SOLUTION BRIEF

ENTERPRISE SOLUTION FOR DIGITAL AND ANALOG VOICE TRANSPORT ACROSS IP/MPLS

IT Organizations Can Reduce Costly TDM Leased Line Fees

Challenge

IP networks were not designed to transport bit-synchronous circuits with timing and clocking requirements end-to-end. In order to transport legacy analog and circuit-based technology across IP/MPLS networks, the critical timing aspects of these technologies need to be preserved.

Solution

Juniper Networks CTP Series Circuit to Packet Platforms provide per-circuit buffering to eliminate packet delay variation (jitter) in the IP network and a clocking tool kit that enables timing synchronization end to end across packet-based networks—delivering a variety of internal, external, and adaptive clocking options that are software selectable on a per-circuit basis.

Benefits

The CTP Series reduces costs by enabling the removal of point-to-point leased lines and the decommissioning of legacy TDM and circuit-based transport equipment. By converging all legacy traffic over one IP/MPLS network, operational cost efficiencies can be realized by managing one transport network instead of multiple legacy networks.

Enterprise organizations in both the public and private sectors are under significant pressure to modernize their networks to next-generation packet-based technologies. Yet many enterprise organizations have legacy analog-based PBX equipment, ATM networks, or other analog/TDM circuit-based networks deployed, and in some cases have made significant prior investments in these legacy technologies.

As these organizations look at the cost associated with upgrading these legacy networks and voice-switching equipment to next-generation packet-based equipment, they are also under pressure to upgrade their existing IP infrastructure, security, and enhance VPN capabilities for a more mobile and decentralized remotely located workforce—and they are looking for ways to modernize their backend infrastructure to better keep up with their competition through modern solutions like cloud computing and data center outsourcing.

Juniper Networks' enterprise network solution for digital and analog voice and circuit-based application transport across packet networks allows public and private sector enterprise organizations to cost-effectively transport these legacy traffic types across lower-cost IP/MPLS networks, allowing for the immediate elimination of costly TDM leased line fees from the IT budget. The cost savings created can then be used to phase out the legacy CPE equipment by upgrading it to modern packet-based technology, or this cost savings can be redirected towards accelerating much needed security, remote access, or cloud computing solutions.

The Challenge

Enterprise IT organizations are under constant pressure to modernize and reduce the cost of their networks. One area of potential savings for IT organizations is telecommunications services fees, which include the transport of voice, data, and video between enterprise sites. Many enterprise organizations still have leased lines, ATM networks, or TDM service lines, and their associated fees, for the purposes of transporting legacy applications between locations. For some organizations, VoIP is a great solution for eliminating legacy analog or circuit switched networks and associated access fees. However, the upgrade cost for VoIP may be prohibitive for enterprise organizations with large analog or digital PBX deployments and VoIP may not be able to eliminate all circuit switched network access fees. There are many other applications, beyond voice, whose real time deterministic nature has required the reliability of circuit based networks for transport. With the Juniper Networks CTP Series Circuit to Packet Platforms, enterprise organizations can consider eliminating all of their leased lines, ATM, and TDM transport services converging all of their analog and digital applications onto next generation IP/MPLS networks.
The Juniper CTP Series Voice Solution

Juniper’s CTP Series products enable small to large enterprises to converge their digital and analog voice traffic over their IP network. This enables significant cost reduction by eliminating point-to-point leased lines and allows enterprises to remove legacy TDM or ATM networks. The CTP Series protects your investment in your current phone systems without requiring any software or hardware upgrades. There is no need to change your operations and maintenance procedures on your phone system and there are no requirements for training on new signaling protocols or adding equipment to support H.323, H.248, or SIP, as you would need for a VoIP solution. All your current phone system features are still supported and are transparent to the CTP Series. The CTP Series keeps the transition of your voice network to your IP network a simple low-risk upgrade with no business disruption.

The CTP Series products use circuit emulation over IP to create pseudowires over the IP network. The digital and analog voice is carried transparently across the IP network. There is no interpretation of signaling or digits dialed. The CTP Series basically creates a wire between the endpoints over the IP network. All features and functions work as they would if the two endpoints were directly connected to each other. Packet delay variation (jitter) caused by queuing delays in the IP networks is removed and many timing and clocking options are supported to ensure end-to-end synchronization across the IP network. The CTP Series is a proven product that has been deployed in networks for many years. Where applicable, the CTP Series supports the IETF RFC-based encapsulation methods, SAToP and CESoPSN. The CTP Series also supports serial and TDM data to be transported across IP/MPLS networks. This capability enables enterprise IT organizations to converge a broad range of circuit-based applications across their IP access network along with their legacy analog and digital voice traffic, resulting in the elimination of most, if not all, leased line services.

Features and Benefits

Digital Voice

Large and small enterprise organizations that have legacy digital PBX switches are paying expensive tie-line fees to link these switches together, when they could be utilizing lower-cost broadband IP access networks. Enterprise organizations that convert their digital TDM traffic can take advantage of the broadband IP access pipes. By doing so, these organizations can leverage the cost savings they receive to invest in other areas of their network—be it security, data center expansion, increasing core network bandwidth, investing in new applications, or even upgrading their legacy digital PBXs to packet solutions. The CTP Series digital voice capabilities enable enterprise organizations to reliably transport PBX traffic across the Internet, so that they can redirect their tie-line cost savings to more critical areas of their business. CTP Series provides the following support for the transport of digital voice over IP/MPLS networks: Full T1/E1, Fractional T1/E1, and DS0 bundling support.

Full T1/E1—The CTP Series supports both T1 and E1 interfaces for interconnecting PBX voice switches. The T1/E1 interfaces support full T1/E1 including framing end to end. The CTP Series transports all signaling protocols transparently. This mode would also support ISDN PRI 23B + D channels. The encapsulation options are CTP, SAToP, CESoPSN, and VCOMP.

Fractional T1/E1—The CTP Series supports fractional T1 and E1 interfaces for interconnecting PBX voice switches. Any number of DS0 channels 1-24 (T1) or 1-31 (E1) are supported. Framing can be transported end to end or terminated locally. The encapsulation options are CTP, CESoPSN, and VCOMP.

DS0 Bundles—The CTP Series supports DS0 bundling and grooming on T1 and E1 interfaces for interconnecting PBX voice switches. Any range of DS0 channels can be a bundle and any number of bundles 1-24 (T1) and 1-31 (E1) are supported. The encapsulation options are CESoPSN and VCOMP.

VOICE ENCAPSULATION TYPES

- **CTP**—This encapsulation supports full T1/E1 and one group of fractional T1/E1. CTP encapsulation minimizes overhead by adding only the Layer 3 IP header to the packet payload.
- **SAToP**—This encapsulation is for Structure-Agnostic TDM over IP and supports full T1/E1 circuits only. This is based on the IETF RFC 4553.
- **CESoPSN**—This encapsulation is for Circuit Emulation Services over a Packet-Switched Network and supports full, fractional, and DS0 bundling on T1/E1 circuits. This is based on the IETF RFC 5086.
- **VCOMP**—This encapsulation is for Voice Compression. This is optional for T1/E1 circuits and is required for all analog voice circuits, 4WE&M, 2WFXS, and 2WFXO. Using this bundle type, one or more voice channels going between the same IP endpoints can be aggregated into a single IP flow.
Analog Voice

Enterprise organizations that use analog trunk lines between PBXs can eliminate point-to-point analog connections or TDM network transport systems by using the CTP Series to transport the analog trunks over the IP network. Analog extensions to remote locations or to local central office locations can also be transported over the IP network to eliminate leased line fees. The ability to transport analog extensions over the IP network enables enterprise organizations to easily add extensions to any location and use the resources of a PBX at a different site. The CTP Series provides the following capabilities for the support of analog voice transport across IP/MPLS networks: 4WE&M Trunks, 2WFXS and 2WFXO.

4WE&M Trunks—The CTP Series supports 4WE&M analog trunks for interconnecting PBX voice switches. The 8-port 4WE&M module supports signaling types I, II, and V. The only encapsulation option supported is VCOMP. 4WE&M channels from a single module or multiple modules can be combined into a single packet to minimize the IP overhead.

2WFXS and 2WFXO Extensions—The CTP Series supports both 2WFXS and 2WFXO modules for extending analog service from the PBX voice switch to remote locations for remote phone extensions or for accessing Central Office (CO) dial-tone or other services. The 24-port 2WFXS and 12-port 2WFXO modules support loop and ground start signaling. The only encapsulation option supported is VCOMP. 2WFXS or 2WFXO channels from a single module or multiple modules can be combined into a single packet to minimize the IP overhead. Normally a 2WFXS would connect to a 2WFXO across the IP network. However, it is also possible for a 2WFXS to connect to another 2WFXS using the Private Line Automatic Ringdown (PLAR) signaling.

4WTO—The CTP Series supports 4WTO analog interfaces for 4Khz / 64 Kbps analog modem and radio connections over the IP network. No signaling is supported. The 4WTO option is available on all CTP Series platforms. Voice compression is not supported or required on the 4WTO connections. The only encapsulation option supported is CTP.

Digital and Analog Voice Interworking

Interworking between digital and analog voice channels is important when remote sites may use analog trunks. However, the central site voice switch location uses digital T1/E1 connections. The ability to aggregate the analog sites into higher-density digital interfaces on the central voice switch is required in many networks. Also, the ability to connect U.S. sites to international locations requires the ability to do companding conversion between T1 and E1 voice interfaces.

T1 and E1 DS0 channels can connect to the analog 4WE&M, 2WFXS, or 2WFXO channels across the IP network. The encapsulation mode must be VCOMP. A-law to mu-law conversion between E1 and T1 is also possible as long as the VCOMP encapsulation method is used.

Voice Compression and Echo Cancellation

The CTP Series voice compression module is supported in the CTP2024 and CTP2056. Voice compression can significantly reduce the amount of IP bandwidth required for each voice call. Combined with silence suppression, IP bandwidth requirements are reduced further. Additional bandwidth savings are realized when multiple T1/E1 DS0s—or channels that are destined to the same location—are bundled together into the same packet, which reduces IP header bandwidth. In VoIP, each voice call is a packet with an IP header. However, with circuit emulation the CTP Series can bundle multiple voice calls into a single packet if the calls have the same endpoints.
Each DS0 or analog channel can be configured differently and independently of any other DS0 or channel configuration. The voice compression algorithms supported include:

- 2.4K MELP
- 8K G.729
- 16K G.728
- 16K G.726
- 32K G.726
- 64K G.711

The voice compression module has no physical interface ports—it is a server module in the CTP Series system. Any T1 or E1 DS0 or analog 4WE&M, 2WFXS, and 2WFXO channel can use any or all of the voice compression module features. The additional software selectable features available on the voice compression module include:

- Echo cancellation with 32 ms end path delay
- Silence suppression
- T.38 Fax detection and demodulation
- Modern detection and bypass
- Tone relay for DTMF and MF tones

### Voice Applications

**SS7 Transport**—Transport point-to-point T1 or E1 SS7 links over the IP/MPLS network. The CTP Series provides T1 and E1 circuit emulation over IP. No hardware or software changes are required on the voice switch.

**PBX Interconnect**—Provides T1, E1, and analog 4WE&M trunks for PBX interconnect over the IP/MPLS network using circuit emulation over IP. No hardware or software changes are required on the PBX voice switch.

**ISDN PRI**—Transport ISDN PRI T1 or E1 between the same IP endpoints. The CTP Series transparently encapsulates the D-Channel end to end without interpretation of signaling states within the signaling channel. All B-Channels can be aggregated into one bundle or multiple bundles with or without compression applied to each bundle.

**PBX Extension**—Provides two-wire analog PBX to phone extensions over IP and two-wire PBX to PBX/CO connections over the IP/MPLS. The CTP Series provides 2WFXO to 2WFXS circuit emulation over IP to enable analog PBX extensions over the IP network. Analog phone extensions can be remotely located over the IP network. Remote two-wire analog access over the IP network to the service provider CO voice switch is also an option. Both loop start and ground start signaling are supported. PLAR is also supported using 2WFXS to 2WFXS circuit emulation.

![Figure 1: Voice network applications](image-url)
Solution Components
Juniper Networks' digital and analog voice transport solution across IP/MPLS networks delivers the capabilities outlined above with the following solution components:

1. CTP Series products for full or fractional T1/E1
   a. CTP1000 line and CTP2000 line
2. CTP Series products for DS0 grooming and bundling T1/E1
   a. CTP2000 line
   b. CTP2000-IM-8P-T1E1 Module
3. CTP Series products for Voice Compression and Echo Cancellation
   a. CTP2024 and CTP2056
   b. CTP2000-COMPRESSION Module
4. CTP Series products for Analog Voice
   a. CTP2024 and CTP2056
   b. CTP2000-COMPRESSION Module
   c. CTP2000-IM-4WEM Module
   d. 4WEM-RTM Module
   e. CTP2000-IM-2WFXS Module
   f. 2WFXS-RTM
   g. CTP2000-IM-2WFXO Module
   h. 2WFXO-RTM

Summary: Transport Voice over Your IP Network
Using Circuit Emulation over IP

Enterprise organizations both large and small are under pressure to reduce network costs by converging all applications onto one IP/MPLS network. Voice applications are a critical component to any business. However, significant savings can be realized by eliminating legacy TDM and ATM networks deployed today to carry voice calls. Using the Juniper CTP Series products to transport voice calls over IP with circuit emulation does not require the voice switch software and/or hardware upgrades and expertise that are required to implement VoIP using SIP, H.323, or H.248. Circuit emulation over IP using the CTP Series is transparent to all signaling protocols and features. It is simple to implement and enables organizations to realize the immediate cost savings associated with transporting their voice calls over IP by eliminating point-to-point leased lines from their IT budget.

The CTP Series is very feature rich with great interface flexibility for serial data circuit requirements. However, the CTP Series voice interfaces and features enable a comprehensive voice circuit emulation solution. All CTP Series interfaces and features are supported in the CTPView for Web-based management of all the CTP Series devices in the network.

Next Steps
For more information, please refer to the Juniper Networks website at www.juniper.net. Please contact the Juniper Sales team to learn how Juniper can help you in your TDM or analog voice transition to your IP/MPLS network.

About Juniper Networks
Juniper Networks, Inc. is the leader in high-performance networking. Juniper offers a high-performance network infrastructure that creates a responsive and trusted environment for accelerating the deployment of services and applications over a single network. This fuels high-performance businesses. Additional information can be found at www.juniper.net.