Cisco Small Business Webinar for Partners

- Technical Track

SBCS Multi-Site for Voice, Data & Video

How to deploy UC520 sites as a Multi Site

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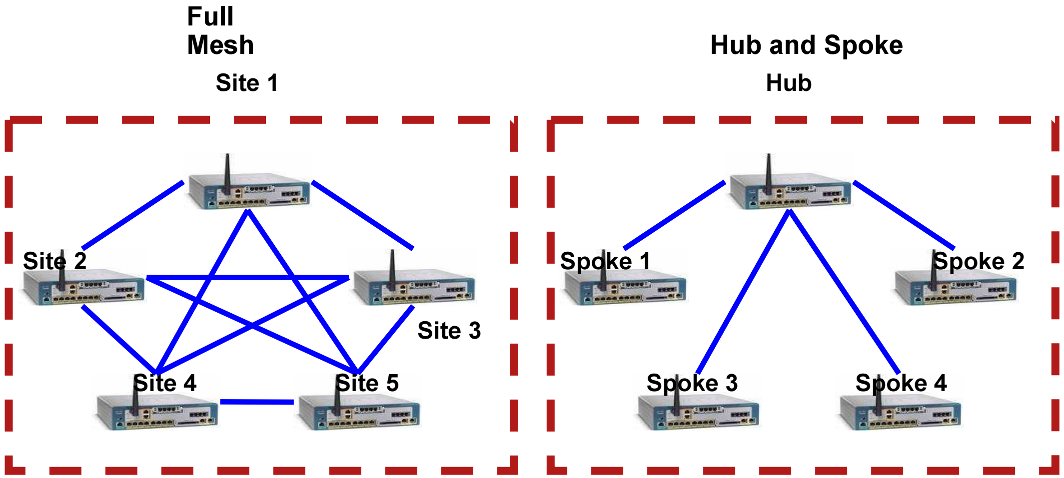
SBCS 1.3 Definition (September FCS)

SBCS	**1.3**
UC500 Software Pack	**UC520-4.2.9.zip**
IOS Image	**12.4(11)XW9** **CME 4.2**
CUE Image	**CUE 3.2**
Phone Loads	**Bug Fixes + new phones**
Default Config	**Bug Fixes + new config**
CCA*	**1.8**

* CCA is not part of UC500 Software Pack but downloadable from CCO free

Two Modes of Multisite

- We will be demonstrating full mesh today, but there are 2 types supported.



	Pros	Cons
Full Mesh	Redundancy/Alternate Paths	Less VPN sessions for Remote Access (EZVPN) More complex configuration and routing
Hub & Spoke	More VPN sessions available for Remote Users Simpler configuration	Bandwidth requirements at the Hub site Single point of failure

Design Guidelines for Typical Multisite Configuration

Site Information	Description/Notes
Number of sites	Maximum of 5 sites
Phones / Users per site	No limit other than UC520 SKU (includes all phones and users). Suggest each site have unique extensions (i.e. 200-220, 221-240,…)
WAN Connectivity (Voice and Data)	Broadband connection required (DSL, Cable, T1, MetroE etc.) at each site supporting VoIP CODEC bandwidth. Each site can have its own SIP Trunk or Legacy telephony interface (FXO, BRI, PRI)
Dial Plan Design	Each site has its own independent dial plan & call control with translation rules and dial peers (OOB CLI) to reach other site.
Call Control	Each site is controlled by its own CME (10.1.1.1) and CUE (10.1.10.1), which maximizes phone users counts and voice mail size.
VPN for Data	Site to Site IPsec VPN with direct encapsulation. Coexists with remote Teleworker EZVPN connections.
Data VLAN subnet	Each site must have a 'unique' data VLAN subnet (192.168.10.0. 192.168.20.0,…)
Site-Site Voice Ext. Dialing	VoIP (SIP or H323 w/ video) Site Index unique for each site (i.e. 81xxx, 82xxx)
Maximum inter-site calls	Per site CAC & CODEC preference configurable based on available bandwidth and design guidelines of Transcoding resources.

Multisite Feature Operation Supported

Features	Supported
Call Transfer	Yes
Conference	Yes
Conference Calls	Yes
Video Calls between sites	Yes (with H323) (384Kbps upstream bandwidth will be required per video call - if bandwidth is not available call will failover to voice only)
Paging & Call Park across sites	Yes & Yes (call pick up can only be accomplished from the site the call was parked in).
Auto Attendant/Voice Mail	Yes/Yes Each site has its own CUE AA and each site can access the others AA and Voice Mail
Fax between sites	Yes
Hunt Groups across sites	Yes* Hunt groups can be accessed site to site, but not mixed members within the hunt group in current SW release.
Shared Directory across sites	No

NOTE With 4.2.9, hunt groups are accessible from each site to the other, In EA package, hunting membership could work across sites using voice hunt cli.

SBCS Multi Site Data

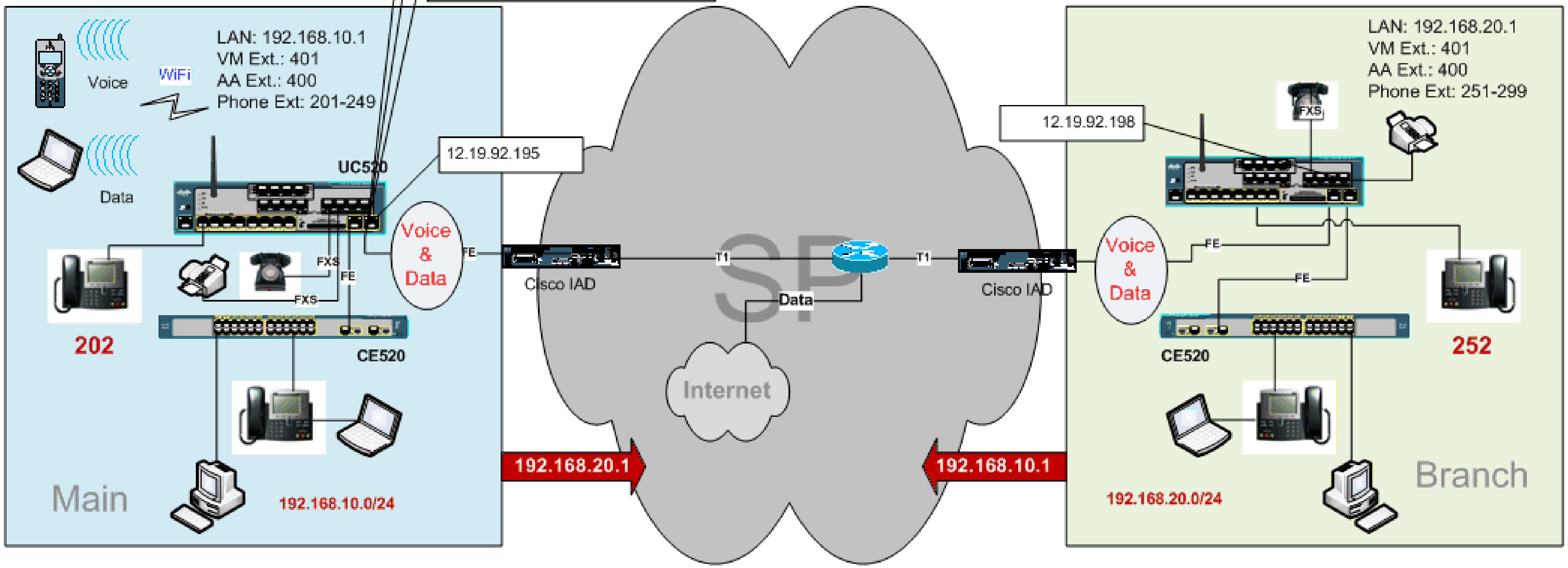
```
crypto isakmp policy 1
 encr 3des
 hash md5
 authentication pre-share
 group 2

crypto isakmp key sbcs address 12.19.92.198

crypto isakmp keepalive 300 periodic

crypto ipsec transform-set MS esp-3des esp-md5-
hmac

crypto map multisite 1 ipsec-isakmp
 set peer 12.19.92.198
 set transform-set MS
 match address 199

ip route 192.168.20.0 255.255.255.0 FastEthernet0/0
```

```
ip nat inside source route-map SDM_RMAP_1 interface FastEthernet0/0 overload
access-list 107 deny   ip 192.168.10.0 0.0.0.255 192.168.20.0 0.0.0.255
access-list 107 permit ip 192.168.10.0 0.0.0.255 any
access-list 107 permit ip 10.1.1.0 0.0.0.255 any
access-list 107 permit ip 10.1.10.0 0.0.0.3 any
!
route-map SDM_RMAP_1 permit 1
 match ip address 107
```

```
access-list 199 permit ip 192.168.10.0 0.0.0.255 192.168.20.0 0.0.0.255
```

```
Interface FE0/0
 crypto map multisite
 access-group 104 in
 ip nat outside
 ip inspect SDM_LOW out
```

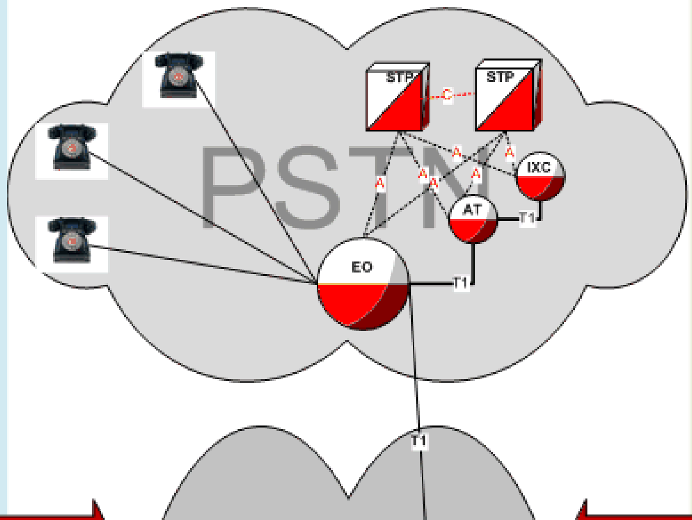
```
crypto isakmp policy 1
 encr 3des
 hash md5
 authentication pre-share
 group 2

crypto isakmp key sbcs address 12.19.92.195

crypto isakmp keepalive 300 periodic

crypto ipsec transform-set MS esp-3des esp-md5-hmac

crypto map multisite 1 ipsec-isakmp
 set peer 12.19.92.195
 set transform-set MS
 match address 199

ip route 192.168.10.0 255.255.255.0 FastEthernet0/0
```

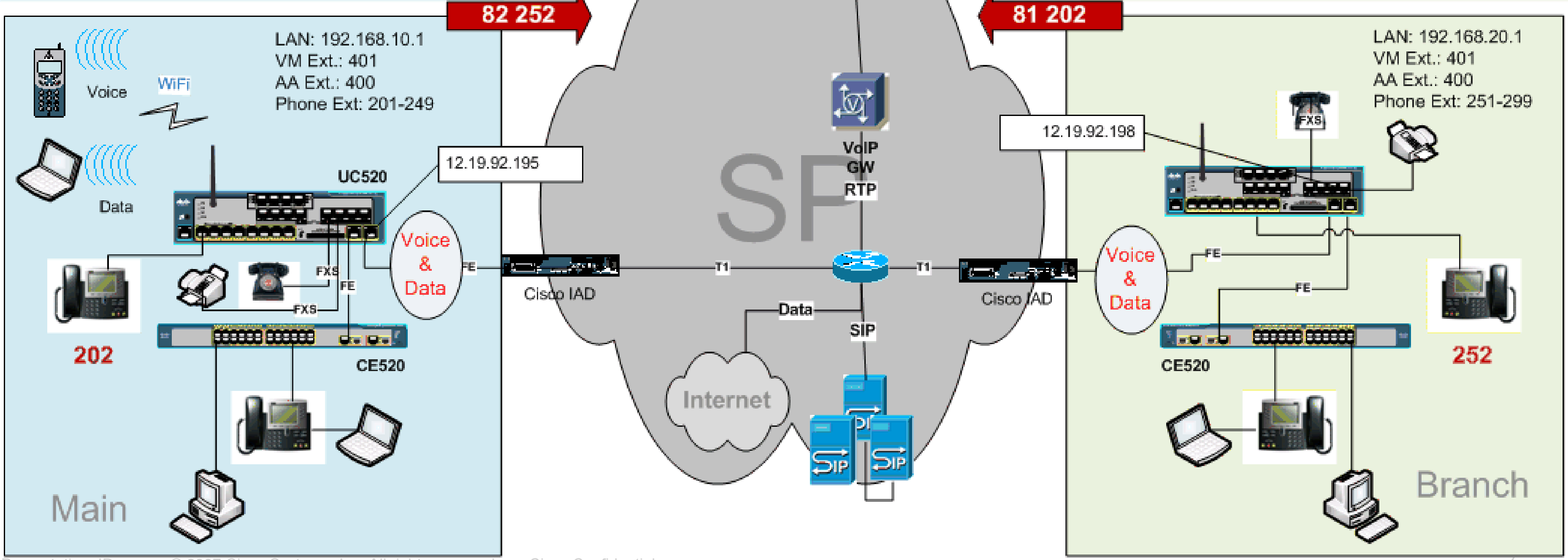


SBCS Multi Site Voice w/o Video (SIP)

```
voice translation-rule 81 ← intersite dial strings to extensions
 rule 1 /^81\(...\)$/ /\1/
!
voice translation-profile main
 translate called 81
!
dial-peer voice 82000 voip ← Intersite Extension Dialing to Branch
 destination-pattern 82...
 session protocol sipv2
 session target ipv4:12.19.92.198
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice 81000 voip ←Intersite Extension Dialing from branch
 translation-profile incoming main
 session protocol sipv2
 session target sip-server
 incoming called-number 81...
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
access-list 104 permit ip 12.19.92.198 0.0.0.255 any <-FE0/0 in
!
voice service voip
  allow-connections sip to sip
```

```
voice translation-rule 82
 rule 1 /^82\(...\)$/ /\1/
!
voice translation-profile main
 translate called 82
!
dial-peer voice 81000 voip ← Intersite Extension Dialing to Main
 destination-pattern 81...
 session protocol sipv2
 session target ipv4:12.19.92.195
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
dial-peer voice 82000 voip ←Intersite Extension Dialing from main
 translation-profile incoming main
 session protocol sipv2
 session target sip-server
 incoming called-number 82...
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
access-list 104 permit ip 12.19.92.195 0.0.0.255 any ← FE0/0 in
!
voice service voip
  allow-connections sip to sip
```





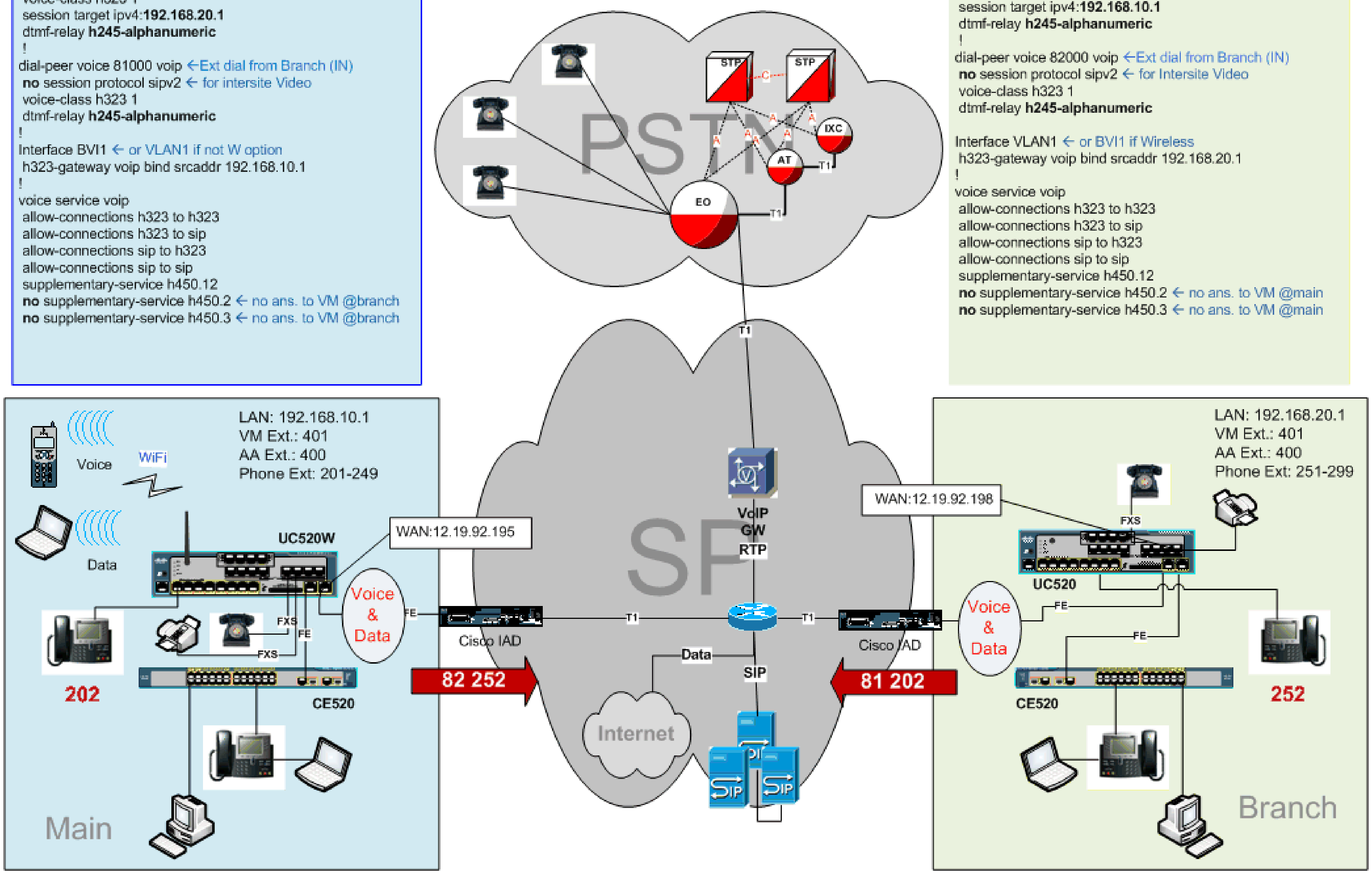
SBCS Multi Site Voice w/ Video (H.323)

Changes and additions shown

```
voice class h323 1 ←For intersite Video
call start slow
!
dial-peer voice 82000 voip ←Ext dial to Branch (OUT)
 no session protocol sipv2 ← for intersite Video
 voice-class h323 1
 session target ipv4:192.168.20.1
 dtmf-relay h245-alphanumeric
!
dial-peer voice 81000 voip ←Ext dial from Branch (IN)
 no session protocol sipv2 ← for intersite Video
 voice-class h323 1
 dtmf-relay h245-alphanumeric
!
Interface BVI1 ← or VLAN1 if not W option
 h323-gateway voip bind srcaddr 192.168.10.1
!
voice service voip
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 supplementary-service h450.12
 no supplementary-service h450.2 ← no ans. to VM @branch
 no supplementary-service h450.3 ← no ans. to VM @branch
```

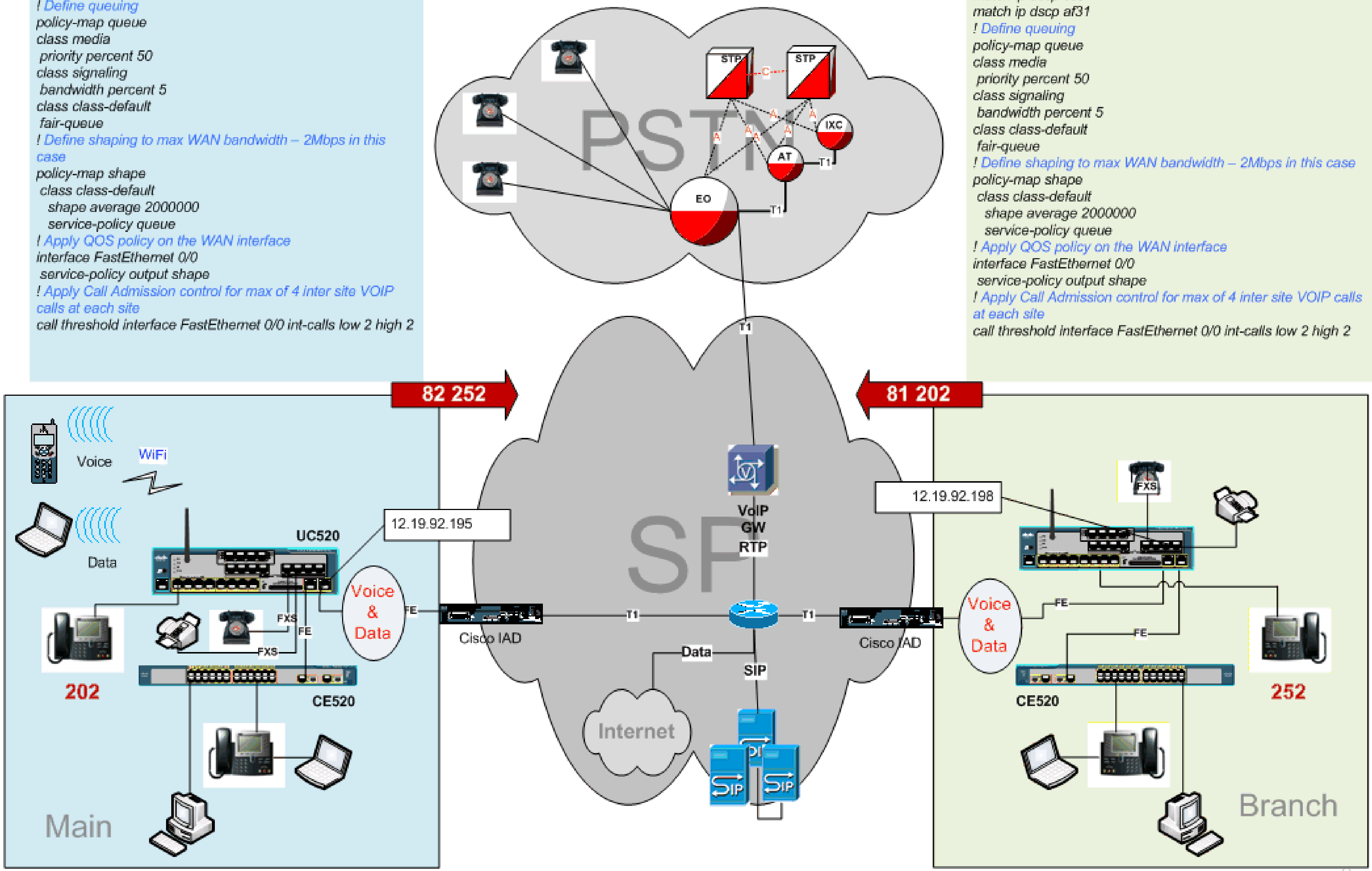
Changes and additions shown

```
voice class h323 1 ←For intersite Video
call start slow
!
dial-peer voice 81000 voip ←Ext dial to Branch (OUT)
 no session protocol sipv2 ← for Intersite Video
 voice-class h323 1
 session target ipv4:192.168.10.1
 dtmf-relay h245-alphanumeric
!
dial-peer voice 82000 voip ←Ext dial from Branch (IN)
 no session protocol sipv2 ← for Intersite Video
 voice-class h323 1
 dtmf-relay h245-alphanumeric

Interface VLAN1 ← or BVI1 if Wireless
 h323-gateway voip bind srcaddr 192.168.20.1
!
voice service voip
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 supplementary-service h450.12
 no supplementary-service h450.2 ← no ans. to VM @main
 no supplementary-service h450.3 ← no ans. to VM @main
```



SBCS Multi Site QoS & CAC

```
! Classify IP traffic
class-map match-any media
match ip dscp ef
!
class-map match-any signaling
match ip dscp cs3
match ip dscp af31
! Define queuing
policy-map queue
class media
 priority percent 50
class signaling
 bandwidth percent 5
class class-default
 fair-queue
! Define shaping to max WAN bandwidth – 2Mbps in this case
policy-map shape
 class class-default
   shape average 2000000
   service-policy queue
! Apply QOS policy on the WAN interface
interface FastEthernet 0/0
 service-policy output shape
! Apply Call Admission control for max of 4 inter site VOIP calls at each site
call threshold interface FastEthernet 0/0 int-calls low 2 high 2
```




```
! Classify IP traffic
class-map match-any media
match ip dscp ef
!
class-map match-any signaling
match ip dscp cs3
match ip dscp af31
! Define queuing
policy-map queue
class media
 priority percent 50
class signaling
 bandwidth percent 5
class class-default
 fair-queue
! Define shaping to max WAN bandwidth – 2Mbps in this case
policy-map shape
 class class-default
   shape average 2000000
   service-policy queue
! Apply QOS policy on the WAN interface
interface FastEthernet 0/0
 service-policy output shape
! Apply Call Admission control for max of 4 inter site VOIP calls at each site
call threshold interface FastEthernet 0/0 int-calls low 2 high 2
```

