



The bridge to possible

# Роль SD-WAN в сервисах совместной работы

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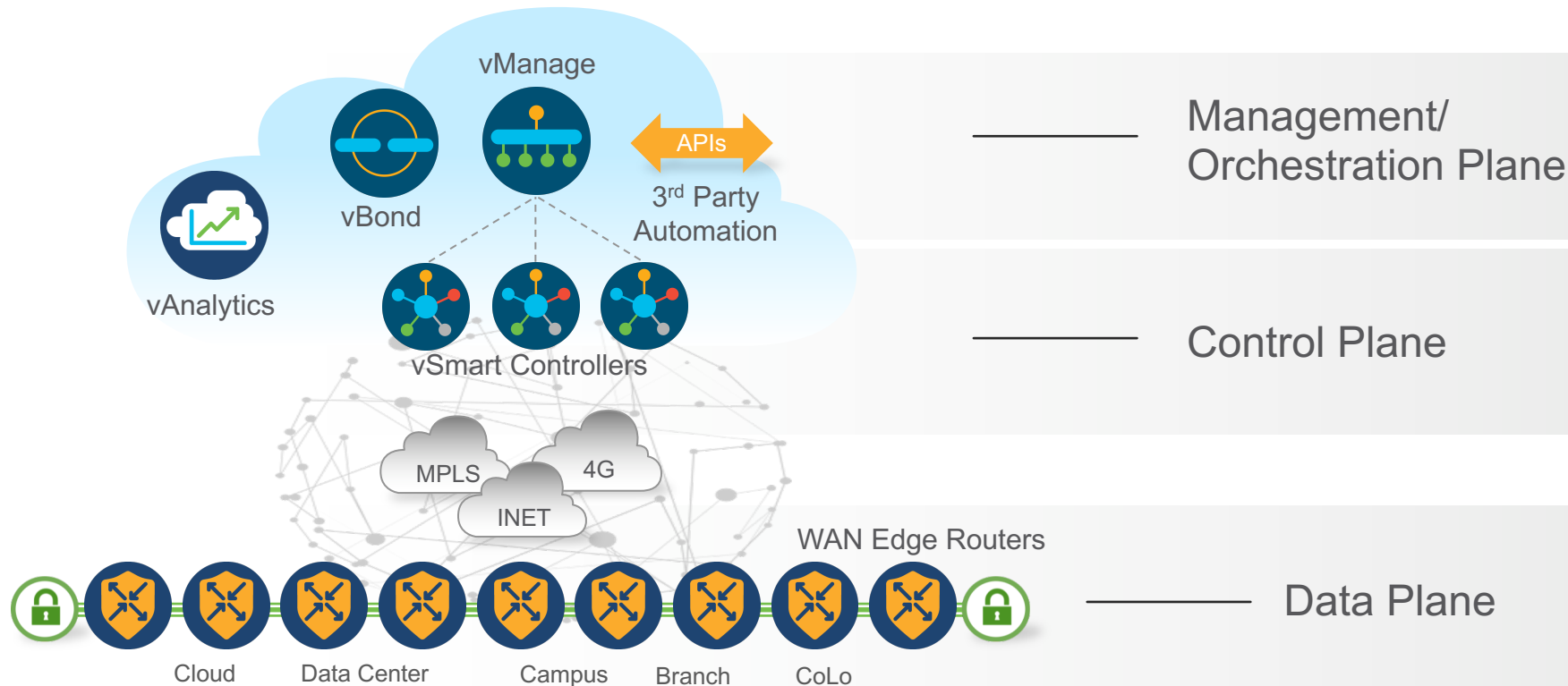
# Содержание

- Обзор Cisco SD-WAN
- Внедрение UC на ISR и Catalyst 8000 в режиме SD-WAN
- Оптимизация унифицированных коммуникаций с помощью SD-WAN
- Ключевые выводы

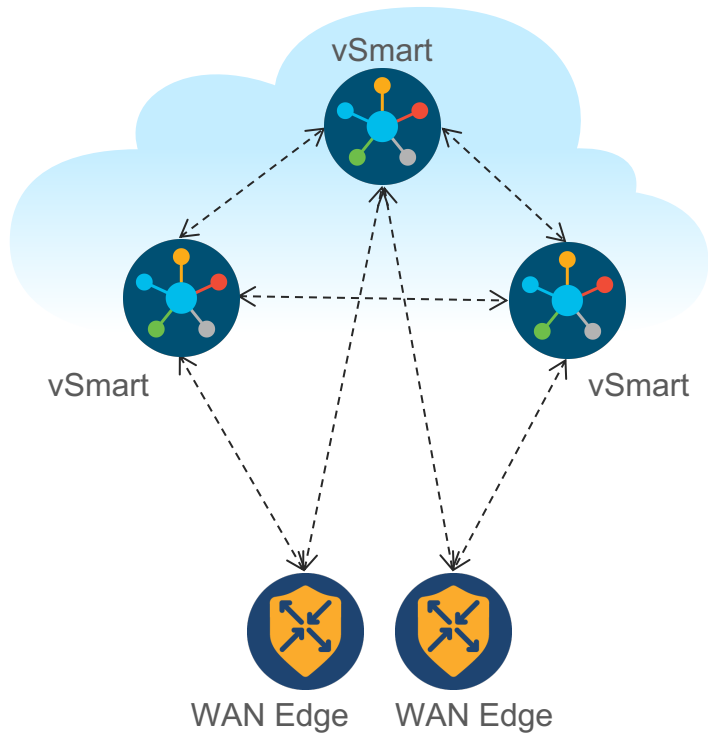
# Обзор Cisco SD-WAN

# Обзор решения Cisco SD-WAN

Применение SDN подходов к WAN-сетям

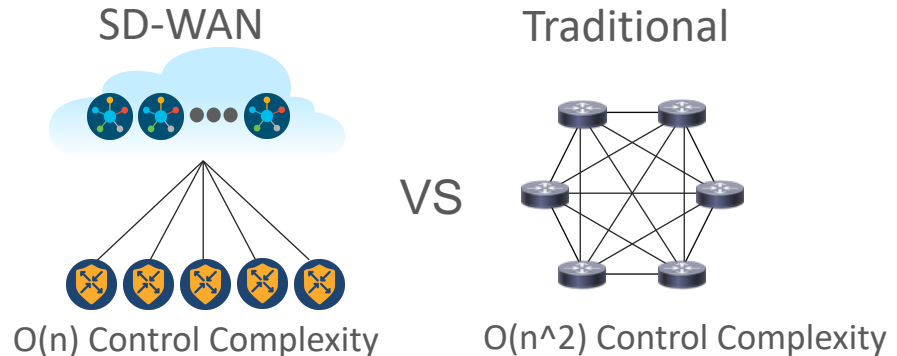


# Overlay Management Protocol

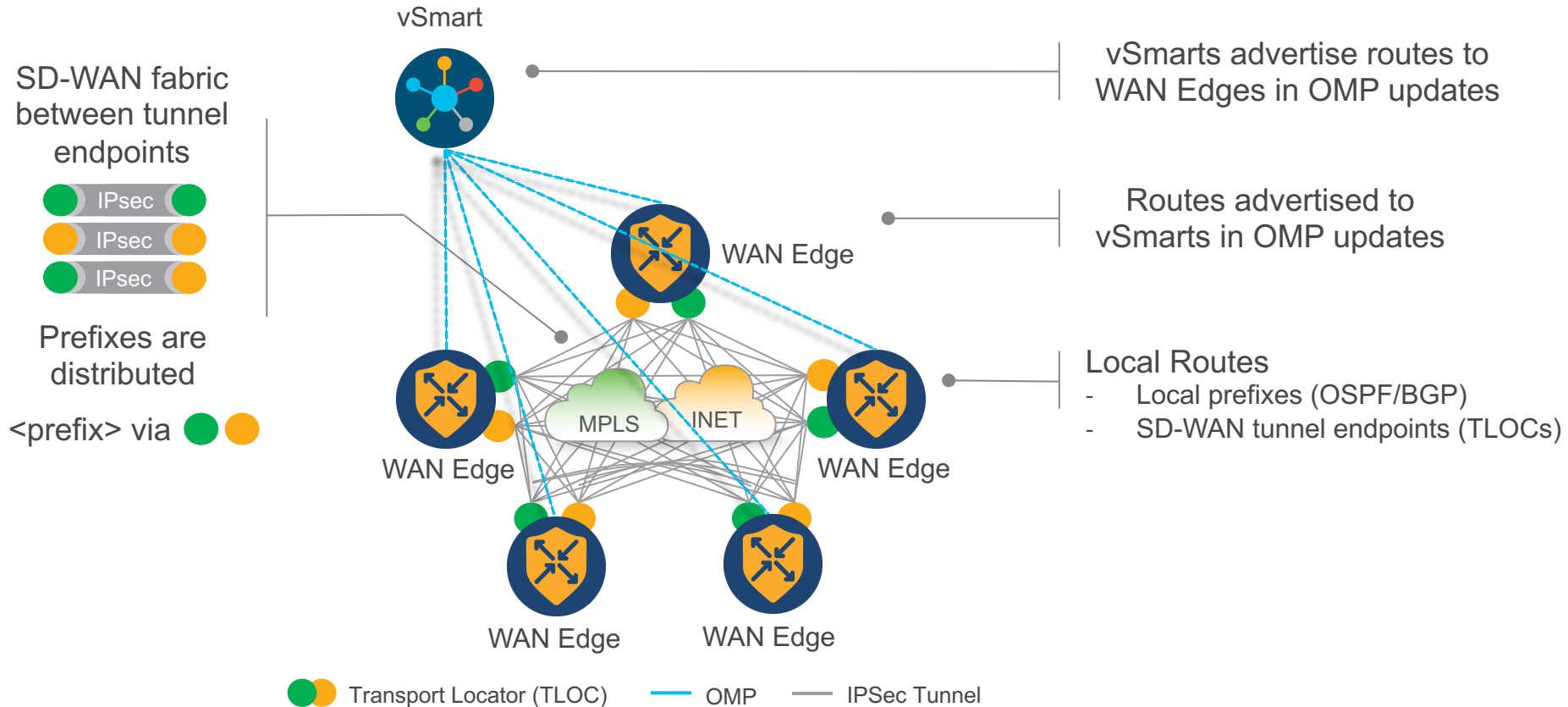


Note: WAN Edge routers need not connect to all vSmart Controllers

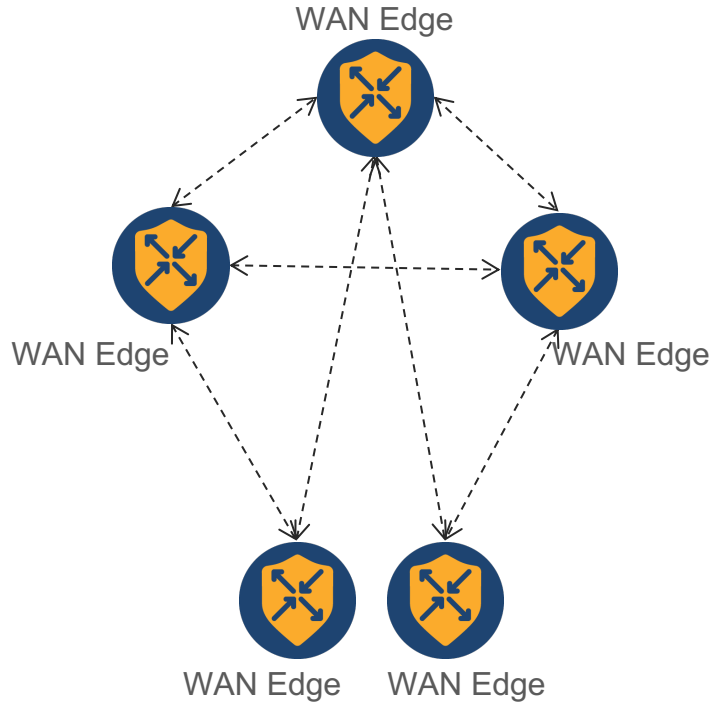
- Overlay Management Protocol (OMP)
- TCP-based extensible control plane protocol
- Runs between WAN Edge routers and vSmart controllers and between the vSmart controllers
  - Inside authenticated TLS/DTLS connections
- Advertises control plane context and policies
- Dramatically lowers control plane complexity and raises overall solution scale



# Data Plane – установка туннелей



# Bidirectional Forwarding Detection (BFD)



Path liveness and quality measurement detection protocol

- Up/Down, loss/latency/jitter, IPSec tunnel MTU

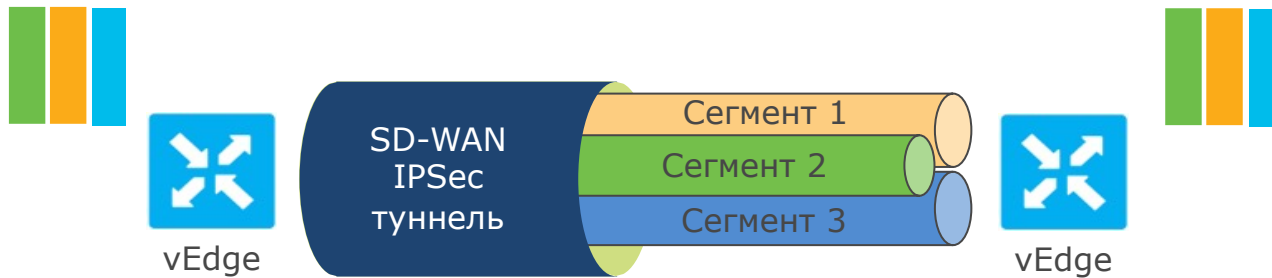
Runs between all WAN Edge and WAN Edge Cloud routers in the topology

- Inside IPSec tunnels
- Operates in echo mode
- Automatically invoked at IPSec tunnel establishment
- Cannot be disabled

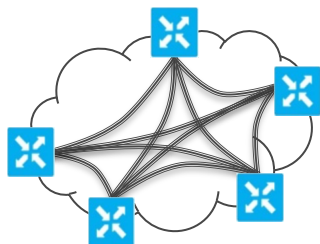
Uses hello (up/down) interval, poll (app-aware) interval and multiplier for detection

- Fully customizable per-WAN Edge, per-color

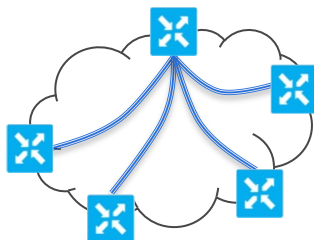
# Безопасная сегментация



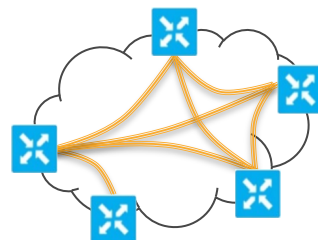
Уникальная топология для каждого VRF



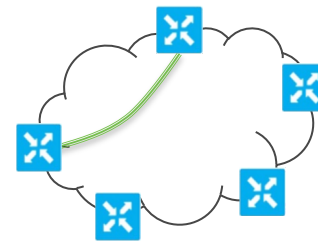
*Full Mesh топология*



Централизованная  
*Hub-and-Spoke*



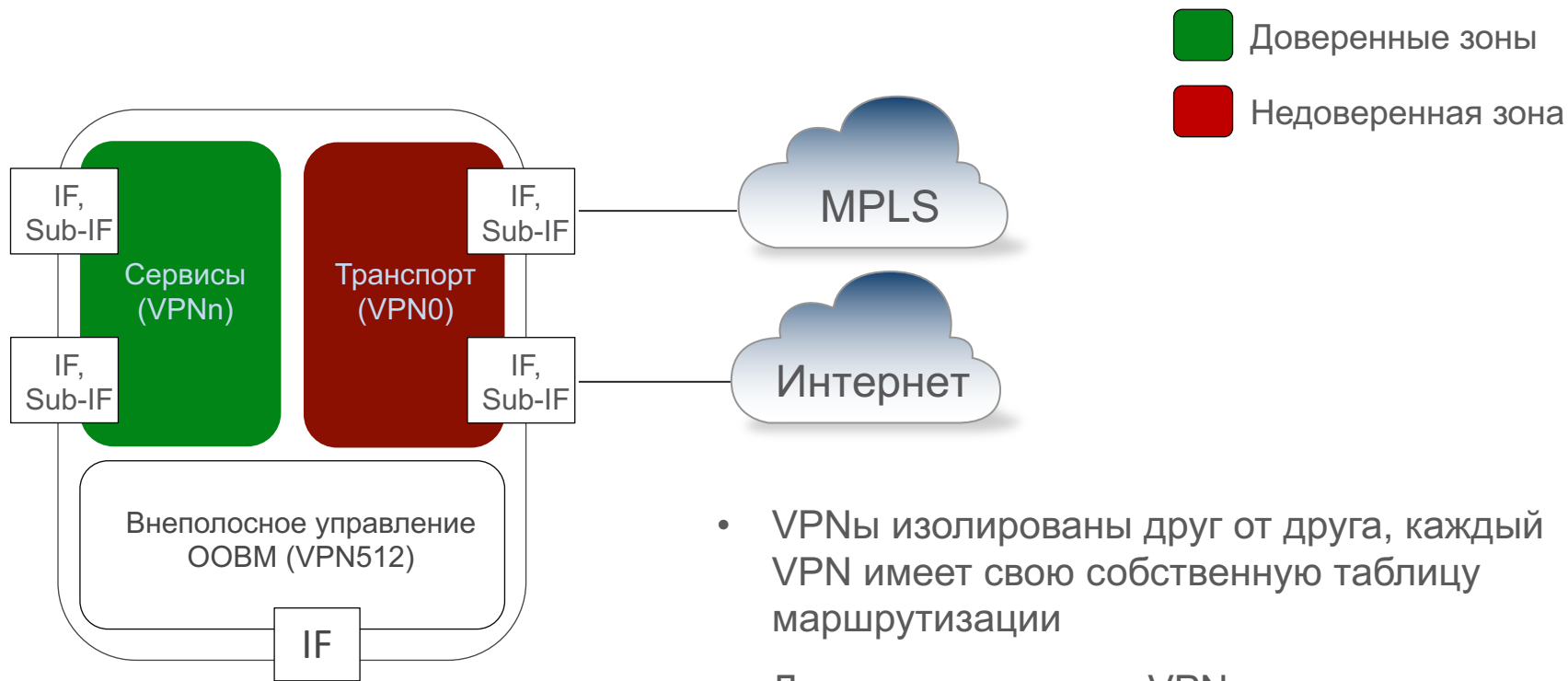
Частично связанная  
*Partial Mesh*



Точка-точка  
*Point-to-Point*

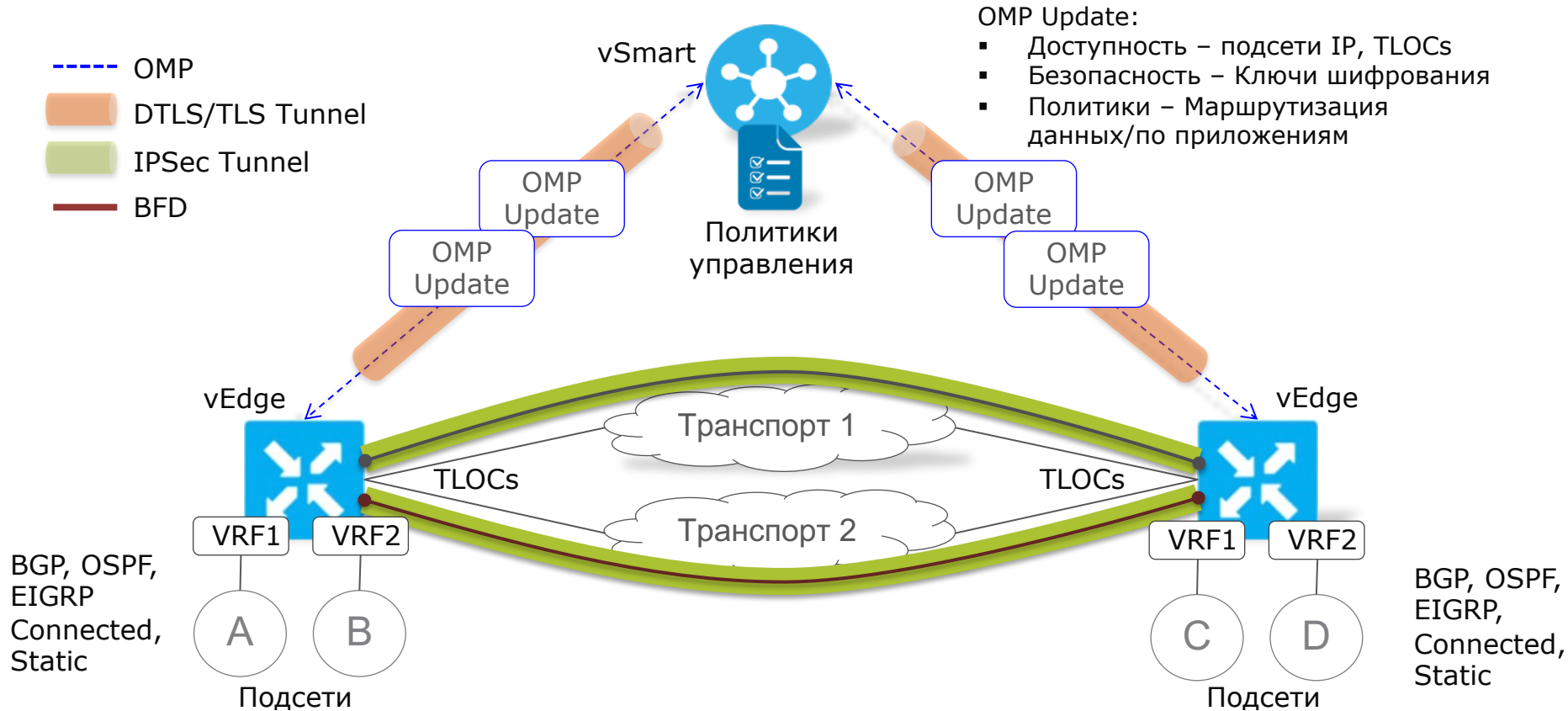


# Понятие VPN (VRF) и зон безопасности в vEdge



- VPNы изолированы друг от друга, каждый VPN имеет свою собственную таблицу маршрутизации
- Доступность внутри VPN автоматически анонсируется через OMP

# Как работает SD-WAN фабрика



# Платформы для Cisco SD-WAN

## Только SD-WAN

**vEdge 100**



50-100M

**vEdge 1000**



175-300M

**vEdge 2000**



**vEdge 5000**



1-2G

**ISR 1100-4G**



125-220M

**ISR 1100-6G**



300-550M

## SD-WAN с сервисами (поддержка UC)

**ISR 1000**



Next-gen  
Performance  
Flexibility

50-200M

**ISR 4000**



Modular Integrated  
services

**Catalyst 8000**



**ASR 1000**



High-performance  
with redundancy

1.6-19G (Basic)

0.7-9G (Medium)

## Виртуализация

**ENCS 5100**



**ENCS 5400**



## Публичные и частные облака



# Внедрение UC на ISR и Catalyst 8000 в режиме SD- WAN

# Поддержка унифицированных коммуникаций (для vManage 20.3-20.4)

## Problem

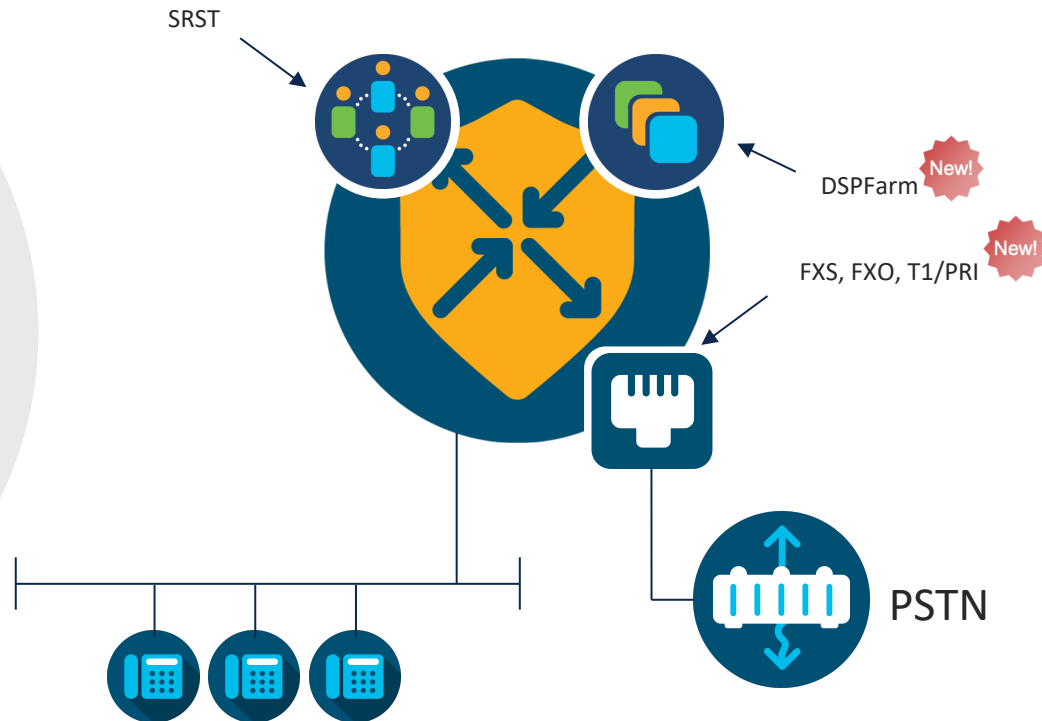
Customers seeking UC and SD-WAN integration were previously forced into a two-box solution at the branch. One box to terminate the SD-WAN fabric and another to handle UC termination. This increased cost, complexity and operational overhead.

## Solution

As of v20.1 and 17.2.1 (Phase 1), Cisco SD-WAN now supports UC and SD-WAN within a single box (analog, basic SIP and SRST). Version 20.3 / 17.3 (Phase 2) adds additional capability for T1/PRI termination, DSPfarming and Fax Passthrough.

## Caveats / Prerequisites

IOS-XE (cEdge) ISR only, 4GB DRAM is supported, CUBE is not supported, H323/MGCP/SCCP are not supported, T1/PRI requires separate PVDM



# FXO/FXS поддержка в SD-WAN



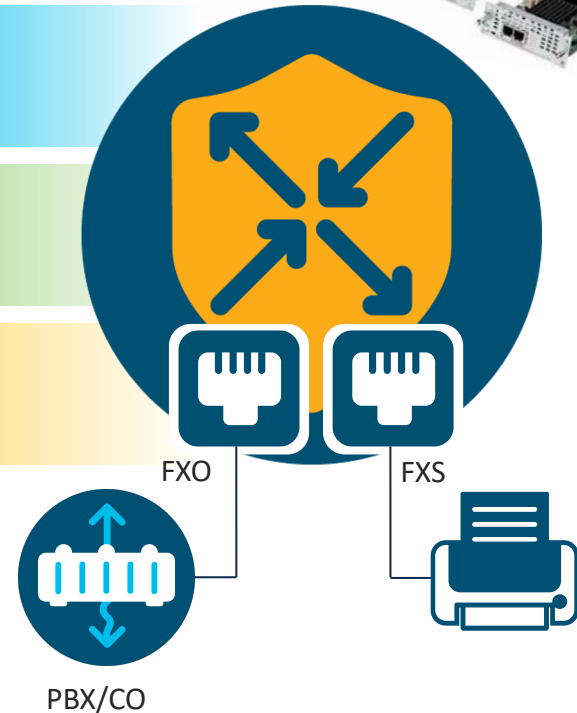
Connect to PBX or key systems, or provide off-premises connections to the public switched telephone network (PSTN)

Built-in DSP with high analog port-density support

NIM-2FXO  
NIM-4FXO  
NIM-2FXSP  
NIM-4FXSP

NIM-2FXS/4FXOP  
SM-X-8FXS/12FXO  
SM-X-16FXS/2FXO  
SM-X-24FXS/4FXO

SM-X-72FXS



# T1/E1 голосовая PRI поддержка в SD-WAN

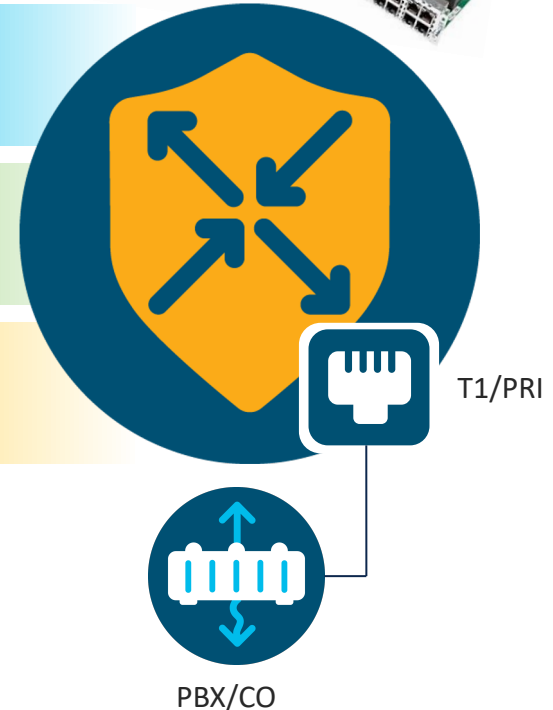


Packet Voice Solutions support (PBX & Central-Office Connectivity)

PSTN termination with multi calls per port: T1 PRI (23) and E1 (30)

NIM-1MFT-T1E1  
NIM-2MFT-T1E1  
NIM-4MFT-T1E1  
NIM-8MFT-T1E1

NIM-1CE1T1-PRI  
NIM-2CE1T1-PRI  
NIM-8CE1T1-PRI



- T1/E1 Voice module contains onboard PVDM4 Slot
- PVDM4 Module **required** for T1/E1 packetization (**purchased separately**)
- Supported ISDN Switchtypes: QSIG, NET5, NTT, 4ESS, 5ESS, DMS100, and NI

# DSPFarm сервисы в SD-WAN

Multi party audio conferencing with (8,16, 32) participants

Save bandwidth with audio codec transcoding

Media Termination Point for IP Calls  
(DTMF Conversion, SIP call bridging, Trusted Relay Point, etc.)



**Form Factor:**

- SM-X-PVDM-500
- SM-X-PVDM-1000
- SM-X-PVDM-2000
- SM-X-PVDM-3000



**Form Factor:**

- PVDM4 – 32
- PVDM4 – 64
- PVDM4 – 128
- PVDM4 – 256



**Form Factor:**

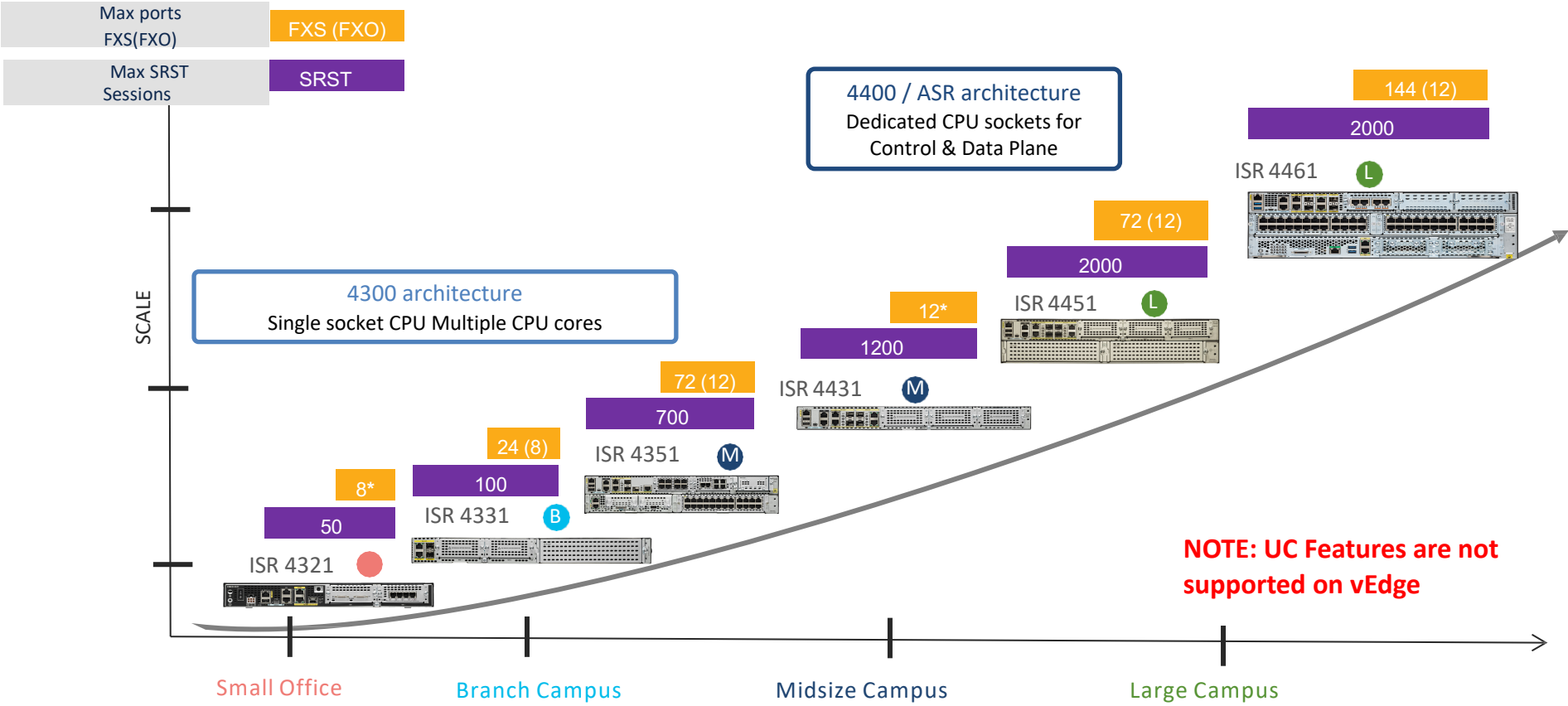
- NIM-PVDM-32
- NIM-PVDM-64
- NIM-PVDM-128
- NIM-PVDM-256



**NIM-PVDM Modules** for IP Voice Services  
**SM modules** for high density DSP usage



# ISR 4000 IOS XE / SDWAN UC Scale



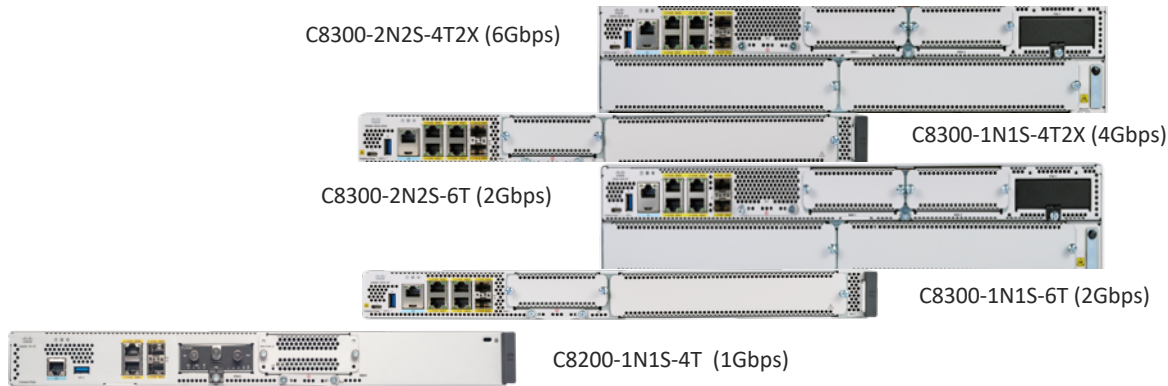
# Cisco Catalyst 8000 Edge Platforms Family

The Leading SD-WAN Edge Platforms with Rich Services



Performance and Rich Services

2500 Phones on  
each platform



Scalable Architecture with x86 and QFP

# UC настройка и политики

## vManage

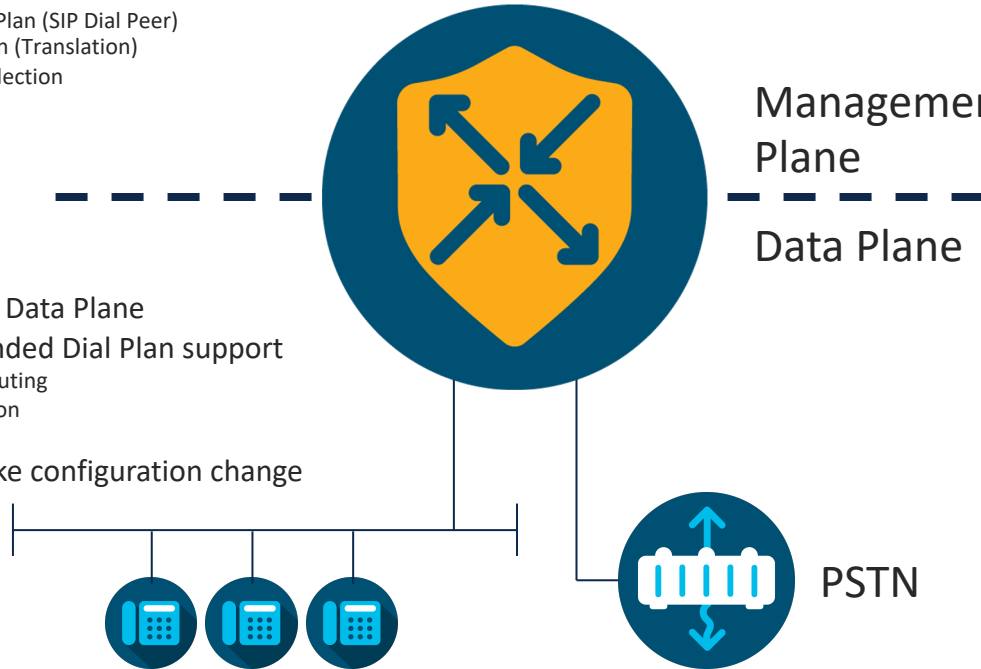


- Does not participate in Call Routing  
Provisions ISR for UC
- Distributed Dial Plan (SIP Dial Peer)
  - Call Manipulation (Translation)
  - Media/Codec Selection
  - SRST



## Call Control (CUCM)

- Participates in Data Plane  
Provides extended Dial Plan support
- Enterprise call routing
  - Media Termination
  - SIP
- Does not invoke configuration change

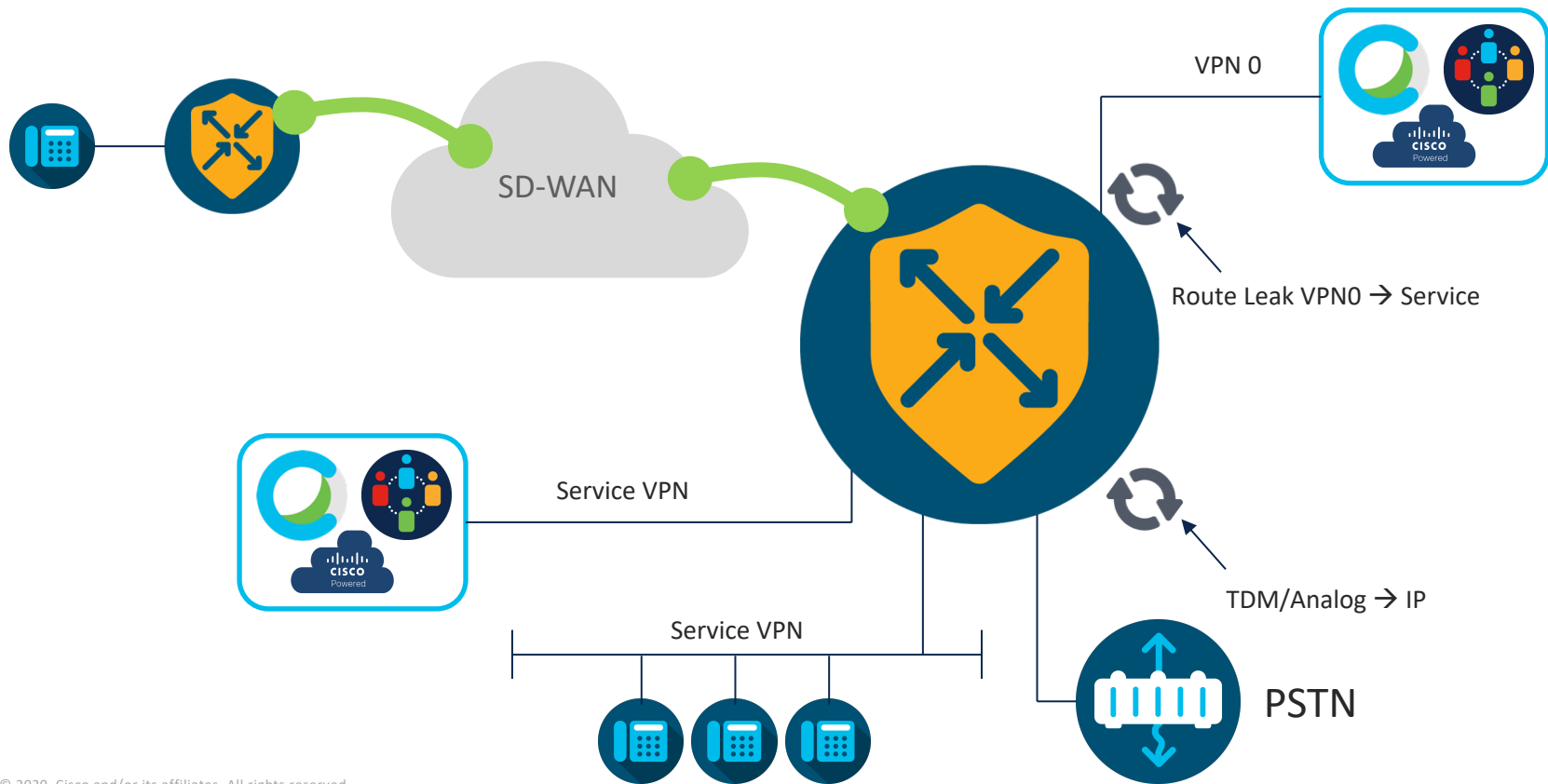


Management/Control  
Plane

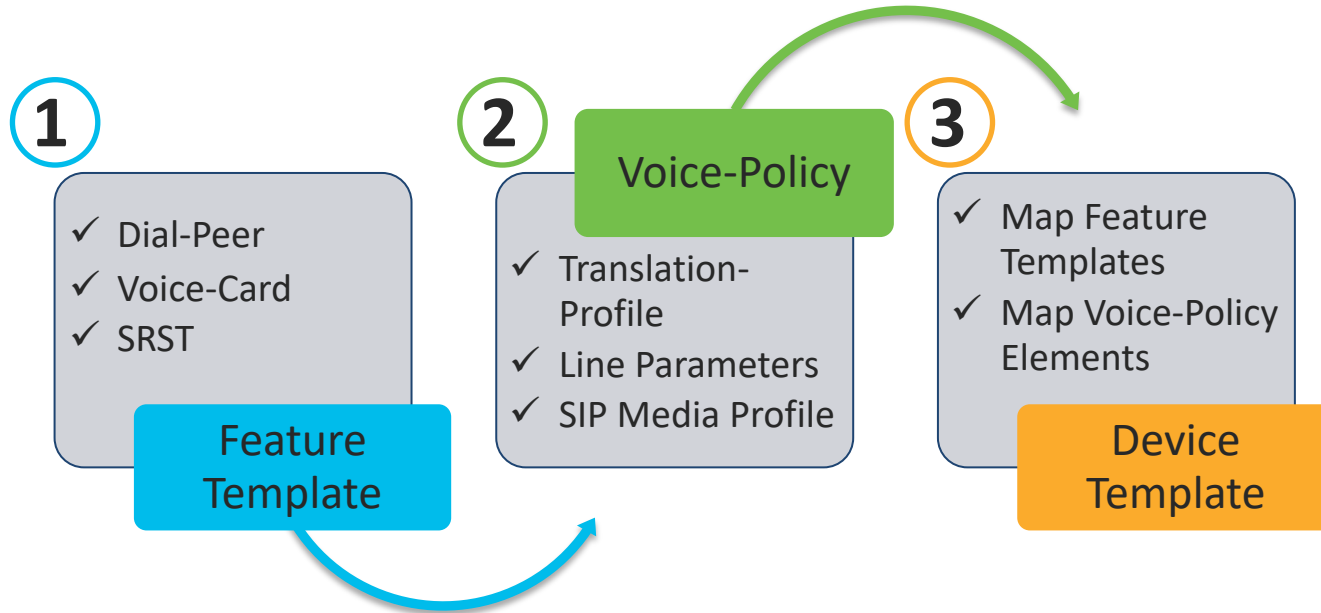
Data Plane

PSTN

# Поддерживаемые сценарии управления вызовами



# Процесс внедрения UC в SD-WAN



DEMO

# Voice Template Configuration

Configuration → Templates → Feature (Tab) → Add New

*Configuration of voice-ports and their associated Dial Plan is handled through vManage Templates. Feature Templates are created first to define physical port parameters (slot, subslot, etc.), POTS/SIP Dial Plan and SRST parameters. These parameters are then attached to a Device Template along with a Voice Policy.*

The screenshot displays the Cisco vManage interface for configuring templates. The main heading is 'CONFIGURATION | TEMPLATES'. Below this, there are tabs for 'Device' and 'Feature', with 'Feature' selected. The breadcrumb trail shows 'Feature Template > Add Template'. On the left, the 'Select Devices' section lists various Cisco ISR models, with 'ISR4321' highlighted in green and a red arrow pointing to it. On the right, the 'Select Template' section is divided into three main categories: 'BASIC INFORMATION', 'UNIFIED COMMUNICATION', and 'VPN'. The 'UNIFIED COMMUNICATION' section is highlighted with a red box and contains a warning message: 'UNIFIED COMMUNICATION (SIP and SRST templates need to be provisioned to enable stand-alone SRST feature)'. Below this warning, there are three template options: 'SIP', 'SRST', and 'Voice Card'. The 'VPN' section below contains options for 'Cisco Secure Internet Gateway (SIG) WAN', 'Cisco VPN', and 'Cisco VPN Interface Ethernet Management | WAN | LAN'.

# Voice Template Configuration

Configuration → Templates → Feature (Tab) → Add New → (Select ISR) → Voice-Card

The screenshot displays the Cisco vManage configuration interface for creating a new analog interface. The breadcrumb trail is: Configuration | TEMPLATES > Device > Feature > Feature Template > Voice Card. The main section is titled "Interface" and contains a "New Analog Interface" button. The configuration fields are as follows:

- Module:** A dropdown menu currently set to "-- Choose --".
- Module Slot/Sub-slot:** A text input field with a globe icon, accompanied by a checked "Use DSP" checkbox.
- Port Type:** A dropdown menu currently set to "-- Choose --".
- Port Selection:** Radio buttons for "All" and "Port Range". The "Port Range" option is selected, with a corresponding text input field and globe icon.
- Description:** A text input field with a globe icon.
- Connection Plan:** A dropdown menu with a checkmark icon.
- Signal Type:** Radio buttons for "Loopstart" (selected), "Groundstart", and "DID".
- Caller-ID Enable:** Radio buttons for "On" and "Off" (selected).
- Shutdown:** Radio buttons for "On" and "Off" (selected).



# Voice Template Configuration

Configuration → Templates → Feature (Tab) → Add New → (Select ISR) → SIP

The screenshot displays the Cisco vManage interface for configuring a Feature Template. The breadcrumb path is Configuration | TEMPLATES. The 'Feature' tab is selected, and the 'Feature Template' is set to 'SIP'. The configuration details are as follows:

Field	Value
Device Type	ISR4331
Template Name	TEST-SIP
Description	TEST SIP

Below the Feature Template configuration, the 'Global' tab is selected under the 'Dial Plan' section. The 'Global' configuration parameters are:

Field	Value
Trusted IPv4 Prefix	10.10.1.1/32
Trusted IPv6 Prefix	[Checked]
Fax Protocol	pass-through
Fax Codec	g711ulaw
Source Interface	GigabitEthernet0/0/3

# Voice Template Configuration

Configuration → Templates → Feature (Tab) → Add New → (Select ISR) → SIP

Dial Plan

[+ Upload Dial Peer List](#) [+ New Dial Peer](#)

Search  Search Options ▼ Total Rows: 3

Tag	Dial Peer Type	Direction	Description	Number Pattern	Forward Digits Type	Transport Protocol	Action
100	POTS	Incoming		,	None		
200	SIP	Outgoing		9T	None	UDP	
101	POTS	Incoming		,	None		

# Voice Template Configuration

Configuration → Templates → Feature (Tab) → Add New → (Select ISR) → SRST

The screenshot displays the Cisco vManage interface for configuring a Feature Template. The breadcrumb navigation is Configuration | TEMPLATES. The 'Feature' tab is selected, and the 'Add New' process is in progress for a device type of 'ISR4331'. The 'Feature Template' is named 'SRST'. The configuration fields are as follows:

Field	Value
Device Type	ISR4331
Template Name	TEST-SRST
Description	TEST SRST

Below the feature template configuration, the 'Global' section is expanded to show the 'Phone Profile' configuration. The 'Global' section includes the following settings:

Setting	Value
System message	SRST MODE!!!
Max phones to support	25
Max Directory Numbers	50
Music on hold	Yes
Music on-hold file	[Selected]

The 'Phone Profile' section is currently empty, with a 'New Phone Profile' button visible at the bottom.

# Voice Policy Configuration

Configuration → Unified Communications → Add Voice Policy

*Configuration of Voice Policy is handled through the Policy workflows of vManage (similar to Localized Data Policy). Voice Policy defines many of the parameters that augment voice-ports and Dial Plan (such as Translation Profiles, Supervisory Disconnect, Station ID, DTMF relay, etc.).*

The screenshot shows the Cisco vManage interface for configuring a voice policy. The breadcrumb navigation is 'CONFIGURATION > Unified Communications > Add Voice Policy'. A header instruction reads: 'Provide a name and description for your voice policy and configure policy profile settings. Click Save Policy to save the policy configuration.'

There are two input fields: 'Voice Policy Name' with a placeholder 'Maximum of 32 characters' and 'Voice Policy Description' with a placeholder 'Description of the policy'.

Below the fields, a section titled 'Select an end point type on the left and start creating your policy profile' contains a sidebar with the following options: 'Voice Ports' (highlighted), 'POTS Dial Peer', 'SIP Dial Peers', and 'SRST Phone'.

The main content area displays a green hexagonal icon with a plus sign and the text 'No Voice Ports policy profiles added.' Below this is a blue button labeled 'Add Voice Ports Policy Profile'.

At the bottom of the page, there are three buttons: 'Preview', 'Save Policy', and 'CANCEL'.

# Voice Policy Configuration

Configuration → Unified Communications → Add Voice Policy → Add Voice Ports Policy Profile

Select the policies from for the list below to start creating your policies.

FXO    FXS    FXS DID

- 
- Translation Profile ⓘ
  - Station ID ⓘ
  - Line Params ⓘ
  - Tuning Params ⓘ
  - Supervisory Disconnect ⓘ

Select the policies from for the list below to start creating your policies.

FXO    FXS    FXS DID

- 
- Translation Profile ⓘ
  - Station ID ⓘ
  - Line Params ⓘ
  - Tuning Params ⓘ

Select the policies from for the list below to start creating your policies.

FXO    FXS    FXS DID

- 
- Translation Profile ⓘ
  - Station ID ⓘ
  - Line Params ⓘ
  - DID Timers ⓘ

# Voice Policy Configuration

Configuration → Unified Communications → Add Voice Policy → Add POTS Dial Peer Policy Profile

The screenshot displays the Cisco vManage interface for configuring a POTS Dial Peer Policy Profile. The main window shows the 'Translation Profile' configuration with the following details:

- Name:** 10-to-4
- Translation Rules:** Calling
- Called - Translation Rule 10:** [edit] [delete]

A modal window titled 'Translation Rules -Called' is open, showing the configuration for a new translation rule. The 'Translation Rule Number' is set to 10. The modal includes an 'Add Rule' button and 'Export'/'Import' options.

The modal also displays a table of translation rules:

Rule	Match	Action	
1	/...../	replace /.../	...

At the bottom of the modal, there are 'Finish' and 'CANCEL' buttons.

# Voice Policy Configuration

Configuration → Unified Communications → Add Voice Policy → Add SIP Dial Peer Policy Profile

The screenshot shows the Cisco vManage interface for configuring a Media Profile. The breadcrumb trail is: CONFIGURATION > Unified Communications > Edit Voice Policy > Edit SIP Dial Peers Policy Profile. The page has a top navigation bar with 'Select Policy Profiles', 'Configure Policy Profiles', and 'Summary' buttons. The main content area is titled 'Media Profile' and contains two sections: 'Codec' and 'DTMF'. Each section has a 'Source' and a 'Target (Drag & drop to reorder)' column. In the 'Codec' section, the source contains 'G711aLaw', 'G722', and 'ilbc', while the target contains 'G729r8' and 'G711uLaw'. In the 'DTMF' section, the source contains 'inband', while the target contains 'rtp-nte', 'sip-notify', and 'sip-kpml'. There are green arrow buttons between the source and target boxes for moving items. At the bottom right, there are 'Save' and 'Cancel' buttons.

Media Profile

Media Profile Number: 1

Codec

Source: G711aLaw, G722, ilbc

Target (Drag & drop to reorder): G729r8, G711uLaw

DTMF

Source: inband

Target (Drag & drop to reorder): rtp-nte, sip-notify, sip-kpml

Save Cancel

# Voice Policy Configuration

Configuration → Unified Communications → Add Voice Policy → Add SRST Policy Profile

The screenshot shows the Cisco vManage interface for configuring a Media Profile. The breadcrumb trail is: CONFIGURATION > Unified Communications > Edit Voice Policy > Edit SRST Phone Policy Profile. The current step is 'Configure Policy Profiles', with 'Select Policy Profiles' completed and 'Summary' pending.

The 'Media Profile' configuration area includes:

- Media Profile Number:** A text field with a range of 1-10000.
- Codec:** A 'Source' list containing G711uLaw, G711aLaw, G722, and ilbc. A 'Target (Drag & drop to reorder)' list contains G729r8. Green arrows indicate the direction of selection.
- DTMF:** A 'Source' list containing rtp-nte, sip-notify, and sip-kpml. A 'Target (Drag & drop to reorder)' list contains inband. Green arrows indicate the direction of selection.

Buttons for 'Add New Media Profile' and 'Copy From Existing' are at the top left. 'Save' and 'Cancel' buttons are at the bottom right of the configuration area. A table at the bottom has columns for 'Media Profile Number', 'Codec', and 'DTMF'. At the very bottom, there are 'BACK', 'Next', and 'CANCEL' buttons.

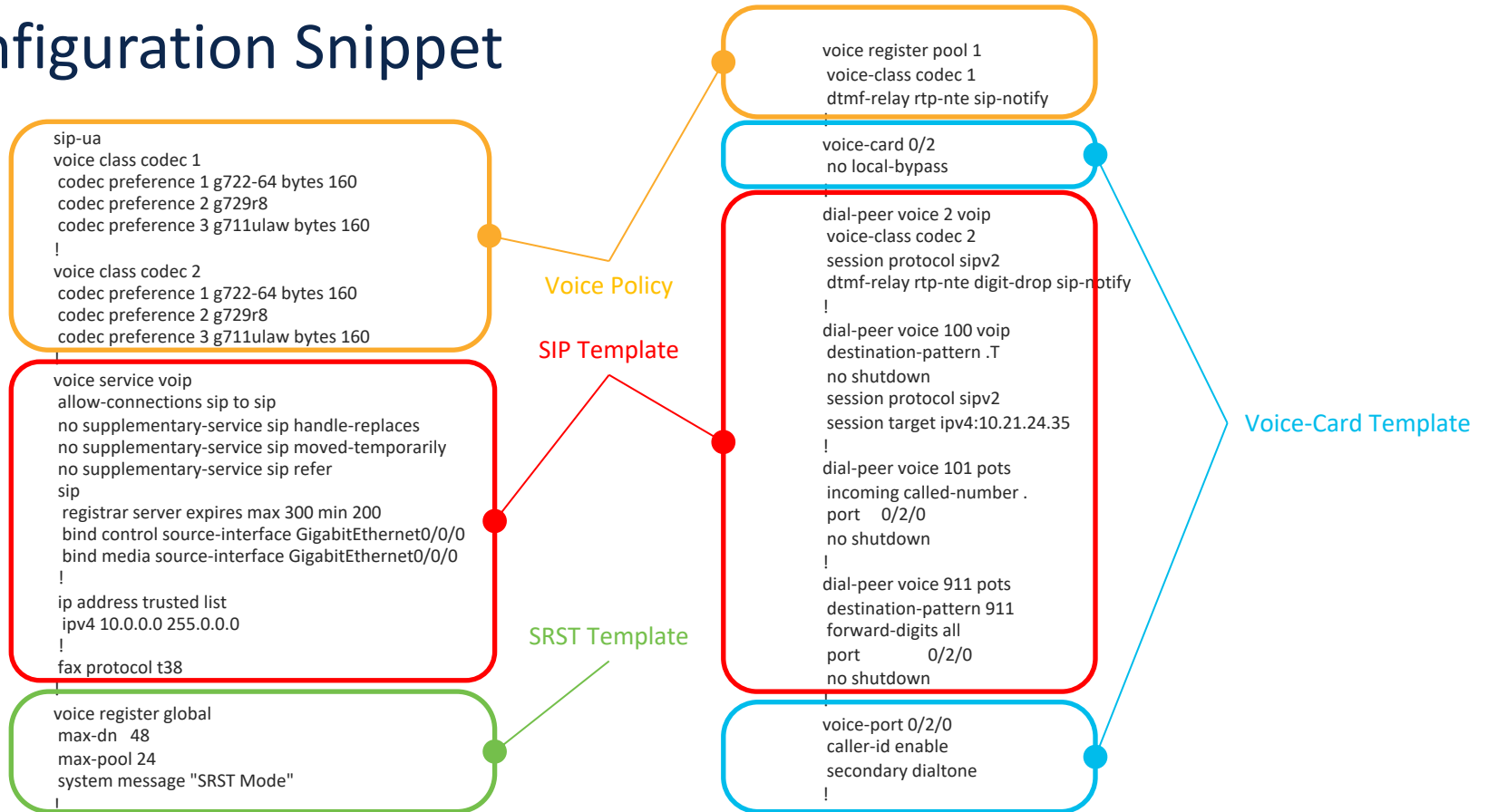


# Voice Template Configuration

Configuration → Templates → Device (Tab) → (Select ISR) → Unified Communications (Section)

The screenshot displays the Cisco vManage interface for configuring voice templates. The top navigation bar shows 'CONFIGURATION | TEMPLATES' with tabs for 'Basic Information', 'Transport & Management VPN', 'Service VPN', 'Cellular', 'Unified Communication', and 'Additional Templates'. The 'Unified Communication' tab is active. A red box highlights the 'Unified Communication' configuration section, which includes four dropdown menus: 'VoiceCard' (TEST-VOICECARD), 'SIP' (TEST-SIP), 'SRST' (TEST-SRST), and 'Voice Policy' (TEST-VOICEPOLICY). A blue 'Mapping' button is located to the right of the 'Voice Policy' dropdown. Below this section is the 'Additional Templates' section, which includes dropdown menus for 'AppQoS' (Factory\_Default\_AppQoS\_ServiceNode\_Tem...), 'Global Template \*' (Factory\_Default\_Global\_CISCO\_Template), 'Cisco Banner' (Choose...), and 'Cisco SIG Credentials' (Choose...).

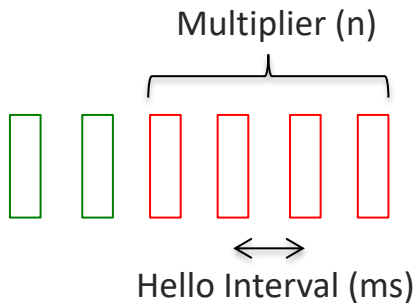
# Configuration Snippet



# Оптимизация унифицированных коммуникаций с помощью SD-WAN

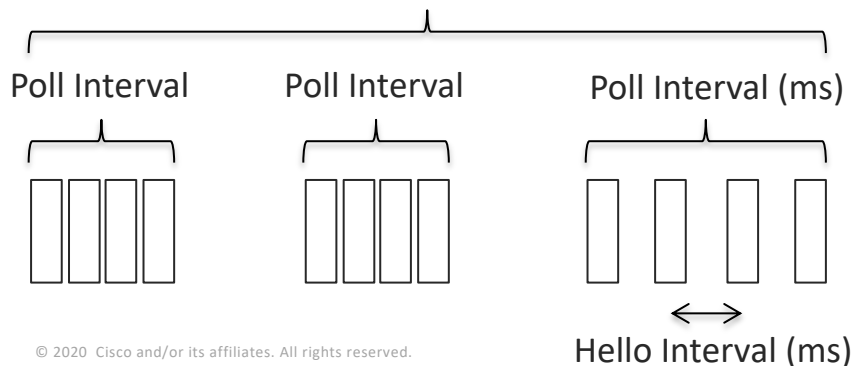
# Bidirectional Forwarding Detection

## Path Quality Detection



Liveliness

Quality



- Each WAN Edge router initiates BFD packet every hello interval
  - Echo mode, no neighbors
  - Tunable to sub-second level
- Poll interval determines the window for calculating path quality
  - Averaged
  - Tunable to sub-second level
- App-route multiplier determines number of poll intervals for establishing overall average path quality
  - Compared against application aware routing thresholds

# Cisco SDWAN Key Features & Cross Architecture Development for Webex

- **Webex Per Flow Type Signatures: Webex Desktop App 39.3+**
  - Video, Audio and High Frame Rate Sharing
  - Additional development needed for Webex Endpoints and MPP Phones
- **SD-Application Visibility and Control**
- **Application Aware Routing**
- **BFD Probes Monitor Transport Health Across SDWAN Fabric**
- **Quality of Service Prioritization**
- **Trackers**

### Webex Desktop App Per Flow Signatures

Type: 0xC003	Length
Flow Type	padding

- **Flow Type** (2 bytes):
  - probe (connectivity check)
  - webex-video
  - webex-audio
  - webex-app-sharing

### NBAR2 Protocol Pack 47

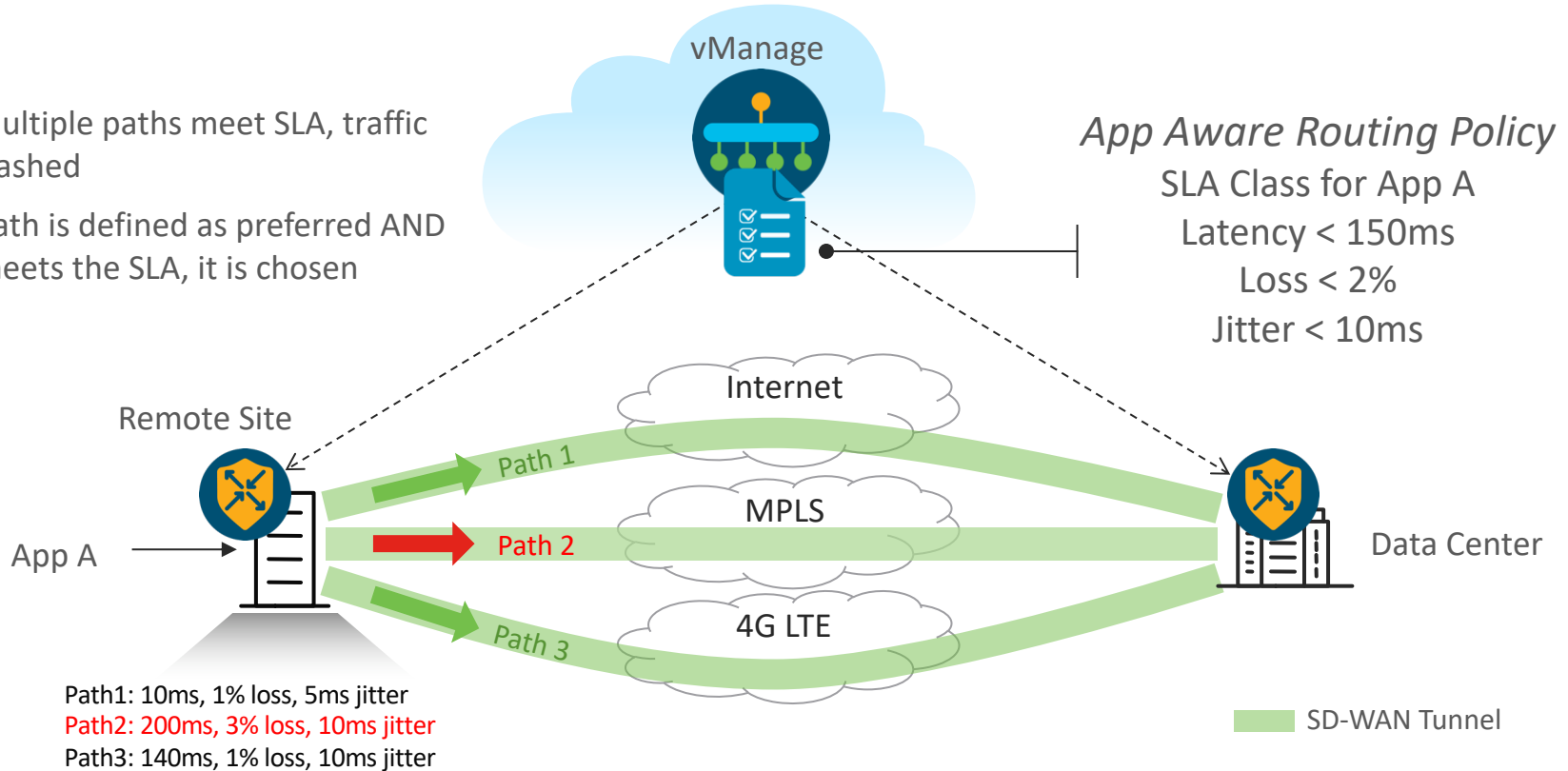
WEBEX-VIDEO	
Name/CLI Keyword	webex-video
Full Name	Webex Video
Description	Webex provides online meeting, web conferencing, and video conferencing services. It supports face-to-face meetings with real-time sharing of data, audio, video, and apps. Webex uses SSL as its underlying protocol.
Reference	<a href="https://www.webex.com/">https://www.webex.com/</a>
Global ID	L7:1250
ID	1250

WEBEX-AUDIO	
Name/CLI Keyword	webex-audio
Full Name	Webex Audio
Description	Webex provides online meeting, web conferencing, and video conferencing services. It supports face-to-face meetings with real-time sharing of data, audio, video, and apps. Webex uses SSL as its underlying protocol.
Reference	<a href="https://www.webex.com/">https://www.webex.com/</a>
Global ID	L7:1251
ID	1251

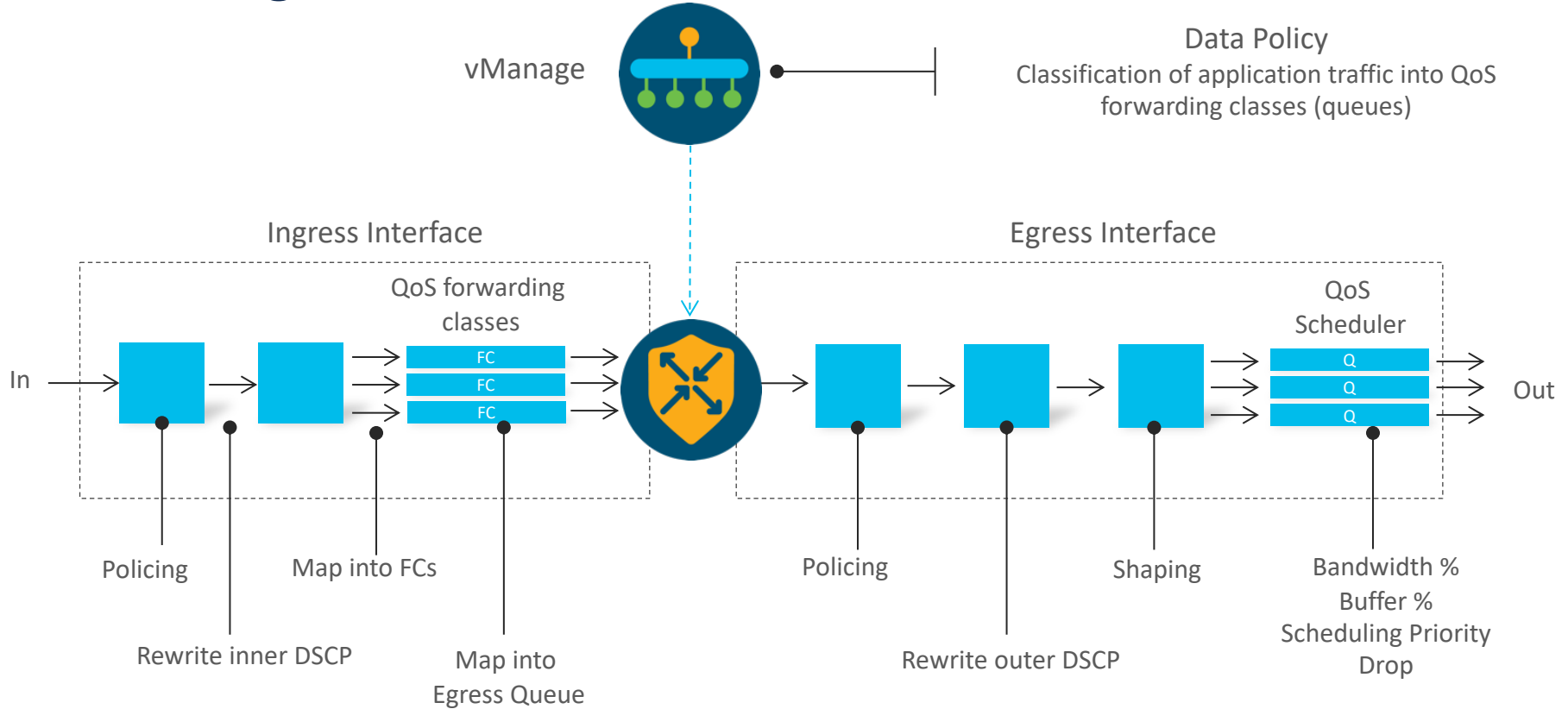
WEBEX-APP-SHARING	
Name/CLI Keyword	webex-app-sharing
Full Name	Webex Application Sharing
Description	WebEx-App-Sharing is granular classification of WebEx protocol application sharing traffic, configured with HTTP-proxy.
Reference	<a href="http://www.webex.com/">http://www.webex.com/</a>
Global ID	L7:546
ID	1480

# Application Aware Routing

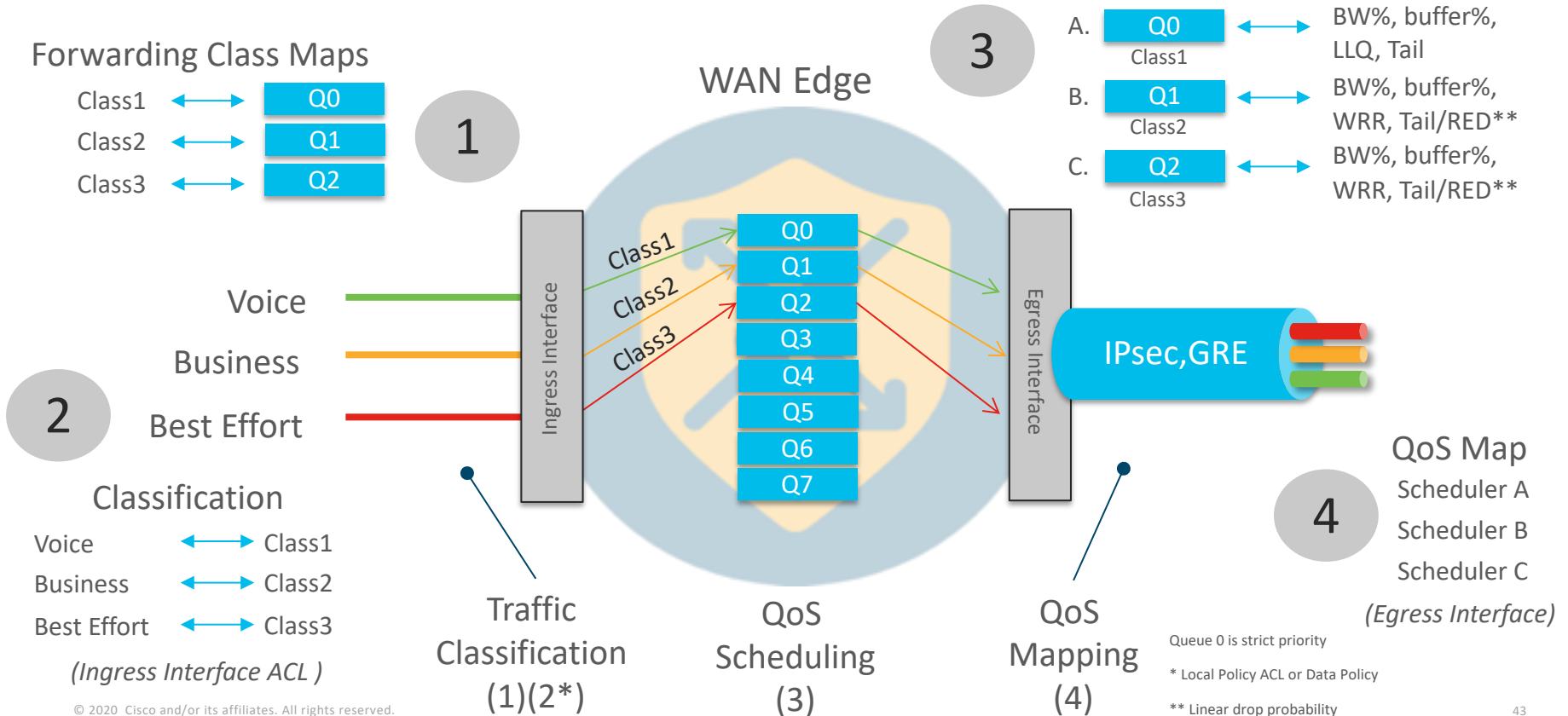
- If multiple paths meet SLA, traffic is hashed
- If path is defined as preferred AND it meets the SLA, it is chosen



# WAN Edge Router QoS Overview

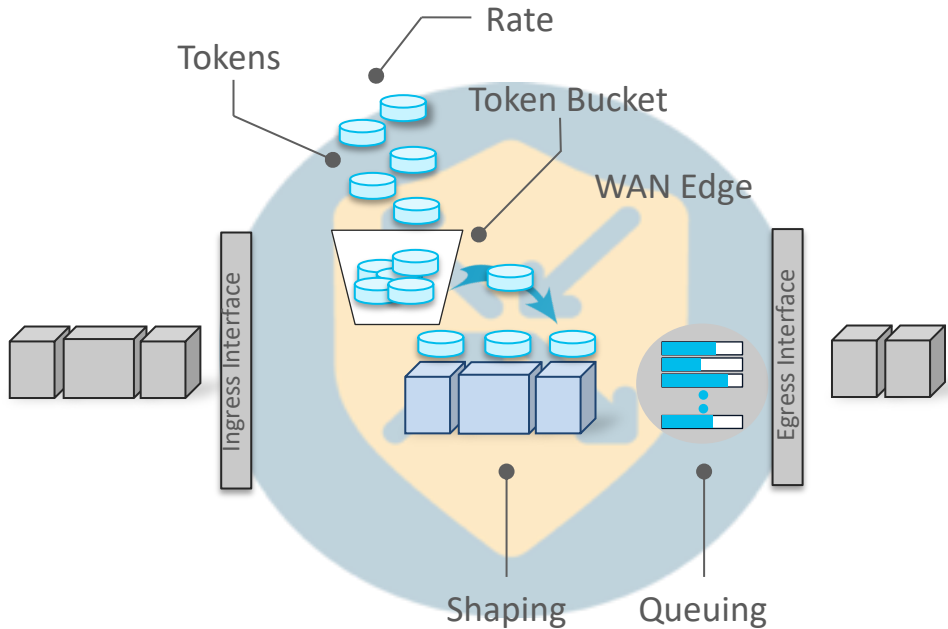


# WAN Edge Router Traffic Prioritization





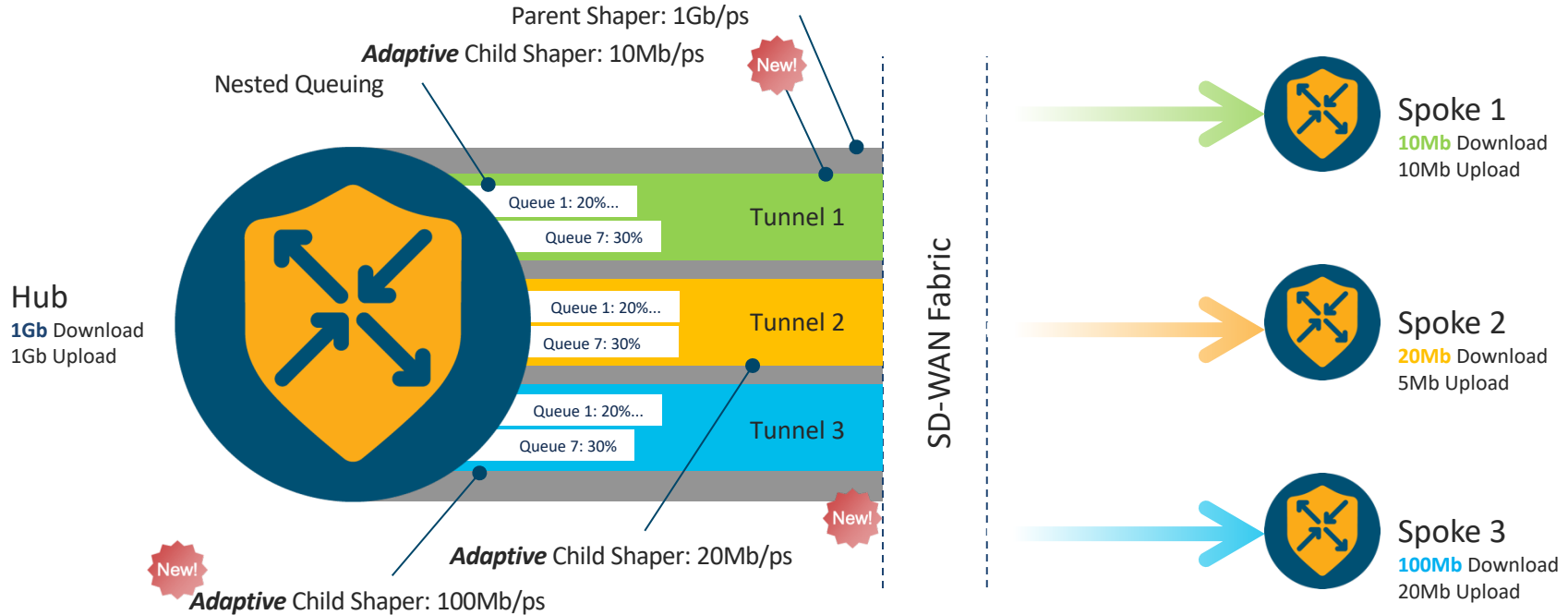
# Shaping



Note: Shaping determines link bandwidth considered for queuing

- Shaping effective on egress physical interfaces
  - Not supported on sub-interfaces
- Forward traffic that conforms to configured shape rate
  - Tokens available in the bucket
- Queue traffic that exceeds configured shape rate
  - Tokens not available in the bucket
- Weighted Round-Robin for queued packets

# Per-Tunnel QoS with Adaptive Shaping



Per-Tunnel QoS allows the Hub site to dynamically adjust the sending rate of its traffic to accommodate lower bandwidth circuits at remote locations. Adaptive shapers measure the **true** circuit capacity at any given moment – rather than relying on static configuration.

# Работа на плохих каналах

**Problem:** transactional data over WAN links, which has few percent packet loss (up to 10-20%).

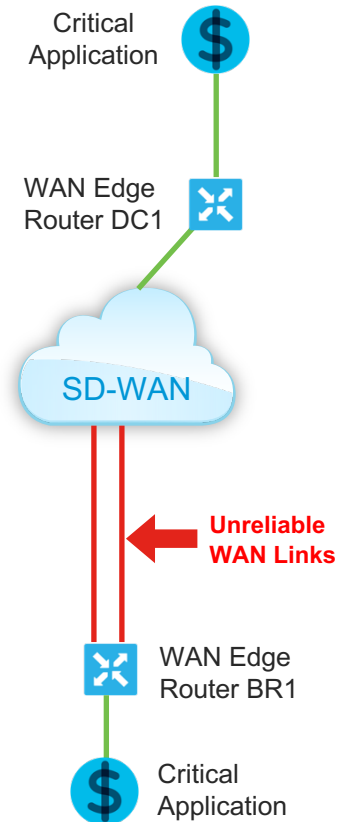
**Main Goal:** 0 packet loss on the application level.

**Solution 1:** **Forward Error Correction (FEC)** send additional parity packet for every 4 data packets, which will be used by the receiving router to reconstruct one lost packet.

**Solution 2:** **Packet Duplication** will duplicate packets for critical apps over both WAN links.



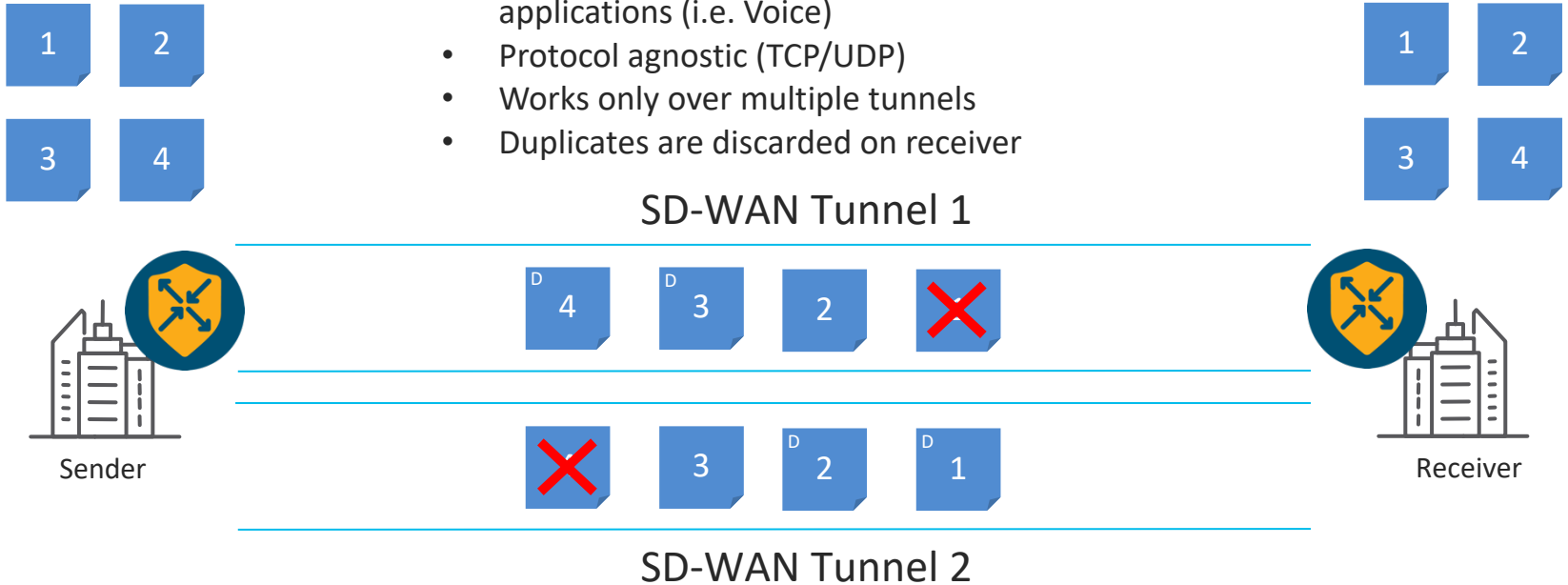
**New in RLS 16.12**



# Packet Duplication

## Highlights:

- Protects against packet loss for critical applications (i.e. Voice)
- Protocol agnostic (TCP/UDP)
- Works only over multiple tunnels
- Duplicates are discarded on receiver



# Dynamic On-Demand Tunnels

## Problem

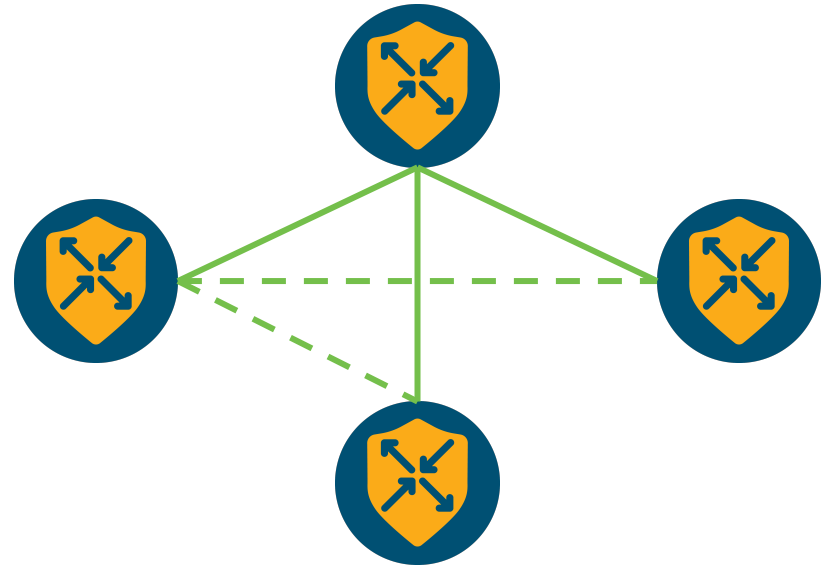
By default, Cisco SD-WAN operates in full-mesh. While topology modification is possible, full-mesh carries a huge computational burden on branch resources and, therefore, becomes difficult to scale. Enterprise customers need full-mesh connectivity, but also need a way to offset the resource burden that full-mesh generally entails.

## Solution

SD-WAN v20.3 / 17.3 now support Dynamic On-Demand Tunneling. Branch routers will maintain an “always-on” tunnel to a hub location, then dynamically build site-to-site tunnels, where necessary.

## Caveats / Prerequisites

Spoke locations must receive TLOC and vRoute of remote, must have backup path and Service TE set (see supporting slides)



— Backup Path  
- - - On-Demand Path

# Route Leaking

## Problem

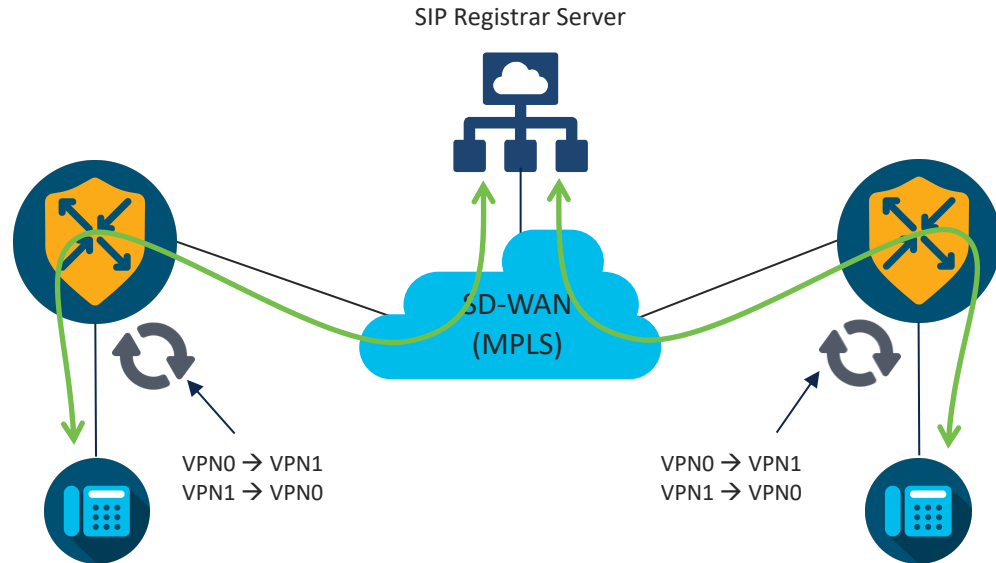
Many customers have expressed a need to expose underlay services within the SD-WAN overlay (such as hosted PBX/VoIP being made available to phones that exist in a Service VPN/VRF). At present (v17.2 / 20.1), SD-WAN only supports this type of route leaking between Service VPNs.

## Solution

Cisco SD-WAN v20.3 / 17.3 now support route leaking between Service VPNs and the Transport VPN.

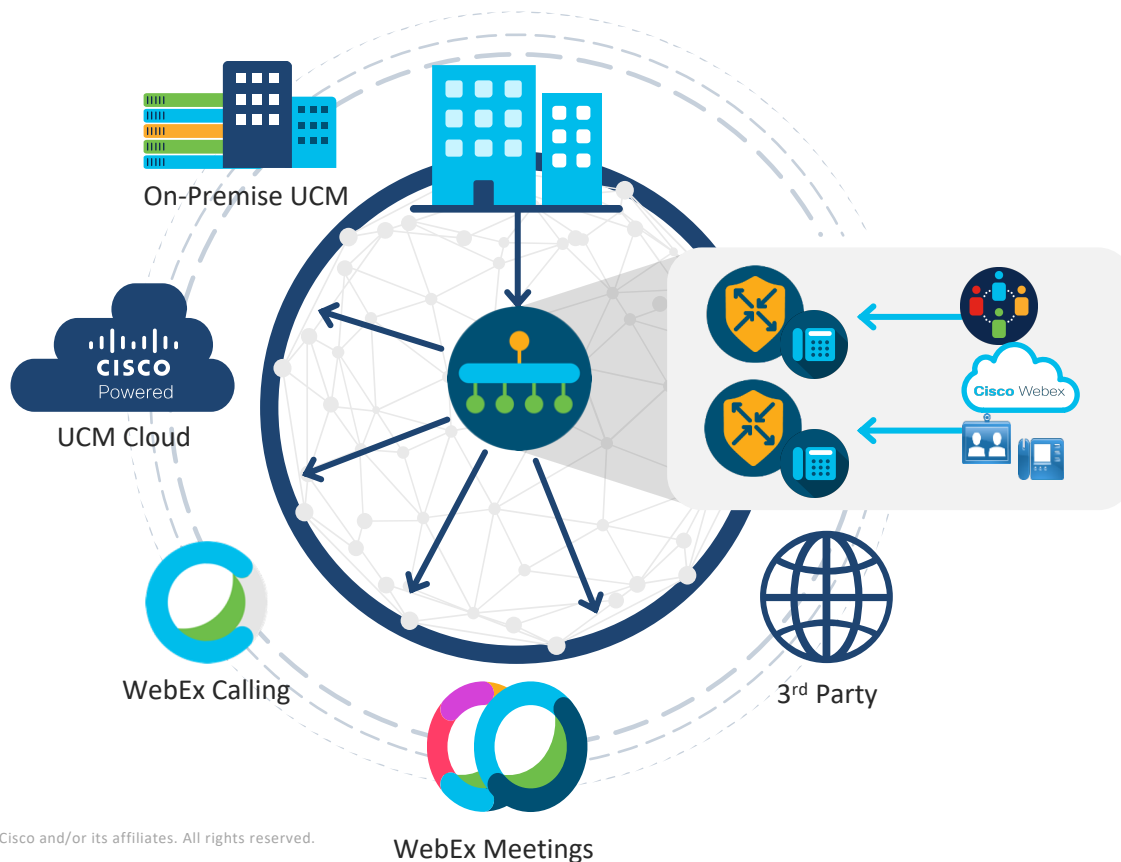
## Caveats / Prerequisites

IPv4 only, no EIGRP support, SSNAT + Route Leak is not supported, VPN0 cannot be transit VPN, cEdge has additional restrictions (see supporting TDM)



DEMO

# Ключевые выводы



## Flexible Connectivity

Directly connect Cloud or On-Premise call control and Webex Meetings services with WAN optimization improving the user experience

## Large Scale VoIP Provisioning

Leverage the power of vManage Templating and Policy orchestration to provision scalable, consistent UC across the enterprise

## Hardware Consolidation

Reduce CapEx and OpEx by consolidating UC and SD-WAN into a single CPE



Спасибо!



The bridge to possible