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Study of the China PTN Industry Standard and Its Key Issues

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Abstract:

In the area of carrier technologies, Packet Transport Networking (PTN) continues to attract significant research interest. Although international standards are not yet complete, China is leading the way in developing PTN equipment and network applications. *General Technical Requirements for Chinese Packet Transport Networks* is the PTN industry standard in China defined by China Communications Standards Association (CCSA). This article introduces the background of PTN technology and standards, then illustrates four important aspects: PTN network architecture, multi-services bearing and data transfer function, PTN network protection, and OAM architecture and functional requirements. Taking the profit motives of Chinese carriers and vendors into consideration, some key issues in PTN technology selection, network applications, and future evolution are discussed.

1 Implementation of Packet Transport Networks

Packet Transport Networking (PTN) is Connection-Oriented Packet-Switched (CO-PS) multi-service unified transport technology. PTN is sufficient for bearing carrier-class Ethernet services, and features standardized services, high reliability, flexible scalability, strict Quality of Service (QoS), and improved Operation, Administration and Maintenance (OAM). Moreover, it supports legacy Time Division Multiplexing (TDM) and Asynchronous Transfer Mode (ATM) services, and inherits management functions such as graphical interface and end-to-end configuration of Synchronous Digital Hierarchy (SDH) network management. PTN is currently applied in Metropolitan Area Networks (MANs) that bear services with QoS requirements—such as mobile backhaul and enterprise

private lines/networks. The MANs of Chinese operators are being transitioned away from TDM to PTN packet switching.

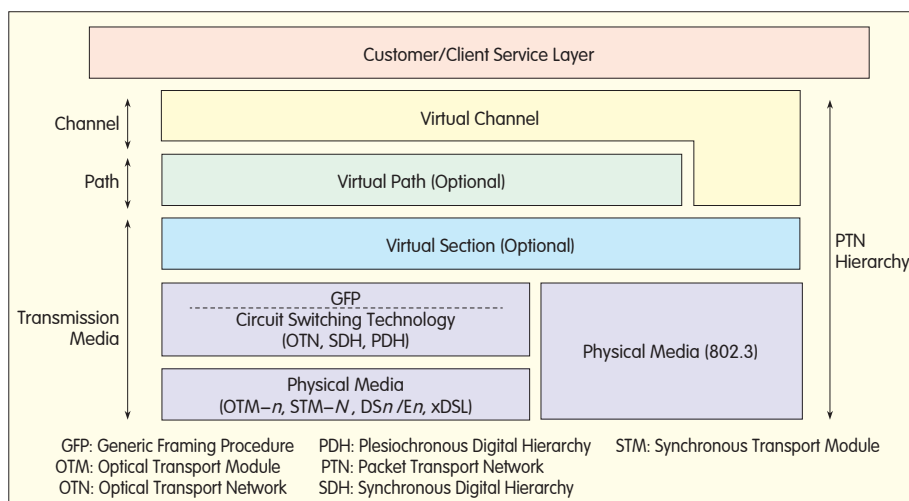
PTN has two implementation technologies: MPLS Transport Profile (MPLS-TP) and Backbone Bridge Traffic Engineering (PBB-TE). Based on Internet Protocol/Multiprotocol Label Switching (IP/MPLS), MPLS-TP discards hop-by-hop forwarding according to IP addresses and establishes transport paths that are independent of the control plane. The connection-oriented end-to-end label forwarding capability of MPLS is retained, and non connection-oriented and non end-to-end features—including Penultimate Hop Popping (PHP), Label Switched Path (LSP) Merge, and Equal-Cost Multi-Path Routing (ECMP)—are eliminated. MPLS-TP therefore supports definite end-to-end transport paths, PTN-style network protection, and OAM capability.

The other implementation technology is Backbone Bridge Traffic Engineering (PBB-TE) specified by IEEE

802.1Qay—a connection-oriented Ethernet transport technology. Also called Media Access Control (MAC) in MAC, Provider Backbone Bridge (PBB) represents an improvement on IEEE 802.1ah, eliminating connectionless Ethernet characteristics such as MAC address learning, spanning tree protocol, and flooding. Traffic Engineering (TE) is also incorporated to enhance QoS. PBB-TE currently supports Point-to-Point (P2P) and Point-to-Multipoint (P2MP) connection service transport and linear protection. However, it does not yet support multipoint-to-multipoint connection-oriented service transport and ring protection.

MPLS-TP and PBB-TE differ in terms of data forwarding, multi-service bearing, network protection, and OAM mechanism^[1-2]. As far as standardization, vendor products, operator applications, and the broader industrial chain are concerned, MPLS-TP seems to have a more promising future than PBB-TE. Therefore, MPLS-TP is currently treated as the leading PTN (4) PTN

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▲ Figure 1. PTN layers.

implementation technology.

2 Main Content of the China PTN Industry Standard

General Technical Requirements for Chinese Packet Transport Networks specifies PTN architecture, multi-service bearing and data forwarding functions, network protection function and performance, OAM capability, QoS, packet synchronization, network interface, network performance, network management, and functions of the control plane. The following introduces main features of PTN.

2.1 PTN Architecture

PTN has the following technical characteristics:

- CO-PS, packet switched kernel, and multi-service bearing;
- It is strictly connection-oriented, and has long-lasting connection that supports manual configuration;
- A Reliable network protection mechanism, capable in various PTN network layers and topologies ;
- Different QoS guarantees for different services;
- Sound OAM fault management and performance management;
- Label-based packet forwarding. Encapsulation, transport, and processing of OAM messages are independent of IP encapsulation and

processing. The protection mechanism does not rely on IP packets;

- Support for bidirectional and unidirectional point-to-point transport paths, as well as TE control of P2P and P2MP transport paths.

As shown in Figure 1^[3], PTN has three layers: Virtual Channel (VC) layer, Virtual Path (VP) layer, and Virtual Section (VS) layer. At the bottom of PTN is the physical media layer, implemented by IEEE802.3 Ethernet/SDH and Optical Transport Network (OTN) Connection-Oriented Circuit-Switched (CO-CS) technologies. In a MPLS-TP system, a Pseudo Wire (PW) layer is equivalent to PTN VC, and the LSP layer is equivalent to PTN VP.

2.2 PTN Multi-Service Bearing and Data Forwarding Functions

The client layer in MPLS-TP architecture^[4] includes PW-based emulation services, services with MPLS labels, and IP services. The standards for this bearing architecture specify PW-based emulation services including TDM services, Ethernet L2 services, and ATM services. However, the functional requirements of a peer-to-peer IP and MPLS service bearing model are still being studied. All service bearing in PTN is connection oriented; a non connection-based mechanism is not included in the standard.

MPLS-TP PTN should meet the

following functional requirements for bearing PW-based emulation services:

(1) Pseudo Wire Emulation Edge-to-Edge (PWE3) encapsulation should be used uniformly to bear emulation services. The PWE3 control word should follow RFC4385.

(2) TDM service emulation and transport should be supported. Specific requirements are as follows:

- TDM service bearing in the Structure-Agnostic TDM over Packet (SAToP) mode should be supported, with the PWE3 encapsulation and control word following RFC4553. The SAToP mode can be applied to any signal structure of TDM circuit emulation.

- Support for TDM service bearing in Circuit Emulation Services over Packet Switch Network (CESoPSN) mode is optional, with the PWE3 encapsulation and control word following RFC5086. CESoPSN is mainly used for Nx64kb/s signal structures of TDM circuit emulation, and is capable of saving bandwidth.

- Support for TDM service bearing in the SDH Circuit Emulation over Packet (CEP) mode is also optional, with PWE3 encapsulation and control word following RFC4842. CEP is mainly used for VC12/VC-4 signal structures of TDM circuit emulation.

(3) Ethernet L2 service emulation and transport should be supported. The specific requirements are as follows:

- Ethernet Line (E-Line), Ethernet Local Area Network (E-LAN), and Ethernet Tree (E-Tree) services are supported, and should follow the related specifications of International Telecommunication Union's Telecommunication Standardization Sector (ITU-T) and Metro Ethernet Forum (MEF).

- Service and VC/CP binding implemented through ports or port+Virtual Local Area Network (VLAN) should be supported.

- Network management-based static configuration of Ethernet services should be supported. Signalling based dynamic configuration of Ethernet services is optional.

- The PWE3 encapsulation and control word should follow RFC4448.

(4) PTN support for ATM service emulation and transport is optional, and the PWE3 encapsulation and control word should follow RFC4717. PTN supports both 1:1 and $N:1$ Virtual Channel Connection/Virtual Path Connection (VCC/VPC) encapsulation, as well as the establishment of unidirectional and bidirectional point-to-multipoint VCC/VPC. Inverse Multiplexing over ATM (IMA) group processing is optional.

MPLS-TP based PTN should meet the following data forwarding requirements:

(1) The MPLS-TP data forwarding mechanism is a subset of the MPLS data forwarding system (RFC3031), and should meet the transport requirements specified in RFC5654. MPLS-TP does not use hop-by-hop forwarding based on IP addresses, and does not support the merged function of ECMP, PHP, and LSP.

(2) MPLS-TP label stacking should follow RFC3032 and RFC5462. The MPLS Time-to-Live (TTL) processing mechanism should follow RFC3443. (Both VC and VP Transmission Convergence (TC) and TTL should adopt pipe models.)

(3) The range of PWs and LSP labels should be $1-1,048,575$ (eg. $2^{20}-1$), from which 0-15 are reserved labels (used or reserved for PTN OAM. RFC3032 defines 4 reserved label values).

(4) MPLS-TP should support the following label switching functions:

Correct pushing of PWs and LSP labels by the source node, enabling multiple PWs to be multiplexed into one LSP;

Correct popping of PWs and LSP labels by the sink node;

Supporting Single-Segment Pseudo Wires (SS-PW) and Multi-Segment Pseudo Wires (MS-PW) switching. SS-PW architecture should follow RFC3985. MS-PW switches PWs of different LSPs into the same LSP.

2.3 PTN Network Protection Functional Requirements

PTN supports three protection types:

(1) Protection within PTN

- Linear protection within PTN

including unidirectional/bidirectional 1+1 path protection, bidirectional 1:1 or 1: N ($N>1$) path protection, unidirectional/bidirectional 1+1 SNC/S protection, and bidirectional 1:1 SNC/S protection. At minimum, a bidirectional 1:1 protection mechanism should be supported.

- Ring protection within PTN including Wrapping and Steering. At minimum, one of these two protection mechanisms should be supported.

(2) Access Link Protection of PTN and Other Networks

- 1+1 or 1: N protection of TDM/ATM access links;

- Protection of GE/10GE access links, eg. Link Aggregation (LAG).

(3) Dual-Homed Protection

Protection within PTN and access link protection are combined for end-to-end service protection when the access link or PTN access node fails. This mechanism is still being studied.

Protection within PTN should meet the following generic functional requirements:

(1) PTN protection switching should support triggering of link/node fault conditions and external commands from the network management system, as well as priority handling of various switching requests. The types of faults triggered include Signal Failures (SF), intermediate node failures of both physical links and VP/VC, and Signal Degrade (SD). External commands that are triggered result in network management orders such as "locked protection", "forced switching", "manual switching", and "clear command".

(2) Protection switching mode: This supports both single-end and dual-end switching types. It should support configuration of "non-revertive operation" or "revertive operation" mode, as well as initiation of Wait to Restore (WTR) function and WTR time setting.

(3) Protection switching time: The service disruption time caused by protection switching—except SD-triggered protection switching—should not exceed 50ms when the delay time is set at 0.

(4) Hold off time setting: When the

PTN bottom network (WDM and OTN) is configured with protection, PTN network protection should support a hold off time, which can be set at 50 ms or 100 ms.

2.4 PTN OAM Architecture and Functional Requirements

PTN OAM includes the OAM mechanism within PTN, service layer OAM, and access link layer OAM.

The OAM mechanism within PTN is divided into alarm-related OAM, performance-related OAM, and other OAM. The OAM requirements of the VC, VP, and VS layers are shown in Table 1. Proactive OAM periodically or continuously reports fault and error code performance, whereas on-demand OAM is manually initiated for the purposes of fault diagnosis and positioning.

3 Key Issues Associated with the China PTN Industry Standard

In the drafting of *General Technical Requirements for Chinese Packet Transport Networks*, selection of technologies for the standard, PTN applications, and sustainable development of PTN were disputed. After extensive discussions, some issues have been solved, but others require follow-up study.

(1) MPLS-TP OAM Implementation Mechanism and Encapsulation Format

There are three available options: T-MPLS G.8114, G-ACh^[5]+Y.1731 OAM PDU^[6], and IETF BFD expansion^[7]+new OAM tools. All Chinese members of the standardization group have agreed to adopt G-ACh+Y.1731 OAM PDU to preserve existing benefits of Chinese operators and vendors, and to facilitate software upgrade in the future. All members wish to promote their preference as the international MPLS-TP standard. However, selecting G-ACh+Y.1731 OAM PDU as the MPLS-TP standard is difficult because the group is led by non-Chinese giants Cisco and Juniper.

(2) Implementation Mechanism of PTN Ring Protection

IETF has approved ring protection

▼ Table 1. OAM requirements in PTN

Type	Function	VS OAM	VP OAM	VC OAM
Proactive OAM	Fault Management	Continuity Check and Connectivity Verification	Required	Required
		Remote Defect Indication	Required	Required
		Alarm Indication Signal	Not Applicable	Required
		Locking	Required	Required
		Client Signal Fault	Not Applicable	Required
	Performance Monitoring	Packet Loss Measurement	Optional	Optional
On Demand OAM	Fault Management and Location	Loopback	Required	Required
		Link Trace	Not Applicable	Required
		Testing	Optional	Required
		Locking	Optional	Required
	Performance Monitoring	Packet Loss Measurement	Optional	Required
		Packet Delay Measurement	Optional	Required
Others	Automatic Protection Switching		Required	Required
	Management/Signaling Control Path		Not Applicable	Required
	Clock		Optional	Not Applicable
	Operator's Custom Functions		Not Applicable	Optional
	Functions Used for Testing		Not Applicable	Optional
OAM: Operation, Administration and Maintenance		VP: Virtual Path		
VC: Virtual Channel		VS: Virtual Section		

requirements, with implementation mechanisms including MPLS-TE Fast Re-route (FRR) applied in ring topology^[8], IEEE multi-segment protection^[9], and ITU-T Wrapping/Steering ring protection^[10]. In consideration of Chinese vendor products and operator applications, the standard adopts ITU-T Wrapping/Steering ring protection. Prior to selection, three technologies were deeply analyzed in terms of service configuration, bandwidth sharing, OAM, and cross-ring protection. The impacts on network operation and maintenance were also taken into consideration. Follow-up standardization work will involve specific mechanisms for point-to-multipoint service protection and cross-ring protection.

(3) PTN and Multiservice Transport Platform (MSTP) Mixed Networking and Interworking Requirements

The mission of PTN is to gradually replace SDH-based MSTP. Therefore, in network deployment, operators must

deal with PTN and MSTP mixed networking, and interworking of two networks. Such a scenario might include PTN convergence ring interworked with the MSTP access ring. However, specifications of the related standard are still open and being debated.

(4) Requirements of PTN Supporting IP/MPLS L3 Functions

With the trend towards all-service operation and Long Term Evolution (LTE) mobile backhaul bearing, should PTN support certain IP/MPLS L3 functions? If so, how should it cooperate with the existing router network? These questions will be studied after operators' specific requirements are made clear. This is an open issue in PTN standardization.

PTN is developing, competing, and converging with IP/MPLS technology. It has so far been applied in IP-based 3G mobile backhaul networks. The next two years will be key in the standardization and industrial application of PTN technology. The PTN

industrial chain is expected to continue developing.

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Biographies

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Standardization Progress of Packet Transport Networks

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Abstract:

Global operators are in the process of constructing next generation IP packet-switched networks, with the view of moving towards full IP-based networks. As data services increase, improvements in bandwidth, Quality of Service (QoS), Operation Administration and Maintenance (OAM), and network reliability are required. Convergence of network data transmission and network communications technology has been driven by converged IP services (including video, voice, and data), and unified multi-service load demand. Packet Transport Networking (PTN) has come into being to meet these requirements, and PTN Standards are being developing rapidly.

There are two leading Packet Transport Network (PTN) technologies: Provider Backbone Bridge Traffic Engineering (PBB-TE) and MPLS Transport Profile (MPLS-TP). The former adapts Ethernet technology to carrier-class transport networks, while the latter is developed from transport networks and Multiprotocol Label Switching (MPLS).

PBB-TE is derived from Provider Backbone Bridge (PBB) technology. PBB-TE eliminates Media Access Control (MAC) address learning, spanning tree protocol and flooding. It has connection-oriented features and improved network scalability.

MPLS-TP is a connection-oriented packet-switched technology capable of Traffic Engineering (TE) and managing network resource use. The MPLS-TP data forwarding plane is a subset of MPLS defined by the Internet Engineering Task Force (IETF). MPLS-TP has MPLS features, but also meets transport requirements. It has a reliable protection mechanism, improved Quality of Service (QoS), and enables scalable Operation Administration and Maintenance (OAM).

In terms of industrialization,

MPLS-TP has won more support from vendors and operators than PBB-TE; therefore, it is more likely to be deployed in large-scale future networks^[1-4].

1 PBB-TE

PBB-TE (IEEE 802.1Qay) is based on PBB (IEEE 802.1ah), which is also called MAC-in-MAC. Based on MAC stacking technology, it encapsulates user MAC within the operator's MAC, enhancing Ethernet scalability and service security by using double encapsulation to isolate user traffic. The critical part of PBB involves the introduction of a 24 bit Service Instance Tag (I-TAG) into the MAC-in-MAC encapsulation.

PBB-TE provides Ethernet with connection-oriented forwarding, which enables service providers to offer dedicated Ethernet links and to guarantee performance. In other words, PBB-TE provides strict QoS for a Metro Ethernet. Certain traditional Ethernet functions such as MAC address learning, broadcasting, spanning tree protocol, and multicasting are eliminated. PBB-TE is therefore

connection-oriented and avoids the flooding of broadcasting packets. This enables connection configuration through network management system or control protocols so that carrier class protection switching, OAM, QoS, and TE are achieved.

PBB-TE uses IEEE 802.1ag as the signaling protocol for PBB-TE tunneling, and continuously monitors tunnel states in the network. If the working tunnel fails, services are automatically transferred to a pre-established backup circuit. This increases network flexibility.

IEEE approved the establishment and authorization application of the PBB-TE program, setting up the 802.1Qay task group in March 2007. The group began standardization of the PBB-TE (IEEE 802.1Qay) data plane in April 2007, and voted through 802.1Qay Draft 1.1 in January 2008. In January 2009, IEEE introduced the draft version of 802.1Qay D5.0 and promoters voted on it. IEEE 802.1Qay-2009 was officially released in August 2009.

Although it has been officially released, IEEE 802.1Qay only specifies the end-to-end protection of Traffic

Engineering Service Instance (TESI). In order to improve reliability and flexibility, several companies applied a PBB-TE segment protection program (a kind of local protection program) and, in a plenary session held in July 2008, debated whether local protection should be implemented in infrastructure or TESI.

The conclusion of that session was to make infrastructure protection for available programs, and to make improvements of TESI local protection as later required. After a year-long discussion, application of the PBB-TE segment protection program (program number IEEE 802.1Qbf) was finally approved at the plenary session in July 2009. The 802.1 working group released the first draft of IEEE 802.1Qbf D0.0 and carried out the first vote in September 2009. The discussion at the plenary session held in November 2009 focused on overlapping protection groups.

As for the PBB-TE control plane, two technologies are available for dynamically configuring PBB-TE tunnels: Provider Link State Bridging (PLSB) and Generalized Multiprotocol Label Switching (GMPLS). PLSB is 802.1aq under the specification process of the IEEE 802.1 working group, while GMPLS is a PBB-TE related standard being developed by the IETF Common Control and Measurement Plane (CCAMP) working group. GMPLS should coordinate with the data plane standardization of the IEEE. Documents on the PBB-TE control plane released by the IETF CCAMP working group include a few on the requirements, architecture, and control protocols of GMPLS controlling Ethernet.

2 MPLS-TP

2.1 Development History

The International Telecommunication Union's Telecommunication Standardization Sector (ITU-T) SG15 started working on Transport MPLS (T-MPLS) standardization as early as 2005. Based on MPLS and transport architecture, T-MPLS simplifies MPLS

by eliminating technical content that is not connection-oriented, abandoning complex protocol families, and adding OAM and protection in the style of legacy transport networks. In 2006, the ITU first approved three standard recommendations for T-MPLS architecture, interface, and equipment functions and features. Following that, the standard recommendations for OAM, protection, and network management were successively proposed. In the ITU-T SG15 Q12+Q14 interim meeting held in Stuttgart in September 2007, the IETF mission indicated that ITU-T's T-MPLS standards conflicted with IETF's related standards. Reserved bytes and reserved numbers of MPLS protocols, for example, were used without any negotiation with IETF, even though IETF defined the core part of MPLS protocols.

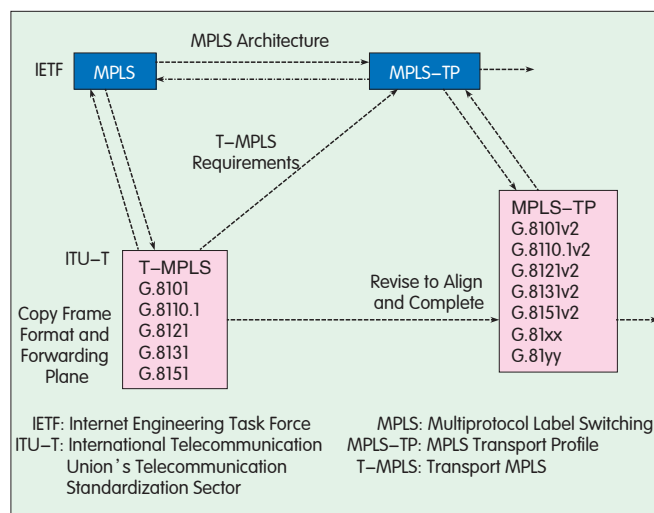
At the February 2008 ITU-T SG15 plenary meeting held in Geneva, Groups Q12 and Q14 reported the debate on T-MPLS protocols. The meeting resulted in a Joint Working Team (JWT) being established to evaluate options for evolving MPLS technology to meet requirements of the transport network. The JWT comprised the T-MPLS Ad Hoc group of ITU-T and MPLS Interoperability Design (MEAD) team. The JWT chairmen were Malcolm Betts, the Q12 rapporteur, and David Ward from IETF. After a series of teleconferences, in April 2008, the JWT decided that ITU-T and IETF would

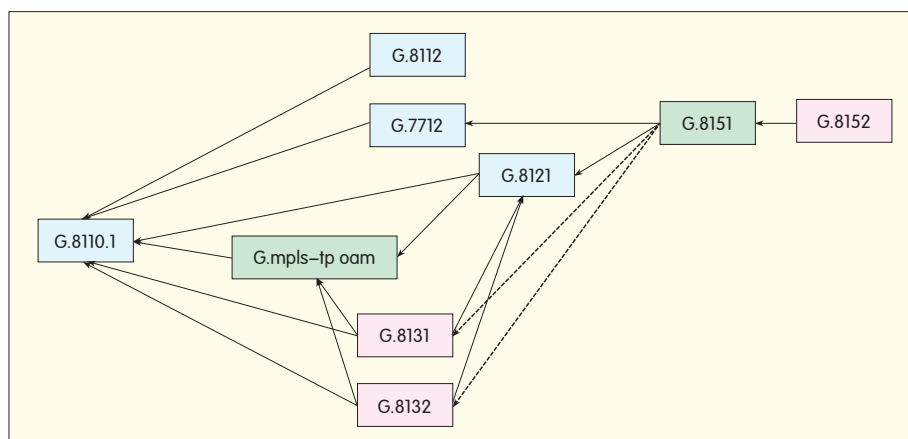
cooperate to develop related standards. ITU-T would provide transport requirements to IETF; and to meet these requirements, would expand MPLS OAM, network management, and control plane protocols through the IETF's standard procedure. The technology was renamed MPLS-TP. IETF's MEAD team was responsible for defining MPLS-TP. In July 2008, the IETF 72nd meeting released MPLS-TP related individual drafts on requirements, framework, MPLS generic associated channel, network management, OAM analysis and requirements, and survivability framework.

WP3 openly discussed T-MPLS/MPLS-TP documents at the February 2008 ITU-T SG15 plenary meeting. The WP3 chairman used pages 7–10 of the powerpoint document provided by the JWT to eliminate bias and make clear the following: MPLS-TP standards would be jointly developed by IETF and ITU-T, as shown in Figure 1; ITU-T would stop work on T-MPLS standards; the released T-MPLS standards would remain unchanged; ITU-T would not make any standard revision or engage in new standard development until IETF's related standards became stable; and the technology name would be changed to MPLS-TP.

Drafts on MPLS-TP framework, network management, MPLS generic associated channel, and OAM that were submitted to the IETF 74th

Figure 1 Evolution of T-MPLS/MPLS-TP standards.





▲ Figure 2. Reference relations among ITU-T standards.

meeting in March 2009 became the working group documents.

ITU-T SG15 Q9, Q10, Q12, and Q14 held a joint meeting to discuss revision of MPLS-TP standards in May 2009. At the meeting, editors provided the revised versions of G.8110.1, G.8110.1 amd1, G.8112, G.8121, G.8131, G.8151, G.8101, the G.8132 draft, and the G.mpls-tpoam draft. At the meeting it was agreed that standards would be approved in the sequence G.8110.1, G.8101, G.7712, G.8112, G.8121, G.8131 and G.8151.

2.2 Progress of Standard Development

2.2.1 ITU-T

The second ITU SG15 meeting in the 2008–2012 study period was held in Geneva between September 28 and October 9, 2009. The joint meeting of MPLS-TP Q9, Q10, Q12, and Q14 discussed IETF mail correspondence, and also created new correspondence.

(1)TD218-WP3 "LS: Restructure of MPLS-TP Work Forums in the IETF."

IETF notified ITU-T that it would dismiss the IETF MEAD team with the purpose of conducting MPLS-TP standardization work according to the normal IETF standardization procedure. It would enable all IETF members to participate in MPLS-TP standardization work earlier. MPLS-TP related tasks would be undertaken by five related IETF working groups. Discussions on MPLS-TP might be conducted through the IETF MPLS-TP mailing list. Adrian Farrel, the chairman of the routing area

and representative of IETF, would be responsible for the general coordination of MPLS-TP work, while the chairmen of the five working groups would execute detailed working plans. A committee would be established to coordinate with ITU, with members including the study subject chairmen, WP3 chairman, the SG15 chairman of ITU-T, as well as the working group chairmen and routing area chairman of IETF. The committee members would convene regularly to exchange information and promote MPLS-TP standardization. Information arising from this committee would be provided to ITU-T participants through the mailing list. The original JWT and Ad Hoc was to remain unchanged, but the Ad Hoc website would be updated to reflect current work.

(2)TD167-WP3 "LS – New version of the MPLS-TP process document available"

This document aimed to explain the normal IETF working procedure and interactions with ITU-T. It is helpful for the ITU-T members to understand the IETF procedure and to participate in IETF work according to the procedure. This document is currently a working group draft, and will be further promoted. It is expected to be released as a history RFC. The draft should be updated according to TD218-WP3, and ITU-T has provided some comments and sent correspondence to IETF.

(3)TD200-WP3 "LS: Progress report on MPLS-TP documents"

This provides a list of MPLS-TP related Requests for Comments (RFC) and standards being developed. It also shows the steps involved in RFC development. The working group's last call is critical in the RFC release procedure. The last call of a RFC usually requires two weeks, and may be repeated once. In total, the last calls of 20 standards require about 80 weeks. Renewing the last call should be avoided in order to save time. Therefore, ITU-T experts are expected to submit informal comments in the early stage of opening an MPLS-TP document, and this can avoid or significantly decrease comments from ITU-T in the formal review period of a last call.

Figure 2 shows reference relations among ITU-T standards.

Figure 2 does not show G.8101 because the development of G.8101 depends on the completion of all the other standards, including the terminologies. Dotted lines in Figure 2 indicate the initial version of G.8151, which may not include the linear and ring protection models.

The MPLS-TP standards to be submitted for approval are divided into three stages: G.8110.1, G.7712, G.8112; G.mpls-tp oam, G.8121, G.8151; and G.8131, G.8132, G.8152.

If a referenced RFC is approved, the corresponding ITU-T standard may also be approved, and IETF should check the corresponding ITU-T standard at the same stage. The original versions of these standards have existed, and they must be in line with developed RFCs. The current plan involves approving the first-stage ITU-T standards at the next SG15 plenary session, and these standards will be discussed through mail and interim meetings before the session.

2.2.2 IETF

At the IETF 76th meeting held in Hiroshima between November 8 and 13, 2009, the MPLS-TP drafts were discussed by the MPLS working group. Since there were many MPLS-TP related personal documents submitted for the meeting, the MPLS working group arranged two MPLS meeting

▼ Table 1. MPLS-TP related RFCs and working group documents

Document State	Document Name
RFCs	RFC 5317: JWT Report on MPLS Architectural Considerations for a Transport Profile
	RFC 5462: EXP field renamed to Traffic Class field
	RFC 5586: MPLS Generic Associated Channel
	RFC 5654: MPLS-TP Requirements
	RFC 5718: An Inband Data Communication Network For the MPLS Transport Profile
	RFC 5704: Uncoordinated Protocol Development Considered Harmful
	RFC 5860: Requirements for OAM in MPLS Transport Networks
	RFC 5921: A Framework for MPLS in Transport Networks
MPLS Working Group Drafts	draft-ietf-mpls-tp-nm-req: MPLS TP Network Management Requirements
	draft-ietf-mpls-tp-cc-cv-rdi: Proactive Connection Verification, Continuity Check and Remote Defect Indication for MPLS Transport Profile
	draft-ietf-mpls-tp-ach-llv: Definition of ACH TLV Structure
	draft-ietf-mpls-tp-survive-fwk: Multiprotocol Label Switching Transport Profile Survivability Framework
	draft-ietf-mpls-tp-nm-framework: MPLS-TP Networks Management Framework
	draft-ietf-mpls-tp-oam-framework: MPLS-TP OAM Framework and Overview
	draft-ietf-mpls-tp-oam-analysis: MPLS-TP OAM Analysis
	draft-ietf-mpls-tp-rosetta-stone: MPLS-TP Rosetta Stone
	draft-ietf-mpls-tp-linear-protection: MPLS-TP Linear Protection
	draft-ietf-mpls-tp-identifiers: MPLS-TP Identifiers
	draft-ietf-mpls-tp-process: IETF Multi-Protocol Label Switching (MPLS) Transport Profile (MPLS-TP) Document Process
	draft-ietf-mpls-tp-fault: MPLS-TP Fault OAM
	draft-ietf-mpls-tp-data-plane: MPLS Transport Profile Data Plane Architecture
	draft-ietf-mpls-tp-on-demand-cv: MPLS on-Demand Connectivity Verification, Route Tracing and Adjacency Verification
	draft-ietf-mpls-tp-loss-delay: Packet Loss and Delay Measurement for the MPLS Transport Profile
	draft-ietf-mpls-tp-lsp-ping-bfd-procedures: LSP-Ping and BFD Encapsulation over ACH
CCAMP Working Group Draft	draft-ietf-ccamp-oam-configuration-fwk: OAM Configuration Framework and Requirements for GMPLS RSVP-TE
	draft-ietf-ccamp-mpls-tp-cp-framework: MPLS-TP Control Plane Framework
PWE3 Working Group Draft	draft-ietf-pwe3-mpls-transport: Application of Ethernet Pseudowires to MPLS Transport Networks
OPSA Working Group Draft	draft-ietf-opsawg-mpls-tp-oam-def: The OAM Acronym Soup

▼ Table 2. Individual documents submitted for the 76th IETF meeting

Document State	Document Name
Individual Documents	draft-koike-ietf-mpls-tp-oam-maintenance-points-00
	draft-sprecher-mpls-tp-oam-analysis-07
	draft-asm-mpls-tp-bfd-cc-cv-01
	draft-nitinb-mpls-tp-lsp-ping-extensions-00
	draft-swallow-mpls-tp-identifiers-02
	draft-sfv-mpls-tp-fault-00
	draft-frost-mpls-tp-loss-delay-00
	draft-sprecher-mpls-tp-survive-fwk-02
	draft-weingarten-mpls-tp-linear-protection-04
	draft-weingarten-mpls-tp-ring-protection-01
	draft-umansky-mpls-tp-ring-protection-switching-00
	draft-liu-mpls-tp-ring-protection-00
	draft-jiang-mpls-tp-ring-fd-00
	draft-he-mpls-tp-csf-01
	draft-zhl-mpls-tp-sd-01
	draft-flh-mpls-tp-oam-diagnostic-test-00
	draft-dai-mpls-tp-lock-instruct-00
	draft-chen-mpls-tp-nm-discovery-req-00

segments and one meeting segment of the Pseudo Wire Emulation Edge-to-Edge (PWE3) working group to discuss them.

Table 1 lists MPLS-TP related IETF RFCs and working group documents, and Table 2 lists the individual documents submitted for the IETF 76th meeting.

3 Conclusion

The emergence of IP services results in IP based networks. Legacy networks using Time Division Multiplexing (TDM) switching are gradually shrinking and being replaced by packet networks. Packet transport networking is an inevitable outcome of IP based networks. The progress of its standardization will significantly impact the development of the entire industrial chain.

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Biography

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Several Issues in the Development of Packet Transport Networks

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Abstract:

Several key issues affect the development and standardization of Packet Transport Networks (PTN) and Multiprotocol Label Switching Transport Profile (MPLS-TP). These include the end-to-end Quality of Service (QoS) mechanism, layered network architecture, introduction of layer-3 functions, and data-plane loopback functions. This paper introduces several views on the construction and maintenance of PTN, and on the requirements of PTN services. After an analysis of Traffic Engineering (TE) based on Multiprotocol Label Switching (MPLS) and Differentiated Service (DiffServ), a service-oriented end-to-end QoS guarantee mechanism is proposed. An alternative to introducing layer-3 functions into PTN is also proposed, based on PTN layered architecture defined in the available MPLS-TP standards and drafts. Requirements of data-plane loopback functions are discussed in conclusion.

There is no standard definition for Packet Transport Networks (PTN). Broadly speaking, PTN can be defined as any transport network that is based on packet switching and meets certain Operation Administration and Maintenance (OAM), protection, and network management requirements. Specific packet-switched technologies include Multiprotocol Label Switching (MPLS), Transport MPLS/MPLS Transport Profile (T-MPLS/MPLS-TP), Ethernet, Backbone Bridge Traffic Engineering (PBB-TE), and Resilient Packet Ring (RPR). Over the past two years, both T-MPLS and PBB-TE have been viewed as the leading PTN technologies. However, as support for PBB-TE from vendors and operators wanes, China has come to view T-MPLS/MPLS-TP as PTN. Therefore, PTNs described in this paper are based on T-MPLS/MPLS-TP.

Transition from T-MPLS to MPLS-TP

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reflects a history of competition and integration of the transport and data fields. MPLS-TP is an achievement preceded by years of competition and negotiation between the Internet Engineering Task Force (IETF) and International Telecommunication Union's Telecommunication Standardization Sector (ITU-T). It can be seen as a balanced exploitation of the benefits of transport and data fields. IETF currently leads the development of MPLS-TP standards, while ITU-T SG15 has a lesser hand and mainly relies on the participation of enterprises and individual expert members^[1-6].

1 QoS in PTN

QoS provides predictable service quality in terms of packet loss ratio, delay, jitter, and bandwidth during network communication. QoS in PTN includes stream classification, labeling, rate limitation, bandwidth guarantee, traffic shaping, and scheduling strategy. MPLS Traffic Engineering (TE) and Differentiated Service (DiffServ) mechanisms are used to implement

QoS in PTN, with the goal of creating a service-oriented end-to-end QoS guarantee^[7-10].

1.1 TE

According to IETF's definition, MPLS-TP must support TE so that network resources can be controlled. The purpose of TE is to optimize resource use for effective and reliable network operation. An important element of TE is the Constraint-Based Routing (CBR) mechanism. In an IP/MPLS network, TE is generally implemented by MPLS-TE, and has two important roles:

(1) Making Service Routing Controllable

A PTN service is first encapsulated by the Pseudo Wire (PW), and then multiplexed to the Label Switched Path (LSP). The LSP is established through network management or the control plane, and LSP routing implemented by either of these is controllable.

(2) Making Service Bandwidth Controllable

PTN can bear E1 emulation services and Ethernet services. The bandwidth

of an E1 emulation service must be fixed and controllable, having high priority, no packet loss, and low delay. Ethernet services can be divided into Constant Bit Rate (CBR) services and Variable Bit Rate (VBR) services. A CBR has the same basic requirements and controllability as E1 emulation services. A VBR service uses Committed Information Rate (CIR) and Excess Information Rate (EIR) to control bandwidth. An operator only guarantees user bandwidth not exceeding the CIR, and may discard EIR traffic when the network is congested. In this way, network bandwidth is controlled.

More specifically, the operator first configures the CIR of the PW (eg. the service) as well as the LSP, and sets the conditions for controllable connection. The sum of all the PW CIRs in one LSP should not exceed the CIR of the LSP; the sum of all the LSP CIRs in one link should not exceed the CIR of the link. Under these conditions, the network can satisfy the CIR bandwidth requirements of all services. Since VBR service involves bursts (eg. the EIR part), network congestion cannot be completely avoided using TE. DiffServ is an effective solution for ensuring CIR bandwidth in such conditions.

1.2 DiffServ

DiffServ is derived from Integrated Service (IntServ), and aims to provide differentiated service levels for Internet traffic. Compared with IntServ, DiffServ has a simpler control system with coarser granularity. It is used to control each QoS type after streaming convergence. IntServ, on the other hand, aims to control each individual stream. DiffServ is therefore scalable, and capable of QoS in large-scale networks.

DiffServ classifies IP streams into different types at the edge of its domain, and assigns one Differentiated Services Code Point (DSCP) to each stream type. The core router in the domain checks the DSCP value, and dispatches packet forwarding according to different types of Per-Hop Behavior (PHB). IETF has defined two kinds of PHB: Expedited Forwarding

(EF) and Assured Forwarding (AF).

(1) EF

EF PHB traffic is unaffected by any other PHB traffic, and this ensures packets are forwarded at the quickest possible rate. Similar to legacy leased lines, EF PHB can guarantee bandwidth service with low packet loss, low delay, and low jitter. CIR is the only bandwidth parameter of an EF-based service; EIR is equal to 0 and traffic greater than CIR is discarded. EF can be applied to E1 emulation and CBR Ethernet services, and should follow RFC3246.

(2) AF

AF provides four levels of packet forwarding, each level with three discarding priorities. The level of a service is determined by PTN equipment—which configures forwarding resources (such as buffer and bandwidth) at different levels, and accounts for the discarding priority. If a service is not congested, AF service performances at different levels are equal. However, if the service is congested, packet loss occurs on all AF levels, and the degree of packet loss correlates with the service level. AF should follow RFC2597.

1.3 DiffServ Supported by MPLS

MPLS-TP based PTN must use the MPLS DiffServ mechanism defined in RFC3270.

After an IP packet has been encapsulated through MPLS, the core router cannot locate DSCP. Accordingly, IETF has proposed a DiffServ-supported MPLS that can map multiple Behavior Aggregates (BA) of DiffServ to a MPLS LSP and forward the traffic on the LSP according to the BA PHB. There are two mapping modes

between LSP and BA:

EXP-inferred-PSC LSP (E-LSP) and Label-Only-Inferred-PSC LSP (L-LSP).

(1) E-LSP

E-LSP uses the MPLS-labeled EXP field to designate multiple BAs to one LSP. The MPLS-labeled EXP field represents the PHB of a packet. Up to eight BAs can be mapped into the EXP field; that is, an E-LSP supports a maximum of eight service levels.

(2) L-LSP

L-LSP designates one LSP to one BA, and uses EXP to represent the packet discarding priority. An L-LSP can only support one service level.

MPLS equipment switches label values per hop, but the management of mapping between label and PHB is difficult. Therefore, compared to L-LSP, E-LSP is more easily controlled because it can determine in advance the mapping relationship between the EXP field and PHB for every packet in the network. PTN equipment currently uses E-LSP.

1.4 End-to-End QoS Implementation in PTN

TE and DiffServ supported by MPLS help PTN guarantee service-oriented end-to-end QoS. MPLS TE is used to control service routing and bandwidth in order to avoid network congestion caused by unbalanced loading. Once the network is congested by burst service or network protection, DiffServ supported by MPLS is used to guarantee CIR.

Table 1 classifies service levels for E-LSP. Peak Information Rate (PIR)=CIR+EIR; and under this condition, the EXP value of LSP and PW

▼ Table 1. PTN service priority instances

Service Levels	Service (PW) Bandwidth Attribute	Experimental Field	Default Hop-by-Hop	Service Type Instances
3	CIR=PIR	111	EF	Control/Network Management Messages/E1 Emulation
2	CIR=PIR	110	EF	Voice/Video
1	CIR, PIR	101 (less than CIR) 100 (more than CIR but less than PIR)	AF	VPN/Private Ethernet
0	CIR=0, PIR	000	DF	Best Effort Data Services
AF: Assured Forwarding CIR: Committed Information Rate DF: Default Forwarding		EF: Expedited Forwarding PIR: Peak Information Rate PW: Pseudo Wire	VPN: Virtual Private Network	

in a data frame are the same one.

Both E1 emulation services and CBR Ethernet services (such as voice and video) adopt EF PHB with a setting of CIR=PIR.

Burst-type services such as virtual private networks and private Ethernet lines use AF PHB. To ensure CIR bandwidth of burst-type services, service streams must be measured, shaped, and labeled at the network ingress according to bandwidth parameters, and Two Rate Three Color Marker (trTCM) must be supported. Moreover, the EXP value of the data frame is set from the mapping relationship, which is used by the LSP follow-up nodes to select suitable PHB.

As for ordinary data services, CIR is set at 0, PIR is set, and Default Forwarding (DF) is used.

Even if the network is congested, the service bandwidth of both EF PHB and CIR part of AF PHB traffic is always guaranteed. Ordinary data services are either discarded or weighed with AF PHB traffic so that certain bandwidth can be obtained during the congestion.

2 Layered Architecture of PTN

IETF RFC5654 divides an MPLS-TP system into transport service layer, transport path layer, and section layer. The transport service layer can be PW or service LSP, similar to Synchronous Digital Hierarchy (SDH) Virtual Channel (VC)-12. PW is used to offer emulation services such as Time Division Multiplexing (TDM), Ethernet, and Asynchronous Transfer Mode (ATM) services. Service LSP is used to offer network-layer IP and MPLS services. The transport path layer refers to the LSP layer, similar to SDH VC-4. The section layer is used to converge information from the transport service layer and transport path layer between two adjacent MPLS-TP nodes. The section layer can be implemented by either MPLS-TP or technologies such as SDH, Ethernet, or Optical Transport Network (OTN). With a layered architecture, PTN can achieve scalability similar to SDH/OTN.

Besides 3-layer MPLS-TP, PTN

should also support related functions of the service layer and section layer. Such functions include OAM of the Ethernet service layer (specified in IEEE802.1ag and Y.1731), OAM of the Ethernet link layer (specified in IEEE 802.3ah), overhead handling of SDH services and links, and protection.

Current PTN equipment uses PW to support various emulation services but does not support IP/MPLS services through LSP. IP/MPLS service implemented by Ethernet PW emulation is highly transparent, but also inefficient—especially for short packets because Ethernet frame headers need to be transported. IP/MPLS implemented by TDM PW emulation has high requirements on network performance, and may increase equipment cost. IP/MPLS implemented by service LSP can avoid the abovementioned problems, but has poor service transparency and is possibly required to handle part of the L3 protocol. Therefore, service transparency, transport efficiency, and cost must be taken into account when selecting IP/MPLS service implementation technology.

In addition, current PTN equipment only supports Single-Segment Pseudo Wires (SS-PW); that is, where the source & sink of PW coincides with that of LSP. SS-PW cannot converge PWs borne by multiple LSPs, and requires PTN equipment to have high LSP capacity. Moreover, current PTN equipment with only end-to-end LSP protection cannot cope with multi-point faults. Multi-Segment Pseudo Wires (MS-PW) can be introduced to solve problems caused by SS-PW, and thereby improve PTN scalability. IETF has listed MS-PW as optional for MPLS-TP.

3 The Support of PTN on L3 Functions and Services

Currently, PTN mostly offers L2 services, including E1/ATM emulation services and E-Line/E-LAN/E-Tree services. It is primarily applied in 3G and Long Term Evolution (LTE) mobile backhaul networks. PTN can satisfy the bearing requirements of 3G networks,

but it is still doubtful whether it can meet the bearing requirements of future LTE systems.

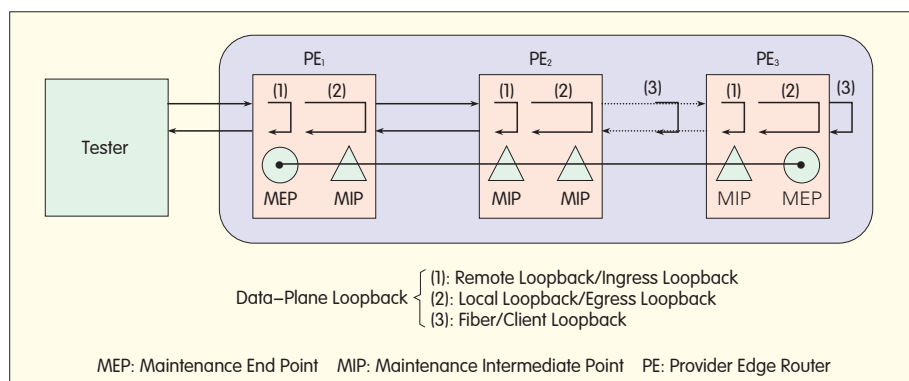
LTE systems have different bearing requirements from 3G networks because they require interconnection between base stations (X2 interface) and multi-homing from base station to Service Gateway (SGW). There are two ways to satisfy the bearing requirements of LTE: end-to-end router networking, and L3+L2 networking (with L3 networking for the core layer and L2 networking for the convergence and access layer). Since end-to-end router networking still has problems with network scalability, manageability, and controllability, L3+L2 networking has gained recognition and support. The core layer in this solution can be implemented by router networking or by introducing L3 functions into PTN.

L3 functions mainly include IP routing and forwarding, L3 MPLS Virtual Private Network (VPN), and L3 multicasting. IP traffic and multicast, with uncertain traffic bandwidth and routing, cannot provide strict QoS guarantee. If these two services are introduced into PTN, they should be set as the lowest level services in order to avoid any impact on L2 services. L3 MPLS VPN can use MPLS TE and DiffServ mechanisms to guarantee QoS. MPLS VPN can also support L3 multicasting with QoS guarantee.

For PTN networks, L3 services with QoS requirements can be offered by L3 MPLS VPN, while L3 services with no QoS requirements may be implemented directly through IP routing and forwarding.

4 Data-Plane Loopback Functions

Current PTN equipment only supports OAM Loopback (LB). OAM LB can be used to verify bidirectional interconnection between the source and sink maintenance endpoints, and to check for inter-node/intra-node faults. However, it cannot locate the specific fault positions. As shown in Figure 1, if there is a fault on the PE₂-PE₃ link, OAM LB cannot determine



▲ Figure 1. Data-Plane Loopback.

whether the fault is in PE₃ or on the link. If PTN equipment can support SDH-like data-plane LB; that is, service LB, accurate fault positioning can be implemented through LB of different points.

Similar to SDH, PTN data-plane loopbacks include Remote/Ingress Loopback, Local/Egress Loopback, and Fiber/Client Loopback. Beside for fault locating, fiber loopback can be used to perform single-end service performance tests on bidirectional time delay, packet loss ratio, and throughput. It also supports testing in live networks.

Supporting both remote and local OAM loopbacks, PTN can implement functions similar to remote and local loopbacks on the data plane. Therefore, fiber loopback should be implemented in PTN to accurately locate faults, and for single-end testing. Further research is still needed to determine whether PTN should support data-plane remote and local loopback functions. Currently,

IETF and ITU-T are discussing the standardization of data-plane loopback functions.

5 Conclusion

PTN is the optimal solution for evolving Multiservice Transport Platform (MSTP) networks bearing 2G mobile backhaul. The goal is to meet requirements of high-quality services such as 3G mobile backhaul and enterprise private line services. From 2008 to 2009, China's three dominant telecom operators trialled PTN bearing 3G mobile backhaul in their live networks. They pushed forward the maturing and commercialization of PTN products. China Mobile began large-scale purchasing of MPLS-TP based PTN equipment in October 2009, marking a turning point in the industrialization of PTN. International MPLS-TP standards have attracted much attention in 2010, and the stability of international MPLS-TP standards will determine how

quickly PTN moves from its introductory stage into large-scale application.

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Biography

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Jing Ruiquan is a senior engineer of China Telecom Corporation, Beijing Research Institute, and is a member of the information communication expert team of the National High Technology Research and Development Program of China ("863" program). His research interests include optical transport networks, packet transport networks, and IP RAN bearing technology. He has long been engaged in planning, construction, and technical research of optical transport networks, and has participated in many SDH, MSTP, and ASON testing tasks. He has published more than 20 academic papers.

Roundup

ZTE France Awarded Best Investor Prize 2010

ZTE Corporation has been awarded the Best Investor Prize 2010 by The Greater Paris Investment Agency, recognizing it for its contribution to the Greater Paris region.

ZTE's headquarters for Europe and North America is based in Paris, helping to position the area as a strategic global business centre for

the high-tech industry.

"We are proud to receive this award from the Greater Paris Investment Agency. ZTE is committed to growing its local office in France and making a contribution to the local community and this award recognises our achievements," said Mr. Zhu Yun, President for ZTE Western Europe,

Eastern Europe and America.

The award ceremony took place on June 22nd in the Chamber of Commerce in Paris and was given personally by the secretary of state for external trade, Ms. Anne-Marie Idrac. In addition to ZTE, two companies in other industries were awarded the Best Investor Price. (ZTE Corporation)

Features of PBB-TE Architecture and GMPLS Control Technology

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Abstract:

Time Division Multiplexing (TDM) transport networks are evolving to packet-oriented, and a variety of carrier-class packet transport technologies have emerged. Provider Backbone Bridge with Traffic Engineering (PBB-TE) is a connection-oriented packet transport technology that provides good scalability and manageability, and guarantees Quality of Service (QoS). Generalized Multi-Protocol Label Switching (GMPLS) is a mature transport network control plane technology that supports multiple data planes with different switching granularity. GMPLS-controlled PBB-TE is a promising solution for Packet Transport Networks (PTN).

Backbone Bridge Traffic Engineering (PBB-TE) is a connection-oriented packet transport technology with good scalability and end-to-end QoS support. It contains an Operation, Administration and Maintenance (OAM) mechanism on the data plane. This enhances the reliability and manageability of telecom networks. PBB-TE is expected to be the preferred solution to metro Packet Transport Networks (PTNs).

1 Features of PBB-TE Architecture

1.1 Evolution of IEEE Ethernet

1.1.1 802.1Q Virtual Local Area Network (VLAN)

In IEEE 802.1Q^[1], a Customer Virtual

LAN Tag (C-Tag) domain based on the frame structure of 802.1 Ethernet is introduced. C-Tag contains a 12-bit Customer Virtual LAN ID (C-VID) and a 3-bit Customer Product ID (C-PID). The C-VID indicates which VLAN the source host belongs to, and C-PID shows the service type of a frame. In a 802.1Q system, physical networks can support up to 4,096 VLANs, and traffic in different VLANs is separated. Depending on the service type indicated by C-PID, the 802.1Q network bridge can offer differentiated services.

1.1.2 802.1ad Provider Bridge (PB)

IEEE 802.1ad PB^[2] is the first provider-oriented Ethernet bridge technology. In PB, a Service VLAN Tag (S-Tag) domain is added to the 802.1Q frame structure for service providers. This domain contains 12-bit provider VLAN identifiers (S-VID) and a 3-bit C-PID. An IEEE 802.1ad bridge network is called Provider Bridge Network (PBN). As shown in Figure 1, an S-Tag is either assigned or removed at the entry node of a PBN. The S-Tag separates provider VLAN

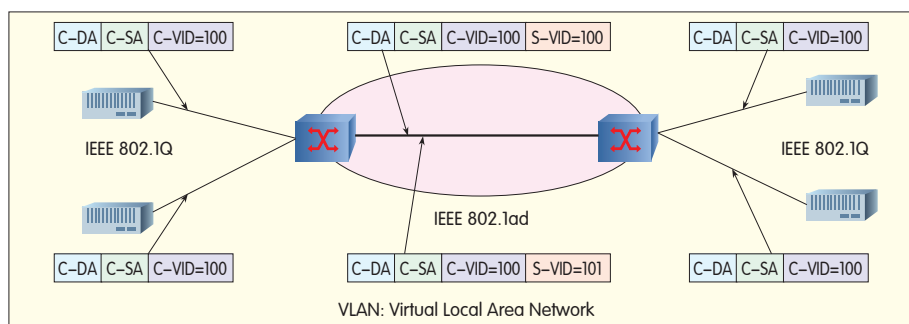
from customer VLAN, and allows multiple customer VLAN services to be transported through the same provider VLAN.

Limited by the length of the S-VID, PBN supports up to 4,096 services. It forwards frames in the format <C-DA+S-VID>, and learns customer Media Access Control (MAC) addresses. Each PBN node therefore maintains a large forwarding table. With these limitations, PBN fails to meet the scalability requirements of telecom networks.

1.1.3 802.1ah Provider Backbone Bridge (PBB)

In IEEE 802.1ah PBB^[3], a Provider Backbone Bridge Network (PBBN) based on PBN is established to improve PB scalability. A PBB frame has a provider frame header <B-DA, B-SA, B-TAG, I-TAG>, which is absent from the PB frame. The PBBN edge node is responsible for adding and deleting the provider frame header. In the header, B-DA and B-SA are the MAC addresses of the PBBN entry and exit nodes; the B-TAG containing a 12-bit B-VID indicates a PBBN

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▲ Figure 1. Provider VLAN is separated from customer VLAN in 802.1ad.

spanning tree or a transport path; and the I-Tag with 24-bit I-SID represents the number of services. A PBBN can support up to 16 million services. The PBBN core node forwards frames in the format <B-DA+B-VID>, and only its edge nodes are required to learn customer MAC addresses. Therefore, the forwarding table of the core node is greatly reduced.

In terms of the number of services and nodes supported, PBB is the first bridge technology to meet the requirements of telecom networks. However, PBB still lacks Traffic Engineering (TE) and OAM features.

1.2 Features of 802.1Qay PBB-TE

A product of PBB and telecom network features, IEEE 802.1Qay PBB-TE^[4] is a type of connection-oriented packet transport technology. PBB-TE network architecture, as shown in Figure 2, has the following features:

(1) Scalability

On the data plane, PBB-TE has the same MAC-in-MAC frame structure as PBB. The core node of a PBB-TE network forwards frames in the format <B-DA, B-VID>. PBB-TE inherits the strengths of PBB in supporting a large number of services and separating provider and customer addresses.

(2) Connection-Oriented and QoS Guarantee

PBB-TE discards the spanning tree protocol and source address learning mechanism, as well as the frames of unknown addresses. Ethernet Switched Path (ESP) is used for transport services and should be established by the control plane or management system. PBB-TE is therefore connection-oriented with each ESP

having definite TE attributes and QoS guarantee.

(3) OAM

Using a Connectivity Fault Management (CFM)^[5] OAM mechanism, PBB-TE can provide carrier-class OAM without assistance from other layers.

(4) End-to-End Path Protection

PBB-TE provides point-to-point and point-to-multipoint ESP with 1:1 path protection. A protection path can be built while simultaneously building a working path, and by pre-configuring the protection path, QoS identical to the working path can be achieved. PBB-TE path fault diagnosis and protection triggering are all completed on the data plane, and protection switching time can reach 50 ms.

(5) Multi-Service Bearing

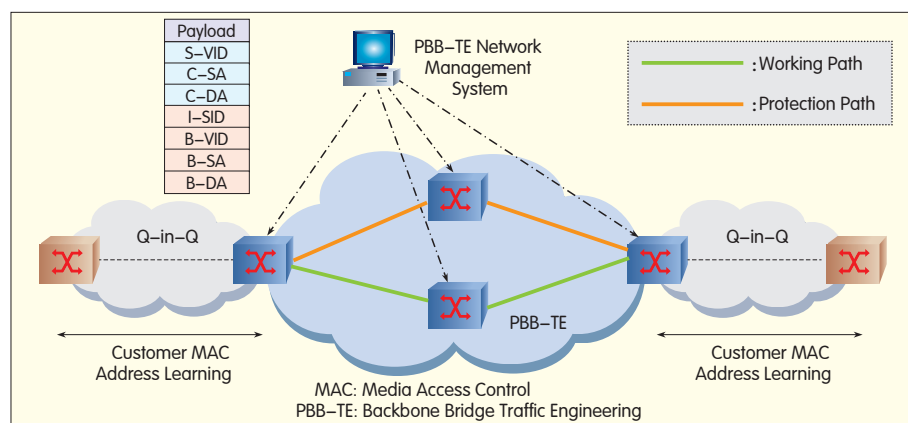
PBB-TE can bear L2 and L3 services, and also supports TDM services. However, compared to Transport Multiprotocol Label Switching (T-MPLS), PBB-TE has weaker Multipoint-to-Multipoint (MP2MP) service support, and its QoS

classification and control plane technology are not developed enough. These weaknesses are expected to be solved during the PBB-TE standardization process. PBB-TE reduces MAN maintenance costs in the long run, and some operators have already tried deploying PBB-TE systems^[6].

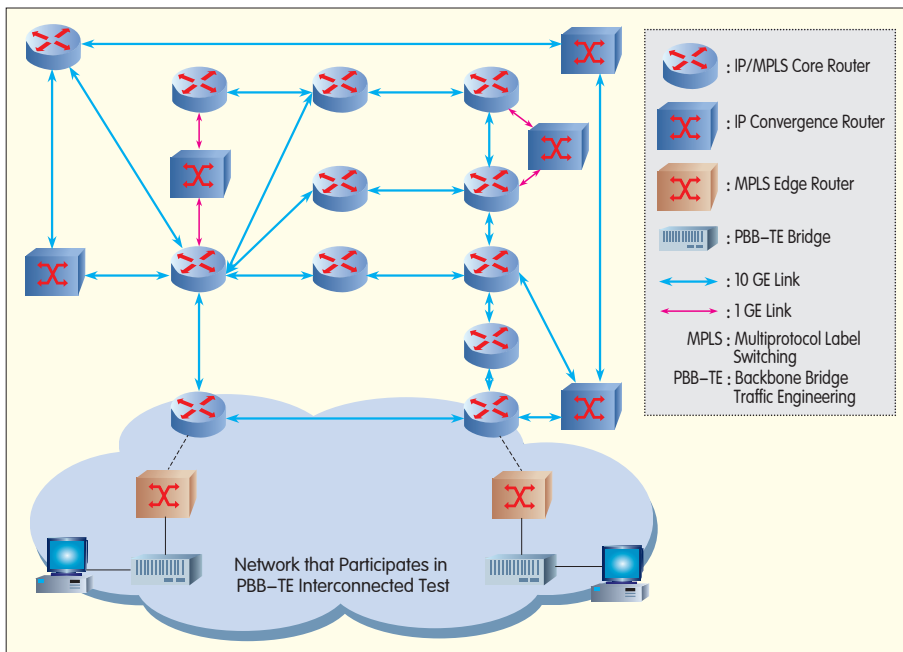
1.3 PBB-TE Equipment Interconnection Test in Shanghai Jiao Tong University Backbone Network

A cost-effective carrier-class packet transport technology, PBB-TE can be applied not only in simple MANs, but also in complex scenarios (such as data centers or R&D backbone networks with intensive services and a large number of nodes). The Network Center of Shanghai Jiao Tong University has attempted to apply PBB-TE into the campus backbone network, and the success of this test will verify the interoperability of PBB-TE and MPLS equipment.

In the campus backbone network shown in Figure 3, the core network consists of a series of IP/MPLS routers interconnected by 10GE or GE routers. The convergence network consists of IP routers. The campus network offers services such as on-demand and multicast intra-campus video, email, File Transfer Protocol (FTP), and P2P file sharing. To provide better service, the network uses MPLS-TE technology in a number of areas. A specific service-oriented MPLS Virtual Private Network (VPN) has also been created to provide university administration with



▲ Figure 2. PBB-TE architecture.



▲ Figure 3. PBB-TE Interconnected Testing System in the Backbone Network of Shanghai Jiao Tong University.

a reliable and confidential network platform.

In the test, the customer MAC frame was encapsulated as the MAC-in-MAC frame after the bridge convergence of PBB-TE. The frame was then encapsulated as an MPLS packet at the MPLS edge node, and transported by the MPLS core network. The frame arrives at the host through the MPLS edge router and PBB-TE bridge at the entry of the MPLS core network. The PBB-TE bridge is a PBB-TE edge bridge with convergence capability.

The test will be deemed successful when a PBB-TE convergence network has been established. The network center intends to evaluate the feasibility of PBB-TE in the campus network by comparing the PBB-TE convergence network with the existing IP convergence network. A study of the PBB-TE control plane will also be conducted on equipment connected to MPLS switches.

2 GMPLS Controlled PBB-TE

Although PBB-TE control plane standards are still being studied,

industry has generally agreed to use GMPLS as the PBB-TE control plane technology. GMPLS extends the meaning of label and label exchange in MPLS, and reuses part of the MPLS protocol. GMPLS functions include signaling, routing, path selection, and link management. With extensions, GMPLS may support data planes such as SONET/SDH, Optical Transport Network (OTN), and Wavelength Division Multiplexing (WDM).

2.1 GMPLS Controlled of Ethernet Label Switching (GELS)

IETF is a leading promoter of standardization of GMPLS-controlled PBB-TE. While standardization is far from complete, two GELS drafts have been released^[7]. These drafts involve GELS architecture and technical specifications. GELS uses as many of the original GMPL components as possible, and makes necessary extensions.

(1) Addressing Mode

The node on the GELS control plane still uses the IP address ID, and control plane messages are exchanged on the IP layer. GELS supports both labeled and unlabeled ports.

(2) Signaling Protocol

A new label format <B-DA, B-VID> is added in GELS, which corresponds to the forwarding table entry of the PBB-TE node. On the data plane, different nodes on the same path correspond to the same forwarding table entry; therefore, the nodes should be assigned the same label. The PBB-TE label is global for the whole network.

(3) Traffic Parameters

GELS uses four bandwidth parameters^[8]: Committed Information Rate (CIR), Committed Burst Size (CBS), Excess Information Rate (EIR), and Excess Burst Size (EBS).

(4) Route and Path Computation

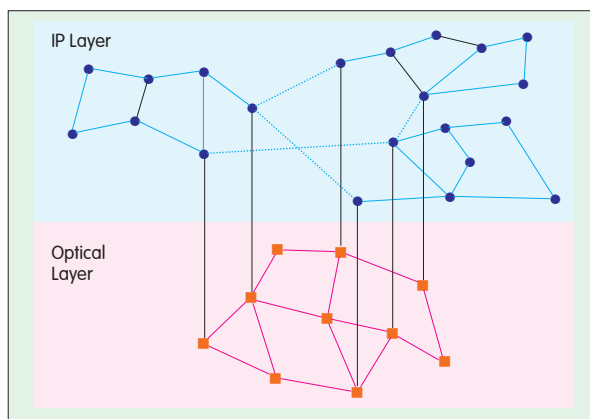
GMPLS does not limit path selection methods. Therefore, it allows computation and selection of any path. Open Shortest Path First with Traffic Engineering Extensions (OSPF-TE) and Intermediate System to Intermediate System Extensions for Traffic Engineering (IS-IS-TE) can still be used to release routing messages of the PBB-TE data plane. Because of labeled or unlabeled data plane ports, routing messages need not carry MAC addresses of ports.

(5) Link Management

GMPLS Link Management Protocol (LMP) and PBB-TE CFM have some overlapping functions. For example, both can implement neighbor discovery, fault diagnosis, fault confirmation, and fault positioning. CFM can work independently without support from other layers, while LMP can allocate numbered/unnumbered interface IDs automatically. CFM and LMP can therefore run together. The two IETF drafts only specify how GMPLS is used to build point-to-point PBB-TE paths. Improvements are required in the specifications of point-to-multipoint path establishment and control-plane-based protection recovery. 802.1aq Provider Link State Bridging (PLSB) is also a solution for PBB-TE control^[9].

2.2 Simulation Platform of GMPLS Controlled PBB-TE

Researchers are likely to use a virtual PBB-TE testing platform rather than a platform built with real PBB-TE



◀ Figure 4.
Screenshots of the Platform for
Large-Scale Optical Network
Validation.

equipment because a virtual platform is more flexible and supports more nodes. Such virtual platforms are of two types:

(1) Emulation

An emulation platform is represented by Finite State Machine (FSM) emulation software such as Network Simulator 2 (NS2). It supports a large number of nodes and has good scalability; however, it lacks details of signaling interaction, and fidelity of the control and data planes is not good enough.

(2) Simulation

The simulation platform is represented by the DRAGON program^[10], and uses computers to replace PBB-TE bridges. The complete GMPLS protocol stack is run in the computers, and data frames are sent through the network cards. Therefore, the PBB-TE control and data planes can be realistically simulated. Scalability is limited because one PBB-TE bridge requires one computer for simulation.

This paper focuses on signaling interworking and cross-layer optimization of GMPLS, and so a compromise is made between emulation and simulation. As shown in Figure 4, the large-scale optical network validation platform simulates a 2-layer network, with PBB-TE on the upper layer and SONET/SDH on the bottom layer. Resource Reservation Protocol-Traffic Engineering (RSVP-TE) and OSPF-TE are implemented completely on the platform, as are the GMPLS extensions to PBB-TE and SONET. The platform does not implement the forwarding

function on the data plane. A node on the platform is only an object in the computer's memory; signaling interaction between nodes as well as routing message updates are implemented by communications between objects, without the need for actual network cards. Both the signaling and routing messages are recorded in logs for offline reading. The platform is intended to successfully demonstrate cross-layer path establishment where tens of nodes are involved. GMPLS extensions are to be implemented on the platform for support of PBB-TE protection switching.

3 Conclusion

PBB-TE is a PTN technology with layered architecture, improved OAM, and QoS guarantee. As a convergence-layer solution, PBB-TE is more cost effective than MPLS. PBB-TE and GMPLS standards are still in progress, and more telecom network characteristics will be introduced into the PBB-TE system. With improved standards, PBB-TE is expected to become an optimal solution for next-generation metro PTNs.

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Ring Protection and Survivability Mechanisms for Packet Transport Networks

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Abstract:

Packet Transport Networks (PTNs) must resolve issues in their protection and recovery mechanisms. These issues include the detection performance of Operation, Administration and Maintenance (OAM), network resource optimization, resource allocation deadlock, and resource deployment blocking. Traditional protection and recovery mechanisms cannot meet the requirements of PTN. In order to improve the protection and recovery performance, network resource use, probability of service recovery, and to decrease the probability of service blocking, this paper introduces overlapped segment shared protection, Pre-configured Multi-Cycle (P-mcycle), conflict-free algorithm, and delay restoration algorithm. Only with appropriate protection and restoration mechanisms is it possible to achieve smooth evolution from TDM networks to integrated packet-based bearer networks and all-IP services.

The development of metro transport networking has been driven by access and transmission demands of broadband data services (e.g. 3-play), Ethernet Private Line (EPL) enterprise services, Layer 2 Virtual Private Network (L2VPN) services, and common broadband services. As service and bearer networks move towards IP, basic transport networks are evolving into Packet Transport Networks (PTNs)^[1].

With good scalability, powerful OAM, and fast protection switching, PTN inherits some of the characteristics of traditional Synchronous Digital Hierarchy (SDH) transport networking. New features such as packet switching, statistical multiplexing, connection-oriented label switching, Quality of Service (QoS) guarantee, and

flexible dynamic control are added for adaptation to data services. These are basic but important technologies for network convergence. In terms of transport mode, PTN not only provides packet data services and traditional Time Division Multiplexing (TDM) services, but also offers Asynchronous Transfer Mode (ATM), Inverse Multiplexing for ATM (IMA)^[2], and Multi-Level Pre-Emptive Priority (MLPPP) to meet transmission demands of 3G mobile systems. PTN devices adopt several encapsulation and adaptation technologies, integrating various TDM and data services on a unified packet transport plane. PTN is thus a unified, packet-switched multi-service transmission platform^[3].

However, integration of various demands and diversified application scenarios creates some critical problems for survivability. This issue directly impacts both the networking mode (ring, mesh, or star) and the service setup mode (1+1 or 1:1 path

protection) of a PTN network, and thus affects services. The survivability of PTN involves blocking during Label Switching Path (LSP) setup, resource utility, and coordination for resource allocation^[4]. Unlike SDH networks, PTN protection mechanisms are still being studied and protection performance is far from adequate.

Protection and restoration strategies of PTN play an important role in the network's service performance, resource utility, and survivability. Because PTN imposes high requirements on survivability, research on its protection and restoration strategies is important. Only with adequate protection and restoration mechanisms is it possible to evolve smoothly from TDM-based transport to packet transport, to achieve network convergence and to deliver all-IP services.

1 PTN Research Status

There are currently two main

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technologies for implementing PTN: Transport Multiprotocol Label Switching (T-MPLS)^[5]/Transport Multiprotocol Label Switching Traffic Engineering (MPLS-TE)^[6], and Provider Backbone Bridge (PBB)/Provider Backbone Transport (PBT)^[7]. Research into survivability strategies for PTNs with these 2 technologies has attracted much attention. Internationally, institutes and organizations such as Alcatel^[8], Nortel^[7], Cisco^[9], Sycamore^[10], European Union Seventh Framework Programme (FP7)^[11], the University of Texas^[12], Stanford University^[13] and Bell Labs^[14] are carrying out research. In China, telecom equipment manufacturers such as ZTE, Huawei, and Fiberhome are studying survivability strategies, while research institutes at Tsinghua University, Peking University, Shanghai Jiaotong University, and Beijing University of Posts and Telecommunications are initiating research into protection and restoration mechanisms from the perspective of resource management, distributed parallel resource allocation, and contention.

The International Telecommunication Union–Telecommunication Standardization Sector (ITU-T) and the Internet Engineering Task Force (IETF) have brought forward their own solutions. ITU-T defines T-MPLS, and supports 1+1 and 1:1 linear protection (G8131) as well as wrapping and steering ring protection (G8132)^[15]. IETF prefers the Fast ReRoute (FRR)^[16] technique of Multiprotocol Label Switching (MPLS) to implement 1:N linear and ring protection. In 2008, IETF and ITU-T established a Joint Work Team (JWT), and currently, experts within this team are studying ring protection requirements of MPLS Transport Profile (MPLS-TP). Limited by its global labels, PBB Traffic Engineering (PBB-TE) supports 1:1 linear protection, but does not support subnetwork protection and connection-based ring protection.

In China, PTN protection and restoration technologies are still in the early stages of research and discussion, with network device manufacturers focused on

carrier-class protection switching strategies. Key technologies being considered include fault detection and fault notification. These technologies aim to achieve an overall switching time of less than 50 ms, supporting MPLS FRR protection (by enabling local protection of important links and network elements in the network topology), and supporting Link Aggregation (LAG) protection of the links.

In 2008, the China Communications Standards Association (CCSA) established a forum specifically focused on transport devices based on a unified switching plane. In 2010, the forum has turned to T-MPLS protection mechanisms.

Overall, current research into PTN protection mechanisms is focused on improving T-MPLS 1+1 and 1:1 linear protection—as it no longer meets the demands of various services. PTN requires multi-level multi-layer protection and restoration. To achieve this, path-level and link-level protection as well as quick routing or rerouting of services by Generalized MPLS (GMPLS) intelligent software is necessary.

In fact, precursors to packet network protection mechanisms already exist. W. Grover, for example, introduced the concept of P-cycle^[17] (designed for IP networks). Before a fault occurs, a set of P-cycles are pre-configured and routed through all nodes and links to be protected, and resources are reserved for them. When a fault occurs, switching is performed according to the pre-designed ring. In this way, ring protection is achieved. In a public patent, Robert Sultan proposes a server trail failure message notification mechanism. This increases the probability of service recovery in cases of network congestion^[18]. Raymond Xie of Sycamore suggests an intelligent protection and restoration mechanism in EtherOptics-based PTN^[19–20].

2 Challenges for PTN Survivability

Survivability is an important issue that must be addressed before deployment

of carrier-class PTNs can occur. Challenges for PTN include survivability evaluation and optimization, distributed deadlock and resource contention, PTN-oriented protection and restoration blocking, as well as fault detection of OAM.

(1) Survivability Evaluation and Optimization

Among these challenges, survivability evaluation and optimization is particularly troublesome. First, determining reasonable criteria for evaluating survivability is difficult. Second, there are few good optimization methods and technologies for PTN survivability. Perhaps due to limitations in human cognitive ability when compared with computing capability, many optimization issues seem particularly difficult to solve. People simply do not know what the best network is. Moreover, limited computing capability makes the method of exhaustion infeasible. Only algorithms such as genetic and greedy algorithms can be used to find a better solution to survivability optimization in distributed networks.

(2) Distributed Deadlock and Resource Contention

To a greater or lesser extent, PTN may encounter distributed deadlock and resource contention problems when establishing a fast LSP, preempting a fast LSP in 1:N protection, or reselecting a quick LSP. At present, the ostrich algorithm is commonly used to solve these problems. This algorithm meets the minimum requirements of IP networks, but cannot realize fast protection and restoration. Distributed deadlock is an awkward problem affecting computer operating systems and networks, and it has yet to be solved. For computer operating systems, Dijkstra proposes banker's algorithm, but its effectiveness has been proven poor. The deadlock problem receives little attention in the research of MPLS; however, in GMPLS research, Zafar Ali et al have begun to think deeply about the problem and have addressed it to the IETF. Currently, studies into deadlock are underway.

(3) Blocking in Deploying

PTN-Oriented Protection and Restoration

Some protection mechanisms similar to SDH have already been adopted in PTN. However, in distributed routing environments rigid protection paths are set up, and blocking with these mechanisms is very high. To solve this problem, lenient protection and restoration mechanisms must be worked out to complement traditional rigid ones, and to meet the specific requirements of PTN.

(4) Fault Detection of OAM

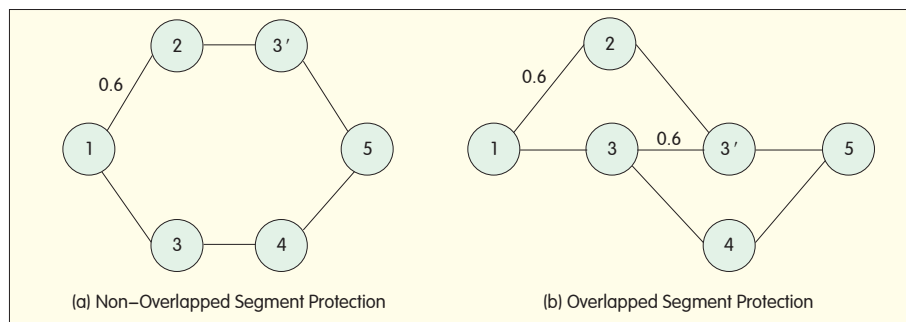
OAM detects Peer-to-Peer (P2P) connectivity status using connectivity detection messages which interact periodically. Along with other methods, it detects packet loss rate, delay, and jitter, and thus achieves loop-back, alarm suppression, and alarm feedback. Moreover, OAM uses Automatic Protection Switching (APS) protocol to protect the two End-to-End (E2E) paths. Whatever the situation, SDH-like protection mechanisms must provide carrier-class protection, and this implies a switching time of no more than 50 ms. The time for the OAM engine to complete path fault detection should not be too long. Specifically, to ensure an overall switching time of less than 50 ms, fault detection and switch triggering must be done 25 ms after a fault occurs. Therefore, OAM fault detection performance is very important.

3 Ring Protection and Survivability Mechanisms for PTN

Most core networks of PTN adopt a mesh structure, so ring protection differs to that of SDH. Here, research carried out by BUPT concerning ring protection and survivability mechanisms of PTN is introduced.

(1) Overlapped Segment Shared Protection Mechanism

After intensive analysis of constraint mechanism and overlapped segment protection mechanism, the 2 mechanisms were integrated, and overlapped segment shared protection algorithm^[21] was used to dynamically adjust the working and protection paths based on link weights. This algorithm



▲ Figure 1. Non-overlapped and overlapped segment protection.

provides multiple overlapped protection segments in the entire working path, enabling reasonable and effective selection of these overlapped protection segments. Compared with traditional protection algorithms, the overlapped segment shared protection mechanism enhances network connection reliability, and improves the utility of network resources by reasonably sharing resources among overlapped protection segments.

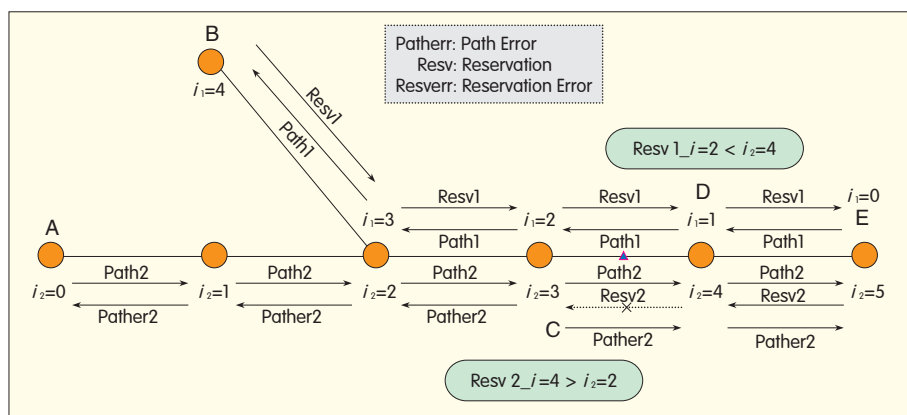
This mechanism protects the entire end-to-end working path using multiple backup protection segments. Compared with path protection, its limited restoration scope is a major advantage. When a link or node fails to work, only its corresponding protection segment is activated. As a result, restoration time is shortened. The mechanism does not provide a protection segment for each link in the working path, so it achieves higher resource utility than link protection. Instead, it provides a backup overlapped protection segment for each segment in the working path, allowing some working links to fall under the protection of several backup segments. This enables flexible network resource allocation and increases the survivability of the network. Traditional non-overlapped segment protection mechanisms cannot protect nodes between segments. Overlapped segment protection mechanism resolves this problem. With neighboring segments overlapped, a faulty node between segments will not lead to failure of both working and protection segments. Overlapped and non-overlapped segment protection is illustrated in Figure 1.

(2) Pre-configured Multi-Cycle (P-mcycle) Scheme

To resolve uncertain restoration of a fault point in mesh networks, a new restoration scheme, P-mcycle, is introduced. This scheme uses Integer Linear Programming (ILP) optimization algorithm (for P-mcycle generation), as well as Sub-cycle secondary routing algorithm. P-mcycle ensures a valid fault restoration time, greatly enhances fault protection success rate, and enhances the efficiency of protection resource reservation. Thus, it improves effectiveness and reliability of restoration in PTN.

P-mcycle generation algorithm is based on ILP. It introduces a P ring for fault-independent path protection. However, the algorithm involves complex computation which often takes a long time. For a small network, P-mcycle generation takes 1 hour; and for a large network, 2 days. In contrast, ILP optimization algorithm leaves variables and constraints such as dynamic service distribution and service symmetry to secondary routing algorithm. A pure ILP model is only used for sourcing optimal P-ring combinations. In this way, the computation involved in P-mcycle generation decreases considerably, and the generation task becomes static planning of network topology.

With the number of variables and constraints decreased, and computational complexity dramatically reduced, an effective pre-selection method can be used to narrow down the number of possible solutions to an ILP problem. In a given network topology, enhanced Depth-First Search (DFS) algorithm is used first to



▲ Figure 2. Conflict-free algorithm.

search all simple rings. Then, the searched rings are checked one by one to determine whether their links satisfy the constraints. M rings with the largest AE values are selected to form a sub-optimal solution space. Finally, ILP optimization algorithm is used to select a group for P-mcycle. P-mcycle obtained in this way can protect all working wavelengths, and ensure the preset maximum reserved wavelength is not exceeded. Service is 100% restored with the least resources.

Secondary routing algorithm simplifies pre-configuration computation in traditional P-cycle, and optimizes protection path selection during switching. With P-mcycle, fault restoration time satisfies the rigid survivability requirements of PTN. Compared with traditional P-cycle, P-mcycle has a higher protection success rate, better resource utility, and better efficiency. Simulation results show that optimization is more pronounced in the Mesh network where the average node density is low.

P-mcycle research will continue to focus on the following aspects: achieving P-mcycle optimization based on node fault restoration and multi-link/multi-node restoration; enabling reasonable distribution of P-mcycle generation resources by taking into account dynamic service distribution changes during candidate ring selection; and studying traffic engineering involved in secondary routing based on the direction of services in order to improve service protection success with fewer network

resources.

(3) Conflict-Free Algorithm

In establishing paths, 2 nodes request the same resources (port or wavelength λ) at both ends of a link and send Reservation (Resv) messages. When the messages reach the other end of the link, the requested resources may already be occupied. In traditional solutions, the node receiving the Resv message treats this as a path error, and returns a Reservation Error (Resvrr) message. If resources at both ends are occupied, the nodes at both ends determine that their paths have not been established. As a result, the resources used for path establishment have been wasted.

Conflict-free algorithm avoids such a problem, as shown in Figure 2. This method works on the principle of adopting a priority comparison strategy for the Resv messages sent at the same time, enabling one path to be established^[22]. Suppose conflict takes place between Node C and Node D, and the resource allocated by Node C for Path 1 is the same as that allocated by Node D for Path 2. In traditional solutions, Resv 1 message sent by Node C arrives at Node D, the requested resource has been occupied, and Node D treats it as a path error. Similarly, Node C registers a path error when it receives Resv 2 message from Node D. Therefore, neither path can be established.

In conflict-release algorithm, a counter i is carried along with a path message. The counter is set to 0 at the source node. A "1" is added whenever

the path message reaches a node, and the new value is stored at the node. When the path message arrives at the destination node, the value of the counter corresponds to the number of links needed to establish a path from the source node to the destination node. As shown in Figure 2, to establish a path from Node A to Node E, 5 links must be passed; from Node B to Node E, 4 links must be passed.

Using conflict-free algorithm for establishing long-distance paths is unlikely to be successful. Suppose there are 2 path establishment requests: Request A involves 7 links, and Request B involves 3 links. If the conflict link is 2 links away from the source node of Request A, and one link away from the resource node of Request B, Request B will succeed in establishing a path but Request A will fail. As a result, the 5 paths that have been established by Request A before the conflict must be released, resulting in resource waste.

(4) Delay Recovery Algorithm

Delay recovery algorithm is used for handling network faults, adopting sequenced delay during network fault recovery.

After a fault occurs, the algorithm allows the source node of the service to send a delayed Notify message. As shown in Figure 3, a time interval t is inserted between two Notify messages. With this method, the probability of resource conflict is reduced in the re-routing and signaling process for fault recovery. When services in two opposite directions are recovered, signaling messages are sent in a certain sequence and no resource conflict occurs.

4 Conclusion

PTN is a future direction in telecom network development. However, ring protection and survivability directly impact its QoS. PTN must resolve issues in fault detection, network deployment, protection and restoration. Existing protection and restoration methods cannot meet the requirements of PTN and new mechanisms must be worked out. These challenges will

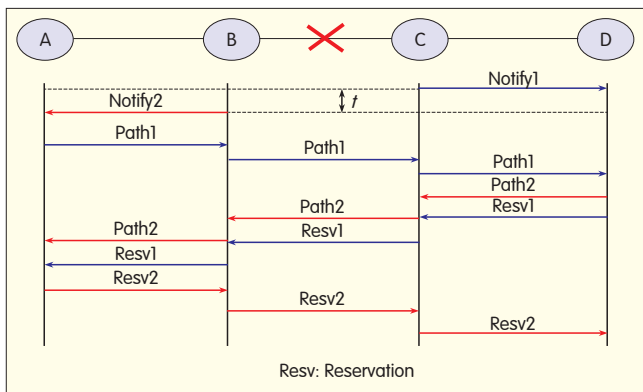


Figure 3.
Delay recovery algorithm.

eventually be solved in the future.

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Biography

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Lu Yueming received his PhD degree from Xi'an Jiaotong University, and is currently a professor at Beijing University of Posts and Telecommunications. He is also a member of the China Computer Federation. He has been engaged in researching computer networks, packet transport networks, and intelligent optical networks for years. He has presided over and participated in 5 nationally funded projects, has been granted 11 patents, and has published over 70 papers.

Roundup

ZTE Tops the Global CDMA Market with a 30% Market Share

IDC's 2010 Global CDMA Market and Prospects research note said that by Q1 2010, ZTE had shipped an accumulated total of over 250,000 base stations, allowing the Chinese company to claim the No. 1 spot in the CDMA market with a global share of 30.3%.

ZTE is the first vendor in the industry to launch the CDMA/LTE dual mode system. ZTE has also steadily enhanced its overall competitiveness

of its CDMA products. It has consolidated its lead in the key emerging markets of China, Indonesia and India, and made breakthroughs in North America. In the Chinese market in 2010, ZTE has undertaken over 80% of the work to migrate Nortel and Motorola CDMA equipment.

By the first quarter of 2010, ZTE's CDMA products were used by more than 120 operators in over 70 countries with an accumulated

wireless capacity of 250 million lines.

A total of four trial and commercial EV-DO Rev.B networks were deployed worldwide including the world's first commercial EV-DO Rev. B network in Indonesia. Meanwhile, ZTE has also cooperated with seven top-tier operators worldwide to deploy commercial LTE networks and almost 50 trial networks in Europe, America, Asia Pacific and the Middle East.

(ZTE Corporation)

Service Adaptation and Label Forwarding Mechanism for MPLS-TP

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Abstract:

Most services borne by transport networks have been transitioned away from Time Division Multiplexing (TDM) towards IP-based. New technology is therefore required to effectively transport packet services. Multiprotocol Label Switching Transport Profile (MPLS-TP) is such a technology which is connection-oriented, capable of multiservice adaptation, and has a flexible label forwarding mechanism. This paper analyzes the characteristics of MPLS-TP, and introduces its latest standardization development. Key issues including data forwarding plane, service adaptation, and label forwarding mechanism are discussed, as well as service implementation of Virtual Private Network (VPN) based on dual-label transport. MPLS-TP applications are summarized in conclusion.

Service networks are undergoing transformation due to the shrinkage of traditional telecom services and the trend towards IP-based services. The future market will require packet transport technology that transmits packet services effectively, but also provides carrier-class Operation, Administration and Maintenance (OAM), as well as protection. Transport devices must change from being adaptable to multiservice interfaces into being adaptable to multiservice kernel. In other words, IP-based services require packet transport networks; but at the same time, traditional services such as voice should be delivered normally. To achieve this, packet technologies and transport technologies must be integrated. And in this context, Packet Transport Network (PTN) is introduced^[1].

There are 2 solutions available for PTN implementation: enhanced Ethernet technologies, and transport technology combined with Multiprotocol Label Switching (MPLS). Provider Backbone Bridge Traffic Engineering (PBB-TE)—or Provider Backbone Transport (PBT)—is representative of the former; MPLS Transport Profile (MPLS-TP)—or Transport MPLS (T-MPLS)—is representative of the latter. Compared with other technologies, these better serve the goals of PTN, and have come into mainstream use. Both are connection-oriented, have performance and reliability similar to Synchronous Digital Hierarchy (SDH), and offer standard, connection-oriented tunnels. Their main differences lie in the way they implement data forwarding, protection and OAM. In the process of standardization, T-MPLS has outpaced PBT. T-MPLS standards have matured, and meet requirements for commercialization, although there is still

room for improvement.

1 MPLS-TP Features and Standardization

1.1 MPLS-TP Standardization

Since 2005, Study Group 15 of the International Telecommunication Union Telecommunication Standardization Sector (ITU-T SG15) has focused on using MPLS technology to define packet transport layer's service function structure (T-MPLS technology). In 2007, ITU-T made recommendations on aspects such as system framework, interface and device specifications, OAM, protection switching mechanism, and service signal adaptation. These recommendations have received support from the Internet Engineering Task Force (IETF). At present, ITU-T is working on detailed definitions of all layer functions, addition of adaptive client signals, and service interconnection and synchronization.

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IETF is preparing T-MPLS Request for Comments (RFC), and defining new labels for T-MPLS services. In 2008, ITU-T and IETF established a Joint Work Team (JWT) to promote the merging of T-MPLS and MPLS. IETF expanded existing MPLS technology into MPLS-TP to better support transport requirements specified by ITU-T. In the future, JWT will develop MPLS-TP standards and ensure consistency between T-MPLS and MPLS-TP standards^[2].

1.2 MPLS-TP Technical Features

MPLS-TP has a clear framework and complete solutions for implementing its key techniques. It simplifies and innovates on MPLS/Pseudo Wire (PW) technology, introducing concepts such as PTN layer, OAM and linear protection. Thus, it meets the demands of PTN^[3]. As a connection-oriented packet transport technology, MPLS-TP consists of a data plane, management plane, and control plane. An end-to-end connection-oriented packet transport tunnel is also established. This tunnel, set up with the network management system or intelligent control plane, is characterized by good operability, and maintenance and protection recovery. MPLS-TP maps client signals into MPLS frames and forwards them using mechanisms such as label switching or label stack. Basic functions of the transport layer—including connection and performance monitoring, survivability and management and control planes—are included^[4-5].

MPLS-TP adopts a dual-label transport mode^[6]. When it provides packet data transport for the client layer, it assigns 2 labels to client data: Common Interworking Indicators (CII) and Transport Label Switched Path (T-LSP). The CII label associates the clients at both ends, and is used by terminals to distinguish client data, while T-LSP label is used for exchange and forwarding client data in the MPLS-TP packet data tunnel.

In short, MPLS-TP is a subset and abbreviated version of MPLS. To support a connection-oriented end-to-end OAM model, many

connectionless features are excluded but protection switching and OAM functions recommended by the ITU-T are present. All these facilitate carrier-class services. MPLS-TP also retains MPLS features that are helpful for data service transport, discards the complicated MPLS control protocol cluster defined by IETF, simplifies the data plane, and removes unnecessary forwarding processing. As a result, it is suited to situations where TDM-based services are being transitioned into IP-based ones.

2 MPLS-TP Service Adaptation Technologies

The unified transport/aggregation platform is a breakthrough in service networking technologies. Service providers need not invest in each network layer to transport mixed services. In MPLS-TP, the transport plane is responsible for packet transport of client data, and also adapting to and forwarding client signals. For different client signals, different signal adaptation and forwarding methods are employed. packet data, cell data and TDM data are quite different in length, format, and multiplexing mode. Therefore, different aggregation, fragmentation, encapsulation, sequencing, timing and multiplexing/demultiplexing methods are used in their adaptive transport^[7].

2.1 Signal Adaptation

Client signals can be directly mapped into T-LSP, or indirectly mapped based on CII. In the dual-label architecture, all types of service signals can be encapsulated in a dual-label structure. The encapsulation layer structures the specified payload signals for transmission over the Virtual Circuit (VC). It consists of 3 sub-layers: payload convergence, timing, and sequencing. The payload convergence sub-layer is tailored to the specific payload type. It can group a number of payload types into a generic class, and then provide a single convergence sub-layer type that is common to the group. The timing and sequencing sub-layers provide generic services to

the payload convergence sub-layer. The functions of the 3 sub-layers are described below.

(1) The payload convergence sub-layer is primarily responsible for encapsulating the payload in VC Protocol Data Units (VC-PDUs). It carries the additional information needed to replay the native data units at the Customer Edge (CE) boundary. When a bitstream is sent to MPLS-TP, one part is peeled off the native service processing module. For example, in structured SDH, fragmentation and circuit overheads may be peeled off.

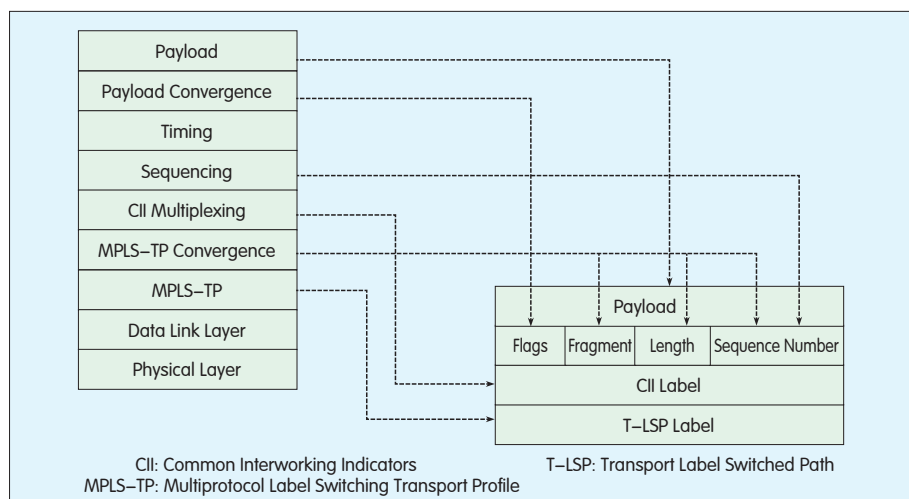
(2) The sequencing sub-layer performs three functions: frame ordering, frame duplication detection, and frame loss detection. Some services must be delivered in order, while other services need not. The choice of strategy for handling incorrect frame sequence, or detected frame duplication or loss depends on the service type. Client signals such as SDH and Frame Relay (FR) may have clock recovery and timed packet delivery timing requirements.

(3) The timing sub-layer performs two functions: clock recovery and timed delivery. Clock recovery involves extracting clock information from the delivered bit stream, and recovering the clock with a phase-locked mechanism. Timed delivery involves delivering non-contiguous VC-PDUs to CEs with a constant phase relation.

IP services can be directly mapped into T-LSP or indirectly mapped by means of dual labels. In dual-label encapsulation, a node does not need L3 forwarding capability. Hence, in networks where nodes do not have L3 forwarding capability, dual-label encapsulation is advantageous. With non-IP service adaptation, VC-based indirect mapping is adopted. As multiple VC multiplexing is transferred in a Label Switched Path (LSP), the service granularity can be less than 2 Mbit/s. Moreover, with CII and label stack added into the services, address space limitation—which occurs in some technologies—can be overcome.

2.2 Service Encapsulation

The generic encapsulation format of



▲ Figure 1. Generic encapsulation format of services.

services is illustrated in Figure 1. The payload containing L2 header or L1 overhead can be IP packet, Ethernet packet, ATM cell, FR cell, or SDH payload. Data information and control words are used for payload convergence. With CII label built in, the VC type in T-LSP can be determined; with T-LSP label built in, MPLS-TP can be determined. Control words often include flags, fragment, length, and sequence number. At the destination end—after terminating the LSP and withdrawing the outer T-LSP label—the device decides which high-layer service instance the data stream belongs to. This is based on the inner CII label.

The transport plane of MPLS-TP has an additional feature: It is transparent to the client layer and service layer. Transparency to the client layer means that any client signal can be carried on MPLS-TP network and transmitted in the form of packets. The client network can be IP, Ethernet, ATM, FR, PDH, or SDH. Transparency to the service layer means that MPLS-TP can use any underlying technology for transport. Before MPLS-TP has its own data link layer protocol, packets can be carried and transmitted via existing Ethernet or SDH networks.

3 Label Forwarding Mechanism

MPLS-TP adopts the dual-label

transport mode in which L3 IP is used to transmit L2 Ethernet data. The 2 labels are CII and T-LSP. To support MPLS-TP networking, T-LSP labels can be infinitely embedded. Therefore, there may be several T-LSP labels. The CII label can be a client signal label; for example, in Figure 2, the CII label refers to VC labels. The multiplexing/demultiplexing module binds several VCs into a VC Group (VCG) and transfers it on the same T-LSP. In this way, the complexity of the transport and switching devices of the network is reduced and less bandwidth is consumed.

3.1 Packet Switching and Forwarding

The service packet switching and forwarding functions of MPLS-TP are mainly involved in exchanging MPLS-TP labels of the packet client data and forwarding the data. Specifically, the functional modules include pre-processing, forwarding and switching, encapsulation, fragmentation, sequencing, timing, multiplexing/demultiplexing, and

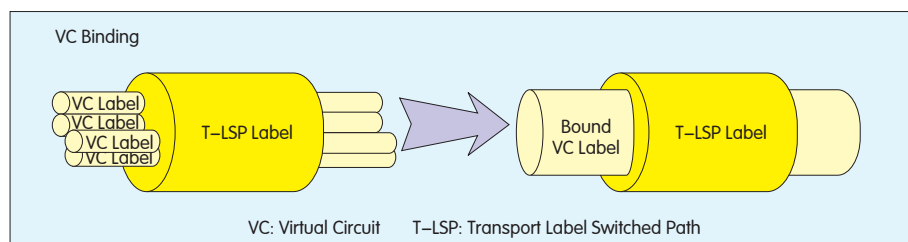
monitoring.

(1) Pre-processing refers to the operations on client data before switching. These operations include data and address conversion, and identification of client data type. Pre-processing simplifies the design of the processing steps that follow.

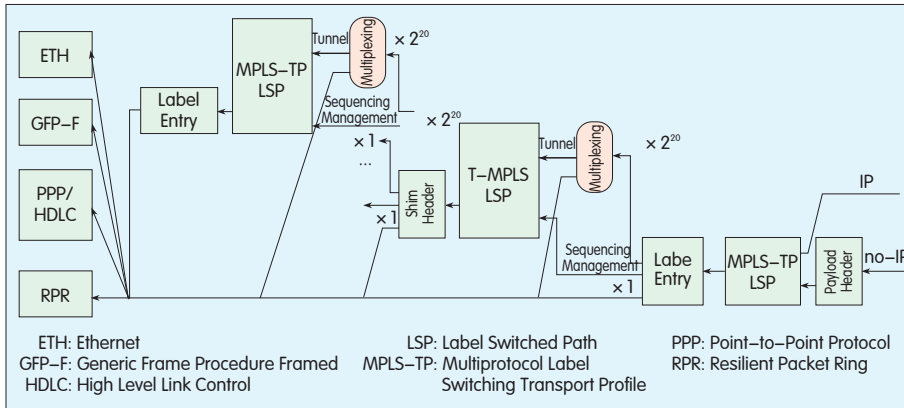
(2) The convergence module is mainly responsible for sorting and converging packets based on type, and importance of client data signals or signaling signals. It dispatches packets to different transport channels, so that different signal types receive different classes of Quality of Service (QoS).

(3) The encapsulation module adapts the signals before they are multiplexed with T-LSP and forwarded. The workings of the module are closely related to the client signal type to be encapsulated. For example, different encapsulation methods are used for packets, cells, and Time Division (TD) signals. Encapsulation is the process of attaching VC and T-LSP labels onto the packets, and inserting OAM information into the packets. Depending on the client signal type, 3 sub-modules may be used: fragmentation, sequencing, and timing. If a client signal exceeds the maximum packet length that can be carried by a service layer network, it must be fragmented. Some clients require sequenced delivery and real-time support, so their signals should be delivered in a sequenced and timed way. A sequencing function provides three services: frame ordering, frame duplication detection, and frame loss detection. SDH and FR client signals may require timed clock recovery and timed packet delivery.

In an MPLS-TP network, the mapping from client signals onto link frames involves client service encapsulation, signal multiplexing, and



▲ Figure 2. An example of MPLS-TP dual-label structure.



▲ Figure 3. MPLS-TP mapping, multiplexing and fragment detection.

mapping of MPLS-TP packets into link frames. Figure 3 illustrates the relationship between information units in an MPLS-TP network.

Client signals can also include OAM packets of T-MPLS networks. Both the data packet and the OAM packets may have a label header for multiplexing. Finally, MPLS-TP packets are mapped onto data link frames, which are transmitted via MPLS-TP topological links. The forwarding and switching module on MPLS-TP terminal devices is responsible for switching the processed client data to related T-LSP and forwarding them. Intermediate transport switching devices continue to forward the packet data with MPLS-TP labels until the data is demultiplexed by the terminal device at the destination. The destination terminal devices forward the data to destination CEs^[8-9].

3.2 Signaling Transfer with Dual-Label Transport Mode

To explain signaling transfer in dual-label transport mode, an example will be provided. Suppose there are 2 Ethernet Virtual Local Area Networks (VLANs): Ethernet VLAN 1 (port 1A, VLAN 100) and Ethernet VLAN 2 (port 1B, VLAN 200). An MPLS-TP VC is established between them based on Label Distribution Protocol (LDP). Figure 4 illustrates a solution for signaling transfer between the 2 Ethernet VLANs.

The process of establishing MPLS-TP VC and transferring signals is described below.

Ethernet VLAN 1 makes a request to

Terminal Device 1 of MPLS-TP network to establish an MPLS-TP VC to Ethernet VLAN 2.

Terminal Devices 1 and 2 of MPLS-TP network negotiate and assign a VC Identifier (VCID) to the VC.

Terminal Device 1 initializes a LDP signaling session to Terminal Device 2 if one is not already present. Once the two terminal devices receive a LDP KEEPALIVE message from each other, they set up the session and are ready to switch the labels bound on the VC.

After the Ethernet VLAN's state becomes "UP", Terminal Device 1 assigns a local CII label (500 in this example) for the VC based on its VCID. It sets up T-LSP 1 for transport over the VC, assigning the VC a T-LSP label

(600 in this example).

Terminal Device 1 converts the T-LSP label into T-LSP tunnel Type-Length-Value (TLV). the local CII label into LABEL TLV, and the CII Identifier (CII-ID) into Forwarding Equivalence Class (FEC) TLV. It then sends a LABEL MAPPING message to Terminal Device 2.

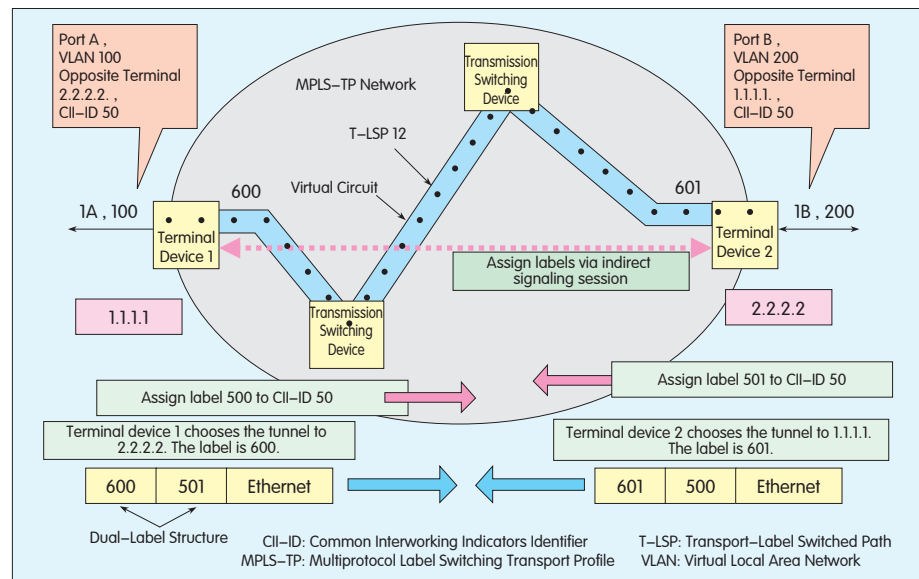
Terminal Device 1 receives a LABEL MAPPING message from Terminal Device 2 and decodes the message to obtain CII label and CII-ID.

At the same time, Terminal Device 2 runs the same steps (1) to (6) independently.

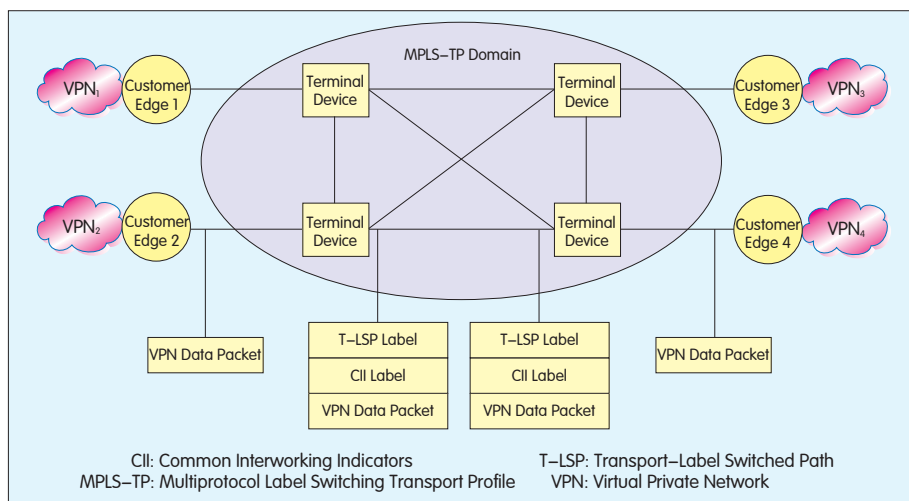
After the two terminal devices have finished label binding, exchanging their labels, and confirming their port parameters are correct, T-MPLS VC with CII-ID 50 is successfully set up. If one Ethernet LAN connection fails to work or is removed, a label cancel message will be sent to the opposite terminal device to cancel the assigned MPLS-TP CII label.

3.3 Dual-Label Transport Mode-based VPN Services

MPLS-TP networks have rich OAM overhead functions^[10] to monitor and manage signals in the network. This improves the operability and security of the entire network, and paves the way for Virtual Private Network (VPN) services in the MPLS-TP network.



▲ Figure 4. Establishment of MPLS-TP VC based on LDP.



▲ Figure 5. Implementation of MPLS-TP VPN.

Traditional VPN technologies (L2 or L3 tunneling technologies) are easy to implement, but are defective in scalability, security, management and maintenance, QoS, and traffic engineering. In particular, when a client adopts multiple access technologies such as Point-to-Point Protocol (PPP), ATM, FR, and Ethernet, the operator must configure several core networks accordingly in order to offer VPN services. VPN services based on MPLS-TP technology can be implemented in a unified network platform that is independent of transport technologies.

MPLS-TP VPN is more easily achieved with point-to-point tunnels set up in the MPLS-TP backbone network. Its CEs do not participate in L3 routing, and the client itself configures routing within the VPN. This makes MPLS-TP VPN independent of the client's L3 network protocol. In MPLS-TP VPN, terminal devices are only responsible for L2 connection, and forwarding between CEs. Functions of L3 or above are performed by CEs. The MPLS-TP network itself can carry different client signals. By means of signal adaptation and encapsulation, a client can use different technologies to access the MPLS-TP VPN, thus satisfying the client's demand for low cost interconnection of VPNs of different layers.

Figure 5 illustrates MPLS-TP VPN. It can be seen that dual labels are

adopted, where a T-LSP label identifies the shared tunnel between terminal devices, and a CII label identifies the dedicated connection between CEs. The CII label, in the form of a MPLS-TP label stack, is multiplexed in the LSP-based tunnel of the MPLS-TP backbone network. The LSP can be regarded as a tunnel carrying several VCs, and a VC is equivalent to a point-to-point route established by the LDP for VPN users.

4 Conclusion

With good service adaptation and label forwarding capabilities, MPLS-TP technology is well-suited for packet transfer. However, it is rivaled by other packet transport technologies such as PBT. As an emerging technology, MPLS-TP needs the joint efforts of standardization organizations, telecom operators, and equipment vendors to hasten its maturity.

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Biographies

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PTN Clock Synchronization Technology and Its Applications

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Abstract:

Clock synchronization is an important issue for packet transport networking. Current clock synchronization technologies include synchronous Ethernet, IEEE 1588v2, and Network Time Protocol (NTP). However, individually, these technologies are beset with certain problems. Synchronization Status Message (SSM) algorithm for synchronous Ethernet standards suffers clock ring, and has difficulty tracing and counting nodes. An extended SSM algorithm can improve clock synchronization. NTP is too imprecise to meet the requirements of telecom networks, yet IEEE 1588v2 alone can lead to slow convergence time and influence time delay precision when the network is heavily loaded. ZTE therefore proposes an IEEE 1588v2 solution based on synchronous Ethernet in order to effectively raise the precision of Packet Transport Network (PTN) time synchronization.

As operators seek to replace their conventional Time Division Multiplexing (TDM) transport networks with Packet Transport Networks (PTN), clock synchronization comes into consideration. There are two requirements for PTN synchronization:

- (1) It must be able to bear TDM services and provide a clock recovery mechanism.
- (2) PTN should have a high-precision network reference clock for synchronization of network nodes.

1 Overview of Synchronization Technologies

Clock synchronization includes frequency synchronization and time synchronization. The former implies identical timing intervals, while the latter implies identical starting time as well as intervals. Different wireless standards have different requirements for clock bearing.

Since 2004, the ITU-T Q13/SG15 has made a number of proposals relating to

PTN, including G.8261 (defining general requirements), G.8262 (defining equipment clock performance), and G.8264 (defining system architectures and synchronization function modules).

IEEE 1588 standards were released in 2002 with the view of defining Precise Time Protocol (PTP) for a LAN multicast environment.

However, standards released specifically for this purpose could not be applied to telecommunications environments with greater complexity. IEEE 1588v2 was therefore issued in 2008 for application in telecom networks^[1-5]. The IETF Network Time Protocol (NTP) implements time synchronization between users and between time servers in the Internet.

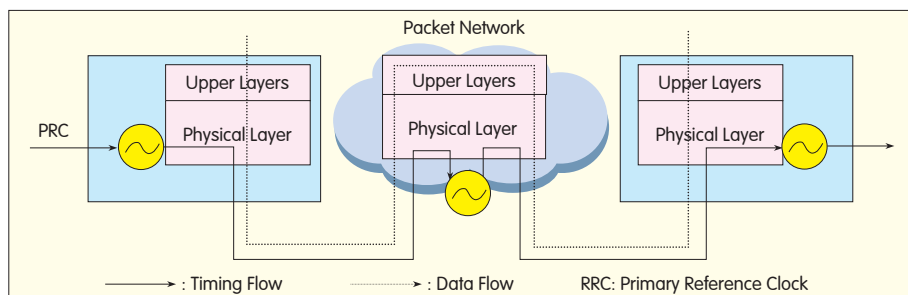
2 Synchronous Ethernet Technology

Physical layer synchronization technology is widely applied in traditional Synchronous Digital Hierarchy (SDH) networks. Each node in an SDH can extract the clock from

the physical links or obtain the clock from external synchronous interfaces. A node selects the best quality clock from among the multiple clock sources, and sets local clocks accordingly. Then the locked-in clock is transported to equipment at lower layers. Each layer of the network becomes synchronized to the Primary Reference Clock (PRC) through level-by-level locking. Similar physical-layer synchronization technology can be used in packet networks, as shown in Figure 1.

2.1 Principle of Synchronous Ethernet

Synchronous Ethernet technology in PTN uses Ethernet link streams to recover clocks. Ethernet physical layer coding adopts 4B/5B (FE) and 8B/10B (GE) technology, in which one additional bit is inserted into every four bits. In this way, four consecutive 1s or four consecutive 0s will not appear in any transported data stream, and the clock information is effectively contained. High-precision clocks are used to transport data from the Ethernet source-end interface, and are recovered and extracted at the



▲ Figure 1. Physical-layer synchronization in PTN.

receiving end.

The principle of synchronous Ethernet is shown in Figure 2. A high-precision clock is embedded into the physical layer chip of equipment at the source side (Node A). This clock is used by the chip to send data out. The physical layer chip of equipment at the receiving side (Node B) extracts this clock from data streams. Precision is not damaged in this process and synchronicity with the source end is maintained. The principle of clock transport in synchronous Ethernet is similar to that of an SDH network—the clock is recovered from the Ethernet physical link. As a result, the quality of recovered clock is not influenced by link service traffic, and clock tree deployment and quality is the same as that of SDH/SONET. This satisfies the timing interface index of G.823.

2.2 SSM Algorithm for Synchronous Ethernet

Synchronization Status Message (SSM) algorithm originates from SDH clock synchronization control. Its application rules and clock selection algorithm conform to the ITU-T G.781 standard, and control of synchronous Ethernet retains SDH network characteristics. Moreover, based on the conventional clock network, SSM supports synchronous Ethernet by building in Ethernet Synchronous Message Channel (ESMC). As described in G.8264, ESMC is a unidirectional broadcast protocol channel of the Media Access Control (MAC) layer used for transporting SSMs between different equipment. The equipment selects the best clock source based on SSM in ESMC messages.

Standard SSM algorithm is effective

for network clock synchronization.

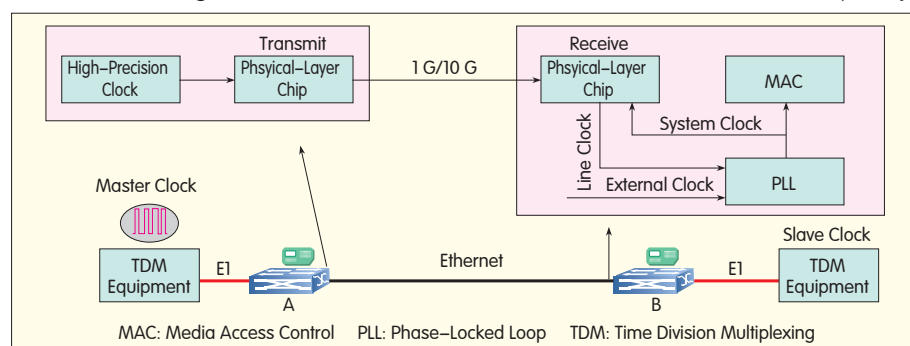
However, it has two weaknesses:

(1) It cannot satisfactorily handle the problem of synchronous clock ring.

(2) It has clock signal attenuation. With an increase of synchronous links, drift caused by noise and temperature changes in synchronization allocation deteriorate timing reference signals. Consequently, the number of synchronized Network Elements (NEs) is limited on the same synchronous link, and standard SSM has difficulty tracing and counting nodes.

For PTN, ZTE proposes an improved and extended SSM algorithm. ESMC packet uses two Type-Length-Values (TLVs) to transport SSM messages. Following the ITU-T standards, the first TLV transports original SSM byte information at the synchronization quality level. The second TLV is used for path protection. The improved algorithm has the following advantages:

- It fundamentally prevents clock ring;
- It automatically selects the optimal (shortest) route in the event of multiple clock paths;
- If there is a route to the master clock, NEs will trace the master clock instead of entering a state of free



▲ Figure 2. Synchronous Ethernet.

oscillation;

- The algorithm is ideal for processing because it supports distributed processing at lower layers and each NE is in an equal position;
- Due to the direct use of standard S1 bytes, there is no problem interworking with other vendors' equipment.

3 Time Synchronization Technology

Time synchronization technology is a progression from frequency synchronization. By using the packet protocol data unit to carry clock/time messages, PTN time synchronization technology can synchronize the master clock and slave clock. Figure 3 shows the basic principle.

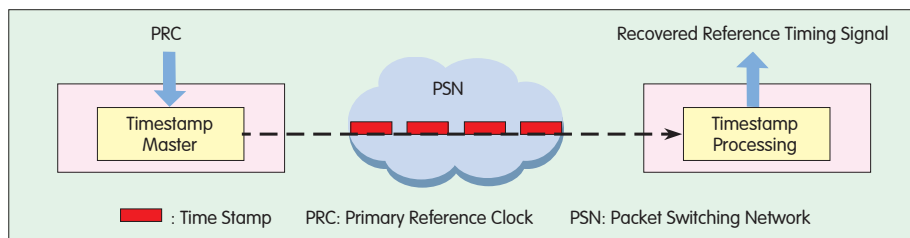
3.1 Network Time Protocol (NTP)

Before the advent of IEEE 1588v2, there were three main types of PTN time synchronization protocol: time protocol, daytime protocol and NTP. NTP is fulfilled purely by low-precision software; the widely used NTPv3, for example, achieves synchronization precision of 10 ms. IETF is now proceeding with NTPv4 standards that support IPv6 and dynamic discovery servers. Synchronization precision of NTPv4 is expected to reach a 10 μ s level. However, the stability and precision of NTP still fails to meet the high requirements of telecom networks.

3.2 IEEE 1588v2 Protocol

3.2.1 Implementation Principle of IEEE 1588v2

IEEE 1588v2 unifies time and frequency



▲ Figure 3. Packet protocol synchronization mode.

synchronization. In the future, it will be suitable for inter-office time-frequency transport of different platforms. It is capable of unidirectional frequency transport using Time over Packet (ToP) based on IEEE 1588v2 timestamp. It also realizes time synchronization by using IEEE 1588v2 protocols.

Therefore, it is widely applied in PTNs.

IEEE 1588v2 time synchronization is achieved by encoding time signals through use of master and slave clocks. Both master and slave time are synchronized by bidirectional message interaction as well as by employing network symmetry and delay measurement technology.

According to the 1588v2 principle shown in Figure 4, $\text{Delay} = (T_2 - T_1 + T_4 - T_3)/2$; $\text{Offset} = (T_2 - T_1 - T_4 + T_3)/2$.

Messages such as Sync, Follow_Up, Delay_Req, and Delay_Resp are delivered between the master and slave clocks. Referring to the four values T_1 , T_2 , T_3 and T_4 , the delay and offset between the master and slave can be determined.

Synchronization messages include general messages and event messages. A general message, such as Follow_Up, does not handle timestamp itself; it can carry accurate delivering/receiving time of event messages such as Sync. In addition, it implements network configuration and management, and fulfills communication among PTP nodes. An event message needs to deal with timestamp, and can carry timestamp or not. Depending on the event message timestamp or general message timestamp, the slave clock can determine the path delay and offset between the master and slave clocks.

3.2.2 Clock Types

Based on protocols such as Ethernet,

IPv4/IPv6, and User Datagram Protocol (UDP), IEEE 1588v2 defines three basic clock types: Ordinary Clock (OC), Boundary Clock (BC), and Transparent Clock (TC).

OC is a single-port device acting as either the master or slave clock. Only one master clock is permitted in a synchronous domain. Frequency, accuracy, and stability of the master clock is directly related to the performance of the entire synchronous network. PRC or GPS is generally adopted for network synchronization. Slave clock performance determines the timestamp precision and sync message rate.

As a multi-port device, BC can connect with many OCs/TCs. Among the multiple ports of BC, one serves as the slave port, connecting to the master port of the master clock or other BCs. The other ports serve as master ports, connecting to the slave port of the slave clock or BC at the next layer. They can also serve as backup ports.

TC is used for connecting the master clock with the slave clock. It forwards interacted synchronous messages between the master and slave clocks in a transparent way. It also calculates local buffering treatment time for synchronous messages—such as Sync and Delay_Req—and writes this time into the CorrectionField byte block of the sync messages. The slave clock calculates delay and offset based on byte values and sync message timestamp in order to realize synchronization. TC can be divided into Edge-to-Edge (E2E) TC, and Peer-to-Peer (P2P) TC.

3.2.3 Delay in IEEE 1588v2

Delay is a major factor affecting precision of an IEEE 1588v2 system.

(1) Timestamp Treatment Delay

IEEE 1588v2 timestamp treatment is performed by hardware. The timestamp processor is located between the physical layer and MAC layer, as shown in Figure 5.

Hardware timestamp treatment compensates for the time taken by IEEE 1588v2 protocol frame to pass through a protocol stack. Therefore, precision of message delivery and timestamp receiving at the port is guaranteed.

(2) Node Buffering and Path Delay

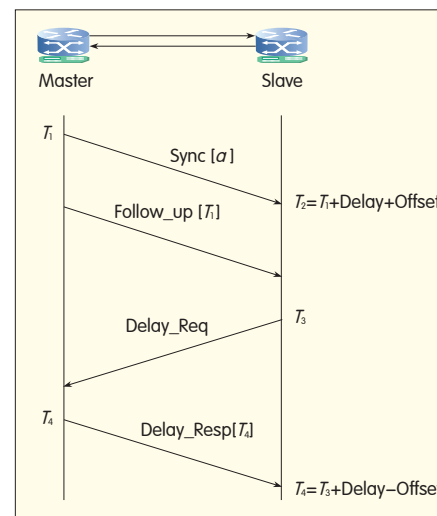
IEEE 1588v2 defines E2E TC and P2P TC for compensating node buffering delay. There are two methods of transport path compensation: delay request-response, and peer delay.

Delay request-response works with E2E TC. This TC marks timestamp treatment in the packet at ingress and egress, while the slave completes all the calculations on time delay compensation.

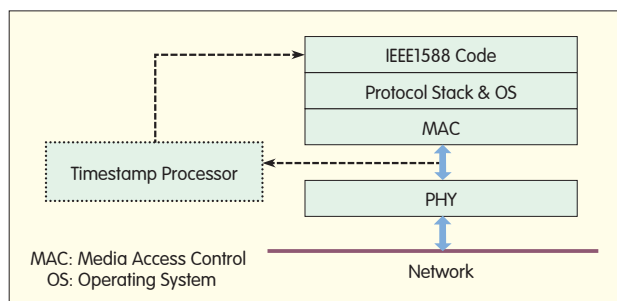
Peer delay works with P2P TC. This TC calculates time delay among endpoints, and each endpoint interacts with the TC respectively to compute P2P time delay. Based on these calculations, the slave then calculates delay compensation.

3.2.4 Fulfillment of IEEE 1588v2 in PTN

IEEE 1588v2 synchronization precision is affected by many factors in real network deployment. Its application in complex network environments, such as hybrids of microwave and switching networks, is still in the research stage.



▲ Figure 4. Principle of IEEE 1588v2 protocol.



◀ Figure 5.
IEEE 1588v2 timestamp
treatment.

IEEE 1588v2 can achieve precision of 100 ns in a pure PTN testing network. However, due to the complexity of network time delay and lack of control over IEEE 1588v2 bidirectional path asymmetry, there are unpredictable risks when relying solely on IEEE 1588v2 protocol and analytic algorithm to adapt to real networks. When a network is heavily loaded, frequently sent IEEE 1588v2 message packets are easily influenced by service packets, and this greatly impacts time delay precision. However, by decreasing the frequency of sending message packets, the time convergence rate may be slowed. In addition, the IEEE 1588v2 algorithm must be compensated with bidirectional path asymmetry—achieved by optical fiber—in actual network construction. Timing error of optical fiber asymmetry is first measured by costly time synchronization testers and oscilloscopes, and is followed by asymmetrical time delay compensation. Owing to the large number of PTN nodes, compensation work requires many engineers. Instruments such as time synchronization testers and oscilloscopes are also cumbersome. It is therefore difficult to apply IEEE 1588v2 widely in real network construction, and opinions still vary about its feasibility.

ZTE Corporation proposes a synchronous Ethernet-based IEEE 1588v2 time transport solution to the problem. The core idea is to establish a highly controllable network where clocks are isolated from time, eliminating unpredictable risk. This is beneficial for fast convergence of time synchronization in order to implement IEEE 1588v2 based on steady frequency synchronization of the

synchronous Ethernet physical layer.

Moreover, it reduces the frequency of IEEE 1588v2 message packet delivery. Time precision is not affected even if the network is heavily loaded, which improves reliability and precision of PTN time synchronization. To solve the engineering problem in PTN asymmetry measurement, ZTE configures its access-layer PTN equipment with a time error measurement function that quickly, easily, and accurately performs measurements without the need for professional instruments.

4 Typical Applications

4.1 Application of Synchronous Ethernet

Similar to SDH, synchronous Ethernet supports both ring networking and tree networking. Generally, RNC provides the clock source, and clock information arrives at every base station via synchronous Ethernet in order to maintain synchronicity throughout the

entire network. Tree networking does not support clock routing protection. However, in ring networking, when current clock routing fails, related NEs can receive messages such as warning and SSM, and trace the source clock from other directions in order to protect clock routing. Figure 6 shows examples of synchronous Ethernet networking.

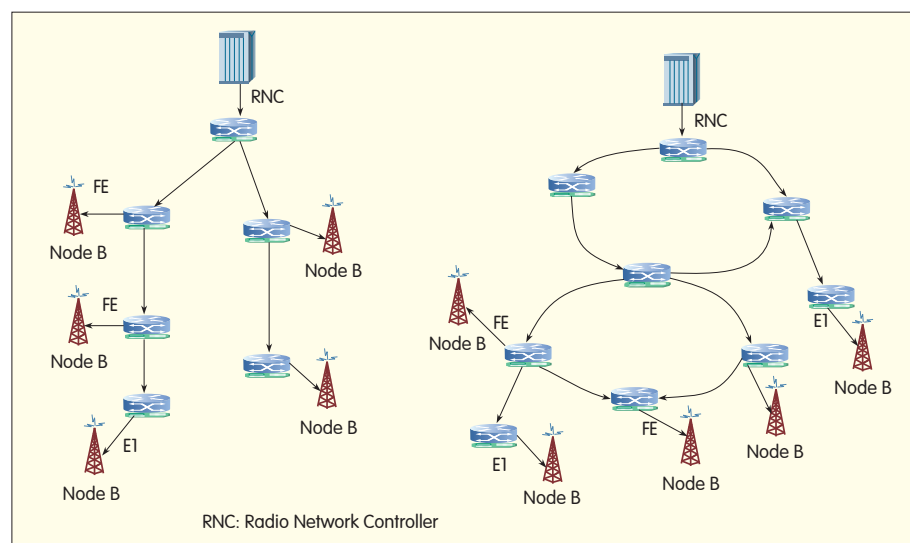
Jittering of synchronous messages is intensified after they have been transported by NEs. Therefore, optimal clock quality is achieved when network equipment is able to trace the clock source using the shortest possible path. ZTE's PTN adopts an extended SSM algorithm which includes the number of nodes the clock has passed through. In any case, NEs are able to trace the clock source using the shortest possible path.

NE C can use point B or D to trace clock information with its source at A. The clock passes through only one node when tracing from point B, and two nodes when tracing from point D. To obtain higher clock quality, ZTE's PTN automatically chooses the clock traced from B.

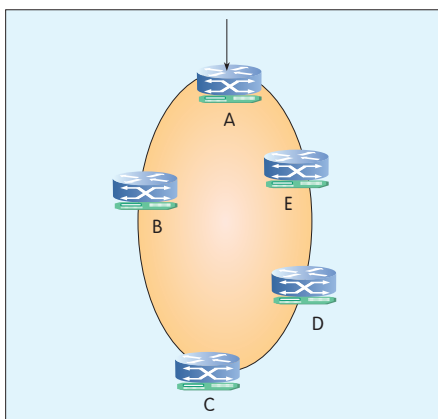
4.2 Applications of IEEE 1588v2

4.2.1 Replacing GPS at Base Stations

Typical IEEE 1588v2 application involves replacing GPS at Radio Access Network (RAN) base stations. Installing a GPS antenna at a



▲ Figure 6. Examples of synchronous Ethernet networking.



▲ Figure 7. An example of clock tracing.

TD-SCDMA or CDMA2000 base station requires an area of 120 degrees to be cleared, and this can be difficult given environmental constraints. GPS is difficult to install indoors and underground. Moreover, GPS is expensive and has a high failure rate. If PTN can provide time synchronization for base stations, it can replace GPS (or

act as the GPS backup) and offer higher security guarantee for RAN.

Figure 8 shows an example of IEEE 1588v2 replacing GPS at base stations. Only one NE in the PTN needs to receive time messages from GPS; for example, through the 1PPS+ToD interface. PTN delivers the time messages to other NEs by IEEE 1588v2 protocol, then the messages arrive at the base stations via Ethernet or other interfaces. Time synchronization among all the base stations is fulfilled accordingly.

Base stations support either IEEE 1588v2 protocol or time interfaces. If supporting IEEE 1588v2 protocol, PTN works in TC mode; if not, PTN works in BC mode.

4.2.2 Frequency Recovery

IEEE 1588v2 is also used to recover frequency by ToP. Most operator networks are ordinary data networks, not supporting synchronous Ethernet.

In this case, IEEE 1588v2 can be used to obtain time frequency in these ordinary networks.

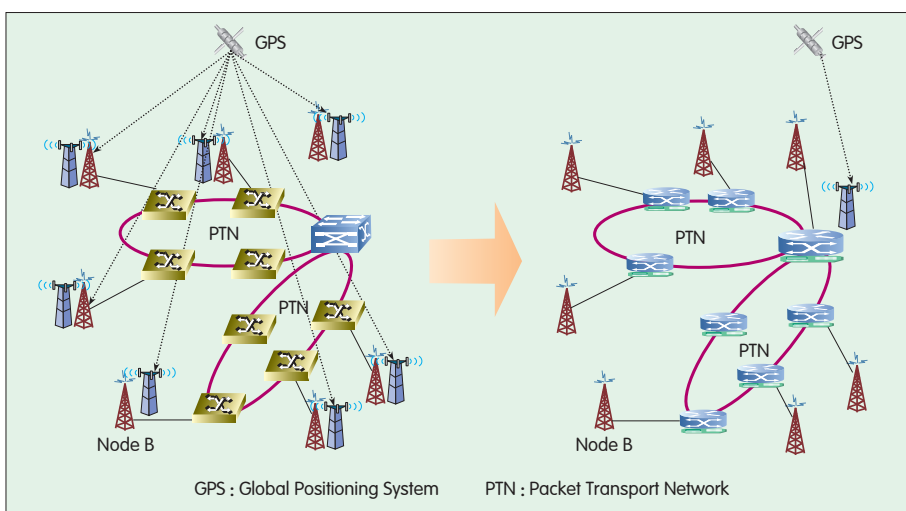
Figure 9 shows an example of frequency recovery 1588v2 networking. When the network between equipment A and equipment B is an ordinary data network, IEEE 1588v2 Sync messages are transported from A to B via the ordinary data network. Equipment B uses IEEE 1588v2 to recover the clock. The recovered clock is taken as the reference source of B, with which the service clock is recovered.

5 Conclusions

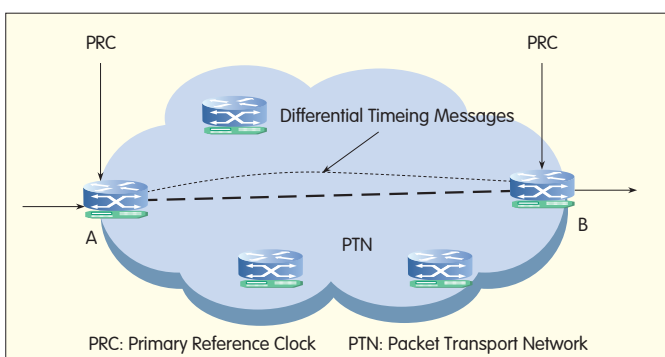
Research into PTN clock synchronization technology is deepening. ZTE proposes an extended synchronous Ethernet SSM algorithm and IEEE 1588v2 solution based on synchronous Ethernet. This solution plays an important role in raising the precision of time synchronization in PTN and reducing the level of engineering complexity. PTN clock synchronization technology has extensive application possibilities, especially in RAN, TDM services, M2M real-time data acquisition, and VIP private networks.

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- [5] MEF 22. Mobile backhaul implementation agreement phase1 [S]. 2009.



▲ Figure 8. An example of IEEE 1588v2 replacing GPS at base stations.



◀ Figure 9. Networking of frequency recovery by IEEE 1588v2.

Biography

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Li Qin received his master's degree from Beijing University of Posts and Telecommunications. He is a PTN solution manager at the Bearer Network Planning System Department of ZTE, and is mainly engaged in the study of 3G/LTE PTN bearer solutions.

PTN and IP-Based Mobile Backhaul

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Abstract:

Packet Transport Networks (PTN) combine the advantages of packet technology and Synchronous Digital Hierarchy (SDH) technology. Taking the packet switch as its core, PTN has a high statistical multiplexing ability, which allows it to become a more efficient packet transfer service. Its strong Operation Administration and Maintenance (OAM)—similar to SDH and carrier-class security protection services—ensures efficient mobile backhaul business management and transmission quality. MPLS Transport Profile (MPLS-TP) packet transport network eliminates connectionless features, such as Penultimate Hop Popping (PHP), label merge and Equal-cost multi-path (ECMP), and is enhanced in terms of OAM, protection, and synchronization. This is ideally suited for carrying IP-based mobile backhaul services and key account services. PTN, original Multi-Service Transport Platform (MSTP), Metro Ethernet, and IP over Wave Division Multiplexing (WDM)/Optical Transport Network (OTN) of organic complexes, contribute to an "all IP" era of telecommunications services.

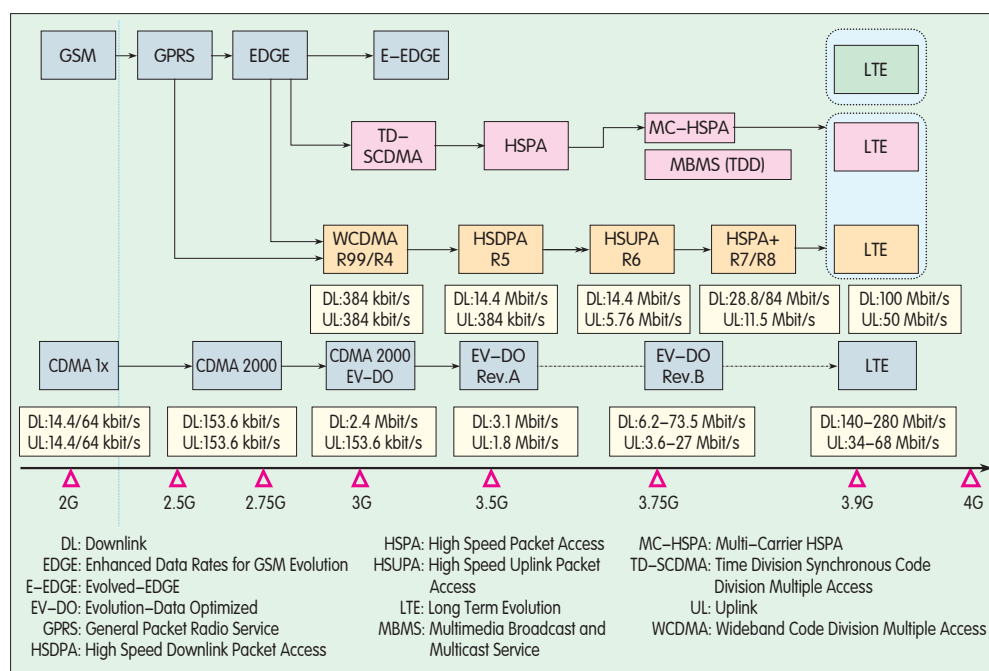
With the rapid development of mobile communication technologies, 3G and Long Term Evolution (LTE) systems exist not just on paper, but are beginning to be deployed in reality. While the development of mobile technologies has bright prospects for the future, it also means new challenges for mobile backhaul networks.

The trend towards all-IP telecommunications services is reflected across the whole mobile communications industry. Services are being transitioned away from Time Division Multiplexing (TDM) voice to IP-based voice and data. Because IP technologies are being widely used in 3G networks and mobile data services continue to grow, pressure is mounting on 3G mobile backhaul networks for better service sensing, higher Quality of Service (QoS), and more efficient statistical multiplexing. The change of interface from E1 to FE has also led to a sharp increase in interface bandwidth. In LTE systems, base stations may even adopt a GE interface with a rate of 1,000 Mbit/s.

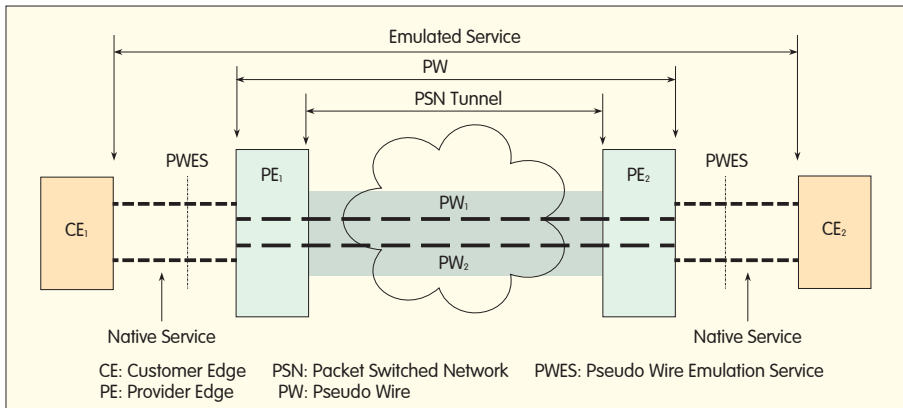
Figure 1 illustrates the evolution of mobile communication technologies. It can be seen that uplink and downlink bandwidths and rates have substantially increased during the evolution from 2G to 3G/LTE. This increase in interface rate means that

much more bandwidth is required for transport networks, and transmission efficiency must be improved in order to reduce costs.

In the 2G era, leading technologies of mobile backhaul networks included Synchronous Digital Hierarchy (SDH)



▲ Figure 1. Evolution of mobile communication technologies.



▲ Figure 2. Edge-to-edge services in PTN.

and Multiservice Transport Platform (MSTP), both designed to aggregate and efficiently transmit TDM-based circuit services. In particular, MSTP was designed to enable IP services to be carried over transport networks. It is not, however, an all-IP solution. Its user interface may be IP-based but at its core, it is still TDM circuit-switching and adopts a fixed tunnel for carrying packet services. As a result, MSTP suffers defects such as low transmission efficiency, high cost, and poor scalability when carrying and transmitting variable-length packets, bursty IP, and Ethernet services. In short, SDH and MSTP cannot meet the requirements of 3G/LTE systems, and they restrain the development of mobile services. Packet Transport Network (PTN)—integrating packet technologies and SDH—is therefore introduced.

1 Technical Features of PTN

3G/LTE systems can be broadband and IP-based broadband. To accommodate these, a mobile backhaul network must have traditional capabilities such as efficient statistical multiplexing, flexible service sensing, and differentiated QoS. As a carrier-class service bearer network, it should also inherit the transmission characteristics of old systems, including end-to-end service management, hierarchical Operation, Administration and Maintenance (OAM), and carrier-class protection. PTN technology has all these

capabilities.

PTN takes connection-oriented packet technologies as its core, and adopts the abovementioned transmission characteristics. As an integrated transport technology, it is used mainly to carry carrier-class Ethernet services as well as TDM and Asynchronous Transfer Mode (ATM) services. Its packet core uses bandwidth efficiently and provides flexible tunnels with powerful statistical multiplexing capability. So PTN is particularly suitable for packet services which are characterized by bursts. It inherits SDH-like transmission characteristics, powerful OAM, carrier-class protection, and Graphical User Interface (GUI) network management. As a result, users feel a sense of continuity and consistency with traditional mobile backhaul networks^[1].

Two technologies are currently available for PTNs: Multiprotocol Label Switching Transport Profile (MPLS-TP)^[2] and Provider Backbone Bridge Traffic Engineering (PBB-TE). MPLS-TP is an extension of core network technologies. Based on Multiprotocol Label Switching Transport (MPLS) of IP-based core networks, it simplifies the transport plane and complex control protocols. Moreover, by eliminating connectionless features, it enhances OAM and protection switching mechanisms, guarantees reliable QoS, and provides statistical multiplexing. PBB-TE is a connection-oriented transport technology and an expansion of Local

Area Network (LAN) technologies. It is based on Media Access Control (MAC)-in-MAC^[3] specified in IEEE 802.1ah. In PBB-TE, the MAC address self-learning function is disabled, but network management and control functions are enabled. At present, MPLS-TP is the de facto mainstream technology for PTNs.

MPLS-TP standardization is led by the International Telecommunication Union-Telecommunication Standardization Sector (ITU-T) and the Internet Engineering Task Force (IETF). These two organizations established a Joint Working Team (JWT) in February 2008, and have made great progress in MPLS-TP standardization. As at February 2009, the JWT had worked out five RFC documents, two recommendations, and 13 drafts. Key standards of MPLS-TP are expected to be released in 2011.

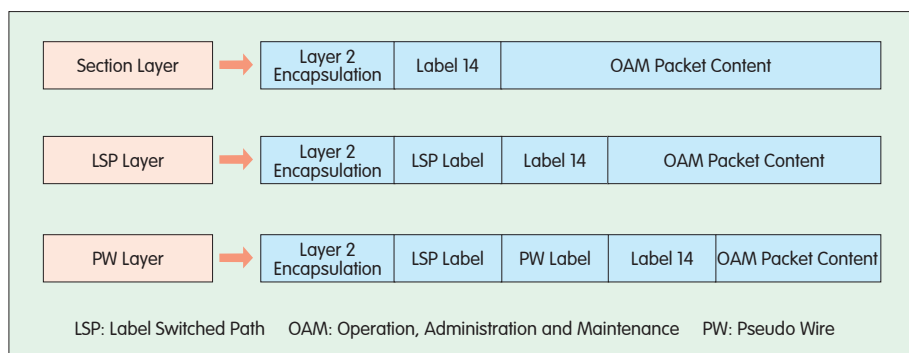
2 Key Technologies of PTN

PTN is an integrated transport network technology featuring both packet and transmission. It has become a mainstream solution for IP-based mobile backhaul networks in 3G/LTE systems due to several key technologies.

2.1 Pseudo Wire Emulation Edge-to-Edge (PWE3)

3G systems are developing at pace but traditional TDM services will continue to be the main source of profit for telecom operators for some time. Therefore, PTN should be capable of carrying multiple services. PWE3 is designed specifically for this purpose. By using PWE3 circuit emulation, MPLS-TP adapts itself to all types of customer services—including Ethernet, TDM, and ATM—and provides dedicated edge-to-edge tunnels for these services.

PWE3 is a service emulation mechanism which encapsulates service data in a special circuit emulation header. This header includes the basic attributes of service data such as frame format, alarm, signaling, and synchronous timing^[4]. Edge-to-edge services in the PTN are illustrated in Figure 2. In PWE3, a Pseudo Wire (PW)



▲ Figure 3. OAM packet encapsulation formats specified in ITU-T G.8114.

is set up and maintained in a Packet Switched Network (PSN) tunnel. A Provider Edge (PE) uses the PW to encapsulate service data for transmission, keeping the attributes and features of the service as much as possible. For a Customer Edge (CE), the PW is merely a link or circuit occupied exclusively by a specific service. This circuit is called virtual circuit. The CE does not recognize the existence of the core network and treats all services it processes as native services.

2.2 QoS

In a SDH-based mobile backhaul network, an exclusive rigid tunnel is provided for all services. This transmission mode is highly reliable, but on the other hand leads to transmission waste because services may have different characteristics. Voice services, for example, with high real-time characteristics, have different transmission requirements to common Internet services. PTN can sense service characteristics and provide transmission as required. For telecom operators, offering suitable transmission for different service flow is cost-efficient, especially when the demand for bandwidth increases substantially.

2.3 Hierarchical OAM and Carrier-Class Protection

The OAM mechanism of PTN basically derives from SDH, and provides alarm and performance management for the section layer, Label Switched Path (LSP) layer, and Pseudo Wire (PW) layer. By supporting hierarchical OAM,

this mechanism can quickly locate a PTN fault, and detect network performance issues such as packet loss rate and delay^[5].

ITU-T specifies the OAM packet encapsulation formats^[6] for Transport Multiprotocol Label Switching (T-MPLS), as shown in Figure 3. The OAM signaling packets of each layer also contain labels encapsulated with MPLS. To distinguish OAM signaling packets from users' service data packets, a special label (14) is defined. By defining a series of OAM protocol packets, G.8114 implements diverse OAM functions.

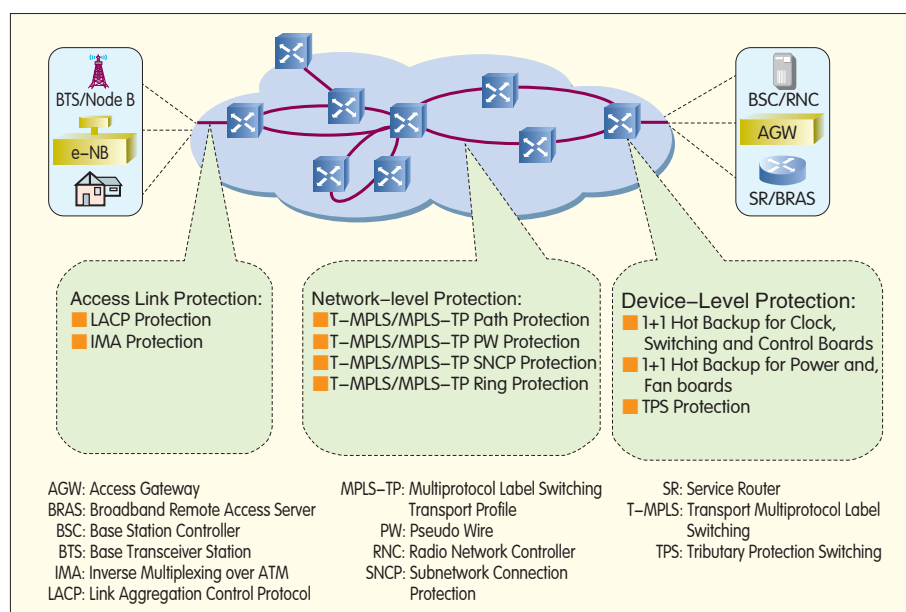
PTN's all-round carrier-class protection is illustrated in Figure 4. PTN provides access link protection, network-level protection, and device-level protection. It also

supports ring, linear, and mesh protection modes. Each mode has its own merits and shortcomings, suitable for different scenarios. Ring protection can provide a high level of protection specifically for ring topology^[7], while linear protection is not specific to any topology and can be implemented with two connections for each path in a fixed network. In application, these modes are often used in combination in order to achieve carrier-class service protection.

OAM and protection are closely related to each other. Using the OAM mechanism to achieve timely fault detection is a premise for carrier-class protection. There are three types of OAM related to the PTN protection mechanism: alarm-related, performance-related, and communication channel-related.

2.4 Synchronization

The development of mobile communication technologies also has implications for time synchronization in mobile backhaul networks. ITU-T, IEEE, and other standardization organizations have proposed numerous time synchronization solutions. Among them, the most attractive is precision clock synchronization protocol suggested in the IEEE 1588v2 standard^[8]. This protocol adopts a



▲ Figure 4. All-around carrier-class protection of PTN.

master/slave clock configuration and encodes the time before transmission. Taking advantage of the symmetry of network links and delay measurement technology, it synchronizes frequency, phase, and the absolute time of master and slave clocks.

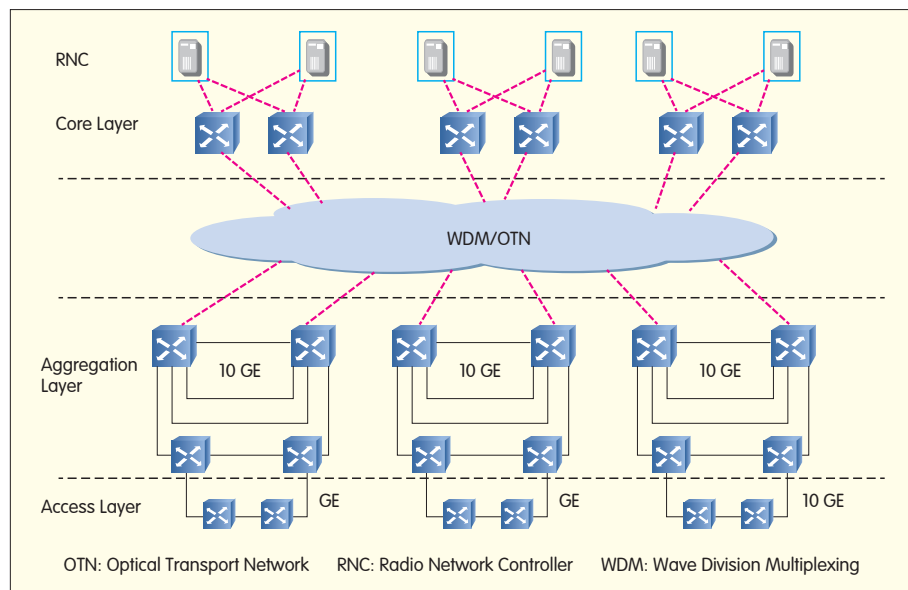
ZTE has been the first to adopt IEEE 1588v2 Precision Time Protocol for clock synchronization based on synchronous Ethernet^[9]. The core idea of this solution is to set up a network with separate but highly controllable clock time in order to avoid unpredictable risks. With synchronous Ethernet at the physical layer, frequency synchronization is implemented among nodes, thus controlling the nodes' phases within a certain range and guaranteeing accurate frequency. IEEE 1588 protocol is only used for phase tuning and time transfer. A clock dedicated network constructed in this way is connection-oriented, strictly controllable, and point-to-point. It avoids interference from asymmetrical IEEE 1588 systems during deployment, and its time synchronization performance is independent of network loader traffic.

3 Relationship between PTN and Other Bearer Networks

3.1 PTN vs. MSTP Network

In the 2G era, SDH/MSTP-based networks provided stable and reliable transmission for mobile services and were widely applied in mobile networks. In 3G/LTE systems where PTN has been introduced, the issue arises of handling the relationship between PTN and the existing MSTP network. Taking into account network complexity, construction expenses, and operation and maintenance costs, an independent PTN plane is able to be set up in addition to the existing MSTP plane.

Although this requires a large investment in the early stages of PTN construction, the advantages are significant. First, existing 2G services can be carried by the MSTP network



▲ Figure 5. Flexible networking of PTN services with WDM/OTN.

without being affected during 3G construction. Second, 3G services can develop smoothly, bandwidth demand of future data services can be easily satisfied, and mobile networks can evolve gracefully to LTE. This solution provides a clear structure of the mobile backhaul network, greatly facilitating future planning, management, and maintenance of the network.

Depending on the availability of MSTP resources in the existing network, a networking mode comprising a mixture of PTN and MSTP can be adopted in some areas of the network as a supplement to the above solution. That is to say, PTN devices can be introduced into the network access layer or aggregation layer of existing MSTP network where there is bandwidth shortage. This not only implements cross-area resource dispatching, but also enables the operator to accumulate experience in low cost PTN planning, operation, maintenance, and management.

In short, 2G networks will continue to be highly profitable for operators for quite some time. This means that MSTP PTN must co-exist. However, in the long term, after all mobile communication networks have become IP-based, MSTP will inevitably be replaced by PTN^[10]. It may even supplement PTN in providing dedicated

access for customers who demand small bandwidth but high security and privacy, or by covering regions temporarily out of reach of PTN.

3.2 PTN vs. WDM/OTN

IP over Wave Division Multiplexing (WDM)/Optical Transport Network (OTN) implements flexible dispatching and protection of large-granule services based on wavelength or Optical Channel Data Unit (ODU). When constructing a Metropolitan Area Network (MAN), IP over WDM/OTN can be applied in the core layer, providing networking, dispatching and protection of large-granule services of core network elements. If a large MAN is used to carry mobile backhaul services, PTN + WDM/OTN networking mode can be used. In this mode, the access/aggregation layer is networked in the form of PTN—which dispatches services upwards to the PTN core dispatching layer in the Radio Network Controller (RNC) equipment room via IP over WDM/OTN. This greatly simplifies the network that contains core layer nodes and aggregation layer backbone nodes, and flexibly dispatches services during adjustment of service homing. Moreover, as the core layer nodes are only connected to the RNC equipment room, the problem is avoided whereby all node devices must be upgraded

because one node has been upgraded to expand its service capacity (a problem sometimes encountered in pure PTN networks). Network costs are therefore saved. WDM/OTN provides flexible dispatching and networking for PTN services, as shown in Figure 5.

3.3 PTN vs. Metro Ethernet

In constructing a MAN, and depending on the nature of services carried on the network, an independent approach is often adopted. Two planes are constructed: a high-value mobile backhaul network and a low-value broadband access network. PTN is primarily used for carrying high value services such as mobile backhaul and services for important group customers. Low-value services such as public broadband access and services for large customers are first carried over a Passive Optical Network (PON), and are then sent to the service control layer, Broadband Remote Access Server (BRAS) or Service Router (SR) through a metro Ethernet. The two planes work independently and do not interfere with each other. Such a strategy takes the following into consideration:

(1) Important services such as Internet services and mobile backhaul services differ in configurability and have different reliability requirements. A shared plane that carries all these services creates greater network complexity. Moreover, Internet services require much more bandwidth than mobile backhaul services. If they share the same plane, MAN access devices must be re-configured for large capacity, high reliability technologies, and this increases costs.

(2) Broadband services are subject to frequent change. So sharing the same plane with mobile backhaul services may affect the stability of mobile services. If the stability of mobile services is to be guaranteed, demand for broadband services cannot be quickly accommodated. Stability and quick response to service demands cannot be realized at the same time.

(3) Internet services are open. If they share the same plane as mobile backhaul services, the network may

become vulnerable.

4 Application of ZTE's PTN Products

With strength in research and development, and with significant investment, ZTE now provides a full range of high-performance PTN products. In bidding tests conducted by China Mobile in 2009, ZTE's products ranked first. Moreover, ZTE has deployed its PTN products in systems of several multinational operators including Telefonica, Telenor, and TIM.

In the second quarter of 2009, China Mobile conducted performance tests on PTN products and IP-based lub interfaces of several vendors. These tests were conducted in its telecom network in Shenzhen and were designed to evaluate PTN products in terms of multi-service bearing capability, QoS, OAM, time synchronization, and longevity. When ZTE's PTN products were used to carry services via IP-based lub interfaces, service indexes and transmission quality met requirements. Compared with MSTP products, ZTE's PTN even excelled MSTP in some indexes and satisfied the TD service requirements of China Mobile.

To date, ZTE's PTN products have been used in China Mobile's PTN network construction in over 20 provinces.

5 Conclusions

Service requirements are the driving force behind the development of PTN technologies. In the evolution to 3G, High Speed Packet Access Plus (HSPA+), and LTE networks, traditional mobile services must become IP-based because emerging mobile services impose greater requirements on synchronization, network delay, reliability, and security. PTN is introduced in this context. As the industrial chain matures, PTN will advance technologically, have greater cost benefits, and will eventually become the mainstream carrier platform for high value services such as

mobile backhaul. The introduction and development of PTN will, in turn, drive the development of service networks and promote the rapid growth of IP-based services.

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Biographies

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communication technologies for many years. In 2004, ZXMP—a product which Zhong oversaw research and development—was awarded first prize at the Science and Technology Progress Awards of Guangdong Province. At present, he is researching and planning PTN products.

Analysis of Time Synchronization in PTN

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Abstract:

For Time Division Synchronous Code Division Multiple Access (TD-SCDMA) and Time Division Long Term Evolution (TD-LTE) wireless systems, using Global Positioning System (GPS) for time synchronization is problematic. In these wireless systems, GPS is costly, insecure, and difficult to deploy. Nowadays, transportation of high precision time/phase synchronization signals via fiber, and based on Precision Time Protocol (PTP) has become mainstream technology. This paper analyzes the main factors affecting time synchronization in a Packet Transport Network (PTN) adopting IEEE 1588v2. Laboratory experiments and field tests prove the feasibility of transporting high precision time synchronization signals through a PTN adopting IEEE 1588v2. A comparison is also made between different networking modes for a PTN adopting IEEE 1588v2.

With the development of mobile telecommunications technologies, precise network time synchronization has become increasingly important. CDMA2000, Time Division Synchronous Code Division Multiple Access (TD-SCDMA), and Time Division Long Term Evolution (TD-LTE) base stations all require high precision time synchronization. In TD-SCDMA systems, for example, the time synchronization index is $\pm 1.5 \mu\text{s}$. Such precision cannot be achieved through a free running oscillator, or even through the frequency synchronization network. On the other hand, installing a Global Positioning System (GPS) for each Time Division (TD) base station is difficult to engineer, costly, and insecure. Therefore, time synchronization protocol has become an important technology for transmitting high precision time synchronization signals in fiber systems.

Using synchronization protocol to transport high precision time synchronization signals via optical fiber systems is expected to be a leading

technology in the future.

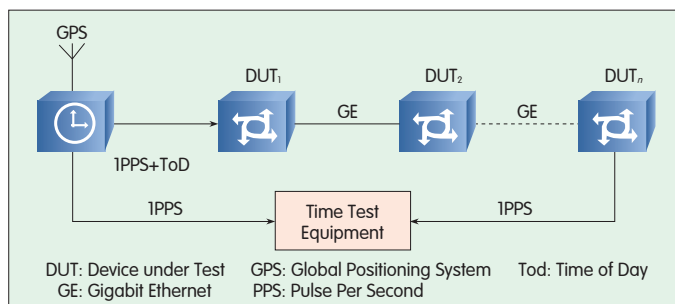
Terrestrial transmission of high precision time synchronization signals involves two key technologies: Precision Time Protocol (PTP), and compensation technology for transmission delay and jitter. Networks using Network Time Protocol (NTP) synchronization are only accurate to dozens of milliseconds, which does not meet the requirements of TD-SCDMA systems. Based on a delay request-response mechanism, IEEE 1588v2, also known as Precision Time Protocol (PTP), uses synchronization messages to calculate the time error between the slave and master clocks. Moreover, by using a hardware-embedded time stamp as well as a Boundary Clock (BC) or Transparent Clock (TC) to compensate delay and jitter incurred by network components or protocol stacks, IEEE 1588v2 achieves precision of sub-microseconds.

IEEE 1588 is a PTP originally designed for synchronization between industrial computers. Before it can be applied to large-scale communication

networks, further research must be undertaken into precision, networking mode, maintenance, and Best Master Clock Algorithm (BMCA). This paper analyzes the factors that may affect time synchronization precision in a Packet Transport Network (PTN) adopting IEEE 1588v2^[1-5].

1 Key Factors Affecting Time Synchronization Precision

IEEE 1588v2 adopts a master/slave clock configuration. The master clock distributes clocks periodically, and the receiver—taking advantage of network link symmetry—measures time offset and delay. Synchronization of frequency, phase, and absolute time of master and slave clocks is thereby realized. Transmission of IEEE 1588v2 packets may require each node to process a time stamp, which incurs delay and jitter. As a result, the number of hops in a transport network impacts the precision of time synchronization signals. Because a store-and-forwarding mechanism is



◀ Figure 1.
Test platform for
evaluating the impact of
hops on time
synchronization.

used in PTN, Packet Delay Variation (PDV) may have a significant impact on precision. Furthermore, network protection switching, signal degradation, temperature, and frequency synchronization may also impact timing precision.

1.1 The Number of Hops

In BC mode, each node terminates PTP packets, so PDV in the last node is not accumulated. The main factor affecting timing precision is the queue at the output port. In addition, because slave clocks at all levels are synchronized to the master clock, drift is incurred during clock recovery. This low-frequency drift is accumulated. Figure 1 illustrates a test platform that is designed for evaluating the impact of the number of hops on timing precision. On this platform, all PTN nodes are configured in BC mode and are interconnected by Gigabit Ethernet (GE) interfaces. The device under Test 1 (DUT1) is connected to the GPS receiver with a 1 Pulse Per Second (1PPS) + Time of Day (ToD) interface. Timing test equipment is used to measure the timing difference between the DUTn and GPS receiver.

Figure 2 shows timing differences in tests with different hops. In a nine hour test with 10 hops, the timing difference ranges from -120.3 ns to 131.5 ns with a peak-to-peak value of 252 ns. In a nine hour test with 20 hops, the timing difference ranges from -61 ns to 192 ns with a peak-to-peak value of 253 ns. In a four hour test with 30 hops, the timing difference ranges from -239.3 ns to 26.8 ns with a peak-to-peak value of 266 ns. These test results show that timing precision changes only slightly in proportion to the number of hops, and the noise model approximates

random distribution.

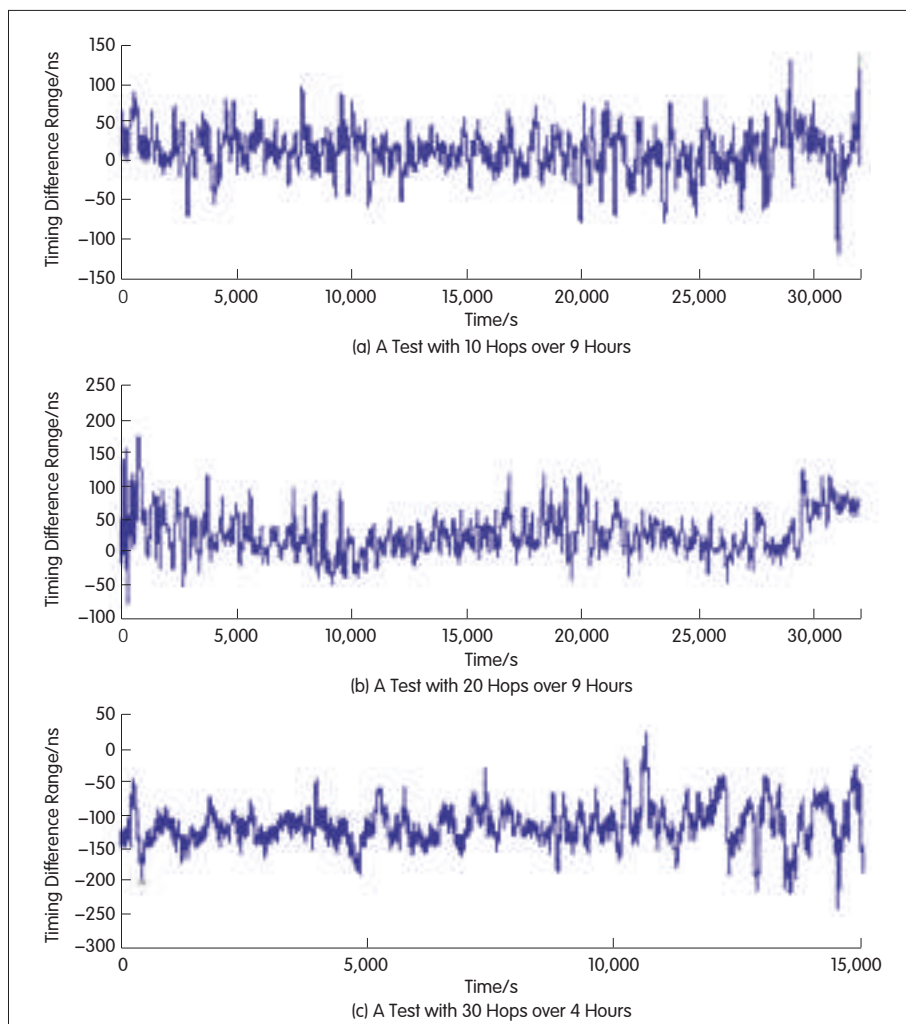
1.2 PDV

PDV mainly impacts timing precision in the purely transparent transport mode. As shown in Figure 3, all PTN nodes in the test platform in Figure 1 are set to transparent transport mode. Timing difference is measured where there is no load, and a 90% load with packet

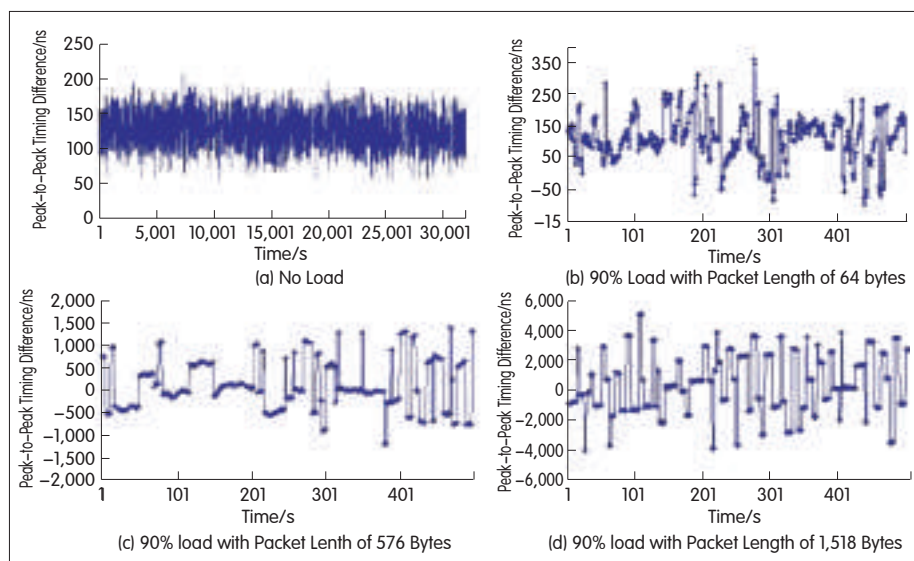
lengths of 64 bytes, 576 bytes, and 1,518 bytes. The peak-to-peak timing differences in each case are 250 ns, 450 ns, 3,200 ns, and 10 μ s respectively. It can be seen that the heavier the load or longer the packet, the greater the impact of PDV on timing precision. Before transparent transport mode can be used to provide high precision time synchronization signals, much optimization must be done.

1.3 Network Changeover

Changeover of time source, link, or clock board may lead to frequency and/or phase shift, affecting timing precision. Test results show that time source changeover causes a timing difference of 6 ns, optical link changeover causes a difference of 26 ns, and clock board changeover



▲ Figure 2. Timing difference with different hops.



▲ Figure 3. Impact of PDV on transparent transport mode.

causes a difference of 13 ns, as shown in Figure 4. Generally, the timing difference arising from network changeover is within 30 ns, enough for a 50 ns allowance in the parameter setting.

1.4 Signal Degradation and Temperature Change

Signal degradation may cause the rate of packet loss to increase, while temperature change may affect clock performance. In either case, timing precision may be affected. When bits are inserted with a Bit Error Rate (BER) of 1×10^{-3} , the peak-to-peak timing difference is within 110 ns, similar to the case where there is no bit error. When the temperature of PTN nodes is raised from -10°C to 50°C , the peak-to-peak timing difference is within 40 ns, and timing does not change proportionally with temperature.

1.5 Frequency Synchronization

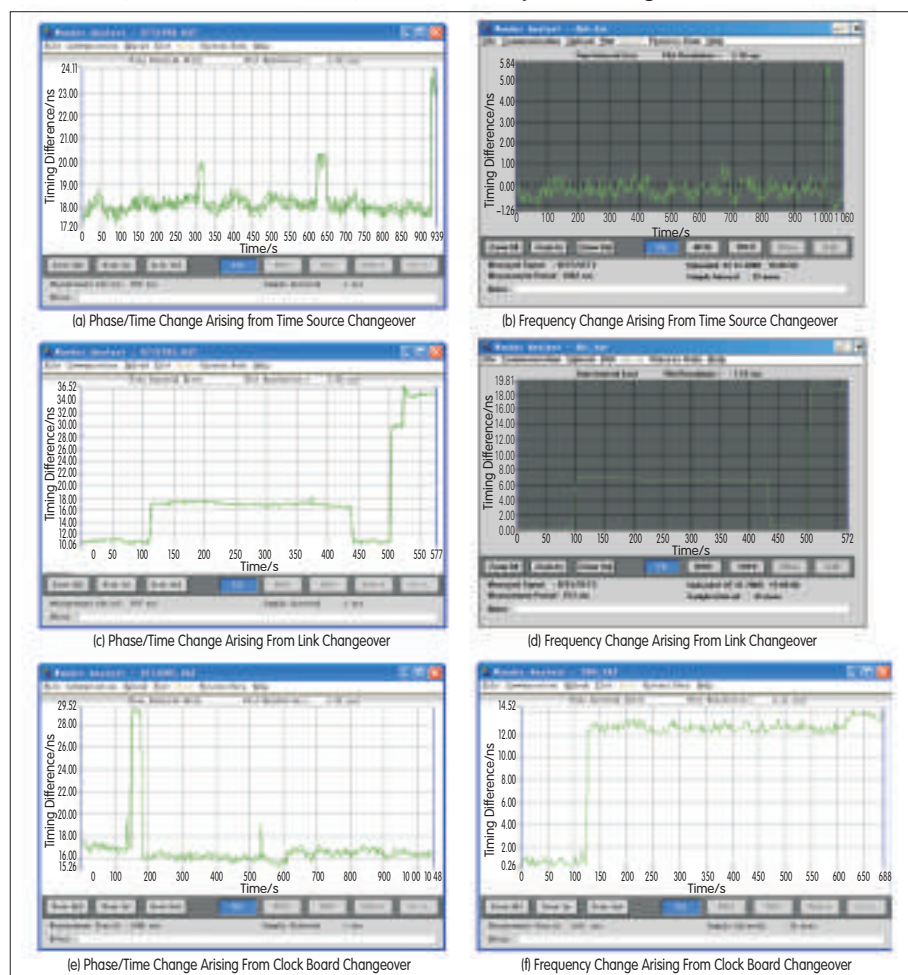
PTN can obtain the clock from the physical layer via synchronous Ethernet, or recover the clock from 1588v2 packets. Tight coupling occurs when PTN realizes frequency synchronization using synchronous Ethernet, and timing and phase synchronization using 1588v2 packets. Loose coupling occurs when PTN realizes frequency, time, and phase synchronization using only 1588v2

packets. Time synchronization in the cases of tight and loose coupling is

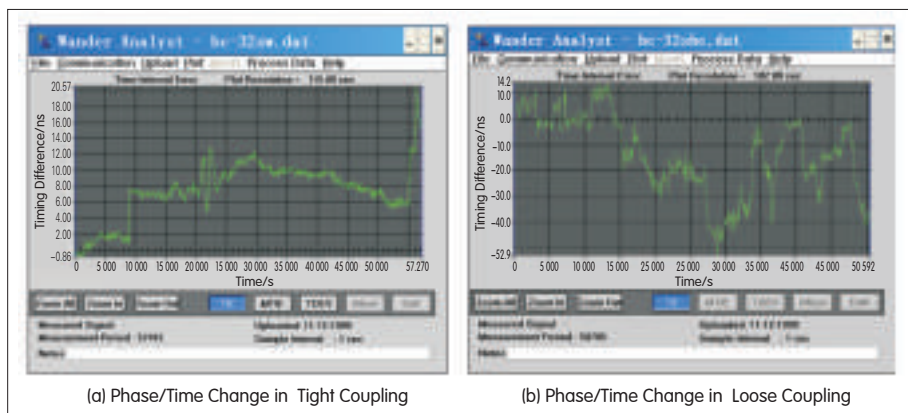
illustrated in Figure 5. Under normal conditions, the peak-to-peak timing difference in tight coupling is 22 ns, while that in loose coupling is 67 ns. Synchronization in tight and loose coupling is therefore similar.

Figure 6 shows the impact of frequency synchronization on timing precision in tight coupling mode. When the frequency is in Keep state, with an offset of 5×10^{-9} , the peak-to-peak timing difference is 100 ns. When the frequency is in Free state, with an offset of 3.8×10^{-8} , time synchronization signals are lost.

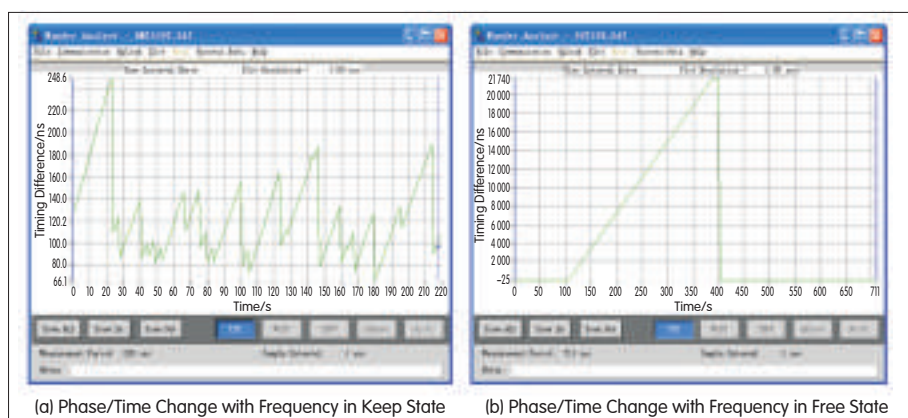
The two coupling modes have their own advantages and disadvantages. In the case of tight coupling, frequency synchronization signals are degraded or lost, which may affect time synchronization signals. In the case of loose coupling, each local PTN clock may take a long time to trace the



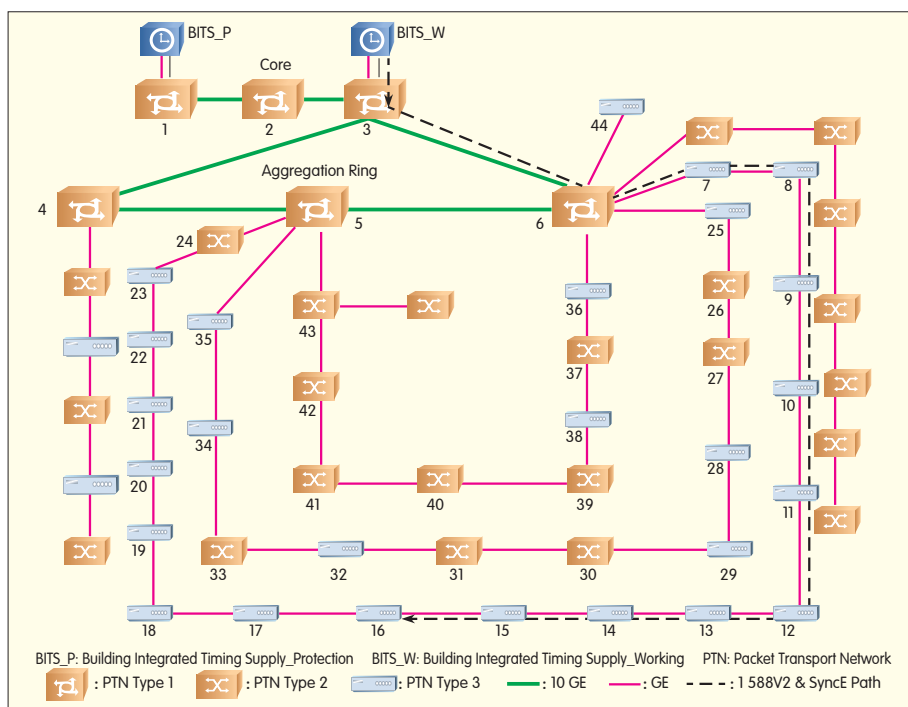
▲ Figure 4. Impacts of network changeovers on timing precision.



▲ Figure 5. Timing synchronization in tight and loose coupling.



▲ Figure 6. Impact of frequency synchronization on timing precision in tight coupling mode.



▲ Figure 7. Network topology for timing synchronization test.

1588v2 clock. Also, in loose coupling mode, the frequencies of 1588v2 packets must be increased, which increases the network load. Tight coupling is therefore preferred.

2 Long-Term Stability of Time Synchronization in BC Mode

To verify the stability of a large PTN adopting 1588v2, part of the existing network is selected for testing. The test network includes three core nodes, three aggregation nodes, and 58 access nodes. Each access node has one TD-SCDMA base station, as shown in Figure 7. All PTN nodes adopt BC mode and all fiber segments compensate for delay asymmetry. Frequency synchronization is implemented using synchronous Ethernet. Time interfaces include 1PPS+ToD interface and FE/GE interfaces.

When FE is used, the peak-to-peak timing difference over 72 hours is 65 ns, as shown in Figure 8. The Maximum Time Interval Error (MTIE) and Time Deviation (TDEV) results are shown in Figure 9. These results show that the timing is stable, there is little noise, and the noise model approximates random distribution.

3 Conclusions

For TD-SCDMA and TD-LTE systems, using GPS to implement time synchronization is costly, insecure, and difficult to engineer. Time synchronization protocol has therefore become the preferred technology for transmitting high precision time synchronization signals in fiber systems. High precision time synchronization in a PTN adopting IEEE 1588v2 has been proven feasible. In existing networks, many fiber segments are asymmetric, so compensation for delay and jitter must be done segment by segment. Both BC and TC eliminate the impact of PDV, but BC is simpler. Standards for time synchronization in telecom networks are still immature, and experience is limited. High precision time synchronization will first

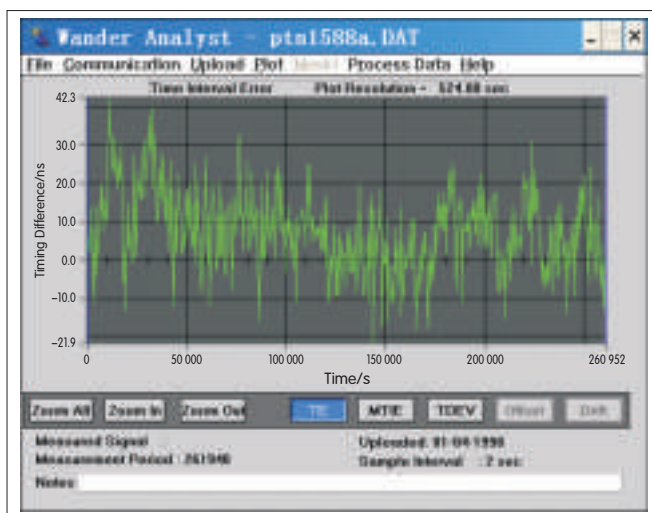


Figure 8.
Long-term stability of time synchronization.

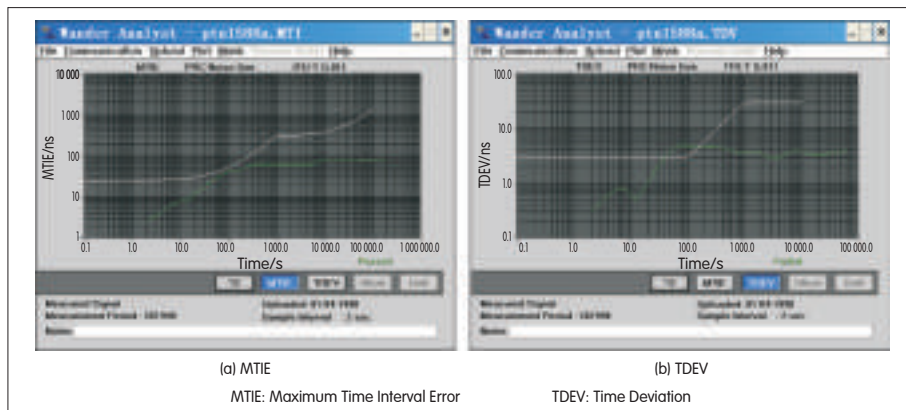


Figure 9. MTIE and TDEV.

be deployed in PTN, and then in Optical Transport Networks (OTN) and Passive Optical Networks (PON) in the future.

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Biography

Li Han



Li Han received his Ph.D. degree from Beijing University of Posts and Telecommunications. He is a deputy principle at China Mobile Research Institute, mainly engaged in the research of transmission, wired access and synchronization technologies. He has published over 50 papers in national and international journals, and applied for more than 20 patents. Being also an ITU-T editor, he has delivered over 60 ITU-T contributions.

Roundup

ZTE Positioned in the Leaders Quadrant in Leading Industry Analyst Firm Report

ZTE Corporation announced on July 14, 2010 that it has been positioned in the leaders quadrant in Gartner's Magic Quadrant for Softswitch Architecture Report.

As described in the report: "Leaders are high-viability vendors with a broad portfolio, significant market share, broad geographic coverage, a clear vision for how service providers' needs will evolve, and a proven track record for delivering products. They are well positioned with their current product portfolios and are likely to continue to deliver leading products. Leaders do not necessarily offer the

best solution for every customer requirement, and their products may not be 'best of breed' throughout their portfolios. However, overall, they provide solutions that offer relatively low risk and high quality."

"We are honored to be one of the Softswitch architecture leaders recognized by Gartner's Magic Quadrant Report. We believe that it is a significant milestone for ZTE's sustainable developments and that it is a clear endorsement of our Softswitch and IMS solutions by the industry," said Mr. Xu Ziyang, Vice President of ZTE.

ZTE's All-IP core network solution covers fixed Softswitch, Mobile Softswitch, IMS and other fields, and the continuous innovative Softswitch and IMS solution promotes the healthy development of industry chain.

ZTE is becoming the leader in driving core network technology development and market application all over the world. Based on continuous investment in research and increasing in emerging markets, ZTE gained 35% revenue growth in the Softswitch market, while the overall market suffered a decline of more than 10%. (ZTE Corporation)

Vehicular Ad-Hoc Networks: An Information-Centric Perspective

Abstract:

Emerging Vehicular Ad-Hoc Networks (VANET) have the potential to improve the safety and efficiency of future highways. This paper reviews recent advances in wireless communication technologies with regard to their applications in vehicular environments. Four basic demands of future VANET applications are identified, and the research challenges in different protocol layers are summarized. Information dissemination is one of the most important aspects of VANET research. This paper also discusses the primary issues in information dissemination from an information-centric perspective, and provides two case studies. Finally, future research directions and possible starting points for new solutions are considered.

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1 Introduction

Vehicular Ad-Hoc Networks (VANET) are becoming an integral technology for connecting daily life to computer networks. They could greatly improve the driving experience both in terms of safety and efficiency. As shown in Figure 1, when multi-hop communication is implemented, VANET enables a vehicle to communicate with other vehicles which are out of sight or even out of radio transmission range. It also enables vehicles to communicate with roadside infrastructure. VANET will likely be an essential part of future Intelligent Transportation Systems (ITS).

Currently, ITS relies heavily on infrastructure deployment. Electromagnetic sensors, for example, are embedded into the road surface; traffic cameras are deployed at major intersections; and Radio Frequency Identification (RFID) readers are deployed at highway entrances. A typical procedure for collecting and distributing traffic information is as follows. First, traffic samples are gathered by road surface sensors and uploaded to a municipal transport center. After data processing, traffic

reports can then be delivered to a user's cell phone via cellular networks. This is an expensive and inefficient way of disseminating location-based information, especially when the information of interest is only a few hundred meters from the user's physical location. With its short-range communication capabilities, VANET may change this paradigm and make generating and disseminating information more straightforward.

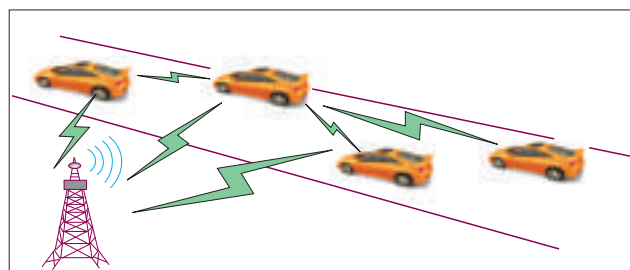
VANET can also serve as a large-scale wireless sensor network for future ITS because every modern vehicle can be regarded as a super sensor node. For example, all new vehicles are usually equipped with exterior and interior thermometers, light sensors, one or more cameras, microphones, ultrasound radar, and other sensory features. Moreover, future

vehicles will also be equipped with an on-board computer, wireless radio, and a GPS receiver, which will enable them to communicate with each other and with roadside units. A wireless sensor network of such magnitude is unprecedented, and perceptive computer systems will extend to every corner of the globe. Information can be generated and shared locally in a peer-to-peer manner without the need for restrictive infrastructure.

The capabilities of future vehicles open up a number of potential applications for use in daily life. The main applications of VANET can be categorized as:

Safety applications: pre-collision warning, electronic road signs, traffic light violation warning, online vehicle diagnosis, and road condition detection. This type of application

Figure 1. ▶
Vehicular ad-hoc networks.



usually takes advantage of short-range communication to perform real-time detection and provide warnings to drivers

Efficiency applications: municipal traffic management, traffic congestion detection, route planning, highway tolling, and public transportation management. This type of application is dedicated to improving both individual and public travel efficiency

Commercial applications: Location-Based Services (LBS) will give rise to a variety of commercial applications such as nearby restaurant specials, cheap gas stations, or even shopping center promotions. Such commercial applications may spur new competition among local businesses.

Infotainment applications: video and music sharing, location-based restaurant or store reviews, carpooling, and social networking. Already, infotainment applications such as Ford Sync^[1] and Kia UVO have become attractive add-ons in the vehicle market. The networking of infotainment systems will surely be a trend in the near future.

An abundance of VANET applications will benefit a wide range of parties: from governments and vehicle manufacturers to local retailers and consumers. Although a few Geographic Information Systems (GIS) companies—such as Google, Garmin, and TomTom—have engaged in collecting and distributing traffic information, traditionally, ITS development and deployment has been the domain of governments. In the future, many more participants will be attracted to VANET and will profit from it. Vehicle manufacturers could predict a boost in their sales by selling VANET-enabled vehicles. Fitting vehicles with a variety of electronic controls and devices is a growing trend, especially fitting electronic safety and information systems. Ford Sync is a very successful example of vehicle infotainment. Moreover, local retailers and service providers will also be interested in promoting their sales via VANET. They could broadcast commercials to passing vehicles and even devise hourly pricing strategies.

Local businesses may gain a competitive advantage or face greater competition. Undoubtedly, consumers will be the beneficiary of enhanced safety and efficiency, cheaper goods, enriched entertainment, and other advantages.

In this paper, the following section reviews recent advances in wireless communication technologies with regard to their applications in vehicular environments. Section 3 identifies four fundamental demands of future VANET applications. Section 4 discusses existing challenges in different network protocol layers. Section 5 further discusses several research topics in information dissemination from an information-centric perspective. Section 6 concludes.

2 Wireless Technologies and Vehicular Communications

Wireless Personal Area Networks (WPAN) using IEEE 802.15 standards have been largely successful in consumer electronics (including vehicular electronics). Ford Sync is a good example. With Bluetooth technology, a cell phone can be connected to the vehicle's audio system enabling a driver to make calls or play hands-free music using voice commands. 802.11 (a/b/g) WLAN technologies have been widely deployed because of their mass production and relatively low cost. Although 802.11 (a/b/g) was not originally designed for vehicular communications, many studies (in particular References^[2-4]) have focused on applying 802.11 to vehicular environments because of the pervasiveness of its technologies. IEEE 802.11p^[5] introduces enhancements to 802.11 which are needed to support Wireless Access in Vehicular Environments (WAVE). This includes data exchange between high-speed vehicles and between vehicles and roadside infrastructure in the licensed ITS band of 5.9 GHz.

Another emerging technology is Wireless Metropolitan Area Network (WirelessMAN), also called Worldwide

Interoperability for Microwave Access (WiMAX) (IEEE 802.16). It is aimed at providing wireless data over long distances in a variety of ways, from fixed point-to-point links to full mobile cellular type access. Currently, the most common form of automobile connectivity is based on cellular telephony and is known as automotive telematics. Typical examples include GM's OnStar system and Ford's RESCU system. Several GIS companies, including TomTom and Garmin, also use cellular networks to transmit real-time traffic information. Usually, cellular-based telematics is a paid service based on user subscription.

In the near future, it is envisioned that architecture of vehicular networks will be hybrid, as shown in Figure 2. In this architecture, long-distance communication techniques, such as cellular networks and WiMAX, will provide vehicles with instant Internet access, while short-distance communication techniques, such as Dedicated Short-Range Communications (DSRC)^[6] and Wireless Fidelity (Wi-Fi), will provide short-range real-time support in an ad hoc manner.

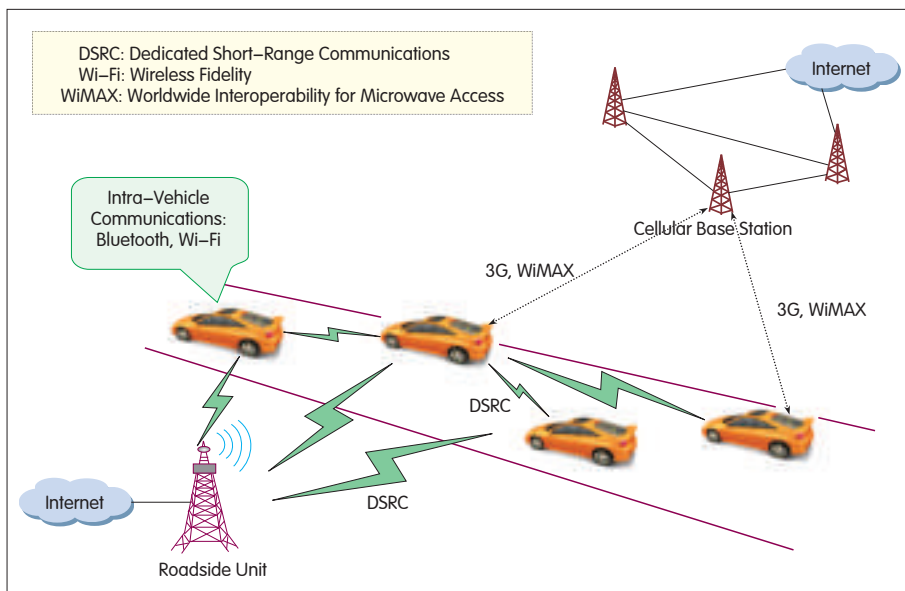
VANET, based on DSRC, Wi-Fi, and other short range communication techniques, will play an important role in future ITS. Compared to infrastructure networks, VANET has two main advantages. It is cheap to deploy and operate, and consumers can enjoy service without subscription. VANET is also essentially a cyber-physical system, which enables communication between two geographically neighboring nodes. This has real-time safety and other applications.

3 Requirements of VANET Applications

Future VANET applications will have four fundamental demands: scalability, availability, context-awareness, and security and privacy.

(1) Scalability

Because of the number of vehicles that could be incorporated into vehicular networks, VANET may



▲ Figure 2. Wireless technologies for future vehicular communications.

become the largest ad hoc network in history. Undoubtedly, scalability will be a critical factor. The advantages of hybrid architecture, together with in-network aggregation techniques and P2P technologies, make information exchange more scalable.

(2) Availability

Due to the real-time interaction between vehicular networks and the physical world, availability is an important factor in system design. This may have a major impact on the safety and efficiency of future highway systems. The architecture should be robust enough to withstand unexpected system failures or deliberate attacks.

(3) Context-Awareness

As a cyber-physical system, VANET collects information from the physical world and may conversely impact the physical world. On the one hand, protocols should be adaptable to real-time environmental changes, including vehicle density and movement, traffic flow, and road topology changes. On the other hand, protocol designers should also consider the possible consequences the protocol may have on the physical world.

(4) Security and Privacy

There is a recent trend of making vehicular on-board computer systems inter-connectable to other systems.

The Ford Sync, for example, connects the vehicle's entertainment system to the driver's cell phone via blue-tooth technology. In the future, vehicular on-board computers could even be open to software developers. These trends may have serious implications for security and privacy due to the cyber physical nature of VANET. Governments and consumers will have very high expectations of VANET safety and security.

4 Research Challenges

This section discusses research challenges from different network protocol layers. The unique properties of vehicular networks give rise to a number of design challenges. These properties also create new opportunities to solve ITS problems from a different perspective.

4.1 Link Layer

In the link layer, the main challenge lies in adapting link layer protocols to unique vehicular environments and to maximize link layer performance. There are three major design objectives for link layer protocols: responsiveness, reliability, and scalability. Link layer protocols are required to be highly responsive to changes in channel conditions and vehicle mobility, while

reliability and scalability are two requirements critical to safety applications. A number of traditional link layer strategies, such as lengthy access point selection, MAC management timeout, and Address Resolution Protocol (ARP) timeout, have been proven inefficient in highly mobile environments. They may lead to increased start-up delays, underutilized bandwidth, or unfair bandwidth allocation. Scalability and reliability are interrelated issues. Reliable broadcasting has been intensively studied for vehicular safety applications. The existing approaches include rebroadcasting, cooperative forwarding, and transmission power adaptation. However, reliability and scalability remain open-ended issues for safety applications because both governments and consumers have extremely high expectations for safety applications.

4.2 Network Layer

In the network layer, the main challenge is to establish a new paradigm for information dissemination in VANET. Ad hoc network routing has been intensively studied in the last decade. In particular, many context-aware routing protocols—such as Mobility-Centric Data Dissemination Algorithm for Vehicular Networks (MDDV)^[7] and Vehicle-Assisted Data Delivery (VADD) in Vehicular Ad Hoc Networks^[8]—have been proposed for VANET. These protocols significantly improve the packet forwarding performance in vehicular environments by taking advantage of vehicle mobility, GPS position, and road layout. They are essentially all packet-based; a packet travels from a source to a destination untouched throughout the entire process. However, this packet-based paradigm no longer satisfies application requirements in VANET from an information-centric perspective. First, for some applications, there is no definite source and destination, which is necessary for packet-based routing. Second, information is altered (or combined) throughout the forwarding process, and this is not a consideration of packet

routing. In a traffic detection application, every vehicle may generate a traffic report that can be combined with other reports as it is disseminated. For all interested vehicles intended to be the recipients of these reports, there is no prior knowledge about how many, when, or where these vehicles might be. Some packet routing approaches such as multicast and geocast can help solve these issues. However, what is needed is a new paradigm for information routing—a replacement for packet routing. The new paradigm will enable information operations such as information generation, aggregation, dissemination, and invalidation.

4.3 Application Layer

In the application layer, the challenge lies in effectively representing, discovering, storing, and updating information throughout the network. Naming and addressing are central problems in vehicular networks. How to index information from the physical world for efficient information storage and dissemination remains an unresolved problem. It is envisioned that the addressing scheme will be a hybrid, multi-level scheme, with context information playing an important role. The naming and addressing policy has a significant impact on other system protocols, such as information discovery and routing. Because vehicles are highly mobile, another challenge is to dynamically map vehicle IDs to position-based addresses. This problem is particularly important for applications across the hybrid network architecture. ARP/Reverse Address Resolution Protocol (RARP)-like mechanisms can be implemented in nodes equipped with both DSRC and infrastructure network interfaces.

Distributed data management is another challenging issue for VANET, impacting data replication, data elimination, and cache replacement. Traditional distributed data management assumes a network is connected with geographically-distributed servers, which is no longer true for VANET. Essentially, VANET can be regarded as

▼ Table 1. Representatives of information dissemination approaches

Information Dissemination Approaches		Representative Work
Macroscopic	Infrastructure-Based	Reference [9]
	Delay Tolerant Routing	References [10], [11], [12] and [13]
	Data Aggregation, Data Caching	References [15], [18] and [24]
Microscopic	One-Hop	References [2], [3] and [4]
	Data Forwarding	References [7], [8] and [20]
	Resource Management	References [21], [22] and [23]

a large-scale distributed database in which each vehicle maintains a local part. Vehicles periodically exchange data to update this global database, and inconsistency cannot be avoided. Therefore, maintaining a relaxed consistency model with minimal overhead is a challenge.

5 Information Dissemination

In this section, research demands and challenges from an information-centric perspective are discussed. VANET can be regarded as an information-centric system where information is collected and disseminated throughout the network, and it is important to identify the system's demands from this perspective. Information dissemination can be classified into two levels: macroscopic and microscopic. Two case studies are presented for these levels respectively. Table 1 lists the major research topics at these two levels and the representative work on each.

5.1 Macroscopic Information Dissemination

Macroscopic information dissemination deals with disseminating information to one node or a group of nodes in a specified geographical area. The destination of information can be a single specified node in the network, a group of specified nodes, or even a group of unknown nodes. The objective of macroscopic information dissemination is to reduce information delivery delay, reduce delivery overhead (including storage overhead and communication overhead), and increase the future query success rate (if the destination is unknown in advance). General research topics for

macroscopic information dissemination include information routing, data caching, and data aggregation.

Information dissemination can be implemented with or without infrastructure support. Jedrzej et al^[9] proposed establishing a P2P overlay network based on cellular network infrastructure. With a reliable connection to internet infrastructure, vehicles can share, discover, and exchange information of non-safety applications in a P2P manner. However, service provided by infrastructure is usually based on paid subscription, which limits the number of consumers. Compared with infrastructure service, ad hoc networks seem to be a more attractive approach. On the other hand, most non-safety applications do not have strict real-time requirements. A recent trend has been to study VANET information dissemination in a delay tolerant manner. Some general purpose DTN routing protocols such as epidemic routing^[10] have been proposed and evaluated. Proactive approaches^[11–12] take advantage of apriori knowledge of geographical location, connectivity pattern, as well as control over movement. Some existing DTN routing protocols assume a predefined source and destination. For example, in Small and Hass's project^[13], a DTN network was established for wild whale monitoring. A sensor node was mounted on the back of a whale, and mobility information was delivered to an infostation hop by hop in a delay-tolerant manner.

Data caching and aggregation have also been studied in VANET. Zhao et al^[14] studied the process of distributing information from a data center into VANETs. Their approach was based on periodical

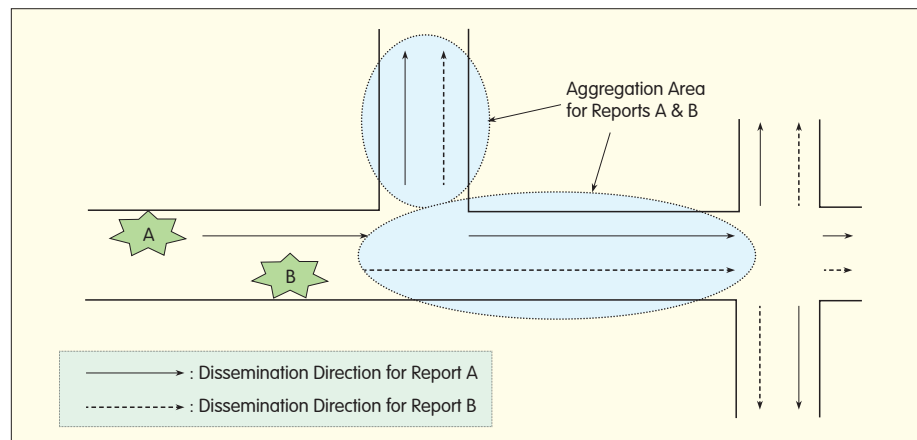
rebroadcasting and buffering. It is a one-way dissemination from a data center to a group of vehicles. Lochert et al.^[15] proposed a hierarchical aggregation scheme that defines a set of landmarks to calculate travel time. They also proposed a roadside unit placement algorithm to optimize aggregation.

Information dissemination, caching, and aggregation have been individually studied in relation to VANET. However, the delay-tolerant dissemination problem has been fused into data query, data caching, and data aggregation issues because, in VANET, for most types of information, there is no a priori knowledge of the destination vehicles. Any vehicle may generate and send out a query in the hope that a response is returned by a neighboring vehicle as soon as possible.

A new paradigm for information routing needs to be established as an alternative to packet routing. First, the destination of information routing must be defined. The dissemination destination is a virtual concept constrained by time, space, and vehicles. In other words, the destination consists of all vehicles which meet the temporal and spatial conditions. There are two basic dissemination operations: pull and push. For pull, a vehicle periodically broadcasts its interest and pulls data from other neighboring vehicles; for push, vehicles intentionally push data to neighboring vehicles so that other vehicles that may be interested in the data can easily obtain it in the future. Since the pull operation is limited to one hop at the initial stage of market penetration, it is more important to devise push strategies. When devising push strategies, the potential impact to data caching and aggregation must be taken into account. Heuristic neighbor information (such as driving direction, speed, frequently visited places, etc.) or even social networking information can be used to predict and control the dissemination.

5.1.1 In-Network Data Aggregation

This sub-section examines the details of macroscopic information



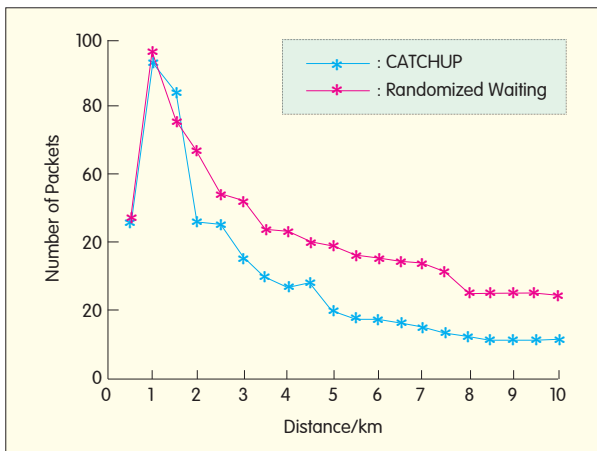
▲ Figure 3. Dissemination tree.

dissemination through an example of in-network data aggregation. As previously mentioned, in the near future, every vehicle will be a super sensor, capable of monitoring its surrounding environment. Each vehicle may generate a traffic report when the vehicle speed is well below the speed limit. However, it is inefficient for every vehicle to generate a report and then broadcast it to the entire network. It is unnecessary to broadcast the speed of individual vehicles on a road which is a few miles away; drivers expect to know general congestion information ahead. Actually, traffic reports can be combined as they are propagating, and the overhead for dissemination can be effectively reduced.

Figure 3 illustrates an example of report dissemination. Two vehicles A and B, which are geographically close to each other, generate two traffic reports. These two reports are intended to be delivered to all vehicles following behind so that those vehicles can take a detour before encountering a traffic jam. A report will be duplicated at an intersection, each copy going in one direction. In this way, a dissemination tree is formed. Since vehicles A and B may not be within each other's signal range, one may not know that a similar report was generated by the other. Instead of being disseminated across the entire network, these two reports can be merged as they propagate. In Figure 3, the dotted circle shows the possible aggregation area for these two reports.

In order to aggregate the two reports, they must be delivered to the same vehicle at the same time. For traditional wireless sensor networks, researchers have proposed a number of structural or semi-structural-based aggregation schemes involving the creation of a transmission schedule to ensure reports meet each other at the fork of a routing tree. However, a fixed routing structure is infeasible in VANETs. Several VANET projects, such as Self-Organizing Traffic Information System (SOTIS)^[16] and TrafficView^[17], use periodic rebroadcasting to collect and redistribute traffic information. Rebroadcasting is a feasible solution for local information exchange and dissemination, but it is difficult to scale it to city-wide dissemination.

Essentially, in-network data aggregation schemes trade off increased delay for reduced redundancy. When a node receives a report, it introduces a delay before forwarding it to the next hop so that it may receive another report for this duration. Structure-based schemes use a transmission schedule to determine this delay. Rebroadcasting schemes use a fixed delay for rebroadcasting. If the delay is more adaptive, however, a packet is more likely to meet other reports. In our previous work^[18], intelligent delay control policies based on local observations were investigated. For example, suppose a node observes that a report has recently passed by; if the node receives another report in a



◀ Figure 4.
Performance of in-network
data aggregation.

short period, it can simply forward it to the next hop immediately in the hope that it can catch up with a previous one. If no report has recently passed, a long delay can be applied in the hope that more reports can be later received by this node. A future reward model is designed to define the benefits of different delay-control policies, and then to establish a decision tree to help a vehicle choose an optimal policy from the perspective of long-term reward. Figure 4 shows the performance of such an aggregation scheme. Our scheme (CATCHUP) is compared with Randomized Waiting^[19]; the results show that the number of packets can be significantly reduced as the distance to the report source increases.

Data aggregation in VANETs has attracted much research attention. However, issues involving scalability, data representation and processing, and delay tolerant routing remain unresolved. Among these, scalability is the most pressing issue. Although a number of data aggregation schemes have been proposed for VANETs, it is still not clear how scalable these schemes are in terms of city-wide communication.

5.2 Microscopic Information Dissemination

Microscopic information dissemination deals with information delivery in one hop or in a few hops. In the initial market penetration stage, a vehicle may rarely encounter another vehicle or roadside unit. Therefore, increasing the efficiency of each encounter between

vehicles is important.

A few recent research projects have paid close attention to one-hop communication in a vehicular environment. Bychkovsky et al studied the techniques to increase one-hop throughput via open WiFi links. They conducted a number of field tests to investigate the performance loss in MAC association, IP address acquiring, and IP route establishing. Hadaller et al conducted a 802.11-based one-hop communication experiment and furnished a detailed experimental analysis. From their experiments, the researchers identified the underlying causes of throughput loss in existing wireless access mechanisms. In sum, these works attempt to analyze and improve the link throughput from the perspective of lower layer protocols (phy, MAC, routing).

Microscopic information dissemination also deals with local multi-hop communication. Usually, the main task of local multi-hop communication is to coordinate local vehicles to disseminate information in a predefined direction. VADD is a forwarding protocol which takes advantage of traffic pattern and road topology to source the best road for delivering a packet. MDDV exploits vehicle mobility for information dissemination, and makes neighboring vehicles collaborate in packet forwarding in order to increase reliability. Zhao et al^[20] studied throughput improvement gained through cooperative relaying to a roadside unit.

Because of the short session duration of a mobile-encounter scenario, efficient management of channel resources is also a practical issue. Chang et al^[21] proposed a scheduling algorithm for downlinks—from a roadside unit to passing vehicles. Zhang et al^[22] proposed another scheduling algorithm which considers both upload requests and download requests. Yu et al^[23] studied the admission control problem when a roadside unit is experiencing (or close to experiencing) overloaded conditions. These studies improve the efficiency of roadside unit access from different resource allocation perspectives.

In general, the main challenge in microscopic information dissemination is how to bind lower layer conditions (mobility, channel, position) and upper layer application requirements together. From the upper layer perspective, DTN applications can tolerate information delays and information inaccuracy; from the lower layer perspective, mobility, channel, and location may change dramatically in a short period of time. Existing work has studied the one-hop communication problem with different network protocols. However, there is still no effective bond between the upper and lower layers. The bond may allow us to take advantage of the lower harsh conditions, rather than being constrained by them.

Three aspects are considered when designing microscopic information dissemination protocols.

(1) Application Requirement

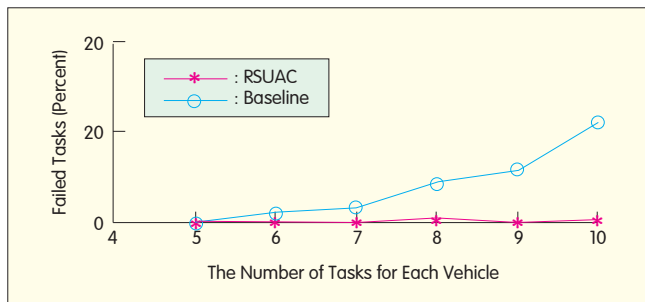
DTN applications do not assume a reliable link, but do prioritize data for transmission. They may specify the information loss tolerance level during the dissemination process.

(2) Resource Management

Problems include how to schedule lower layer resources (such as transmission channel and transmission rate); how to schedule upper layer tasks; and how to allocate resources to ensure fairness.

(3) Cooperation

Here, focus is on cooperation between vehicles within signal range. Potential techniques include multi-task scheduling, relaying, multi-party



◀ Figure 5.
Performance of admission control.

network coding, and others.

5.2.1 Roadside Unit Admission Control

This subsection provides a case study for microscopic information dissemination. Roadside Units (RSU) play an important role in future ITS. They are usually deployed at highway ramps or road intersections, and can communicate with passing vehicles with DSRC technologies. RSUs can provide these vehicles with a variety of potential services. For example, a passing vehicle may download digital maps, commercials, and traffic reports from an RSU. However, an RSU is a sparse resource. In the initial phase of market penetration, only a limited number of RSUs can be deployed at highway ramps or major intersections. Even when a vehicle encounters a unit, it may take less than half a minute to move out of signal range. Moreover, multiple vehicles may concurrently compete for services from the RSU. Therefore, it is important to efficiently manage RSU access.

Transmission integrity is also important for RSU access, since the services provided may be time or location sensitive. If downloading a task like a digital map or traffic report cannot be completed before the vehicle moves out of signal range, the downloaded part would be meaningless, not to mention a waste of bandwidth. Admission control is a potential approach to guaranteeing transmission integrity. The task of admission control is to determine whether to admit a new upload or download task. Once a task has been admitted, a transmission schedule is calculated to guarantee completion of the task.

Admission control has been the subject of intensive study. Traditional

admission control schemes mainly focus on long-term sessions; for example, VoIP and multimedia services. Some real-time systems simply characterize transmission tasks by average rate, peak rate, or burst size. However, roadside unit access is mainly focused on short sessions, and the transmission rate for a moving vehicle may vary dramatically. A dedicated admission control scheme is therefore desirable for roadside unit access.

In our previous work, an admission control scheme was proposed for roadside unit access. The scheme calculates a transmission schedule for all tasks (including current and new tasks) based on a channel prediction model and a vehicle mobility model. If a feasible schedule is found, new tasks will be admitted; otherwise, new tasks will be rejected to guarantee the success of current tasks. The problem was treated as a linear-programming optimization problem and a set of algorithms were designed to calculate the bandwidth allocation schedule. All concurrent transmission tasks share the bandwidth according to the schedule, thereby maximizing the success rate of these tasks. In a NS2-based simulation, our scheme was compared with a baseline admission control method. The baseline method allocated bandwidth based on a minimum required rate. Figure 5 demonstrates that the RSUAC scheme effectively reduced the percentage of failed tasks even when the workload (number of tasks per vehicle) increased.

6 Conclusions

VANET is a promising area for future ITS, and has the potential to become

the largest ad hoc network in history. In the past few years, it has attracted much attention from academia, industry, and government. However, there are fundamental issues that remain unresolved. Better paradigms are needed for information dissemination and distributed data management. Undoubtedly, the number of research contributions will continue to increase in the near future.

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Biographies

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Roundup

ZTE and Three Release Android Éclair Handset in the UK

ZTE Corporation and Three have launched the ZTE Racer, a low cost Android 2.1 (Éclair) powered handset.

The Racer is the first Android handset in the UK from ZTE and is Three's first Android handset to break the 100 barrier, selling at 99.99.

Android could account for the second highest market share of mobile operating systems by 2012, according to a recent report from Gartner. However, it can take over a year to develop a new smartphone which means that device manufacturers are struggling to keep up with developing Operating Systems (OS) and new devices are often launched with older versions of the OS.

The Racer is a ZTE and Three

co-branded handset that combines a 2.8" QVGA touchscreen, 3.2 megapixel camera, 256 MB of internal memory, Bluetooth and 7.2 Mbit/s HSDPA capabilities.

The handset is aimed at consumers who require access to a wide range of applications. The Android OS allows users to download thousands of apps from the Android market with quicklink access to services including Facebook, Spotify, YouTube, Google Maps and Google Talk.

ZTE expects to increase its share of the UK carrier handset market from three percent in 2009 to eight percent in 2010. In Q2 this year alone, ZTE mobile devices sales in the UK passed the million unit mark for the first time. (ZTE Corporation)



Applying Analytical Hierarchy Process to a WiMAX Performance Evaluation Model

Abstract:

Evaluating performance of individual features of WiMAX technology is a topic of widespread discussion. Currently, there is no quantitative way of measuring WiMAX technology so that wireless operators can meet their design objectives. This paper outlines a set of design criteria for WiMAX and provides a decision-making aid that ranks the importance of criteria using Analytic Hierarchy Process (AHP). This ranking should sufficiently reflect market expectations of the relative importance of various design criteria. A model integrating AHP priorities with enhanced Data Envelopment Analysis (DEA) is the basis for formulating a technological value in simple, comparable format. A case study is provided to show how this technological value is used to evaluate a three year network deployment plan. In the future, this model could be extended to WiMAX equipment suppliers for the purpose of validating performance targets of individual criteria, and enhancing supplier roadmaps for future network development.

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1 Introduction

To satisfy the need for multi-media data access any time and anywhere, demand has grown for high speed wireless data transmission. Expansion of the smart phone and embedded device market requires high speed data networks to cope with different application suites.

Worldwide Interoperability for Microwave Access (WiMAX) is based on the IEEE 802.16 standard. It was initially designed for fixed and nomadic connection; and since 2001, has been continually updated and enhanced. With the introduction of 802.16e in 2005, it started to evolve towards support for fixed and mobile connection. Important technologies such as Multiple-Input Multiple-Output (MIMO), Orthogonal Frequency Division Multiplexing (OFDMA), and Media Access Control (MAC) have been introduced to enable high data rate,

enhanced spectral efficiency, and to improve efficiency and coverage in Non-Line of Sight (NLOS) situations.

A number of studies^[1] have been carried out to evaluate different WiMAX techniques, and to individually assess uplink/downlink data rate, spectral efficiency, and Quality of Service (QoS). However, such studies have failed to reference overall design objectives. These assessments do not address the fact that WiMAX technology should support multiple types of applications. Thus, evaluation of any individual feature does not help engineers improve design effectiveness in a quantitative manner.

As a result, there is no consensus about how good WiMAX technology and network performance can actually be. There is a lack of systematic and analytical method to improve overall network performance so that long term high-speed data network development plans can be met.

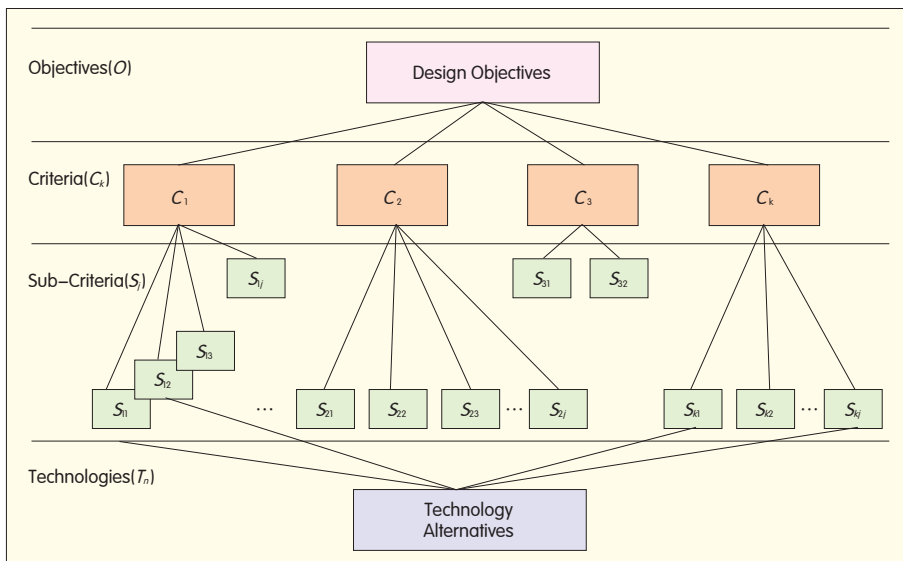
Analytical Hierarchy Process (AHP)

and Data Envelopment Analysis (DEA) can be used to develop a new WiMAX performance model. The purpose of such a model is to establish an agreed-upon performance index (defined as a technological value) that can be measured and compared with design objectives. It is built on the features of broadband wireless access networks, but is not only applicable to WiMAX technology. Indeed, the model can easily be modified for Evolved High Speed Packet Access (HSPA+) and Long Term Evolution (LTE) technologies.

2 Methodology of AHP-Based WiMAX Performance Evaluation Model

2.1 AHP Modeling

In the field of technology management, many studies^[2] have been carried out in which AHP has been adopted to



▲ Figure 1. An AHP hierarchical model using relative values.

evaluate or assess technology by prioritizing different criteria. AHP is a comprehensive framework designed to cope with intuitive, rational, and irrational factors when multi-objective, multi-criterion decisions must be made, and a number of alternatives are possible.

The AHP approach was designed to help decision makers integrate qualitative and quantitative aspects of complex problems. Complex problems are solved by decomposing the structure of the problem into hierarchies, and pair-comparisons are made to determine the priorities in each hierarchy.

Typically, an hierarchical model comprises three or four levels—Objectives, Criteria, Sub-Criteria, and Alternatives. Objectives are on the top level while alternatives are on the lowest level, as shown in Figure 1. Pair-comparisons determine the importance of design criteria to the design objective. The results are represented as a relative value which indicates the technical value of a criteria compared to others. For example, technology 1 may be twice as important as technology 2 in meeting design objectives. These results are useful but insufficient because they do not provide a holistic measurement of the overall performance of WiMAX technology.

They do not show how the technology should progress in meeting objectives.

As a result, the conventional approach of constructing an AHP hierarchical model with technological alternatives at the lowest level is insufficient for measuring overall WiMAX performance. A new hierarchical approach called DEA^[3] is proposed in which the lowest level is replaced with technological efficiency, as shown in Figure 2. Technological efficiency is a semi-absolute value with

a maximum of 100%. This enables technologies to be measured in a simple and comparable format.

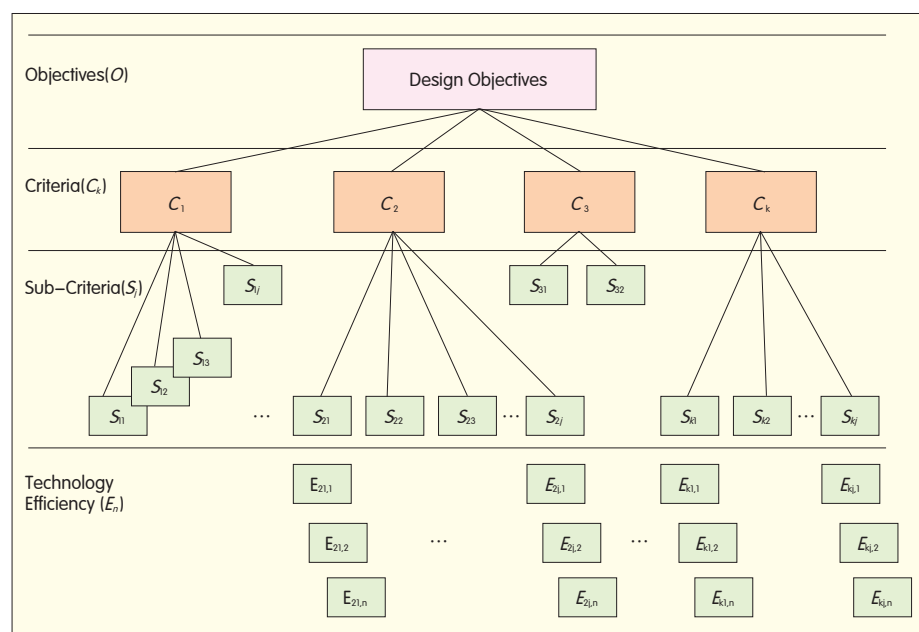
2.2 Integrating DEA into AHP Model

DEA is one of the key methods for measuring technological efficiency of WiMAX or Broadband Wireless Access (BWA). It is a multi-factor efficiency analysis model for determining the relative efficiencies of a set of decision making units or criteria.

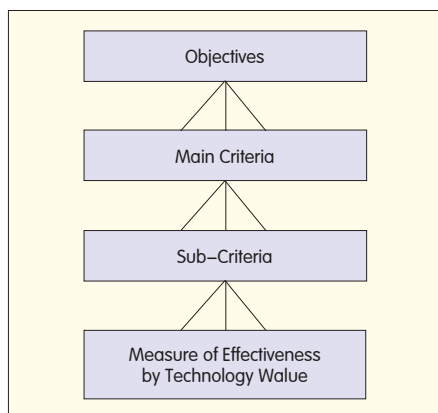
Technological efficiency is represented as a ratio of the maximum sum of output to the relevant input bundle. With this approach, technological criteria are evaluated according to a semi-absolute value; ideal technological efficiency in terms of network performance is 100.

Telecom systems—including wireless networks—can be compared in this way. However, never before has DEA been applied to WiMAX or BWA technology where multiple design criteria are used to assess efficiency.

Unlike traditional wireless communications in which capacity or spectral efficiency is the benchmark for assessing technology, multiple design criteria must be considered when measuring the performance of WiMAX technology that supports not only voice, but also converged data, multi-media,



▲ Figure 2. A proposed hierarchical model using technological efficiency measurement.



▲ Figure 3. An AHP hierarchical model using technological value for measurement.

and other IP applications.

By combining AHP and DEA methodology, design criteria can be weighed and prioritized before being integrated into the DEA framework for deriving technological efficiency. The technological efficiencies for each design criteria are then aggregated and transformed into the technology value, which is a quantifiable yardstick for measuring overall WiMAX performance.

The methodology is as follows:

(1) Performance measuring criteria for WiMAX technology must be identified and categorized into main performance groups. Main design criteria and sub-criteria are used to build the AHP hierarchical model as shown in Figure 3.

(2) Using AHP, the basic elements of each level—such as the measuring criteria and sub criteria—are then compared in pairs with respect to the design objectives. The comparison is based on a 9 point scale that indicates the relative importance of each criteria/sub-criteria.

(3) DEA framework is used to determine the technological efficiency for each design criteria. These technological efficiencies can then be used to evaluate the technology based on the degree to which its design criteria satisfy the desirability on the measure of effectiveness of each design factor.

(4) By integrating AHP priority with technological efficiency within the DEA framework, the technological value of

WiMAX can be derived. This is a measure of WiMAX performance in relation to technological expectations, and can be represented by the mathematical model:

$$TV_n = \sum_{k=1}^K \sum_{j=1}^J f_{k,j} \times V_{(t_n,k,j)} + W_k \times V_{(t_n,k)} \quad (1)$$

In Equation (1),

- TV_n : Technological value for measuring the performance of WiMAX in relation to technological expectations;
- W_k : Relative priority of main criterion (k) with respect to the company's design objective;
- $f_{k,j}$: Relative priority of sub criterion (j) with respect to the main criteria (k);
- t_n, k, j : Performance and physical characteristics of design factor (n) along with main criteria (k) and sub criterion (j);
- t_n, k : Performance and physical characteristics of design factor (n) along with main criteria (k);
- $V_{(t_n,k,j)}$: Desirability value of the performance and physical characteristics of technology (n) along with sub criterion (j) for main criterion (k);
- $V_{(t_n,k)}$: Desirability value of the performance and physical characteristics of technology (n) for main criterion (k).

3 Characteristics of WiMAX Technology

WiMAX is a BWA technology, and BWA is defined as wireless access with broadband connection capabilities. Broadband should have instantaneous bandwidths greater than 1 MHz and support data rates greater than 1.5 Mbit/s (IEEE standard for Local and Metropolitan area networks Part 16 2009). This is the bare minimum for widespread success. Broadband wireless systems must deliver multi-megabit per second throughput to end users, and have robust QoS to support voice, data, and multimedia services. Indeed, it should be able to support IP-based applications and services in a highly efficient manner.

To meet stringent service requirements and to overcome constraints imposed by wireless, there are several technological challenges

that BWA must overcome^[4].

(1) Data Rate

Reliable transmission and reception schemes need to be developed to push broadband data through hostile wireless channels. Typically, services require high data rates and high-speed mobility. Wireless communication relies on complex radio wave propagation mechanisms so that signals can traverse through the intervening environment.

Several factors affect or impair transmission:

- Decay of signal power in an NLOS environment;
- Inter-symbol interference in a multipath environment when time delay between various signal paths becomes significant;
- Multipath fading caused by large variations in the amplitude of the received radio signals;
- Doppler effect due to relative motion between transmitter and receiver.

(2) Spectral Efficiency

High spectral efficiency and broad coverage must be achieved because only a limited amount of spectrum is available to deliver broadband services to a large number of users. The capacity of a wireless system is closely related to its frequency usage. For higher capacity and spectral efficiency, frequency reuse must be maximized. But this can significantly increase the potential for interference. The challenge, therefore, is to design a transmission scheme that can operate with a low Signal to Interference plus Noise Ratio (SINR).

The signal level depends on cell radius and the number of base stations within the coverage area. In order to maintain a certain spectral efficiency within the coverage area, the following should be considered:

- The cell radius of the base station;
- The number of base stations and cost required for the coverage area;
- An effective method of ensuring reliability under low signal to SINR.

(3) Policy Based QoS

Broadband Wireless networks must support a variety of voice, data, and multimedia applications. Each



application has a different traffic pattern and QoS requirement^[5]. Therefore, policy-based QoS should be enforced so that service provision levels are appropriate to different subscriber plans.

QoS must be delivered end to end across both the wireless link and underlying IP network. The challenge lies in accommodating variable QoS requirements across application, services, and user profile, while maintaining the quality levels as defined for all users. Design factors affecting QoS include:

- Throughput that handles multiple traffic flow;
- Packet loss of different applications;
- RAN latency in handover between cells;
- Jitter in the time between idle to transition.

(4) Mobility Management

Mobility is a key element for wireless services. However, support for a subscriber station moving over a large coverage zone leads to two important design considerations^[6],

- Roaming: A means of reaching inactive users for session initiation and packet delivery must be provided regardless of their location within the network;
- Handoff: A continuous, uninterrupted session must be maintained with the base station while on the move—even at vehicular speeds.

(5) Portability and Device Management

Portability requires the subscriber device to be lightweight, battery powered, and energy efficient so that

re-charging is not necessary for an extended period of time^[7].

To reduce power consumption, WiMAX technology should support a power-efficient transmission scheme, power saving protocol for fast switching technology, signal processing algorithm, and it must be interoperable with other radio channels (e.g. EVDO, Wi-Fi) within

the subscriber station.

(6) Security

Security is always a major consideration in wireless communication systems. End users are often concerned about privacy and data integrity while network providers are primarily concerned about preventing unauthorized use of network services. WiMAX networks should support:

- Data encryption to avoid eavesdropping over the communication link;
- Authentication of legitimate end users who access the network;
- Access control over subscription applications or services.

(7) Converged IP-Based Architecture

IP-based architecture can facilitate rapid development of new services and applications because there is a large development community for leveraging. An IP-based system also tends to be cheaper because there are more supplies in markets that have already been adopted by wired communication systems.

Although IP-based protocol is simple and flexible, it is not necessarily efficient and robust enough to operate with limited bandwidth in wireless environments. Thus, making IP protocol more bandwidth efficient and guaranteeing QoS is of major concern in WiMAX systems.

(8) Integration with Legacy Infrastructure

For service operators with existing wireless infrastructure, it is important to leverage the existing platform and roll out wireless broadband services in a timely and cost effective manner.

Although WiMAX employs a different RF access technology, integration of existing base stations, Access Services Network (ASN), and Connectivity Service Network (CSN) can bring about seamless upgrade and easy migration of the existing customer base^[8].

4 AHP Hierarchical Model for WiMAX Performance Evaluation

AHP prioritizes and assesses network performance criteria/sub-criteria in relation to design objectives. It is particularly useful where certain design criteria such as converged IP architecture and security are not quantifiable. Paired comparisons between criteria support logical and rational decision making.

AHP formulates a model by structuring network performance criteria in hierarchies consisting of three levels: services operator's design objectives at the highest level, followed by categorized performance criteria, and then performance sub-criteria at the lowest level.

5 Case Study—An AHP Model for Evaluating Performance of an Operator's WiMAX Network

The AHP model was adopted by a leading WiMAX service provider to measure their network performance against their desired target. The data—which was collected in 2009—included three years (2009–2011) desired performance data for each design criteria/sub-criteria. The model converted performance data into technological efficiency, which became an integral part of the technological value of the company's WiMAX network. The technological value was an index for measuring the extent to which the WiMAX network was meeting the provider's desired performance level.

5.1 Prioritization of the AHP Model

Technological characterization was

▼ Table 1. An AHP model with relative priority from a group of industrial experts

Main Criteria (C_i)	Relative Priority of main Criterion (W_i)	Sub-Criteria (S)	Relative Priority of Sub-Criterion ($f_{k,j}$)
C1: High Data Rate	0.312	S11: Non Line of Sight Coverage	0.12
		S12: Mitigate Intersymbol Interference	0.108
		S13: Diversity Management for Multipath Fading	0.058
		S14: Suppression of Doppler Effect	0.026
C2: High Spectral Efficiency	0.163	S21: Coverage by Increasing the Cell Radius	0.076
		S22: Reliability Under Low Signal to SINR	0.056
		S23: Cost of Building Base Station	0.031
C3: Policy Based QoS	0.095	S31: Throughput Handling Multiple Traffic Flow	0.036
		S32: Packet Loss for Support Various Applications	0.024
		S33: Latency	0.025
		S34: Jitter	0.01
C4: Mobility Management	0.107	S41: Roaming	0.042
		S42: Handoff Across Multiple Sites and Network	0.064
C5: Portability and Device Management	0.09	S51: Power Efficient Modulation to Support Low Power Device	0.046
		S52: Power Saving Protocol for Fast Switching Algorithm	0.025
		S53: Signal Processing Algorithm	0.011
		S54: Interoperability with Other Radio Channel	0.009
C6: Security	0.076	S61: Encryption to Protect Privacy	0.03
		S62: Authentication for Identification	0.033
		S63: Access Control	0.013
C7: Low Cost Convergence IP Architecture	0.127	S71: IP Protocol Bandwidth Efficiency	0.04
		S72: Delivery QoS Limited Bandwidth	0.087
C8: Integration with Legacy Infrastructure	0.031	S81: Integrate with Mobile Station	0.016
		S82: Integrate at Access Service Network (ASN)	0.008
		S83: Integrate at Connectivity Service Network (CSN)	0.007

▼ Table 2. Technological metrics of the service provider's WiMAX performance

Main Criteria	Sub-Criteria	Measurement Unit	Yr2009	Yr2010	Yr 2011
C1: High Data Rate	S11: Non Line of Sight Coverage	UL/DL (Mbit/s)	UL:4	UL:4.8	UL:5.2
	S12: Mitigate Intersymbol Interference		DL:18	DL:21.6	DL:23.4
	S13: Diversity Management for Multipath Fading				
	S14: Suppression of Doppler Effect				
C2: High Spectral Efficiency	S21: Coverage by Increasing the Cell Radius	UL/DL (Mbit/s)	UL:1.6	UL:1.92	UL:2.08
	S22: Reliability Under Low Signal to SINR		DL:2.4	DL:2.88	DL:3.12
	S23: Cost of Building Base Station				
C3: Policy Based QoS	S31: Throughput Handling Multiple Traffic Flow	UL/DL (Mbit/s)	UL:1.6	UL:1.92	UL:2.08
	S32: Packet Loss for Support Various Applications		DL:2.4	DL:2.88	DL:3.12
	S33: Latency	Percent	<3%	<2%	<1%
	S34: Jitter	<ms	120ms	100ms	80ms
C4: Mobility Management	S41: Roaming	Compare with 3G/HSDPA (5-Very Good, 4-Good, 3-Average, 2-Poor, 1-Very Poor)	AVe	G	VG
	S42: Handoff Across Multiple Sites and Network	5 point scale (5-Very Good, 4-Good, 3-Average, 2-Poor, 1-Very Poor)	AVe	G	VG
	S51: Power Efficient Modulation to Support Low Power Device	dB	10	20	20
	S52: Power Saving Protocol for Fast Switching Algorithm	mW	200	150	100
C5: Portability and Device Management	S53: Signal Processing Algorithm	No of different radio channel (1 to 5)	0	1	2
	S54: Interoperability with Other Radio Channel				
C6: Security	S61: Encryption to Protect Privacy	Compare with 3G/HSDPA (5-Very Good, 4-Good, 3-Average, 2-Poor, 1-Very Poor)	VG	VG	VG
	S62: Authentication for Identification				
	S63: Access Control				
C7: Low Cost Convergence IP Architecture	S71: IP Protocol Bandwidth Efficiency	Relative cost compared with 3G/HSDPA (5-Very Good, 4-Good, 3-Average, 2-Poor, 1-Very Poor)	VG	VG	VG
	S72: Delivery QoS Limited Bandwidth				
C8: Integration with Legacy Infrastructure	S81: Integrate with Mobile Station	Backward compatibility (5-Very Good, 4-Good, 3-Average, 2-Poor, 1-Very Poor)	AVe	G	VG
	S82: Integrate at Access Service Network (ASN)				
	S83: Integrate at Connectivity Service Network (CSN)				

defined by a group of WiMAX service providers, who also provided judgement on the relative priority of design criteria/sub-criteria. An online evaluation tool was used during face to face interviews. Operators prioritized the importance of each criteria by paired comparisons with other criteria on the same level. The comparison warranted the relative importance of each criteria and sub-criteria, and the results are summarized in Table 1.

The relative priority for the main criteria (W_i) and the sub-criteria ($f_{k,j}$) is an integral part of the AHP model.

5.2 Technological Assessment with Desired Performance Targets

The WiMAX operators estimated their desired network performance levels from 2009 to 2011, and this became the desirable output under the DEA framework. The data was then integrated into the AHP model to become the technological metrics for each of the measuring criteria, as shown in Table 2.

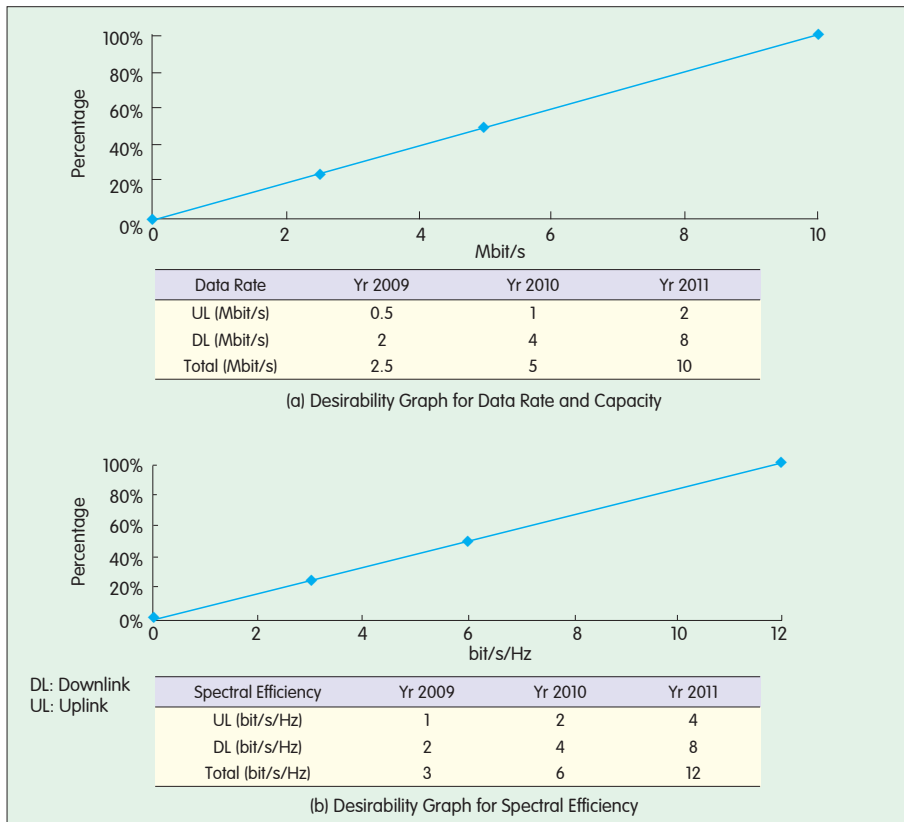
Desirability graphs representing operators' preferences for the technological metrics of each design criteria were drawn. Figure 4 shows examples of desirability graphs for data rate and spectral efficiency.

The technological value represents the overall performance of the AHP model after integrating all the technological efficiencies of the design criteria. For the WiMAX study in 2009, this was 50%. As WiMAX technology continues to improve, the technological value is expected to improve to 70% by 2010, and then to 100% by 2011.

5.3 Formation of the Performance Evaluation Model

The growth rate of technological value shows WiMAX service providers how design criteria should be improved in order to meet desired performance targets.

High data rate (0.312) is still the dominant factor affecting overall performance. Any improvement in NLOS coverage and/or interference cancellation would also greatly improve technological value. For example, technological value increases from 8%



▲ Figure 4. Desirability graphs for data rate and spectral efficiency.

to 16% between 2009 and 2010, and reaches 31% by the end of 2011.

High Spectral efficiency (0.163) is the second most important factor. Given that frequency spectrum is the most limited resource in a wireless network ecosystem, operators are concerned about enhancing efficiency and maximizing the coverage and number of end users. Technological value for the operator increased from 4% to 8% between 2009 and 2010, and will reach 16% by 2011.

The contribution of the above two criteria alone affects the overall WiMAX network performance target by 47%. The other six design factors affect the overall performance target by 53%. This model provides a clear indication of what resource investments need to be made into the network, and monitors performance improvement in a quantitative way.

6 Conclusions

The performance evaluation model

provides a quantitative approach to measuring an operator's WiMAX network performance against their desired targets. It is not designed for comparing one mobile operator's network performance with another's, as performance targets differ according to market strategy and investment model.

However, achieving targets very much depends on equipment suppliers delivering products that align with the operator's timetable. This model can also be extended to equipment suppliers for the purpose of validating performance targets of individual criteria over some years of study, as well as to enhance supplier roadmaps for future network development. The enhanced model could strengthen partnerships between mobile operators and equipment suppliers.

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Biographies

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Research on Convergence Network of EPON and WiMAX Based on ROF

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Abstract:

Convergence of optical and wireless networks is a promising development for future access network architecture. A Radio over Fiber (ROF)-based network that converges Ethernet Passive Optical Network (EPON) and Worldwide Interoperability for Microwave Access (WiMAX) technologies makes it possible to simultaneously transmit EPON baseband signals and WiMAX wireless Radio Frequency (RF) signals. This article elaborates on uplink and downlink transmission, redundancy protection, and roaming features of such a network.

and QoS performance^[5]. However, the BSs are required to convert digital baseband signals into wireless RF signals, which may be very expensive especially when wireless signals are high frequency. Large numbers of BSs are required for wide coverage and to compensate for the quick fading of wireless signals.

The other solution is Radio over Fiber (ROF) technology^[6]. An ROF system uses optical fiber as the transmission link between BS and Central Station (CS) and uses an optical carrier to transmit RF signals. Optical fiber is only used for transmission, while the CS is responsible for switching, control and signal regeneration. The BS performs optical-electrical conversion. In this way, complex and expensive devices can be placed at the CS to be shared by multiple remote BSs, and power consumption costs can be reduced. Many technical papers have been written on WLAN and ROF convergence, as well as on successful commercial cases of GSM and ROF convergence. Research into transmission of WiMAX in Synchronous Optical Networks (SONET) is currently being undertaken^[7-8].

Broadband access technology has come into widespread use in line with wideband-consuming services such as Internet Protocol Television (IPTV) and Video on Demand (VoD). To further optimize resources, cut down costs, and provide wider and more flexible broadband access services, convergence of optical and wireless networks is possible in future architecture^[1-2]. At present, application areas of Ethernet Passive Optical Network (EPON)^[3] are being expanded because of its low cost, wide bandwidth and structural Ethernet advantages. On the other hand, Worldwide Interoperability for Microwave Access (WiMAX) technology specified by IEEE 802.16^[4] is becoming a mainstream wireless broadband access technology. Compared to Wireless Local Area Network (WLAN) access technology, WiMAX provides wider bandwidth, broader coverage and better Quality of Service (QoS). It also provides better data access service compared to cellular technology.

Convergence of EPON and WiMAX therefore avoids the drawbacks within each individual technology, making full use of the wide bandwidth of optical access technology and flexibility of wireless technology. This gives users a better experience, and also decreases construction and maintenance costs across the entire network.

1 Current Solutions to EPON and WiMAX Convergence

There are currently 2 solutions to EPON and WiMAX convergence. One solution is baseband optical fiber transport technology, which transports digital baseband signals directly through optical fiber. The Optical Network Units (ONUs) of EPON and WiMAX Base Stations (BSs) are directly connected through the Ethernet interface, and Ethernet frames are transmitted via optical fiber. This scheme can be easily implemented, and has wide coverage. It also has good bandwidth allocation

2 Architecture of a ROF-Based EPON and WiMAX Converged System

In contrast to a standard EPON system, a new Optical Line Terminal (OLT) and an Optical Network Unit Base Station (ONU-BS) are introduced in a ROF-based EPON and WiMAX converged system.

For the sake of reducing costs and concentrating processing, the devices originally on the BS WiMAX physical layers are moved to the central node OLT. The IEEE 802.16 physical layer

defines a working frequency of 2–66 GHz and 2 modulation modes: single-carrier modulation and Orthogonal Frequency Division Multiplexing (OFDM) modulation. OFDM is used on 2–11 GHz frequency. Supporting Non-Line of Sight (NLOS) transport, OFDM technology resists attenuation and multipath. OFDM modulation and demodulation devices are used in this architecture.

There are 3 usable frequencies under 11 GHz: 2.5 GHz, 3.5 GHz, and 5.8 GHz. Considering WiMAX deployment, this architecture adopts 3.5 GHz. Since there is no specifically defined carrier bandwidth, a WiMAX system can work at a bandwidth between 1.25–20 MHz. This architecture adopts a bandwidth of 20 MHz, but ignores the problem of co-channel interference. In actual application, frequency reuse and sector partitioning technologies can make better use of frequency resources and system throughput. In the deployment of an ROF system, direct modulation is usually adopted if the RF signals are below 10 GHz, and external modulation is above 10 GHz. Direct modulation is used in this architecture.

In the downlink direction, this architecture adopts Subcarrier Multiplexing (SCM) technology at the OLT end in order to simultaneously transmit EPON wireline baseband signals and wireless RF signals, and also to distinguish RF signals that belong to different BSs. This architecture defines the EPON baseband signals at 0–2.5 GHz, and up-converts the radio signals to subcarriers above 3.5 GHz. The bandwidth of each subcarrier is 20 MHz, and the central frequency interval is 0.1 GHz. One subcarrier corresponds to one BS; for an EPON system with a branching ratio of 1:16, there are 16 subcarriers in total. The laser is modulated after the baseband signal and modulated subcarrier have been integrated. Since the baseband signal and subcarrier remain at different frequencies, no interference is produced. Baseband and radio signals are simultaneously transported in the downlink. When the remote ONU-BS

receives mixed signals from the OLT, it demultiplexes the baseband and subcarrier signals, uploads the baseband signal to related devices for processing, and performs down-conversion of the subcarrier signal. The frequency of the local oscillation and the converter used is the same as that of OLT. After moving through the band pass filter, the radio RF signal—originally belonging to the local BS—is demodulated and sent out through the antenna. That is, the BS only performs down conversion and requires no other devices. Costs are therefore lowered.

The ONU-BS uses Time Division Multiple Access (TDMA) mode to upload data in the uplink direction. In this mode, the baseband and radio RF signals are mixed together. The EPON MAC and WiMAX MAC cooperate with each other to allocate transmission timeslots of uplink data for every ONU-BS. When a designated timeslot appears, the ONU-BS sends data according to the authorized OLT window size. In this way, uplink data of all ONU-BSs can be transported in a pre-defined sequence without conflict after they have reached the shared optical fiber. Performing up-conversion on the radio RF signal to convert it into a subcarrier signal at the ONU-BS end is unnecessary. Different ONU-BSs are located in different transmission

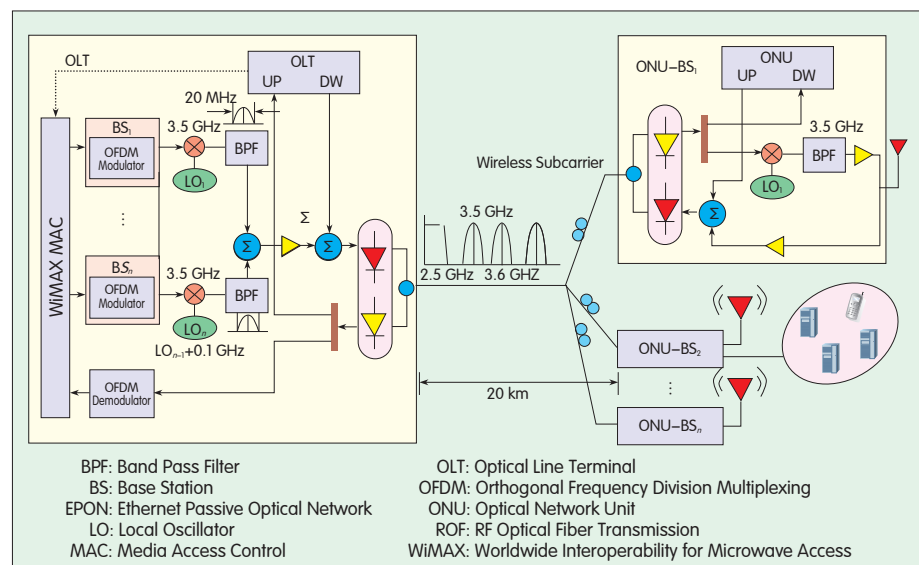
timeslots and their uplink RF signals do not conflict with each other. Moreover, the radio RF signal has a frequency of 3.5 GHz while that of the baseband signal is below 2.5 GHz. Therefore, the baseband signal does not conflict with the radio RF signal, and they can be transmitted at the same time. With this design, the OLT receiver can be simplified by using only an electrophotonic detector and a WiMAX OFDM demodulator. No local oscillation or converter is needed since there is no cross-interference between subcarriers.

3 Features of the ROF-Based Convergence Solution

The architecture of this ROF-Based EPON and WiMAX converged system has the following features:

(1) The BS performs down conversion only. Devices responsible for radio signal processing and some upper-layer functions are placed at the OLT end. For ease of description, the OLT end shown in Figure 1 uses N OFDM modulators (one for each BS), which in practise can be combined into one. At the OLT end, all BSs can share one set of OFDM modulation and demodulation devices.

(2) $N+1$ (rather than $N+N$) protection is implemented because all devices are



▲ Figure 1. ROF-based EPON and WiMAX converged system architecture.

moved to the OLT end. NBSs share just one spare BS. In current radio networks, BSs cannot support redundancy protection, or can only provide 1+1 protection. In other words, one spare BS is needed for every BS. In an ROF-based converged network, system stability is improved and cost is reduced.

(3) OLT serves as a central controller that manages the wireless resources of all BSs and knows the real-time radio resource information of all BSs. In standard WiMAX architecture, wireless resource management is performed at different BSs and information cannot be exchanged within the necessary timeframe. This limits the system's work efficiency. With the help of an algorithm, OLT dynamically allocates wireless resources for every BS in order to make full use of the wireless resource and to balance the loads among BSs. System efficiency is accordingly improved, and can be further improved if Multiple-Input and Multiple-Output (MIMO) and adaptive modulation technologies are employed.

(4) Roaming among BSs is much simpler. OLT processes all data and knows all information, so the control channel between OLTs and every BS is no longer necessary. To support user mobility, roaming in different WiMAX BSs must be taken into consideration. BSs cannot connect with each other, and in any ordinary networking scheme, a control channel is used—between OLT and every WiMAX BS—to exchange control information in real time. When a user roams from one BS to another, OLT sends a command via this control channel to make the original BS disconnect the user and at the same time add them to the local BS. However, in converged architecture, when OLT senses that a user is roaming between different BSs, it simply switches over to a new subcarrier frequency to send data to the user.

The converged system, however, is not without some weaknesses. It is susceptible to fiber attenuation, dispersion, and non-linear effect in optical fiber, as well as cross interference between subcarriers. It also places high requirements on

modulation and detection technology, especially when the radio frequency is high. Moreover, as it takes time for the radio signal to be transported over optical fiber, the longer transmission distance impacts MAC performance^[9] more significantly. Further research is required into the bandwidth allocation algorithm so that uplink transmission timeslots can be better allocated for wireline baseband data and radio RF signals.

4 Conclusion

Architecture that converges optical fiber and wireless networks has a bright future. An ROF-based EPON and WiMAX converged network simultaneously transmits the EPON baseband signal and WiMAX radio RF signal via optical fiber. Such a system combines fixed line and mobile networks, and brings about more satisfactory user experience. Moreover, it can greatly reduce network construction and maintenance costs.

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Cloud Computing (3)

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Editor's Desk:

In the preceding two parts of this series, several aspects of cloud computing—including definition, classification, characteristics, typical applications, and service levels—were discussed. This part continues with a discussion of Cloud Computing Open Architecture and Market-Oriented Cloud. A comparison is made between cloud computing and other distributed computing technologies, and Google's cloud platform is analyzed to determine how distributed computing is implemented in its particular model.

services and solutions vendors, partners, and end users that provide or consume shared resources within the cloud computing environment. Collaboration between vendors and their partners is emphasized in the cloud computing value chain.

(2) Virtualization for Cloud Infrastructure

Hardware virtualization involves managing hardware equipment in plug-and-play mode; software virtualization involves using software image management or code virtualization technology to enable software sharing. Dynamic code assembly and execution is another software virtualization technology. In an Internet application, some JavaScript code elements can be dynamically retrieved and inserted into an Ajax package to create new functions or features for a web client.

6.4 Open Architecture

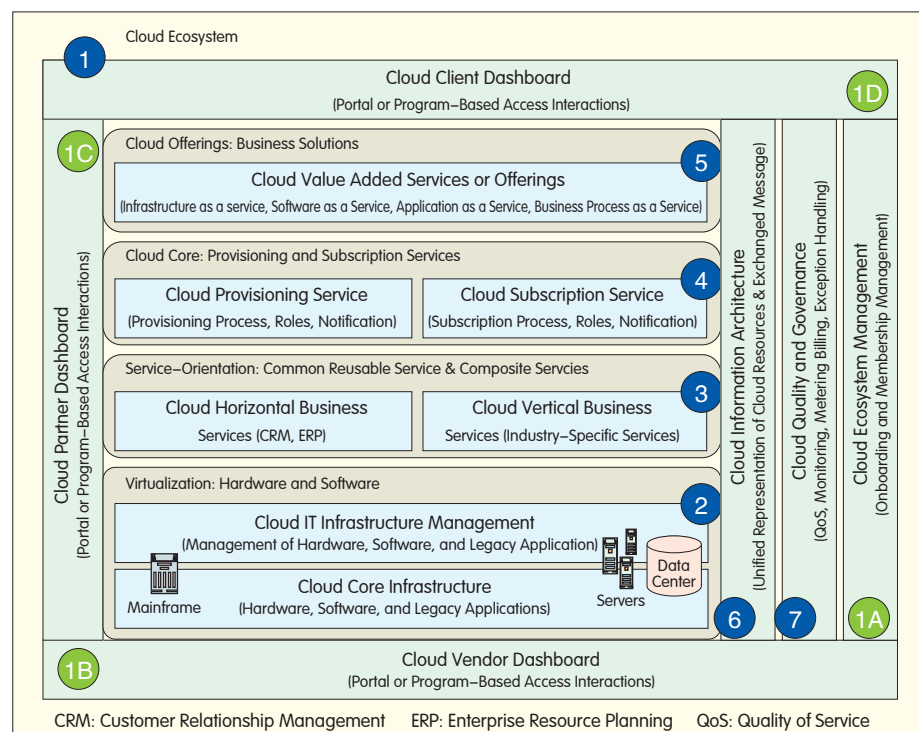
Virtualization is a core technology for enabling cloud resource sharing, and Service-Oriented Architecture (SOA) enables flexibility, scalability, and reusability. By combining these two technologies, researchers have developed an open cloud computing architecture based on the Open System Interconnection (OSI) model, and have used it as the reference model for implementing the cloud computing system, as shown in Figure 6.

This architecture encompasses cloud ecosystem, cloud infrastructure and its management, service-orientation, core provisioning and subscription, composite cloud offerings, cloud information architecture and management, and cloud quality analytics. In designing the architecture, seven basic principles are adopted:

(1) Integrated Management for Cloud Ecosystem

An architecture must support cloud computing ecosystem management.

Such an ecosystem includes all



▲ Figure 6. Cloud computing open architecture.

(3) Service–Orientation

Service–orientation is a driving force that gives cloud computing business value in terms of asset reusability, composite applications, and mashup services. Common services can be reused to enable the cloud's core provisioning and subscription services as well as to build cloud offerings in Infrastructure as a Service (IaaS), Software as a Service (SaaS), and even Business Process as a Service (BPaaS).

(4) Extensible Service Provisioning

This feature is unique to cloud computing systems. Without extensibility, the provisioning part of the cloud architecture can only support a certain type of resource sharing. Free use and paying users can periodically change their roles as service providers or consumers and this change can occur at three levels of service provisioning.

(5) Configurable Enablement for Cloud Offerings

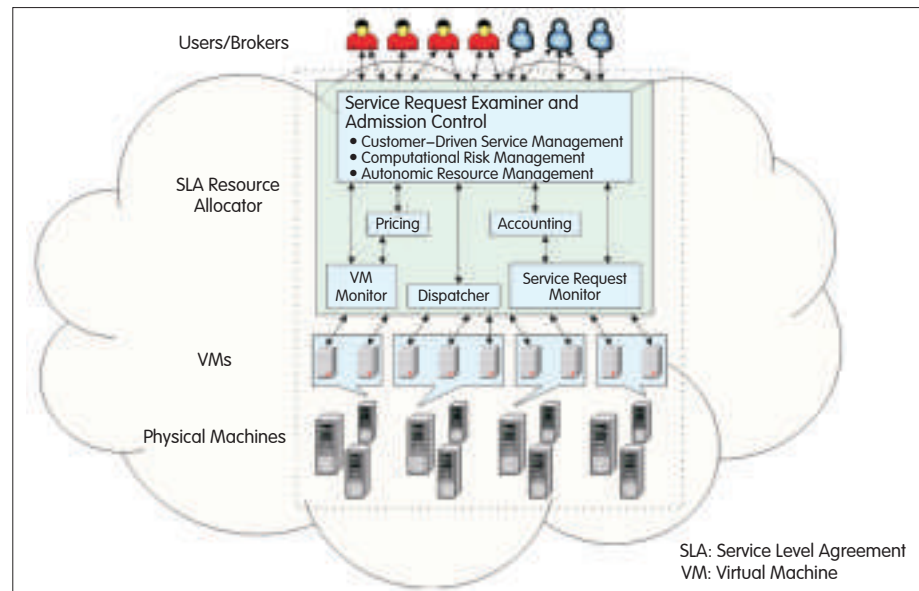
The architecture must ensure configurability of the cloud computing platform and services. The modularized ecosystem management, virtualization, service–orientation, and cloud core form a solid foundation to ensure a computing platform that is configurable, combinable and manageable.

(6) Unified Information Representation and Exchange Framework

The collaborative feature of cloud computing comprises information representation and message exchange between cloud computing resources. Cloud computing resources include all business entities (e.g. cloud clients, partners, and vendors) and supporting resources such as virtualization related modules, service–orientation related modules, cloud core, and cloud offerings. The cloud information architecture module enables representation of cloud entities in a unified cloud computing entity description framework. Message routing and exchange protocols as well as message transformation capability form the foundation of cloud information architecture.

(7) Cloud Quality and Governance

This module identifies and defines



▲ Figure 7. High-level market-oriented cloud architecture.

quality indicators for the cloud computing environment and a set of guidelines to govern the design, deployment, operation, and management of cloud offerings.

In short, the objective of such an architecture is to combine SOA and virtualization technologies in order to exploit the business potential of cloud computing.

6.5 Market–Oriented Cloud

Cloud computing is a new Internet–based resource sharing mode particularly focused on its business model. How, then, does this feature impact cloud computing? Researchers have proposed a market-oriented cloud architecture, global cloud exchange and market infrastructure for trading services, which have been investigated intensively.

6.5.1 Market–Oriented Cloud Architecture

In the article *Cloud Computing and Emerging IT Platform: Vision, Hype, and Reality for Delivering Computing as the 5th Utility*, researchers from the Cloud Computing and Distributed Systems (CLOUDS) Laboratory of the University of Melbourne presented a market-oriented architecture. This architecture supports Quality of Service (QoS) negotiation and Service Level

Agreement (SLA)–based resource allocation in the context of cloud computing, as shown in Figure 7.

In this architecture, there are four main entities:

(1) Users/Brokers

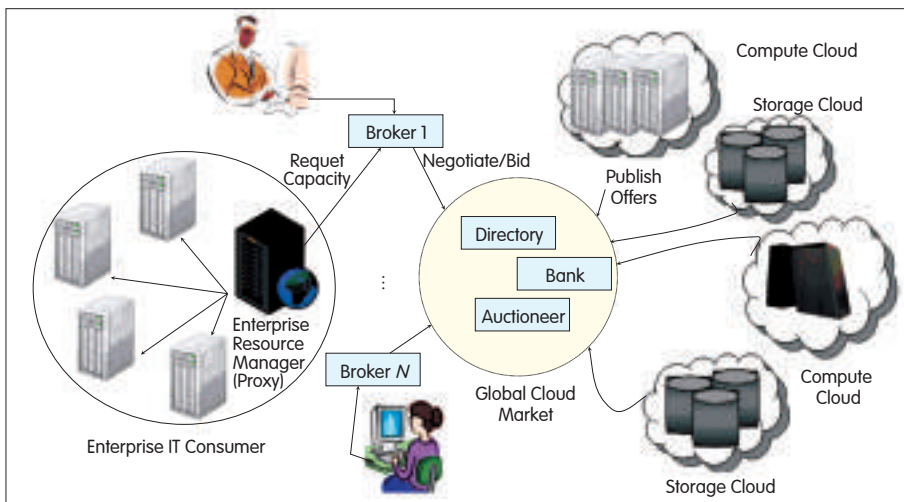
Users (or brokers acting on their behalf) submit service requests from anywhere in the world to the Cloud Computing Center to be processed.

(2) SLA Resource Allocator

The SLA Resource Allocator acts as the interface between Cloud service provider and external users/brokers. It requires the interaction of the following mechanisms to support SLA-oriented resource management.

- Service Request Examiner and Admission Control

When a service request is first submitted, the Service Request Examiner and Admission Control mechanism interprets it for QoS requirements before determining whether to accept or reject it. The mechanism also requires updated status information on resource availability from the Virtual Machine (VM) Monitor mechanism and workload processing from the Service Request Monitor mechanism in order to make effective resource allocation decisions. It then assigns the request to a VM and determines resource entitlements for the allocated VM.



▲ Figure 8. Global cloud exchange and market infrastructure for trading services.

- Pricing

The Pricing mechanism determines how service requests will be charged based on submission time (peak/off-peak), pricing rates, or resource availability.

- Accounting

The Accounting mechanism meters the actual usage of resources by each request so that the final cost can be calculated and charged to the user.

- VM Monitor

The VM Monitor mechanism oversees the availability of VMs and their resource entitlements.

- Service Request Monitor

The Service Request Monitor mechanism oversees the execution progress of service requests.

(3) VMs

Multiple VMs can be activated or stopped dynamically on a single physical machine to meet accepted service requests.

(4) Physical Machines

Multiple computing servers form a resource cluster to meet service demands.

Commercial market-oriented cloud systems must be able to:

- Support customer-driven service management;
- Define computational risk management tactics to identify, assess, and manage risks involved in the execution of applications;
- Devise appropriate market-based resource management strategies that

encompass both customer-driven service management and computational risk management in order to sustain SLA-oriented resource allocation;

- Incorporate autonomic resource management models that effectively self-manage changes in service requirements in order to satisfy both new service demands and existing service obligations;

- Leverage VM technology to dynamically assign resource shares according to service requirements.

6.5.2 Cloud Service Exchanges and Markets

Enterprises currently employ cloud services to improve the scalability of their services and to deal with bursts in resource demand. However, at present, the proprietary interfaces and pricing strategies of service providers prevent consumers from swapping one provider for another. For cloud computing to become mature, services must follow standard interfaces. This would enable services to be commoditized and would pave the way for the creation of a market infrastructure for trading in services.

In cloud computing markets, service consumers expect their specific QoS requirements to be met with minimal expense, and service providers hope to retain their clients while achieving the highest possible Return on Investment (ROI). To achieve this, mechanisms,

tools, and technologies must be developed to represent, convert, and enhance resource value. Figure 8 illustrates a cloud exchange and market system model based on real-world exchanges.

In this model, the market directory allows participants to locate providers or consumers with suitable offers. Auctioneers periodically clear bids and requests received from market participants, and the banking system carries out financial transactions.

Brokers perform the same function in such a market as they do in real-world markets: they mediate between consumers and providers by purchasing from the provider and sub-leasing to the consumer.

Consumers, brokers and providers are bound to their requirements and related compensations through SLAs. An SLA specifies the details of the service to be provided in terms of metrics agreed upon by all parties, and penalties for violating these expectations, respectively. Such markets can bridge disparate clouds, allowing consumers to choose a suitable provider by either executing SLAs in advance or by purchasing capacity on the spot. Providers can set the prices for a resource based on market conditions, user demand, or current level of utilization of the resource. The admission-control mechanism at the provider end is responsible for selecting the auctions to participate in or the brokers to negotiate with. The negotiation process continues until an SLA is formed or the participants decide to break off. Brokers profit from the difference between the cost of leasing the resource, and what they charge consumers to gain a share of the resource. A broker, therefore, must choose both consumers and providers. Consumer demands include deadlines, fidelity of results, turnaround time of applications, and budget limitations. Enterprise consumers can deploy their own limited IT resources into clouds as guarantees for enterprise computing, or they can lease providers' resources to upscale their applications.

The idea of utility markets for

computing resources has been around for a long time. Recent research projects have particularly focused on trading VM-based resource allocation by time slices. In the above model, a resource broker can negotiate with resource providers. Based on enterprise Grid, Melbourne University's CLOUDS Laboratory implements a market-oriented platform called "Aneka", which is also a NET-based service-oriented resource management platform. Aneka exhibits many of the properties of the cloud computing model.

6.6 Comparison of Cluster, Grid, and Cloud Computing

The first part in this series briefly introduced some characteristics of cloud computing that can be directly experienced by users. This part, however, discusses some of the technical characteristics that distinguish cloud computing platforms from cluster and grid computing.

Although cloud platforms share some common characteristics with clusters and Grids, they have their own unique attributes and capabilities. These include support for virtualization, services with Web Service interfaces that can be dynamically composed, and support for the creation of third-party, value-added services by building on cloud compute, storage, and application services. Table 1 compares the key characteristics of cluster, grid and cloud computing systems.

7 Cloud Computing Model

Although enterprises and academic researchers have proposed various cloud system models, most of these do not reveal the computing paradigm for problem solving in a cloud. To enable communication and collaboration between server clusters within a cloud, Google has introduced Google File System (GFS), BigTable, and MapReduce technologies—so-called the "three sharp weapons" of cloud computing. With these technologies, Google has formed a cloud with thousands or even millions of

▼ Table 1. Key characteristics of clusters, grids, and cloud systems

Characteristics	Systems		
	Clusters	Grids	Clouds
Population	Commodity Computers	High-End Computers (Servers, Clusters)	Commodity Computers, High-End Servers, and network Attached Storage
Size/Scalability	100	1,000	100 to 1,000
Node OS	Linux, Windows	Mainly Unix	Multiple OSs Running on Virtual Machines
Ownership	Single	Multiple	Single
Interconnection Network/Speed	Dedicated, High-End Network with Low Latency and High Bandwidth	Mostly Internet with High Latency and Low Bandwidth	Dedicated, High-End Network with Low Latency and High Bandwidth
Security/Privacy	Traditional Login/Password-Based; Medium Level of Privacy (Depends on User Privileges).	Public/Private Key Pair Based Authentication and Mapping a User to an Account; Limited Support for Privacy.	Each User/Application is Provided with a Virtual Machine; High Security/Privacy is Guaranteed; Support for Setting Per-File ACL.
Discovery	Membership Services	Centralized Indexing and Decentralized Information Services	Membership Services
Service Negotiation	Limited	Yes, SLA Based	Yes, SLA Based
User Management	Centralized	Decentralized, VO-Based	Centralized or Delegated to Third Party
Resource Management	Centralized	Distributed	Centralized/Distributed
Allocation/Scheduling	Centralized	Decentralized	Centralized/Decentralized
Standards/Inter-Operability	VIA-Based	Some Open Grid Forum Standards	Web Services (SOAP and REST)
Single System Image	Yes	No	Yes, but Optional
Capacity	Stable and Guaranteed	Varies, but High	Provisioned on Demand
Failure Management (Self-Healing)	Limited (Failed Tasks/Applications are Generally Restarted).	Limited (Failed Tasks/Applications are Generally Restarted)	Strong Support for Failover and Content Replication; VMs can be Easily Migrated from One Node to Another
Service Pricing	Limited, in Non-Open Market	Mainly Internal Pricing	Pricing According to Effects, with Discounts for VIP Clients
Internetworking	Multi-Clustering within an Organization	Limited	Highly Potential; Support Third-Party Solution Providers' Loose Coupling of Services from Different Clouds
Application Driven	Science, Business, Enterprise Computing, Data Centers	Collaborative Scientific and High Throughput Computing Applications	Dynamically Provisioned Legacy and Web Applications, Content Delivery
Potential for Building Third-Party Value-Added Solutions	Limited	Limited, Mainly Scientific Computing Oriented	Highly Potential; Create New Services by Dynamically Provisioning of Computing, Storage, and Application Services and Offer Individual or Combined Cloud Services to Users
ACL: Access Control List OS: Operating System REST: Representational State Transfer			SOAP: Simple Object Access Protocol VIA: Virtual Interface Architecture VM: Virtual Machine

computers, creating a powerful data center.

7.1 GFS File System

Desktop applications differ from Internet applications in many respects. GFS is a proprietary distributed file system developed by Google Inc. It is designed to allow efficient and reliable access to data by using large clusters of commodity hardware. GFS is optimized for Google's core data storage and usage needs (primarily the search engine), which can generate enormous amounts of data that needs to be retained. Google's Internet search computing learns from the functional programming paradigm in which operations do not modify original

data but generate new computing data. Therefore, one feature of GFS is that it generates a large number of very large files mainly for reading. These files can be appended but rarely re-written. GFS is also characterized by high data throughput.

There are two types of GFS nodes: one master node, and a large number of chunkservers. Chunkservers store data files, with each individual file broken up into fixed-sized chunks of 64 megabytes. Each chunk is assigned a unique 64-bit label to maintain logical mappings of files to constituent chunks. The master node only stores metadata associated with the chunks, such as the tables mapping the 64-bit labels to chunk locations and the files they make



up, the locations of the copies of the chunks, what processes are reading or writing to a particular chunk, or taking a "snapshot" of the chunk pursuant to replicating it. This metadata is kept current by the master node as it periodically receives updates from each chunk server.

Modification permissions are handled by means of time-limited "leases". The master node grants permission for a process to modify a chunk within a given period. The modifying chunkserver, which is always the primary chunk holder, then propagates the changes to chunkservers with backup copies for synchronization. With several redundant copies, reliability and availability are guaranteed. Programs access the chunks by first querying the Master server for the locations of the desired chunks, the Master replies with the locations, and the program then contacts and receives the data from the chunkserver directly.

Google currently has over 200 GFS clusters, each of which consists of 1,000 to 5,000 servers. Using GFS, Google has proven that clouds built on cheap machines can also deliver reliable computing and storage.

7.2 BigTable Database System

BigTable is a compressed, high performance proprietary database system built mainly on GFS and Chubby Lock Service. It is also a distributed system for storing structured data. A BigTable is a sparse, distributed, multi-dimensional sorted map, which can be indexed by a row key, column key, and a timestamp. By allowing a client to dynamically control

data layout, storage format, and storage location, BigTable meets application demands for localized access. Tables are optimized for GFS, being split into multiple tablets of about 200 megabytes. The locations in the GFS of tablets are recorded as database entries in multiple special tablets, which are called "META1" tablets. META1 tablets are found by querying the single "META0" tablet. The META0 tablet typically has a machine to itself which is queried by clients by clients for the location of the META1 tablets; and consequently, the location of the actual data.

BigTable is designed for databases of petabyte scale with data across thousands of servers. It is also designed to accommodate more machines without the need for reconfiguration.

7.3 MapReduce Distributed Programming Paradigm

GFS and BigTable are used by Google for reliable storage of data in a large-scale distributed environment. Google's MapReduce is a software framework designed to support parallel computing on large data sets (often greater than 1 terabyte) on a large cluster. It is therefore a computing model specifically designed for cloud computing.

7.3.1 Software Framework

The MapReduce software framework design is inspired by two common programming functions: "Map" and "Reduce". It was developed within Google as a mechanism for processing large amounts of raw data; for example, counting the number of occurrences of

each word in a large set of documents. In functional programming, map and reduce are tools for constructing higher-order functions.

Map applies a given function to a list of elements (element by element) and returns a new list. These new elements are the products of the function applied to each element in the original list. For example, $\text{Map } f[v_1, v_2, \dots, v_n] = [f(v_1), f(v_2), \dots, f(v_n)]$. In this way, the functions can be computed in parallel. The MapReduce computing model is suitable for applications requiring high-performance parallel computing. If the same computing is required on a large set of data, the data set can be divided and assigned to different machines for computing.

Reduce involves combining elements of a list using a computing approach (function). To unfold a binary operation (function) into a n -ary operation (function), the reduce function is used: $\text{Reduce } f[v_1, v_2, \dots, v_n] = f(v_1, (\text{Reduce } f[v_2, \dots, v_n]) = f(v_1, f(v_2, (\text{Reduce } f[v_3, \dots, v_n])) = f(v_1, f(v_2, f(\dots f(v_{n-1}, v_n) \dots)))$. MapReduce computing model combines the intermediate results obtained from Map operations by Reduce operations until the final result is calculated.

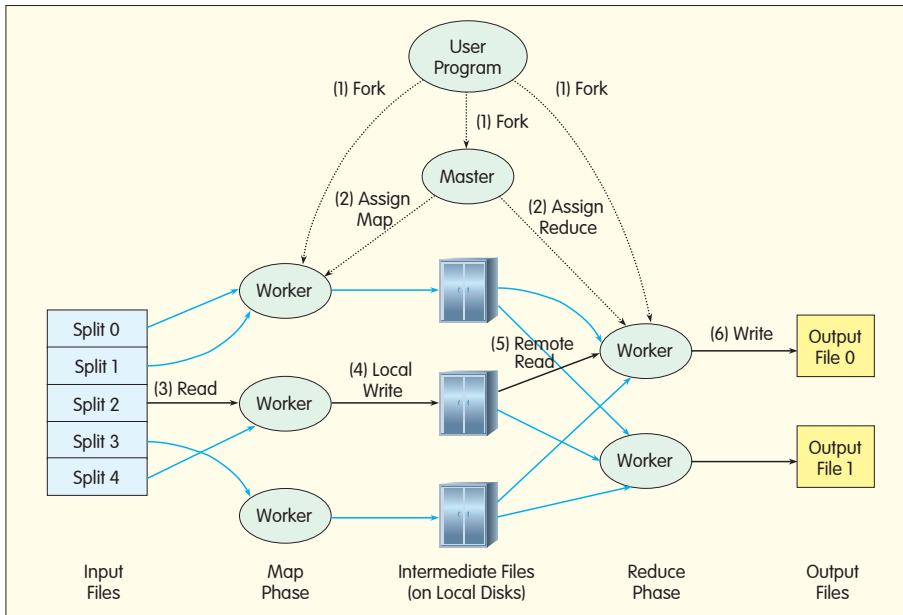
7.3.2 Execution Procedure

Map invocations are distributed across multiple machines by automatically partitioning the input data into a set of splits or shards. Reduce invocations are distributed by partitioning the intermediate key space into pieces using a partitioning function. When the user program calls the MapReduce function, the overall operation flow is illustrated in Figure 9.

The MapReduce library in the user program first splits the input files into M pieces. It then starts up many copies of the program on a cluster of machines.

One of the copies of the program—the master—is special; the rest are workers. The master picks idle workers and assigns each one a map task or a reduce task.

A worker that is assigned a Map task reads the contents of the corresponding input split. It parses key/value pairs out of the input data and



▲ Figure 9. Google MapReduce execution procedure.

passes each pair to the user-defined map function. The intermediate key/value pairs produced by the Map function are buffered in memory.

The buffered pairs are periodically written to local disk and partitioned into R regions by the partitioning function. The locations of these buffered pairs on the local disk are passed back to the master, which is responsible for forwarding these locations to the Reduce workers.

When a Reduce worker is notified by the master about these locations, it uses remote procedure calls to read the buffered data from the local disks of map workers. When a Reduce worker has read all intermediate data for its partition, it sorts it by the intermediate keys so that all occurrences of the same key are grouped together.

The Reduce worker iterates over the sorted intermediate data and for each unique intermediate key encountered, it passes the key and the corresponding set of intermediate values to the user's

reduce function. The output of the reduce function is appended to a final output file for this reduce partition.

When all Map and Reduce tasks have been completed, the master wakes up the user program. At this point, the MapReduce call in the user program returns back to the user code.

Upon completion, the output of the MapReduce execution is available in the R output files. Typically, users do not need to combine these R output files into one file; they often pass these files as input to another MapReduce call or use them from another distributed application.

7.4 Apache Hadoop Distributed System Infrastructure

Google's GFS, BigTable, and MapReduce technologies are open to the public but their implementation is private. A typical implementation of these technologies in the open source community involves the Apache Hadoop project. Inspired by Google's

MapReduce and GFS, Hadoop is an open-source Java software framework consisting of functional programming-based concurrent computing model, and a distributed file system. Hadoop's HBase, similar to BigTable distributed database, supports data-intensive distributed applications to work with thousands of nodes and petabytes of data.

Hadoop was originally developed to support distribution for the Nutch search engine project. Yahoo has invested a great deal of money into the project and uses Hadoop extensively in areas such as web search and advertising. IBM and Google have launched an initiative to use Hadoop to support university courses in distributed computer programming. All these have been instrumental in promoting and popularizing cloud computing worldwide.

(To be continued)

Biographies

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Wang Bai is a professor and vice dean of the School of Computer Science and Technology, Beijing University of Posts and Telecommunications (BUPT). Her research interests include next generation telecom operation supporting system, distributed computing technologies, and visualization analysis of complex networks. She has published more than 60 papers and 3 books.

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AD Index

A1, Back Cover:
ZTE Corporation

Abbreviation Index

A

ACL: Access Control List
AF: Assured Forwarding
AGW: Access Gateway
AHP: Analytical Hierarchy Process
APS: Automatic Protection Switching
ARP: Address Resolution Protocol
ASN: Access Services Network
ATM: Asynchronous Transfer Mode

B

BA: Behavior Aggregates
BC: Boundary Clock
BER: Bit Error Rate
BITS_P: Building Integrated Timing Supply_Protection
BITS_W: Building Integrated Timing Supply_Working
BMCA: Best Master Clock Algorithm
BPaaS: Business Process as a Service
BPF: Band Pass Filter
BRAS: Broadband Remote Access Server
BS: Base Station
BSC: Base Station Controller
BTS: Base Transceiver Station
BWA: Broadband Wireless Access

C

CBR: Constant Bit Rate
CBS: Committed Burst Size
CCAMP: Common Control and Measurement Plane
CDR: Call Detailed Record
CE: Customer Edge
CEP: Circuit Emulation over Packet
CESoPSN: Circuit Emulation Services over Packet Switch Network
CFM: Connectivity Fault Management
CII: Common Interworking Indicator
CIR: Committed Information Rate
CO-CS: Connection-Oriented Circuit-Switched
CRM: Customer Relationship Management
CS: Central Station

CSN: Connectivity Service Network

D

DEA: Data Envelopment Analysis
DF: Default Forwarding
DFS: Depth-First Search
DL: Downlink
DSCP: Differentiated Services Code Point
DSRC: Dedicated Short-Range Communications
DUT: Device Under Test

E

E2E: Edge-to-Edge
EBS: Excess Burst Size
ECMP: Equal-Cost Multi-Path Routing
EDGE: Enhanced Data rates for GSM Evolution
EF: Expedited Forwarding
EIR: Excess Information Rate
EPON: Ethernet Passive Optical Network
ERP: Enterprise Resource Planning
ESMC: Ethernet Synchronous Message Channel
ESP: Ethernet Switched Path

F

FEC: Forwarding Equivalence Class
FR: Frame Relay
FSM: Finite State Machine
FTP: File Transfer Protocol

G

GE: Gigabit Ethernet
GFP: Generic Framing Procedure
GFP-F: Generic Frame Procedure Framed
GFS: Google File System
GIS: Geographic Information System
GMPLS: Generalized Multiprotocol Label Switching
GPRS: General Packet Radio Service
GPS: Global Positioning System
GUI: Graphical User Interface

H

HDLC: High Level Link Control
HSDPA: High Speed Downlink Packet Access
HSPA: High Speed Packet Access

I

IaaS: Infrastructure as a Service
IMA: Inverse Multiplexing over ATM
IPTV: Internet Protocol Television
IS-IS-TE: Intermediate System to Intermediate System Extensions for Traffic Engineering
ITS: Intelligent Transportation Systems

L

LACP: Link Aggregation Control Protocol
LAG: Link Aggregation
LAN: Local Area Network
LBS: Location-Based Service
LMP: Link Management Protocol
LSP: Label Switched Path
LTE: Long Term Evolution

M

MAC: Media Access Control
MAN: Metropolitan Area Network
MBMS: Multimedia Broadcast and Multicast Service
MC-HSPA: Multi-Carrier HSPA
MDDV: Mobility-Centric Data Dissemination Algorithm for Vehicular Networks
MIMO: Multiple-Input Multiple-Output
MLPPP: Multi-Level Pre-Emptive Priority
MMS: Multimedia Messaging Service
MP2MP: Multipoint-to-Multipoint
MPLS: Multiprotocol Label Switching
MPLS-TE: Multiprotocol Label Switching Traffic Engineering
MPLS-TP: Multiprotocol Label Switching Transport Profile
MS-PW: Multi-Segment Pseudo Wires

Abbreviation Index

MSTP: Multiservice Transport Platform
MTIE: Maximum Time Interval Error

N

NE: Network Element
NLOS: Non-Line of Sight
NTP: Network Time Protocol

O

OAM: Operation, Administration and Maintenance
OC: Ordinary Clock
ODU: Optical Cannel Data Unit
OFDM: Orthogonal Frequency Division Multiplexing
OLT: Optical Line Terminal
ONU: Optical Network Unit
OS: Operating System
OSI: Open System Interconnection
OSPF-TE: Open Shortest Path First with Traffic Engineering Extensions
OTM: Optical Transport Module
OTN: Optical Transport Network

P

P2MP: Point-to-Multipoint
P2P: Peer-to-Peer
PBB: Provider Backbone Bridge
PBBN: Provider Backbone Bridge Network
PBB-TE: Provider Backbone Bridge Traffic Engineering
PBT: Provider Backbone Transport
PDH: Plesiochronous Digital Hierarchy
PDV: Packet Delay Variation
PE: Provider Edge
PHB: Per-Hop Behavior
PHP: Penultimate Hop Popping
PIR: Peak Information Rate
PLL: Phase-Locked Loop
PLSB: Provider Link State Bridging
PON: Passive Optical Network
PPP: Point-to-Point Protocol
PPS: Pulse Per Second
PRC: Primary Reference Clock

PSN: Packet Switched Network
PTN: Packet Transport Network
PTP: Precision Time Protocol
PW: Pseudo Wire
PWE3: Pseudo Wire Emulation Edge-to-Edge
PWES: Pseudo Wire Emulation Service

R

RAN: Radio Access Network
RARP: Reverse Address Resolution Protocol
REST: Representational State Transfer
RFID: Radio Frequency Identification
RNC: Radio Network Controller
ROF: Radio over Fiber
RPR: Resilient Packet Ring

S

SaaS: Software as a Service
SAToP: Structure-Agnostic TDM over Packet
SCM: Subcarrier Multiplexing
SD: Signal Degrade
SDH: Synchronous Digital Hierarchy
SF: Signal Failure
SINR: Signal to Interference plus Noise Ratio
SLA: Service Level Agreement
SMS: Short Message Service
SNCP: Subnetwork Connection Protection
SOA: Service-Oriented Architecture
SOAP: Simple Object Access Protocol
SONET: Synchronous Optical Network
SOTIS: Self-Organizing Traffic Information System
SR: Service Router
SSM: Synchronization Status Message
SS-PW: Single-Segment Pseudo Wire
STM: Synchronous Transport Module

T

TC: Transparent Clock
TDEV: Time Deviation
TDM: Time Division Multiplexing

TDMA: Time Division Multiple Access
TD-SCDMA: Time Division Synchronous Code Division Multiple Access
TE: Traffic Engineering
T-LSP: Transport Label Switched Path
TLV: Type-Length-Values
T-MPLS: Transport Multiprotocol Label Switching
ToP: Time over Packet
TPS: Tributary Protection Switching
TTL: Time-to-Live

U

UDP: User Datagram Protocol
UL: Uplink

V

VADD: Vehicle-Assisted Data Delivery
VANET: Vehicular Ad-Hoc Network
VBR: Variable Bit Rate
VC: Virtual Channel
VCC: Virtual Channel Connection
VIA: Virtual Interface Architecture
VLAN: Virtual Local Area Network
VM: Virtual Machine
VO: Virtual Organization
VoD: Video on Demand
VoIP: Voice over Internet Protocol
VP: Virtual Path
VPC: Virtual Path Connection
VPN: Virtual Private Network
VS: Virtual Section

W

WAVE: Wireless Access in Vehicular Environments
WCDMA: Wideband Code Division Multiple Access
WDM: Wavelength Division Multiplexing
Wi-Fi: Wireless Fidelity
WiMAX: Worldwide Interoperability for Microwave Access
WLAN: Wireless Local Area Network
WPAN: Wireless Personal Area Network
WTR: Wait to Restore