White Paper

Software-Defined Networking and Network Programmability: Use Cases for Defense and Intelligence Communities

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Abstract

This paper examines how recent advances in software-defined networking (SDN) and network programmability can be used to simplify operations, enhance agility, and meet new mission requirements within U.S. Department of Defense (DoD) and intelligence community networks. SDN and network programmability have emerged to address trends in IT by providing greater automation and orchestration of the network fabric, and by allowing dynamic, application-led configuration of networks and services. To deliver these requirements, networks must be open, programmable, and application aware. Networks must evolve to meet these emerging trends without compromising their current resiliency, service richness, or security, and without disrupting previous organizational investments.

This paper demonstrates, by means of use cases, how current networking issues within federal agencies, such as the DoD and intelligence community, may be solved through SDN and network programmability.

Problem Statement

Introduction

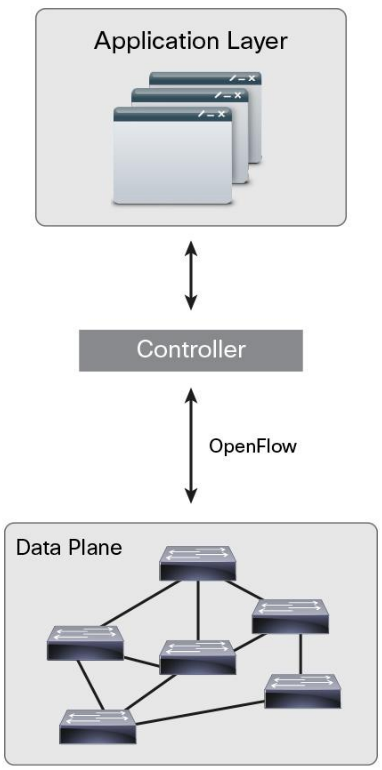
Traditionally, network architectures within corporate and government networks use network devices that combine control plane and data plane functions in a single device, typically a router or switch. The control plane is an element of a router or switch that determines how one individual device within a network interacts with its neighbors. Examples of control plane protocols are routing protocols, such as Open Shortest Path First (OSPF), Border Gateway Protocol (BGP), and Spanning Tree Protocol (STP). These protocols determine the optimal port or interface to forward packets (that is, the data plane). While these control plane protocols scale very well, and provide a high level of network resiliency, they pose limitations. For example, routing protocols may only be able to determine the best path through a network based on static metrics such as interface bandwidth or hop count. Likewise, control plane protocols do not typically have any visibility into the applications running over the network, or how the network may be affecting application performance.

Data plane functionality includes features such as quality of service (QoS), encryption, Network Address Translation (NAT), and access control lists (ACLs). These features directly affect how a packet is forwarded, including being dropped. However, many of these features are static in nature and determined by the fixed configuration of the network device. There is typically no mechanism to modify the configuration of these features based on the dynamic conditions of the network or its applications. Finally, configuration of these features is typically done on a device-by-device basis, greatly limiting the scalability of applying the required functionality.

SDN and Network Programming Models

The Open Network Foundation (ONF) defines SDN as a decoupling of the control plane and the data plane. Traditional network devices have an integrated control plane and data plane. In this classic SDN architecture, however, network intelligence is logically centralized in a controller, and there is a physical separation between control plane and the data plane. OpenFlow is the protocol that specifies the interactions between the control plane running in the controller and the infrastructure (Figure 1). The intent of this architecture is agility, automation, and a decrease in overall costs of the network.

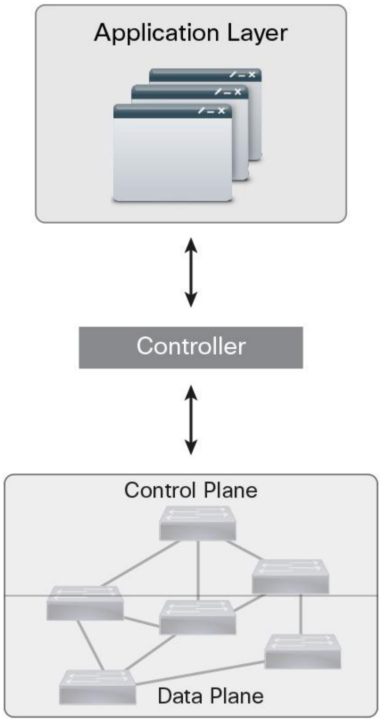
Figure 1. ONF Model for SDN



While the classic SDN approach may hold promise for some customer segments and use cases, it may not be the best fit for others, and is not without its limitations. This complete separation of the entire control plane from the data plane in a classic SDN model, removing it from the network to a centralized controller, is a radical departure from the design and operation of most networks today. The classic SDN approach to network architecture introduces potential challenges in availability, performance, scale, and security. When all control plane functionality is moved to a central controller, the controller must be highly available. The performance should meet the rapid requests and extreme load conditions on the network. The controller must be able to scale to very large networks, with thousands of nodes. Finally, it must be highly secure, with only approved applications able to modify or change the underlying network.

While the classic SDN model takes advantage of network programmability, SDN is not a requirement for programmability. Network devices can be exposed to the application layer through application programming interfaces (APIs). This hybrid approach to network programmability and SDN deployments can take advantage of hardware intelligence as well as existing feature sets within the network operating system. This can be accomplished without a complete decoupling of the control plane and the data plane (Figure 2). Moreover, in a classic SDN model, one must forgo the features of the native operation system, as those capabilities must be re-created in a controller. The hybrid SDN model allows many of the benefits of a classic SDN model with a centralized controller, while still allowing users to benefit from the existing network capabilities many have come to rely upon. In many cases, the network operating system can simply be upgraded, rather than a complete upgrade of the network infrastructure. This allows an evolutionary approach to network programmability, as opposed to the revolutionary approach of a classic SDN model. This evolutionary method allows users to choose which applications and features are migrated out of the network device and onto a controller or application server. Finally, a hybrid model allows users to use their existing fault management or network management systems. The hybrid SDN model is able to minimize some of the potential challenges that face a classic SDN approach to network architecture.

Figure 2. Hybrid SDN Model

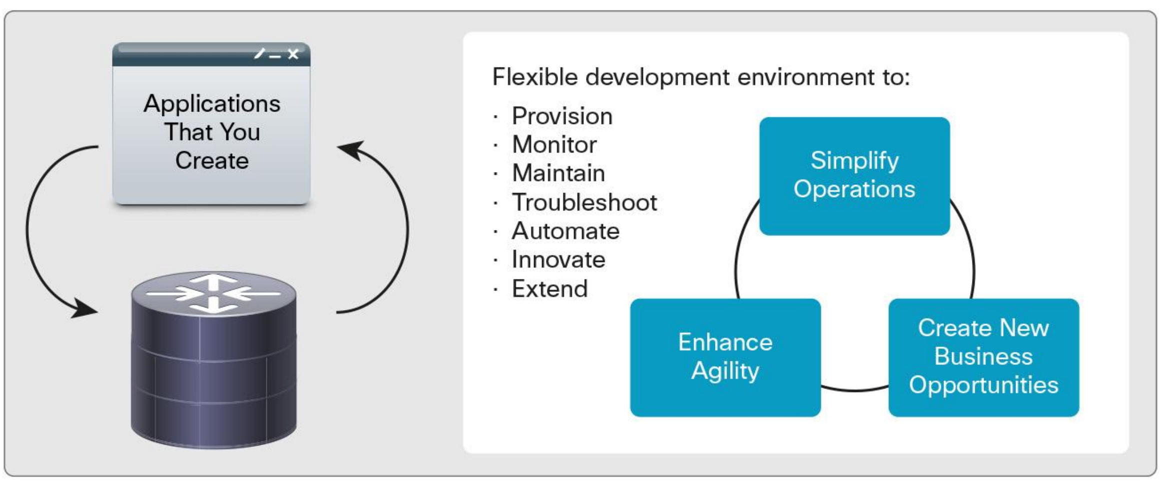


Introduction of Cisco ONE and onePK

Overview

Cisco® Open Network Environment (ONE) is a comprehensive solution that allows networks to become more open, programmable, and application-aware. It is a portfolio of complementary technologies, which includes platform APIs, controller and agent technologies, and overlay network technologies. The platform APIs are a key element within the Cisco ONE strategy and part of a Software Development Kit (SDK) called ONE Platform Kit (onePK). onePK is a comprehensive set of platform APIs, providing full-duplex programmatic access to Cisco devices. It is a toolkit, which allows developers to build custom applications that interact with Cisco network devices in ways that have never before been possible (Figure 3). onePK enables developers to build automation and orchestration into the network. The benefits of onePK can be achieved through a software update to existing routers and switches, providing investment protection and an evolutionary path to a software-defined network.

Figure 3. The onePK Toolkit



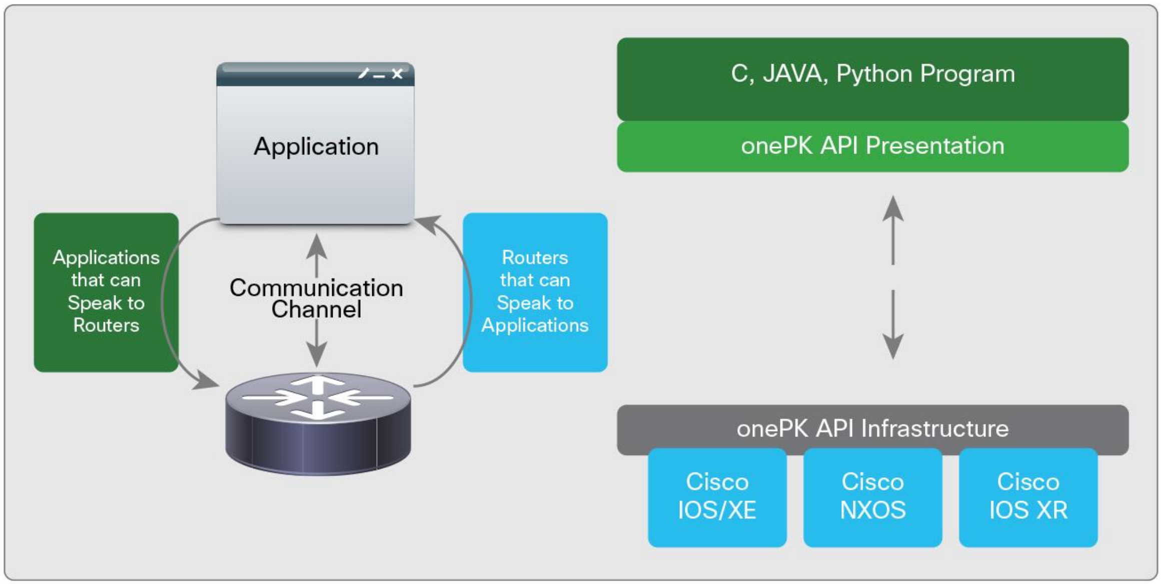
onePK simplifies operations by allowing automation of functions, thereby creating efficiency. It provides a framework for innovation, providing access to network devices, which allow new services to be created. Finally, onePK increases agility by allowing new features and extending device functionality, allowing you to move faster. The result is a rich interaction between applications and the underlying infrastructure.

The capabilities of onePK represent a hybrid SDN model. However, these capabilities may be used to facilitate the construction of a classic SDN-based system. For example, onePK can be used to implement OpenFlow agents, or be used by a network controller, each of which represents typical elements in a classic SDN architecture.

Cisco onePK Architecture

Cisco ONE Platform Kit is a development toolkit for major Cisco platforms. onePK is a highly flexible development environment, providing a presentation API which supports C, Java, and Python languages (Figure 4).

Figure 4. The onePK Architecture



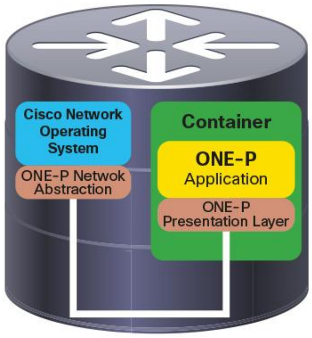
Central to the onePK architecture is a single set of API libraries for all Cisco major software platforms: Cisco IOS® and Cisco IOS XE Software, Cisco IOS XR Software, and Cisco NX-OS Software. This infrastructure layer provides a level of abstraction for platform-specific implementations. This helps ensure that the application developer does not have to be concerned with the specifics of the network or underlying infrastructure, greatly increasing the scope and scale of the application. Developers can use the same APIs across the entire network, even when devices are running different network operating systems. It is this abstraction of the infrastructure that greatly enhances the agility and time to market for the application developer.

Between the presentation and infrastructure layer is a communication channel, which provides a typical client/server model between the application and the network device.

Deployment Models

A deployment model describes where the onePK applications will run. The applications can run on Cisco devices, such as routers and switches, processing blades, or on an external server. Developers and network operators can decide which model fits their mission and the requirements of the applications they are running. Each model is source code portable. Discussed below are the three main deployment models.

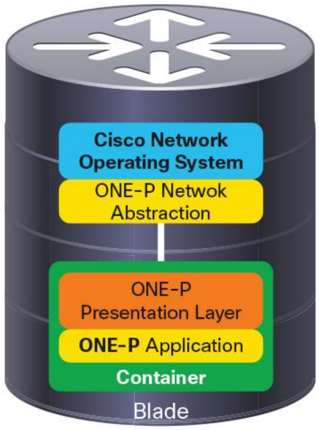
Figure 5. Process Hosting Model



Process Hosting

In the process hosting model, all onePK applications are run in a process container directly on a Cisco network device such as a switch or router (Figure 5). The supported containers are Linux containers (LXC) and kernel-based virtual machine (KVM). The process hosting model helps enable application developers to install and run applications within a network element. This provides low-latency communication and a single footprint for applications. Containerization helps ensure proper resource allocation between each application.

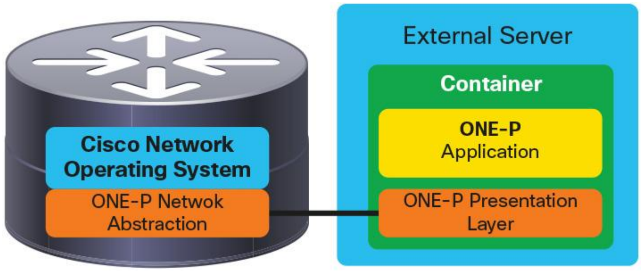
Figure 6 Blade Hosting Model



Blade Hosting

In the blade hosting model, the applications are run on a process blade (Figure 6). This helps enable an application to run close to the control and data planes. This model means applications can have dedicated resources to perform their tasks. The blade hosting model also provides a high degree of isolation.

Figure 7. End-Node Hosting Model



End-Node Hosting

In the end-node hosting model, applications are run on an external server connected to the network (Figure 7). This method gives users the flexibility to use the platform of their choice. The platforms can range from multicore Linux or Windows servers to small mobile devices based on Android or iOS operating systems. These platforms can run the applications in either containers or on the host operating systems, and they provide the highest degree of isolation.

Security

The security model for onePK involves multiple security measures, which together help ensure that applications are run highly securely, without introducing vulnerabilities to the network. Central to the security model, however, is the network administrator, who plays a critical role in helping to ensure the security of the entire network. The administrator authorizes which users can run which applications, and can enable onePK services for the entire network or for specific users and devices. No applications can be run on network elements without the active consent of the network administrator.

Authentication, authorization, and accounting (AAA) can also be enabled on the network devices. This helps ensure that only authenticated and authorized users can run the application. Finally, the operations performed by the applications on the network element may be logged through process accounting. This provides auditing of all applications developed using onePK.

When applications are deployed to run in a containerized manner, the application developer has two options: trusted containers and untrusted containers. Trusted containers are provided as part of the Cisco infrastructure to run on network elements or processing blades. Trusted containers allow applications to use a private trusted communications channel. Untrusted containers allow applications to run within commodity containers on servers. Programmed applications are required to be authenticated and communicate using highly secure channels such as Transport Layer Security (TLS). The network administrator can enforce that the application running from an untrusted container has valid credentials to access the network element. Cisco recommends that all applications be run in a container and that network administrators enforce resource limits.

In addition to deploying and running applications in a highly secure manner, the onePK security model provides security measures for authenticity of the applications themselves. For example, the administrator can work to ensure that a chain of trust exists between the developer of the application and the device it is installed upon. A signature can be created by the publisher of the application package and verified during the time of install. It is up to the network administrator to decide which application publishers are trusted. Finally, the internal interprocess communication (IPC) mechanism performs strong type check and prevents buffer overrun conditions. Any IPC access that originates from an untrusted host is authenticated and encrypted, using TLS-based security.

Use Cases

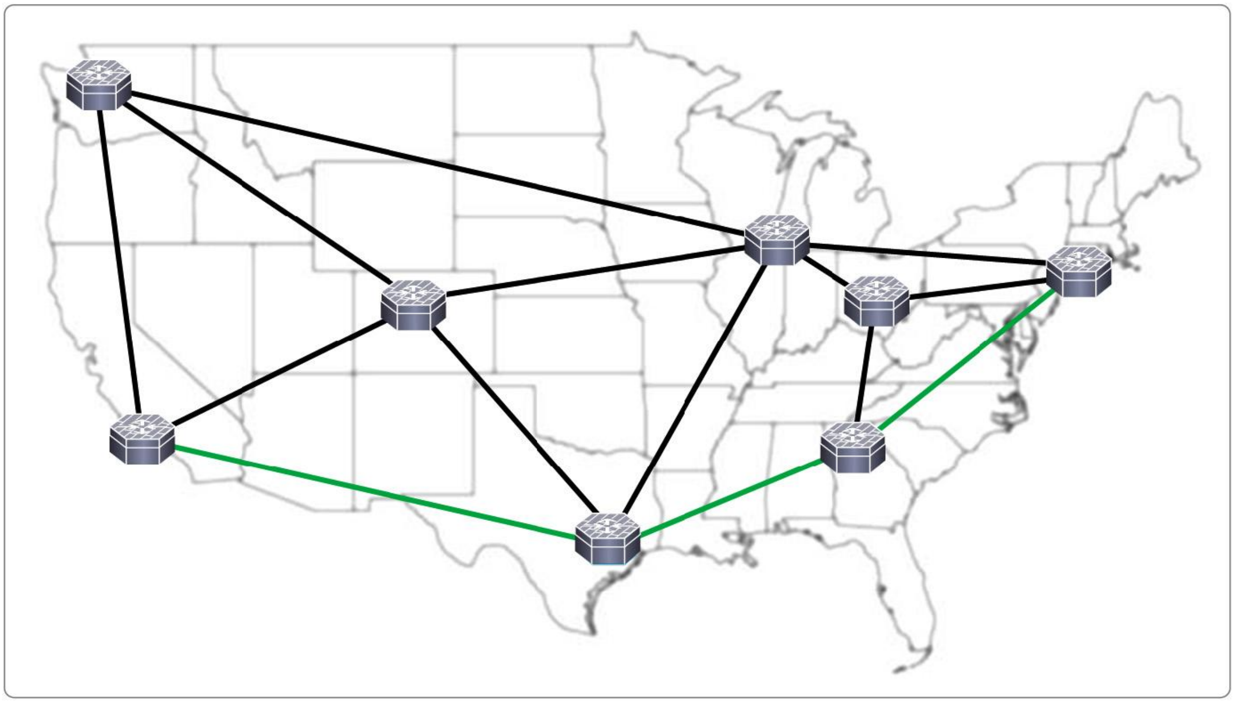
The power of SDN and network programmability is best illustrated through use cases. While there are a very large number of potential use cases, here we focus on issues that face the DoD and intelligence communities in particular, although enterprise companies may find them of value as well. Finally, the use cases explore and highlight what is both reasonable and feasible, providing detailed examples and working code. While the code snippets shown are in the C programming language, there is also feature parity with the Java and Python languages. Please note that this is sample code and subject to change before the General Availability release of the API.

Use Case 1: Latency-Based Routing

The goal of a routing protocol, such as BGP, OSPF, Intermediate System-to-Intermediate System (IS-IS), or Enhanced Interior Gateway Routing Protocol (EIGRP), is to find an optimal path to a given destination within the network. The metric used to determine the optimal path, however, is often fixed and static, and does not vary based on the load placed on the network or the utilization of its links. For example, in the case of OSPF or IS-IS, the cost used to calculate the best path is the interface bandwidth, which is typically fixed. In the case of BGP, while there are metrics that can be set by the administrator, the protocol, again, is relatively static in nature.

In an era where cloud computing, big data, and surging video traffic are becoming the norm, networks are reaching a saturation point, and unable to keep up with demand. In some cases, a path through a network may become oversubscribed, while an alternate path may be underutilized, or not used at all. Because network traffic tends to be “bursty” in nature, utilization of a set of links may fluctuate greatly over time. Traditional routing protocols may not be able to react efficiently to the dynamic nature of today’s networks. Figure 8 shows a path through a network, from Los Angeles to New York. This path is fixed - the lowest cost is based upon the bandwidth of each interface. The network is unable to take into account the load on each of its links.

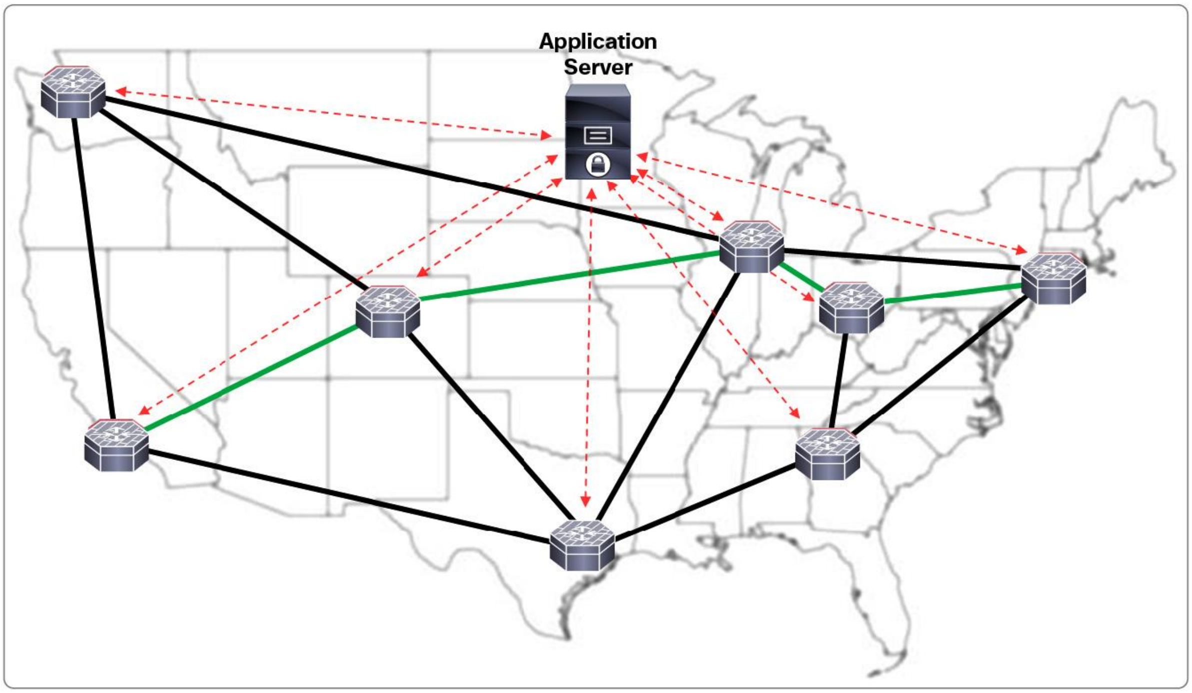
Figure 8. Network Path Based on OSPF Metrics



What is required is a mechanism that can dynamically react to the latency and utilization of the network in real-time. For example, as links on a router get congested, the latency for traffic along that path increases, caused by queuing delays in interface egress buffers. Thus, latency can be used as a metric for each path within a network, which allows for real-time response to oversubscribed links in a network. Applications that require minimal delay and jitter, such as mission-critical real-time traffic or voice traffic, could take a low-latency path through the network, while all other traffic (such as bulk and transactional data) could take a path derived from the underlying routing protocol. This allows real-time application traffic to be directed over links with the lowest latency to the destination.

As shown in Figure 9, a latency-based metric follows a different path through the network: one that has the least latency or that is currently experiencing less congestion. This allows real-time application traffic to follow a path through the network that provides the optimal response times.

Figure 9. Network Path Based on Low-Latency Metrics



To help enable this solution, each node in the network discovers the topology of the network, and the set of links interconnecting each device. Next, each node in the network samples the latency to each of its connected neighbors, and the result is sent to an application server. That latency data is fed into Dijkstra's algorithm to determine the lowest cost path, where cost in this case is the latency measurements. From the output of Dijkstra's algorithm, a new set of routes can be applied to the network, routes that provide the shortest path tree based on latency in the network. These routes are "overlaid" on top of the routes derived from the existing routing protocol.

Figure 10 show sample code of the creation of a local application route, which can then be applied to each node in the network through onePK.

Figure 10. Sample Code - Creation of a Local Application Route

```
/*
 * Create an L3 unicast route object.
 */
routing_l3_ucast_route_t *
Create_Route(char *prefix, unsigned int mask, char *nh_addr, network_interface_t *nh_if, uint32_t admin_dist, uint32_t metric) {

   onep_prefix_t *onep_prefix = NULL;
   routing_l3_ucast_next_hop_t *nh = NULL;
   routing_l3_ucast_route_t *route = NULL;
   onep_status_t rc = ONEP_OK;

   // Create new L3 unicast route from prefix
   rc = onep_routing_l3_ucast_route_new(onep_prefix, &route);
   if (rc != ONEP_OK)
      return (NULL);

   // Set the metric for the next hop, append the next hop to the route.
   rc = onep_routing_l3_ucast_next_hop_set_metric(nh, metric);
   rc = onep_routing_l3_ucast_route_append_next_hop(route, nh);
   if (rc != ONEP_OK)
      return (NULL);

   // Set the admin distance for the route.
   rc = onep_routing_l3_ucast_route_set_admin_distance(route, admin_dist);
   if (rc != ONEP_OK)
      return (NULL);

   return (route);
}
```

Routes that are applied to a network node are known as application routes, as they are derived and installed by applications. The application server installs the set of application routes on each node in the network. Figure 11 shows the routing table from a node in the network with a set of application routes.

Figure 11. Application Routes (Truncated Output)

```
LAX-GW# show  ip route
Codes: L - local, C - connected, S - static, R - RIP, M - mobile, B - BGP
     a - application route
     + - replicated route, % - next hop override

10.0.0.0/8 is variably subnetted, 4 subnets, 2 masks
a       10.100.2.0/24 [10/0] via 192.168.1.2, 00:01:18, GigabitEthernet3/0
a       10.100.3.0/24 [10/0] via 192.168.1.6, 00:01:03, GigabitEthernet2/0
a       10.100.4.0/24 [10/0] via 192.168.1.10, 00:01:16, GigabitEthernet1/0
a       10.100.5.0/24 [10/0] via 192.168.1.6, 00:01:17, GigabitEthernet2/0
a       10.100.6.0/24 [10/0] via 192.168.1.6, 00:00:04, GigabitEthernet2/0
a       10.100.7.0/24 [10/0] via 192.168.1.10, 00:00:33, GigabitEthernet1/0
a       10.100.8.0/24 [10/0] via 192.168.1.6, 00:01:04, GigabitEthernet2/0
```

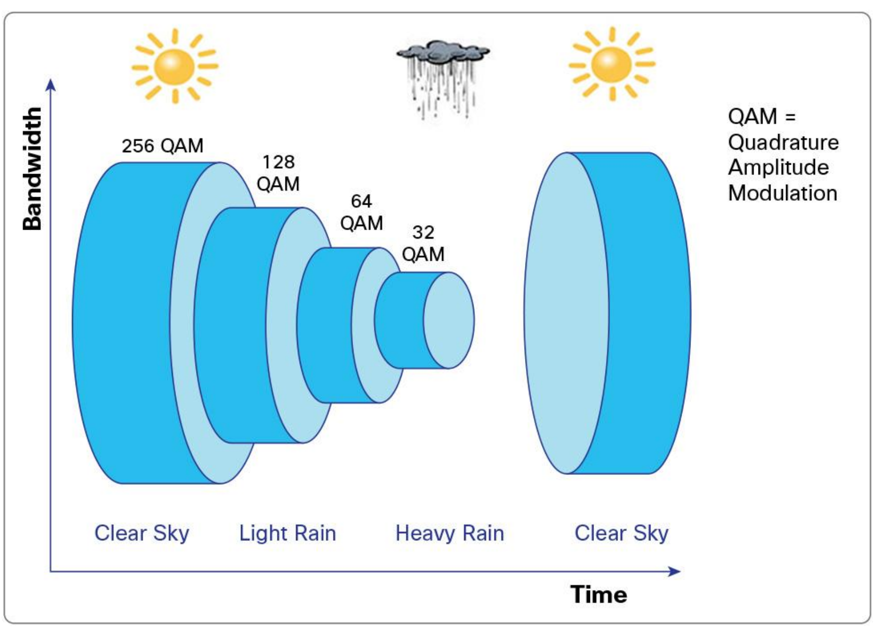
This example provides a solution for low-latency routing, by overlaying a low-latency path through the network over an existing routed path. Route table stability can be increased by modifying the latency data collection and averaging mechanisms, and by implementing route dampening to minimize route flapping and prevent sustained route oscillations. This solution can provide a routing algorithm that is able to dynamically react to the delay and utilization of the network in real time, providing low-latency paths through a network for mission-critical application traffic. However, great care must be taken while implementing such a solution. This example of latency-based routing does not provide the loop-avoidance and black-hole avoidance mechanisms of mature routing protocols. Moreover, centralization of the control plane can lead to scalability issues as well as transient micro-loops. Finally, this solution is not intended as a replacement for existing routing protocols, as they provide the highest level of stability and scalability. This example does, however, demonstrate the potential capabilities available in a hybrid SDN model, highlighting the augmentation of the control plane where a subset of control plane functionality is placed in a centralized application server.

Use Case 2: Dynamic QoS

Quality of service (QoS) refers to the ability to prioritize some applications, users, or data flows, at the detriment of other data. It is used to help guarantee a certain level of performance to an application. For example, voice over IP (VoIP) traffic may be prioritized and provided a fixed guarantee for delay, jitter, or bandwidth. However, such guarantees are statically assigned to a port or interface on a router or switch, which has a fixed port speed. For example, on a 100 Mbps Fast Ethernet interface, there may be a fixed 10 Mbps guarantee for voice traffic.

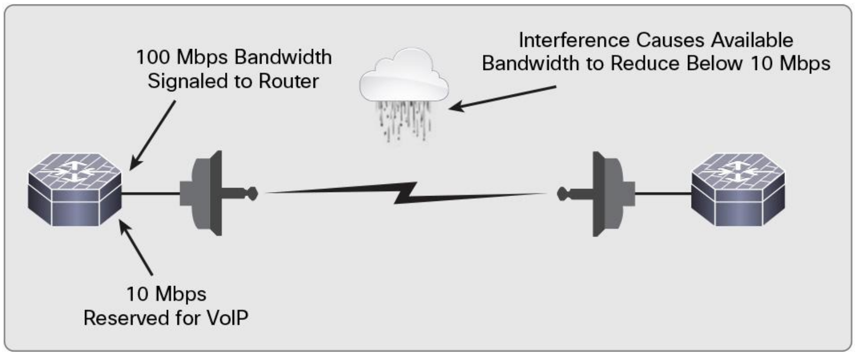
There are cases, however, where the bandwidth on an interface is not fixed, but varies over time. Traditional radios or modems provide fixed modulation, and the bandwidth on the connected interface is also fixed. Newer wireless or microwave radios and satellite modems, however, may provide adaptive modulation and coding mechanism. With this technology, changes due to fluctuating weather conditions, attenuation due to obstacles, or limited line-of-site operations may cause signal degradation and inferior signal quality, resulting in a reduction in the available bandwidth over the link (Figure 12).

Figure 12. Impact of Signal Strength on Bandwidth



This change in bandwidth is signaled to the connected router or switch, which in turn changes its interface bandwidth. However, it is possible that the interface bandwidth may be decreased below the reserved bandwidth for a traffic class in the QoS policy. For example, if 10 Mbps of bandwidth were reserved for VoIP traffic, it may be possible for the interface bandwidth to drop below this level (Figure 13). The result is inconsistent behavior, often resulting in the removal of the QoS policy from the interface altogether.

Figure 13. Dynamic Reduction in Interface Bandwidth



What is required is the ability to actively monitor the signaled interface bandwidth, and modify the QoS parameters based on the current bandwidth available over that interface.

The onePK toolkit brings that ability, through programmatic interfaces to network devices. Through the onePK API, the interface bandwidth can be monitored, and a different QoS policy can be applied to an interface in real time, based on the thresholds determined by the mission requirements. This feedback loop of monitoring the interface and applying a QoS policy is done purely through software and without user intervention.

Figure 14 shows an example of this. Here, the available bandwidth over the wireless link drops to 5 Mbps, which is signaled to the router interface. The onePK process on the router detects this change in bandwidth, and notifies the onePK process running on a server. The server, based upon user-defined thresholds for interface bandwidth, sends back a new QoS policy, which is applied to the router interface connected to the wireless modem. While an external server is shown here, any of the three hosting models could be applied to this scenario.

Figure 14. Adaptive QoS Policy Applied to Router

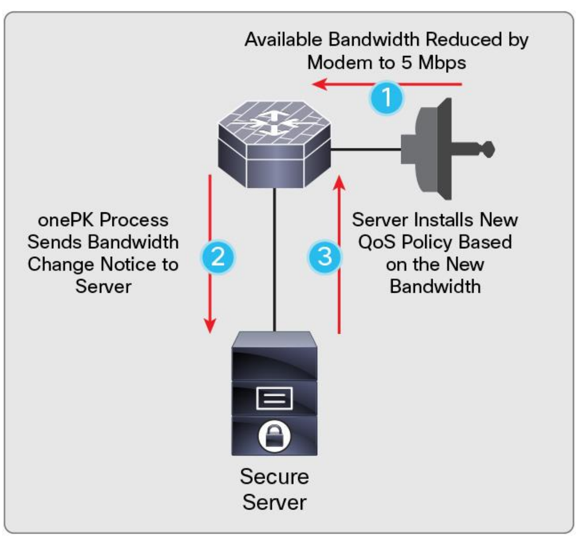


Figure 15 shows sample code of a QoS policy being applied to a router interface. An interface target is first created, and then a predefined QoS policy is applied to that target.

Figure 15. Sample Code - Applying a QoS Policy to an Interface

```
/**
 * Apply the QoS Policy to an Interface
 */
onep_status_t  Apply_QoS_Policy(struct Interface_Info *Router_Interface_Info, struct QoS_Policy *QoS_Policy) {

   onep_status_t rc = ONEP_OK;

   /*
    * Create Target
    */
   rc = onep_policy_create_interface_target(Router_Interface_Info->interface,
                ONEP_TARGET_LOCATION_HARDWARE_DEFINED_OUTPUT,  &QoS_Policy->int_target);
   if (rc != ONEP_OK) {
      fprintf(stderr, "Cannot create interface target. Error: %s\n", onep_strerror(rc));
      return ONEP_FAIL;
   }

   /*
    * Apply the QoS Policy to Target
    */
   rc = onep_policy_apply_to_target(QoS_Policy->policy_map, QoS_Policy->int_target);
   if (rc != ONEP_OK) {
      fprintf(stderr, "Cannot apply QoS Policy to target.  Error: %s\n", onep_strerror(rc));
      return ONEP_FAIL;
   }

   return ONEP_OK;
}
```

This use case demonstrates the value of onePK and network programmability by showing how the network can react to dynamic external conditions and modify its state based on those conditions. This use case, in particular, shows how network policy is no longer static, but is able to react to the current conditions and state of the network.

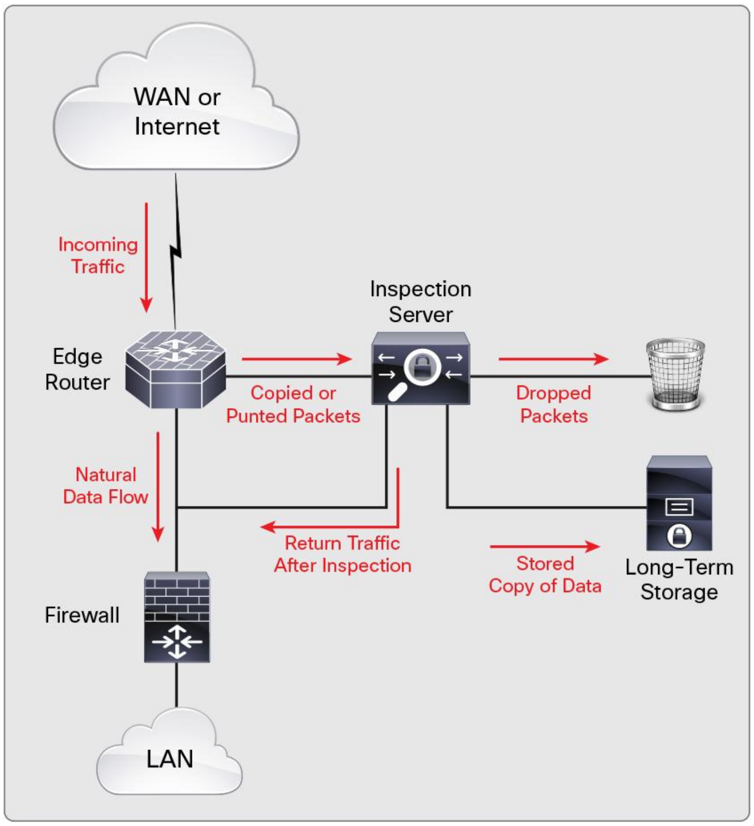
Use Case 3: Diversion Networking

Cyber security threats have evolved over the past decade. Attackers are increasingly improving their approaches with greater funding, plus attacks are increasingly innovative. Defense tools must also change and adapt to this evolving landscape. While intrusion detection and malware detection systems are essential elements of an overall network security strategy, it is not enough to modify policies or controls. The security paradigm must be focused on the threats themselves, and have the ability to protect against both the sophistication of the attacks as well as against the sheer number of new attack vectors.

The new network security model must be able to detect threats and remove malicious flows in the network in real time. However, with the migration to high-bandwidth links such as 10 Gb and eventually 100 Gb in the LAN and even the WAN or to the Internet, traditional security devices are unable to process this amount of data. What is needed is the ability to divert a subset of the overall network traffic off the naturally routed or switched path to a device that can both offload the processing of the diverted traffic and provide a deeper level of inspection or analytics.

Figure 16 shows this model. A subset of the incoming traffic from the WAN or the Internet is diverted to an inspection server. This diversion could be copies of the packet or the original packet itself. The subset of the traffic that is diverted could be based upon source address, destination address, protocol, port number, TCP flags or even a pattern or signature within the packet payload itself. Furthermore, the decision to divert traffic could be automated and based upon predefined policies or triggers.

Figure 16. Diversion Networking



In the case of a copy of the data traffic, the original traffic flow still proceeds via its original path and a copy of the packet is sent to the inspection server. In the case of a punt, the original flow is diverted or redirected to the inspection server.

The inspection server is able to provide forensics or higher-level analytics of the traffic. Based on the results of that inspection, the data flow could be either dropped, or sent back to its original path. Furthermore, the inspection server could instruct the edge router to drop all subsequent packets in that flow. Finally, the data could be sent to a long-term storage cluster for further post-hoc forensics.

Central to the concept of diversion networking is the fact that, unless the data is dropped, the sender or source of the traffic has no way of knowing that the traffic flow has been diverted.

Figure 17 shows sample code of onePK's DataPath Service Set (DPSS). DPSS provides application developers hooks into the packet flow through a Cisco switch or router to extract packets from that flow. In this case, the packet is diverted (that is, punted) from its routed path and sent to the application, based upon a predefined source and destination IP addresses in an ACL. The application obtains relevant information about the packet, such as IP version number, source and destination addresses and ports, and Layer 4 protocol. In this sample code, if the packet is an IPv4 TCP packet, the Layer 4 payload is sent to a function which does additional processing of the TCP payload, such as advanced malware detection or intrusion detection, based upon predefined signatures.

Figure 17. Sample Code - Packet Processing for Deep Packet Inspection

```
/*
 * Packet callback - TCP payload processor.
 * If IPv4 && TCP, sends payload to Process_TCP_Payload() for deep packet inspection
 */
void dpss_pak_inspect_callback(onep_dpss_traffic_reg_t *reg, struct onep_dpss_paktype_ *pak, void *client_context, bool *return_packet) {

   < Variables removed for brevity >

   rc = onep_dpss_pkt_get_flow(pak, &fid);
   if( rc != ONEP_OK ) {
      fprintf(stderr, "Error getting flow ID. code[%d], Error[%s]\n", rc, onep_strerror(rc));
      return;
   }
   Get_IP_Version(pak, &ip_version);
   Get_IP_Packet_Info(pak, &src_ip, &dest_ip, &src_port, &dest_port, &l4_protocol, &ip_version);
   Get_Flow_State(pak, fid, &l4_state);

   if (ip_version == IPV4_VERSION && l4_protocol == TCP_PROTOCOL) {
      rc = onep_dpss_pkt_get_payload_size(pak, l4_payload_size);
      if( rc != ONEP_OK ) {
         fprintf(stderr, "Error getting L4 Payload Size. Code[%d], Error[%s]\n", rc, onep_strerror(rc));
         return;
      }
      rc = onep_dpss_pkt_get_payload(pak, &l4_payload);
      if( rc != ONEP_OK ) {
         fprintf(stderr, "Error getting L4 Payload. Code[%d], Error[%s]\n", rc, onep_strerror(rc));
         return;
      }
      rc = Process_TCP_Payload(l4_payload, l4_payload_size, &l4_state);
      if( rc != ONEP_OK ) {
         fprintf(stderr, "Error processing L4 TCP Payload. Code[%d], Error[%s]\n", rc, onep_strerror(rc));
         return;
      }
   }
   return;
}
```

Based upon the results of the payload processing, the packet could be sent back from the application to the original destination, or dropped.

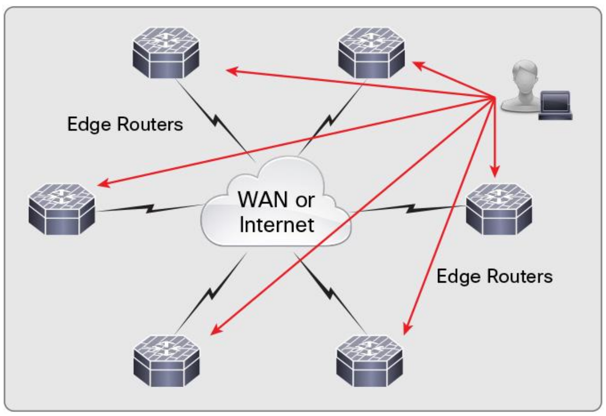
This example shows how one can simply divert network traffic from its original path to an application, which both offloads the processing from the router and provides a mechanism for deep packet inspection and analytics.

Use Case 4: Central Policy Management for Access Control Lists

An ACL is a mechanism by which traffic that is entering (inbound) or leaving (outbound) an interface on a router or switch may be filtered. This traffic may be allowed or denied, based on the configuration of the ACL. ACLs in this context are primarily used as a network security mechanism, in which traffic from a specific host or subnet is blocked or dropped. While ACLs can be applied at many places in the network, they are most often installed on either the WAN edge, which connects to a separate network or autonomous system, or to routers that connect directly to the Internet itself. In these cases, ACLs applied to the network edge provide the first line of defense against cyber attacks.

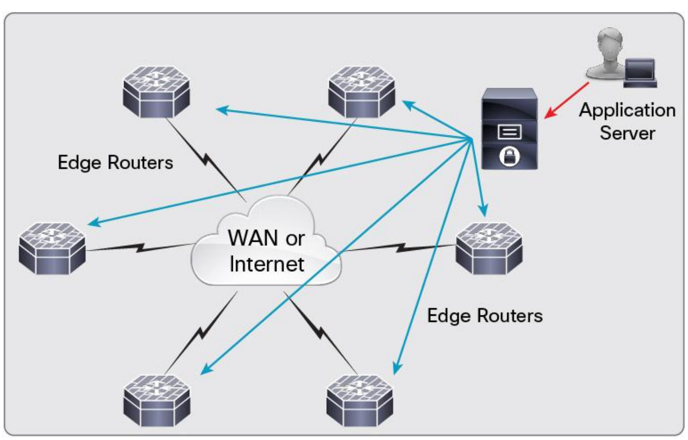
Installation of ACLs on network devices is often a manual process. A network operator moves from device to device, installing what is often the exact same ACL (Figure 18). The number of devices can be in the hundreds, and the size of the ACLs can be as large as 100,000 entries or larger.

Figure 18. Installation of ACLs on WAN or Internet Edge Devices



What is needed is a scalable approach to ACL management, where the ACL policy is centralized. Programmatic interfaces through onePK allow direct programmatic access to network devices from a central server, allowing an operator to run a single application and install an ACL on many devices simultaneously. This is shown in Figure 19.

Figure 19. Installation of ACLs from a Centralized Application Server



This approach simplifies operations, allows explicit error checking, and reduces the possibility of human-induced errors in the process. The central policy management of ACLs also allows additional logic, such as ACL expiration or time-of-day policies, to be applied. Finally, a centralized ACL policy allows the immediate blocking of malicious data traffic from all the routers at network edge in real time, should a new threat be detected.

Sample code of installation of a preconfigured ACL on a set of routers is shown in Figure 20.

Figure 20. Sample Code - Applying an ACL on a Group of Routers

```
/**
 * Apply the ACL to an interface for each router in the network.
 */
onep_status_t Install_ACL(struct Router_Info **router_info, struct Network_Info *network_info) {

  int i;
  onep_status_t rc = ONEP_OK;

  for (i = 0; i < network_info->num_routers; i++) {
    rc = onep_acl_apply_to_interface(router_info[i]->acl, router_info[i]->interface, router_info[i]->acl_direction);
    if (rc != ONEP_OK) {
      fprintf(stderr, "\nFailed to Apply ACL to interface %s for address %s\n",
              router_info[i]->if_name, router_info[i]->element_addr);
      fprintf(stderr, "Return Code: %s\n", onep_strerror(rc));
      return ONEP_FAIL;
    }
  }

  return ONEP_OK;
}
```

This use case demonstrates the power of onePK to simplify operations by automating functions to increase efficiency.

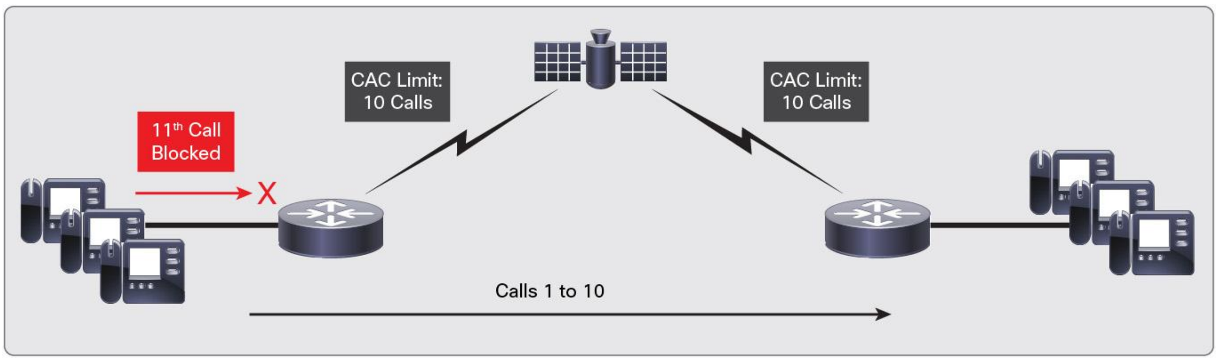
Use Case 5: Call Admission Control in Multiple Security Domain Networks

VoIP and unified communications are a group of technologies which allow both voice and data traffic to run over a single infrastructure. VoIP utilizes network bandwidth extremely efficiently, as packet switching is much more effective than transmission over a circuit-switched network. For these reasons, VoIP can both reduce total network and infrastructure costs, as well as circuit costs.

Networks, however, can support a limited number of voice calls at any given time. Real-time traffic such as voice requires sufficient bandwidth, and voice quality may suffer because of latency, jitter, and loss. These constraints are especially pronounced in tactical networks, where connections between two locations are over satellite links. In these scenarios, WAN links are typically well below 10 Mbps. Because of this, the concept of Call Admission Control (CAC) is typically used, which limits the number of simultaneous voice calls over a given link or portion of the network. CAC prevents voice traffic oversubscription of WAN links. For real-time traffic such as voice, it is better to deny network access under congestion conditions than to allow traffic onto the network which could be dropped, resulting in poor voice quality.

Figure 21 demonstrates a simplified view of the concept of CAC in a tactical environment. In this example, the network supports a total of 10 VoIP calls. When the eleventh call is attempted, it is rejected because the network is at maximum voice (VoIP) capacity. That is, CAC works in the call set-up phase, and protects voice traffic from other voice traffic. Should the network allow the eleventh call through the network, the voice quality and user experience of all of the calls would suffer. CAC makes a decision based on whether the required network resources are available to provide suitable QoS for the new call.

Figure 21. Call Admission Control in a Tactical Network

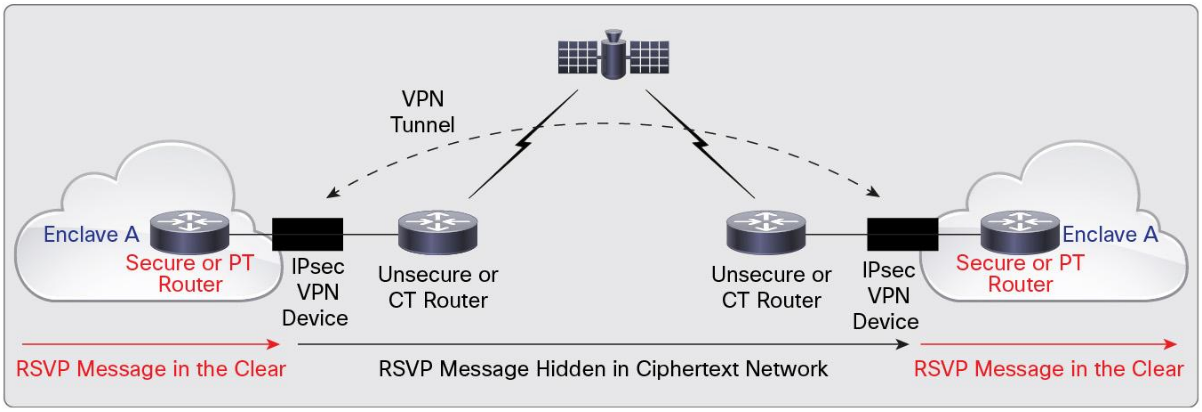


One common CAC mechanism is Resource Reservation Protocol (RSVP). RSVP is a signaling mechanism that is designed to reserve resources across a network. This gives RSVP the unique advantage of not only providing CAC for voice, but also helping to guarantee the QoS against changing network conditions for the duration of the call. RSVP can be initiated by either hosts or routers, which can request specific QoS guarantees, such as bandwidth or performance, for voice calls. RSVP is aware of the topology of the network because the RSVP reservation is end to end and installed on every interface the call will traverse through the network.

The end-to-end nature of RSVP introduces a challenge in tactical networks. Tactical networks are often deployed with multiple security domains or enclaves. Data within the enclave is unencrypted or “in the clear,” and each enclave has a “secure” or plaintext (PT) router. Each enclave is protected by an IP Security (IPsec) VPN device (IVD), which provides packet-based encryption at the IP layer. The IVD within an enclave encrypts packets received from the PT router and forwards it to the destination IVD through a single “unsecure” or ciphertext (CT) router. In this context, “secure” refers to routers within the encryption boundary of the IVD. “Unsecure” refers to routers that are outside the encryption boundary, which carry only encrypted traffic but are outside the security domain.

Within a particular enclave, the IVDs build VPN tunnels between themselves. Because the RSVP messages originate from within each enclave on the secure or PT side (which then enters the VPN tunnel), the RSVP traffic is hidden from the CT network. As such, each PT router has no knowledge of the number of calls (that is, the amount of reserved bandwidth) the CT router is able to support. Figure 22 highlights the issue of RSVP over an encrypted or ciphertext network, although only a single enclave is shown for illustrative purposes.

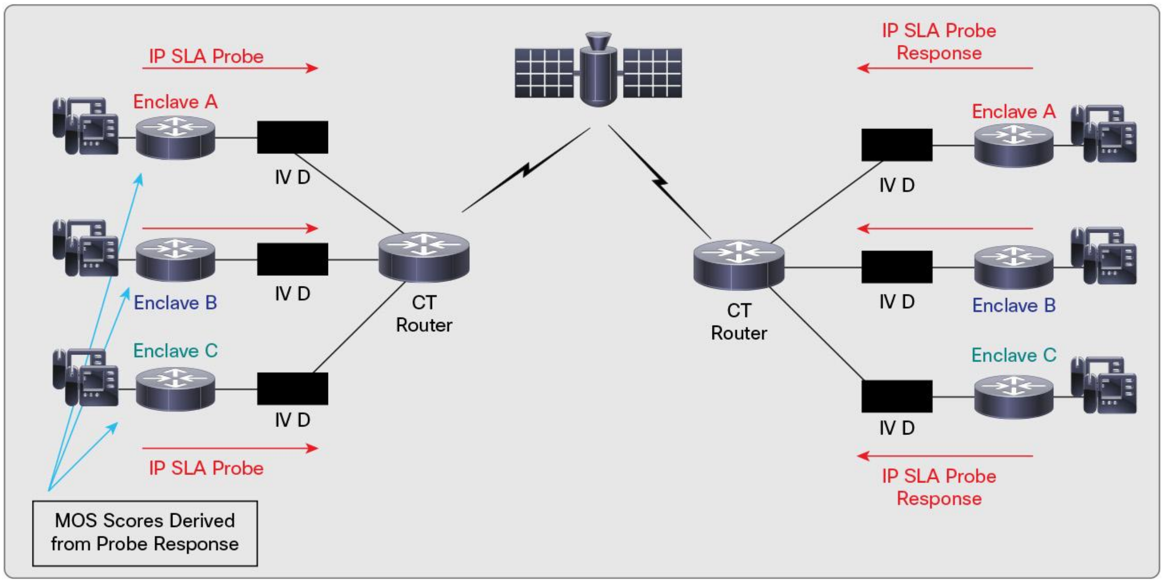
Figure 22. RSVP Messages Across a Ciphertext Network (Single Enclave Shown)



Because the RSVP messages are hidden from the ciphertext network, one must infer the state of the ciphertext core from the PT routers. This environment is additionally challenging when multiple security enclaves are connected to a single CT core.

To resolve this, the PT routers in each enclave may send probes across the ciphertext core in order to determine if the CT network has enough bandwidth to support new calls. This solution utilizes Cisco IOS IP Service Level Agreement (IP SLA), which is a Cisco IOS mechanism that is able to diagnose network suitability for real-time traffic applications such as VoIP. IP SLA is able to proactively monitor VoIP quality levels across the ciphertext network by calculating consistent voice quality scores (mean opinion scores or MOSs) between PT routers within an enclave. That is, each PT router sends an IP SLA probe at a specified frequency. The response from those probes allows each router to determine if the ciphertext network can support additional voice calls at an acceptable quality level. Figure 23 highlights this process with three security enclaves.

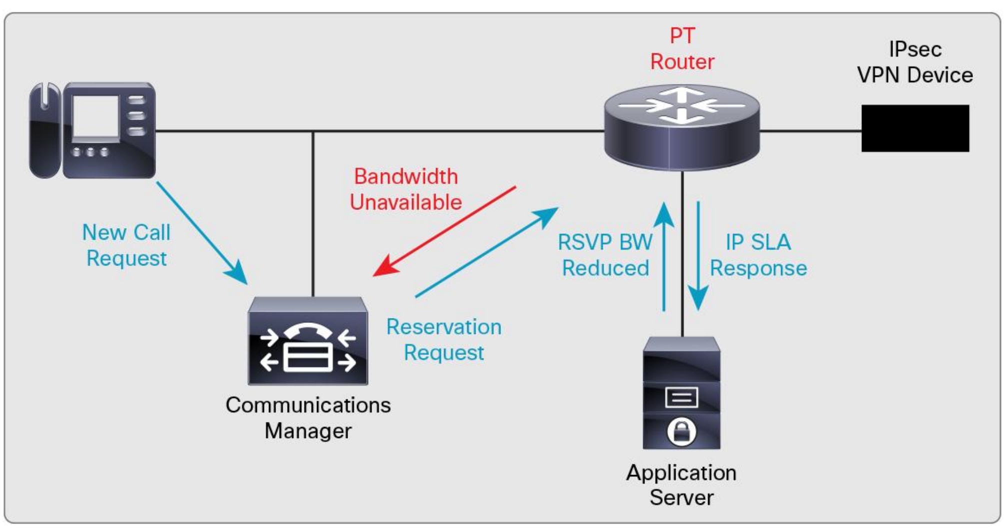
Figure 23. IPSLA Probes from Multiple Security Enclaves Across the Ciphertext Network



Within each enclave, a onePK application running on a server within the enclave is signaled at the return of an IP SLA probe with the MOS score that represents the voice quality along the end-to-end path. The application checks the MOS value against a user-defined threshold, such as a MOS score of “3” or “fair.” If the value from the response falls below this predefined threshold, the RSVP bandwidth defined on the PT router is reduced.

When a new call request comes into the communications manager, it sends a reservation request to the PT router. In this case, the request is rejected with a bandwidth unavailable code. This helps to ensure that no new calls are placed from within the enclave, protecting voice quality for all current calls. Figure 24 represents this process. While an external server representing the end-node hosting model is shown here, a blade or process hosting model within the PT router itself could be used.

Figure 24. Call Blocked Based on IP SLA Response Metric



Finally, the PT router will continue to send and receive probes. When it receives a probe with the MOS score above the predefined threshold, the RSVP bandwidth is incremented, allowing new calls within the enclave.

Figure 25 is sample code of an event handler that is called after the receipt of an IP SLA probe. This involves processing the probe and applying a new RSVP bandwidth value to the router if the MOS value is below the predefined threshold. The processing and post-processing functions provide analytics to maximize bandwidth utilization and minimize oscillations of blocked calls.

Figure 25. Sample Code - Event Handler for IP SLA Probe

```
/* Handler for Application Publish Event
 * This is invoked when an IP SLA application event (probe) is received.
 * Client_data is of type struct call_data *
 */
int appl_handler(appl_event_t *event, void *client_data)
{
   event_handle_t eh;
   uint32_t data_index = 2;
   char *ipsla_mos_value = NULL;
   unsigned int IPSLA_Event_Code = 0;
   onep_status_t rc = ONEP_OK;

   rc = onep_appl_event_get_event_handle(event, &eh);
   if (rc != ONEP_OK) {
      fprintf(stderr, "Failed to get event handle: %s\n", onep_strerror(rc));
      onep_appl_event_destroy(&event);
      return rc;
   }

   rc = onep_appl_event_get_data(event, data_index, &ipsla_mos_value);
   if (rc != ONEP_OK) {
      fprintf(stderr, "Failed to get event data: %s\n", onep_strerror(rc));
      onep_appl_event_destroy(&event);
      return rc;
   }

   Set_Current_MOS_Score((struct call_data *)client_data, atof(ipsla_mos_value));

   IPSLA_Event_Code = Process_IPSLA_Event((struct call_data *)client_data);

   if (IPSLA_Event_Code == UPDATE_RSVP_BW )
      Set_RSVP_BW((struct call_data *)client_data);

   Post_Process_IPSLA_Event((struct call_data *)client_data, IPSLA_Event_Code);

   return rc;
}
```

This solution is not without its challenges. Because there is no true visibility into the state of the ciphertext network, the system must infer and react to congestion and potential voice quality issues. Furthermore, because there are multiple enclaves with no central control, enclaves may compete for the same bandwidth on the ciphertext network. This may lead to oscillation and tuning issues such as underutilization of available bandwidth, or calls being blocked when bandwidth is actually available.

Regardless, there are several advantages to this solution. It is simple in design, and can be deployed without any modification to the ciphertext network. Likewise, this solution can be implemented without the need for a network guard, either between the secure and unsecure networks, or between the secure enclaves themselves. This solution may also be deployed in many network topologies, such as ciphertext core architectures that provide load sharing, because the probe traffic and voice bearer traffic follow the same path through the network. Finally, factors such as probe frequency as well as averaging and hysteresis algorithms may minimize oscillations while providing optimal bandwidth utilization. This use case demonstrates the ability of network programmability to react to dynamic conditions within a network and modify the state of the network based on those conditions.

Conclusion

In the defense and intelligence communities, the network is mission-critical. Software-defined networking provides a new architecture for this network, an architecture that brings the application layer and the networking layer closer together. It is a shift in technology that will change the networking landscape by providing greater automation and orchestration of the network fabric, and by allowing dynamic, application-led configuration of networks and services. SDN allows the network to respond to requests from an application in real time, based upon the current state and condition on the network. It is this agility which will transform networks as we know them.

With options for both classical and hybrid approaches to software-defined networking, you have a choice. By introducing network intelligence in a centralized controller, the classical SDN model provides agility, automation, and a decrease in costs. However, it poses challenges of high availability, scalability, and performance. The hybrid SDN model minimizes some of these challenges by allowing you to choose which applications and features to migrate to a central controller. The hybrid model continues to provide many of the benefits of the classical approach to defense and intelligence communities.

Cisco's approach to software-defined networking and network programmability is through the Cisco Open Network Environment (ONE). This is an encompassing solution that allows network programmability and application awareness. Programmability is achieved by exposing platform APIs to developers through the ONE platform kit (onePK). The benefits of ONE and onePK include automation, orchestration, and customization of the network, over an infrastructure layer that is abstracted from the platform. This, in turn, allows flexibility and simpler management, both of which are key attributes of a mission-ready network.

The use cases described in this paper are examples of how SDN and onePK can provide benefits to defense and intelligence networks. The use cases provide real-world examples as well as snippets of code, which developers can use as guides when customizing their own networks. Developers can choose from the variety of examples and decide which use case most closely matches their needs. Several different programming languages can be used, providing flexibility of choice. In addition, because several different deployment models can be used, network administrators have multiple options in incorporating SDN and network programmability into their environment. Finally, network security is an integral part of the onePK solution, and operators can be confident that onePK can be deployed, in a highly secure manner, into their current environment. Cisco's approach to software-defined networking and network programmability provides a path to a new architecture for defense and intelligence communities.

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