per each stream (see Figure 2.8).
- Look for errors such as FCS, IP checksum, PCR jitter, and continuity
- Provide QoS measurement for each stream (e.g., MDI for video)
**Troubleshooting**

- Ability to monitor traffic and provide indication of the transport errors

- Ability to analyze several streams simultaneously and verify the impact of packet loss and jitter per stream, as well as across streams. (This can point out whether the source of the problem is the aggregation network or another source.)

- Ability to provide statistics such as throughput, delay, loss, and jitter (see Figure 2.8).

---

**Perform End-to-End Connectivity Testing**
- Verify physical layer with fiber characterization tools
- Perform physical layer BERT

**Perform RFC2544 Automated Testing**

**Perform Traffic Emulation and CoS Testing**
- Verify:
  - Throughput
  - Frame Loss
  - Latency
  - Packet Jitter
- Ensure SLAs are met

---

Figure 2.8 The JDSU T-BERD® 8000 provides comprehensive and easy-to-interpret reports on critical service parameters including throughput and delay.
Service Assurance Solution

(Note: Service assurance solution requirements for overall triple-play service delivery are described in detail in Chapter 10. This section identifies key components, requirements, and capabilities that a system should conform to when testing triple-play services in the Ethernet aggregation network.)

Discrete functions of a service assurance solution with respect to Ethernet transport may include all or subsets of the following depending upon application needs.

<table>
<thead>
<tr>
<th>Characteristic</th>
<th>Requirement</th>
</tr>
</thead>
<tbody>
<tr>
<td>Fault Isolation</td>
<td>Rapid and automated fault segmentation and/or isolation</td>
</tr>
<tr>
<td>Troubleshooting</td>
<td>Reactively determining what faults exist and what causes them</td>
</tr>
<tr>
<td>Automated Testing</td>
<td>Pre-scheduled routines to proactively test target areas</td>
</tr>
<tr>
<td>Passive Monitoring</td>
<td>Non-intrusive measurement of actual customer traffic</td>
</tr>
<tr>
<td>Active Testing and Monitoring</td>
<td>Measurements of performance based on synthetic traffic</td>
</tr>
<tr>
<td>Turn-Up</td>
<td>Test tools to assist the accuracy of initial service delivery to end customer</td>
</tr>
<tr>
<td>Reporting</td>
<td>Providing users actionable information in performing their roles</td>
</tr>
<tr>
<td>Data Collection</td>
<td>Gathering key metrics or measurements to aid in analysis</td>
</tr>
<tr>
<td>Data Correlation</td>
<td>Correlating related data to better assess and analyze information</td>
</tr>
<tr>
<td>Performance Management</td>
<td>Data assessment to improve or optimize performance</td>
</tr>
</tbody>
</table>

A centralized test and management strategy is essential for supporting the varying requirements of a rapidly growing customer base and maintaining an Ethernet network capable of delivering triple-play services that meet stringent QoS agreements. Fundamental strategy requirements are QoS monitoring and data consolidation. Ultimately, the system should examine customer traffic; gather, capture, and access data from multiple points in the network; and, isolate faults between the customer premises and the service provider as part of the service assurance process. Using distributed IP test heads—located at aggregation switches or at routers in the network as close to the edge as possible—can provide in-depth testing and data analysis with respect to the requirements stated in the SLA. An IP test head that supports multiple, simultaneous services testing enables installation and commissioning to be carried out in parallel, thus reducing testing time and increasing efficiency. If the process is automated from a
centralized location, the ability for multiple technicians to work simultaneously will produce significant cost savings when new, high density locations are activated. Furthermore, an increasing number of companies operate from multiple physical locations. Therefore, it is essential that service providers have the ability to provide the additional assurance of end-to-end service verification. This can be achieved by installing test heads at two or more sites to prove full end-to-end connectivity and perform detailed fault finding in both directions of the service.

The Total Solution

The challenges service providers face after turn-up arise as a result of the need to adhere to SLAs. Through remote testing, automated data capturing, and active and passive monitoring, service providers can gain a coherent view of network performance, customer traffic, and usage patterns. Complemented by full-featured portable test equipment, the centralized solution can reduce unnecessary technician dispatches by isolating faults when network problems occur. Network technicians can determine quickly whether the problems exist between the core and the edge, between the access point and the edge, or between any two points on the network. Effective portable tools can also allow a single technician to identify and analyze the faults once isolated. Ongoing analysis of the consolidated data provided by the test solution will reveal usage trends, link statistics, and traffic distributions. Such test solutions allow service providers to identify degradation before QoS reaches unacceptable levels and, most importantly, before customers are negatively impacted.
Understanding FTTx networks and associated optical testing

The challenge of providing the increased bandwidth needed to support triple-play service delivery is on-going and can be daunting. Running optical fiber much deeper into the access network, in some cases all the way to the customer premises, is an important part of the strategy of nearly every service provider. The appeal is that a fiber optic infrastructure offers the potential for practically unlimited bandwidth and also facilitates greater control over the operation, administration, and provisioning of the access system.

FTTx Architectures

The general penetration of fiber into the access network is often referred to as Fiber-to-the-X (FTTx). The actual architecture can vary depending on the depth of penetration (see Figure 3.1). In an FTTx architecture, the optical line terminal (OLT) houses the laser transmitters which are dedicated to individual users in a point-to-point (P2P) network or shared in a passive optical network (PON). The optical fiber running to the users includes several components: the feeder cable which terminates at the OLT, the distribution cable which comprises the majority of the access network, and the drop cable which connects users or remote distribution points. In FTTH, the optical network terminal (ONT) receives the signal from the OLT and converts it into usable electronic signals for voice, video, and data at the customer premises. If the last mile is provided by a copper access network, the fiber is terminated at an optical network unit (ONU), often consisting of a digital subscriber line access multiplexer (DSLAM), which services the copper network and is located at a remote central office (RCO).
While providing fiber directly to the home (FTTH) is a very attractive offering for the delivery of high bandwidth services to customers, it is not always the most cost-effective solution for providers. Therefore, for cost-saving alternatives, providers must utilize existing plants to deploy different types of fiber architectures. For new homes and new residential areas where no network is yet available, providers will deploy complete Fiber-to-the-Home (FTTH) technology. These new areas are termed “greenfield” applications.
For “brownfield” or overlaid/overbuilt applications, providers end the fiber before reaching the customer premises in order to leverage the existing network infrastructure. Stopping just short of the customer premises allows providers to bypass the cost of pulling fiber under sidewalks and driveways. Following is a chart of the technology distinctions that have been defined to clarify where the fiber infrastructure switches to copper.

<table>
<thead>
<tr>
<th>Architecture</th>
<th>Abbreviation</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Fiber-to-the-Cabinet</td>
<td>FTTCab</td>
<td>Fiber extends to a street-side cabinet or digital loop carrier (DLC) and uses ADSL2 technology to access customers. Typical distance is up to 12,000 feet (4 km).</td>
</tr>
<tr>
<td>Fiber-to-the-Node/Neighborhood</td>
<td>FTTN</td>
<td>Fiber extends to a large street-side cabinet or optical network unit (ONU) and uses ADSL2 or ADSL2+ technology to access customers. FTTN typically serves about 200 residential or small business customers with a radius of 3,000 to 8,000 feet (1 to 2.5 km).</td>
</tr>
<tr>
<td>Fiber-to-the-Curb</td>
<td>FTTC</td>
<td>Fiber extends to an outdoor cabinet beside homes or office buildings approximately 1,000 to 2,000 feet (300 to 600 m) from the customer premises. FTTC may use VDSL technology to access customers.</td>
</tr>
<tr>
<td>Fiber-to-the-Building/Basement/Business</td>
<td>FTTB</td>
<td>Fiber extends to a building. FTTB is similar to FTTH with the exception that it serves multiple customers. It can also serve multi-dwelling units, where FTTH serves only single-family units.</td>
</tr>
</tbody>
</table>

Table 3.1 FTTx architectures defined

For large residential areas, as is the case for the first installations of FTTH in the United States, the optical distribution network will consist of 1:32 splitters. For smaller residential areas, the splitters may be distributed in order to be closer to other smaller areas using a single FTTH architecture. A 1:4 splitter may be located at one FDH followed by four 1:8 splitters at other FDHs, or there may be a 1:8 splitter followed by eight 1:4 splitters.
An FTTH network can follow a star/tree (standard), ring, or bus topology, with the possible use of active components depending upon the locations of the customers. However, an FTTH network can also be a simple, dedicated P2P network. This is the case in Europe, for example, where there is a high density population and where Ethernet switches or active components are used instead of splitters.

**FTTP Infrastructure and Technology**
As discussed, the two basic configurations for fiber deployment are P2P and PON. In P2P networks, a fiber strand and laser is dedicated to each customer. In PONs, each fiber and laser is shared among users. P2P networks provide the ultimate in bandwidth but require active devices or switches at the customer premises. There are several different types of PONs including broadband passive optical network (B-PON) using mainly ATM, gigabit passive optical network (G-PON), and Ethernet passive optical network (E-PON). The specifications and characteristics of each technology are listed in the following table.

<table>
<thead>
<tr>
<th></th>
<th>B-PON</th>
<th>G-PON</th>
<th>E-PON</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Span</strong></td>
<td>20 km</td>
<td>60 km max, 20 km differential</td>
<td>10 km today, 20 km planned</td>
</tr>
<tr>
<td><strong>Maximum insertion loss</strong></td>
<td>20/25/30 dB</td>
<td>15/20/25 dB</td>
<td>15/20 dB</td>
</tr>
<tr>
<td><strong>Maximum number of branches</strong></td>
<td>32</td>
<td>64 (128 Considered)</td>
<td>32</td>
</tr>
<tr>
<td><strong>Traffic mode</strong></td>
<td>ATM</td>
<td>ATM, Ethernet, TDM</td>
<td>Ethernet</td>
</tr>
<tr>
<td><strong>Architecture</strong></td>
<td>Asymmetric or Symmetric</td>
<td>Asymmetric or Symmetric</td>
<td>Ethernet</td>
</tr>
<tr>
<td><strong>Video overlay</strong></td>
<td>Yes</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td><strong>Applicable standard</strong></td>
<td>ITU-T G.983.x</td>
<td>ITU-T G.984.x</td>
<td>IEEE 802.3ah</td>
</tr>
<tr>
<td><strong>Chipset support</strong></td>
<td>Available</td>
<td>Available</td>
<td>Available</td>
</tr>
<tr>
<td><strong>Upstream burst time</strong></td>
<td>Fixed 56 bytes (ATM) Guard: 25.6 ns Preamble: 35.2 ns (typical) Delimiter: 16.9 ns</td>
<td>Laser turn on/off: 512 ns (max) AGC setting and CDR look: 400 ns</td>
<td></td>
</tr>
</tbody>
</table>

Table 3.2 PON standards
BPON uses time division multiple access (TDMA) over two downstream wavelengths and one upstream wavelength. It typically provides 622 Mbps shared among up to 32 users. EPON uses one wavelength for downstream and one wavelength for upstream traffic and offers up to a symmetric 1.25 Gbps service. BPON has evolved into GPON which utilizes the same wavelength plan but delivers up to 2.5 Gbps of bandwidth shared among up to 64 users.

**Point-to-Multipoint Networks**
As previously discussed, a PON network usually is a point-to-multipoint network (P2MP). This means that although there is one fiber at the OLT, the other end of the distribution network can have up to 32 fibers under the same optical network. This is facilitated in the PON through use of a passive optical component called a splitter.

![Schematic diagram of a splitter](image)

The splitter allows for one port at one end and up to 64 ports at the other end. Therefore, a signal sent from one end possibly can be distributed to 64 customers simultaneously. This application is ideal for video distribution. For data and voice, the use of time division multiple access (TDMA) enables customers to receive and send exactly what they choose, without knowing what other customers are receiving and sending. Additional technologies such as WDM PON are under investigation for point-to-multipoint networks. The most attractive alternate is wavelength division multiple access (WDMA) which dedicates a particular wavelength to each customer.
A splitter’s primary benefit is that it is a passive component, requiring no maintenance and no power activation. Its primary drawback is high rate of insertion loss. Insertion loss is defined as $10\log(1/n)$, where $n$ is the number of ports (2 to 64). Table 3.3 shows the expected insertion loss that corresponds to a specific number of ports.

<table>
<thead>
<tr>
<th>Number of ports</th>
<th>Insertion loss</th>
</tr>
</thead>
<tbody>
<tr>
<td>2</td>
<td>3 dB</td>
</tr>
<tr>
<td>4</td>
<td>6 dB</td>
</tr>
<tr>
<td>8</td>
<td>9 dB</td>
</tr>
<tr>
<td>16</td>
<td>12 dB</td>
</tr>
<tr>
<td>32</td>
<td>15 dB</td>
</tr>
<tr>
<td>64</td>
<td>18 dB</td>
</tr>
</tbody>
</table>

Table 3.3 Expected insertion loss based on number of ports

This insertion loss factor has a high impact on the transmission distances for FTTH, essentially limiting the achievable span length to a distance of approximately 20 km (30 dB maximum network insertion loss). This distance limit also takes into consideration that low-cost components are often used at the OLT and ONT.

**PON Signals and Network Structure**

In a standard PON system (see Table 3.2), also called an optical access network as defined by ITU-T G.983.1, the central office (CO) is interfaced to the public switched telephone network (PSTN) using DS-X/OC-Y signals and is connected to ATM or Ethernet interfaces. Data and voice signals use the 1490 nm wavelength for downstream signals and the 1310 nm wavelength for upstream signals. Video services enter the system from a cable television headend or from a satellite feed. Video signals are distributed via the 1550 nm wavelength (downstream signals only).
In a PON network (see Figure 3.4), the signals are combined into a single fiber using WDM techniques at the CO with an OLT. At the CO, a fiber distribution frame (FDF) integrates a number of OLTs together with splicing trays and connectors, which, in turn, connect the OLTs to the fiber network. COs may contain optical networks. The type of connectors at the OLTs is most often ultra polished connectors with a typical ORL value from 50-55 dB. More modern networks utilize angle polished connectors (APC) with an ORL value of 67-70 dB.
Because customers may be located far apart along the feeder, splice enclosures may be located in aerial or underground environments. In this scenario, one cable may be divided into a number of fibers going to different directions. Some typical feeder cable distances can extend up to 30,000 ft. (10 km).

Figure 3.4  A PON network

Corning aerial splice enclosure (Courtesy of Corning)
From the splice enclosure, signals are transmitted along the complete feeder to the fiber distribution hub (FDH), where the signal distribution occurs through the use of passive optical splitters. Thirty-two branches per splitter and 10 splitters per FDH are typically the maximum split levels implemented. To accommodate this, the FDH houses multiple splitters as well as splicing trays and connectors. No power is required at this location because all of the components are passive components. This hub is also called the primary flexibility point (PFP) or the fiber cross connect (FCC). If different splitters exist along the cable, then the other splitters are located at a fiber distribution panel (FDP).

After the FDH in the network, one fiber in the distribution cable is earmarked per customer. When a customer is designated, the splitter is connected to the fiber of this specific customer. Before a customer is designated, the fibers coming from the distribution cable and from the splitters are left open. They are positioned, unconnected, in a ‘parking lot.’ The term “parking lot” defines a staging or waiting area for connectors. In order to satisfy the constraints of the ORL, APC connectors are used here.
The cable is distributed to the customer locations, usually with a distance spanning no more than 20,000 ft (6 km). Access terminals close to the customer premises enable some fibers (four to 12) to branch out of the distribution cable and be terminated with the use of a splice tray. APC connectors are used here, also. In most cases, the other fibers go through the splice tray using slack loops.

From the access terminals, a drop cable, with a maximum of a several hundred feet (10-300 m), spans to the customer premises where the ONT is located. Again, APC connectors are used. At the ONT, the optical signal is converted into an electrical signal using an optical-electrical converter (OEC). This converter splits the signal into the services required by the customer. Various interfaces are used including RJ-11 twisted pair jacks for POTS, Category 5/6 RJ-45 10/100/1000 Base-T Ethernet jacks for high-speed data (IP interface), and 75 ohm coaxial ports for CATV and digital broadcast services (DBS). The 75 ohm coaxial port is connected to the set top box, which is connected to the TV.
Implementing a PON test strategy

Test Strategy for characterizing connectorized passive optical networks

For a PON network, different opinions exist regarding the need whether to connectorize the elements along the link or splice directly to splitters, terminals, and the CPE. Using connectors adds loss along the network but enables easy evolution of the network and provides more test access points. This chapter describes the deployment stages of fully connectorized networks only. Specific paragraphs detail the test procedure differences when other networks principles are used.

<table>
<thead>
<tr>
<th>Stage</th>
<th>Task</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Feeder cable construction</td>
</tr>
<tr>
<td>2</td>
<td>Distribution cable construction</td>
</tr>
<tr>
<td>3</td>
<td>Frame installation and connection</td>
</tr>
<tr>
<td>4</td>
<td>Feeder cable acceptance test</td>
</tr>
<tr>
<td>5</td>
<td>Distribution cable acceptance test</td>
</tr>
<tr>
<td>6</td>
<td>Equipment installation and test</td>
</tr>
<tr>
<td>7</td>
<td>Troubleshooting</td>
</tr>
</tbody>
</table>

Table 3.4 PON Deployment Stages

Figure 3.6 Tests to perform on a PON

Central Office Tests
- OLT output power
- Video output power
- ONT + link input power
- OLR
- Patch cord loss
- WDM coupler loss

Feeder, distribution, and drop fibers
- Attenuation
- Splice loss
- Connector loss
- Connector reflectance
- OLR
- Fault location
- Continuity check

Premises Tests
- ONT output power
- Video/data + link input power
- WDM coupler loss
Stages 1 and 2: PON Feeder and distribution cable construction
For a PON application, the optical cable containing the fibers is laid using one of three methods.

1. Direct burial installation—The optical cable is inserted in direct contact with the soil. It is laid either in a pre-dug ditch or via simultaneously plowing a slot and inserting the cable. This is the most expensive method and is used only for high-density population areas.

2. Duct installation—The optical cable is placed inside a pre-installed duct that runs between access points.

3. Aerial installation—The optical cable is placed on poles or towers, allowing routing of the optical transmission path above ground. This is the method of choice for network overlaid (brownfield) applications.

Feeder and the distribution sections then may be spliced in an enclosure, either to join two cables or to divide one large cable into multiple smaller cables to diverge to different locations. Cables and fibers can be of any type, depending on the application and the density, including loose tube or ribbon fiber. Light, battery operated fusion splicers—some with the capability to estimate splice loss—have been designed specifically for FTTx applications. However, fusion splicing is still complex and time-consuming because it requires managing the fiber in the splice tray, cleaning and cleaving the fiber, then splicing the fiber as well as closing the splice enclosure. Another method of deploying the distribution cable is to pre-engineer and build the entire fiber section, which is placed on deployment reels. The cable is then laid out and terminated with limited need for additional field splicing.
Depending on the construction method used the enclosure is located either underground in a manhole or vault or on the aerial. A bi-directional OTDR measurement from the CO at 1310/1550 nm is recommended to qualify the splices (see Figure 3.7).

For the majority of installations, chromatic and polarization mode dispersion measurements are not required due to the low bit rate and short span length. However, recently in several countries, providers have expressed an interest in the ability to deliver 10 Gigabit Ethernet services over FTTH architectures which may necessitate characterization of the fiber for some service providers.

Figure 3.7 Fiber splice test points
Fusion Splicing

Fiber splicing (Courtesy of Piani)

<table>
<thead>
<tr>
<th>Step</th>
<th>Task</th>
<th>Objective</th>
<th>Test Tool</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Lay fiber/cable for the feeder and</td>
<td>Provide fiber between the CO and customers</td>
<td></td>
</tr>
<tr>
<td></td>
<td>distribution network</td>
<td></td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>Position splice enclosures and splice</td>
<td>Perform continuity along the feeder</td>
<td>Fusion splicers and OTDR</td>
</tr>
<tr>
<td></td>
<td>fibers for the feeder</td>
<td></td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>Position splice enclosures and splice</td>
<td>Perform continuity along the distribution</td>
<td>Fusion splicers and OTDR</td>
</tr>
<tr>
<td></td>
<td>fibers for the distribution network</td>
<td>network</td>
<td></td>
</tr>
</tbody>
</table>
Stage 3: PON Frame Installation and Connection
After feeder and distribution cable construction, frame installation can be performed. For PONs, several frame types are installed. The first frame is the fiber distribution hub (FDH), housed in an outside cabinet which also contains the splitters used in the network. While other splitter combinations may be used, FTTH networks frequently employ a single FDH with a 1x32 splitter structure or multiple FDHs with 1x8 and 1x4 cascaded splitters. At the FDH all fibers coming from the CO are spliced to the splitters.

An OTDR measurement, performed from the CO, is recommended to verify splice quality. All connectors from the splitter output should be placed in the parking lot to store the pigtails in a separate path to reduce fiber congestion. All fibers going to the distribution network are connectorized. A measurement at 1310/1550 nm is then performed from the hub to check the splice quality.

The second frame, the access terminal, is located close to the customer premises, and it consists of a splice enclosure located either on a pole or in a manhole. A set of fibers (usually 4, 6, 8, or 12) coming from the cable are extracted and spliced to be connectorized to drop cables. An OTDR measurement from the hub is recommended to check the splice quality.

The third frame to be installed is usually the fiber distribution frame (FDF) cabinet which is located at the CO. Like the other frames, the fiber is spliced to a pigtail in order to be connected to the patch panel. An OTDR measurement is then performed from the CO to check the splice quality. For large networks, other FDFs may be located on the feeder in order to distribute the different cables. The same process applies to those frames.
Chapter 3: Deploying and Troubleshooting Fiber Networks

Frame 1: Fiber distribution hub (Courtesy of ADC)

Frame 2: Access Terminal (Courtesy of ADC)

Frame 3: Distribution Frame (Courtesy of ADC)
At the end of this process, the feeder and the distribution network are complete and ready for end-to-end acceptance testing consisting of overall distance, insertion loss, and ORL. Additional tests such as 1310/1550 nm OTDR testing may also be performed at this point to verify splice and connector distances, losses and reflectances.

<table>
<thead>
<tr>
<th>Step</th>
<th>Task</th>
<th>Objective</th>
<th>Test Tool</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Check connectors on jumpers, test equipment, fiber links on distribution hub</td>
<td>Verify connector quality</td>
<td>Fiber inspection scope</td>
</tr>
<tr>
<td>2</td>
<td>Install FDH and splitter and distribution network</td>
<td>Verify continuity between feeder and distribution network</td>
<td>Fusion splicer, OTDR</td>
</tr>
<tr>
<td>3</td>
<td>Install terminals</td>
<td>Verify continuity between the distribution network and the drop cable and customer premises equipment</td>
<td>Fusion splicer and OTDR</td>
</tr>
<tr>
<td>4</td>
<td>Install CO FDF</td>
<td>Verify continuity between the CO equipment and the feeder</td>
<td>Fusion splicer and OTDR</td>
</tr>
</tbody>
</table>

The Fiber Distribution Hub (FDH) is installed first. Then, the terminal and the Fiber Distribution Frames (FDFs) are installed. The splitter is spliced, and the CO and terminal are spliced and connected. Testing is performed with an OTDR during splicing. Connectors at the splitter location are left open.
**Stage 4: PON feeder cable acceptance test**

Before acceptance testing can begin, the technician must ensure that both the connectors and patch cords (those used to test and those in use on the network) meet network operator requirements for insertion loss and reflectance. The tests may be performed either with an OTDR (1310/1550 nm) or with a VFL operating in the 635-670 nm range and a source/power meter/ORL meter combination.

With an OTDR, the test is performed from the CO or from the hub or from both for bi-directional OTDR testing. To simplify testing, the feeder cable acceptance test typically does not include the splitters. If the does include the splitters, the process is complex and requires a high performance (high dynamic range, short dead zone) OTDR with the capability to test through splitters. The inputs of the splitters are connected to the feeders, and one output of the splitter to be tested is connected to a 2 km receive cable. Testing from the CO (or eventually from the hub) then qualifies the feeder part and the input/output of the splitter. To test the next output of the splitter, the technician can move the receive cable from one output to a different one and repeat the test from the CO (or the hub).

To expedite feeder cable acceptance testing, technicians may select to test only one or a few outputs from the splitter. When the feeder fiber link for a given splitter has been completely tested, the test equipment at the CO and the hub can be moved to a different fiber and the process is then repeated. It should be noted that receive cable is the same as launch cable—it is simply used at the receive end instead of the launch end.

If a source/power meter/ORL meter combination is employed, two operators with test equipment are needed, one at each end of the feeder cable. When the feeder cable acceptance test does not include the splitters, one test set is connected to the CO and another is connected to the hub connector just before the splitter. If the feeder cable acceptance test does include the splitters, then the process is more complex. In order to take the splitters into account, the inputs of the splitters should be connected to the feeders. The
The technician then measures the fiber at the CO and one output of the tested splitter at the hub. Without disconnecting the test equipment at the CO, the technician must move the test equipment to a different output of the same splitter and perform another test. This process is continued until the entire splitter is tested.

Technicians may opt to test only one or a few outputs rather than testing all of the outputs of the splitter. Test equipment is connected at the CO to the fiber which is linked to the next splitter. Then, the test equipment is connected at the hub to this splitter and testing is performed. This process is continued until all splitters are tested.

In all cases, measurement results should be compared to the specific requirements for loss distance, and ORL as specified in the design of the fiber cable network. Test results are compared to these requirements and corrective action is taken when needed by testing with the OTDR for the same fiber. Each result is recorded in the unit and also on a customer network database for maintenance purposes. Both manual and automated testing options are available.

<table>
<thead>
<tr>
<th>Step</th>
<th>Task</th>
<th>Objective</th>
<th>Test Tool</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Check connectors on jumpers, test equipment, fiber links</td>
<td>Verify connector quality</td>
<td>Fiber inspection scope</td>
</tr>
<tr>
<td>2</td>
<td>Continuity check/Insertion loss/ORL testing of the feeder cable</td>
<td>Acceptance test of the feeder cable</td>
<td>Source/Power meter, ORL, or OTDR</td>
</tr>
</tbody>
</table>
Stage 5: PON distribution cable acceptance test

Prior to performing acceptance testing on the distribution cable, the technician must ensure that both the connectors and the patch cords (used for both the test and the network) are in good working condition. Tests then can be performed with either an OTDR or a source/power meter/ORL meter combination. If the technician uses an OTDR, the test should be performed from either the hub or from the customer premises, or from both if bidirectional testing is required. If the source/power meter/ORL meter combination testing option is used, two technicians are required—one at each end of the link. A VFL also may be used to check continuity. The results are compared to the design specifications of the fiber network.

Figure 3.9  Acceptance testing between the CO and the splitter
Specific operator requirements exist for loss, distance and ORL based on the fiber cable network design. Test results are compared to these requirements and corrective action is taken when needed. If the issue involves the connection or the jumpers, the corrective action should be tested with the same loss test tools. Each result should be recorded in the unit and also on a customer network database for maintenance purposes. Manual and automated test solutions are available.
### Step 1: Check connectors on jumpers, test equipment, fiber links
- **Objective**: Verify connector quality
- **Test Tool**: Fiber inspection scope

### Step 2: Continuity check/Insertion loss / ORL testing of the distribution cable
- **Objective**: Acceptance test of the distribution cable
- **Test Tool**: Source/Power meter, ORL, or OTDR

For the overall link (feeder and distribution), the international standards provide the minimum design requirements as outlined below.

<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Class A</td>
<td>5 dB</td>
<td>20 dB</td>
<td>32 dB, all port terminated (20 dB otherwise)</td>
</tr>
<tr>
<td>Class B</td>
<td>10 dB</td>
<td>25 dB</td>
<td>32 dB, all port terminated (20 dB otherwise)</td>
</tr>
<tr>
<td>Class C</td>
<td>15 dB</td>
<td>30 dB</td>
<td>32 dB, all port terminated (20 dB otherwise)</td>
</tr>
</tbody>
</table>

### Stage 6: PON Equipment Installation and Testing
OLT and ONT installation are two of the key steps in building the PON infrastructure. OLTs are installed at the central office, while ONTs are installed at customer premises. ONUs are typically installed at remote central offices. As described in Stage 1, the drop cable connecting the terminal and the ONT is installed at the customer premises using either aerial or underground methods depending upon greenfield or brownfield application. The ONU is connected to the customer premises using copper cabling.

The drop cable is usually pre-connectorized to expedite installation. If the connector caps remain in tact, no further testing during installation is required. With the drop cable installed, the technician can connect the drop at the terminal, then, at the FDH, connect the specific fiber for the customer. In some cases the drop cable may not be pre-connectorized. Here, connectors are added and tested in the field.
Upon completion, the technician may opt to perform insertion loss testing again by connecting one unit to the FDH and the other unit to the fiber at the ONT. An alternative method is testing with an OTDR from the ONT location.

<table>
<thead>
<tr>
<th>Step</th>
<th>Task</th>
<th>Objective</th>
<th>Test Tool</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Insertion loss testing of the distribution network with drop cable</td>
<td>Acceptance testing for the distribution network with drop cable</td>
<td>OTDR or source/Power meter</td>
</tr>
</tbody>
</table>

Note that not all OTDRs are equipped to disable the test if customer traffic is present. When turning up a residence on a link that is already in service, the technician must never send a test signal from the premises back upstream unless the test system is equipped with an out-of-band wavelength and its performance has been verified with respect to the network.

Once customers have been established, ONTs and OLTs are installed. Next, the drop cable is installed. When the ONTs and OLTs are activated, testing with a selective power meter (SPM) is performed at the ONT and OLT locations. At the OLT location, ORL testing may be performed.

![Figure 3.12 PON equipment installation testing](image)

ONT on customer premises
Photonic Layer Testing of OLT and ONT Signals

The following discussion assumes that the OLT itself, typically consisting of a 1490 nm transmitter, a 1590 nm transmitter with amplifier, a 1310 nm photodiode/receiver, WDMs and patch cords, has been fully tested internally. The ONT, which usually consists of a 1310 nm transmitter, 1490 nm photodiode/receiver, 1550 nm photodiode/receiver, WDMs and patch cords, is also assumed to be fully tested. In such instances, the following tests may then be performed.

The first step in characterizing the OLT is to measure the output level of the OLT with a power meter and to verify the performance against the manufacturer’s suggested requirement for signal output level. If 1550 nm video service carried on the line, then a selective 1490/1550 nm power meter is needed. If a basic power meter is used, the OLT will activate the 1490 nm and 1550 nm wavelengths independently. In instances where the video service uses the 1550 nm transmission with analog signals, the ORL measurements of the feeder part should be taken from the CO at the feeder cable location. Then the technician should proceed to the ONT location. On the ONT side, the following tests are required.

A selective 1310/1490/1550 nm power meter operating in Through mode is used to measure the output level at the ONT while the OLT transmits at 1490 and 1310 nm downstream. This will trigger the ONT to transmit 1310 nm bursts upstream. Technicians may elect to measure only the 1490 nm and 1550 nm and not the 1310 nm wavelengths using the selective 1490/1550 nm meter. This test validates overall signal level. If the OLT displays a warning that there is no 1310 nm signal present, then the ONT must be replaced.
When there is a physical layer failure at a single ONT, the ONT is disconnected and a power measurement is performed.

<table>
<thead>
<tr>
<th>Step</th>
<th>Task</th>
<th>Objective</th>
<th>Test Tool</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Power level measurements at OLT location</td>
<td>Verify the levels at the OLT with pass/fail information</td>
<td>Selective power meter at 1490 &amp; 1550 nm</td>
</tr>
<tr>
<td>2</td>
<td>ORL measurements from OLT</td>
<td>Verify the ORL in case of analog video transmission with pass/fail information</td>
<td>ORL meter</td>
</tr>
<tr>
<td>3</td>
<td>Power level measurements at ONT location</td>
<td>Verify the levels at the ONT with pass/fail information</td>
<td>1310, 1490, 1550nm selective power meter with through mode or 1490, 1550nm selective power meter</td>
</tr>
</tbody>
</table>

International standards provide the minimum and maximum values for the transmitters and receivers:

**Tx Output Power**

between +9 dBm max and -7.5 dBm min, depending on standards

**Rx Receiver**

between -3 dBm max and -33.5 dBm min, depending on standards

(will vary according to bit rate, network class, and PON type. See ITU-T standards G.983.1 and G.984.2 for further details)

**Stage 7: PON Troubleshooting**

When there is a physical layer failure at a single ONT, the ONT is disconnected and a power measurement is performed.
Analyzing customer premises or Remote Central Office (RCO) equipment outages when not all RCOs or OLTs are out of service

When some—but not all—RCOs are working, this is an indication that the OLT and feeder sections are working properly. It further indicates that the problem is either in the distribution network or the RCO. In this case, the technician should go to the out-of-service RCO to test the power levels of the link. If the wavelength power level is correct, this indicates that this RCO has failed and should be replaced.

If the wavelength power level is incorrect, the distribution network may have failed, and the wavelength power level should be measured at the terminal. If the level is correct, then the drop cable should be changed. If changing the drop cable does not correct the level, the technician should disconnect the connector at the FDH of this fiber link and take an OTDR measurement from either the FDH or the RCO.

<table>
<thead>
<tr>
<th>Step</th>
<th>Task</th>
<th>Actions if results are Good</th>
<th>Actions if results are Bad</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Measure power level at RCOs and verify connections</td>
<td>Go to step 2</td>
<td>RCO failure. Repair RCO</td>
</tr>
<tr>
<td>2</td>
<td>Measure power level at terminal and verify connections</td>
<td>Change the drop cable</td>
<td>Go to step 3</td>
</tr>
<tr>
<td>3</td>
<td>Disconnect connector on FDH and perform OTDR measurement of distribution network as well as verify connection</td>
<td>Not applicable</td>
<td>Identify break and repair the fiber</td>
</tr>
</tbody>
</table>
Analyzing an RCO outage when all the RCOs are out of service

When all RCOs are out of service, the technician should check the OLT to verify whether it is transmitting the correct power levels. If it is not, then OLT should be changed. If the OLT is transmitting correctly, this indicates an outage in the fiber network. An OTDR measurement should be performed from the OLT connection toward the FDH to locate a possible break or bend in the feeder. The portion of the OTDR trace just before the high loss on the display indicates the location of the splitter. If a break or bend is located, steps must be taken for repair. If the feeder is performing properly, this indicates that the problem may be at the FDH. The technician should check the connectors and splitters at this location. Additionally, the technician should use an OTDR to test in either direction to identify a bend or break.

<table>
<thead>
<tr>
<th>Step</th>
<th>Task</th>
<th>Actions if results are Good</th>
<th>Actions if results are Bad</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Disconnect and measure power levels at OLTs</td>
<td>Go to step 2</td>
<td>Change OLT</td>
</tr>
<tr>
<td>2</td>
<td>Measure the feeder from the OLT location with OTDR</td>
<td>Go to step 3</td>
<td>Outage in fiber network.</td>
</tr>
<tr>
<td>3</td>
<td>Go to the FDH and check connections</td>
<td>Go to step 4</td>
<td>Fix connection</td>
</tr>
<tr>
<td>4</td>
<td>Test distribution network with OTDR</td>
<td>NA</td>
<td>Outage in fiber network</td>
</tr>
</tbody>
</table>
Spliced PON testing variation versus connectorized PON
When a connector is replaced by a splice, then the test access point disappears at that location. During fiber construction, where splices are used a bare fiber adapter will be required to connect the test equipment to the fiber link for most tests. During troubleshooting, due to the lack of mechanically connectorized ends, a clip-on fiber identifier may also be used to verify whether traffic is present on the fiber section under investigation. However, a fiber identifier cannot be used for acceptance testing, as the loss generated by the clip-on mechanism is not clearly defined and the power readings are not accurate.

For example, during installation when there is no connector after the splitter, the technician must position a bare fiber adapter at the splitter output to qualify the feeder during construction. Once this is completed, the splitter output can be spliced to the distribution fiber. The test then can be performed from the other end of the distribution fiber using the same process.
Acceptance testing is always performed end to end, so the fiber links to be analyzed must be connectorized at both ends. If there is only one overall zone for the complete FTTx optical network (connectors at each end of the network only), then an overall acceptance test can be performed. If the fiber link is split into two zones with connectors, then acceptance testing should be carried out on each of the two zones.

It is particularly challenging to test through the splitter with an OTDR when there are multiple fibers connected after the splitter. When performing an OTDR test from the CO to the customer premises, the best approach is to disconnect all of the splitter outputs except the one running to the customer premises. This makes it much easier to interpret the OTDR test results; however, a large loss—up to 19 dB with a 1x64 splitter—will still be seen at the splitter location. For further information about testing through the splitter with an OTDR, please refer to Appendix A.

The source/power meter/ORL combination provides an end-to-end test which only offers overall measurements of the fiber link. For troubleshooting purposes, a clip-on fiber identifier can be used to identify the faulty segment. As an example, moving from the customer premises to the CO, at each location where the

Figure 3.15  OTDR trace sample
independent fiber can be accessed, the operator can perform a power level measurement with the fiber identifier. If the power level is low, this indicates that the fault is between the test location and the CO. If the power level is normal, this indicates the fault is between the test location and customer premises. By performing tests at multiple locations, the location of the fault can be identified easily within a given zone.

Alternatively, the OTDR can be used to identify the faulty section. To avoid interrupting customer traffic, the technician should use an out-of-band test wavelength such as 1625 nm. If there is a failure on the distribution network, then the fiber at the distribution network is connecting only the customer experiencing service disruption. The technician should disconnect this fiber at a given connectorized test access and perform the 1625 nm test from this location. The fault can then be localized, even if the complete FTTx network is in service.

**Point-to-Point FTTx testing variation versus PON testing**

Point-to-point FTTx networks are much easier to install and test, and the processes are similar to the deployment of metro or long haul optical networks. However, the fiber count is higher and the distances are much shorter with a P2P network, thereby increasing the installation cost.

### P2P Network Deployment Stages

<table>
<thead>
<tr>
<th>Stage</th>
<th>Task</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Fiber cable construction</td>
</tr>
<tr>
<td>2</td>
<td>Frame installation and connection</td>
</tr>
<tr>
<td>3</td>
<td>Fiber cable acceptance test</td>
</tr>
<tr>
<td>4</td>
<td>Equipment installation and test</td>
</tr>
<tr>
<td>5</td>
<td>Troubleshooting</td>
</tr>
</tbody>
</table>
The fiber patch panels are located only at the fiber ends—a fiber distribution frame at the central office and the remote fiber distribution frame at the remote central office, or alternatively the terminal close to the customer premises.

During the fiber construction and for the complete acceptance tests, there are only point-to-point fiber links. Therefore, the process is straightforward, and either dual wavelength OTDRs operating at 1310/1550 nm or a source/power meter/ORL meter combination is used.

The transmission equipment is not as complex and includes Ethernet DSLAMs, or ONU. Frequently, only one or two wavelengths (1310/1490 or 1490/1550 nm) are used thereby requiring the use of only standard power meters.

Because there is no frame along the complete distribution fiber, troubleshooting tests are performed from the CO first, then at any accessible test point on the network to the customer premises. OTDRs or a source/power meter/ORL meter combination is used.
### Troubleshooting process with an OTDR:

<table>
<thead>
<tr>
<th>Step</th>
<th>Task</th>
<th>Actions if results are Good</th>
<th>Actions if results are Bad</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Check connectors on jumpers, equipment, test equipment</td>
<td>Go to Step 2</td>
<td>Clean connectors and repeat Step 1</td>
</tr>
<tr>
<td>2</td>
<td>Check CO equipment output signal</td>
<td>Go to Step 3</td>
<td>Repair CO equipment — Reconnect equipment to the fiber network — Bring the network back into service</td>
</tr>
<tr>
<td>3</td>
<td>Check fiber link with an OTDR</td>
<td>Go to Step 4</td>
<td>Repair the link then perform acceptance testing — Reconnect equipment — Bring the network back into service</td>
</tr>
<tr>
<td>4</td>
<td>Check remote CO equipment output signal</td>
<td>NA</td>
<td>Repair remote CO equipment — Reconnect equipment to the fiber network — Bring the network back into service</td>
</tr>
</tbody>
</table>

### Troubleshooting process with a source/power meter/ORL meter combination:

<table>
<thead>
<tr>
<th>Step</th>
<th>Task</th>
<th>Actions if results are Good</th>
<th>Actions if results are Bad</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Check connectors on jumpers, equipment, test equipment</td>
<td>Go to Step 2</td>
<td>Clean connectors and repeat Step 1</td>
</tr>
<tr>
<td>2</td>
<td>Check CO equipment output signal</td>
<td>Go to Step 3</td>
<td>— Repair CO equipment — Reconnect equipment to the fiber network — Bring the network back into service</td>
</tr>
<tr>
<td>3</td>
<td>Check remote CO equipment output</td>
<td>Go to Step 4</td>
<td>— Repair remote CO equipment — Reconnect equipment to the fiber signal network — Bring the network back into service</td>
</tr>
<tr>
<td>4</td>
<td>Check fiber link with source/power meter (link repair made with an OTDR)</td>
<td>None</td>
<td>— Repair the link, then perform acceptance testing — Reconnect equipment — Bring the network back into service</td>
</tr>
</tbody>
</table>
**Recommended test tools**

Whether the user is deploying a point-to-point network or a PON, the final acceptance test is ultimately the process of characterizing the complete optical network. It includes continuity checks, insertion loss and optical return loss measurements of the end-to-end network through analysis of the transmission optical wavelengths. A minimum of two wavelengths are selected (usually 1310/1550 nm) in order to identify any macrobends along the network, a common fault when installing short distance and high fiber count networks. The loss generated by a bend increases as the test wavelength increases. For example, if a localized section has a loss of 0.2 dB at 1310 nm and a loss of 0.5 dB at 1550 nm, then this can be clearly identified as a bend. Other point losses do not vary significantly with wavelength.

![Screen capture of OTDR trace with bending](image)

*Figure 3.17  Screen capture of OTDR trace with bending*
Visual Fault Locators and Fiber Inspection Microscopes
Most fiber link problems occur at the extremities and involve components such as the connectors and patch cords. These problems often can be resolved with tools such as visual fault locators (VFLs) (635-650 nm) and fiber inspection scopes. The visual fault locator tests continuity of a short fiber link up to 5 km and identifies any critical bends or breaks in patch cords or inside fiber cabling. When a bend or break does occur, the visible red light leaks from the fiber and can be detected visually by the operator.
The fiber inspection microscope enables the visual quality inspection of connectors located on patch cords or inside the patch panels or distribution frames. Damaged or dirty connectors cause high losses that degrade the transmission quality. For safety reasons, operators may prefer video display fiber scopes over simple fiber microscopes because the video display scope does not place the operator’s eye in a direct line with the output of the fiber when inspecting the connector.

A video inspection scope with a handheld display

These photos illustrate a damaged, dirty connector on the left and good quality connector on the right.
Optical Time Domain Reflectometers (OTDRs)

OTDRs are also used for acceptance testing. They measure the link distance; locate bends, high splice losses, high connector losses, and high connector reflectances; measure end-to-end loss and end-to-end optical return loss (ORL); and perform continuity checks. The OTDR is a single-ended tester meaning that it can be used to perform all tests from any test point location.

An ideal continuity check is performed by adding a network element beyond the remote end of the link. The network element can be a specific known length of fiber such as a launch cable, a specific known reflective event, or a specific known or predetermined OTDR fiber signature. The network element is detected directly by the OTDR to validate the fiber link continuity. To obtain the best accuracy in splice loss, connector loss, ORL measurements, and automatic continuity checks, technicians may choose to perform OTDR testing from both ends. This cancels any OTDR test effects which could provide inaccurate readings if fibers that do not have the same optical characteristics are spliced together.

OTDRs are the main tools used to troubleshoot a fiber link which may lead to the assumption that an OTDR could be used to test the complete FTTx optical network. However, with the use of PON technology, OTDR traces become difficult to interpret. Traditional OTDR analysis uses both the Rayleigh backscatter effect and the Fresnel back reflection effect to characterize events and fiber ends. The OTDR then displays a trace that is the signature of the fiber link. The Rayleigh backscatter effect takes into consideration that part of the transmitted light is reflected back to the transmitter due to fiber impurities. The Fresnel back reflection effect relies on index discontinuities. It has the same effect as Rayleigh backscatter effect, but its effect is due to fiber-to-air connections.
These effects result in an easy-to-understand signature of the link as long as there is only one branch along the link. Because a PON network provides multiple branches along the link, the PON OTDR signature becomes very complex to interpret. This is because the OTDR is not able to differentiate the reflected light coming from a default of one branch location to another branch location, since they are located at the same distance to the OTDR. The result is an aggregate trace that does not display the effect of each branch individually. Rather, it displays the effect of all of the branches at the same time.

PON Reflectometer Trace Analysis takes into consideration the backscatter response after the splitter that is used in PON networks. It allows for the identification of the optical branch, the localization of faults, and attenuation estimation.

Therefore, in order to analyze a PON network, a specific test procedure, described in Appendix A, is required.
Source, Power Meter, and ORL Meter Combination

With a combination of source, power meter, and ORL meter, the operator can perform automatic continuity checks and measure end-to-end insertion loss as well as end-to-end ORL. ORL testing is particularly important if analog video at 1550 nm is used, because this technology is sensitive to reflectance. Several different test configurations can be used.

In one approach, the source is placed at one end and a power meter at the other end. The power meter reads the power level coming from the source. If there is no reading, this indicates the presence of a crossover along the link or a discontinuity. When the fiber count is high, then it becomes more difficult to identify crossovers. The technician then can use either a VFL or move the power meter along the fiber ports to identify the one that is receiving the signal from the source. The insertion loss is measured by activating the source at one end and the power meter at the other end. Some users conduct an ORL measurement at both ends.
Another approach uses a loss test set, consisting of a source and a power meter within the same unit, at each end. In some cases one or two ORL meters are also used. This approach improves repeatability by performing bi-directional loss measurements over two fibers.

**Bi-directional Loss Test Sets**
A final option is to use two bi-directional loss test sets containing built-in ORL measurement capabilities at each end of the link. This approach automates bidirectional loss and ORL measurements for faster handling and test time and for improved repeatability. Some bidirectional loss test sets can also provide link distance information.
Remote Test Systems and Clip-Ons
When the fiber link is not in service, a test access point easily can be obtained by disconnecting the fiber. Transmission wavelengths can be used for the test. For in-service testing or out-of-service testing without fiber disconnection, the operator can add a specific network element that enables the test equipment to be connected to the network. Out-of-transmission band wavelengths, most commonly 1625 nm, are used for the test. Complete optical network management systems (ONMS) are available to serve as optical fault management tools.

The clip-on fiber identifier/power meter can be used to perform tests at any location where the individual fibers can be accessed. By slightly bending the fiber, the clip-on device provides an indication of the directivity and the power level of the optical signal. However, it becomes difficult to obtain an accurate measurement when multiple wavelengths are transmitted over the same fiber.
**Selective Optical Power Meters**
A 1310/1490/1550 nm selective power meter with Through mode is needed for FTTH applications with ONTs. The 1490 nm and 1550 nm wavelengths are used for downstream and the 1310 nm wavelength for upstream traffic. By connecting one port of the unit to the OLT fiber port and activating the OLT, simultaneous 1490/1550 nm power readings are provided. The power meter can also be used to measure the signals coming from both the OLT and ONT. The 1310 nm wavelength provided by the ONT is activated by the 1490 nm signal coming from the OLT. The technician can connect one port to the fiber coming from the OLT and a second port to the fiber coming from the ONT. Then the power meter can simultaneously measure 1310 nm, 1490 nm, and 1550 nm power levels.
<table>
<thead>
<tr>
<th>Test Equipment or tool</th>
<th>Function</th>
<th>Use</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inspection Scope</td>
<td>Visual inspection of connectors</td>
<td>Fiber link construction and troubleshooting</td>
</tr>
<tr>
<td>VFL (Visual Fault Locator)</td>
<td>Continuity check up to 5 km, break/bend visual identifier for fiber along patch panel/hub areas</td>
<td>Fiber link construction and troubleshooting, at locations where fibers are accessible</td>
</tr>
<tr>
<td>Optical Talk Set</td>
<td>Enables communication between operators using cable link</td>
<td>When two operators are required for the end-to-end test</td>
</tr>
<tr>
<td>Light Source / Power meter Or bidirectional loss test set</td>
<td>Measures the fiber link insertion loss (end-to-end loss), tests continuity</td>
<td>Fiber link construction, acceptance testing and troubleshooting</td>
</tr>
<tr>
<td>Power Meter only</td>
<td>Measures the output power of equipments</td>
<td>Equipment and fiber link turn-up and troubleshooting</td>
</tr>
<tr>
<td>Power Meter with clip-on device</td>
<td>Estimates the output power along the link</td>
<td>Equipment and fiber troubleshooting, at any location where fibers are accessible, even when connectors cannot be accessed</td>
</tr>
<tr>
<td>Clip on fiber identifier</td>
<td>Identify traffic and traffic direction on fiber, may also estimate output power along the link</td>
<td>Equipment and fiber troubleshooting, at any location where fibers are accessible, even when connectors cannot be accessed</td>
</tr>
<tr>
<td>1310/1490/1550 selective power meter with Through mode</td>
<td>Measures the power levels of equipments and fiber link when OLT/ONT are connected</td>
<td>Fiber link and equipment (ONT/OLT) turn-up and troubleshooting</td>
</tr>
<tr>
<td>ORL meter</td>
<td>Measure the overall ORL loss</td>
<td>Fiber link construction and troubleshooting</td>
</tr>
<tr>
<td>OTDR</td>
<td>Measures all the characteristics of the fiber link</td>
<td>Fiber link construction, acceptance testing and troubleshooting</td>
</tr>
</tbody>
</table>

For further information about fiber optic test tools and methodologies, refer to the *JDSU Reference Guide to Fiber Optic Testing*.
Optical fiber offers nearly unlimited bandwidth, low attenuation and distortion, is not sensitive to radio frequency interference, and does not interfere with other systems. There is no doubt that fiber will continue to penetrate further into the access network. However, the cost of replacing each of the hundreds of millions of twisted copper pairs that are used to deliver services to consumers and businesses around the world is enormous. Service providers making the tough decision on how quickly to upgrade the local loop to fiber must consider the ability of the existing copper plant to meet current and future market demands.

Service providers have shown remarkable ingenuity in extending the life of the existing copper plant. The evolution of xDSL technologies, from ADSL to ADSL2+ to VDSL2, has demonstrated the ability to deliver ever increasing data rates—enough to deliver triple-play voice, video, and data services. However, one of the greatest challenges of xDSL technologies is that the limitations of copper make it difficult to achieve long range and high data rates at the same time. Service providers have addressed this challenge by migrating toward more advanced data transmission technologies and by rolling out fiber closer to subscribers—all in an effort to enable higher data rates by reducing the copper loop length. VDSL2 is particularly suitable for these ‘deep fiber’ rollouts through its ability to offer up to 100 Mbps symmetric service over loops shorter than 100 meters and 38 Mbps over loops shorter than 1000 meters.

<table>
<thead>
<tr>
<th>Type of Service</th>
<th>Max Download (Typical)</th>
<th>Typical Service</th>
</tr>
</thead>
<tbody>
<tr>
<td>ADSL1</td>
<td>8 Mbps</td>
<td>3 Mbps</td>
</tr>
<tr>
<td>ADSL S=1/2</td>
<td>12 Mbps</td>
<td>6 Mbps</td>
</tr>
<tr>
<td>ADSL2+</td>
<td>24 Mbps</td>
<td>6-8 Mbps</td>
</tr>
<tr>
<td>ADSL2+ bonded (2)</td>
<td>48 Mbps</td>
<td>12 Mbps</td>
</tr>
<tr>
<td>VDSL1</td>
<td>52 Mbps</td>
<td>20 Mbps</td>
</tr>
<tr>
<td>VDSL2</td>
<td>100 Mbps</td>
<td>30-40 Mbps</td>
</tr>
<tr>
<td>BPON</td>
<td>622 Mbps</td>
<td>Shared by 32</td>
</tr>
<tr>
<td>EPON</td>
<td>1 Gbps</td>
<td>Shared by 32</td>
</tr>
</tbody>
</table>

Table 4.1  Triple-play access network speeds
As IPTV deployments increase, operators are focusing on the bandwidth that will be needed for triple-play IP television (IPTV), both HDTV and standard definition television (SDTV). Major European and North American service providers believe that VDSL2 coupled with newer video compression technologies, such as MPEG-4 AVC, will provide the necessary bandwidth to offer triple-play services as well as Internet access and VoIP. For example, a major European telco has committed €3 billion to a fiber to the neighborhood deployment in fifty of the largest cities in Germany, involving shortening loops by deploying remote DSLAMs closer to end users. VDSL2 will be used to deliver triple-play services in the last mile. The largest US service provider is also deploying a similar VDSL-based FTTx triple-play network. Investments such as these indicate that the future of the copper plant is still bright and that the challenges of dealing with the installed copper plant will be with us for decades to come.

But, these new DSL technologies are placing new demands on copper pairs which were never specifically designed to deliver triple-play services. Copper’s limitations, such as the potential for attenuation and crosstalk problems, increase the challenge of triple-play service delivery. As a result, providers delivering these new bandwidth intensive services over copper face new and difficult turn-up and troubleshooting challenges.

The Physical Challenges of Triple Play over Copper

The high data rates, and associated higher signal bandwidth used to deliver broadband services such as IPTV, place far more demands on the existing copper plant than plain old telephone service (POTS). Use of a much wider frequency spectrum breaks new ground unfamiliar to providers. Isolating the cause for packet loss is often the most difficult part of the troubleshooting challenge. Correlating physical impairments to issues seen at the data layers—such as packet loss which in turn causes visual impairments for video programs—requires multilayer analysis.
**Foreign voltage**
Typically, the first physical layer test checks for the presence of foreign voltage, either DC or AC, on an open pair. Excessive foreign voltage on an open pair can be a safety hazard and is also a sign that the loop is in poor health.

**Foreign DC voltage**
DC voltage on an open pair typically comes from the central office battery. This happens when a pair is crossed with one or more working pairs as a result of physical cable damage, water in the cable, or faulty splices. This cable damage can result in excessive noise or attenuation on an xDSL circuit.

To accurately test for DC voltage, the technician must use a 100K Ohm impedance digital voltmeter and test on a dry pair, with the circuit open on the far end. Ideally an open pair should have no DC voltage, but xDSL can tolerate some amount—generally less than 3V DC tip to ring, tip to ground, and ring to ground is considered acceptable. If a DC voltage measurement is taken on a wet pair connected to the central office, it should range between 48V and 52V DC tip to ring, -48V to -52V DC ring to ground, and less than 3V tip to ground.
The lower the resistance of the path between the pair under test and the working pair, the greater the measured voltage will be. A cross can be the result of physical damage to the cable, water ingress in the cable, or faulty splices. If excessive VDC exists, use the insulation (leakage) resistance and opens measurements to further refine the source of the problem, which could be a resistive fault, wet section, etc. When the cause of the problem has been determined, the technician should use the time domain reflectometer (TDR) or resistive fault locator (RFL) functions, to locate the fault for removal.

**Foreign AC voltage**
The presence of electrical energy from nearby power lines can induce energy into an open pair resulting in AC voltage on the line. This AC voltage can act as a noise source, degrading the quality of xDSL transmission. To measure foreign AC voltage, the technician must use a digital voltmeter. Typically the tip-to-ring voltage should be less than 3V AC and the tip-to-ground and ring-to-ground voltage should be less than 10V AC.

**Tip and Ring**
In telephony, the terms that represent the conductors that comprise a circuit are known as tip and ring or A and B. The tip and ring term stems from the early days of telephony when operators made telephone connections using .25 inch phono plugs similar to those used today for stereo headphones. Early systems also carried a third wire which was a ground. The “tip” (A) was the tip of the plug and was the positive (+) side of the circuit. The “ring” (B) was a conductive ring right behind the tip of the plug and was the negative (-) side of the circuit. Right behind the ring was the “sleeve” which was the ground connection.

Siemens 3-pin
The ground (sleeve) is no longer used today for individual pairs.
Balance

Good balance is critical for a pair to be able to cancel noise or interference coupled in from sources inside or outside the cable binder. Balance measurements qualify the overall health of the pair. Theoretically, the tip and ring conductors should have the same electrical characteristics. The more alike (balanced electrically) the conductors are, the better the pair is at rejecting noise, both internal and external to the cable. A balance measurement quantifies the amount of similarity. If a pair is out of balance, it will not reject noise as well, resulting in reduced xDSL data rates or even loss of synch.

The major characteristics of a pair are resistance, capacitance, and inductance. A well-balanced pair has these electrical characteristics closely matched on tip and ring which means the same amount of leakage to ground is present. No series resistive faults, and no half taps, loads or other factors that would make one of the main characteristics of tip different from ring should be present. When the tip and ring conductors are very close to the same electrical balance, the circuit can reject interference. The reason for this is that any noise coupled into the pair will be coupled in equally on each wire, thus cancelling each other. The noise will then be “rejected.” On a pair with poor balance, the noise is coupled in on one wire stronger than the other thereby establishing a differential signal across the tip and ring. This unwanted signal appears as noise damaging the DSL signal at the receiver. The more noise, the lower the quality of the DSL signal.
Chapter 4: Troubleshooting the Copper Plant for IP Services

Typical Copper Pair Faults

**Short**
Continuity between two wires without an external jumper or loop back plug in place.

**Cross**
One or both wires in a pair cross with one or both wires in a second pair. Identify by use of third pair individual wire.

**Ground (Earth) Fault**
One or both wires in a pair read continuity with external ground or shield.

**Intermittent**
A break or “open” which appears and disappears as the wire or connector is flexed. Identify by same method as open pairs.

**Open Pairs**
A break or “open” fault appears. Use a split-pair loop back plug or individually use one wire in each of two pairs to determine which individual wire is open.
Reversed
The termination of a wire pair is reversed, for example, when a telephone circuit tip at one end appears as ring at the opposite end.

Split
Between two pairs, one wire of one pair is reversed with one wire of the second pair. Identify by use of third pair individual wire or by visual inspection of the color coding at the termination.

Circuit balance
The conventional method of measuring circuit balance is to measure the amount of power influence or noise to ground on a pair. The technician then measures the amount of noise that develops across the pair due to this power influence. Subtracting noise from power influence gives the circuit balance. Power influence is AC power line induced interference and is measured in dBnC—that is dB reference to noise through a C message filter. The noise this causes on the pair is also measured in dBnC. The measurement result is in dB and the higher the number, the smaller the amount of noise voltage across tip and ring and the better the pair balance. A result of 60 dB or greater indicates the pair has good balance and should be acceptable. For example, if a circuit has a balance of 60 dB and has 72 dBnC of power influence, it can reject 60 dB of that which leaves only 12 dBnC of noise on the pair. This is a very small amount of noise and would not affect the quality of the circuit.
To get accurate readings, these measurements must be made with the line dialed into a quiet termination state reached by dialing a quiet termination number. No termination or the wrong termination may result in a higher than actual reading of noise on the circuit. And there are limitations to the conventional balance test as well. The most significant is that this technique requires a certain amount of power influence to obtain a good balance reading. Too little power influence results in balance results that are lower than the actual values. For example, if the power influence is only 50dBm, the best possible balance reading would be 50dB even if the noise is 0.

Longitudinal balance
Longitudinal balance is similar to circuit balance, except the power influence limitation mentioned above is overcome because the test set applies a signal to the pair that takes the place of power influence in measuring the pair’s ability to reject noise. During the longitudinal balance test, a 1,000 Hz (1 KHz) tone is applied by the test set to the pair under test from tip to ground and ring to ground, then the set measures the level of induced noise on the pair. A calculation of the balance is performed and displayed. This method also enables accurate measurements on unterminated lines. The measurement is given in decibels, with 60 dB or greater generally considered acceptable, 50 to 60 dB marginal, and less than 50 dB unacceptable.

Resistive balance
Balance can also be measured using a resistive measurement by grounding both conductors at the far end and comparing tip-to-ground and ring-to-ground measurements. These measurements should be within 5 \( \Omega \) or 1% of each other, whichever is less. This measurement requires intervention at the far end of line, so it is often replaced by a longitudinal balance measurement.
Capacitive balance
Another method to compare the similarity between tip and ring is with an opens meter. If you measure the loop length on an open pair, the tip-to-ground measurement should be within 1% of the ring-to-ground measurement. A measure of greater than 1% can indicate faults on one of the conductors.

![Capacitive balance measurement](image)

Figure 4.1 Assessing capacitive balance using opens measurement

Insulation resistance
An open loop should have no electrical path from tip-to-ring, ring-to-ground, or tip-to-ground. Cable damage, wet sections, or bad splices can create a resistive path for current to flow. Minor resistive faults affect the balance of a pair, which leads to noise. Major resistive faults create a hard short on the circuit and a complete signal loss.

To identify resistive faults, the technician must use a digital ohmmeter. When conducting this test, it is important to simulate conditions that will exist on the line when it is turned up for service. This will help to ensure that the stresses applied by the working line will not affect the pair condition. The ohmmeter used should be designed specifically to measure cable pair insulation resistance. In general, a resistance greater than 3.5 MΩ is considered acceptable for xDSL.
Faults with a very low resistive value such as shorts can be located using a TDR. More highly resistive faults (high resistive fault value) may not be visible on a TDR trace. To locate more highly resistive faults, use a resistive fault locator (RFL).

**Load coils**
The quality of voice calls is degraded by the capacitance on a very long loop, typically greater than 18,000 feet. The attenuation caused by this capacitance can be counteracted by adding inductance to the line, in the form of load coils—devices placed into a telephone circuit between the end office and the subscriber to step up the voltage and compensate for signal loss due to bridged taps. The load coil is an inductive device that acts as a high-frequency choke. They typically are placed 6,000 feet apart, but may be spaced at longer or shorter intervals depending on the loading scheme used.

Load coils, however, act as a low pass filter, causing very high attenuation for signals above 4 kHz. This attenuation is too great for xDSL signals, so any loads on a loop intended to carry xDSL must be removed. This includes loads on bridged taps.

A simple way to check for load coils is to use a load coil detector. A load coil detector sends a frequency sweep to find the attenuation characteristic of a load coil. It then returns the number of loads on the line. These test sets often require that there be at least 1,000 feet of line after each load. Some test sets provide the capability to identify distance to first load coil, a key metric.

Another way to locate load coils is to use a TDR. A load coil shows up on a TDR in the same manner as an open—an upward spike. The leading edge of the spike can be used to approximate the distance to the load. A TDR can accurately show the distance to the first load only, so if more than one load exists, a new TDR trace should be made after each load is removed.
Loop length
Attenuation of a loop increases as loop length increases. If the loop is too long, the modems may not synch at all, or will have a low data rate. Because of this, there is a maximum length for a pair to be acceptable for ADSL. This length depends on wire gauge, gauge changes, and overall line condition. The approximate maximum length is 18,000 feet for ADSL, 15,000 feet for ADSL2, and 18,000 feet for ADSL2+. To obtain rated levels of bandwidth associated with ADSL2+, however, the loop must be shorter than 8,000 feet. Loop lengths for VDSL2 are considerably lower. One major service provider has set 2,300 feet as a conservative starting point for its rollout but has considered extending to 3,000 feet or more if conditions are acceptable. Between 2,300 and 3,000 feet, results of other measurements are closely examined to be sure they are well within limits. A fault condition that may be tolerable on a 2,000 foot loop could impair performance on a 3,000 foot loop.
Figure 4.3  xDSL downstream speed as a function of loop length

One way to test loop length is to use an opens meter on a disconnected pair. An opens meter measures capacitance, and since cable specifications in North America require that all wiring used for outside plant have a constant capacitance of .083 mf per mile between tip and ring, an opens meter can measure the capacitance of an open pair and convert the measurement to distance. The capacitance between tip and ground or ring and ground, however, is dependent upon the cable fill type. As a result, to accurately measure opens tip to ground or ring to ground, the test set must be configured for the proper fill type. Since bridged taps add capacitance, this length includes any bridged taps for all opens results. Wet sections of cable also add capacitance and if present, will increase the length for opens results. Some test sets allow the entry of custom values for capacitance per distance unit for unique cable types.
Loop length also can be measured by placing a short on the far end, and then taking a distance to short measurement. This measures the resistance of the loop and will not include the length of bridged taps. Resistance is dependant upon both cable gauge and temperature, so these values must be set for a test set to accurately determine resistive length. This measurement of loop length is not affected by the presence of bridged taps. Therefore, if the loop length measured resistively to the short at the far-end (distance to short) is substantially shorter than the loop length measured previously using the opens meter, this can be a possible indication of bridged tap on the line.

Finally, a TDR can also be used to measure loop length by opening the line at the far-end and locating the open on the TDR trace.

Figure 4.4  Measuring loop length using opens measurement
Wideband Noise

High frequency noise can have a negative impact on xDSL service, causing bit errors or in the worst case complete loss of synchronization between the ATU-R and ATU-C. The table below lists common noise sources and the corresponding frequencies and tones. To measure noise, use a test set capable of wideband noise measurements rather than one designed only for voice-band noise measurements. It should support noise measurement across the wideband range up to 30MHz to cover the ADSL/2/2+ and VDSL1/2 frequency spectrums.

<table>
<thead>
<tr>
<th>Noise Source</th>
<th>Noise Frequency (kHz)</th>
<th>ADSL Tone #</th>
<th>Tone Frequency (kHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>ISDN BRI</td>
<td>40</td>
<td>9</td>
<td>38.8125</td>
</tr>
<tr>
<td>HDSL passband center</td>
<td>196</td>
<td>45</td>
<td>194.0625</td>
</tr>
<tr>
<td>HDSL passband max</td>
<td>392</td>
<td>91</td>
<td>392.4375</td>
</tr>
<tr>
<td>HDSL2</td>
<td>274</td>
<td>64</td>
<td>276</td>
</tr>
<tr>
<td>T1 passband center</td>
<td>772</td>
<td>179</td>
<td>772</td>
</tr>
<tr>
<td>E1 passband center</td>
<td>1024</td>
<td>237</td>
<td>1022</td>
</tr>
</tbody>
</table>

Table 4.2 Noise sources and tones

To filter only for noise that will affect ADSL, the technician must set the tester use a “G” Filter. For VDSL2, use of a test set that has selectable filters applicable to the VDSL band plan in use is essential. However, the standards bodies have not yet defined filters that adequately cover the VDSL spectrum. JDSU has defined a set of “J” filters which do cover the VDSL spectrum. For example, the J-20K8 filter, which provides a 20KHz to 8MHz band width, should be selected for noise testing of VDSL pairs employing this band plan. To ensure accuracy, the far end should be terminated with a 100 Ω resistance for any noise measurement. Wideband noise is measured in dBm, and is usually a number below zero. For example, the maximum noise depends on the attenuation and desired line rate. A typical requirement for VDSL is that measured wideband noise should be below -50 dBm.
<table>
<thead>
<tr>
<th>Filter</th>
<th>Technology</th>
<th>Lower 3dB</th>
<th>Upper dB</th>
</tr>
</thead>
<tbody>
<tr>
<td>E-Filter¹</td>
<td>ISDN BRI, DDS-LL</td>
<td>1 kHz</td>
<td>50 kHz</td>
</tr>
<tr>
<td>F-Filter¹</td>
<td>HDSL</td>
<td>4.9 kHz</td>
<td>245 kHz</td>
</tr>
<tr>
<td>G-Filter¹</td>
<td>ADSL</td>
<td>20 kHz</td>
<td>1.1 MHz</td>
</tr>
<tr>
<td>G2-Filter</td>
<td>ADSL2/2+</td>
<td>20 kHz</td>
<td>2.2 MHz</td>
</tr>
<tr>
<td>J-20K8</td>
<td>VDSL 2-8</td>
<td>20 kHz</td>
<td>8.5025 MHz</td>
</tr>
<tr>
<td>J-20K12</td>
<td>VDSL 2-12</td>
<td>20 kHz</td>
<td>12.0025 MHz</td>
</tr>
<tr>
<td>J-20K17</td>
<td>VDSL2-17</td>
<td>20 kHz</td>
<td>17.0025 MHz</td>
</tr>
<tr>
<td>J-20K30</td>
<td>VDSL2-30</td>
<td>20 kHz</td>
<td>30.0025 MHz</td>
</tr>
<tr>
<td>J-640K17</td>
<td>VDSL2</td>
<td>640 kHz</td>
<td>17.6 MHz</td>
</tr>
<tr>
<td>J-17M25</td>
<td>VDSL2</td>
<td>17.6 MHz</td>
<td>25 MHz</td>
</tr>
<tr>
<td>J-17M30</td>
<td>VDSL2</td>
<td>17.6 MHz</td>
<td>30 MHz</td>
</tr>
<tr>
<td>J-25M30</td>
<td>VDSL2</td>
<td>25 MHz</td>
<td>30 MHz</td>
</tr>
</tbody>
</table>

¹ Defined per IEEE standard 743

Table 4.3 xDSL Filters by Technology

Figure 4.5 Wideband noise measurement (VDSL frequency range)
Wideband noise can be the result of radio frequency interference from disparate digital services in the same cable, AM or shortwave radio stations, and other sources. If excessive wideband noise is detected, use of a test set with a wideband spectral analysis capability is recommended as the next logical troubleshooting step. This provides more in-depth analysis of the noise on the line to determine the detailed characteristics of the interference and aid in identification of the potential noise source. Intermittent bursts or spikes of noise, commonly referred to as impulse noise may also be present as discussed in detail in the next section. Due to its intermittent nature, impulse noise may or may not be detected using the wideband noise measurement. If impulse noise is suspected, the impulse noise measurement is recommended for further isolation and analysis of the problem.

While it is critical to make the determination whether or not the level of noise present is sufficient to impair the service and to attempt to ascertain the noise profile and potential source as described above, it is also important to consider potential root causes. Though a noise interference source can be strong enough to impair xDSL service by itself, in many cases pair imbalance and/or poor bonding and grounding of the cable are significant contributing factors. Specifically, unbalanced pairs and cable bonding and grounding issues reduce the noise immunity of the pair making it far more susceptible to interferers.
Impulse Noise

Environmental influences and electromechanical devices can cause short duration energy bursts, noise events that are not continuously present, often making them difficult to detect with a standard wideband noise measurement. This is called impulse noise. To measure impulse noise, use a test set that can measure (count) noise spikes over an extended time frame. This will increment a counter for every instance that the noise crosses a user-defined threshold. The test set should support wideband impulse noise testing across the applicable DSL technology spectrum. To fully address the ADSL, ADSL2/2+ and VDSL2 frequency spectrums, up to 30 MHz is required. Impulse noise testing should be performed with selectable filters appropriate for the xDSL service and band plan in use (ADSL: G, ADSL2+: G2, VDSL: J).

It is beneficial to have a test set that provides additional analysis capability to show the number of impulse events at 3dB points above and below the prescribed threshold. This is useful in helping to isolate potential impulse noise problems. Noise levels can fluctuate with the time of day, the nature of the source itself or worsen over time do to degrading plant conditions. Coupled with the intermittent nature of impulse noise, these factors can mask the presence of a significant problem. For example, if there no recorded events presents at the defined threshold but just 3dB below it, accepting the pair without further investigation could lead to issues later with an increase in the noise source level or worsening plant conditions.

Figure 4.6  Impulse noise measurement and impulse event capture and analysis
Another beneficial feature is Impulse Capture. This feature is helpful to analyze the “signature” in amplitude versus time of the captured impulse noise event. The Impulse Capture feature can be set up to trigger on a detected impulse noise event which exceeds the defined threshold and automatically, graphically captures the impulse noise event for on screen viewing or storage and export for later analysis.

**Spectral Analysis of Wideband Noise**

As mentioned previously, interference from sources such as other digital services or AM or shortwave radio stations can reduce the performance of a copper loop. Spectral analysis provides further analysis and source identification once the presence of a noise problem has been determined to be present. Spectral analysis provides a graph of where noise disturbers exist in frequency and amplitude across the spectrum of interest and aides in correlation with potential interferences sources. For example, if a T1 circuit in the same binder group is interfering with the xDSL service, a spectrum analysis will show the noise pattern on the line and allow the technician to overlay a mask of the frequency characteristics for a T1 signal. If the characteristics match up with the noise on the circuit, then the interferer can be identified as a T1 source. In this example, spectral analysis provides a quick and simple means to identify the interference source.

![Figure 4.7 Spectral analysis with ADSL mask active](image_url)
For providers deploying higher-speed data or triple-play services, a test set that is capable of performing a spectral noise analysis up to 30 MHz should be used in order to support analysis of noise across the ADSL and VDSL frequency spectrums. This is critical for two reasons. First, the frequency ranges supported on many older test sets typically top out at or just beyond the upper limit of the ADSL frequency spectrum and do not fully address the ADSL1/2/2+ and VDSL1/2 spectrums. Second, and more importantly, the potential negative impact of excessive noise on ADSL and VDSL circuits carrying triple-play services is far greater than for traditional lower-speed or “best effort” Internet data service over xDSL. While some loss of data rate due to excessive noise was tolerable or even unnoticeable to traditional xDSL Internet data service users, packet loss or slower data rates due to noise on IPTV service can damage the QoS causing pixelization and frame freezes leading to a negative customer experience.

Ideally, the spectrum analysis measurement should also support analysis down to a noise floor that approximates or approaches normal ambient noise conditions on twisted pair plant—typically around -140dBm/Hz. This is important because noise levels are rarely static and often fluctuate with time, the source, or varying environmental and plant conditions. For example, it is very useful to be able to identify a potential source of interference that is present even if its magnitude at the time of testing is not significant enough to impair the xDSL service. As additional xDSL pairs are brought on line within the same cable or plant conditions worsen, the impact of the interference source could begin to be noticed in the form of errors, lost packets, and/or lower data rates.
Finally, the spectral analysis measurement should support both continuous and maximum hold (Max Hold) spectral graphing functions. In the normal or continuous viewing mode, noise values are plotted against frequency across the sampled bands, allowing a near-real time continuous view. With the Max Hold feature enabled, the maximum noise values detected over time are plotted against frequency. This provides a real troubleshooting benefit over snapshot graph views of noise amplitude vs. frequency because noise level values are not static and typically fluctuate over time. The Max Hold feature greatly enhances analysis as opposed to the user trying to visually discern the maximum values, in real time, across multiple frequencies.

![Spectral analysis without Max Hold feature enabled and with Max Hold feature enabled](image)

Figure 4.8  Spectral analysis without Max Hold feature enabled and with Max Hold feature enabled
TDR capabilities
A time domain reflectometer (TDR) is used to identify faults, such as bridged taps, wet sections, load coils, shorts, opens, and splices, and their location on copper cabling. Functioning much like sonar on a submarine, the TDR sends an electrical pulse down a copper pair and analyzes the energy that is reflected back. Reflection occurs when the launch pulse detects any changes in the impedance characteristics on the copper pair. The shape of the reflection identifies the nature of the fault. Any impairment that lowers the impedance creates a downward bump in the trace while an impairment that increases the impedance creates an upward bump.

The open end of a pair or a hard short will show the largest reflections on a trace. Opens, load coils, the end of a bridged tap, and wet sections, produce an upward trace. Shorts and the first portion of a bridged tap signature produce a downward trace. Splices typically produce a combination signature: upward followed immediately by a downward trace. Because the launch pulse attenuates as it travels down the pair, the further an event is from the test set, the weaker the reflection. Because cable gauge affects attenuation, a TDR's range is reduced on smaller wire gauges.
Resistive fault location

Highly resistive faults can be very difficult to detect and correct during initial testing and almost always worsen over time due to corrosion and other causes. Resistive fault location (RFL) provides a reliable and very accurate means for locating resistive faults before a cable repair is attempted. RFL is a technique that uses a series of resistance measurements to accurately calculate the distance to a fault. The test set uses built-in circuits to measure resistance on a faulted lead and a known good lead in order to calculate the distance from the tester to the fault. It also displays the total end-to-end loop length and distance from the far-end strap or short back to the fault. RFL is known as “strap-to-fault” for this reason. As stated above, RFL requires a known good wire or wires to use as reference. This can be the other wire in the pair or both wires of a separate pair. For a single pair measurement, the far end tip and ring must be shorted together, and for a separate pair measurement, the tip and ring of the good pair must be shorted to the faulted wire of the bad pair. The accuracy of an RFL depends on correctly setting the cable gauge and temperature, as this affects resistance.

Figure 4.10 Single pair RFL analysis showing fault at 1087 feet
Bridged taps
Bridged taps, or laterals, are excess lengths of wire that extend past the subscriber or are spliced in along the span. Many un-used bridged taps remain from when party lines were deployed and two or more taps were made on every line. Over time, the extra taps were cut and taken off the termination block, but left connected at some point to the loop. They became buried within the maze, and can be difficult to locate.

While they do not interfere with normal POTS service, bridged taps cause undesirable reflection that can distort the high-frequency signals in modern transmission technologies. They degrade the performance of an xDSL circuit by causing signal reflections at the splice point and at the end of the tap. These reflections become noise in the circuit degrading DSL performance. Bridged taps also add noise to the circuit by acting as antennas, picking up external noise along the tap, and attenuate the signal at the wavelength that corresponds to the tap length.
xDSL modems can tolerate some level of noise and attenuation, thus some bridged taps are acceptable for most types of xDSL service. Certain bridged taps may render the loop unacceptable. For ADSL service, the length of all bridged taps on the span should total less than 2,500 feet with no single tap exceeding 2,000 feet. Taps of 300 to 800 feet in length have a very large impact. For instance, several short taps (under 100 feet) might be acceptable, but one long tap may be unacceptable. However there should be no bridged taps close (within 1,000 feet) to the modem. The closer the bridged tap is to the modem, the higher the energy level of the reflected signals. If the tap is very close to one modem, the reflection may contain more energy than the signal from the other modem. At this point, the modem close to the tap may be unable to distinguish between data and unwanted reflections.

The expanded spectrum of VDSL2 makes it more susceptible to shorter bridged taps in the range of 20 to 100 feet. These short taps may be enough to cause a substantial rate reduction. Typically, much more stringent requirements, including in some cases removal of all bridged taps, are required for VDSL2 than for ADSL.
The most effective way to test for the presence of bridged taps is with a TDR. The impedance of a bridged tap shows up on a TDR trace as a downward spike followed by an upward spike. The beginning of the downward spike gives the approximate location of the tap’s splice point. The distance between the start of the downward trace and start of the upward trace will give the approximate length of the tap.

Bridged taps can also be identified by measuring the capacitive loop length and comparing that to the resistive loop length. Since bridged taps act as extra capacitance, the capacitive loop length will include the length of any bridged taps, while the resistive loop length will not. This is a two-ended test, requiring that the line be open for the capacitance measurement, and then shorted for the resistive measurement. So, it is not as simple as the TDR, nor does it indicate the location of each tap.

Figure 4.13  Capacitive loop length and resistive loop length
Opens
Opens are conditions in which tip and ring are completely disconnected because of a cable cut, storm damage, or damage from animals. Finding opens is relatively easy. No receive signal is detected and the DC path is broken. As a result, current is not present. One-sided opens differ from complete opens in that one wire, either tip or ring, is not broken. They are usually the result of corrosion caused by a degrading splice or moisture on a cable. However, a TDR sees a one-sided open as a complete open. Since the wire is still connected, the TDR shows additional faults downstream.

Wideband Testing Considerations
With the increase in deployment of ADSL2+ and now VDSL2 in order to offer high bit rate video services, the ability to address testing across the full ADSL2+ and VDSL2 frequency spectrums is a critical component when formulating installation and troubleshooting test strategies and plans. Traditional, lower-speed, ADSL Internet data service may not have always driven as strong a case for wideband testing. Packet loss, packet jitter, momentary loss of synch, or failure to maintain adequate bandwidth to support IPTV can severely degrade video performance and, with it, customer QoE. VoIP, while less sensitive to packet loss than IPTV and less bandwidth intensive, is nonetheless delay sensitive. In short, IP QoS problems (loss, delay, jitter, etc.), due to excessive wideband noise or attenuation on the physical medium, may have been acceptable for traditional ADSL-based Internet data service, but now represent a critical area of risk when deploying and maintaining triple-play service over DSL.

As noted throughout the preceding troubleshooting section, traditional wideband test equipment may not address the requirements of today’s ADSL2+ and, especially, VDSL2 circuits. Test sets should support measurements across the full wideband frequency range up to 30MHz for VDSL2. Additionally, the appropriate filters for ADSL, ADSL2/2+, and the different VDSL2
frequency bands should be selectable in order to support measurement of noise energy within the frequency range of the circuit/service being deployed.

In summary, the following installation and troubleshooting procedures should be considered when exploring best practices for wideband test approaches:

- **Wideband Noise** – Ideally should be measured during prequalification, installation, and as a first step in troubleshooting of potential noise sources. The correct filter for the xDSL service in use should be applied (including the new J filter previously defined). This measurement should be used as the first step to determine the presence of excessive wideband noise. If a noise problem exists, then further troubleshooting is warranted.

- **Impulse Noise** – An impulse noise problem may or may not be readily apparent from the results of the wideband noise test due primarily to the intermittent nature of impulse noise. A short-duration impulse noise test should be run during prequalification and, if possible, installation testing. If an impulse noise problem is suspected, longer duration testing may be necessary.

- **Spectral Analysis** – Spectral analysis capability is an essential troubleshooting tool that should be used for further analysis of noise problems including identification of the interference source.

- **Next Steps for Noise Problems** – In some cases, identification of the interference source causing excessive noise provides the path to resolution. For example, a situation in which another digital service in the same cable binder is not spectrally compatible and can be moved or addressed directly. In many cases, however, external noise sources such as AM radio stations cannot be directly controlled. Under these circumstances, investigation of the root cause is necessary. Pair imbalance and bonding and grounding issues can degrade the noise immunity of the pair to the point where external noise sources that would not normally have an impact become a significant problem.
Recommended Test Set Capabilities

**Video 1 STB**
- Video Stream 1 (Active)
  - MPEG-2 TS Broadcast-RTP
  - 192.168.0.152:7007
- Status: Stream Up
- Error Msg: Ok

**Video 2 QoS**
- Current: NA
- Max: NA
- Score: NA
- Hist: NA
- PCR Jitter: 2mS
- Latency: 31mS
- Cont. Err: 0.00% 0.00%
- Err. Ind.: 0

**VDSL Summary**
- Show Time: UP DOWN
- Actual Rate: 6364 K 51592 K
- Max Rate: 7312 K 74649 K
- Capacity: 87.0% 69.1%
- Noise Margin: 6.1 dB 16.1 dB
- Attenuation: — 3.7 dB

**VDSL Graphs**
- Tone: 1411 Bits 9 SNR 45.60
- Frequency: 5106 5382 5658 5934 kHz

**WB2 Impulse - Running**
- Elapsed: 00:00:45
- Count: 11
- Threshold: +3 dB
- Termination: 100 Ω
- Filter: J-20K8
- Dead Time: 200 ms

**Spectral - Running**
- ADSL
- dBm: 40
- kHz: 1200 1500

---

**Chapter 4: Troubleshooting the Copper Plant for IP Services**
While physical layer testing is important for xDSL service, application specific QoS testing is also important. Therefore, a test set that supports voice, video, and data applications, in addition to copper testing, provides the most efficient solution and enables more seamless correlation of potential problems between protocol layers. For example, excessive loss in data rate, attenuation, and/or noise margin seen at the VDSL layer can be more quickly isolated to cable faults, such as bridged taps, wet sections, or excessive noise caused by pair imbalance or poor bonding and grounding.

As an example, if in-service monitoring of the IPTV service reveals excessive bursts of lost packets and VDSL errors (code violations, CRC, uncorrected FEC errors, etc.) are observed, the problem often can be isolated to excessive noise or impulse noise on the line caused by pair imbalance or poor bonding and grounding. Recent lab and field testing has shown this to be a major source of IPTV degradation. Again, having the capability in a single instrument to make this diagnosis quickly and accurately is essential.
Developing a Test Strategy

A comprehensive test strategy to ensure that the copper plant is qualified to deliver reliable triple-play services must ensure that field technicians utilize best practices to deliver fast problem resolution and avoid call-backs. The best approach for turn-up typically begins at the physical layer and works through to the service layer for installation and the reverse for troubleshooting. The copper plant is tested first during turn-up because the metallic layer must be qualified as a prerequisite for delivery of triple-play services. For troubleshooting, the service is tested first and then xDSL tests are performed. This enables fast identification of network versus content problems that are usually outside of the technician’s control and avoids wasting time sectionalizing the problem.

There may be exceptions to this recommended flow. For instance, if the pair(s) to be put in service were exhaustively pre-qualified and checked ‘good’ immediately prior to installation, it may not be necessary to begin with the full battery of recommended metallic/physical layer tests. Rather, beginning with a VDSL test at the serving terminal or network interface device may suffice. The technician would then drill down to physical layer testing if a problem was detected. The following diagram shows a complete installation testing flow.
Chapter 4: Troubleshooting the Copper Plant for IP Services

Physical Layer Test
- Foreign VAC
- Foreign VDC
- Leakage Resistance
- Opens
- Capacitive Balance
- Longitudinal Balance
- Power Influence
- Circuit Noise
- Wideband Noise
- Impulse Noise
- Load Coil

Identify faults such as shorts, opens, grounds, wet sections, bridged taps, more highly resistive faults, loads, etc. Isolate as required and then locate using TDR, distance to short function, or RFL.

VDSL Test
- Current Bit Rate
- Max Attainable
- Relative Capacity
- Noise Margin
- Attenuation
- Line Errors

Identify WB noise and impulse noise issues and isolate to either pair imbalance or bonding and grounding issue.

IPTV Test
- PCR Jitter
- Latency
- Continuity Error
- Transport Error Indication
- Packet Loss
  - Rate
  - Period (Hole Size)
  - Distance

Based on error indications, isolate to network problem or physical layer related problem on the F2 plant or in-home wiring.

Network Issue
(Upstream of DSLAM)

COMPLETE ESCALATE/
HAND OFF

Figure 4.15 Typical copper plant installation testing flow for IPTV service delivery
Troubleshooting the Premises Wiring

Chapter 5
New Frontiers for Premises Testing

Traditionally, the customer premises wiring has been purely the responsibility of the homeowner. But with today’s proliferation of triple-play services, installation and repair technicians must cross the threshold of the customer’s home to ensure successful service deployment and trouble resolution. In this new model, the technician must find and eliminate improper wiring and sub-standard components—all while troubleshooting a mixture of wiring types including coax, twisted pair, and Ethernet cabling. Challenges are further complicated by the fact that homeowners may have distinct and non-negotiable preferences on aspects of the installation such as the location of the residential gateway (RG).

Figure 5.1 Today’s home networking configuration for triple-play customers is likely to include a hybrid of coax, twisted pair, wireless, and Ethernet.
Understanding the Role of the Premises Architecture

The triple-play access network delivery method chosen by the service provider has great implication with respect to the performance requirements placed on the customer’s home network infrastructure. Different methods of triple-play delivery inevitably require different home networking technologies.

In the FTTH model, the fiber is terminated at the home by an optical network terminator (ONT) that provides interfaces to serve analog and digital video over coax, data over Ethernet, and phone service over twisted pair wiring. Service providers use this model to provide digital video through quadrature amplitude modulation (QAM) or Internet protocol television (IPTV) or a combination of both. For the premises architecture that uses both QAM for broadcast video and IPTV for on-demand, the IPTV video shares the coax with the QAM digital video and is typically delivered using the Multimedia over Coax Alliance (MoCA) standard.

The FTTC and FTTN models offered by many service providers use VDSL2 or ADSL2+ as the network access technology. These too can involve a different home networking set-up. The DSLAM in the central office or remote DSLAM in the neighborhood uses copper and a DSL technology to connect to the DSL modem or RG. A range of network technologies is used to reach STBs. In this situation, digital video is deployed strictly as IPTV because twisted pair does not have the bandwidth necessary to carry a QAM cable TV signal. Service providers using xDSL are deploying IPTV using existing wiring in the home. HPNAv3 is often used in the xDSL architecture to deliver IPTV and data since it can run on existing twisted pair telephone lines or coax. Yet others deliver video and data over MoCA especially if the access is a QAM cable TV technology.

The goal in the home is “no new wires”—re-use existing infrastructure for the most economical deployment of services. Each service provider must evaluate which technology can provide the most advantageous balance between economic and performance benefits for triple-play services deployed in the home.
Emerging Home Networking Standards: MoCA and HPNAv3

Multimedia over Coax Alliance (MoCA)

MoCA is a home networking standard that occupies a 50 MHz band operating above the typical cable TV signal band. It can provide multiple high speed channels that coexist with up to 8 nodes per channel. MoCA provides a raw data rate of up to 270 Mbps and has a typical data rate of greater than 100 Mbps in a majority of the installations. A common approach is to use one frequency for a 100 Mbps data network and a separate frequency for a 100 Mbps IPTV network. A key benefit of the MoCA approach is the ability to work over existing coax and through multiple splitters in a home. MoCA can be configured to scan automatically for active channels so it can train itself to find the best frequency. Figure 5.2 illustrates the manner in which MoCA will adapt to and overlay with existing services.

![MoCA spectrum options](image)

**HPNAv3**

HPNAv3 can be run over twisted pair phone lines or coax or a combination of both. It is designed to provide guaranteed QoS and priority service flows. These factors combine to make HPNAv3 a technology well suited for QoE sensitive applications such as live HD television or VoIP. HPNAv3 operates in the 4-48 MHz range so it does not easily coexist with a DOCSIS cable modem or an interactive cable STB. In an IPTV installation HPNAv3 frequencies can be adjusted, such as 12-28 MHz, to co-exist with other services using frequencies below 12 MHz. HPNAv3 operates with a master-slave relationship with all communications being coordinated by the master. Direct peer-to-peer communications can occur but they must be scheduled through the master.
Testing HPNA and MoCA

Testing HPNA and MoCA requires testing the physical layer and the service layer. Leading-edge test sets provide a full suite of physical layer tests to find impairments in the coax or twisted pair network and offer a mode that will measure the performance of the separate nodes in the network. The purpose of HPNA drop testing is to verify that the HPNA link to each drop (outlet) is operating correctly and has the capacity to consistently carry data. This test should be performed at every drop within the premises to verify the functionality of the HPNA network, with only the host device installed. This test can be performed on either coax or telephone cable. Final HPNA network testing is recommended after all HPNA devices are installed and fully provisioned to verify that the entire HPNA network is operating correctly. The HPNA devices within the network should all be de-activated for this test to be completed correctly. In an HPNAv3 network, the optimal data rate for each link is 112 Mb. If the packet rate is zero and the data rate is over 96 Mb, the network should perform properly. If the packet error rate is non-zero or the data rate is below 96 Mb, the operator should verify that the network has been provisioned correctly.

The HPNA or MoCA mode tests also measure signal and data performance to ensure proper operating headroom. Future truck rolls may be reduced and customer satisfaction increased by fully qualifying each coax outlet in the home.
Residential network performance metrics, such as loss, latency, and jitter can have a significant impact on service quality. It is particularly important to consider the interwoven ramifications of correcting the individual effects. While jitter can be overcome by increasing the jitter buffers of the receiving equipment, this action will increase latency as well as the cost of the equipment. Additionally, if the jitter buffer is too small, then packet loss can be caused by noise in the system or service capacity issues. A real-time protocol such as VoIP or IPTV does not retransmit lost packets. The result is an impairment that directly affects the perceived quality of service and the customer’s QoE.
### Table 5.1 Recommended IPTV network transport thresholds.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Threshold</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Loss</td>
<td>&lt;0.1%</td>
</tr>
<tr>
<td>Jitter</td>
<td>&lt;100 ms</td>
</tr>
<tr>
<td>PCR</td>
<td>&lt;100 ms</td>
</tr>
<tr>
<td>IGMP Latency</td>
<td>250 ms</td>
</tr>
</tbody>
</table>

### Table 5.2 Data rates and QoS requirements.

<table>
<thead>
<tr>
<th>Service</th>
<th>Data Rate</th>
<th>QoS Requirements</th>
<th>Max BER</th>
<th>Effects of data errors</th>
</tr>
</thead>
<tbody>
<tr>
<td>VoIP</td>
<td>64 kb/s</td>
<td>Yes</td>
<td>1x10^-6</td>
<td>Garbled or robotic voice, dropped call</td>
</tr>
<tr>
<td>Standard Definition Television (MPEG-2)</td>
<td>8 Mb/s per channel</td>
<td>Yes</td>
<td>1x10^-6</td>
<td>Picture freeze or tiling</td>
</tr>
<tr>
<td>High Definition Television (MPEG-2)</td>
<td>20 Mb/s per channel</td>
<td>Yes</td>
<td>1x10^-6</td>
<td>Picture freeze or tiling</td>
</tr>
<tr>
<td>Standard Definition Television (MPEG-4 H.264)</td>
<td>4 Mb/s per channel</td>
<td>Yes</td>
<td>1x10^-6</td>
<td>Picture freeze or tiling</td>
</tr>
<tr>
<td>High Definition Television (MPEG-4 H.264)</td>
<td>10 Mb/s per channel</td>
<td>Yes</td>
<td>1x10^-6</td>
<td>Picture freeze or tiling</td>
</tr>
<tr>
<td>Online Gaming</td>
<td>1-4 Mb/s</td>
<td>Maybe</td>
<td>1x10^-6</td>
<td>Poor performance</td>
</tr>
<tr>
<td>Best Effort Data</td>
<td>2-15 Mb/s</td>
<td>No</td>
<td>1x10^-6</td>
<td>TCP-IP provides Retransmission transparent to user</td>
</tr>
</tbody>
</table>
In addition to loss, latency, and jitter emanating from the core network, a range of in-home issues, including phone line problems, Ethernet wiring mis-configuration or faulty termination, poor coax cabling integrity, and noise impairments can combine and compound to degrade the customer’s triple-play QoE. The service provider must put a testing plan into place to ensure that technicians pay careful attention to testing the premises network. The plan must facilitate a fast and easy way to identify and resolve these issues at the time of service deployment in order to avoid costly return trouble calls.

**Phone line issues**
Phone lines (twisted pair) in the premises often carry both voice service and data services using HPNA. In the FTTH architecture, the ONT or RG emulates the POTS network by providing all of the battery voltages, ring tones, and dial tones that were provided by the central office in the past. Consequently, troubleshooting VoIP carried over the phone wiring is very similar to troubleshooting POTS (see Chapter 8, Troubleshooting VoIP). Common errors affecting in-home wiring installations include opens, shorts, crossed wires, or broken wires. In-home technicians require tools that can help identify these and other problems quickly. For troubleshooting difficult phone wiring problems in the premises, the same techniques that are used on the local loop can be used. These are described in Chapter 4.
**Toning and Tracing**

It is often necessary to trace to the far end of a data, voice, or coax cable run to determine its location and to which CPE device it should be connected. Two methods are commonly used. One is to use a tone generator to produce a low voltage tone that can be traced through the cable jacket with a simple inductive tone tracing probe. Some tone generators have multiple functions such as built-in continuity tests and the ability to generate signals that will work through coax splitters and filters so that a video system can be traced while it is active and carrying video signals. Other tone generators have special power levels that allow the technician to trace cables through walls and under flooring.

Another, even faster, way to discover the far end of a cable run is to use remote identifiers that emit a unique signature when a main unit is attached at the near end. These identifiers are numbered and display this number on simple-to-use testers. This method allows a single technician to wiremap, log, and label unknown cable systems from one end.
<table>
<thead>
<tr>
<th>Typical Symptom</th>
<th>Fault Condition</th>
</tr>
</thead>
<tbody>
<tr>
<td>No dial tone – dead line</td>
<td>Clean Open Fault</td>
</tr>
<tr>
<td>Scratchy or intermittent dead line</td>
<td>Dirty Open Fault (Intermittent)</td>
</tr>
<tr>
<td>No dial tone – all incoming calls are busy</td>
<td>Short (Continuity)</td>
</tr>
<tr>
<td>Multiple symptoms (double dial tone, cross-talk, can't break dial tone)</td>
<td>Cross</td>
</tr>
<tr>
<td>Consumer equipment can not get dial tone</td>
<td>Reverse</td>
</tr>
<tr>
<td>Constant hum on line</td>
<td>Ground Fault</td>
</tr>
<tr>
<td>Multiple symptoms – cross talk, can not break dial tone</td>
<td>Split</td>
</tr>
</tbody>
</table>

Table 5.3 Phone line symptoms and fault conditions
Identifying Ethernet wiring issues
Many homes are now pre-wired with twisted pair wiring suitable for Ethernet data services. The EIA/TIA 568A standard offers two accepted color code standards, T-568A and T-568B. The actual wiring and connectivity is exactly the same. Both work equally well but they should not be mixed in the same install. Notice the difference in Figure 5.4. A-Standard uses orange for line 2 and green for line 3 while B-Standard uses green for line 2 and orange for line 3.

![Figure 5.4 T-568A and T-568B standards.](image)

Verification of proper termination is very important. Between 75% and 85% of the time in-home technicians spend troubleshooting can be attributed to efforts to resolve improper terminations. The most common termination faults can be found by a wiring verifier. Continuity tests include verification of pin-to-pin connections and the ability of the wire to carry a signal. The verification tester also tests for shields and voltage on line. This is a basic connectivity test, not a stress test.
### Fault Common Cause Diagnosis

<table>
<thead>
<tr>
<th>Fault</th>
<th>Common Cause</th>
<th>Diagnosis</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Good Cable</strong></td>
<td></td>
<td>![T568A/B Passing Cable (unshielded)]</td>
</tr>
<tr>
<td><strong>Opens</strong></td>
<td>An electrical discontinuity (break) in one or both wires in a pair</td>
<td>![OPEN](12345678 ID 345678 1)</td>
</tr>
<tr>
<td></td>
<td>![1. Are man-made](12345678 ID 345678 1)</td>
<td>Pass</td>
</tr>
<tr>
<td></td>
<td>![2. Occur in some type of enclosure](12345678 ID 345678 1)</td>
<td>Pass</td>
</tr>
<tr>
<td></td>
<td>![3. Are either fully open or partially open](12345678 ID 345678 1)</td>
<td>Pass</td>
</tr>
<tr>
<td><strong>Shorts</strong></td>
<td>An electrical connection between conductors of the same pair or any wire in the shield</td>
<td>![SHORT](12345678 ID 123456x5 1)</td>
</tr>
<tr>
<td></td>
<td>![1. Insulation breakdown](12345678 ID 123456x5 1)</td>
<td>Fail</td>
</tr>
<tr>
<td></td>
<td>![2. Physical damage](12345678 ID 123456x5 1)</td>
<td>Fail</td>
</tr>
<tr>
<td></td>
<td>![3. Faulty installation](12345678 ID 123456x5 1)</td>
<td>Fail</td>
</tr>
<tr>
<td></td>
<td>(staple punctures, nicked or cut cable, etc.)</td>
<td>Fail</td>
</tr>
<tr>
<td><strong>Mis-wire</strong></td>
<td>A wire or both wires of a pair not connected to the correct pins at the other end on the cable</td>
<td>![MISWIRE](12345678 ID 12345678 1)</td>
</tr>
<tr>
<td></td>
<td>![1. Man-made](12345678 ID 12345678 1)</td>
<td>Fail</td>
</tr>
<tr>
<td></td>
<td>![2. Typically appear at the termination point](12345678 ID 12345678 1)</td>
<td>Fail</td>
</tr>
<tr>
<td><strong>Split Pair</strong></td>
<td>An error in the twisting of the wires within the cable</td>
<td>![SPLIT WIRE](12345678 ID 12345678 1)</td>
</tr>
<tr>
<td></td>
<td>![1. Terminated in the wrong order](12345678 ID 12345678 1)</td>
<td>Fail</td>
</tr>
<tr>
<td></td>
<td>![2. Improper pulling technique](12345678 ID 12345678 1)</td>
<td>Fail</td>
</tr>
<tr>
<td></td>
<td>![3. Kinked and twisted cables](12345678 ID 12345678 1)</td>
<td>Fail</td>
</tr>
<tr>
<td></td>
<td>(1 not twisted with 2; 3 not twisted with 6)</td>
<td>Fail</td>
</tr>
</tbody>
</table>

Figure 5.5 Verification tester identifies common cable faults

For higher bandwidth applications and for high reliability communications, additional wiring testing or certification is needed to ensure that the service can be carried properly.
Traditional certification testers, based on TIA-568, certify the ability of the wiring plant to carry a specific frequency, which in theory corresponds to a data rate. Newer certifiers, based on IEEE 802.3 specifications, test the wiring and perform live data tests to certify the quality of the data connection and the connection rate.

Ethernet specifications up to and including 100 MHz Fast Ethernet are based on physical parameters and measurements that are linked to frequency signals that infer performance against a set standard of bandwidths. When the standards were developed, there was no clear roadmap depicting the network speeds or data requirements that would be necessary to support triple-play service deployment. Now, gigabit Ethernet uses all four pairs simultaneously, and requiring measurement of the other pairs’ influence on the pair under test. This has resulted in the creation of a measurement for power sum near- and far-end crosstalk (PSNEXT and FENEXT).

Traditional certifiers use frequency sweep signals to measure likely conditions for failure on a cable, and to measure the results of these signals against a set of pre-determined limits on each measurement with respect to TIA-568 specifications. The frequency sweep based certifiers are useful for testing the construction of the data cable. New certification tools use digital signal generation to measure the signal-to-noise ratio (SNR) and add the bit error rate test (BERT) as a demonstration of error free transmissions of actual digital data over that line. The BER test measures actual data flow rates from both ends of the cable run while FENEXT and PSNEXT try to infer data carrying capability by simulation with transient frequency signals.

This new class of certifiers is very applicable and useful for premises testing since they test the systems actual data performance. Speed standard testing, is performed by the newer certifiers, accurately determines how a cable will work in real network conditions. In addition, these new certifiers incorporate TIA-568 test specifications because the SNR contains all of the noise measurements of TIA-568 and the SKEW measurement is the same.
**Locating and resolving coax problems**
Existing coaxial home networks present a variety of challenges. Constructed by the home builder, the owner, or perhaps a previous service provider, the quality and routing of the network is rarely known. A high-quality coax installation should provide at least 30dB noise isolation to the outside world. This isolation is referred to as noise immunity. However, these networks often contain splitters, pinches, breaks, bad cables, un-terminated ends, bad connections, or amplifiers—all of which may lead to network problems and QoS issues. Proper grooming of the network to remove and repair or replace portions of the network to meet the triple-play service provider standards is critical to providing reliable services.

One of the biggest impediments to installing interactive services that utilize the existing coaxial network is determining the configuration of the distribution network. Further, the in-home technician must identify any potential problems with the network before turning up service. The technician may spend valuable hours trying to discover the network architecture manually through process of elimination. Locating faulty or undesired components is a tedious and lengthy process which often requires the technician to enter attics, crawlspace, and/or basements.

![Figure 5.6  Complexities of mapping existing networks.](image-url)
Traditional tools such as basic TDRs do not work through splitters and show only the distance to the first fault. More advanced TDRs require navigation and interpretation of a graphical trace which requires formal training and experience to perform correctly.

A new generation of test sets based on frequency domain reflectometry (FDR) provide a simple, single-ended method to map the coax network, to identify the components (barrels, splitters, and splices) and fault areas, and to determine associated distances to each. This mapping of the network can eliminate unnecessary trips into other areas of the home. It will also quickly direct the technician to the location of faults and components that require attention. The test set evaluates the coax network from the network interface device (NID) and identifies any splitters, open connections, cable faults, or mismatched impedances. This test works through and identifies multiple splitters and shows their distances. Interpretation of graphical traces is eliminated so the test may be performed by less experienced technicians.
A table view or graphical network representation provides the technician with a quick and precise overview of the elements within a home’s coaxial network. Simply connecting the test-set at the NID, the technician can determine the topology and location of any components and faults in the network. For coax networks incorporating devices such as splitters, FDR technology is superior because it will work through splitters and identify any items that may hamper the RF performance. FDR can also help locate other impairments of the network, including open cables, splitters, unterminated ports, and pinched or damaged cables.

Figure 5.7  The table or graphical view provides quick, easy-to-interpret results.
Graphical FDRs identify impedance mismatches and can measure opens, shorts, and structural damage to passive elements accurately, but they require extensive interpretation and understanding of splitter behavior in order for the technician to be able to properly use this information.
Using advanced signal processing, the single-ended map test takes the interpretation out of the measurement and provides a table view of the network. The table view identifies the common elements and shows the return loss and distance associated with the impedance mismatch.

Figure 5.9 The distance to each fault is shown.

With the quick view of the network, the technician can take action on any undesired issues, such as removing nested splitters and terminating open cables.

**Noise-causing service impairments**

Ingress is unwanted, over-the-air signals or electrical noise that seeps into the coaxial network. Coaxial cables and connectors are designed to be shielded and prevent over-the-air signals from interfering with the signals carried over the center conductor. However, coaxial home wiring networks are susceptible to ingress noise if the shielding, connectors, or terminations are substandard or damaged. Since off-air-signals and electrical noise are commonly
present in home environments at the same frequency as desired signals, ingress noise can disrupt or disturb operator-provided analog or digital services. Sources of disruptive ingress noise include household items with electrical motors, cordless phones, Home Amateur Radio (HAM), machinery, microwave ovens, etc. All of the in-home distribution signals including HPNA, DOCSIS, DSL, MoCA, and video can be affected.

The ingress noise challenge is complicated by the fact that noise sources are not always present or constant in level or frequency. Most installations are performed in the day time when fewer noise sources may be present in the home. This creates scenarios in which services can be installed and working, but later degrade when noise sources are turned on. The result is repeat service calls and dissatisfied customers. Network operators need to identify poorly shielded cable or connectors proactively in coaxial networks that have susceptibility to ingress noise so that weak spots can be repaired without a callback.

The ingress noise resistance test was developed to help reliably identify and pinpoint coaxial home wiring that is overly sensitive to ingress. An external noise source is used so that weaknesses can be detected even when local noise sources are turned off. Local transmission in the range where ingress noise is common is not permitted by the FCC for test purposes. But the existing FM carriers that operate between 88 and 108 MHz provide a good constant source of external energy. Measuring the received signal strength of FM carriers on a disconnected coaxial home network makes it possible to determine ingress noise resistance—the ability to shield against service disturbing ingress. The effectiveness of the coaxial shielding and connections to block ingress in the FM band is directly correlated to the ability to block ingress in the frequency range of DC to 45 MHz where the coaxial attenuation is the lowest.
Home shielding effectiveness in the FM band
Cable Labs® performed a study of digital transmission characteristics in a home in 1994. This study was performed on multiple homes served by various cable systems in different geographical regions. In that study one factor examined was home shielding effectiveness in the FM band. Table 5.3 presents the results as a histogram. The study showed that 99% of the homes have a shielding effectiveness of greater than 27 dB and 90% have a shielding effectiveness of greater than 42 dB. On the other hand, 10% will attenuate an external signal by less than 42 dB and 5% by less than 36.

<table>
<thead>
<tr>
<th></th>
<th>Average</th>
<th>50%</th>
<th>90%</th>
<th>95%</th>
<th>99%</th>
</tr>
</thead>
<tbody>
<tr>
<td>Home Wiring Shielding (dB)</td>
<td>67</td>
<td>58</td>
<td>42</td>
<td>36</td>
<td>27</td>
</tr>
</tbody>
</table>

Table 5.3  Histogram of shielding effectiveness

Figure 5.10  Plots of home shielding effectiveness
The ingress resistance/noise immunity test looks at a disconnected cable system to determine if any external carriers are present in the FM Band. If there are carriers present above a predetermined level, then the household will be susceptible to allowing ingress into the plant, which may cause video or data transmission problems.

The fact that ingress in the DC-45 MHz band is intermittent and requires more than a cursory glance with a spectrum analyzer is supported by experience with the JDSU PathTrak™ spectrum monitoring system. This system continuously scans the 5-65 MHz band and provides statistical reports of the return path ingress and noise performance over time. Figure 5.11 shows a performance history scan of a reverse path node that had intermittent ingress problems. Note the time dependency of noise. The noise in the 5-45 MHZ band is much higher and problematic from 4:00 to 8:00 in the evening—typically service technicians are not troubleshooting at this time.

![Figure 5.11 Time summary of ingress on a coaxial network. Performance summarized by 1 hour averages over 24 hour period.](image-url)
In poorly shielded or grounded coaxial network, time-of-day noise floor variations can occur repetitively and are commonly linked to increased activity in home networks. Figure 5.12 illustrates noise floor variations at 17 MHz over a 72 hour period collected with a performance logging spectrum analyzer. Figure 5.13 illustrates a properly performing coax network at the same frequency over same time period.

Figure 5.12 Average, max, min hold traces for a single frequency (17 MHz) over 72 hour time period collected in 15 minute time increments

Figure 5.13 Average, max, and min hold traces for a single noise frequency (17 MHz) over 72 hour time period collected in 15 minute time intervals
A properly shielded home coaxial network should not allow any off-air signals to ingress into network. Figure 5.14 illustrates a scan of frequencies in the FM radio band in a home network performed at the main junction point outside the home. This home network is well shielded with no FM signals penetrating into the network. Comparatively, Figure 5.15 illustrates a poorly shielded home network where FM signals compromise the coax shielding and ingress into the home network.

Figure 5.14  Frequency scan of 88-108 MHz on a well shielded home collected on a JDSU SmartClass Home meter

Figure 5.15  Frequency scan of 88-108 MHz on a poorly shielded home collected on a
Interpreting results
The noise immunity test may be used to characterize individual coaxial runs within a single home network and to identify the shielding effectiveness. The goal of these tests is to enable technicians to rapidly identify poorly performing runs and take the steps necessary to bring performance up to the provider’s specifications prior to leaving an installation. Ultimately, the goal is to prevent future service callbacks and improve customer satisfaction.

The following figures (5.16 through 5.19) illustrate shielding effectiveness variations within a single home network. All data were taken on field instruments using the ingress resistance/noise immunity test. First, data was collected on local off-air signal strength in the FM radio band by connecting a dipole antenna directly to the field test set and saving the results. Figure 5.16 illustrates the results of off-air signal strength local to the home under test.

Next, individual noise immunity tests were taken at three separate coax outlets in different rooms of the house (Figures 5.17 though 5.19). The shielding effectiveness of each run of coax to the separate rooms was calculated from these scans. The 45 dB of shielding effectiveness of one run (Figure 5.17) is good and should provide sufficient shielding against ingress noise sources. However, another run in the same house (Figure 5.18) provided only 11 dB of shielding effectiveness and was unlikely to provide resistance to ingress noise to reliably provide services over time.

The remaining coax run (Figure 5.19) was marginal in performance and provided 23 dB shielding effectiveness.
Figure 5.16 Spectrum scan of local off-air FM radio frequency band using dipole antennae connected directly to test instrument.

Figure 5.17 Noise Immunity Test scan of FM band in living room. 45dB of shielding effectiveness at 96.1MHz.

Figure 5.18 Noise Immunity Test scan of FM band in bedroom. 11dB of shielding effectiveness at 96.1MHz.

Figure 5.19 Noise Immunity Test scan of FM band in living room. 23dB of shielding effectiveness at 96.1MHz.
The noise immunity test simplifies the interpretation of results by allowing Pass/Fail thresholds to be set. These thresholds ensure that technicians make consistent and clear decisions. Pass/Fail indicator thresholds are programmable by the end user and can be fixed for field technicians. Programmability is necessary because end users may be deploying different equipment and services over time with different signal types and protocols. Certain protocols and services are more tolerant of noise than others.

Modern home coaxial networks built with good quality cable, components, and workmanship typically will have minimum 50 dB of shielding effectiveness in the FM band by design. Substantially less than this is indicative of a significantly damaged cable run, splitter, or connector or poor workmanship. Damaged cable and connectors are not likely to improve over time and are more likely to degrade. CableLabs' formal studies on home networking shielding in real networks indicate 95% of homes have greater than 36 dB of shielding effectiveness.

The ingress resistance/noise immunity test has been in practice for more than one year at cable operators. Noise immunity tests can be performed either as an absolute test where the absolute signal levels on the coax are examined, or as a calibrated test where a comparative/relative shielding effective number is calculated from the difference between the local off-air strength and the signal strength on the closed coaxial network. Typical off-air signal strength is between -10 and +10 dBmV. On average, operators have converged on 30 dB of shielding effectiveness as a decision threshold for replacement of connectors, splitters, or coaxial runs. For an absolute reading JDSU recommends a -30dBmV threshold. Any signal greater than -30dBmV should be considered a failure and warrants corrective action for proper network operation. Calibrated tests should be performed in areas where FM signal strength is very high (over +20dBmV) to avoid replacing cable that
may be performing well and providing adequate isolation. For calibrated measurements JDSU recommends creating a pass or fail conditional threshold based on 30 dB of shielding effectiveness.

Note: It is best to perform the noise immunity test outside the premises at the NID where FM radio signals are strongest. A FM reference signal from outside the premises is necessary to baseline premises measurements indoors. In order to ensure good noise immunity in a cable plant, the technician must ensure that all cable connections are tightened, and that all cables with damaged shielding as well as low quality splitters and barrels are replaced. Additionally, any non-terminated pieces of coax must be terminated.

**Importance of In-Home Test Strategy**
The challenge of reliable triple-play service delivery extends from the headend through the home. Historically, providers have been concerned with the transport of services from the core network to the customer’s network interface device. Through the evolution of the in-home network and its potential impact on customer QoE, providers now are asked to extend provisioning and service assurance strategies from the headend to the customer’s set top box.

The quality of service—and the provider’s ability to ensure service quality—relies not only on effective provisioning but also on the quality of the existing in-home wiring. While providers strive to maintain a philosophy of “no new wires” during service installations, in order to reduce costs both for the provider and customer, technicians must optimize performance across scenarios that can include multiple wiring interfaces such as coax, phone wiring, wireless, and Ethernet. Thorough, appropriate testing and verification of the in-home wiring and the service are key in ensuring that the installation will meet customer expectations and prevent future trouble calls.
Over a decade ago, the advent of digital television in the broadcast industry prompted a complete paradigm shift in both broadcast equipment and test practices. What once was a linear, continuous analog signal became a fragmented, nonlinear string of digital packetized data that contained a fraction of the original video information. Modulators were pushed to the perimeter of the broadcast rack—almost an afterthought to digital broadcast’s essential digital encoders and multiplexers. The antenna and television found their once direct union completely separated by a revolutionary device—the set top box. The network itself effectively went “dark” because programs were no longer visible at any point in the broadcast chain—they were hidden in transport streams, obscured by MPEG, ATSC, and/or DVB transport protocols. Technicians managing this network learned to rely on sophisticated test equipment more like protocol analyzers than video monitors or RF devices. The broadcast video network transformed into one that looked more like an IP network. Monitoring and troubleshooting this network required a new approach that carried with it a steep learning curve.

Today, digital television service delivery is again prompting sweeping changes as traditional telecom service providers strategize to include IPTV in their triple-play service offerings. Bundled service delivery is bringing with it a new class of test equipment and supporting test methods. While telecommunication networks historically have relied on protocol analysis for efficient packet-based transport, IPTV delivery introduces new challenges and considerations. Delivery of packet-based video over these networks is not necessarily a natural progression. Examined more deeply, IPTV service delivery entails more than the primary obstacle of getting the sheer volume of video payloads
down a DSL pipe. The maturation of Advanced Video Coding (AVC) technology has not shrunk video bandwidth use to insignificant levels, thereby removing barriers to implementation, as originally predicted. And, as providers are finding, there are greater challenges than the size of the video payload. Indeed, the very nature of that video payload creates a much larger complication, and digital television is once again the impetus for a steep learning curve.

The challenge of delivering IPTV service lies in the essential aspect of digital video: its packets are more than simple payload. Rather, they are a transport layer unto themselves. There is a common misconception that IPTV is the streaming of video over an Ethernet network—TV over IP. In reality, IPTV is the streaming of MPEG transport streams over an Ethernet transport network. The “TV” (audio and video program components) is the payload of that MPEG transport stream, which is the payload of the IP transport network. Thus, the video is more accurately considered a secondary payload to the Ethernet network, while the MPEG transport stream is the primary payload. That MPEG transport stream is both intricate and sensitive, and requires a much higher quality of service than the traditional traffic on a service provider network. Video is not voice or broadband data. For it to work, each packet must arrive on time and in a specific order.
Understanding the IPTV Headend

Delivery of IPTV requires turning the telecom service provider into a video network operator, not turning video into a telecom service. Delivering real-time video services entails implementing a significant amount of dedicated equipment and processes. Every network operator, regardless of the nature of their distribution network, has to have a location to house that equipment and process. The headend has come to be recognized within the digital video industry as the central video-processing facility, and in the service provider network all aspects of video belong to the broadcast IPTV headend. This is where programming is collected from content providers; processed, reformatted, or repackaged for the IPTV network; combined with middleware, Electronic Program Guide (EPG) or other interactive services; encrypted; and ultimately encapsulated into IP for deployment into the service provider distribution network.

Figure 6.1   The IPTV Head-end takes in video from content providers, processes and repackages it for use in the IPTV system, then sends the video services into the Core Network.
Figure 6.1 illustrates the IPTV headend. Broadcast video services are acquired in the ingress stage shown here on the left. Typically this is via satellite downlink, terrestrial antenna, or over a fiber network. The video programming as it enters the IPTV headend is basically the same content that any other network operator receives from content providers, such as, national feeds from major networks and regional content from off-air broadcasters and local affiliates. Those services go through significant processing within the headend, and then pass to the egress stage, where they are encapsulated in IP packets and transmitted to the service provider core network.

**Ingress and Video Compression**

Ingress to the IPTV headend is the point of demarcation between content provider and network operator. Once that content crosses into the headend, the operator is responsible for any degradation of quality. This introduces a challenge because the operator cannot make use of that content without making significant changes to it. The first is video compression. Digital television is made possible through compressing the video to a manageable size. For traditional video networks—cable, satellite and terrestrial operators—MPEG-2 video compression was sufficient. For IPTV deploying over DSL, the bitrates that can be supported are limited and additional compression is needed. In most cases, the deployment of triple-play video requires conversion of incoming video content from MPEG-2 compressed video to an MPEG-4 AVC variant.

Both MPEG-2 and MPEG-4 compression utilize the same fundamental principles. Analog video has been digitized, and then a range of complex algorithms involving aspects including motion estimation, inter-frame prediction, identification of spatial and temporal redundancies, etc. are applied to reduce the bandwidth required to transport the video with minimal reduction in quality. The end products of the compression process are audio or video
elementary streams (ES). Each digital program is composed of a pair of audio and video ESs. That ES pair must then be delivered together to the set top box for decoding and presentation. By definition, compression suggests an end product that is smaller than the original. To accomplish this reduction, MPEG compression permanently removes data from the original file to get the smaller output version. Much of the complex algorithms used in the compression exist so the decoder can figure out how to rebuild all the data that has been removed, and all of them require an ability to make predictions of what the missing data is by looking at video frames before and after the frame in question. For this to work, the decoder must decode certain future video frames in advance, and it must retain certain older frames to use as reference in rebuilding intermediate video frames. This is accomplished by the encoder generating two sets of time stamps for the video, one for the presentation sequence (PTS) and another for the decode sequence (DTS).

**Presentation Order** *(coming out of encoder, and as seen on TV)*

```
I1 B1 B2 P1 B3 B4 P2 B5 B6 P3 B7 B8
```

**Decode Order** *(going into set-top box buffer)*

```
I1 P1 B1 B2 P2 B3 B4 P3 B5 B6 ...
```

Figure 6.2 MPEG compression removes data from the video frames, requiring the decoder to make predictions between reference I and P frames to rebuild the intermediary P and B frames. This requires a different decode sequence then the presentation sequence.
Reference Frames

Another critical element in this decode process is the use of a true reference frame. The predictions the decoder must make will be limited by the accuracy of its reference points. MPEG compression creates three types of video frames, based on the level of compression applied to each frame.

- I-Frames (Intra-coded Frame): These undergo minimal compression, and the decode of an I-frame has no dependence on any other frame. This is a true frame, a complete reference for the decoder to use.

- P-Frames (Predicted Frames): These undergo moderate compression, and the decoder must look backward to a previous I or P frame to apply the differentials and rebuild these.

- B-Frames (Bi-directionally predicted Frames): These undergo extensive compression, and the decoder rebuilds them in a similar fashion from both previous and future I and P frames.

The encoder will divide the entire video file into groups of these frames, each called a Group of Pictures (GOP). Each GOP starts with a new full reference I-Frame. This division of frames helps mitigate the propagation of errors as they are limited to one GOP. With the next GOP comes a new I-frame, and the picture sequence begins again. Traditionally, the GOP would be 12-15 frames, but the GOP size is variable; there can be different settings for more or less frames in the GOP. Adjusting the GOP settings directly affects the presentation picture quality by adjusting the level of compression, and indirectly affects the user quality of experience. The GOP size affects the subscriber’s set top box workload, impacting factors including latency and channel change time. For example, the set top box must wait for an I-Frame to start a program presentation. This wait time is a significant factor in channel change time. Therefore, the GOP settings will impact that aspect of the user experience.
Preparing Video for DSL Delivery

While the triple-play service provider does not perform the original compression (this is carried out by the content provider), the provider must achieve a more efficient bit rate for those programs to enable delivery in a DSL network. This is accomplished either through transrating or by converting the video to an MPEG-4 AVC format. The conversion can be carried out either by re-encoding (decoding the video back to an uncompressed format, and then re-encoding it), or transcoding (converting a compressed MPEG-2 video directly into a compressed MPEG-4 video).

In either case, the operator introduces some risk to the quality of the video and, as a result, must have the ability to test the new product to ensure the quality has not been compromised. This is an important step, as the content providers maintain stringent quality standards for their programming. Content providers will not allow an operator to deliver their services if they cannot guarantee the end product suffers no degradation in quality.

However, it is more of a qualification exercise than operational test practice. Compression is effectively an isolated, “closed-system” process with repeatable outcomes for the same input. The operator can play known good content into the compression stage and then use an Elementary Stream Analyzer to validate the output picture quality of the new MPEG-4 video file. Figure 6.3 depicts an analysis screen from an Elementary Stream Analyzer, segmenting the video frame into macro blocks and showing the quantization (level of compression) and motion vectors per macro block.
An Elementary Stream Analyzer can be used to test the compressed video frame by frame to identify violations of encoding specifications and provide detailed information about the frames, macro blocks, motion vectors, and other compression parameters.

The ES analyzer will examine the video at the micro level of frames, slices, and macro blocks. Where compression was too heavy, resulting picture quality defects will be evident. The ES analyzer thus allows the operator to thoroughly test the various encoding and transcoding solutions and settings available and determine the optimum equipment model and configuration for their network. A vigorous qualification of this compression stage is important, as the hand-off from the encoder or transcoder to the MPEG transport stage is another demarcation point. Real-time monitoring of the compression product (the content itself) is neither practical nor feasible, so the operator would like to have a high degree of confidence that any problems with the programming originated at some point after the compression stage.
MPEG Technology

Compressed content must be transported to a receiver for presentation. For network operators, this stage of service delivery can present many troubleshooting challenges. Compression techniques developed in the mid 1980s, introduced a new era of digital storage in the format of the audio CD. This was essentially an audio ES on disc. The logical next step was to apply this technology to video, but doing so required a way to synchronize the audio and video ES pair at the decoder. With the help of the Moving Pictures Experts Group (MPEG), a method for capturing the encoder’s System Time Clock (STC) was defined and the audio and video ES pair could be transferred to a decoder. The decoder could lock its clock to the STC reference and enable the audio and video to be presented in tight synchronization. The result was better than expected. There was no loss in presentation quality with significantly lower storage requirements. Visionaries immediately began to contemplate ways to move this technology to the next frontier—broadcast television where the increased bandwidth savings had incredible potential.

Applying this technology to broadcast television introduced challenges, one of which was delivering multiple programs. This required a means of uniquely identifying each program and each of its components, and constantly sending an accurate index of this information to the decoder so that it could find the desired program’s components and decode them. Equally important, and significantly more challenging, it meant transferring the encoders STC reference across time and space, so that the programs could be decoded with tight synchronization anytime, anywhere.

After considerable effort the MPEG standards body delivered the solution in the MPEG-2 standards, a series of specifications\(^1\) that included enhanced compression technologies optimized for

transport and the MPEG-2 transport standard for delivering compressed video and associated audio in real-time. This packet based MPEG-2 transport standard was revolutionary. As IP transport was gaining momentum at this time, IP was a proven and widely accepted means of transporting packetized data, yet it had no means for providing needed synchronization. The MPEG-2 transport standard accomplished this with such efficiency that it remains the global medium for delivery of digital television. Even today, every set top box is an MPEG-2 receiver, regardless of the type of network or the video codec used to decode the video ES itself. Every piece of broadcast equipment that modifies the program stream in any way today is an MPEG-2 device, creating a new MPEG-2 transport stream at its output so as to preserve the unique identification of program elements and the integrity of the Program Clock Reference (PCR), the mechanism used to transfer the encoder’s STC.

MPEG-2 Transport
MPEG-2 transport begins by taking the compressed ES and packetizing it at one video frame per packet, as illustrated in Figure 6.4. The packet payload is the video frame and the header holds the Presentation and Decode Time Stamps (PTS and DTS). The audio ES is similarly packetized, although no DTS is used in the header. Each of these packets then is divided into uniform MPEG-2 transport packets of 188 bytes. The payload can be up to 184 bytes, the header at least 4 bytes. Within the header is a number called the Packet Identifier (PID). PIDs are used to identify the type of payload in each packet, allowing any downstream device to rapidly sort the incoming packets. The process of combining these 188 byte packets to create an MPEG-2 transport stream is performed in an MPEG-2 multiplexer.
Figure 6.4 The MPEG-2 System is comprised of three layers: the Elementary Stream, the Packetized Elementary Stream consisting of large, variable length packets, and the MPEG-2 Transport Stream, composed of uniform 188 byte packets.

The transport stream can hold one or more programs, either as an MPEG-2 Single Program Transport Stream (SPTS) or MPEG-2 Multi-Program Transport Stream (MPTS). The main differences are the number of PIDs in the stream, the transport stream bit rate, and the volume of metadata needed to index the program(s) in the transport stream. In all cases, the bulk of the MPEG-2 transport stream are video PIDs, followed by audio PIDs, and possibly Null PIDs used to fill unused space and achieve a specified transport stream bit rate. But the most important data in the transport stream is the index the decoder must use to identify the PIDs in that stream. Without it the video and audio components will be unrecognizable. This information is contained in the Program Association Table (PAT) and Program Map Table (PMT). The PAT is always located on PID 0x0000 and lists the programs being carried in that MPEG-2 transport stream and the PID number of the PMT for each program. The PMT, in turn, lists the PID numbers for the audio and video components as well as any additional program components in the stream, such as secondary audio, metadata, etc. Assigning PIDs and the subsequent logging in the PAT and PMT
tables is a critical task performed in an MPEG-2 multiplexer. If the PIDs are incorrectly identified in the tables, the program components are lost to the set top box. Because the PIDs for each program component in the stream must be unique, the PIDs must be remapped anytime one or more programs are combined into a new transport stream.

**MPEG-2 Synchronization**

The second half of the MPEG-2 transport process is the synchronization of program elements. At the PES layer, the headers contain the PTSs for the audio components and both PTS and DTSs for the video components. These time stamps are created against the encoder's system time clock (STC), and have meaning only if that STC exists at the decoder. This is accomplished by the periodic stamping of a program clock reference (PCR) in the MPEG-2 transport packets. The decoder will use the PCR to lock its own 27 MHz clock to the encoder's STC. Additionally, it will use a phase lock loop to track the PCRs and keep its own STC in sync. That 27 MHz clock is very sensitive to small inaccuracies, so the jitter of PCR arrivals and the accuracy of the PCR are of paramount importance to the set top box.

**Testing MPEG-2 Transport**

Digital video test is based heavily on MPEG-2 transport test as this is the end-to-end medium. The audio and video content has been created by an encoder in the compression stage, and whether that was done in-house or by a content-provider, it is a stand-alone stage and distinctly separate from transport. Good or bad, the compression stage hands the content to transport, and from the encoder throughout the network and right to the set top box, all digital video programming is running in an MPEG-2 transport stream. This is particularly true within the headend, where the video services have not yet been exposed to the risks inherent to Ethernet transport.
MPEG-2 Transport Errors
The most common errors that can occur within MPEG-2 transport fall into one of the following categories.

PID/PSI errors
An error in the assigning of PIDs and/or the cataloging of PID number and type to the PAT or PMT. Either of these problems will result in the set top box, or any downstream device for that matter, not being able to identify and sort the contents of that transport stream. Initially this has been done by the original encoder and multiplexer, but through the course of the content providers network, and subsequently the IPTV headend, the MPEG transport multiplex may have changed several times. Each additional multiplexer inserted within the video delivery system increases the probability of additional errors.

Continuity Counter Errors
Continuity errors are the MPEG equivalent to a dropped transport packet. In actuality, this does not have to be a dropped packet. Given that the decoder needs to receive packets in a specific order, any loss of continuity (a dropped packet, a packet out of sequence, or a repeated packet) is effectively a dropped packet to an MPEG receiver. The severity of this problem depends on the type of packet affected. The effect of a dropped video I-Frame packet would be worse than an audio packet, for instance. Before the video services have reach the IPTV headend, they have been subjected to risk of MPEG packet loss by means of Error Rate (BER and/or MER) in the content providers RF distribution network.

Program Clock Reference (PCR) impairments
The PCR is a complex and sensitive device. It is a counter, a time value, and a clock all in one. The spacing between consecutive PCRs in the MPEG transport stream must be consistent and within MPEG specifications, but that spacing can easily be impacted by
any addition or deletion of some element of the MPEG-2 transport stream. Even dropped packets would create some errors in the PCR spacing. Once the decoder has locked to a PCR, it expects to see each subsequent PCR at regular intervals. The delta between the expected and actual arrival can be significantly impacted by the network, and this jitter can cause many problems for a decoder. Likewise, inaccuracies in the PCR value itself will cause similar effects as jitter. The PCR is the most critical element for a high-quality presentation at the set top box. Many picture quality issues are related to some aspect of PCR performance, including basic errors such as pixelization as well as more complicated ones such as chroma/color problems.

In addition to these errors, several composition and timing/spacing errors can occur, as can application-specific errors. Because packet spacing and order, as well as rate and bandwidth, are so critical in MPEG, any unrecognized PID in the stream can cause trouble, as can PIDs that are present but with negligible bit rate. Likewise, the PSI table data must be accurate and it must also be systematically present in the transport so that any receiver can access it almost immediately after tuning to the MPEG transport stream.

**MPEG Testing Guidelines**

All of these factors combine to yield a cumulative effect—MPEG transport testing is an inescapably complex and broad task. Technicians must interpret and confirm protocols, identify packet flows and track the rates for each of them, decode header information, parse tables, read descriptors, check references and cross-references, replicate a system time clock and make countless iterative calculations. And, this covers testing the basics only. There are literally hundreds of errors or events that could transpire in just a few minutes, and each has a temporal relevance. When they happen and how long they persist is important to know when trying to troubleshoot a problem.
With regard to digital video, MPEG testing cannot be reduced to “go/no go” LEDs. Rather, MPEG relies on guidelines laid out by the Digital Video Broadcasting (DVB) Project. The DVB Project is a consortium of public and private television entities who came together to define an open standard for the delivery of MPEG-2 based digital television. Predominantly European in origin, it has grown to be a global body and the DVB open standard is fairly global in adoption as well. Where the MPEG specifications for video and audio transport were focused on just the transport task, with an eye on being flexible and forward compatible, the DVB standards were thorough and provided copious definition of the methods for implementing television services over MPEG. In May of 1997, the DVB Project created a set of measurement guidelines for MPEG/DVB systems which focuses on the MPEG-2 transport stream and includes details on DVB specific tests. One of those was this set of measurement guidelines, as recorded in ETR 290 and later expanded in TR 101-290.

**TR 101-290**

TR 101-290 takes a series of tests for the MPEG-2 transport stream, MPEG-2 PSI, and DVB SI (Service Information: these are in-band tables containing DVB Electronic Program Guide data) and groups them into three categories according to their importance. The result is a simple matrix that reflects the instantaneous “health” of the MPEG-2 transport stream. Any comprehensive MPEG test set will provide some form of TR 101-290 evaluation. Figure 6.5 illustrates the TR 101-290 view of the JDSU DTS-330 MPEG Analyzer.
Figure 6.5 TR 101-290 provides a simple, yet comprehensive assessment of the MPEG-2 transport stream’s instantaneous health. In this analyzer screenshot, each category has an LED to indicate current status and a counter for historical performance in that category.

Each table represents the parameters for health at a specific priority of MPEG performance. Left to right they are Priority 1, 2, 3, and a fourth column labeled Miscellaneous.

- Priority 1 parameters are critical for the MPEG Transport Stream to be capable of being decoded. Note that sync, continuity count, and PAT/PMT tests are represented in Priority 1.
- Priority 2 parameters are recommended for continuous monitoring. In application most of these parameters directly affect the quality of Program presentation. Note that all the PCR tests are represented in this group.
- Priority 3 parameters are defined as optional. These are tests specific to the DVB SI data and would be irrelevant to any system not deploying a DVB Electronic Program Guide (EPG) in-band to the transport stream.
- The Miscellaneous column is not part of TR 101-290. It is an extension unique to the JDSU DTS-330, providing an additional tier of performance metrics that are pertinent to the concept of an MPEG health check.
Within each table, each parameter has an LED to reflect the current status (red for *in error*, green for *no error*) and, in this solution, a counter to indicate how many errors have been observed in this test session.

Testing against a TR 101-290 template is an excellent way to get a rapid yet comprehensive assessment of the health of an MPEG transport stream, as each parameter in these tables is the cumulative status of one or more tests. For instance, the PAT category will reflect an error status if the time between PAT arrivals has exceeded 0.5 seconds, but it will also reflect an error status if something other than the PAT table is on PID 0x0000, as that is reserved for the PAT only, or if the PID 0x0000 has been scrambled, as that is invalid and would prevent the decoder from being able to use the PAT.

Ultimately, TR 101-290 provides a validation tool, which is the first stage of MPEG test. For purposes of rapid QA, turn-up and provisioning of equipment, validation allows the technician to confirm that the MPEG system is compliant and be reasonably confident that there are no obvious MPEG failures at that point in the network.

**MPEG Troubleshooting**

The second stage of MPEG testing is troubleshooting. MPEG transport is highly dynamic and very volatile. The errors that occur are often transient. Diagnosing problems from the symptoms alone is difficult as there is rarely a one-to-one relationship. Often there are multiple problems that will trigger the same symptom, and there may be multiple potential root causes that could create one problem. For instance, a loss of audio at the set top box could mean the audio component has been lost and is not present at the set top box. But, it could also be caused when the audio PID has not correctly been identified in the PMT table, or when the PTS values for the audio and video have been corrupted and are too
far apart to synchronize. Incorrect descriptor tags for secondary audio when the set top box tries to switch could also be problematic. The error-causing scenarios are infinite. Turning up a series of equipment with known good content and validating reliable operation is a basic task; however, it is quite complex to go live with real-time content from outside sources, and ramp up to a high volume of services where real-time rate-shaping and bandwidth management is being performed.

**MPEG Transport Test and Visibility**

TR 101-290 is an excellent tool, ideal for monitoring and validation. But it lacks the functionality to support in-depth troubleshooting. The use of error counters, as illustrated in Figure 6.5, helps extend its usefulness as a troubleshooting tool as it provides visibility into historical performance. For those sporadic, short-lived errors in the MPEG transport that have a drastic effect on a decoder, it can be very helpful to see the historical TR 101-290 performance. In this instance, however, TR 101-290 provides only an idea of where to look. For instance, TR 101-290 alarms in the PCR fields suggest looking more closely at PCR performance. But, those problems could be caused by a number of events directly and indirectly connected to the PCR itself. The PCR values may have been entered or modified incorrectly either at the encoder or at some remux later in the broadcast chain; there may be excessive network jitter; or packet loss may have disrupted the PCR spacing in the stream.

Troubleshooting the MPEG stream with active analysis is required to identify the source of these problems. The same is true for structural items, including PID presence and identification in the PAT and PMT tables. TR 101-290 will tell us if a PID referenced in the PAT/PMT is not present in the stream, but without visibility into the stream itself, it is impossible to know if that element is truly missing or whether it is there and stamped with a different PID number.
Figure 6.6 The PAT and PMT tables are how the set top box identifies the programs and the individual program elements. An analyzer that can decode the tables allows the user to verify that the program elements have properly been identified by the encoder or any device that has created a new multiplex of the MPEG programs.

Figure 6.6 illustrates an analyzer decode of the PAT and PMT tables. With this protocol decode, a technician can confirm the navigational information the set top box will use for this transport stream. Descriptors (small tags of additional detail information) for each program are also displayed here. Descriptors communicate important program details to the receiver, such as audio type, language, closed captioning information, copy control presence, ratings, and more.

Figure 6.7 illustrates analysis of the PIDs present in a transport stream. Here the technician can verify the components in the stream and get details such as PID type and PID rate. Used in conjunction with the PAT and PMT decodes, a technician can cross-check the MPEG PSI to the PIDs in the stream, confirming if the PIDs a set top box will be expecting to see in the stream, are actually present.
Figure 6.7 Visibility of all PIDs in the MPEG transport stream gives the user an ability to validate that the right data is on the right PID, as well as a means of identifying individual rates for audio, video and Null PIDs.

**Beyond TR-290**

Further, there are issues that exist beneath the surface—problems that technically are not *illegal* to an MPEG transport stream, and therefore will not trigger an alarm in TR 101-290. A common assumption is that digital programs contain only audio and video pairs. While this is the most common scenario, MPEG transport could also carry digital radio services, or datacasts, among others. In those cases, it is valid for a program to have a PCR PID mapped to an audio PID, or programs with an audio element and no video. If by chance a traditional program (video and audio) should lose the video PID and the PMT reference to video, there will be no alarming against TR 101-290. This is also true when a PCR has been inserted on both the video and audio components. There will, however, be noticeable presentation issues at the set top box. These problems all require comprehensive visibility into the MPEG transport stream to check the structure of the stream and confirm the presence and even the rate of each element. The following examples illustrate the steps of MPEG troubleshooting.
Troubleshooting Scenarios for Audio and Video Problems

Missing Audio
If a customer reports no audio at the set top box, to test this in the MPEG layer, the technician must connect to the network and view the transport stream carrying the afflicted program. It is recommended to first verify that TR 101-290 performance is within parameters. For this specific issue, the technician would check the PTS parameter in Priority 2, and Referred PID parameter in Priority 1. The PAT and PMT parameters in Priority 1 are the tables that tell the decoder where the audio can be found. Since the video is working in this scenario, the technician should assume those tables are valid and registering at the decoder. The technician should also look for PCR errors—not for specific audio relevance but generically because this is a key to the program presentation. Several scenarios may present to the technician:

- **Scenario A: PSI/PID references are incorrect.** Upon checking TR 101-290, the technician may observe that there are no PAT errors or PMT errors, but a red indicator in the Referred PID parameter of the Priority 1 table is not present. The MPEG analyzer provides event logs for the test results of each parameter. Therefore, the technician is able to confirm that the Referred PID error is for PID 0x0567. The technician can further confirm this by checking the PID view of the analyzer which lists every PID in the stream. The analyzer also provides a decode of the PAT and PMT tables, so the technician is able to see the same list of Programs, PMT, Audio, and Video PIDs that the set top box will receive. Checking the PMT for the suspect program indicates that the video should be on PID 0x566, and the audio on PID 0x0567. The technician now knows that the issue is a missing audio PID.

Before the technician assumes the worst and begins to search for the location where audio PID was lost, the technician should check the TR 101-290 list again to determine whether there are rogue, unreferenced PIDs present in the stream. If the technician determines that events are being reported for an Unreferenced PID on PID 0x0568, further investigation of the PAT and PMT will indicate that none of the programs are using PID 0x0568. Because the technician has determined that the rate of this PID matches an audio PID rate, it is safe to assume that an MPEG multiplexer has either mapped the audio to the wrong PID, or entered the wrong audio PID in the PMT table for this program.
Scenario B: PTS errors. Upon checking TR 101-290, the technician finds that no errors are being reported in any of the Priority 1 or 2 parameters. The stream is compliant. This would suggest that the issue is either occurring downstream of the technician’s test point or is content specific, in other words, native to the audio. If the technician’s charter is to maintain the headend, the technician should not be concerned with events happening downstream. To be thorough, the technician should check the PAT and PMT information and verify that there is an audio PID in the transport for this program when it leaves the headend. When this is accomplished, the technician may look more deeply into the PTS for the audio. If the technician immediately notices that there is a significant delta between the PTS value for the video and the PTS value for the audio, the problem will be identified. These are the time codes for the presentation of these components, and they should be very close in value. With the problem identified, the technician’s task becomes determining the origin of the problem.

PTS values are created by the encoder and stored in the PES header, so the technician should test at the input to the transcoder to determine whether the PTS delta exists there. If it does not, this indicates a problem originating in the transcoder. If it does, the next step is to test at the ingress stage. Testing directly off the antenna or satellite is ideal because this enables the technician to eliminate the IPTV headend from suspicion altogether. Assuming that the MPEG analyzer has the appropriate RF interfaces, the technician must tune to the program coming off air. If the PTS delta exists there, the technician can ascertain that the source of the problem is the content provider. The technician should record the stream, generate text reports, make screen captures, and then contact the content provider immediately with the evidence of this issue. Optimally, the technician’s analyzer will provide remote access capability (VNC, Remote Desktop, etc), to connect immediately to the Internet and transmit evidence of the problem directly to the content provider.
Stat Mux and Missing Video
To examine video troubleshooting, it is helpful to begin with a similar symptom—the video is not present at the set top box. Again, the technician must check the TR 101-290 performance of the MPEG transport stream carrying the suspect program. If no alarms are noted, the next step is to use the analyzer to check the PAT and PMT and identify which PID the video should be on. The technician should check the PID list, first to confirm that the Video PID is present and, second, to confirm the data rate on the PID is valid.

The technician may find that the video PID is indeed present, but the data rate seems low. When compared to the data rate of other programs with no presentation issues, it is indeed much lower than the others. This indicates that at some point in the broadcast chain, the video bit rate has been constricted. By performing the same testing at each step upstream, the technician systematically can clear each piece of equipment from suspicion until the trouble is located.

Figure 6.8 This analyzer screenshot displays the volume of Video PIDs (blue and orange) and the NULL PID in a transport stream. The bit rate of the Video PIDs fluctuates based on the volume of video data in the frames being transported at that moment. The NULL PID (0x01FF) fills in the additional bit rate needed to hit the transport stream’s specified overall bit rate.
This problem is increasingly more likely with the proliferation of bandwidth maximizing technologies. Rate shaping, transrating, and stat-muxing are processes designed to maximize the use of bandwidth. As the video element is always the largest bandwidth consumer in the MPEG stream, it often is the focus of bandwidth optimization. In fact, in a Constant Bit Rate (CBR) MPEG transport stream, most of the stream consists of video PIDs, and the balance is filled with stuffing of null PIDs to accomplish the constant output bit rate. If a stat mux experiences an interruption in its process, rather than drop the video PID altogether, it will likely continue to push it out with some nominal rate, and the Null PID rate will jump to cover the gap. The resulting program presentation will either have degraded video or none at all. TR 101-290 has no provision for alarming on his scenario, but if the technician can monitor the PID flows it is possible to detect when the video PID rate and null PID rates change.

In an IPTV system, bit rate is at such a premium that Variable Bit Rate (VBR) transport streams may be used in lieu of CBR streams. The VBR stream has no Null PID, but the issue with Video bit rate being constrained too heavily persists. An analyzer that can track the rate of individual PIDs or Programs can provide valuable visibility to an IPTV headend engineer. The better analyzers will extend an ability to set rate thresholds, allowing for active rate monitoring and alarming when the rate falls too low.

**PCR Analysis**

As the primary synchronization device for the set top box, the PCR is critical to a quality program presentation. It essentially consists of a time code that has been stamped into the outgoing transport stream at an interval of 27 MHz. In the MPEG environment, the PCR must be accurate to within 500 nanoseconds. That threshold applies to the accuracy of the time code within each successive PCR, as well as the jitter on arrival of each PCR at the receiver. Thus, there is an element of accuracy and network jitter. There are multiple threats to the integrity of the PCR integrity as it travels through the MPEG and IP system. Sophisticated MPEG multiplexers such as stat muxes, transcoders and rate shapers make making significant changes to the MPEG transport streams. While the video programming is intended to look identical going in as coming out of these devices, the MPEG transport streams often look completely different. Yet, the PCR integrity must be preserved. Often these devices will author the PCR values in concert with re-spacing them at the output, and that presents significant risk to the PCR integrity.
There is no single method for measuring PCR performance, but to be accurate the test device must have a 27 MHz clock signal, just as a set top box does, so that it can lock to a PCR and track the jitter and accuracy of each new PCR arrival.

Figure 6.9 The MPEG Analyzer lists the PCR PIDs in the MPEG transport stream, and the real-time measurements for jitter, accuracy and drift.

Figure 6.9 depicts an MPEG analyzer’s tracking of PCR performance. In the graph, the PCR jitter for PID 0x0800 has been displayed both in real-time as well as in a histogram. Using these graphs, the headend technician can rapidly identify if the PCR jitter of this MPEG stream has exceeded 500 nanoseconds, which is the defined minimum requirement for an MPEG receiver such as a set top box. By being able to test both jitter and accuracy, the user can isolate PCR problems caused by network jitter or by the handling of the PCR by some piece of network equipment. It is important to note that PCR is heavily impacted when MPEG is encapsulated into IP packets and placed in Gigabit Ethernet transport. At that point, the MPEG streams are no longer in an MPEG environment, but testing the PCR still has validity. This is discussed in greater detail in the discussion regarding video over IP, but it is important to recognize that phenomena now as it directly relates to PCR measurements and it can be a confusing aspect of IPTV.
Interoperability and Segmentation

The preceding examples have illustrated how testing digital video in the headend primarily consists of testing MPEG transport streams. Access to the TR 101-290 status, PSI decodes, PCR measurements, and visibility to the PIDs and programs gives the technician a comprehensive toolbox for validating the integrity of the digital video while in transport, and troubleshooting the issues that have been observed. But even equipped with this toolset, the technician is limited if the ability to apply it at various points across the broadcast chain is not present. MPEG transport is a horizontal technology. From encoder to set top box, digital video programming runs in an MPEG-2 transport stream. When those programs enter the IPTV headend at ingress, they arrive in MPEG transport. Leaving at the egress, they are in MPEG transport. These programs can be monitored at the egress of the IPTV headend, proactively detecting MPEG problems. TR 101-290 is an excellent foundation for that monitoring, but this indicates only when the problems exist and what they are. It cannot identify where the problems originated. The best way to accomplish this is to use an MPEG test to segment the broadcast chain and test at each node where the MPEG programs have been modified or otherwise handled. Figure 6.10 illustrates a generic broadcast chain common to an IPTV system.

Figure 6.10 Test points for testing interoperability
Interoperability and segmentation Test Case

Testing interoperability of equipment and isolating the source of MPEG problems are common practices in digital video test. Two useful procedures are comparative analysis of the MPEG stream at multiple test points, and simultaneous analysis of multiple test points.

Digital video programs are created in an encoder and delivered via satellite to an IPTV Headend by the content provider. The video is MPEG-2, and the IPTV operator must convert this to H.264 for delivery over their DSL system. The provider demodulates the RF satellite distribution and sends the MPEG-2 over DVB-ASI into a transcoder. The output of the transcoder consists of MPEG transport streams, but the video has been converted from MPEG2 to H.264 (MPEG-4). In this example, the operator has observed occasional audio loss and tiling of the video on the presentation of these programs. When TR 101-290 performance of these programs at the egress of the IPTV headend is checked, consistent alarming for continuity counter errors is observed. It may be assumed that the presentation issues are due to data loss at the MPEG transport layer. The question to then be answered is from where are the continuity counter errors originating?

To isolate the source, the technician must analyze the MPEG transport layer at multiple test points back upstream (see Figure 6.10). With an MPEG analyzer that supports a satellite RF interface, the technician can test the MPEG streams from the dish before entering the IPTV headend. If the MPEG transport is suffering the same errors from the dish, the technician can assume the problems are due to the content provider’s network (encoder, multiplexer and RF distribution). If not, this may indicate that problem originates in the IPTV headend. Working back towards the egress, the technician can test at the input to the transcoder if the MPEG analyzer has a DVB-ASI interface. Then, the technician may test again at the output if the analyzer supports on Ethernet interface.
This comparative analysis is a very effective way to segment the broadcast chain and isolate problem sources to a specific piece of equipment or stage of the network. It requires the MPEG analyzer to have a broad range of interfaces to provide the test access across the broadcast chain. Additionally, if the analyzer supports simultaneous analysis, this task becomes even more efficient, as the technician can literally compare the same video programs at two separate points in the chain at the same time. For example, in Figure 6.10, if the MPEG analyzer supports simultaneous analysis, the headend technician may look simultaneously at the same program on the ASI monitor port from a satellite receiver and at the GigE output of a video IP switch.
Conflicts of Interpretation

Video is a very dynamic service, and it requires flexibility in the application of any rules. This begins at the encoder, where compression that works for one type of programming may be completely unacceptable for another. It continues at the transport level—the previously illustrated case of CBR vs. VBR is an example. This creates an environment where two vendors may have a conflict of interpretation, or implementation, of the MPEG transport. To continue the CBR vs. VBR example, it is perfectly valid for MPEG transport streams to use a Null PID – common in fact. It is equally valid for the vendors to not have a Null PID, perhaps less common but not “illegal.” If equipment vendor A outputs CBR streams with a Null PID, and equipment vendor B expects to receive streams without the Null PID, the operator encounters a conflict of interpretation. Neither vendor is technically incorrect, but the service cannot be delivered while the debate over adjustments needed to resolve the issue takes place.

Middleware is an area susceptible to this risk, as it is tempting here to tweak the MPEG transport or compression to enable value-added features. The tweaks may not be against the MPEG standards, but they may create conflicts with other video equipment, or even with the decode process in general. Comparative analysis is the most useful tool for identifying the issues at the heart of these conflicts of interpretation because nothing is technically wrong with the video. When the technician cannot test to determine what is wrong, comparing on either side of the conflict to identify differences is the best alternative approach.
MPEG over IP

Thus far, this chapter has addressed testing MPEG compression and MPEG transport, but has not addressed testing as it relates to IP transport. This is due to the fact that, ultimately, the video processing in an ITPV headend is very much the same as a digital cable headend or a satellite uplink. The encoding, content provider distribution, ingress, re-encoding and multiplexing are generic video functions, and exist independent of the distribution technology. As we near the egress edge of any headend, the MPEG transport is prepared for the distribution network. Digital cable will modulate MPEG transport to a RF QAM carrier. Satellite will modulate MPEG transport to an RF PSK carrier. Providers will encapsulate MPEG transport to an IP “carrier.”

This is where everything finally begins to look more like IP. AVC compression technology has reduced video bandwidths to the point where single video programs can be delivered at bit rates sustainable on a DSL network. By packaging each video program as an MPEG single program transport stream (SPTS), and leveraging the switching and routing capabilities of the IP network, the video delivery system can be molded into a point-to-point model that better fits the existing IP network model. Instead of sending many programs down a fat pipe and switching between channels locally at the set top box, a handful of specific programs are sent down a smaller pipe, and programs are switched remotely. Although outwardly this seems to be a straightforward process, there is a twist: the true enabler of digital television is the MPEG-2 transport stream. To successfully deliver IPTV, the provider must be able to deliver these MPEG2 transport streams intact over their IP network.
This is challenging for two reasons:

- MPEG-2 transport is a complex and vulnerable system which works well provided that all the MPEG transport packets arrive at the set top box in a specific order, and within a specific window of time. This is a challenge in and of itself, even when a relatively simple RF modulation is used for the distribution network.

- An IP network by nature is not designed to deliver a high QoS—even in a higher-layer transport. Rather, it is designed to deliver static payload in a relatively unrestricted timeframe. It is a best-effort delivery service. Higher-layer protocols and intelligent hosting/routing can improve performance, but to a certain extent these efforts are almost counter-intuitive to the nature of IP transport.

As a result of these factors, triple-play service delivery essentially requires the provider to maintain two transport networks: IP and MPEG. The former is a familiar task, and although there are additional test requirements with the advent of IPTV, they primarily are enhancements to existing IP test practices. Because MPEG-2 transport streams have a tight window for packet arrival, any MPEG packet loss incurred by dropped IP Packets or excessive spacing (from inter-frame delay or re-sequencing) in the IP transport network will present significant risk to the program quality. Buffering at a network node will allow for a certain window for retransmissions and for delayed frames to catch up, but buffering also contributes to latency. Once the program presentation has begun, the stream is constrained to the much smaller buffer of the set top box. Therefore, an operator can look to monitor the IP jitter and IP packet loss over an Ethernet transport segment to gauge the performance of that IP pipe specific to video services. This enables monitoring of the threat-level to the digital video programs from the IP transport over that leg of the IP network. More sophisticated analysis will allow monitoring these IP parameters on a per flow basis as opposed to the generic performance of the entire IP pipe.
Cross-Layer IP and MPEG Analysis

Initial IPTV test practices focused on this method. However, headend technicians found they were limited to responding to IP performance alarms. In this case, if there had been IP packet loss the operator would have been alerted. The operator would know that the video services could be at risk, but did not know the severity of the impact, if any, on the video services.

When the MPEG SPTS are encapsulated to IP, seven MPEG transport stream packets are packaged in one IP packet. This is the maximum that can fit without fragmenting the MPEG packets. Each MPEG packet is carrying data for some part of the MPEG SPTS. It could be video, audio, PSI table data, or something else entirely.

Figure 6.11  When MPEG transport is encapsulated in Ethernet, seven 188 byte MPEG-2 transport packets are packaged in one Ethernet frame. (The maximum without fragmentation.)

So while losing an IP packet means losing seven MPEG packets, it does not mean losing seven MPEG packets with video information. Losing an MPEG packet containing payload for a video I-Frame is significantly more damaging to the program presentation than losing one containing payload for the audio background channel. The former could cause very noticeable picture defects; the other could go completely unnoticed to even the most trained ear. For the operator to know how the MPEG
transport stream was affected by IP layer impairments, technicians must have test visibility to the MPEG transport protocol, and it must be real-time and simultaneous. Ideally this will include the ability to perform simultaneous cross-layer analysis of both IP and MPEG transport protocols to enable correlation of impairments at both layers. Figure 6.12 below illustrates such a simultaneous cross-layer analysis, as provided in the JDSU DTS-330 analyzer.

The IP flows in the Gigabit Ethernet pipe are listed at the top. Because this is a video over IP application, each flow in the list is an MPEG MPTS or MPEG SPTS. Each entry includes an indication of the Inter-Frame Delay (IP jitter) experienced on that flow, as well as several Priority 1 MPEG measurements including continuity counter, transport and PAT errors. The bottom window lists the PIDs of the highlighted MPEG transport stream, with current continuity counter errors and bit-rate listed per PID. With this analysis, the technician can immediately assess if there has been any negative impact to the MPEG transport streams during IP transport, and rapidly identify not only which MPEG transport streams suffered a packet loss from IP impairments, but also which element of those MPEG programs was impacted.
This simultaneous cross-layer analysis has great value in troubleshooting as well, especially if complete MPEG analysis is available in addition to this MPEG/IP monitoring display. For example, if video is suffering from tiling, the IPTV headend technician can check this SimulTrack view to confirm whether there have been any obvious IP or MPEG layer problems on that program. If none have been reported, the technician may pursue a deeper MPEG inspection of that program to confirm the status of the PCR. If none is detected there, the technician can record a section of the MPEG transport stream, extract the video elementary stream, and use an elementary stream analyzer to confirm whether the content has native problems. The method is the same as described earlier for the MPEG transport streams. It has simply been extended to the Ethernet layer by way of GigE test access to the MPEG analyzer. Ultimately the goals are the same: determine if the problem is due to transport (MPEG or IP) or content, identify the source (equipment or provider) and enact steps for resolution.

Managing Video Test in the IP Realm

It is important to note that MPEG test at the Gigabit Ethernet layer is different from MPEG test in an MPEG environment. The moment MPEG transport streams are put in a Gigabit Ethernet stream, they are effectively “broken.” The tight spacing and network jitter requirements of MPEG are violated instantly; the IP jitter alone is order of magnitudes greater than PCR jitter. However, different does not mean invalid. The MPEG protocol remains intact. Therefore, tests of PSI and descriptors, PIDs, and continuity counters all remain valid. Even PCR measurements have validity. In fact, they continue to be very telling to the health of the MPEG transport streams. PCR accuracy and spacing are non-temporal measurements, meaning the rate of PCR arrival is irrelevant. PCR jitter becomes noise to the IP jitter. But, if this is tracked per flow in the Ethernet pipe, it is indicative of the level of buffering required...
by the edge equipment that must take the video over Gigabit Ethernet in and then reconstitute and output the MPEG-2 transport streams with proper spacing and sequence. By analyzing the MPEG streams from a Gigabit Ethernet test point at the egress of the headend, the technician has visibility to the final video product, allowing problem identification not only from the IP transport, but from multiplexing or encoding at some point upstream. Thus, MPEG-2 transport analysis remains an integral element of video test in the IPTV headend, even after the MPEG has been encapsulated to IP packets.

Deploying and maintaining IPTV service has proven to be more challenging than many anticipated. But, with the proper tools, service providers can be prepared to deliver this part of the triple play reliably. An MPEG transport analyzer that provides real-time Ethernet test access, ASI or RF test access, and an elementary stream analyzer, will enable the IPTV headend engineer to have visibility into the video across the entire facility. Additionally, it will provide the core tools necessary to troubleshoot dynamic video problems at egress, ingress, and the content layer. Nothing downstream of the headend will improve the quality of those video services. However, if the technician has the capability to validate that video services are good upon leaving the headend, it is reasonable to conclude that video problems experienced at the STB most likely emanate from distribution network or home wiring issues. This level of sectionalization will enable the technician to pass on the trouble resolution quickly for appropriate dispatch.
Video service quality is ultimately determined by the end user or subscriber. The quality of experience (QoE) is a subjective concept whose components are nearly impossible to measure in a practical, operational manner. Yet, a service provider can make objective measurements on a set of parameters that can be used to judge the performance of the network. A model for mapping objectively measurable metrics to QoE is the basis for good installation and troubleshooting procedures.

Mapping of objective measures of quality for video services—video quality of service (QoS) parameters—cannot be made in a one-to-one, direct correlation manner, nor can all subjective issues be measured directly. This is true especially of certain video artifacts which may be present in the video payload. In order to help with this correlation and add structure to measurement approaches, quality in this context can be organized into a logical model with four parts:

- **Content Quality**: the actual video and audio payload
- **Video Stream Quality**: the video transport stream packet flows
- **Transport Quality**: the IP packet flows
- **Transactional Quality**: the interaction between the user and the service

### Components of the Video Quality Experience

The video quality experience includes transactional items and program content specific items. Transaction items include the availability of program material, responsiveness of the electronic program guide (EPG), video on demand (VoD) billing transaction delay, channel change delay, and VoD feature responsiveness to pause/play commands. Program content items encompass picture quality and audio quality, including the absence of blurred images, edge distortion, pixelization, tiling, or frozen frames.
The following diagram depicts how objectively measurable metrics can be mapped to subjective QoE issues, and organizes these issues into four quality parts: content quality, video stream quality, transport quality, and transaction quality.

Figure 7.1 QoS and QoE Mapping for Video Services

**Content quality**
The quality of the content is the starting point. Decisions made in the video headend, where the content is acquired, determine the variations in quality. The initial quality of the video stream is established by decisions including which content sources are used, which compression algorithms are implemented, which encoders are employed, and by the source quality monitoring system. The data output of the encoders starts the video packet flow.

There is one critical source quality parameter that can be measured in MPEG-2 transport stream video flows at the customer premises and in the last-mile access network: the video transport packet error indicator count. Figure 7.2 shows a schematic diagram of the transport stream packet header. The components of the transport stream packet header are defined in ISO/IEC 13818.
The error indicator is a bit that is set by the encoders in a transmitted video packet when they detect corrupted source content. The presence of packets with this indication is strictly an issue related to content quality. It is not related to the performance of the distribution network. Monitoring of video encoder output streams in the headend can detect this condition and provide an early opportunity for problem resolution. Error indicator counts seen at the customer premises reveal a source quality problem.

![Figure 7.2 A schematic diagram of the transport stream packet header](image-url)

**Error indicator**

The error indicator is a bit that is set by the encoders in a transmitted video packet when they detect corrupted source content. The presence of packets with this indication is strictly an issue related to content quality. It is not related to the performance of the distribution network. Monitoring of video encoder output streams in the headend can detect this condition and provide an early opportunity for problem resolution. Error indicator counts seen at the customer premises reveal a source quality problem.

<table>
<thead>
<tr>
<th>Sync Byte</th>
<th>Error Indicator</th>
<th>Payload Unit Start Indicator</th>
<th>Transport Priority</th>
<th>PID</th>
<th>Transport Scrambling Control</th>
<th>Adaptation Field Control</th>
<th>Continuity Counter</th>
<th>Adaptation Field</th>
<th>Payload</th>
</tr>
</thead>
<tbody>
<tr>
<td>8</td>
<td>1</td>
<td>1</td>
<td>1</td>
<td>13</td>
<td>2</td>
<td>2</td>
<td>2</td>
<td>4</td>
<td></td>
</tr>
</tbody>
</table>

Fixed length of each packet = 188 bytes
Optional field = [Blank]

**PCR** = Program Clock Reference
**DTS** = Decoding Time Stamp
**PTS** = Presentation Time Stamp
Mean Opinion Score (MOS)
It is suggested that the concept of a mean opinion score (MOS), traditionally used in voice, be extended to video services. In this case, a V-MOS (video MOS) value can be determined for the video portion of a program and an audio MOS similar to MOS used for voice services can be determined for the audio portion of the program. This combination of an audio and video MOS value or score will enable an estimation of the subjective QoE for the video program video and audio content to be made from objectively measurable metrics.

Video Stream Quality

Continuity Error
A continuity error indicates a problem with the MPEG-2 transport stream packets themselves defined as any one of the following conditions: a lost packet, an out of sequence packet, or a duplicate packet on a per packet identifier (PID) basis. The sequence number used to identify each packet has a small value range of 16. Thus, in cases where more than 16 packets are lost in a row, the sequence counter will not represent the true picture of the event. It is important to analyze packet loss at the RTP level as well, where much larger sequence number range will detect the full loss event.

PCR jitter
Timing in the transport stream is based on the 27 MHz system time clock (STC) of the encoder. To ensure proper synchronization during the decoding process, the decoder’s clock must be locked to the encoder’s STC. In order to achieve this lock, the encoder inserts a 27 MHz time stamp into the transport stream for each program. This time stamp is referred to as the program clock reference (PCR). Video decoders use the timing signal to synchronize to the encoded data stream so that they can derive
two timing parameters embedded within each audio and video program. These timing parameters, the decode time stamp (DTS) and the presentation time stamp (PTS) are used in the decoding process to properly present the decoded video to the display unit (e.g. the television).

If excessive PCR jitter is present, the decoder cannot synchronize itself correctly to the data stream. The end result is visual impairments, such as pixelization, frame freezes, and loss of color. The amount of PCR jitter that is considered excessive is not a constant. It is determined by various parameters, including the input buffer sizes of the decoder and the design of the set top box software. However, in today’s typical packet video networks, PCR jitter should be less than 150 ms. Optimally, it should be less than 100 ms, depending on the specific decoder/STB design. Some designs may handle much more packet jitter based on extra large jitter buffers, so any specific metric must be established with respect to the particular network design. Note that some designs may handle much more packet jitter based on extra large jitter buffers.

Several factors can cause PCR jitter. The mostly likely causes include:

- Overall network packet jitter
- Transcoding problems in the encoder
- Local ad insertion issues

If packet jitter is not excessive when PCR jitter is present, the cause is specific to the particular program flow. In this example, an encoder may not be performing to specifications. If this is the case, PCR jitter will be constantly excessive. If PCR jitter is not constant, a momentary problem from inserting local programming may be the cause.
**PSI Data Error**

Video programming using MPEG-2 compression technology create an elementary stream for the audio and video portions of a program. And, there may be more than one audio flow included if different audio formats are included such as surround sound, 2-channel stereo, or different audio language selections. Each of these packet flows is assigned a PID number. Each program also contains additional data inserted into the flow called program specific information (PSI) which is organized into tables of data. Two critical items are the PMT and PAT table data items which tell the decoder how to identify and organize all of these individual flows.

The table data must be present in the streams at minimum intervals as required in a given network design. In addition, the table data can be scrambled by an encoder. Scrambling must be off. Analyzing streams for these items is critical to determining the video stream quality.

**Transport Stream Quality**

Transport quality refers to the operations of a video distribution network regarding delivery of the video material or programming as it relates to IP packets. Depending upon the network architecture, this may include analyzing packet flows within various protocols such as RTP or TCP over IP. The most critical metrics are packet loss and packet jitter. These two metrics provide a direct measure of network performance, but are difficult to map to subjective QoE quality.
An IP packet typically carries seven video packets. A lost IP packet event will normally create some kind of visible impairment. But, a subscriber’s subjective response to, for example, minor pixelization events caused by packet loss varies with the pattern or distribution of the events. For the same loss rate different distributions of loss will impact perceptions differently. Thus, adding a metric to measure the number of packets lost in an event—the period—and a metric for the number of times these events are separated by at least one packet in the event—the distance—will allow a better analysis of the spread of these loss events (RFC3357). This can help in looking at the “bursty” nature of the loss, which in turn can be helpful in determining the source of the loss: buffer over runs, forward error correction (FEC) depth setting threshold crossings, etc. Another packet analysis approach defined in RFC4445 called Media Delivery Index (MDI) provides a way to look at packet loss and jitter by calculating two metrics: delay factor (DF) and media loss rate (MLR). DF is a jitter analysis that indicates how long a data stream must be buffered, at its nominal rate, to prevent packet loss. MLR is the count of lost or out-of-sequence packets over time. However, mapping this kind of analysis to perceived quality, or QoE is best accomplished by a MOS algorithm.

**Transaction quality**

Transaction quality focuses on the availability and responsiveness of the video service to the expectations of the user. For example, the user may consider: “If I have subscribed to a set of broadcast TV channels, can I gain access to them? If I can gain access, how long does it take to change the channel? If I am watching a VoD movie program and I want to pause the program for some reason and then resume play, how responsive is the service to my commands?” These perceptions of responsiveness can be measured with an elapsed time approach.
The focus of QoS is upon metrics that can be instrumented and provide direct insight into variables in network performance. Consider the elapsed time from when the new channel is selected on the remote to when the new program first appears on the display. In this example, total elapsed time would be difficult to instrument and it would not provide direct measurement of network variables. The time to fill up the video buffers and decode the new channel program is a delay or time component fixed by the network design: STB decode buffer size, Group of Picture (GoP) settings in the video flow, and delivery bite rates for the program. Thus, a more revealing metric is one that measures the time from sending the message to obtain the new channel to receipt of the first video packet of the new channel. This metric measures the network performance directly, the variable parameter of Internet Group Management Protocol (IGMP) latency.

IGMP is the signaling protocol used to access broadcast video services that use a multicast network design to efficiently manage network bandwidth. IGMP enables each STB to obtain only the programming that the viewer is interested in watching, conserving bandwidth in the access network. In this implementation, a join message is sent from the STB to the network. The join message asks the network to send the requested program/channel to the STB by joining a multicast group carrying the desired broadcast channel. IGMP latency is the period between the time the join message is sent and the time the first video packet is received by the STB. Thus, IGMP latency is a measure of both service provisioning and the network’s response performance.

Figure 7.3 shows an example of an IGMP message flow when a broadcast program, Channel 2, is requested and then a channel change to another program, Channel 3, is requested.
The DSLAM, in this example, has access to the requested programs. It performs a snooping function, looking at the requested program material. If the requested program is directly available at the DSLAM, the join is performed. If the requested program is not available at the DSLAM, the IGMP messages continue upstream to a location, such as a video hub office, where the material is available. The join is performed at this location, and the program is routed to the DSLAM and on to the STB. The measurement of the variable network performance aspect of total channel change time is the critical parameter for measuring actual network performance.
In a similar manner, the delay associated with VoD services can be measured by establishing a latency for the Pause/Play commands. It is suggested that once a VoD program flow has been received, elapsed times can be measured, as outlined in the diagram below, establishing these two latencies providing additional insight into the transactional quality of the service.

Again, this objective measurement reveals the performance of the network directly. Attempting to measure the delay in obtaining the start of a VoD program would include a billing cycle or authentication of the subscriber for pre-paid services. The latency associated with these transactional quality items would vary more with software operations determined by network design than a network performance variable. Even more important, the subscriber’s perception would vary over a wide range degrading the value of such a metric. Further, it would be more difficult to instrument due to the variability of specific middleware software operation.

VoD services require that program content be stored and accessible from servers specifically designed for such service. A signaling protocol, typically Real Time Streaming Protocol (RTSP), is used in conjunction with software generically called middleware to enable the service. The middleware provides the billing and access control necessary for on-demand services. The following diagram illustrates this process.
Figure 7.4 Parameters for measurement of elapsed time for VoD request processing

From a test support viewpoint, it is the flow of video packets from the various server locations (sometimes called server farms) across the distribution network to the premises or subscriber that needs to be validated, not the operation of the middleware. Thus, a VoD test implementation that allows a simple RTSP signaling communication between a tester and a VoD server, bypassing a middleware negotiation, enables the VoD flows to be tested, but without the complexity of supporting the middleware. Since middleware is based on a universal standard, each vendor's version is different requiring the tester to support each version. Even new releases of code would potentially impact the tester. Thus, this simple approach avoids the cost and complexity of full middleware support and reduces costs while maintaining a robust test methodology.
Video Service Provisioning and Turn-up Testing

An Overview of on-site procedures
New service provisioning or turn-up testing can be accomplished most effectively at the customer premises where the accumulated effects of all the elements of the distribution network are seen. The installation test procedure should address test access, test methodology, and problem resolution.

Test access
A test access point close to or at the STB location reveals metrics that involve all segments of the network including the in-house distribution portion. However, due to network segment ownership and responsibilities, testing at the interface to the access network, xDSL, or customer side of a residential gateway, may also be important.

Test methodology
At the appropriate access point, it may be possible to have the tester work in a Terminate mode in which the unit can be configured as necessary to emulate the STB. While there are numerous configuration items which would be required to support a given network design, the goal is to achieve a data–layer-up status such that application streams (in this case video) can flow to the tester using IGMP or RTSP signaling as applicable.

In those cases where a tester cannot or is not allowed to emulate the STB, the tester could function in a Monitor mode. When a Monitor mode, such as “xDSL to Ethernet Through mode” or “Ethernet Bridge mode”, is used, the video flows can be successfully accessed. In this monitor methodology, the STB conducts the activities to bring up various program flows.
**Problem resolution**

Packet loss (continuity error) problems typically are seen on all channels/programs coming to the customer premises because they are not source- or content-related. If packet loss is present, analysis of the physical layer at the xDSL interface or Ethernet interface will aid in sectionalization. If no physical layer errors are present, then packet loss is most likely being caused by the distribution network, not by the access network. More than likely, congestion is the issue.

Looking further into the temporal component is important. Are packets being lost during known peak traffic times during the day? Are the packet loses coming in bursts with intervals with no loss? Or, are they random, single, or small packet loss events? Bursts of loss are symptomatic of buffer overflows related to heavy traffic. Random single or small events are more likely to be caused by noise hits on the access network that are impacting packet flows.

PCR jitter problems may be caused by content quality problems as outlined previously or overall network packet jitter. This can be determined by evaluating more than one channel/program at a time. If excessive PCR jitter is present on more than one channel, network jitter is most likely at fault. If excessive PCR jitter is present on only one channel, then a source problem, as described above, is typically the cause.

Error indicator analysis will help diagnose content problems. Since the error indicator can be set only by the encoder, it specifically indicates content-only problems. Usually this affects only one program or channel. However, if a multiple program feed in the headend is experiencing problems, multiple programs or channels can be affected. In this case, analyzing a channel from another source, or different feed, is useful.

IGMP latency measures the network’s performance. Typically, IGMP latency is similar for multiple channels. However, if network topology and network management place access to certain program material deeper in the network, then differences may be seen. Testing multiple channels/programs to exercise this network design is useful.
# Video Service Quality Testing Procedures

**Specific steps to ensure quality service delivery**
Following is a concise overview of the actions a technician should take to test video service quality thoroughly.

## Analysis

<table>
<thead>
<tr>
<th>Step</th>
<th>Actions</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Select a broadcast channel (IP address and port number) or select a VoD program (URL).</td>
</tr>
</tbody>
</table>
| 2    | Initiate the signaling sequence:  
|      | (IGMP – Broadcast)  
|      | (RTSP – VoD) |
| 3    | Receive the requested program flow.  
|      | Analyze video and transport stream packets for QoS metrics:  
|      | Packet loss, jitter; lost packet distance and period; PCR jitter; PSI error count;  
|      | error indicator count |
| 4    | Compare metrics to thresholds established for network under test.  
|      | *Note: if all pass this was a successful test. If any fail go to Step 5.* |
| 5    | Initiate a second channel or VoD program flow. |
| 6    | Measure IGMP latency or RTSP latency with Pause/Play commands. |
| 7    | Receive second video flow, and analyze video QoS metrics.  
|      | *Note: The two streams are being measured simultaneously* |
| 8    | While analyzing video quality metrics analyze physical layer metrics on interface under test. |
| 9    | Compare packet loss events to physical layer error events.  
|      | *Note: If no physical layer errors, packet loss is due to events northbound of network segment under test.* |
| 10   | Compare stream one metrics to stream two metrics.  
|      | *Note: If any QoS thresholds fail in one stream, but not in the other, content and stream specific issues are present.* |

## Problem Resolution

<table>
<thead>
<tr>
<th>Step</th>
<th>Actions</th>
</tr>
</thead>
<tbody>
<tr>
<td>11</td>
<td>Initiate a third stream.</td>
</tr>
</tbody>
</table>
| 12   | Compare QoS metrics to thresholds for network.  
|      | *Note: Thresholds are set to match a particular network.* |
| 13   | Compare QoS metrics for stream 3 to streams 1 and 2. Look for differences related to items pointing to content versus overall packet flow.  
|      | *Note: Packet flow problems will affect all streams. All Channels versus One Channel is the key concept to separate content issues from network issues.* |
Video Service Troubleshooting: A practical example

Using a JDSU HST-3000, four parameters are measured: PCR Jitter, Latency, Continuity Error, and Error Indicator Count. Latency is the IGMP latency, which defines the channel changing time not including the decoder buffer file and the decode times. Continuity error is the lost packet rate. Error Indicator count is the number of MPEG-2 TS packets received in which the bit was set indicating the encoder had a problem with the data it received. These four parameters summarize the video quality of the selected program flow (Figure 7.5).

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Current</th>
<th>Max</th>
<th>Score</th>
<th>Hist</th>
</tr>
</thead>
<tbody>
<tr>
<td>PCR Jitter</td>
<td>160mS</td>
<td>2488mS</td>
<td>Fail</td>
<td>Fail</td>
</tr>
<tr>
<td>Latency</td>
<td>0mS</td>
<td>728mS</td>
<td>Pass</td>
<td>Marg</td>
</tr>
<tr>
<td>Continuity Err</td>
<td>0.00%</td>
<td>0.00%</td>
<td>Pass</td>
<td>Pass</td>
</tr>
<tr>
<td>Error Ind.</td>
<td>0</td>
<td>NA</td>
<td>Pass</td>
<td>Pass</td>
</tr>
</tbody>
</table>

**Overall:** Fail

Figure 7.5 A measure of video quality of service

Thresholds are set for each specific network. In this example, PCR jitter is excessive, and IGMP latency was marginal at one point but is now OK.

Additional data can be obtained by analyzing the stream in more detail. Because each stream includes video, audio, and data (program table data) portions, packet statistics can be obtained for each stream (Figure 7.6). If PAT or PMT data is missing or not present at specific intervals, the decoder cannot function properly and picture quality can be impacted.

Figure 7.6 MPEG-TS stream PID (Packet Identifier) detail for the video, audio, and data portions of the stream.

The PAT and PMT data must be in any video flow. Error conditions exist if it is not present at intervals supported in a specific network design.
Figure 7.7 Detailed bandwidth analysis of the portions of a video program flow.

The bandwidth for the portions of a program flow is important in analyzing performance, especially in the access network. The total may be fixed, but the video portion may vary with the motion content.

Figure 7.8 Packet flow visibility for both the IP and RTP flows.

Much of today's video service uses Real Time Protocol (RTP) because its use of sequence numbers and time stamps makes it possible to deliver a higher quality packet flow. Analysis at this level is very revealing.
Standard troubleshooting steps are summarized through these graphic examples:

**Critical Diagnostic Fault Conditions:**

**Trouble call:**
- Video impairments
  - All channels?
  - One channel?
- Time of day specific
  - A pattern?
  - Random throughout the day?
- Temporal component
  - Bursty?
  - Randomly spaced?

The key question is: are the problems visible on more than one channel?

**IP Fault Resolution**

**Step 1**

**Video pixelizations**
- All channels? = yes
  - Lost packet events will be seen on more than one stream

**Lost packets**
- Analyze packet stats
  - RTP stats
  - Pkt loss analysis: RFC 3357 stats
- Sectionalization
  - If no errors are seen at the physical layer of both I/F’s, then packet loss is up-stream of the DSLAM
  - Continuity errors (lost, duplicate or out of sequence packets) are counting up

If problems are present on all channels, problems are most likely packet transport issue related. Period Distance (RFC 3357) analysis is critical. All error recovery mechanisms have limits as to maximum period and minimum distance (interval between loss periods).
Step 2

Video pixelizations
  – Physical layer problems? = yes
  – Check VDSL stats for errors
    • FEC errors: Un-correctable FEC errors
    • If yes, impulse noise most likely cause, if noise margin at least 6 dB.
    • Review Bits/Tone graphs for notches
      – Notches = Noise problems
  – Check Packet stats
    • RTP Stats
    • RFC 3357
    • Check physical stats
  – Video MOS
    • Summarize video picture quality with video MOS score

With more than one channel being affected, the issue is identifying the source of the packet problems. Evaluating physical layer metrics over an access link such as VDSL will help sectionalize the problem.

These numbers are very low as they should be for such a poor picture. V-MOS should be about 3.5.

This shows an example of how this correlation can be easily made. If there was a packet loss count over the ADSL interface under test, and no counts for ATM and ADSL layer items, then the packet loss would be upstream of the DSLAM or remote IP device terminating the ADSL link. The same analysis can be accomplished at the Ethernet interface.
Dropped packets indicate a problem over the Ethernet link. If the test access point was in the home using a home networking technology, this could indicate a problem with that segment of the network.

Video R-Factor scores the particular packet loss rates on a scale 1 to 120 (120 being the best performance). V-MOS scores, on a scale of 1 to 5 (5 being the best), evaluate the overall perceptual quality based on a complex algorithm that includes not only packet loss, but codec type, distribution of loss, etc. to estimate how a customer would score the picture quality. It provides an overall rating that better describes the user experience than other network packet performance metrics.

**Step 3**

Video pixelizations
- All channels? = no

Lost packets
- Analyze lost packet stats
  - Typically none seen when trouble is one channel

Analyze PCR jitter stats
- PCR jitter problems are typically source issues:
  - Headend trans-coding problems
  - Local Ad insertion problems

Analyze PSI error stats
- Typically a content issue

Error Indicator Count
- Analyze count
  - Corrupted content at encoder

Take the network design into account when evaluating QoS metrics. For example, networks that use an error recovery mechanism that enables lost packet data to be replaced by a re-transmission mechanism will be more tolerant of lost packet events. However, excessive loss causing high re-transmission rates can overload the network. Counts of re-transmission activity is an additional metric of particular importance in such a network.
The Business Case for Ensuring Video Quality

Determining whether poor video QoS performance is discernable on more than one channel will help sectionize problems, and simultaneous analysis of the key QoS parameters on multiple video flows will further refine this analysis. Ultimately, this leads to efficient and effective trouble resolution for content versus packet network performance. If no physical error or problem exists on the physical interface of the test point, yet network packet issues exist, that network segment can be eliminated as a possible cause of the problem. Further, the VMOS score concept can greatly help in supporting the QoE analysis specific to the subscriber. It provides objectivity to the normally subjective area of picture quality which helps improve process efficiency. Tools, processes, and technology are available to ensure service delivery quality, yet additional factors must be considered to ensure business success with video service delivery.

The Importance of Service Assurance

Delivery of video services to retail customers is specifically designed to increase average revenue per use (ARPU) in the face of existing strong competition from cable operators or other service providers. In today’s competitive landscape, the efficient turn-up of new service and delivery of support for existing service is critical. A service assurance program designed to ensure high levels of video quality from the headend or VoD to the subscriber including integrated field operations is essential. A service assurance partner that can provide software, probes, and portable field tools to deliver the necessary service provisioning, monitoring, trending, and fault resolution in a coordinated, efficient manner can play an important role in accomplishing these goals.
The emergence of Voice over IP (VoIP) as the new standard for business telephone systems has been confirmed by marketing data that shows revenue generated by IP-based PBX lines has surpassed conventional time domain multiplexing (TDM) lines in percentage of worldwide revenue. The penetration of VoIP in the residential market is deepening quickly. A key driver of this rapid adoption rate is triple-play service offerings from providers offering phone service at a low incremental cost. Potential customers are initially attracted to VoIP by its low price but tend to stay with the service when they grow accustomed to its advanced features such as Web-based call logs, click-to-dial and virtual phone numbers.

**Understanding VoIP Equipment and Protocols**

The customer premises equipment (CPE) for a VoIP subscriber can consist of either a traditional POTS phone that plugs into a terminal that converts signals to VoIP or a phone with native VoIP support that can connect directly to an IP network. The server supports the routing of calls across the network and provides management functions. The IP network provides connectivity between the terminals.
Data plane protocols
When a call has been set up, the speech is encapsulated in two data plane protocols, Real-time Transport Protocol (RTP) and User Datagram Protocol (UDP), and transmitted in an IP frame. The control plane or signaling portion of the VoIP protocol manages the traffic that is required to connect phone calls and maintain the overall network operation. There are three main control plane standards used for VoIP today, H.323, SIP, and MGCP.

RTP is the protocol that supports voice content traffic. Each RTP packet contains a small portion of the voice conversation. The size of the packet and the size of the voice sample inside the packet depend on the codec used. If an RTP packet is lost or dropped by the network it will not be retransmitted to avoid delay in the conversation due to the network or the phones requesting lost packets. The network should be designed to minimize lost packets.
**Control plane protocols**

Real Time Control Protocol (RTCP) allows the end points to communicate directly concerning the quality of the RTP packet stream. RTCP gives the end-points the ability to adjust the call in real time to increase its quality.

The control plane provides signaling protocols that register VoIP phones and connect phone calls among other functions. H.323 was the first widely adopted and deployed VoIP protocol suite. The standard was developed by the International Telecommunications Union’s Telecommunication Standardization Sector (ITU-T) for transmitting audio and video over the Internet. Over the last 10 years this protocol has gone through several revisions and additions to provide more features and increase scalability and stability. H.225 is the media control portion of the H.323 protocol suite. It establishes a logical channel for each call. H.225 represents the basic signaling messages including setup, alerting, connect, call proceeding, release complete, and facility messages.

![H.323 protocol stack](image-url)
The fastest growing control plane protocol is Session Initiation Protocol (SIP). SIP is published in RFC 2543. SIP messages can be broken up into two major categories—messages from clients to servers and messages from servers back to clients. Each message has a message header that identifies the message type, calling party, and called party. There are four basic message types: general headers, entity headers, request headers, and response headers.

The other common signaling choice for residential based VoIP services is Media Gateway Control Protocol (MGCP). MGCP is a combination of SGCP (Simple Gateway Control Protocol) and IPDC (IP Device Control Protocol). The main feature of the MGCP protocol suite is the capability of breaking a telephony gateway into two basic part—a call control and a media element.

**VoIP codecs**

Codecs are used to convert analog voice data to digital format for transmission across a packet network and transform it back to analog at the destination. Codecs can be implemented in software, hardware, or both. Voice data generally uses one of two types of codecs: waveform encoders and vocoders. Waveform encoders attempt to accurately reproduce the input signal at the expense of a higher bit rate. Vocoders represent the signal by varying several parameters and use lower bit rates. As a rule, the higher the bit rate used, the better the voice quality. Higher bit rate codecs, on the other hand, generate more network traffic and thereby reduce the overall call capacity of the network.

A number of different codec implementations are currently in use. The only internationally agreed-upon standard—and the only one required for compliance with the ITU-T’s H.323 standards—is G.711 Pulse Code Modulation (PCM). Two variations of the standard, called mu-law (for US use) and a-law (non-US use), are in widespread use. All versions of G.711 signals operate at 64 Kbps.
Since the 64 Kbps bandwidth requirements for G.711 exceed the capabilities of most POTS modems, G.723.1 is emerging as a low bandwidth standard. G.723.1, a combination of G.721 and G.723, operates at 6.3 Kbps and 5.3 Kbps respectively. The voice quality of G.723.1 is similar to PCM.

<table>
<thead>
<tr>
<th>Codec Type</th>
<th>Voice Samples per Frame (Default)</th>
<th>Voice Samples per Frame (Maximum)</th>
</tr>
</thead>
<tbody>
<tr>
<td>PCM/PCMA</td>
<td>2</td>
<td>10</td>
</tr>
<tr>
<td>G.723</td>
<td>1</td>
<td>32</td>
</tr>
<tr>
<td>G.726-32</td>
<td>2</td>
<td>20</td>
</tr>
<tr>
<td>G.729</td>
<td>2</td>
<td>64</td>
</tr>
<tr>
<td>G.728</td>
<td>4</td>
<td>64</td>
</tr>
</tbody>
</table>

Table 8.1 Voice samples per frame for VoIP codecs

**VoIP Service Quality**

**Objective measures for a subjective experience**
Packet networks were designed originally for bursty, non-real-time applications such as file transfers and email. This type of traffic tolerates delays and usually has built-in retransmit capabilities in the event of data loss or corruption. Retransmission enables error-free performance at the application level. However, VoIP service delivery places more stringent requirements on these networks that were structured to support data payloads. As a result, new class of service (CoS) mechanisms must be enabled in the packet network to achieve acceptable levels of performance to support toll quality VoIP.
Ultimately, as with video, VoIP service quality is determined subjectively by the end user. Yet, a service provider can make objective measurements to judge the performance and quality of experience (QoE) the network will deliver. A model for mapping objectively measurable metrics to important QoE metrics is the basis for good installation and troubleshooting procedures, which in turn play an important role in an overall service assurance program.

Figure 8.3 outlines the way in which objective metrics can be mapped to subjective QoE issues by organizing the issues into four parts.
Measuring VoIP service quality
Many ways exist to measure voice quality on a VoIP network. Active methods involve sending known voice samples across the network from one endpoint to a receiving endpoint. The receiving endpoint does a comparison analysis of the degraded sample to the original. Due to the complexity of the signal comparisons, intrusive testing algorithms are computationally intensive and not viable for real-time quality measurements.

Passive methods passively calculate voice quality without a reference voice sample. These methods are most commonly used to turn-up and test actual networks. The most popular passive method, mean opinion score (MOS), assigns a value to the overall quality of the delivered voice through a network. MOS scores range from 1 (bad) to 5 (excellent).

A true MOS score is determined by people listening to the same call and rating it from 1 to 5. Test devices can measure a MOS score through complicated algorithms that are based on the data from large groups of listeners rating calls. The test devices can then provide overall and per call MOS scoring to give a good view of network performance.

The E-model produces a single value called an R value. This value is derived from a variety of factors including delay and other network impairments. Originally the E-Model was intended to be used for network planning and design. R values range from 0 (extremely poor) to 100 (high quality). Any R value below 50 is unacceptable. TDM based phone calls have a maximum R value of 94.
Common VoIP service degradation factors
A number of effects can occur on the service delivery network that will negatively impact the perceived quality of service. Echo and packet loss, slip, and/or delay are a few of the most common that providers must identify and eliminate prior to releasing VoIP service to the customer.

Echo effect on voice quality
The most common cause of echo is an impedance mismatch between local and long distance circuitry. Telephone handsets use 4 wires to carry a call while the long distance network uses a single 2 wire pair. To interface between these wiring schemes a device called a hybrid is used. However, a hybrid causes signal reflection due to an impedance mismatch. Unfortunately there is no way to balance the long distance to local connection because looking from the local side, the long distance network’s loop impedance varies depending on the length of the local loop, number of phones, type of phones, presence of load coils, and other factors. A secondary, but potentially significant, source of echo is acoustic feedback from certain types of phones such as speaker phones.

The echo effect of can be addressed through suppression or cancellation. Suppression is the simpler method and simply turns off ‘receive’ while ‘transmit’ is active. With the cancellation method, the transmitted sound is saved. Then, if a weakened version of that sound appears on the receive side, it is deleted from the signal. Echo cancellation is usually implemented in a digital signal processor (DSP).
**Packet loss effect on voice quality**
Packet loss can occur for a variety of reasons inside a network. Under difficult transmission conditions, bit errors may exceed correctable levels and the packet is discarded. This is most often the case with wireless networks but is true to a lesser extent on any transmission media. A packet can also be misrouted or exceed its time-to-live quota due to network topology changes or network congestion.

Packet loss can occur in bursts due to transient network occurrences. Packet loss can also occur periodically, for example, due to consistent network congestion at certain times of the day. Depending on its magnitude and frequency, packet loss can be annoying and make conversation difficult. Periodic losses in excess of 5-10% of all voice packets transmitted degrade voice quality significantly.

**Packet slip/jitter effect on voice quality**
Packets transmitted at equal intervals from their source can arrive at the destination at irregular intervals. This is generally referred to as jitter although precise definitions for this term vary. Some define jitter as the difference between the longest and the shortest delay over some period of time. Others define jitter as the maximum delay difference between two consecutive packets in some period of time.

Excessive jitter makes speech choppy and difficult to understand. Jitter is calculated based on the inter-arrival time of successive packets. For high-quality voice, the average inter-arrival time at the receiver should be nearly equal to the inter-packet gaps at the transmitter and the standard deviation should be low. Jitter buffers (buffers that hold incoming packets for a specified amount of time) are used to counteract the effects of network fluctuations and create a smooth packet flow at the receiving end.
Packet delay effect on voice quality
Circuit switched networks guarantee delays on the PSTN will not be greater than a few tens of milliseconds. When delays increase beyond a threshold (250 milliseconds is often cited), people talk over each other as a result of the time it takes to realize another party is speaking. Packet networks typically have delays approaching the 250 millisecond threshold. Beyond constraints imposed by transmission speeds, most delay occurs in a packet network due to packetization, encoding, and processing overhead.

Network Assessment and Pre-qualification

Before VoIP service can be installed at the customer premises, the service provider must assess the wiring and networks at the customer premises to determine their suitability for transporting VoIP traffic. At the enterprise, testing and assessment confirms the ability of the LAN’s network elements—switches, routers, and cabling—to provide CoS treatment for delay-sensitive VoIP traffic. Additionally, it determines the load planning for the LAN; helps to establish a baseline service level agreement (SLA); and determine the need for network equipment or WAN interface upgrades. At the consumer customer premises, home wiring and home networks must be assessed.

This phase of testing is typically carried out with handheld equipment, software products, or passive test devices depending upon the scale of the VoIP installation. The devices look at all of the traffic on the LAN/WAN and home network to assess the quality of the network before VoIP service is added. The testing tools determine whether packets are being dropped or lost as they traverse the entire network.

It is important to note that the network can provide error-free data service while performing very poorly in terms of lost and dropped packets. High throughput can be used to make up for problems such as jitter, latency, and packet loss. However, a network that carries data perfectly may not provide satisfactory quality for voice traffic because VoIP is not capable of compensating for these problems.
Network Equipment Installation and Provisioning

In this phase of deployment, the service provider installs or upgrades and tests the integrated access devices (IADs), voice gateways, routers, firewalls, or home networking equipment that will be used or installed to support VoIP. The installer performs physical layer testing of WAN links and gateways. Ping tests are used to confirm that connectivity is established between network elements that have been installed or upgraded.

Typically, most of the testing at this phase is performed at Layer 2. If a virtual LAN (VLAN) or other IP level is to be installed, it is performed at this stage. This task should be carried out by an experienced Level 2 or Level 3 technician. This technician may do troubleshooting on a case-by-case basis to isolate problems and also may perform VoIP quality testing to ensure the network elements have been both installed correctly and configured with correct QoS settings. Handheld devices and bulk call generators are used to do these tests.

Figure 8.4 VoIP Service delivery network
Service Turn-up and Provisioning

Prior to completing service turn-up, field technicians must verify connectivity to signaling gateways, service provisioning, and call quality. Terminal adapters or VoIP phones are installed and applications are tested in this phase. Field technicians rely heavily on handheld testing devices to troubleshoot problems. Terminal adapters or IP phones are set up and plugged in to the LAN and their IP addresses are provisioned. Handheld test sets are normally used because they can assume unique aliases to mimic an end device on the network.

Handheld test sets can also be plugged in at any point in the network to help isolate problems. For instance, a handheld device can be used to determine if a specific end device’s alias has been provisioned correctly. It can help identify errors in provisioning network equipment during installation. It can also be used to verify the installation of specific equipment in the VoIP network.

Voice quality issues are resolved at this stage. The technician must place and receive calls through the network to ensure that the link is properly provisioned with the correct signaling protocol. Calls should be placed within the VoIP cloud and from the VoIP cloud to the PSTN. Local and long distance calls should be placed to multiple exchanges. Case-by-case trouble shooting is used to resolve any issues before service turn-over. By confirming that all of the possible calls can be placed, a technician can confidently connect the CPE knowing that any signaling issues will not be within the carrier’s cloud. Another important practice is to capture all test records gathered at this phase for baseline/SLA reference and for use during future trouble calls.
While checking the various call options, the technician will be able to monitor the quality of the RTP stream. Field technicians listen to calls using handheld units to determine voice quality. They also look at the MOS of each connection to help make an objective analysis of the voice quality. The two main values a technician will want to examine are the R value (derived from the E-model) and the MOS value. The technician can compare those values with the SLA defined by the carrier who owns the service.
Troubleshooting and Maintenance

Technicians will be called upon to perform two types of troubleshooting: catastrophic failure and intermittent issues. When a circuit is non-operational, the troubleshooting task is very similar to turn-up. The technician will be able to terminate the circuit back into a test device and begin the process of checking connectivity to the local elements through pings, trace routes, and call placement. The problem can be sectionalized to the CPE or carrier, and remedied by the appropriate party.

Intermittent issues can present challenges. If the customer’s phone system works reliably most of the time, the customer may not allow the circuit to be taken out of service for testing. In this instance, the technician must monitor the circuit to determine which calls are experiencing trouble, when the trouble occurs, and note environmental factors during the trouble occurrences. Environment, in this case, refers to the traffic riding on the network at the time of the failures. For example, other applications may use router processing time or bandwidth that can cause calls to drop or voice quality to transmit un-intelligibly to the receive end. The only means to determine whether CPE traffic is the cause is to monitor the whole circuit while the problem is occurring.

Network management and passive monitoring tools work well for data networks but lack the detail needed to identify many problems in VoIP networks. They also lack the ability to segment problems to the CPE, customer network, or service provider network. Furthermore, dispatching a technician with a comprehensive network analysis instrument to the customer premises usually indicates sending an engineer with more skills—at a higher expense.
Using handheld IP phone emulators for active testing enables the service provider to dispatch lower level field technicians for customer-facing problems. Using these handheld devices, technicians can qualify a customer complaint. Then the technician can perform active testing at strategic points in the network to help segment the point of origin of the problem. Testing can be coordinated to leverage NOC resources while in the field and help prevent problems from escalating to the next tier.

To improve the troubleshooting process, service providers can minimize their dispatches for data gathering by using packet capture agents (PCAs). The PCAs empower the end user to gather vital information that can be forwarded to the network operations center (NOC) for analysis. Many problems can be identified and resolved remotely using PCAs.
### Figure 8.8 Sample PCA report

**Call Quality**

<table>
<thead>
<tr>
<th>Source</th>
<th>MOS Value</th>
<th>R Factor</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>CQ</td>
<td>LQ</td>
</tr>
<tr>
<td>192.168.53.77:16568 (G.729)</td>
<td>3.88</td>
<td>3.91</td>
</tr>
<tr>
<td>10.1.1.6:18328 (G.729)</td>
<td>2.12</td>
<td>2.31</td>
</tr>
</tbody>
</table>

**Voice Quality**

<table>
<thead>
<tr>
<th>Quality</th>
<th>Excellent</th>
<th>Good</th>
<th>Fair</th>
<th>Poor</th>
<th>Bad</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>5</td>
<td>4</td>
<td>3</td>
<td>2</td>
<td>1</td>
</tr>
</tbody>
</table>

**Call Degradation Factors**

- Code: 32.29%
- Codec: 0.09%
- Error: 7.16%
- Jitter: 0.09%
- Packet Loss: 0.09%
- Packet Error: 0.09%
- Frame Error: 0.09%
- Signal Error: 0.09%
- Total: 100.09%

**Call Statistics**

<table>
<thead>
<tr>
<th>Source</th>
<th>Jitter Max</th>
<th>Jitter Avg</th>
<th>Jitter Abs</th>
<th>Packet Loss Max</th>
<th>Packet Loss Avg</th>
<th>Packet Loss Total</th>
</tr>
</thead>
<tbody>
<tr>
<td>192.168.53.77:16568 (G.729)</td>
<td>7.68</td>
<td>1.21</td>
<td>41.44</td>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>10.1.1.6:18328 (G.729)</td>
<td>16.76</td>
<td>5.48</td>
<td>43.04</td>
<td>3</td>
<td>0.42</td>
<td>158</td>
</tr>
</tbody>
</table>

**Jitter**

- Median: 0.00
- 95th Percentile: 10.00
- Maximum: 18.00
- Average: 1.00

**Packet Loss**

- Maximum: 3.22
- Maximum: 3.22
- 95th Percentile: 3.22
- Maximum: 3.22
- Average: 3.22
- Average: 3.22

---

8/11/2004 4:24:37 PM
**Provisioning Tests**

**Purpose:** To make sure that the IP phone is set up properly and can place/receive calls

**Tests**
1. Registration with Gatekeeper/proxy server - Ping
2. Place calls on and off the network
3. Trace the call route - Traceroute
4. Assess MOS/R-Factor for call quality

**Troubleshooting Tests**

**Purpose:** To determine the appropriate causes for poor call quality

**Tests**
1. Handset, volume, processor/DSP performance, mic/earpiece acoustics and echo canceller performance
2. TDM voice quality, echo
3. Network packet performance
   a. Packet throughput
   b. Packet loss
   c. Packet delay/latency
   d. Packet jitter

1. Call won't connect
   Likely trouble scenarios: Configuration errors, Packet Loss
   Common Causes:
   a. Registration problem with Call Manager
   b. Gateway provisioning problem

2. Can make calls on, but not off the network
   Common Causes:
   a. Codec improperly configured

3. Call connects but can't hear anything

4. Call connects but hear clicking/popping noise to the end user, garbled speech.
   Likely trouble scenario: Packet Loss, Packet Jitter
   Common Causes
   a. Network Congestion
   b. Buffer overruns in network components
   c. Priority queuing and traffic shaping problems
   d. Priority queuing and traffic shaping problems in combined VoIP / data networks

5. Call connects but then drops
   Likely trouble scenario: Packet Loss

6. Call connects but hear over talk and line echo
   Likely trouble scenario: Packet Delay
   Common Causes:
   a. Priority queuing and traffic shaping problems (CoS/QoS)
   b. Choice of CODEC selection (packetization, buffering, compression)
   c. Network congestion

**Packet Loss**
- Customer Experience – popping/clicks, dropped calls in extreme cases
- Best Test– exact measurement based on IP packet Sequence Stamp
Isolating VoIP problems

The following steps detail some of the ways that service providers might respond to a customer’s complaint that a phone does not work. The first step is to determine if the problem is in the network, a provisioning issue, CPE related issue, or a customer issue. To accomplish this task, the technician must perform an IP phone emulation test. The next step is to attempt registration of the test device with the gatekeeper/proxy server. The test set conducts signaling decodes and displays signaling error messages. It proves connectivity to the gatekeeper/proxy server and verifies provisioning of the customer’s unique alias. The technician also uses the device to place test calls on and off the network.

Another set of tests verifies the IP phone’s connectivity. A ping test suite can help determine if a registration problem exists. The test pings the gatekeeper/proxy server IP address. If off-net calls fail, gateway provisioning may be causing the trouble or there may be a connectivity issue. The latter can be identified by pinging the gateway device’s IP address. A trace route test suite is performed if the ping test fails with the gateway or gatekeeper/proxy server. The trace route helps isolate path/device connectivity problems.
Solving Voice Quality Problems

Even when phones are working, the quality may be unacceptable. The problem may be either conversation quality or voice quality. Conversation quality is impacted by dropped or incomplete calls, voice clipping and silence suppression. Voice quality is impacted by pops, clicks, garbled words, and echo. Poor quality is usually caused by problems with the CPE, TDM, or network packet performance.

When attempting to resolve quality issues, it is important to distinguish between controllable factors and non-controllable factors. For instance, field technicians have limited control over CPE performance, but they do have control of network performance and its impact on CPE performance.

If a customer indicates that voice quality is bad, excessive echo is present, and there are garbling and clicking sounds on the line, the first step the technician must take is to verify the customer’s complaint by evaluating quality from the end user’s perspective. The test set is used in IP phone emulation mode to place test calls. Based on whether or not same problem is observed on the test set, the CPE can be ruled out or established as the cause of the problem.

To establish subjective voice quality, the technician must listen to a live test call for imperfections and monitor and correlate network metrics to call imperfections. The handheld test set enables the technician to test the line from multiple locations to isolate the source of the trouble. In the enterprise, calls can be placed from the next router down all the way through to the ingress router to diagnose whether the problem is on the LAN or on the WAN link. On the residential side, calls can be placed outside the home where the copper pair terminates onto the home wiring system or home networking system or the back of the customer modem in order to further segment the problem.
Objective voice quality is established by generating a MOS, which is then compared to maintenance and provisioning and SLA requirements. Measurements are taken at the packet interface of live calls. The MOS is calculated based on network performance and transport measurements such as packet delay, packet jitter, packet loss, and out-of-sequence packets. The customer quality complaint is based on an actual MOS score derived through the customer’s ear. The technician’s MOS values are based on a MOS score calculated by a test device. If the test set shows a good MOS score, the call must be captured and listened to. To take this step, the technician must use a test set with the ability to play back the voice and emulate the actual VoIP phone.

**VoIP Testing Strategy**

Providers rolling out VoIP in the triple play must address a tough challenge from the beginning. Potential customers have a preconceived expectation of the quality of VoIP services. These expectations are based on years of experience with legacy landline voice services. Achieving this level of service and voice quality is possible for hosted residential and business VoIP services with careful testing at all phases of service deployment. For a service provider, making a bad first impression with a new service, which often involves losing the customer, is usually more expensive implementing proactive and comprehensive test procedures.

Experience gained from the deployment of other new services demonstrates the importance of sound installation methods and procedures supported by effective field test tools in achieving deployment success. This combination can enable the field technician to turn-up new services and to troubleshoot problems quickly yet efficiently. In this model, deployment costs are minimized by enabling less experienced technicians to isolate and solve problems at the premises. Providers will see the impact on the bottom line—future trouble calls, trouble escalation, and dispatch of higher level support can all be greatly reduced.
A fragmented approach to testing is not effective for supporting a packet-based service with the stringent performance requirements of VoIP. A comprehensive end-to-end testing strategy must be developed to ensure customers’ QoE. It should be comprised of custom-developed bundled sets of handheld test instruments, test systems, and test software. The testing strategy should provide an integrated approach that is coordinated with each stage of VoIP service delivery.

In addition to providing technicians with the ability to see and hear how VoIP is performing, test suites should help service providers anticipate and prevent problems before they occur and escalate into larger problems—ultimately to reduce expensive backtracking and troubleshooting. VoIP networks deployed using these processes and procedures and test methods will provide reliable consistent service to end customers and operate cost effectively and profitably for the service providers implementing them.
Troubleshooting High Speed Data Service

Traditional telecom providers face enormous challenge, but also great opportunity, in addressing today’s market for the high speed data service component of triple play. As carriers adopt FTTx strategies that can make triple play a reality, a considerable hurdle remains: optimizing the use of existing copper facilities to meet performance and QoS expectations. This must be accomplished in a competitive environment which is already seeing some providers lose market share.

Over the last few years, with the advent of VoIP and IPTV, high speed data service has gone from the most to the least critical service from a performance standpoint. Yet it still requires fast and effective turn-up and troubleshooting in order to retain customers and keep costs at a level conducive to profitable service delivery. Despite the fact that VoIP and IPTV applications inherently require higher network performance, high speed data service delivery is more closely scrutinized because users tend to be the most technically savvy and demanding members of the customer base. Providers have found that data service, too, must perform as advertised, to retain customers. Furthermore, with advanced networked video peer-to-peer sharing technology beginning to emerge, the demand for increased up-stream bandwidth is a new and challenging paradigm shift.

![Figure 9.1 Internet data service delivery network](image-url)
Providers typically offer service level agreements (SLAs) for high speed data service only for business class customers. Yet with consumers becoming increasingly demanding, it is likely that business-style SLAs may soon migrate to residential customers. In the meantime, some service providers already measure their performance on residential accounts by their ability to meet typical business level SLAs with a particular focus on throughput.

**Understanding xDSL**

Delivery of high speed data service over copper is possible largely due to advances in digital subscriber line (DSL) technology. ADSL/2/2+ places increased demands on physical loop performance to enable service delivery at data rates necessary for triple-play services. VDSL technologies further increase loop performance requirements.

**ADSL**

Delivery of ADSL-based data services requires a single copper pair to carry a standard voice circuit as well as the data service. Three information channels are created: a plain old telephone service (POTS) channel for voice, a medium speed upstream channel, and a high speed downstream channel. The POTS channel is separated from the ADSL channels via a passive, low-pass/high-pass filter that separates the low frequency POTS signals from the high frequency ADSL signals. The splitter also protects the DSL signal from POTS signal transients originating from events such as on-hook and off-hook and ring tones.

ADSL service may be installed without a splitter by using micro filters in-line with the phone jack at each telephone device, enabling the customer to install the customer premises equipment (CPE). Data rates depend upon several factors including length of the copper pair, wire gauge, presence of bridged tap, proper bonding and grounding and the overall
health of the loop as assessed by pair balance. Line performance increases as the loop length is reduced or the wire gauge is increased. Additionally, removal of bridged taps as well as ensuring the line has good pair balance and is free of cross-coupled interference are key steps to optimizing data rate performance.

The modem (ATU-R) located at the subscriber’s premises communicates with the modem (ATU-C) at the central office (CO). The CO modem bank consists of cards mounted in the DSLAM. A residential or business customer connects a PC and modem to an RJ-11 telephone outlet. The existing house wiring usually carries the ADSL signal to the NID located on the exterior of customer’s home.

At the CO, a main distribution frame collects the cables from many subscribers and uses splitters to distribute the data traffic to a DSLAM and route the regular telephone traffic to the public switched transport network (PSTN). The DSLAM mixes DSL services from different subscribers onto high speed edge network circuits, either ATM or Ethernet based, depending on the DSLAM configuration. Often, a DSLAM concentrator is used in cases where an ILEC or CLEC has many DSLAMs distributed over a large geographic area.
**ADSL signal encoding**

Traditional POTS uses a narrow 4-kHz base band frequency to transmit analog voice signals. This means that even with sophisticated modulation techniques, current dial-up modem technology can only achieve data throughput up to 56 kbps. DSL obtains much higher throughput by using a much wider frequency spectrum. Frequency division multiplexing (FDM) creates multiple frequency bands to carry the upstream and downstream data.

![Figure 9.3 Frequency bands](image)

The American National Standards Institute (ANSI) and the International Telecommunications Union (ITU) chose discrete multi-tone modulation (DMT) as the standard line code for ADSL. DMT divides the frequency spectra into parallel channels where the center of each channel is represented by a modulated quadrature amplitude modulation (QAM) sub-carrier. Each carrier is orthogonal to the other sub-carriers so there is no interference between sub-carriers. ADSL is a two-band system where one part of the frequency spectrum is used for upstream transmission and the other part (up to 1.1 MHz) is used for downstream transmission. QAM employs a combination of amplitude modulation and phase shift keying. For example, a signal that
transmits at three bits per baud requires eight binary combinations to represent the signal. This example assumes two possible measures of amplitude and four possible phase shifts, which allow for eight possible waves. Using this method, a large bit stream is broken down into a few bit words per DMT channel or tone.

**ADSL2+**

In July 2002, the ITU completed the ADSL2+ standard which provides major improvements in data rate and reach performance, rate adaptation, and diagnostics. ADSL2+ specifies a downstream frequency spectrum of up to 2.2 MHz providing a substantial increase in downstream data rates of over 25 Mbps at shorter loop lengths. ADSL2+ also extends the lower frequency spectrum which increases upstream data rates to about 1 Mbps. In addition to increased spectrum usage, other enhancements are made. As opposed to ADSL where overhead bits are fixed and always consume 32 kbps per frame of transmission capacity, in ADSL2+ the overhead data rate can be reduced to 4 kbps, providing an additional 28 kbps for payload data. Improved Reed-Solomon error correction coding yields higher coding gain, and the modifications to the initialization state machine deliver data rate increases.
VDSL
In the meantime, VDSL1 and VDSL2 have been certified as standards by the ITU, using the same DMT concept as ADSL and ADSL2+ with QAM line coding in the tones as before. VDSL and VDSL2 use a much larger frequency spectrum making multiple bands for upstream and downstream transmission possible. This enables a greater degree of flexibility for higher data rates and service configurations supporting symmetry between upstream and downstream rates. VDSL2 uses up to 4,096 tones which are spaced 4 kHz or, in some cases, 8 kHz apart.

A frequency spectrum of up to 30 MHz in VDSL2 (as opposed to 12 MHz with VDSL1) results in 100 Mbps symmetric data rates to users within 1,000 feet of the central office. On the other hand, once reach exceeds a mile, performance degrades to about 24 Mbps, approximately the same level as ADSL2+.

While this migration to VDSL1 and VDSL2 is occurring, packet-based transport over VDSL is also replacing traditional ATM technology in order to facilitate Ethernet service all the way to the customer. Migration to an Ethernet-based architecture such as VLAN per-service can greatly simplify the network architecture and reduce network operating costs significantly.

While VDSL offers significant potential advantages such as the capability to provide much greater bandwidth as well as symmetric service configurations, new challenges also exist. Early-stage technology does not guarantee interoperability amongst differing modem and chip set designs. Further, since VDSL2 uses a higher frequency spectrum (up to 30MHz), new noise environments are present which must be understood and managed. New noise sources, such as short wave radio stations, and the requirement to not interfere with existing spectrum usage, such as amateur radio transmitters, makes VDSL deployment complex. Reach and rate performance not only depends on the copper loop, but also on spectrum management. There is still a considerable amount to be learned about VSDL
performance over various types of copper pairs as well as its susceptibility to radio frequency interference. Perhaps the most significant unknown is the precise rate and reach performance on a given loop.

Figure 9.5 summarizes rate and reach performance of several DSL technologies.

![Figure 9.5 DSL Rate and Reach Performance by Band Plan](image)

Table 9.1 summarizes the amateur radio bands that must be protected from interference by VDSL deployments as specified in ITU G.993.2. DSLAM configurations allow the removal from usage any of these bands based upon a given geographic territory’s requirements for protection. These radio bands or “notches” impact potential maximum data rate performance by their removal of usable spectrum.
ITU standard G.993.2 defines several band plans for VDSL deployment. Figure 9.5 below outlines two. The bands (“U” for up-stream and “D” for down-stream) show how the larger VDSL spectrum is broken into multiple bands. The U0 band is used when no POTS or ISDN service is to be carried on the same copper loop or pair. Based upon the modem design, some or all of these bands can be used to achieve the desired rate reach performance.
In order to simplify some VDSL complexity, the standards introduce the concept of “profiles.” According to ITU G.993.2, “profiles are specified to allow transceivers to support a subset of all the settings and still be compliant” with the standard. Multiple profiles allow vendors and service providers to limit implementation complexity and target specific service requirements. Profiles mentioned include 8a, 8b, 8c, 8d, 12a, 12b, 17a, and 30a where the number indicates the upper limit of the frequency spectrum to be used. Table 9.2 below provides a few examples.

<table>
<thead>
<tr>
<th></th>
<th>8b</th>
<th>17a</th>
<th>30a</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum aggregate downstream transmit power (dBm)</td>
<td>+20.5</td>
<td>+14.5</td>
<td>+14.5</td>
</tr>
<tr>
<td>Maximum aggregate upstream transmit power (dBm)</td>
<td>+14.5</td>
<td>+14.5</td>
<td>+14.5</td>
</tr>
<tr>
<td>Subcarrier spacing (kHz)</td>
<td>4.3125</td>
<td>4.3125</td>
<td>8.625</td>
</tr>
<tr>
<td>Minimum net aggregate data rate (Mbit/s)</td>
<td>50</td>
<td>100</td>
<td>200</td>
</tr>
<tr>
<td>Typical use case</td>
<td>CO</td>
<td>FTTN</td>
<td>FTTB Japan</td>
</tr>
</tbody>
</table>

Annex A, Annex B (998)

| Index of highest supported downstream data-bearing subcarrier (upper band edge frequency in MHz (Informative)) | 1971 (8.5) | N/A | N/A |
| Index of highest supported upstream data-bearing subcarrier (upper band edge frequency in MHz (informative)) | 1205 (5.2) | N/A | N/A |

Annex C

| Index of highest supported downstream subcarrier (upper band edge frequency in MHz (informative)) | 1971 (8.5) | 4095 (17.664) | 2098 (18.1) |
| Index of highest supported upstream subcarrier (upper band edge frequency in MHz (informative)) | 1205 (5.2) | 2782 (12) | 3478 (30) |

Table 9.2 VDSL Profile Examples
VDSL may be deployed in the same binder groups as ADSL. Two examples are outlined in Figure 9.7. Cross talk problems from VDSL service affecting ADSL service can be a significant issue in such mixed environments. In the figure, the case where VDSL and ADSL are serviced from the CO or Exchange, the VDSL signal can affect the ADSL signal (see Point A in Figure 9.7.) In the case where VDSL is served from a remote node, cross talk can be even more of a problem. This occurs where the ADSL signal has been attenuated and is weaker and the VDSL signal is stronger since it is launched at the remote node (see Point B in the Figure 9.7). Management of the PSD for VDSL, specifically reducing power on a pair by pair basis to minimize the interference, is both possible and needed. In these cases the use of different profiles to reduce spectrum overlap is also beneficial.

Figure 9.7 VDSL crosstalk analysis
High Speed Data Service Delivery Test Strategy

With this extensive list of xDSL caveats and considerations, network operators and service providers face a variety of testing challenges to install and maintain robust and cost-effective high speed data services. Providers need a test strategy that ensures connectivity and quality from the time of turn-up and fast root cause identification and resolution of problems that arise on the live network.

The strategy begins with establishing the readiness of the physical plant to support high speed data service (as explained in Chapters 3 and 4). Connectivity of the service to the DSLAM in the local exchange must also be assured. Beyond the DSLAM, connectivity and routing, both to the ATM or IP network and ultimately to the service provider, need to be verified to ensure the customer’s expected level of service.

There is also the challenge of providing and maintaining different DSL variants simultaneously such as ADSL, ADSL2+, VDSL, and VDSL2 to meet the demands of business and residential customers using fiber and copper for the last mile. To complicate matters, all of these challenges must be addressed within an operating environment constrained by reduced budgets, smaller workforces, and tighter deadlines.

As a result, a comprehensive test strategy that addresses each of these aspects is critical. The test solution must deliver comprehensive physical layer, data, and service testing at the twisted pair/metallic, DSL, ATM, IP, and application layers. It must also support multiple DSL variants. In addition, it should be capable of delivering process improvement features that can be utilized to improve productivity and efficiency. Portable test tools with Web browser and FTP throughput test capabilities are required to verify DSL physical layer performance, ISP connectivity, and ISP and data service throughput.
Turn-up testing

Before a provider commissions DSL service, technicians must check DSL performance from the CPE through the DSLAM, then validate and confirm several protocol layers to verify that the network can transport voice, video, and data service payloads. A failure anywhere across any of the different layers—transport, network or service—will cause the service to fail. Therefore, it is important to check not only data rates between the NID and DSLAM, but also to test the application-specific QoS metrics to validate service quality. An efficient method to accomplish this task is to use a test set that emulates the ADSL modem and establishes connectivity with the DSLAM. The tester should also emulate the subscribers’ CPE to validate and confirm voice, video, and data services. Experience shows that many problems result from CPE configuration errors as well as network provisioning. Quickly identifying and proactively addressing these problems saves time and expense that would otherwise be spent on repeat service calls.
The JDSU HST-3000 provides complete DSL testing capabilities. During turn-up testing, the test set connects to the line and establishes synchronization with the DSLAM, then confirms DSL layer performance by emulating DSL modems. In this mode of operation, the test set verifies the actual DSL data rate for the current or actual rates, and the connection’s maximum possible rates. Line capacity is calculated by comparing actual and maximum rates as an indication of robustness or spare capacity. When capacity exceeds 80 percent, service can become very sensitive to transient noise conditions. The measured signal-to-noise margin should be evaluated to assess the DSL signal strength over the loop, pointing to future performance capabilities and limitations. SNR and attenuation/gain per tone are analyzed. When noise problems are suspected or detected, SNR and attenuation/gain per tone data (graphs) should be analyzed as necessary to identify the impact, scope, and potential source of the interference or noise. Once these steps have been completed at a given location on the loop, it is often beneficial to connect to the DSL service at other points on the loop to further sectionalize potential fault conditions.

Troubleshooting

Qualification and Synchronization Problems
If the service qualification fails, or if the system does not synchronize, the following systematic trouble isolation procedures are recommended:

- The technician must replace the DSL modem with the test set to perform emulation and attempt to synchronize with the DSLAM.
- If the technician is able to synchronize with the test set but not with the DSL modem, a check of the minimum bit rate setting at the NMS is recommended. If the minimum rate is set too high, the DSL modem does not synchronize. If the NMS settings are correct, the DSL modem should be replaced.
If synchronization at the test set cannot be gained, a check for dial tone will determine if the network is connected to the central office.

If there is no dial tone, it is necessary to check for an open using a loop-troubleshooting tool with a TDR to locate the fault.

If a dial tone is available, then a check for load coils on the span is needed. If none are present, the technician must move up the loop to the next access point, typically the pedestal.

The technician can disconnect the circuit from the pedestal to the customer premises equipment and connect the test set to the loop facing the central office or digital loop carrier.

If synchronization is successful, the problem is on the loop between the pedestal and the NID.

The technician may continue this process, as needed, along the complete copper span, to check the splice case, cross box, and main distribution frame.

If synchronization is successful at the NID, the in-home wiring to the modem or residential gateway (RG) most likely is the cause of the problem. Typically, this problem is isolated with a simple digital volt ohmmeter test. An alternative solution is to install a new cable directly from the NID to the wall outlet at the modem/RG location. If synchronization is possible in the house but the performance is poor, the technician must move the test set to the NID to check for improved performance. If performance shows an improvement, the cause of the problem likely is faulty in-home wiring. If synchronization is possible at the NID but performance is poor, the source of the problem most likely is on the span. The technician should look at a bits carried per DMT tone graph, and then compare the bits per DMT tone to the SNR per DMT tone data. Unusual dips in
the graphs may indicate impairments on the line such as wet sections, bridged taps, or resistive faults. The pair should be examined for the presence of fault conditions as discussed in the physical layer testing chapter.

Excessive noise on the line usually is the result of either pair imbalance or bonding and grounding issues. In addition to lower than acceptable DSL bit rates, a second potential indicator of this would be a noise margin failure condition. The excessive DSL line errors, especially when they occur in bursts, are often another indicator of a potential noise problem. If DSL testing indicates a suspected noise problem, the pair should be tested for the presence of excessive wideband noise or wideband impulse noise. If results are above established thresholds, the technician should isolate the root cause, typically a pair imbalance or bonding and grounding issue as described in the previous section.

**Connection problems**

If a ping test to the ISP address is not successful, the ISP may be experiencing service problems, or the address is incorrect. This procedure verifies network and ISP connectivity. The choice of a destination address determines the results to be expected from the ping statistics. If a ping test is conducted on the broadband remote access server (BRAS) or primary domain name server (DNS) server, a better than 90 percent success rate and a short delay is expected. However, if a ping test is conducted on a public IP address outside the domain, results may vary. The tester used for this application must show these statistics to provide a complete picture of the network environment.
**Service verification**
A true test of end-to-end service requires verifying access to that service. Technicians must use a test instrument in IP ping mode to verify the routing connectivity across the network to an IP host or server while assessing packet loss rates and packet delay to and from the ping destination. The instrument should check the IP layer by verifying whether another host device is alive and able to echo back; use a flood mode to gauge network congestion; and, determine the minimum, maximum, and average packet delay time of IP packets. Tracking packet delay and loss helps determine whether delays and slow service are due to provider error or CPE problems.

Since users can only reach the ISP end of the service with the correct username, password, and encapsulation, test tools must support the appropriate IP encapsulations and authentication protocols for effective IP ping support.

**High Speed Data over FTTx**
Validating data service over an FTTx access network is similar to DSL access topology testing. The technician must:

- Establish connectivity, to the ISP
- Provision necessary network elements for increased data flow and class of service treatment
- Reconfigure DSLAM ports for dual latency path support for the mixed IP application environment

To complete the installation process, field personnel must verify DSL physical layer performance, ISP connectivity and ISP and data service throughout. FTP throughput testing with selectable file sizes in both upload and download direction establishes the performance of the link in a way that much more closely matches actual use cases than a simple download test. HTTP testing using a Web browser must be completed to ensure that the end users’ ISP access/connectivity is working properly.
Process Automation Yields Measurable Improvement

High-speed data service turn-up—done properly—is an exacting and time-consuming task. To address the many potential obstacles, leading providers are adopting automated workforce management solutions that interface directly with smart instruments capable of addressing all scenarios technicians may encounter.

The JDSU HST-3000 Handheld Services Tester, for example, provides automated testing capability that includes copper physical layer, xDSL, and IP service layer checks. This combination allows technicians to more easily distinguish between core and access network problems. In addition, the automation allows rapid problem isolation to the appropriate layer, (e.g., quickly identifying a packet loss issue and correlating it to a noise problem caused by pair imbalance at the physical layer).

During the test, a vast array of data, including IP, DSL, and copper loop performance, is gathered for further analysis and record keeping. These instruments can be integrated with Web-based tools to tie the efforts of field technicians with the tasks of central office staff, yielding unprecedented advancements in operational efficiency.
Select and initiate test and the instrument automatically progresses through a sequence of tests to verify DSL sync & performance, PPP authentication, IP connectivity and IP data rate.

Figure 9.8 HST 3000 automated DSL testing
Measurable Improvement
Measurable process improvement is achieved in this new approach via advanced scripting that anticipates typical scenarios. As a result, technicians no longer spend time on repetitive tasks. Rather, the built-in intelligence manages the instrument through the test set-up, execution, results storage, and results upload processes. Complex processes are simplified and in some cases reduced to single button pushes. If needed, the technician can move quickly to manual troubleshooting techniques. This automation ensures that each technician performs tests consistently and to the provider’s established standards at each job.

Adherence to provider-set thresholds for service performance is guaranteed. Automatic software updates allow the technician to keep pace with test technology changes and new test scripts. Such customized solutions and procedures are offering providers a new venue to lower operational costs and reduce customer churn.

As consumer demand for bandwidth pushes copper to its limitations, the migration to fiber to the x (FTTx) has gained concentrated momentum. The advent of fiber in the loop is reducing operating expenses related to the copper plant. But, the need for smart test instrumentation and a systemized approach to workforce management is still critical. Implementing automated processes that incorporate centralized systems and smart instruments will establish the foundation to efficiently deliver services that meet customers’ QoS expectations.
Sustained Performance Improvement

Highly configurable, customer-specific process automation is vital for the success of an automated workforce management solution. Working closely with service providers across the globe, JDSU has developed a framework for sustained performance improvement. By adapting the TechComplete™ application to facilitate adherence to established methods and procedures while automating and expediting service turn-up and repair, JDSU is helping providers to meet four key business objectives:

Reduce costs
Automation introduces efficiency that sharply reduces three cost factors: Repeat failure rates that drive costs up and drive the need for additional truck rolls; lengthy repair times that increase operational costs; extensive training to bring technicians up to speed on process and technology.

Increase Revenue
Service outages and slow turn-ups eat away at profits. By introducing the efficiency of process automation, providers can reduce outages and technicians can bring more customers on line faster.

Increase Service
With reduced repeats customer satisfaction improves, and customers are more likely to add services.

Decrease Complexity
Today’s workforce model requires increasingly intelligent test solutions that reduce complexity. Automation not only provides one-button execution for tests, but also centralizes test set management so that providers can quickly deploy new methods and procedures which must evolve as services and technologies change.

By the Numbers
Statistics gathered in actual practice indicate that dramatic achievements in efficiency and reduction in costs can be achieved. Results include:

- A reduction in repeat rates from 24% to 5%
- A 30% drop in incident reports during the first month of deployment
- A 25% reduction in DSL repair dispatches
- A 12% reduction in technician time spent per installation dispatch
- A 15% reduction in technician time spent per repair dispatch
Service Assurance for Triple-Play Services

Triple-play service delivery presents challenges from concept inception. Not only must providers roll out new network infrastructures and upgrade existing ones with exacting detail to ensure that inherent bandwidth requirements can be met, but technicians must master new terminology, technology, and testing techniques to ensure that services are correctly provisioned and installed. And once the new subscriber’s service is established, the challenges do not diminish on-going service assurance is critical.

While triple-play service delivery is a core business strategy for today’s providers, they are vying to offer services that are already available from a range of competitors. New entrants to the market find an aggressively competitive landscape where they must meet exacting service quality standards. In every case, customers have previous experiences—and therefore set expectations—of acceptable service delivery for voice, video, and data because virtually everyone already talks on the phone, watches subscription-based TV, and surfs the Internet.

With these mitigating factors, on-going, pro-active service assurance for the packet-based network that delivers the triple play is crucial. As providers entice customers away from traditional service delivery methods, they must prepare to deliver services more efficiently and economically to satisfy a sophisticated clientele. A poorly executed installation and/or maintenance plan can result in permanent loss of customers. With a multitude of options available, disappointed customers can simply choose other voice, video, and data providers.
For successful market entry, providers must deliver reliable service that spans the entire network from headend to customer. To accomplish this, solutions are needed that give the provider true, end-to-end customer QoE service visibility. Visibility into network performance at varying levels of bandwidth consumption will be important. With limited predictability in dynamic network performance, the provider must break down traditional organizational boundaries, share information, and manage service delivery end to end. An effective, comprehensive service assurance system must be in place to prevent potential service quality issues. With customer acquisition costs ranging up to thousands of dollars per user, the impact of the loss of a single subscriber is significant.

An effectively planned and implemented service assurance strategy will address this and more. It will enable the provider to decrease costs and complexity and ultimately increase revenue. Satisfied customers will remain loyal and are likely to add new services as they become available.

**Technical and Operational Challenges to Triple Play**

Providers rolling out triple play are rapidly discovering a significant number of challenges—both technical and operational—that must be overcome to successfully deliver voice, video, and data services. The technical challenges include the reality that:

- Voice, video, and high speed data place different burdens on the network delivering the service.
- The traditional network—core, access, and home—is not optimized for these new services.
- New service deliveries usually involve complex IP-protocol interactions among home, access, network, and source elements.
The operational challenges include the need to:

- Manage the service not just the network.
- Share information and responsibility end to end.
- Accommodate the dynamic nature of IP service delivery.

Operationally, many providers’ organizational structures are separated into distance groups such as engineering, network operations, field operations, and provisioning. This organizational structure may hinder ease of information sharing among groups which can result in inefficiencies and duplication of effort. With the new services being delivered over IP, it is indeed challenging to properly distinguish between content issues, customer premises issues, and network performance.

QoE is the key differentiator in today’s competitive marketplace. Providers understand they cannot afford to have quality problems such as long IPTV channel change times, video pixelization, no dial tone, dropped VoIP calls, bad call quality, slow data speeds, or any of the other myriad problems that can result from trouble on multi-service IP networks. It is apparent that conventional network focused tools, methods of operations, and operational organizational structures designed for traditional services are insufficient for maintaining and understanding true end-to-end QoE for triple-play service delivery.

Service providers now require end-to-end service assurance solutions that provide a continuous view of application specific QoS metrics and that will make it possible to quickly and efficiently troubleshoot, identify, and resolve any quality degradation.
Service Assurance Challenges for Triple Play

Providers are making use of the functionality of existing network element and service reporting and fault management systems. Yet, these systems must be augmented in order to deliver the required QoS for new triple-play services. While the required additions vary, essentially at issue is the need to add video tools to the legacy voice provider and voice tools to the legacy video provider.

For providers new to the video market (IPTV service delivery), systems must be implemented to:

- Ensure the quality of video services as launched from the headend.
- Deliver video services error free across the optical network.
- Manage the performance of xDSL loop to optimize the bandwidth and service offering.
- Monitor the service performance end to end including the loop and home environments.

For providers new to the voice market (VoIP service over hybrid fiber/coax), systems must be implemented to:

- Ensure quick identification and isolation between loop and network and handoff service issues.
- Monitor the performance of the upstream ingress to ensure loop quality is sufficient for VoIP.
- Ensure total voice customer experience including managing delay, jitter, packet loss and echo.

By adding components to address these issues to an existing service assurance portfolio, the provider can ensure the ability to deliver the new service with high availability.
Components of an Integrated Service Assurance Solution

A comprehensive service assurance solution should address the correct mix of business processes as defined by the Enhanced Telecom Operations Map (e-TOM) model:

- Service Problem Management
- Service Quality Analysis/Action and Reporting
- Resource Problem Management
- Resource Quality Analysis, Action and Reporting

Many service providers make the assumption that the network equipment providers, or outsourced solution providers, will provide the majority of this functionality with the aim of leveraging their existing investments in their OSS infrastructure and tools. However, as time passes and service deployment volumes increase, many providers express the following concerns:

1. What happens when I have multiple vendors with different approaches?
2. How do I get more visibility of the service performance (higher layer applications), not just the network performance (layer 1-2 applications)?
3. How do I build an operations support environment that scales independent of vendor-specific element solutions?
4. How do I create a common toolset across the organization to optimize available resources while avoiding vendor-specific specialization?

A comprehensive service assurance solution will combine elements of the e-TOM model, described in more functional terms in the following Service Assurance Model.
**Figure 10.1** The NetComplete Service Assurance Model for end-to-end coverage

**Required functions and capabilities**
Discrete functions of a service assurance solution in this model may include all or subsets of the following depending upon application needs:

1. **Fault Isolation**—rapid and automated fault segmentation and/or isolation. E.g., Correlation in fault management systems to quickly isolate fault locations end to end.

2. **Troubleshooting**—providing reactive, in-depth analysis tools that allow in-depth testing and inspections to help determine what causes the faults.

3. **Automated Testing**—pre-scheduled routines to proactively test target areas.

4. **Passive Monitoring**—non-intrusive measurement of actual customer traffic.

5. **Active Testing and Monitoring**—measurements of performance based on synthetic traffic.
6. **Turn-Up**—test tools to assist the accuracy of initial service delivery to end customer.

7. **Reporting**—providing users actionable information in performing their roles.

8. **Data Collection**—gathering key metrics or measurements to aid in analysis.

9. **Data Correlation**—correlating related data to better assess and analyze information.

10. **Performance Management**—data assessment to improve or optimize performance.

In pulling all of these requirements together, a successful service assurance solution should:

1. Verify the integrity of content as it is entering and leaving the network.
   a. Verify the incoming content as it enters the network and at key hand off points to ensure it is good using metrics such as MOS, V-MOS, and audio MOS that provide an objective indicator of quality.
   b. Continuously measure the customer experience to verify the service is delivered as expected.

2. Verify the integrity of the network and the services running on top
   a. Service and Network Performance Management continuously measures the ability of the network to deliver the services as promised.
   b. Correlate Network performance to service performance to distinguish between systemic network and isolated customer issues.
   c. Use Fault Management to identify and correlate network faults and alarms.
d. Perform Capacity Management to understand network utilization and performance for proactive engineering and planning.

3. Provide the ability to quickly and remotely perform service-centric fault sectionalization.
   a. Fully diagnose faults using automation, expertise and common toolsets across functional groups, making it possible to dispatch to fix, not to find problem.

A robust, yet cost-effective service assurance solution will provide a centralized solution to accomplish these objectives by:

1. Fully utilizing and leveraging the inherent capabilities of the network elements, their associated EMSs, and customer premises equipment to:
   b. Manage the individual customer experience by obtaining key metrics from CPE devices and reporting the individual customer’s experience for proactive fault identification.

2. Complementing network element solutions with remote hardware probes and software agents deployed at key points in the network to:
   a. Provide end-to-end service visibility by continuously monitoring key quality indicators and degradation factors as defined for each type of service.
      i. Passively monitoring real customer traffic and performing QoS Analysis and Reporting.
      ii. Actively injecting synthetic traffic to proactively assess service performance.
b. Correlate network performance to service performance by providing a single view accessible across groups and organizations.

c. End-to-end problem detection, segmentation and troubleshooting – extend reach to customer premise and content injection points.

3. Adapt to the unique needs of the provider, applications, and services.

   
   b. Carrier-Class Scalability – designed for 24/7 (24 hours a day/7 days a week) network operations center applications.
   

**Reporting and troubleshooting**

A complete triple-play service assurance solution performs a multitude of reporting and troubleshooting tasks—active and passive service monitoring and network performance reporting as well as active service emulation. Following is an overview of each task with sample reports and example screen captures provided to demonstrate use case scenarios for monitoring and troubleshooting service QoS.
Active service monitoring
A mesh of transactions (IPTV channel change requests, VoIP calls, HTTP downloads) can be created using probes and software agents as endpoints and QoS measurements made for all synthetically generated transactions. By placing software agents at the customer premises and distributing hardware probes at key points in the network, active transactions can be placed end to end at various points within and across the network. Each of these performance test measurements then can be aggregated and correlated to provide a proactive measure of the service quality.

Figure 10.2  Example VoIP Active Performance Monitoring Showing Monitored Paths through the Network
Passive service monitoring
Hardware probes can be placed at strategic points in the network to passively monitor the content entering the network and network traffic. The data generated by these probes can be used to calculate QoS metrics such as MOS, PCR Jitter, packet loss, jitter, and latency and to identify poor quality based on user defined thresholds.

Figure 10.3 VoIP Call Quality Detail Report

Figure 10.4 Top Conversations Monitoring
Red LED indicates an MPEG2TS with errors. Green indicates an MPEG2TS with no errors. No LED indicates it is not an MPEG2TS.

Grey streams are streams for which data is no longer received. These are called “Aged” or “Non-active” streams.

Red indicates the IFD Avg is lower/greater than the bounds set in the preferences.

This will display all the PIDS on a selected stream. “Events” is selected. The events associated with a particular stream are displayed.

Figure 10.5 Multi-channel IPTV MPEG2 Monitoring
Network performance reporting

Network performance reporting involves collection of raw usage and performance data from multi-vendor, multi-technology network segments, such as access, cable, data/IP, optical transport, and wireless service providers. This approach provides an end-to-end view of the quality and performance of the network and correlates it to the service performance being monitored both actively and passively. The following chart provides an overview of typical network performance reports and their associated functions.

<table>
<thead>
<tr>
<th>Report Name</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>Traffic Operations Reports</td>
<td>Increase customer service and network performance satisfaction by ensuring optimal service levels are met</td>
</tr>
<tr>
<td>Traffic Engineering Reports</td>
<td>Aid network dimensioning, help avoid costly over-engineering and identify the profitability of special customer features</td>
</tr>
<tr>
<td>Traffic Separations Reports</td>
<td>Support revenue collection from possible sources by identifying toll call volumes between service providers</td>
</tr>
<tr>
<td>Specialized Customer Reports</td>
<td>Provide increased revenue potential based on network usage information</td>
</tr>
<tr>
<td>Switch Performance and Network</td>
<td>Provide flexibility required to meet changes in regulatory requirements</td>
</tr>
<tr>
<td>Performance Reports</td>
<td></td>
</tr>
<tr>
<td>Trending and Forecasting Reports</td>
<td>Support forecasting requirements and capital expenditures with the ability to forecast customer service loads and plan equipment deployment accordingly</td>
</tr>
<tr>
<td>Exception Reports</td>
<td>Provide notification required when certain collected data or summarized data exceeds preset thresholds</td>
</tr>
</tbody>
</table>
Active service emulation

Active service emulation offers the service provider the capability to reproduce customer or network problems. The technician can emulate the customer scenario by mimicking traffic priority settings, codec selection, etc., in order to quickly validate the degradation. Hardware probes and software agents can be deployed at key points in the network such as content injection points and key handoff points and at the customer premises. This makes it possible to quickly sectionalize the network by running service emulation tests between the various identified end points along the service delivery path.

Figure 10.6 IPTV IGMP channel change test results

Figure 10.7 VoIP two-way call test results
Measurable Benefits

Successful triple play deployment requires managing multiple services which place increasing bandwidth demands on the network with differing traffic patterns. Any service provider offering triple play cannot risk customer dissatisfaction and resulting churn. Developing an effective service assurance strategy that enables continuous monitoring and rapid reaction to QoS issues will be a key service differentiator.

By implementing a service and network monitoring solution, analysis has shown that a 15% reduction in truck rolls and 20% reduction in MTTR can be achieved by proactively monitoring QoS and accurately isolating the problem prior to dispatch. This 15% reduction translates into significant cost benefits. The following example quantifies these benefits for VoIP service based on the following assumptions:

VoIP Cost Savings Scenario

15,000 VoIP Subscribers
1 truck roll for every 10 VoIP subscribers per month
1 trouble ticket registered for every 7 VoIP customers per month
MTTR = 7 days x average time spent per trouble ticket per day (30 minutes) = 3.5 hours
$75 per truck roll
235 x 7 productive hours/12 months = 137 hours per month
$36 per hour network operations center tech (assumption of $60,000 salary divided by annual productive hours)

Cost/month

1,500 truck rolls (15,000 subscribers/10) x $75 per truck roll = $112,500
2,143 trouble tickets per month x 3.5 hours per trouble ticket x $36 per hour NOC tech rate = $270,000.

Conservative Cost Benefit /Month

15% reduction in truck rolls translates to $16,875 savings per month
20% reduction in MTTR translates to $54,000 savings per month

Service providers can add to these savings the economic benefits of increased customer satisfaction and reduced customer churn.
Sustained Viability

Gaining—and sustaining—market viability in the triple-play arena demands that providers evolve their networks to withstand the rigors of bandwidth-intensive voice, video, and data service delivery. While the deployment of broadband access networks that bring optical fiber near or to the home and in-home distribution networks are integral to this evolution, just as critical is service assurance for the applications operating over the network. Successful triple-play service deployment is possible only with test strategies that address each aspect of the infrastructure as well as the customer experience.
PON Reflectometer Trace Analysis

**Typical Reflectometer Traces**

This section compares two typical scenarios, a core network and a distribution network, to explain the importance of PON Reflectometer Trace Analysis.

For the core network, the OTDR trace shows the splices and connectors along the link. The analysis of this trace is simple, and it is done automatically by most of the OTDRs currently on the market. The OTDR provides a signature of the link and a table of events, which can be stored for maintenance purposes.

![Figure A.1 OTDR trace of a core network](image-url)
For the PON network, the OTDR trace shows the splitter, splices, and connectors along the link. Prior to the splitter, the analysis is simple and is similar to the analysis of the core network. But following the splitter, the analysis becomes more complex. With only the trace and the table of events given directly by the OTDR, it is impossible to locate and measure the different events of the different branches that are located after the splitter. The OTDR trace does not take into account the fact that there are different branches. It only analyzes the light that reflects from the entire network (see Figure A.2). All the information is available on the trace itself. But, it is necessary to decode the information coming from the OTDR. Further software analysis is required for this decoding procedure.

Figure A.2  OTDR trace of a point-to-multi-point network
Theoretical Method of PON Reflectometer Trace Analysis

The difficulty in analyzing a PON network is directly linked to the use of the splitter. When a splitter is used, analysis of the backscatter response after the splitter is necessary. Figure A.3 shows a PON reflectometer trace analysis of a simulated network with one splitter. The splitter can have any number of branches with fiber of different lengths. Figure A.4 shows the PON reflectometer trace given by the OTDR of a simulated network with one splitter.

Figure A.3  Simulated point-to-multi-point network with one splitter

Figure A.4  OTDR trace of a simulated point-to-multi-point network with one splitter
Displayed attenuations on a PON reflectometer trace
There are two types of attenuation on a multi-branch backscattered signal. The first type, \( A_c \), is located at the splitter. The second type, \( A_{FEi} \), is located where fiber branch ‘i’ ends, corresponding to a fiber end. These attenuations are closely dependent on the adjacent fiber parameters (backscatter coefficients) as well as the splitter parameters (forward and reverse insertion losses).

In order to more fully understand the analysis, a simplified version of the theoretical formula is defined. It requires the following assumptions:

1. The different fibers connected to the splitter have very similar characteristics, including bidirectional symmetry behaviors, backscatter coefficients, and attenuation.
2. The splitter has the same characteristics for both directions.

Taking into account these assumptions, the different types of attenuation are calculated by the following formulas:

\[
A_c = 5 \times \log(m) + E.L. \\
A_{FEi} = 5 \times \log\left(\frac{m-i+1}{m-i}\right)
\]

where E.L. is the excess loss of the splitter, \( m \) is the number of outputs, and \( i \) is the analyzed branch.

Analysis Method of PON Reflectometer Trace Analysis
This example uses a 1:8 splitter with the same assumptions as in the theoretical method. A standard OTDR is used to obtain the signature of the link. The analysis method requires a learning acquisition phase. This phase uses an optical network OTDR measurement simulator. This innovative simulator, developed by JDSU, can be used with both point-to-point and point-to-multi-point networks. In addition to classical fiber, connector, splice, and attenuator simulation, the software algorithms integrate \( n \) to \( m \) splitter synthesis based on an attenuation behavior formula similar to the calculations of \( A_c \) and \( A_{FEi} \).
Learning Acquisition Phase

The learning acquisition phase can be divided into three main steps:

1. OTDR acquisition simulation with construction data (Figure 3.26).
2. OTDR acquisition in the field under real conditions (Figure 3.27).
3. Comparison between the two acquisitions, including distance tuning and locking.

Figure A.5  OTDR acquisition simulation with construction data

Figure A.6  OTDR acquisition in the field under real conditions
This learning acquisition phase allows for the generation of a reference pattern (RP) and an event reference table (ERT). The RP corresponds to the trace, and the ERT includes the reflective and nonreflective events list. The analysis does not require all of the non-reflective events in order to run. After this phase, the analysis can be performed using real conditions in the field.

**Analysis Phase**

Analysis can be performed during installation, monitoring, or maintenance applications. During in-service monitoring, if the system detects an OTDR signature deviation, then analysis can be performed. During maintenance, if a failure occurs during traffic, then the fault location can be determined. In addition, the failure level is established, if possible. In both cases, the analysis method consists of a comparison between the current pattern (CP) and the reference pattern (RP).

The following sections discuss the different events that can occur during analysis.

**Optical Branch Identification**

During optical branch identification, two different possibilities can appear:

1. Fresnel reflection extinction or attenuation

   When analyzing the pattern comparison, if a fiber end reflective event deviation is detected, then the affected optical branch can be directly identified (Figure 3.28).
2. Attenuation deviation

When analyzing the pattern comparison, if an attenuation deviation along a fiber section is detected, then the affected fiber is the one that ends at the same distance that the deviation stops appearing (Figure 3.29). In this example, branch number six is affected.
Fault localization and attenuation estimation

After locating the distance where the deviation begins to appear, the location of the fault can be identified.

Eventually the fault can be associated with an event recorded in the reference table. Due to the mixing of multiple branches, fault attenuation cannot be directly calculated from the attenuation deviation between the two curves (Figure A.9).

![Figure A.9 Identifying the location and attenuation of a fault](image)

Using a pattern simulator, though, a virtual attenuator can be inserted into the affected branch and the attenuation can be increased until the same deviation from the reference pattern is achieved. This technique allows for an approximation of the attenuation level of the fault.

Accuracy and Limitations of the Method

Because this type of analysis involves measurements, it is necessary to point out its accuracy and limitations. As far as accuracy is concerned, the main source of error lies in the reality of the simulator models and data entry uncertainties. For example, optical splitters reverse and forward parameters may need to be entered. The fiber backscatter coefficient deviations add some uncertainty on the attenuation estimation and could be integrated into the theoretical formula.
Moreover, when P2MP networks are measured with a high level of splitting, the sensitivity depends on the amount of backscatter contribution that is lost. Therefore, sensitivity depends on the location of the fault compared to the fiber ends.

As in other OTDR measurements, the signal-to-noise ratio (the dynamic range, for example) can disturb the acquisition. If the network has a large total loss, then the dynamic range of the OTDR may not be enough to provide a backscatter trace along the entire link. Therefore, it is important to have enough dynamic range to go through a splitter with the OTDR.

If the network has branches of similar distances, then the OTDR may not differentiate the different events and branches. For this reason, new OTDRs are being developed, providing event spatial resolutions of 1 m or better.

In any case, if the concept of resolution is taken into consideration during the construction of a network to avoid reflection coincidence, then the maintenance of the network will be easier to perform.
<table>
<thead>
<tr>
<th>Term</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAL5</td>
<td>ATM Adaptation Layer 5</td>
</tr>
<tr>
<td>Absorption</td>
<td>In an optical fiber, loss of optical power resulting from conversion of power into heat.</td>
</tr>
<tr>
<td>AC-3</td>
<td>Audio compression standard adopted by ATSC and owned by Dolby.</td>
</tr>
<tr>
<td>Actual Down Rate</td>
<td>Bearer channel rate downstream</td>
</tr>
<tr>
<td>Actual Up Rate</td>
<td>Bearer channel rate upstream</td>
</tr>
<tr>
<td>ADC</td>
<td>Analog to Digital Converter</td>
</tr>
<tr>
<td>ADSL</td>
<td>Asymmetric Digital Subscriber Line. A digital subscriber line (DSL) technology in which the transmission of data from server to client is much faster than the transmission from the client to the server. The difference between ADSL1 and ADSL2/2+ is the standard that defines them: ADSL1: ITU G.992.2 ADSL2: ITU G.992.3 and .4 ADSL2+: ITU G.992.5 ADSL2</td>
</tr>
<tr>
<td>ADSL2+ (ITU G.992.3 and G.992.4)</td>
<td>Adds new features and functionality targeted at improving performance and interoperability, and adds support for new applications, services, and deployment scenarios. Among the changes are improvements in data rate and reach performance, rate adaptation, diagnostics, and stand-by mode.</td>
</tr>
<tr>
<td>ADSLAM</td>
<td>Advanced Digital Subscriber Line Access Multiplexer. Concentrates and multiplexes signals at the telephone service provider location to the broader wide area network.</td>
</tr>
<tr>
<td>APD</td>
<td>Avalanche Photodiode. Photodiode which operates in the avalanche mode, providing internal gain that is advantageous in reception.</td>
</tr>
<tr>
<td>ASCII</td>
<td>American Standard Code for Information Interchange. Defines a conversion from letters, numbers, and symbols (128) to a digital form of ones and zeros for use by computers and data communications.</td>
</tr>
<tr>
<td>ASI</td>
<td>Asynchronous Serial Interface. A standard DVB interface for a transport stream. See also DVB Asynchronous Serial Interface</td>
</tr>
<tr>
<td>ATM</td>
<td>Asynchronous Transfer Mode. A network standard that specifies fixed length cells to transmit voice, video, and data information. ATM is scalable in that it can operate at different transmission speeds such as 51, 100, 155, 622 Mb/s and beyond.</td>
</tr>
<tr>
<td>ATSC</td>
<td>Advanced Television Systems Committee. The digital broadcasting standard developed in the US and used in several countries worldwide. Digital broadcasting standard developed in North America</td>
</tr>
</tbody>
</table>
### Attenuation
The decrease in power of a signal. In optical fibers, loss of average optical power due to absorption, scattering and other radiation losses. It is generally expressed in dB without a negative sign.

### Attenuation Coefficient
The rate of optical power loss with respect to distance along the fiber, usually measured in decibels per kilometer (dB/km) at a specific wavelength. The lower the number, the better the fiber’s attenuation. Attenuation is specified at 850 and 1300nm for multimode fiber and 1310 and 1550nm for single mode fiber, over a temperature range of -60°C to +85°C.

### Attenuation Dead Zone
For a reflective or attenuating event, the region after the event where the displayed trace deviates from the undisturbed backscatter trace by more than a given vertical value _F_ (usually 0.5 dB or 0.1 dB). Telcordia specifies a reflectance of -30 dB, a loss of 0.1 dB and gives different locations. In general, the higher the reflected power sent back to the OTDR, the longer the dead zone. The attenuation dead zone depends on the pulse width, the reflectance, the loss, the displayed power level and the location. It usually indicates the minimum distance after an event where the backscatter trace can be measured.

### ATU-C
ADSL Transceiver Unit—Central Office

### ATU-R
ADSL Transceiver Unit—Remote

### ATV

### Auto-Negotiation
Procedure for adjusting line speeds and other communication parameters automatically between two devices during data transfer

### Backscattering
Portion of scattered light which returns in a direction generally reverse to the direction of propagation

### Bandwidth
In digital communications, this is the measure of the data rate, in bits per second, of a data flow in a given content such as a channel, or physical interface. In relation to a physical transmission medium, the difference, expressed in Hertz (Hz), between the highest and the lowest frequencies passing through the fiber or the frequency spectrum used to carry data over a copper medium.

*Note*: This term is often used to specify the bandwidth (MHz x km) of a multimode fiber.

### Bend Radius (minimum)
The amount a fiber can bend as measured by the radius of the bend, before increased loss or mechanical damage occurs.

### BER
Bit Error Ratio

### B-frame
A bidirectionally predicted picture, or a picture created by reference to preceding and subsequent pictures used in MPEG compression technologies.

### Bitrate
The rate at which a bit stream arrives at a given point typically measured in bits per second.

### Bits per Tone
Used in the context of DSL technologies, it is the number of data bits assigned to a given discrete multi-tone (DMT). Measurement and display allows analysis of the bits assigned per DMT; significant dips can reveal the presence of AC interference.
<p>| <strong>Block</strong> | A set of 8x8 pixels used during discrete cosine transform (DCT); used in MPEG compression technologies. |
| <strong>Bonding</strong> | In the context of DSL, the use of two copper loops to carry a DSL service, thus nearly doubling the effective bit rate. ADSL2 standards support the ATM Forum’s inverse multiplexing for ATM (IMA) standard (af-phy-0086.001) developed for traditional ATM architectures to bond two or more (typically up to 4) ADSL loop together at the ATM layer. Through IMA, separate copper pairs appear to be one loop. Bonding is also being defined for VDSL services. |
| <strong>Bouquet Association Table (BAT)</strong> | A DVB table that describes a set of services grouped together by a broadcaster and sold as a single entity. |
| <strong>Broadband</strong> | In data communications broadband has evolved into a marketing term referring to access circuits with data rates greater than dial-up services and not a technical term. It can denote technologies which use a wide range of frequencies for the signaling method used. In television networks it can denote a large number of channels carried by an antenna vs. one that carries a smaller number. |
| <strong>Broadcaster</strong> | A person or entity that provides a sequence of scheduled events or TV programs to the consumer. |
| <strong>Buffer Tube</strong> | A thermoplastic tube which is a component of fiber optic cables serving to segregate the fibers into groups and to mechanically decouple mechanical forces on the cable from the fibers by permitting the fibers to float in the tube. The tubes can be filled (in outdoor cables) or unfilled (in indoor cables). |
| <strong>Building Distributor</strong> | A distributor in which the building backbone cable(s) terminate(s) and at which connections to the campus backbone cable(s) may be made. |
| <strong>C Message Filter</strong> | A filter established for analysis of the voice frequency spectrum, typically 690-3000 Hz |
| <strong>Cabling System</strong> | The physical media components, such as connectors and cables, used in a local area data communications networks. |
| <strong>Campus Distributor</strong> | The distributor from which the campus backbone cabling emanates. |
| <strong>CA</strong> | Conditional Access |
| <strong>CAP</strong> | Carrierless Amplitude Phase. A line coding used for early versions of ADSL. |
| <strong>CAT</strong> | Conditional Access Table. This table identifies EMM streams with a unique PID value. The CAT is always found on PID 0x0001. Allows service providers to control subscriber access to programs and services |
| <strong>CATV</strong> | Community Access Television, otherwise known as Cable TV |
| <strong>CD</strong> | Chromatic Dispersion. A type of dispersion that causes broadening of input pulses along the length of the fiber. Chromatic dispersion is due to the different wavelengths of light traveling at different speeds through the fiber. It is at a minimum value at the fiber zero dispersion wavelength. |
| <strong>Cell</strong> | A fixed length unit of information. Most other data units can vary in length, but a cell is fixed in size. This helps cut down on network delays and variations in the delay through a network. Typically used in ATM technologies. |</p>
<table>
<thead>
<tr>
<th>Term</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>Channel</td>
<td>A digital medium that stores or transports an MPEG-2 transport stream. In the context of broadcast TV, a program channel.</td>
</tr>
<tr>
<td>Cladding</td>
<td>The glass layer surrounding the core of an optical fiber. The lower index of refraction of the cladding as compared to the core causes the light within the core to be totally internally reflected and remain in the core.</td>
</tr>
<tr>
<td>CO</td>
<td>Central office</td>
</tr>
<tr>
<td>Coating</td>
<td>An acrylate polymer material put on a fiber during the draw process to protect it from the environment and rough handling.</td>
</tr>
<tr>
<td>Compression</td>
<td>Reduction of the number of bits needed to represent an item of data.</td>
</tr>
<tr>
<td>Conditional Access</td>
<td>A system used to control viewer access to programming based on subscription.</td>
</tr>
<tr>
<td>Continuous Monitoring</td>
<td>The monitoring method that provides continuous real-time monitoring of all transport streams in a network.</td>
</tr>
<tr>
<td>Core</td>
<td>The central region of an optical fiber through which light is transmitted.</td>
</tr>
<tr>
<td>Coupling ratio/loss (Cr, Cl)</td>
<td>Ratio/loss of optical power from one output port to the total output power, expressed as a percent.</td>
</tr>
<tr>
<td>CPE</td>
<td>Customer Premises Equipment. Devices or equipment that the customer provides to interface with the service provider.</td>
</tr>
<tr>
<td>CRC</td>
<td>Cyclic Redundancy Check. 32-bit field used to verify the correctness of table data before decoding.</td>
</tr>
<tr>
<td>Crosstalk</td>
<td>The interference caused by signals on adjacent circuits in the same binder or cable group in a network.</td>
</tr>
<tr>
<td>Cutoff Wavelength</td>
<td>In single mode fiber, the shortest wavelength at which a single mode can be transmitted. Beyond this wavelength, several modes transmit simultaneously, and the fiber becomes multimode.</td>
</tr>
<tr>
<td>CW</td>
<td>Continuous Wave</td>
</tr>
<tr>
<td>D/A</td>
<td>Digital to Analog Converter</td>
</tr>
<tr>
<td>DBS</td>
<td>Direct Broadcasting Satellite or System</td>
</tr>
<tr>
<td>Dead Zone</td>
<td>In the context of a TDR, the time occupied by the launch of the measurement pulse resulting in a distance over which measurement of reflections cannot be made. Also known as a “blind spot.” The pulse width used directly affects the size of the dead zone.</td>
</tr>
<tr>
<td>Decoding Time Stamp (DTS)</td>
<td>Time stamp found in the PES packet header that indicates the time at which an audio or video frame is to be decoded.</td>
</tr>
<tr>
<td>Delay</td>
<td>The length of time required for bits to find their way through a network. Delay is a contributing measure of the carrying capacity of a link.</td>
</tr>
<tr>
<td>Dense Wavelength Division Multiplexing (DWDM)</td>
<td>Technique used to multiplex several signals on the same fiber within a narrow wavelength band.</td>
</tr>
</tbody>
</table>
Digital Television
A general term used to describe television that has been digitized. It can refer to Standard Definition Television or High Definition Television.

Digital Video Broadcasting (DVB) Project
A European consortium that has standardized digital TV broadcasting in Europe and in other countries.

Dispersion
The cause of bandwidth limitation in a fiber. The spreading (or broadening) of a light pulse as it spreads along a fiber. Major types are:

- modal dispersion caused by differential optical path lengths in a multimode fiber
- chromatic dispersion caused by a differential delay of various wavelengths of light passing through a fiber

Distributed Feedback (DFB) Laser
Laser with a Bragg reflection grating in the active region in order to suppress multiple longitudinal modes and enhance a single-longitudinal mode.

Distributor
The term used for the functions of a collection of components (e.g., patch panels, patch cords) used to connect cables.

DMT
Discrete Multi-Tone. A line coding used for DSL technologies.

Downlink
Communication link from a satellite to a terrestrial receiver.

Downstream Rate
The line rate, typically in bits per second, of data transfers in the direction from the network machine to the user's machine.

DS
Downstream; also called Dn.

DSL
Digital Subscriber Line. A generic name for a family of standards allowing high-speed data transfer over copper telephone lines.

DSLAM
Digital Subscriber Line Access Multiplexer. The line cards in the DSLAM terminate the network end of a DSL line.

DTE
Data Terminal Equipment

DTMF
Dual Tone Multi-Frequency. A voice-band tone based method of signaling, typically used for the PSTN voice network.

DTS
Decoding Time Stamp

DTV
Digital Television. A general term used to describe television that has been digitized. It can refer to standard-definition TV or high-definition TV

DVB
Digital Video Broadcasting.

DVB Asynchronous Serial Interface (ASI)
A standard coaxial DVB interface for MPEG-2 transport streams.

DVB Synchronous Parallel Interface (SPI)
A standard DVB interface for a video transport stream.

DVB-C
Digital Video Broadcasting-Cable. The DVB standard for broadcasting digital TV signals by cable. The RF spectrum in digital cable TV networks has a frequency range of (approx) 46MHz to 850MHz.
<table>
<thead>
<tr>
<th>Term</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>DVB-S</td>
<td>Digital Video Broadcasting-Satellite. The DVB standard for broadcasting digital TV signals via satellite.</td>
</tr>
<tr>
<td>DWDM</td>
<td>Dense Wave Division Multiplexing</td>
</tr>
<tr>
<td>Dynamic Range</td>
<td>IEC Dynamic Range (introduced by Bellcore, now Telcordia) The difference between the extrapolated point of the backscatter trace at the near end of the fiber (taken at the interception between the extrapolated trace and the power axis) and the upper level of the noise floor at or after the fiber end. The upper level of the noise is defined as the upper limit of a range which contains at least 98% of all noise data points. The dynamic range is expressed in decibels (dB). This measurement is performed for 180 seconds usually with largest pulse width of the OTDR.</td>
</tr>
<tr>
<td>RMS Dynamic Range</td>
<td>The difference between the extrapolated point of the backscatter trace at the near end of the fiber (taken at the intersection between the extrapolated trace and the power axis) and the RMS noise level.</td>
</tr>
<tr>
<td>Echo</td>
<td>The reflecting of a portion of a transmitted signal back toward its source due to a variety of reasons specific to the technology involved.</td>
</tr>
<tr>
<td>ECM</td>
<td>Entitlement Control Message. A message in the transport stream that carries the keys used by the decoder to descramble the audio, video, and data for a program.</td>
</tr>
<tr>
<td>EDFA</td>
<td>Erbium Doped Fiber Amplifier. Device which amplifies an optical signal without employing O/E and E/O conversions.</td>
</tr>
<tr>
<td>EIT (ATSC)</td>
<td>Event Information Table. An ATSC PSIP table that carries program guide information including titles and start times.</td>
</tr>
<tr>
<td>Electromagnetic Spectrum</td>
<td>Term used to describe the entire range of light radiation, from gamma rays to radio.</td>
</tr>
<tr>
<td>Elementary Stream (ES)</td>
<td>A bit stream that includes video, audio, or data. It represents the preliminary stage of the Packetized Elementary Stream (PES).</td>
</tr>
<tr>
<td>Encapsulation</td>
<td>The technique used by layered protocols in which a layer adds header information to the protocol data unit (PDU) from the layer above.</td>
</tr>
<tr>
<td>EMM</td>
<td>Entitlement Management Message. A message in the transport stream used to update the subscription options or pay-per-view rights for an individual subscriber or for a group of subscribers.</td>
</tr>
<tr>
<td>EOC</td>
<td>Embedded Operations Channel (between a VTU-O and VTU-R or ATU-C and ATU-R). Used to carry communications between the two devices regarding their operation.</td>
</tr>
<tr>
<td>EPG</td>
<td>Electronic Program Guide. Display that describes all programs and events available to the viewer. It functions as an interactive TV guide that allows users to view a schedule of available programming and select an event for viewing.</td>
</tr>
<tr>
<td>Ethernet</td>
<td>A local area network (LAN) communications scheme defined by IEEE 802.3. Ethernet has also become a generic term for data flows using Ethernet framing for the data units for WAN circuits.</td>
</tr>
<tr>
<td>ETR</td>
<td>ETSI Technical Report</td>
</tr>
<tr>
<td>Term</td>
<td>Definition</td>
</tr>
<tr>
<td>------</td>
<td>------------</td>
</tr>
<tr>
<td>ETR 290</td>
<td>ETSI recommendation regarding measurement of MPEG-2/DVB transport streams. Full number is ETSI TR 101 290 vx.x.x</td>
</tr>
<tr>
<td>ETSI</td>
<td>European Telecommunication Standard Institute</td>
</tr>
<tr>
<td>ETT</td>
<td>Extended Text Table</td>
</tr>
<tr>
<td>Event</td>
<td>A collection of elementary streams with a common time base and an associated start time and end time. An event is equivalent to the common industry usage of “television program.”</td>
</tr>
<tr>
<td>Event Dead Zone</td>
<td>Minimum distance on the trace, where two separate events can still be distinguished. The distance to each event can be measured, but the separate loss of each event cannot be measured. This parameter gives an indication of the minimum distance in order to distinguish between reflective events which occur in close proximity.</td>
</tr>
<tr>
<td>Event Information Table (EIT)</td>
<td>The ATSC PSIP is a table that carries event information including titles and start times for events on all the virtual channels within the transport stream. ATSC requires that each system contain at least 4 EIT tables, each representing a different 3-hour time block.</td>
</tr>
<tr>
<td>Extended Text Table (ETT)</td>
<td>The optional ATSC PSIP table that carries long descriptions of events and channels. There are two types of ETTs: Channel ETTs, which carry channel descriptions, and Event ETTs, which carry event descriptions.</td>
</tr>
<tr>
<td>FDM</td>
<td>Frequency Division Multiplexing</td>
</tr>
<tr>
<td>FEC</td>
<td>Forward Error Correction. A method for protecting the transport stream against error. FEC adds error control bits before RF modulation. With these bits, errors in the transport stream up to certain limits may be detected and corrected prior to decoding.</td>
</tr>
<tr>
<td>Ferrule</td>
<td>A mechanical fixture, generally a rigid tube, used to confine and align the polished or cleaved end of the fiber in a connector. Generally associated with fiber-optic connectors.</td>
</tr>
<tr>
<td>Fiber Distributed Data Interface (FDDI)</td>
<td>A standard for 100 Mbit/s fiber optic local area network.</td>
</tr>
<tr>
<td>Fiber Optic Span</td>
<td>A series of one or more terminated optical fiber elements which may contain complex passive components.</td>
</tr>
<tr>
<td>Floor Distributor</td>
<td>The distributor used to make connections between the horizontal cabling, other cabling subsystems, and active equipment.</td>
</tr>
<tr>
<td>Term</td>
<td>Definition</td>
</tr>
<tr>
<td>--------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Forward Error Correction (FEC)</td>
<td>A method for protecting the transport stream against error. FEC adds error control bits before RF modulation. With these bits, errors in the transport stream up to certain limits may be detected and corrected prior to decoding.</td>
</tr>
<tr>
<td>Frame</td>
<td>Lines of spatial information for a video signal. A generic term for organizing data into packets, the details specified in the particular technology associated with the term.</td>
</tr>
<tr>
<td>FTTB</td>
<td>Fiber to the Building</td>
</tr>
<tr>
<td>FTTC / FTTK</td>
<td>Fiber to the Curb / Kerb</td>
</tr>
<tr>
<td>FTTH</td>
<td>Fiber to the Home</td>
</tr>
<tr>
<td>FTTN</td>
<td>Fiber to the Node</td>
</tr>
<tr>
<td>FTTO</td>
<td>Fiber to the Office</td>
</tr>
<tr>
<td>Fusion Splice</td>
<td>A permanent joint accomplished by the application of localized heat sufficient to fuse or melt the ends of the optical fiber together, forming a continuous single fiber.</td>
</tr>
<tr>
<td>G Filter</td>
<td>A filter established for the analysis of the voice frequency spectrum, typically 20 KHz to 1.1 MHz</td>
</tr>
<tr>
<td>G.Lite</td>
<td>A lower-speed version of ADSL. It offers the advantage of being an ITU standard.</td>
</tr>
<tr>
<td>Gateway</td>
<td>A system which does translation from one native format to another.</td>
</tr>
<tr>
<td>GMQ</td>
<td>General Membership Query. An IGMP message asking the host to respond with all multicast channels that it wants to join.</td>
</tr>
<tr>
<td>GOP</td>
<td>Group Of Pictures. A set of pictures, or video frames usually 12-15 frames long, used for temporal encoding of MPEG-2 video defining the number of frames from one I-frame to the next.</td>
</tr>
<tr>
<td>Graded-Index Fiber</td>
<td>Fiber design in which the refractive index of the core is lower toward the outside of the core and increases toward the center with the peak at the centerline. This multimode fiber design reduces the time difference between the arrival of different modes, minimizing modal dispersion and maximizing bandwidth.</td>
</tr>
<tr>
<td>Group Index</td>
<td>The factor by which the speed of light in vacuum has to be divided to yield the propagation velocity of light pulses in the fiber.</td>
</tr>
<tr>
<td>Groups of Pictures (GOP)</td>
<td>Set of pictures, or video frames usually 12-15 frames long, used for temporal encoding of MPEG-2 video defining the number of frames from one I-frame to the next.</td>
</tr>
<tr>
<td>GSQ</td>
<td>Group Specific Query. An IGMP message used to learn if a particular group has any members on an attached network.</td>
</tr>
<tr>
<td>HDTV</td>
<td>High Definition Television. HDTV's resolution is approximately twice as high as that of Standard Definition Television (SDTV) for both horizontal and vertical dimensions. HDTV has an aspect ratio of 16x9 as compared to the 4x3 aspect ratio of SDTV</td>
</tr>
<tr>
<td>HPNA</td>
<td>Home Phone Networking Alliance. Organization formed to establish standards for home networking technology over coax and twisted pair copper phone lines within the home.</td>
</tr>
<tr>
<td>Hub</td>
<td>A point in a network where circuits are connected. As a device, a Hub connects circuits with very little intelligence providing a physical bus only and does not set up routing paths.</td>
</tr>
<tr>
<td><strong>I/F</strong></td>
<td>Interface</td>
</tr>
<tr>
<td><strong>ICMP</strong></td>
<td>Internet Configuration Message Protocol. The protocol used to handle errors and control messages at the IP layer. ICMP is part of the IP protocol</td>
</tr>
<tr>
<td><strong>IEC</strong></td>
<td>International Electrotechnical Commission.</td>
</tr>
<tr>
<td><strong>IEEE</strong></td>
<td>Institute of Electrical and Electronics Engineers</td>
</tr>
<tr>
<td><strong>I-frame</strong></td>
<td>Intra-coded frame. A picture encoded without reference to any other picture. I-frames provide a reference for predicted and bidirectionally predicted frames in a MPEG-2 compressed video stream.</td>
</tr>
<tr>
<td><strong>IGMP</strong></td>
<td>Internet Group Management Protocol. A signaling protocol typically used in Broadcast IP video service.</td>
</tr>
<tr>
<td><strong>IGMP Latency</strong></td>
<td>Elapsed time from sending an IGMP Join message until receipt of the first video packet of the target broadcast video program stream.</td>
</tr>
<tr>
<td><strong>IMA</strong></td>
<td>Inverse multiplexing for ATM as defined by the ATM Forum</td>
</tr>
<tr>
<td><strong>INP</strong></td>
<td>Impulse Noise Protection</td>
</tr>
<tr>
<td><strong>Index of Refraction</strong></td>
<td>See refractive index.</td>
</tr>
<tr>
<td><strong>Insertion Loss</strong></td>
<td>The increase in the total optical attenuation caused by the insertion of an optical component in the transmission path.</td>
</tr>
<tr>
<td><strong>Integrated Receiver Decoder (IRD)</strong></td>
<td>A receiver with an MPEG-2 decoder, also known as set top box (STB).</td>
</tr>
<tr>
<td><strong>Inter-Frame Prediction</strong></td>
<td>An MPEG-2 compression technique that periodically encodes a complete reference frame and then uses that frame to predict the preceding and following frames.</td>
</tr>
<tr>
<td><strong>Interoperability</strong></td>
<td>The ability of a system to use the parts or equipment of another system.</td>
</tr>
<tr>
<td><strong>IP</strong></td>
<td>Internet Protocol. The network layer protocol for the Internet Protocol suite.</td>
</tr>
<tr>
<td><strong>IP Address</strong></td>
<td>The 32-bit address (IPv4) assigned to hosts that want to participate in a TCP/IP Internet communication. A 128 bit address in IPv6.</td>
</tr>
<tr>
<td><strong>IP Video</strong></td>
<td>Packet based (IP) video service; sometimes called IPTV.</td>
</tr>
<tr>
<td><strong>IRD</strong></td>
<td>Integrated Receiver Decoder. A receiver with an MPEG-2 decoder, usually located in a headend. While the function is also in an STB, it is called a video decoder.</td>
</tr>
<tr>
<td><strong>ISDN</strong></td>
<td>Integrated Services Digital Network</td>
</tr>
<tr>
<td><strong>ISO</strong></td>
<td>International Standardization Organization</td>
</tr>
<tr>
<td><strong>ITU</strong></td>
<td>International Telecommunications Union (UIT)</td>
</tr>
<tr>
<td><strong>Jitter</strong></td>
<td>In video services, the variability in latency or delay in a data network of cells or data packet follows. Jitter can be measured in many ways, but is typically a measure of the packet delay variation at a given measurement point. In the context of MPEG-2 video, see PCR Jitter.</td>
</tr>
<tr>
<td><strong>Jumper</strong></td>
<td>A cable unit or cable element without connectors used to make a connection on a cross-connect.</td>
</tr>
<tr>
<td><strong>L2</strong></td>
<td>Low power mode where the transceiver modifies power based on traffic. L2 low-power mode enables statistical power savings at the ADSL transceiver unit in the central office (ATU-C) by rapidly entering and exiting low power mode based on Internet traffic running over the ADSL connection.</td>
</tr>
<tr>
<td><strong>L3</strong></td>
<td>Low power mode where each end enters sleep mode when the connection is used for extended periods. Enables overall power savings at both the AUTC and the remote ADSL transceiver unit (ATU-R).</td>
</tr>
<tr>
<td><strong>LAN</strong></td>
<td>Local Area Network. A geographically limited communications network intended for the local transport of voice, video, and data.</td>
</tr>
<tr>
<td><strong>LASER</strong></td>
<td>Light Amplificated by Stimulated Emission of Radiation. A device that produces monochromatic, coherent light through stimulated emission.</td>
</tr>
<tr>
<td><strong>Launch Fiber</strong></td>
<td>A length of fiber used to create an equilibrium modal distribution in multimode, and to measure the first connector of the network in both multimode and single mode systems.</td>
</tr>
<tr>
<td><strong>LED</strong></td>
<td>Light Emitting Diode. A semiconductor device used to transmit light into a fiber in response to an electrical signal. It typically has a broad spectral width. Its spectral width typically is 50 to 60 nm.</td>
</tr>
<tr>
<td><strong>Line Quality</strong></td>
<td>Equivalent to SNR. Referred to in this manner to be equivalent with the Cisco modem.</td>
</tr>
<tr>
<td><strong>Line Rate</strong></td>
<td>The speed by which data is transferred over a particular line type, expressed in bits per second (bps).</td>
</tr>
<tr>
<td><strong>Local Transmit Power</strong></td>
<td>Transmit power in dBm in the upstream direction. Note that this can be (and often is) negative.</td>
</tr>
<tr>
<td><strong>LOF</strong></td>
<td>Loss of Frame</td>
</tr>
<tr>
<td><strong>Loopback</strong></td>
<td>A diagnostic test mode that returns the transmitted signal back to the sending device after it has passed through a network or across a particular link. The returned signal can then be compared to the transmitted one. The discrepancy between the two aids in determining fault location.</td>
</tr>
<tr>
<td><strong>LOS</strong></td>
<td>Loss of Signal</td>
</tr>
<tr>
<td><strong>LPF</strong></td>
<td>Low Pass Filter</td>
</tr>
<tr>
<td><strong>MAC</strong></td>
<td>Media Access Control</td>
</tr>
<tr>
<td><strong>Macroblock</strong></td>
<td>A group of 16x16 pixels used for motion estimation in temporal encoding of MPEG-2 video.</td>
</tr>
<tr>
<td><strong>MDF</strong></td>
<td>Main Distribution Frame</td>
</tr>
<tr>
<td><strong>MDU</strong></td>
<td>Multi-Dwelling Unit</td>
</tr>
<tr>
<td><strong>Mechanical Splice</strong></td>
<td>A fiber splice accomplished by fixtures or materials rather than thermal fusion.</td>
</tr>
<tr>
<td><strong>MFD</strong></td>
<td>Mode Field Diameter. A parameter that expresses for a single mode fiber the section where the majority of the light energy passes. It can be expressed as the diameter of optical energy in the fiber. Because the MFD is greater than the core diameter, MFD effectively replaces core diameter in practice.</td>
</tr>
</tbody>
</table>
MGT  Master Guide Table. An ATSC PSIP table that defines sizes, types, PIDs, and version numbers for all of the relevant tables within the transport stream. The PID value for this table is 0x1FFB.

MIB  Management Information Base

Micro Bend  Small distortion of a fiber caused by external factors such as cabling.

MIP  Megaframe Initialization Packet. This packet is used by DVB-T to synchronize the transmitters in a multi-frequency network.

MOS Score  Mean Opinion Score. A numerical indication of the perceived quality of received voice or video service. MOS is expressed on a scale of 1 to 5, with 5 representing the highest perceived quality. The estimation is based upon algorithms for a specific service, such as analog voice, packet-based voice (VoIP), and video that analyze signals, and associated impairments which estimate what humans would score the same experience. In video services, both video and audio scoring is made.

MP@HL  Main Profile at High Level. MPEG-2 specifies different degrees of compression vs. quality. Of these, Main Profile at High Level is the most commonly used for HDTV.

MP@ML  Main Profile at Main Level. MPEG-2 specifies different degrees of compression vs. quality. Of these, Main Profile at Main Level is the most commonly used.

MPEG  Moving Picture Experts Group. A working group of the International Organization for Standardization (ISO). MPEG standards focus upon interoperability and generally are rules about interaction between systems and media files. The standards define a set of tools that can be used to conduct the compression and decoding functions. Defining only the interfaces and behavior of the system facilitates innovation in implementation. For example, as MPEG1 technology has evolved, the encoders and decoders have improved and offer greater efficiency and compression, offer faster performance, and deliver better-looking video, and better-sounding audio.

MPEG-1  An official standard for encoding audio and video adopted in 1992. The simplest of the MPEG standards, it describes a way to encode audio and video data streams, along with a way to decode them. While MPEG1 may seem outdated. The default size for an MPEG1 video is 352x240 at 30fps for NTSC (352x288 at 25fps for PAL sources). These were designed to give the correct 4:3 aspect ratio when displayed on the pixels of rectangular TV screens. For a computer-based viewing audience, 320x240 square pixels gives the same aspect ratio.

MPEG-2  ISO/IEC 13818 standard that defines MPEG-2 motion video and audio compression. It applies to all layers of transmission (video, audio and system).

The MPEG2 standard builds upon MPEG1 to extend it to handle the highest-quality video applications. MPEG2 is a common standard for digital video transmission at all parts of the distribution chain. Broadcast distribution equipment, digital cable headends, video DVDs, and satellite television all employ MPEG2; as do point-to-point streaming devices.

MPEG2 requires approximately 6 Mbps to provide the quality viewers are accustomed to seeing on DVDs, although data rates up to 19 Mbps are supported. 720x480 is the typical 4:3 default resolution, while 1920x1080 provides support for 16:9 high-definition television.
### Glossary

**MPEG-4**
ISO/IEC 14496 standard that defines MPEG-4 video and audio compression. It specifies simultaneous coding of synthetic and natural objects and sound. The same definitions are embodied in the ITU recommendation H.264.

While MPEG2 was designed to scale up to broadcast and high-definition quality and operating requirements, MPEG4 is designed to scale down to accommodate dial-up Internet bandwidths and to small devices including cell phones and PDAs while remaining viable for high-quality desktop streaming up to 1 Mbps. The AAC audio codec is the root of the MP4 file type, recently popularized by Apple iTunes, among others.

MPEG4 is much more than just an audio and video compression and decompression scheme. It is a container for media objects (images, text, video, animation, interactive elements like buttons and image-maps, etc) and a way to choreograph them into a synchronized, interactive presentation. MPEG4 also has standard interfaces to allow plugging in a DRM scheme called Intellectual Property Management and Protection (IPMP).

MPEG4 is still at the frontier of media technologies. The specification is extensive, and each vendor implements it in its own way by selecting from a set of tools which are important for the target application. Many MPEG4 tools include incompatibilities. The Internet Streaming Media Association (ISMA) is an industry consortium dedicated to interoperability among MPEG4 products and services. Essentially, any implementation that is ISMA-compliant will work with any other.

In MPEG-4 compression each channel of standard definition TV consumes about 1.5 to 2 Mbps, and enhanced versions such as Microsoft's Media Player 9 or VC-1 based approach can achieve about 1 Mbps per channel.

**MPEG-7**
Multimedia Content Description Interface that standardizes descriptions for searching, filtering, selecting and handling audiovisual content.

MPEG-7 is not a video or audio coding scheme or delivery mechanism. Officially called the Multimedia Content Description Interface, it is a set of rules and tools for describing content, and its focus is metadata. The ability to identify, search, index, and publish information about content is critical. Metadata schemes may include descriptions of semantic elements (shapes, colors, people, objects, motion, musical notation); catalog elements (copyright and access rules, parental ratings, title, location, date, etc); or structural elements (technical stats about the media).

**MPEG-21**
MPEG-21 focuses upon content distribution: control over content in all parts of the delivery chain and in all kinds of networks and devices. Its intent is to enable digital media content to interact with end devices to facilitate seamless access to media across a variety of devices and networks, regardless of bandwidth or client capability.

**MPTS**
Multiple Program Transport Stream. An MPEG-2 transport stream containing several programs that have been multiplexed into one stream.

**Multimode Fiber**
An optical fiber in which light travels in multiple modes

**Multiplex**
Combining two or more signals into a single bit stream that can be individually recovered.
**Multiplex** (n)  A digital stream including one or more services in a single physical channel. *(v)*—To sequentially incorporate several data streams into a single data stream in such a manner that each may later be recovered intact.

**Mux**  Multiplexer

**NIT**  Network Information Table. A DVB table that contains information about a network's orbit and transponder; always located on PID 0x0010. DVB specifies two types of NITs, the NIT Actual and the NIT Other. The NIT Actual is a mandatory table containing information about the physical parameters of the network actually being accessed. The NIT Other contains information about the physical parameters of other networks. The NIT Other is optional.

**NID**  Network Interface Device. The demarcation point where the public network ends and the private network within a home or office begins. All wiring and user devices inside the premises (such as a modem) are controlled and operated by the owner.

**Node**  A point of flexibility and/or interconnection within the fiber optic cabling system.

**NSC**  Number of Sub-Carriers

**NT**  Network termination

**NTR**  Network Timing Reference

**NTSC**  National TV Standard Committee

**Numerical Aperture**  The number that expresses light gathering capacity of a fiber related to the acceptance angle. The sine of 5% optical power angle (corresponding to -13 dB) is used to measure the Numerical Aperture.

**NVoD**  Near Video on Demand. This service allows for a single TV program to be broadcast simultaneously with a few minutes of difference in starting time. For example, a movie could be transmitted at 9:00, 9:15 and 9:30.

**OAM**  Operations, Administration and Maintenance

**OLR**  Online Reconfiguration

**ONU**  Optical Network Unit

**Optical Loss budget**  The amount of signal loss that can be tolerated in a system before errors occur.

**ORL**  Optical Return Loss. The ratio (expressed in dB) of the reflected power to the incident power from a fiber optic system or link ORL = \(-10 \log (P_r/P_i)\) or ORL = \(10 \log (P_i/P_r)\).

**OTDR**  Optical Time Domain Reflectometer. An instrument used to characterize a fiber optic link. Useful in estimating fiber link attenuation, attenuation coefficient, discrete reflections, splice/connector loss, and point defects, all as a function of fiber distance.

**P2MP**  Point-to-Multi-Point

**P2P**  Point-to-Point. A connection established between two specific locations or devices such as a hub and a workstation or between two buildings.
<table>
<thead>
<tr>
<th>Term</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet</td>
<td>See Transport Packet</td>
</tr>
<tr>
<td>PAL</td>
<td>Phase Alternating Line</td>
</tr>
<tr>
<td>PAP</td>
<td>Password Authentication Protocol</td>
</tr>
<tr>
<td>PAT</td>
<td>Program Association Table. An MPEG-2 table that lists all the programs contained in the transport stream and shows the PID value for the PMT associated with each program. The PAT is always found on PID 0x0000.</td>
</tr>
<tr>
<td>Patchcord</td>
<td>A cable permanently assembled at both ends with connector components principally for cross-connection within a patching facility.</td>
</tr>
<tr>
<td>Payload</td>
<td>All of the bytes in a packet that follow the packet header.</td>
</tr>
<tr>
<td>PCR</td>
<td>Program Clock Reference. A time stamp in the MPEG-2 transport stream that sets the timing in the decoder. The PCR is transmitted at least every 0.1 seconds</td>
</tr>
<tr>
<td>PES</td>
<td>Packetized Elementary Stream. This type of stream contains packets of undefined length. These packets may be comprised of video or audio data packets and ancillary data</td>
</tr>
<tr>
<td>PES Packet</td>
<td>In MPEG-2, the structure used to carry elementary stream data (audio and video). It consists of a header and a payload.</td>
</tr>
<tr>
<td>PES Packet Header</td>
<td>The leading bytes of a PES packet, which contain ancillary data for the elementary stream.</td>
</tr>
<tr>
<td>P-frame</td>
<td>In MPEG-2 the Predicted frame, or a picture coded using references to the nearest previous I- or P- picture.</td>
</tr>
<tr>
<td>Physical Layer</td>
<td>Handles transmission of raw bits over a communication channel. The physical layer deals with mechanical, electrical, and procedural interfaces.</td>
</tr>
<tr>
<td>PID</td>
<td>In MPEG technology, the Packet Identifier. This unique integer value identifies elements in the transport stream such as tables, data, or the audio for a specific program.</td>
</tr>
<tr>
<td>Pigtail</td>
<td>A short length of optical fiber permanently attached to a connector and intended to facilitate jointing between that connector and another optical fiber or component</td>
</tr>
<tr>
<td>PLL</td>
<td>Phase Lock Loop. This locks the decoder clock to the original system clock through the PCR.</td>
</tr>
<tr>
<td>PMS-TC</td>
<td>Physical Media Specific -Transmission Convergence</td>
</tr>
<tr>
<td>PMT</td>
<td>Program Map Table. An MPEG-2 table that specifies PID values for components of programs. It also references the packets that contain the PCR.</td>
</tr>
<tr>
<td>POTS</td>
<td>Plain Old Telephone Service; one of the services using the voice band; sometimes used as a descriptor for all voice band services.</td>
</tr>
<tr>
<td>PPP</td>
<td>Point-To-Point-Protocol. The successor to SLIP, PPP provides router-to-router and host-to-network connections over both synchronous and asynchronous circuits.</td>
</tr>
<tr>
<td>PSD</td>
<td>Power Spectral Density</td>
</tr>
<tr>
<td>Term</td>
<td>Description</td>
</tr>
<tr>
<td>------------</td>
<td>---------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>PSI</td>
<td>Program Specific Information. PSI refers to MPEG-2 table data necessary for the de-multiplexing of a transport stream and the regeneration of programs within the stream. PSI tables include PAT, CAT, PMT, and NIT.</td>
</tr>
<tr>
<td>PTS</td>
<td>Presentation Time Stamp. This stamp indicates the time at which an element in the transport stream must be presented to the viewer. PTSs for audio and video are transmitted at least every 0.7 seconds. The PTS is found in the PES header.</td>
</tr>
<tr>
<td>QAM</td>
<td>Quadrature Amplitude Modulation. This is a type of modulation for digital signals used in CATV transmission (DVB-C). Amplitude and phase of a carrier are modulated in order to carry information.</td>
</tr>
<tr>
<td>QPSK</td>
<td>Quadrature Phase Shift Keying. A type of modulation for digital signals used in satellite transmission (DVB-S).</td>
</tr>
<tr>
<td>RBN</td>
<td>Regional Broadband Network</td>
</tr>
<tr>
<td>Reed-Solomon Uncorrected Errors</td>
<td>Bit errors that were not corrected by the Reed-Solomon coding.</td>
</tr>
<tr>
<td>Reflectance</td>
<td>The ratio of reflected power to incident power of an event or connector ( R = 10 \log(Pr/Pi) ).</td>
</tr>
<tr>
<td>Refractive index</td>
<td>A property of light transmitting materials defined as the ratio of the velocity of light in vacuum ( (c) ) to its velocity in a given transmission medium ( (v) ). ( n = c/v )</td>
</tr>
<tr>
<td>Remote Address</td>
<td>The IP address of a remote server.</td>
</tr>
<tr>
<td>Remote Gain</td>
<td>Receiver gain in the upstream direction in dB. Note that this can be negative but usually is not.</td>
</tr>
<tr>
<td>Remote Transmit Power</td>
<td>Transmit power in dBm in the downstream direction. Note that this can be (and often is) negative.</td>
</tr>
<tr>
<td>Repeater</td>
<td>A device used to regenerate an optical signal to allow an increase in the system length.</td>
</tr>
<tr>
<td>RFC</td>
<td>Request for Comments</td>
</tr>
<tr>
<td>RFI</td>
<td>Radio Frequency Interference</td>
</tr>
<tr>
<td>Route</td>
<td>The path that network traffic takes from its source to its destination. The route a datagram may follow can include many gateways and many physical networks.</td>
</tr>
<tr>
<td>RS</td>
<td>Reed-Solomon Protection Code. This refers to the 16 bytes of error control code that can be added to every transport packet during modulation.</td>
</tr>
<tr>
<td>RST</td>
<td>Running Status Table. A DVB-SI table that indicates a change of scheduling information for one or more events. It saves broadcasters from having to retransmit the corresponding EIT. This table is particularly useful if events are running late. It is located on PID Oxo013.</td>
</tr>
<tr>
<td>Scattering</td>
<td>A property that causes light to deflect out of the core area of the fiber, thereby contributing to attenuation.</td>
</tr>
</tbody>
</table>
**SDT**

Service Description Table. This DVB SI table describes the characteristics of available services. It is located on PID 0x0011. Two types of SDTs are specified by DVB, the SDT Actual and the SDT Other. The SDT Actual is a mandatory table that describes the services within the transport stream currently being accessed. The SDT Other describes the services contained in other transport streams in the system.

**SDTV**

Standard Definition Television. SDTV refers to television that has a quality equivalent to NTSC or PAL.

**Section**

A syntactic structure used for mapping PSI/SI/PSIP tables into transport packets of 188 bytes.

**Service**

A collection of one or more events under the control of a single broadcaster. Also known as a Program.

**SI**

Service Information. A DVB protocol that specifies all data required by the receiver to de-multiplex and decode the programs and services in the transport stream. Mandatory DVB SI tables include TDT, NIT, SDT, and EIT.

**Single Mode Fiber**

An optical wave guide (or fiber) in which the signal travels in one mode.

**SLA**

Service Level Agreement

**SMPTE**

Society of Motion Picture and Television Engineers

**SNG**

Satellite News Gathering. Refers to the retransmission of events using mobile equipment and satellite transmission.

**SNMP**


**SNR**

Signal to Noise Ratio. The ratio of the received optical signal power divided by the RMS noise floor for the detector.

**Softswitch**

Device in a telephone network which connects one phone line to another entirely through software

**SONET**

Synchronous Optical NETwork. A transport interface that enables the public network to carry services. SONET is the North American optical fiber standard that supports transmission rates that start at 51.84 Mb/s and reach to 2.488 Gb/s.

**SPI**

Synchronous Parallel Interface. A standard DVB interface for a transport stream.

**Splice**

A permanent junction between optical fibers.

**Splitter (Optical)**

A passive device which devises optical power among several output fibers from a common input.

**Splitter**

A device used in DSL to allow continued use of POTS or ISDN services while at the same time supporting a DSL technology on the same copper pair or loop.

**SPTS**

Single Program Transport Stream. An MPEG-2 transport stream that contains one unique program.

**SRA**

Seamless Rate Adaptation. ADSL2 feature enabling the system to change data rate while in operation without any service interruption or bit errors.
<table>
<thead>
<tr>
<th>Abbreviation</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>ST</td>
<td>Stuffing Table. An optional DVB-SI table that authorizes the replacement of complete tables due to invalidation at a delivery system boundary such as a cable headend. This table is located on PID 0x0014.</td>
</tr>
<tr>
<td>STB</td>
<td>Set top box. A digital TV receiver (IRD).</td>
</tr>
<tr>
<td>STD</td>
<td>System Target Decoder. A hypothetical reference model of the decoding process defined by MPEG-2.</td>
</tr>
<tr>
<td>Step-index fiber</td>
<td>A fiber whose index of refraction (n) changes sharply at the interfaces of its core and cladding.</td>
</tr>
<tr>
<td>STT</td>
<td>System Time Table. An ATSC PSIP table that carries time information needed for any application requiring schedule synchronization. It provides the current date and time of day and is located on PID 0x1FFB.</td>
</tr>
<tr>
<td>Sync Bits</td>
<td>Synchronizing bits (more properly bytes or characters) used in synchronous transmission to maintain synchronization between transmitter and receiver.</td>
</tr>
<tr>
<td>Syncs</td>
<td>Number of times the unit has trained.</td>
</tr>
<tr>
<td>Table</td>
<td>A format in which service information is transmitted. Tables are further divided into subtables, then into sections, before transmission. Several types of tables are specified by MPEG, DVB, and ATSC. Refer to the JDSU MPEG Pocket Guide for more information on types of Service Information tables and their functions.</td>
</tr>
<tr>
<td>Telecommunications Closet</td>
<td>An enclosed space for housing telecommunications equipment, cable terminations, and cross-connect cabling. The telecommunications closet is a recognized cross-connect between the backbone and horizontal cabling subsystems.</td>
</tr>
<tr>
<td>Telecommunications Closet</td>
<td>A fixed connecting device where the horizontal cable terminates. The telecommunications outlet provides the interface to work area cabling.</td>
</tr>
<tr>
<td>Time-stamp</td>
<td>An indication of the time at which a specific action must occur in order to ensure proper decoding and presentation.</td>
</tr>
<tr>
<td>TOT</td>
<td>Time Offset Table. An optional DVB SI table that supplies the UTC time and date and shows the difference between UTC time and the local time for various geographical regions. The PID for this table is 0x0014.</td>
</tr>
<tr>
<td>TP</td>
<td>Twisted Pair</td>
</tr>
<tr>
<td>TR 101 290</td>
<td>ETSI recommendation priorities for monitoring MPEG-2/DVB transport streams.</td>
</tr>
<tr>
<td>Transmitter</td>
<td>An electronic package used to convert a signal carrying electronic information to a corresponding optical signal for transmission by fiber. The transmitter can be a light emitting diode (LED), laser diode, or vertical cavity surface emitting laser (VCSEL).</td>
</tr>
<tr>
<td>Transponder</td>
<td>Transmitter and (re)sponder. Refers to the equipment inside a satellite that receives and re-sends information.</td>
</tr>
<tr>
<td>Transport Packet</td>
<td>188-byte packet of information in a transport stream. Each packet contains a header and a payload.</td>
</tr>
<tr>
<td><strong>Term</strong></td>
<td><strong>Definition</strong></td>
</tr>
<tr>
<td>----------</td>
<td>----------------</td>
</tr>
<tr>
<td>Transport Stream</td>
<td>A stream of 188-byte transport packets that contains audio, video, and data belonging to one or several programs.</td>
</tr>
<tr>
<td>Tx</td>
<td>Transmitter</td>
</tr>
<tr>
<td>Uplink</td>
<td>Communication link from earth to a satellite</td>
</tr>
<tr>
<td>Upstream Rate</td>
<td>The line rate for message or data transfer from the source machine to a destination machine on the network.</td>
</tr>
<tr>
<td>US</td>
<td>Upstream</td>
</tr>
<tr>
<td>UTC</td>
<td>Universal Time Co-ordinated</td>
</tr>
<tr>
<td>VC</td>
<td>Virtual Channel</td>
</tr>
<tr>
<td>VCI</td>
<td>Virtual channel identifier. A unique numerical tag as defined by a 16 bit field in the ATM cell header that identifies a virtual channel, over which the cell is to travel.</td>
</tr>
<tr>
<td>VCT</td>
<td>Virtual Channel Table. ATSC table that describes a set of one or more channels or services. Information in the table includes major and minor numbers, short channel name, and information for navigation and tuning. There are two types of VCTs, the TVCT for terrestrial systems and the CVCT for cable systems.</td>
</tr>
<tr>
<td>VDSL</td>
<td>Very High Speed Digital Subscriber Line. Enhancement to DSL that permits transmission of data up to 200mb/sec or more on a twisted copper pair utilizing a bandwidth or frequency range up to 30MHz.</td>
</tr>
<tr>
<td>VFL</td>
<td>Visual Fault Locator. The visual fault locator is a visual light source used to locate breaks or points of excess loss in fiber cable. The common wavelengths are 635 nm, 650 nm and 670 nm.</td>
</tr>
<tr>
<td>VLC</td>
<td>Variable Length Coding. This refers to a data compression method (Huffmann).</td>
</tr>
<tr>
<td>VoD</td>
<td>Video on Demand, also VOD</td>
</tr>
<tr>
<td>VoIP</td>
<td>Voice over Internet Protocol</td>
</tr>
<tr>
<td>VPI</td>
<td>Virtual Path Identifier. An eight bit field in the ATM cell header which indicates the virtual path over which the cell should be routed.</td>
</tr>
<tr>
<td>VSB</td>
<td>Vestigial Sideband Modulation. The terrestrial modulation method used in the ATSC. It can have either 8 (8 VSB) or 16 (16 VSB) discrete amplitude levels.</td>
</tr>
<tr>
<td>WAN</td>
<td>Wide Area Network. A data communications network that spans any distance and is usually provided by a public carrier.</td>
</tr>
<tr>
<td>WDM</td>
<td>Wavelength Division Multiplexer. Passive fiber optic component which combines optical channels on different wavelengths.</td>
</tr>
<tr>
<td>ZAP Time</td>
<td>Elapsed time from sending an IGMP Join message until receipt of the first video packet of the target broadcast video program stream.</td>
</tr>
</tbody>
</table>
Notes:
All statements, technical information and recommendations related to the products herein are based upon information believed to be reliable or accurate. However, the accuracy or completeness thereof is not guaranteed, and no responsibility is assumed for any inaccuracies. The user assumes all risks and liability whatsoever in connection with the use of a product or its application. JDSU reserves the right to change at any time without notice the design, specifications, function, fit or form of its products described herein, including withdrawal at any time of a product offered for sale herein. JDSU makes no representations that the products herein are free from any intellectual property claims of others. Please contact JDSU for more information. JDSU and the JDSU logo are trademarks of JDS Uniphase Corporation. Other trademarks are the property of their respective holders. ©2007 JDS Uniphase Corporation. All rights reserved. 30149199 000 0907  TRIPLEPLAY.BK.TM.AE