



Encryption keys

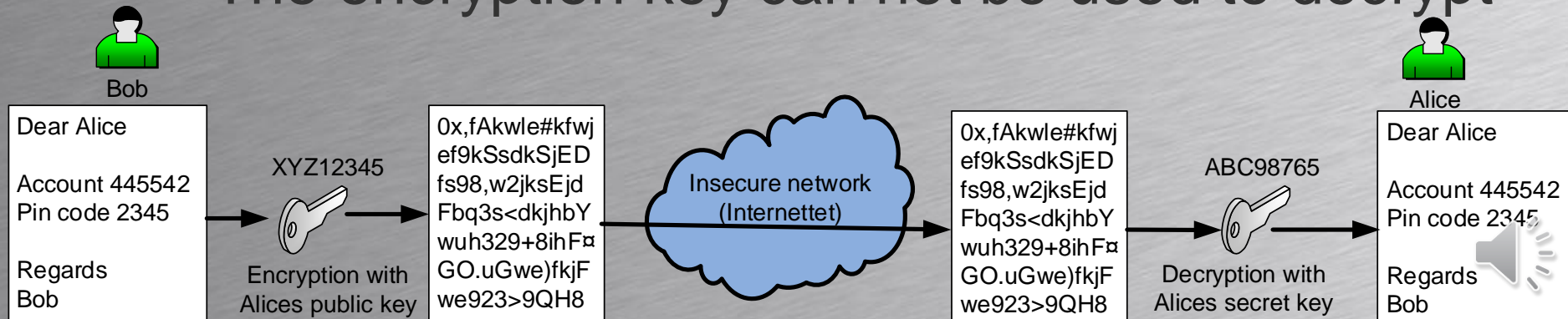
- Symmetrical keys
 - Same key used for encryption and decryption
 - Exchange of symmetrical keys between parties difficult without risk of interception
- Asymmetrical keys
 - One key for encryption and another for decryption - called a key pair.
 - Encryption key can not be used to decrypt
 - Exchange of encryption key without risk





Asymmetrical keys

- Alices computer generates a key pair
 - A public key: XYZ123345 (Used to encrypt)
 - A secret key: ABC98765 (Used to decrypt)
- Alice transmit her public key to Bob
- Bob uses Alices public key to encrypt
- If a hacker intercept the messages
 - The encryption key can not be used to decrypt





MPLS VPN

Multi Protocol Label Switching

- From a ISP's MPLS brochure
 - The customers locations are connected together in a closed private network
 - Transport via the Internet in a closed group
 - Internet access not possible through MPLS
 - Speeds from 512 Kbps to 1 Gbps
 - Existing customer IP address plan preserved
 - Normally private IP addresses are used by customers
 - 10.0.0.0/8
 - 172.16.0.0/12
 - 192.168.0.0/16

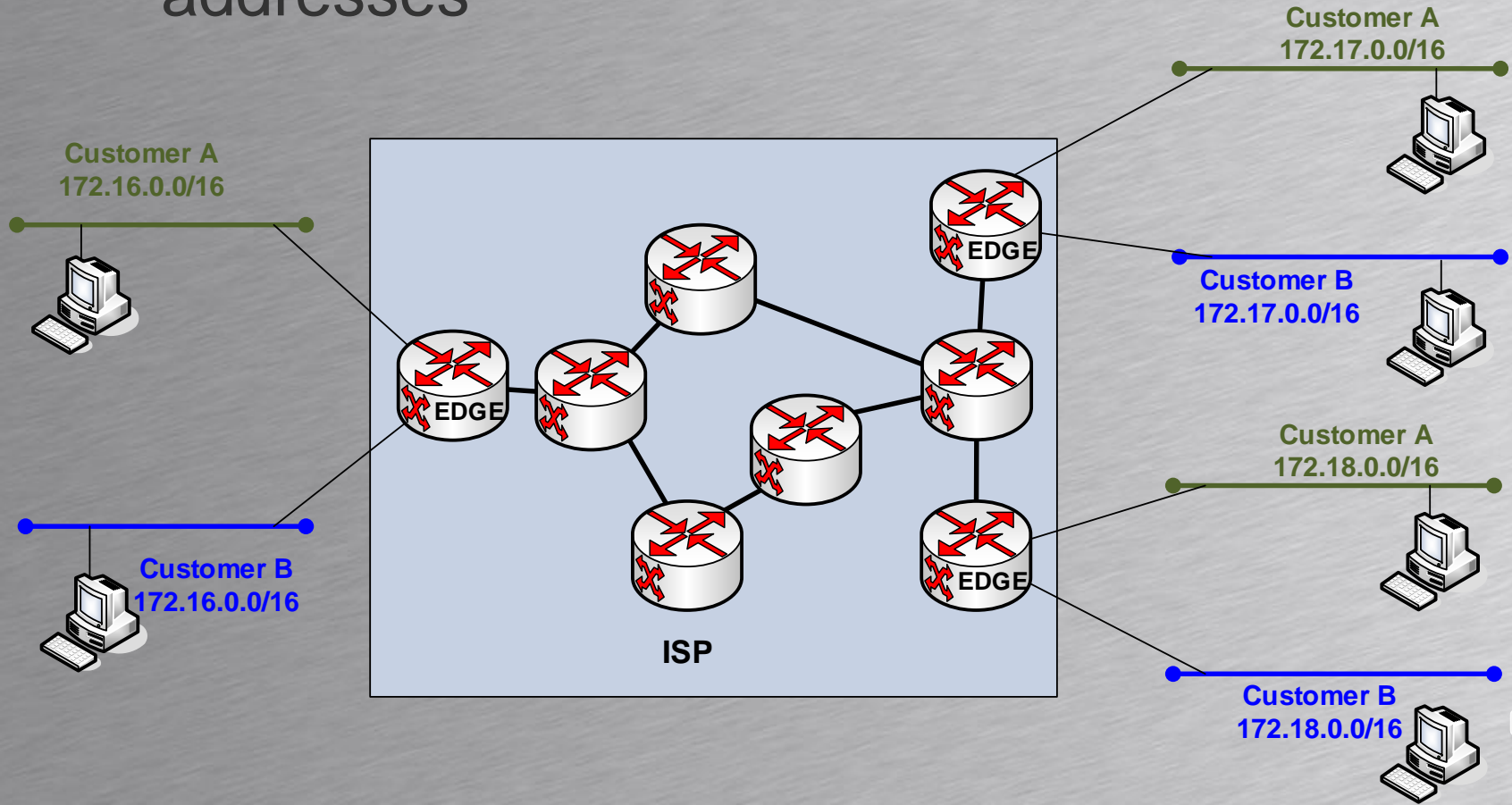




MPLS VPN

Multi Protocol Label Switching

- Physical network as seen from the ISP
 - Both customers “accidentally” uses same IP addresses

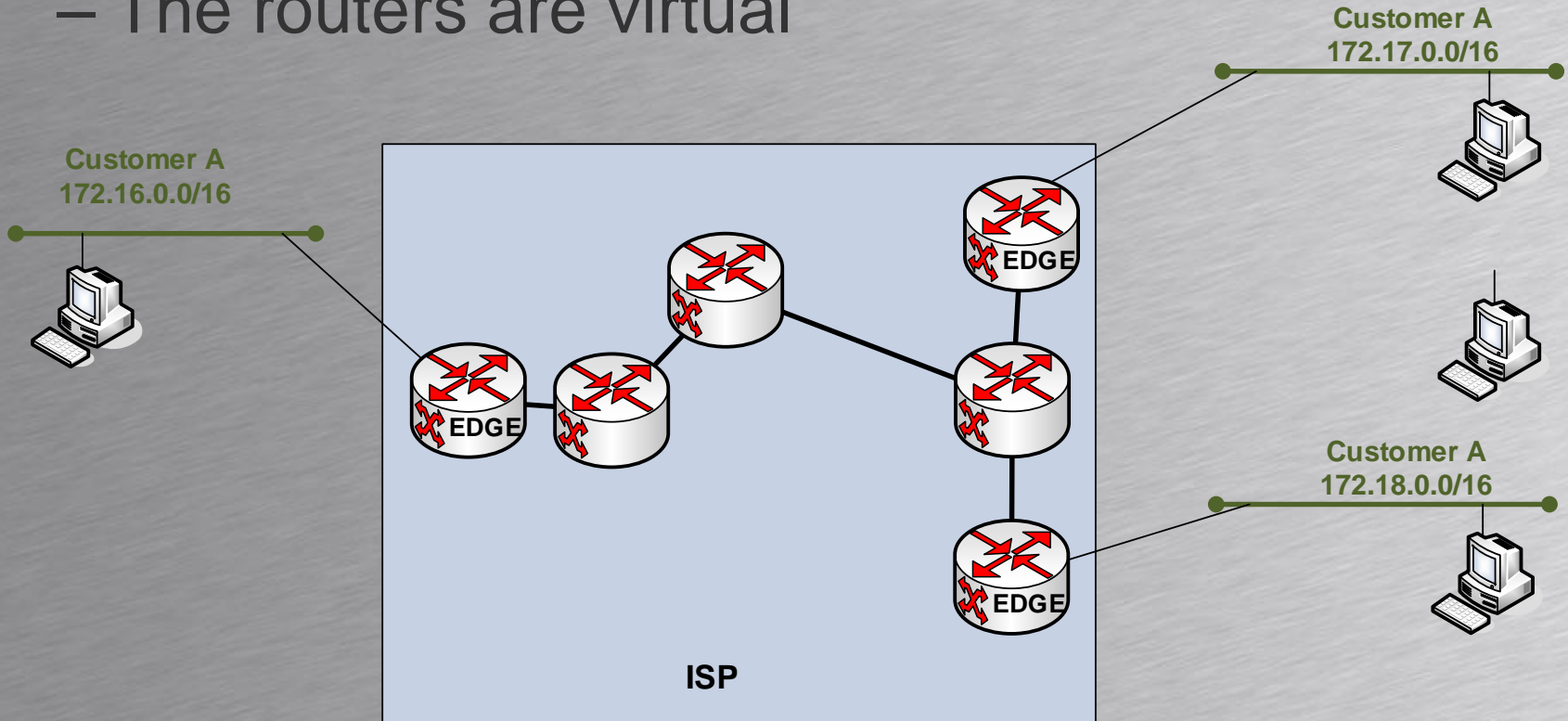




MPLS

Multi Protocol Label Switching

- Physical network as seen from Customer A
 - Customer A sees “his own network”
 - The routers are virtual

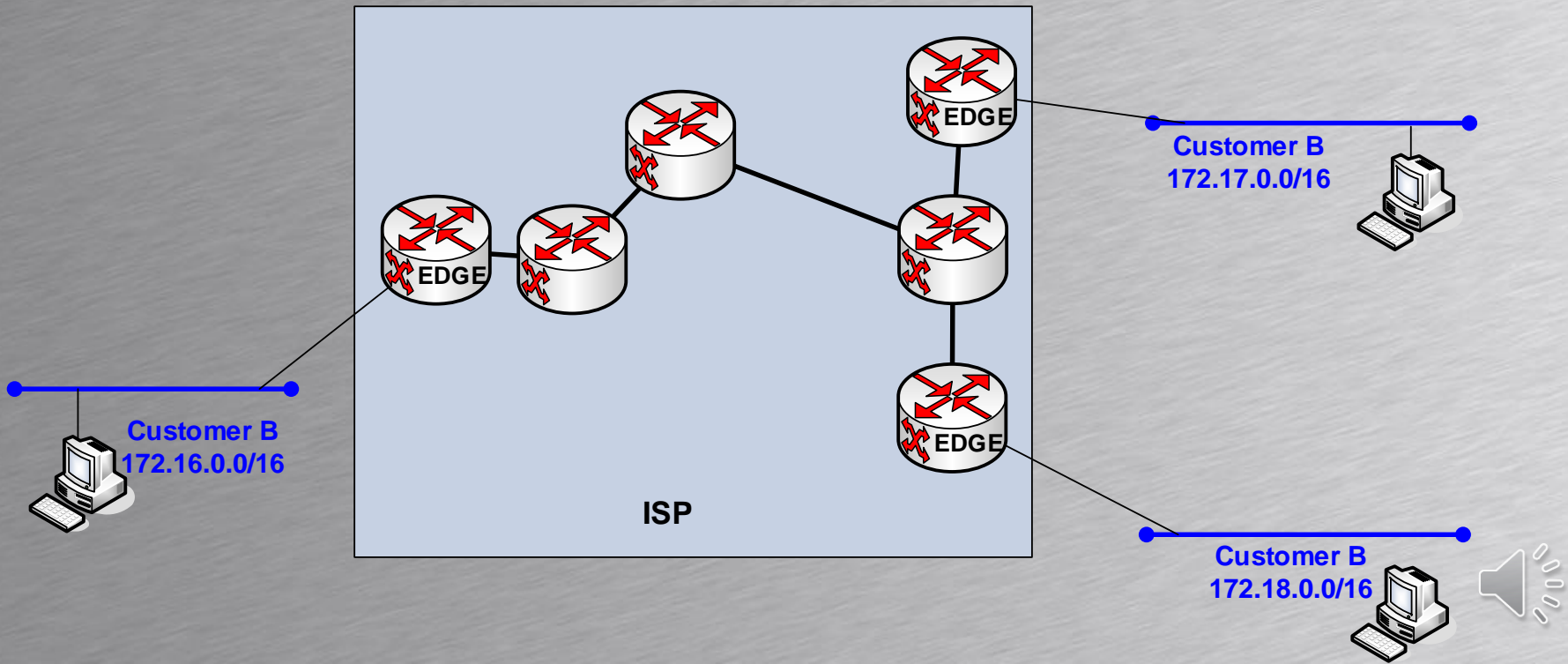




MPLS VPN

Multi Protocol Label Switching

- Physical network as seen from Customer B
 - Customer B sees “his own network”
 - The routers are virtual





VPLS

Virtual Private Lan Service

- VPLS is another VPN type using MPLS technology
- MPLS VPN is a routed VPN (OSI layer 3)
 - Each customer site having different IP networks
 - Virtual Routers
- VPLS VPN is switched VPN (OSI layer 2)
 - Each customer site have different MAC addresses



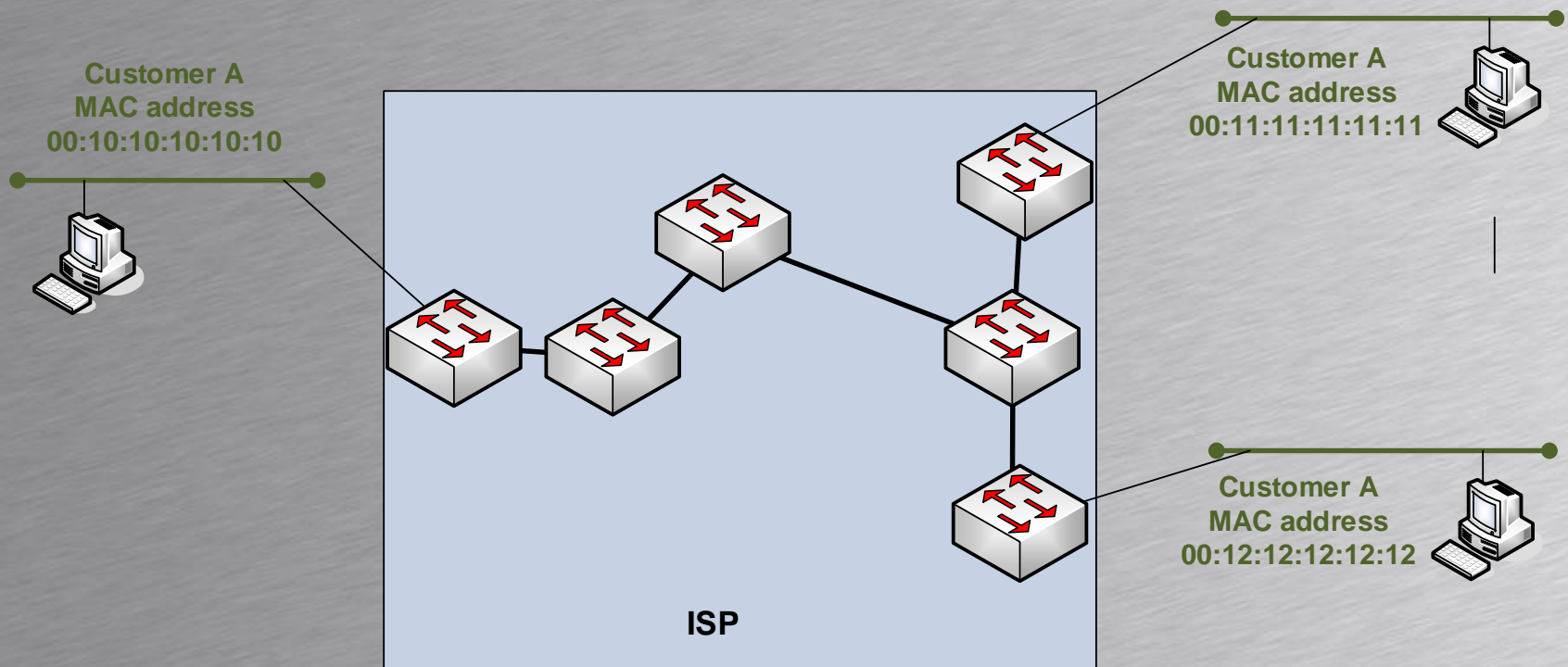


VPLS

Virtual Private Lan Service



- Physical network as seen from Customer A
 - Switching between remote sites





IP ToS to IP DiffServ

	Class 0	Class 1	Class 2	Class 3	Class 4	Class 5	Class 6	Class 7
Class Selector	000000	001000 (CS1)	010000 (CS2)	011000 (CS3)	100000 (CS4)	101000 (CS5)	110000 (CS6)	111000 (CS7)
Assured Forwarding Low Drop Precedence								
Assured Forwarding Medium Drop Precedence								
Assured Forwarding High Drop Precedence								
Expedited Forwarding						(EF) IP voice		

If a router or switch experience congestion it will start to drop packets in configured classes.

Within each class it will drop packets according to drop preference.

High drop preference = high probability the packet is dropped

Classes can be allocated to different drop preferences

...
 Class 0: 00₂ = 0 = lowest drop preference
 ...
 Class 3: 11₂ = 3 = highest drop preference





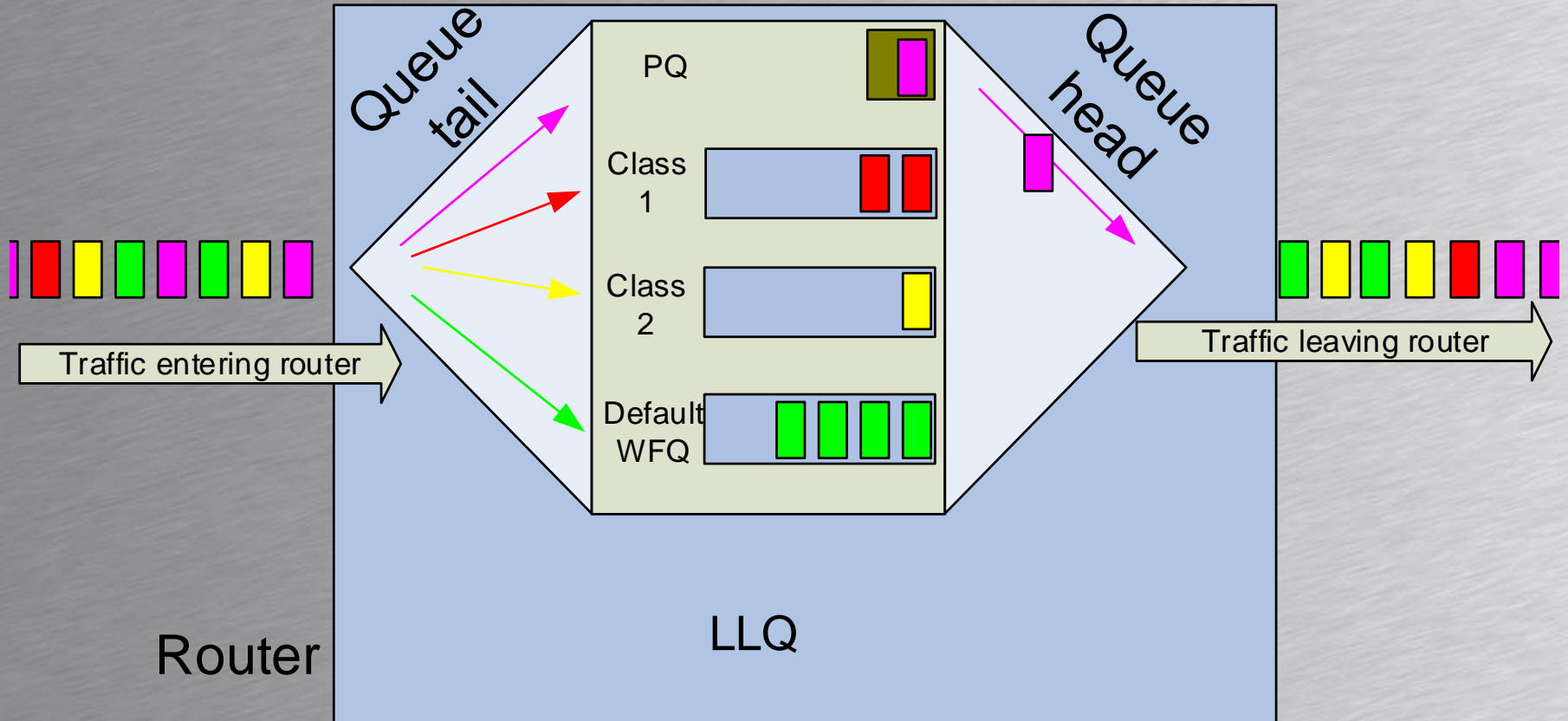
LLQ: Low Latency Queuing

- LLQ takes the best from priority queuing, round robin and weighted fair queuing giving
 - 1 priority queue used for VoIP
 - Up to 256 round robin queues
 - Weighted fair queuing for traffic not classified





LLQ: Low latency queuing





VoIP SECURITY



Encryption of voice and signaling



SIP Security

- SIP Register authentication vulnerability
 - A SIP phone registers with its proxy using username and password
 - If the username and password are transmitted in clear text, identity theft is possible
- SIP register authentication security
 - The server sends a ‘nonce’ to the client
 - A nonce is a random number
 - The client adds the nonce to the password and calculate a hash value returned to the server



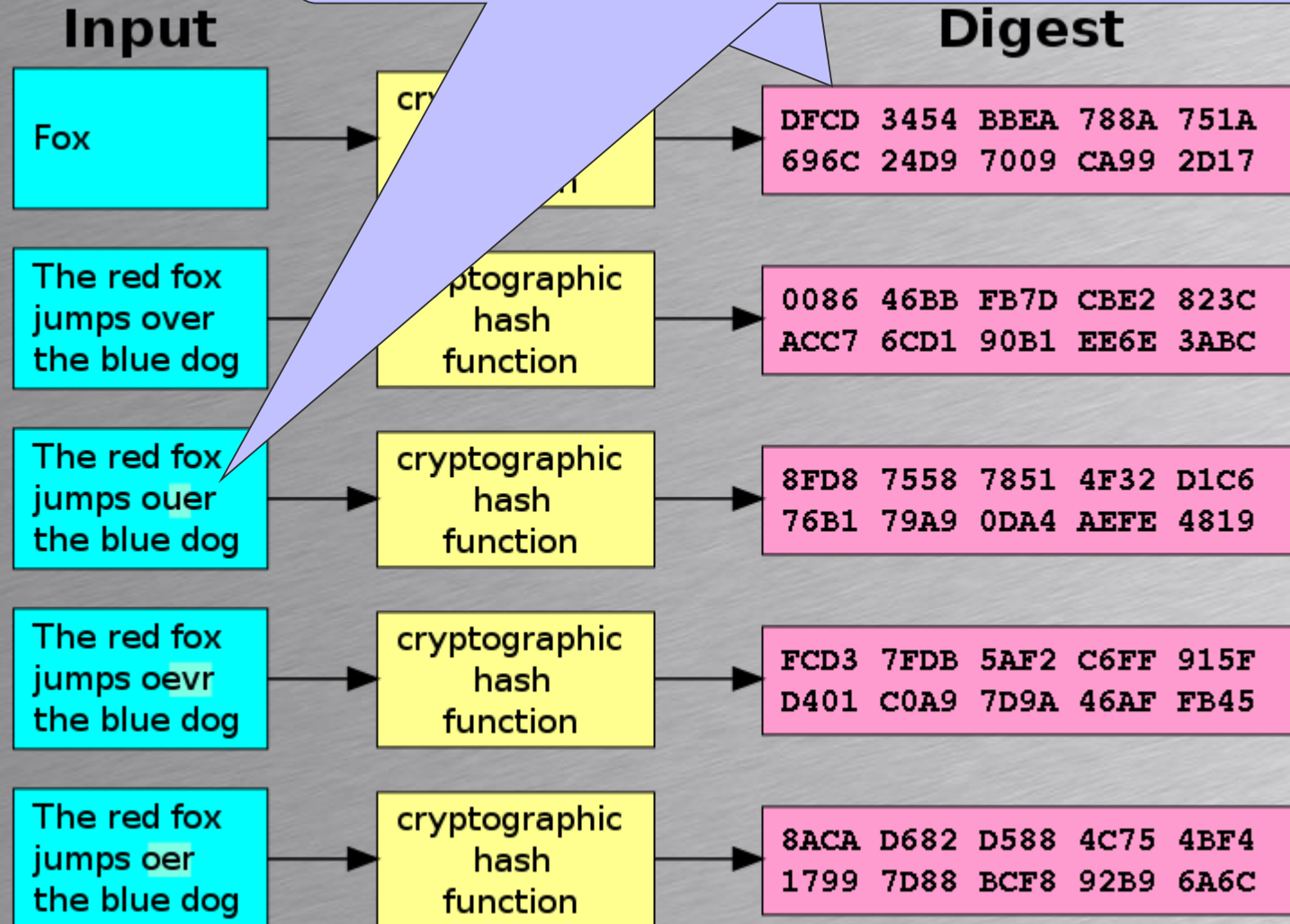
hash

- A hash is a mathematical function
- Maps variable length data to fixed length data
- Used to protect passwords
- MD5 is presently the most used hash function
 - MD5 hash is considered compromised
 - Other hashes such as SHA-1, SHA-2 and SHA-3 are more secure. SHA-3 the most secure.
 - We will properly see them in SIP soon



Note the output of the hash function all have the same length – regardless of the input value size
 The hash value is called a digest

A small change in the input value yields a totally different digest





Basic principle

- The password is stored on the server – The “hash”

The hacker has learned
The username – the public URI
The nonce
The hash'ed password+nonce
Next time the server will choose a new random nonce

Username: john@domain.com
Password: ABC123



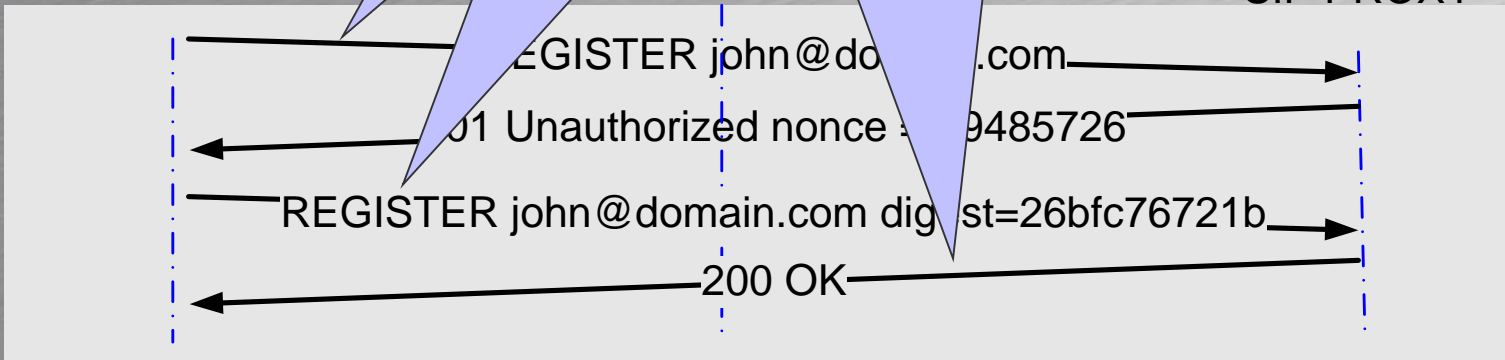
UA



Username: john@domain.com
Password: ABC123



SIP PROXY





Wireshark capture

No.	Source	Destination	Protocol	Info
160	10.197.0.104	87.48.131.54	SIP	Request: REGISTER sip:vk102113.
161	87.48.131.54	10.197.0.104	SIP	Status: 401 Unauthorized (0
Frame 161: 540 bytes on wire (4320 bits), 540 bytes captured (4320 b				
Ethernet II, Src: Motorola_be:4c:84 (00:24:37:be:4c:84), Dst: LnSrit				
Internet Protocol Version 4, Src: 87.48.131.54 (87.48.131.54), Dst:				
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)				
Session Initiation Protocol (401)				
Status-Line: SIP/2.0 401 Unauthorized				
Status-Code: 401				
[Resent Packet: False]				
Message Header				
From: "5401 heth"<sip:henrikth@...ip.dk>;tag=95859cb8-ac5				
To: "5401 heth"<sip:henrikth@vk1...ip.dk>;tag=5e13d931038de3				
Call-ID: 68656e72696b-aabb-7065-...23b0-0-2eba@10.197.0.104				
CSeq: 1 REGISTER				
Via: SIP/2.0/UDP 10.197.0.104:5060 branch=z9hg4bK-33-c7e4-4abde8b4				
Content-Length: 0				
WWW-Authenticate: Digest nonce="3B75025A1DDC2D5100000000F79C7455"				
163	87.48.131.54	10.197.0.104	SIP	status: 200 OK (1 bindings)

The server adds a nonce in the packet to the client

Digest nonce="3B75025A1DDC2D5100000000F79C7455"

Filter: sip Expression... Clear Apply Save New Label

G Y

● ○

ntec*

No.	Source	Destination	Protocol	Info
162	10.197.0.104	87.48.131.54	SIP	Request: REGISTER sip:vk102113.hvoip.dk SIP/2.0 (1 binding)
163	87.48.131.54	10.197.0.104	SIP	Status: 200 OK (1 binding)

Frame 162: 817 bytes on wire (6536 bits), 817 bytes captured (6536 bits)

Ethernet II, Src: LnSriatha_ab:23:b0 (00:1a:7e:ab:23:b0), Dst: Motorola_b

Internet Protocol Version 4, Src: 10.197.0.104 (10.197.0.104), Dst: 87.48.131.54

User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)

Session Initiation Protocol (REGISTER)

Request-Line: REGISTER sip:vk102113.hvoip.dk SIP/2.0

Message Header

From: "5401 heth"<sip:henrikth@vk102113.hvoip.dk>; tag=95859cb8-ac50068

To: "5401 heth"<sip:henrikth@vk102113.hvoip.dk>

Call-ID: 68656e72696b-aabb-7065-01a7eab23b0-0-2eba@10.197.0.104

CSeq: 2 REGISTER

Via: SIP/2.0/UDP 10.197.0.104:5060;branch=z9hg4bK-33-c826-5b7b8d92

Max-Forwards: 70

Support

User-A

Expires: 0

[truncated] Authorization: Digest user=henrikth@vk102113.hvoip.dk

Authentication Scheme: Digest

username="henrikth@vk102113.hvoip.dk"

realm="hvoip.ip.tdk.dk"

nonce="3B75025A1DDC2D5100000000F79C74"

uri="sip:vk102113.hvoip.dk"

response="2c881a030008dd77a29ada104d3992ec"

algorithm=MD5

The hashed password and nonce

dk

ngs)

dk



Wireshark capture

- Packet 160 – Client register request
 - No password attached
- Packet 161 – Register rejected
- Packet 162 – Client register request
 - Hash digest included
- Packet 163 – Server registers client
 - The client is online

No.	Source	Destination	Protocol	Info
160	10.197.0.104	87.48.131.54	SIP	Request: REGISTER sip:vk102113.hvoip.dk
161	87.48.131.54	10.197.0.104	SIP	Status: 401 Unauthorized (0 bindings)
162	10.197.0.104	87.48.131.54	SIP	Request: REGISTER sip:vk102113.hvoip.dk
163	87.48.131.54	10.197.0.104	SIP	Status: 200 OK (1 bindings)



SRTP

Secure Real Time Transport Protocol

- SRTP provides
 - Confidentiality: Encryption of voice
 - Authentication: Identity of parties
 - Integrity: Data not changed in transit
 - Replay protection: Packets cant be replayed
- SRTP can be used with unicast and multicast
- SRTP is described in RFC 3711



SRTP

Secure Real Time Transport Protocol

- RFC 3711 does not cover key exchange between end-points
- A master key must be exchanged securely between end-points
- The master key is used to generate the all the necessary session keys
- Key exchange implemented using public or proprietary methods
 - Different vendors different method ☹️



S RTP

Secure Real Time Transport Protocol

- Keys could be exchanged using
 - MIKEY: Public RFC 3830
 - Multimedia Internet Keying
 - ZRTP: Public RFC 6189
 - Zimmermann RTP
 - KEYMGT: Public RFC 4567
 - Key Management Extensions
 - SDMS
 - Session Description Protocol Security Descriptions for Media Streams



SRTP with ZRTP

Secure Real Time Transport Protocol

- ZRTP is a cryptographic key-agreement protocol to negotiate keys for encryption
- Uses Diffie-Hellman key exchange
- Uses same UDP ports as SRTP
 - No extra UDP or TCP ports necessary
- ZRTP can be used with SIP and H.323

Diffie and Hellman



- Dr. Whitfield Diffie
- Bachelor of science mathematics
- Retired but studying security in grid computing



- Martin Hellman
- Professor Emeritus from Stanford University
- Retired



Diffie-Hellman key exchange

- Uses mathematical one-way functions
- Security based on huge prime numbers

A 1024 bit prime:

1797693134862315907708391567937874531978602960487560
1170644442368419718021615851936894783379586492554150
2180565485980503646440548199239100050792877003355816
6392295531362390765087357599148225748625750074253020
7744771258955095793777842444242661733472762929938766
8709205606050270810842907692932019128194467627007

e

- DH Group 2 = 1024 bit
- DH Group 5 = 1536 bit
- Higher group numbers are more secure



Alice



Common paint

+



Secret colours

=

Bob

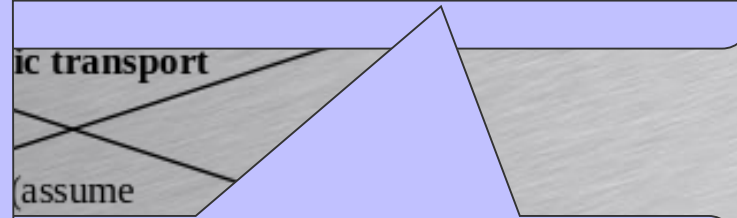


+



=

They both have a common secret consisting of $\frac{1}{3}$ yellow + $\frac{1}{3}$ orange + $\frac{1}{3}$ blue = Common colour



Mixing paint is a many-to-one function. Impossible to separate into individual colours

+



=



Common secret

+



=





SIP and ZRTP flow

Encrypted SRTP media stream. Keys exchanged with diffie-hellman directly between the endpoints. No involvement from SIP proxies



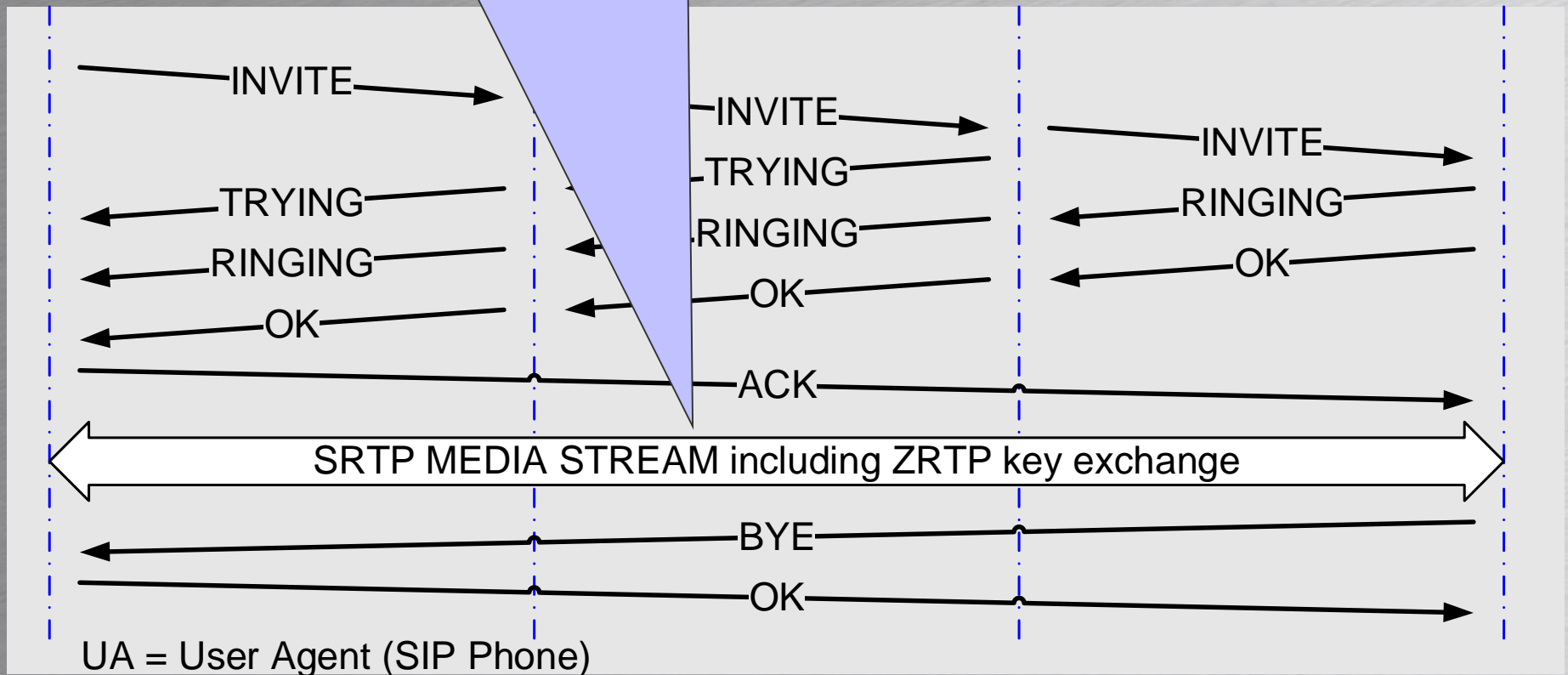
UA



SIP PROXY



UA

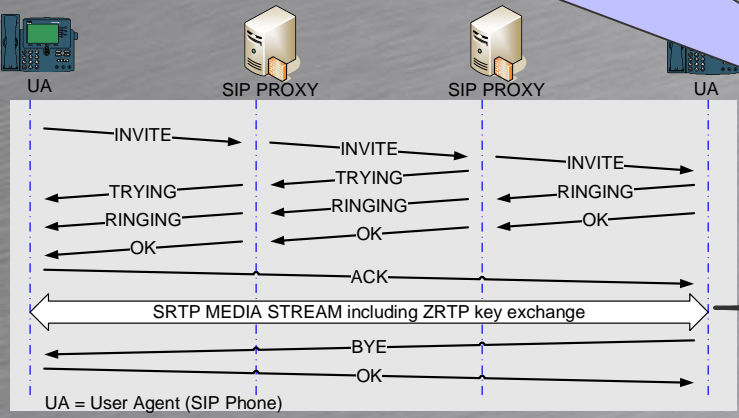
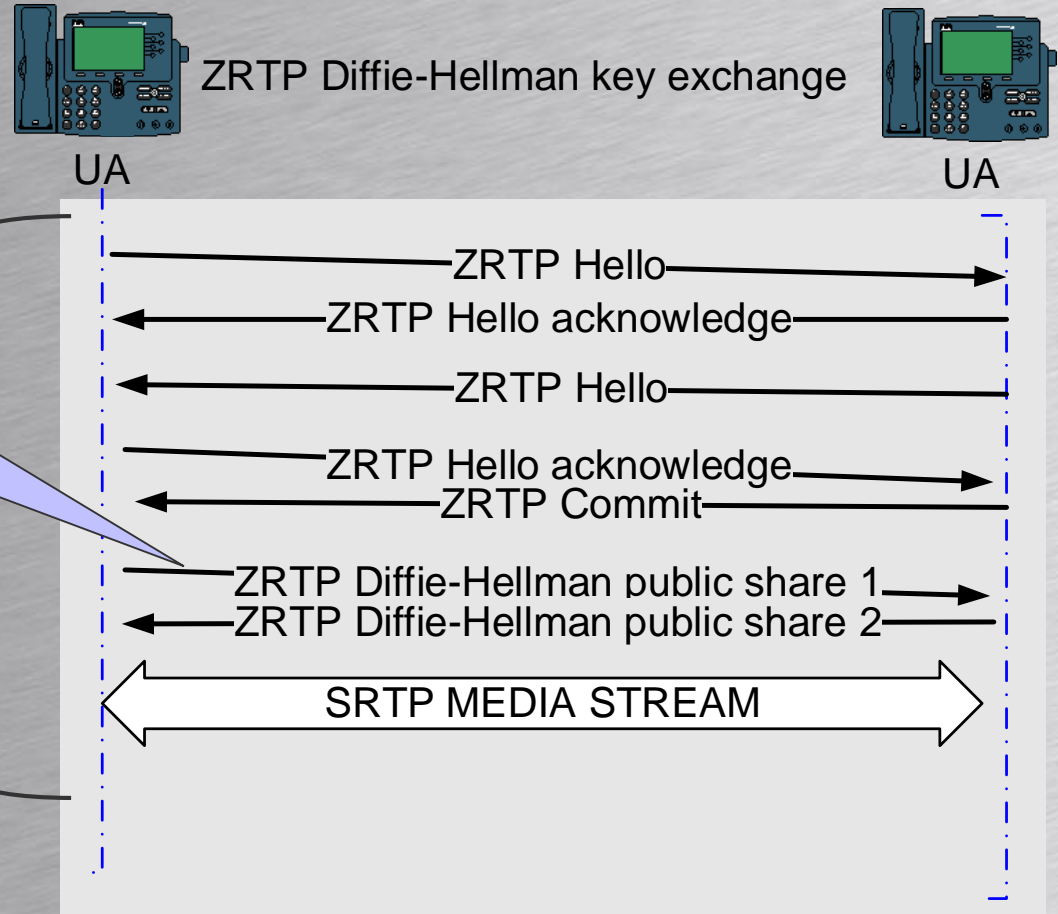


UA = User Agent (SIP Phone)



RTP, ZRTP and SRTP

ZRTP exchanges Diffie-Hellman keys using the RTP protocol and SRTP uses the keys for encryption





Secure SIP

- As known from web surfing
 - HTTP is unencrypted transport on TCP port 80
 - HTTPS is encrypted transport on TCP port 443
 - HTTPS uses SSL/TLS for security
- SIPS signaling or SIP over SSL/TLS gives
 - SIP is unencrypted transport on TCP port 5060
 - SIPS is encrypted transport on TCP port 5061
 - SIPS uses SSL/TLS for security



SSL/TLS

- SSL – Secure Sockets Layer
 - Older but still used
- TLS – Transport Layer Security
 - New version of SSL giving better security
- SSL and TLS can use different security protocols and key sizes
 - Client and server agree on which security settings to use. Also called Cipher setting



SSL/TLS

- When a client initiates a SSL or TLS connection to a server it list the possible Cipher settings it supports
- The server responds with the cipher setting it prefers
- A cipher setting typically include
 - Exchange of public keys (Asymmetric keys)
 - An encryption standard and key size
 - An HASH algorithm to use



SSL/TLS

```
Transmission Control Protocol, Src Port: 50438 (50438), Dst Port:
Secure Sockets Layer
  TLSv1 Record Layer: Handshake Protocol: Client Hello
    Content Type: Handshake (22)
    Version: TLS 1.0 (0x0300)
    Length: 191
  Handshake Protocol: Client Hello
    Handshake Type: Client Hello
    Length: 187
    Version: TLS 1.1 (0x0302)
  Random
    Session ID Length: 0
    Cipher Suites Length: 72
  Cipher Suites (36 suites)
    Cipher Suite: TLS_ECDHE_ECDSA_WITH_AES_256_CBC_SHA (0xc00a)
    Cipher Suite: TLS_ECDHE_RSA_WITH_AES_256_CBC_SHA (0xc014)
    Cipher Suite: TLS_DHE_RSA_WITH_CAMELLIA_256_CBC_SHA (0x0088)
    Cipher Suite: TLS_DHE_DSS_WITH_CAMELLIA_256_CBC_SHA (0x0087)
    Cipher Suite: TLS_DHE_RSA_WITH_AES_256_CBC_SHA (0x0039)
    Cipher Suite: TLS_DHE_DSS_WITH_AES_256_CBC_SHA (0x0038)
    Cipher Suite: TLS_ECDH_RSA_WITH_AES_256_CBC_SHA (0xc00f)
    Cipher Suite: TLS_ECDH_ECDSA_WITH_AES_256_CBC_SHA (0xc005)
    Cipher Suite: TLS_RSA_WITH_CAMELLIA_256_CBC_SHA (0x0084)
```

In this example the client lists 36 different Cipher settings/suites the server can choice from



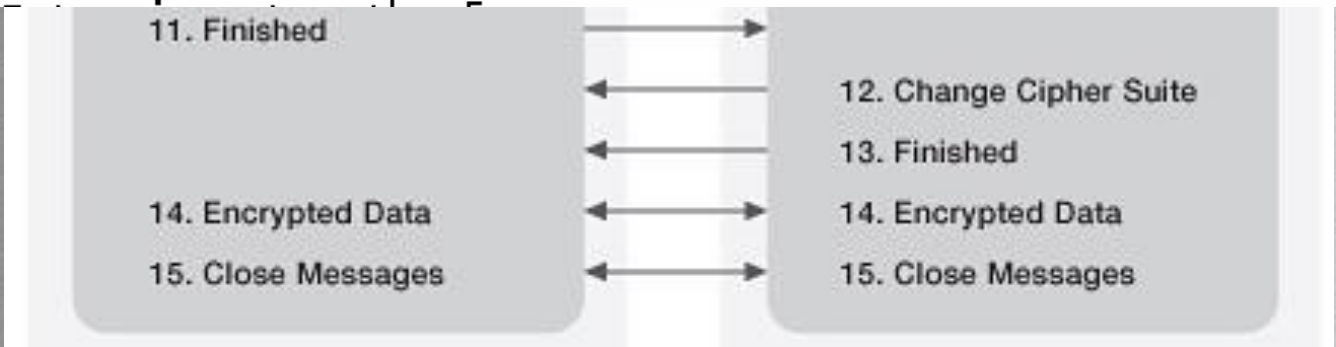
SSL/TLS

In this example the server have chosen the Cipher setting

- TLS: Use TLS
- RSA: Key exchange protocol
- RC4_128: RC4 encryption with 128 bit key
- SHA: Hash algorithm

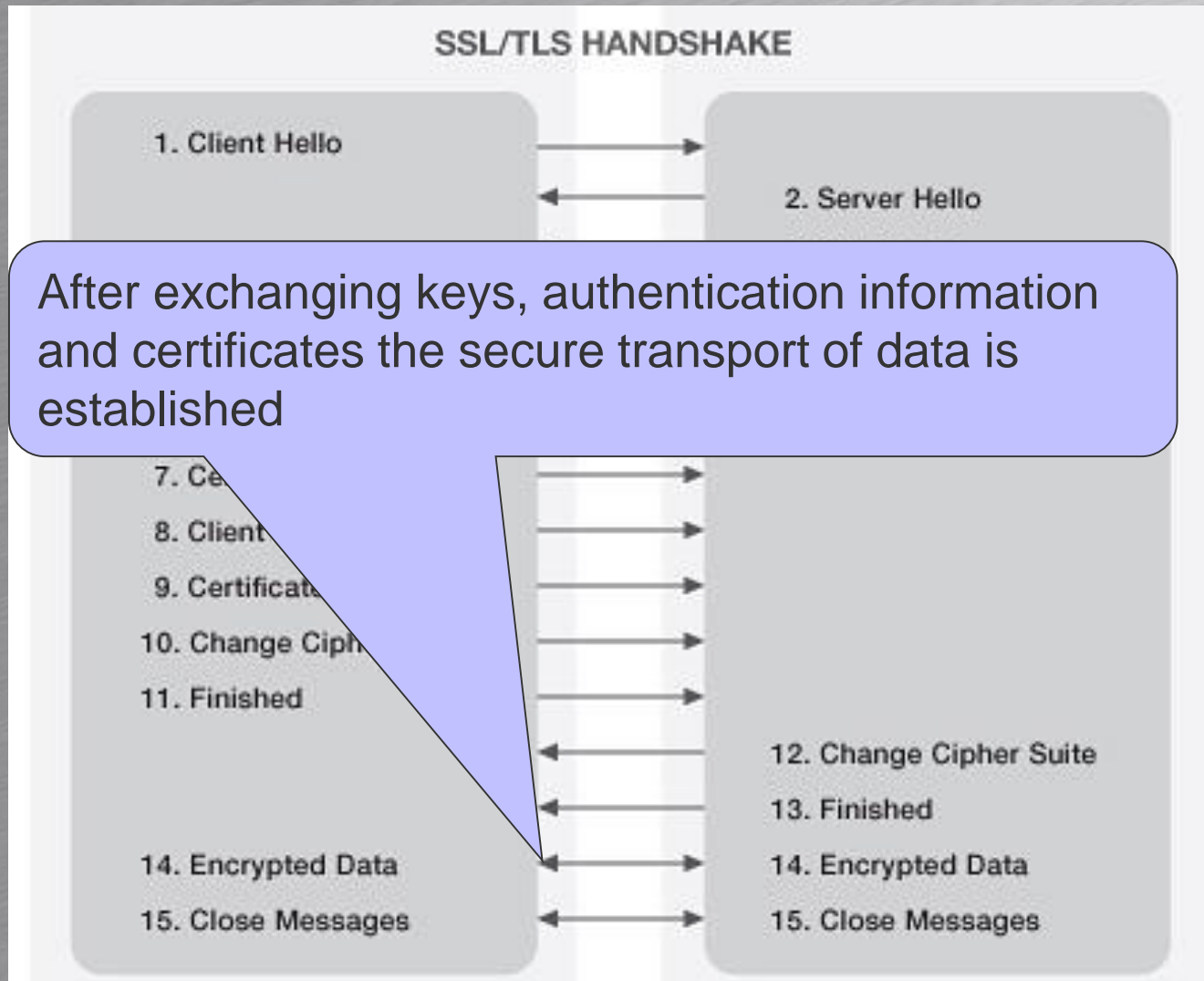
443), Dst Port: 50438
30), #14(1136)]
e Handshake Messages

```
Length: 45  
Handshake Protocol: Server Hello  
Handshake Server Hello (2)  
Length: 45  
Version: TLS 1.0 (0x0301)  
Random  
Session ID Length: 0  
Cipher suite: TLS_RSA_WITH_RC4_128_SHA (0x0005)  
Compression Method: null (0)
```





SSL/TLS





VoIP AVAILABILITY



When things go wrong



Things that might fail

- IP PBX fails
 - All phones registered fail
- Power outage
 - All devices without battery backup fail
- Network device failure
 - All devices dependent on that device fail
- PSTN/ISDN connection fails
 - No incoming or outgoing calls possible
- VPN connection between sites fail
 - No calls between sites



Redundancy and failover

- Redundancy
 - When the primary device fails a redundant secondary device takes over the load and ensures connectivity
 - Important an alert is transmitted if the primary or secondary device fails
 - No impact on normal service



Redundant power supply for server





Types of redundancy

- Hot standby
 - Secondary device ready to offload primary
 - Heartbeats transmitted between secondary and primary
 - If primary device don't answer heartbeats for a given time period secondary device takes over
 - Alarm transmitted to alert IT-Staff
 - If primary device don't receive heartbeats from secondary device for a given time period
 - Alarm transmitted to alert IT-Staff



Types of redundancy

- Load balancing
 - Workload distributed between two or more redundant devices
 - Heartbeats transmitted between devices
 - If one device don't answer heartbeats for a given time period the workload are distributed to the remaining
 - Alarm transmitted to alert IT-Staff

RAID: Redundant Array of Independent Disks





Virtual IP address

- A virtual IP address is a IP address shared between two or more devices
 - A virtual IP address uses a virtual MAC address
- Only one device will normally use the IP address
 - Called the Active device
- The active device does all the workload
- If the active device fails the standby device becomes active and takes over the virtual IP address and the virtual MAC address

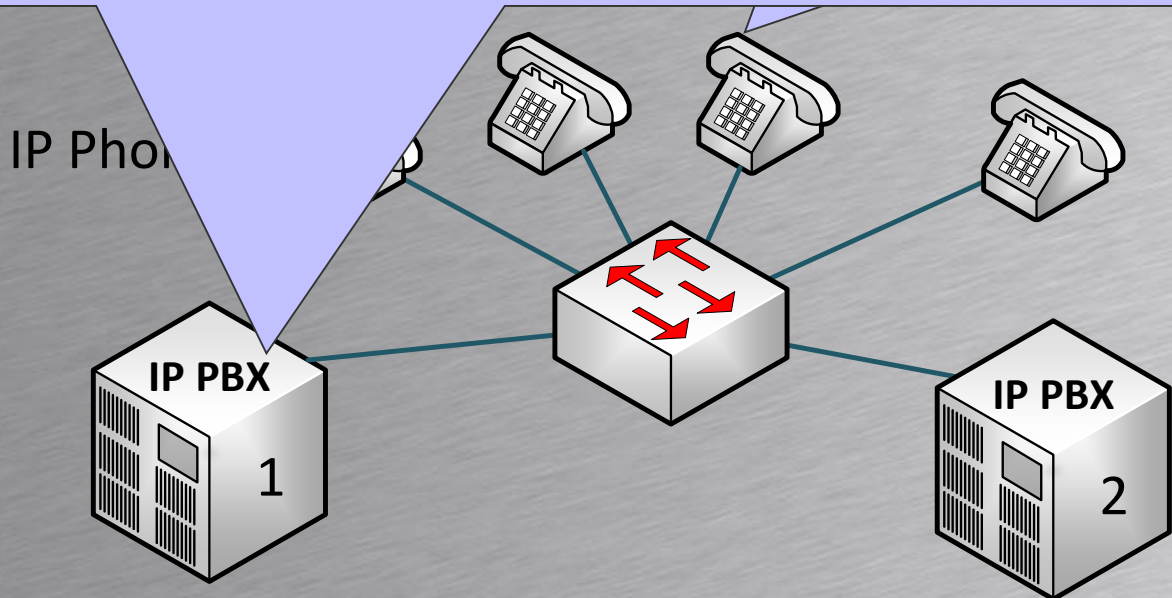


Virtual IP address

Example

The phones register to 192.168.1.2

IP PBX 1 transmits all information included registered phones on-going calls and other information to IP PBX2 on 192.168.1.11



IP PBX 1 configuration

IP Address	: 192.168.1.10
State	: Active
Virtual IP addr.	: 192.168.1.2
Virtual MAC addr.:	00-10-20-30-40-50

IP PBX 2 configuration

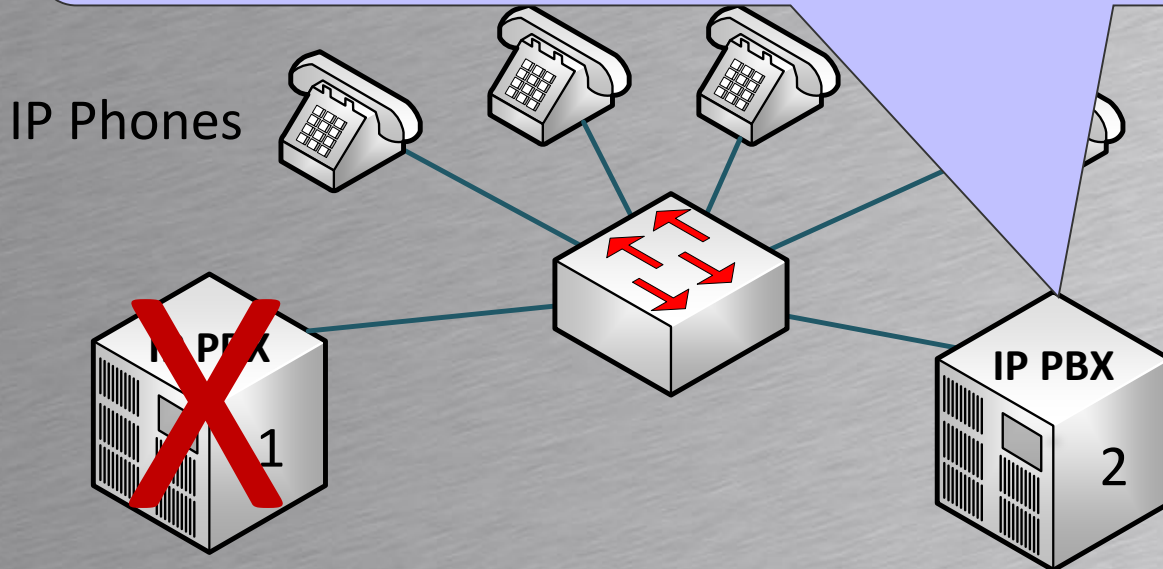
IP Address	: 192.168.1.11
State	: Standby
Virtual IP addr.	: -
Virtual MAC addr.:	-



Virtual IP address

Example

IP PBX 2 takes over as active when no replies from heartbeats received. The state of all phones known and on-going calls still in progress



IP PBX 1 configuration

IP Address	: 192.168.1.10
State	: PBX service down
Virtual IP addr.	: -
Virtual MAC addr.:	-

IP PBX 2 configuration

IP Address	: 192.168.1.11
State	: Active
Virtual IP addr.	: 192.168.1.2
Virtual MAC addr.:	00-10-20-30-40-50



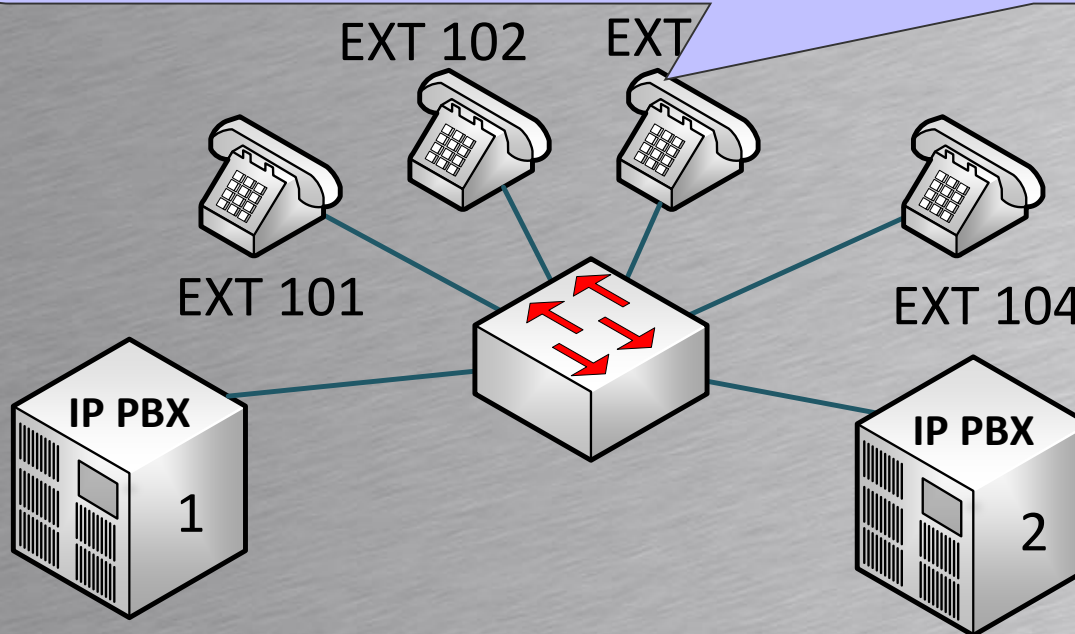
SIP proxy redundancy 1

- Client based failover
- SIP phones – UA – register with two SIP Proxies
 - NOTE: Not all SIP phones can register twice
 - A primary and a backup proxy
 - All phones in the SIP domain register with two proxies
 - If the primary fails the phones use the backup proxy



SIP proxy redundancy 1

Each SIP phone can initiate a call using IP PBX 1 or IP PBX 2.
If the first tried IP PBX is unavailable the SIP phone will try the other IP PBX



IP PBX 1 configuration

IP Address : 192.168.1.10

Registered phones

EXT: 101, 102, 103 and 104

IP PBX 2 configuration

IP Address : 192.168.1.11

Registered phones

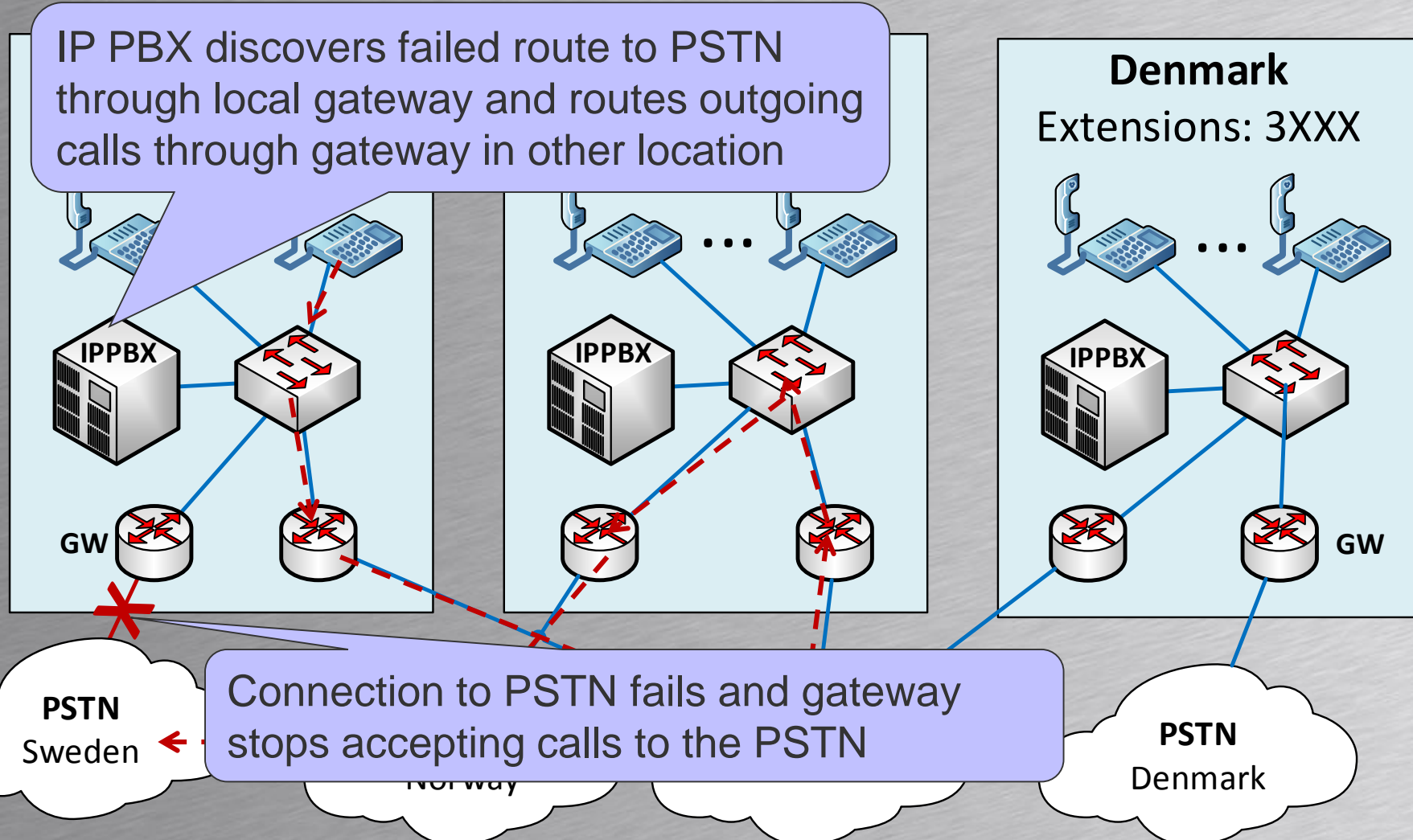
EXT: 101, 102, 103 and 104



PSTN failover

- Rerouting calls to the PSTN

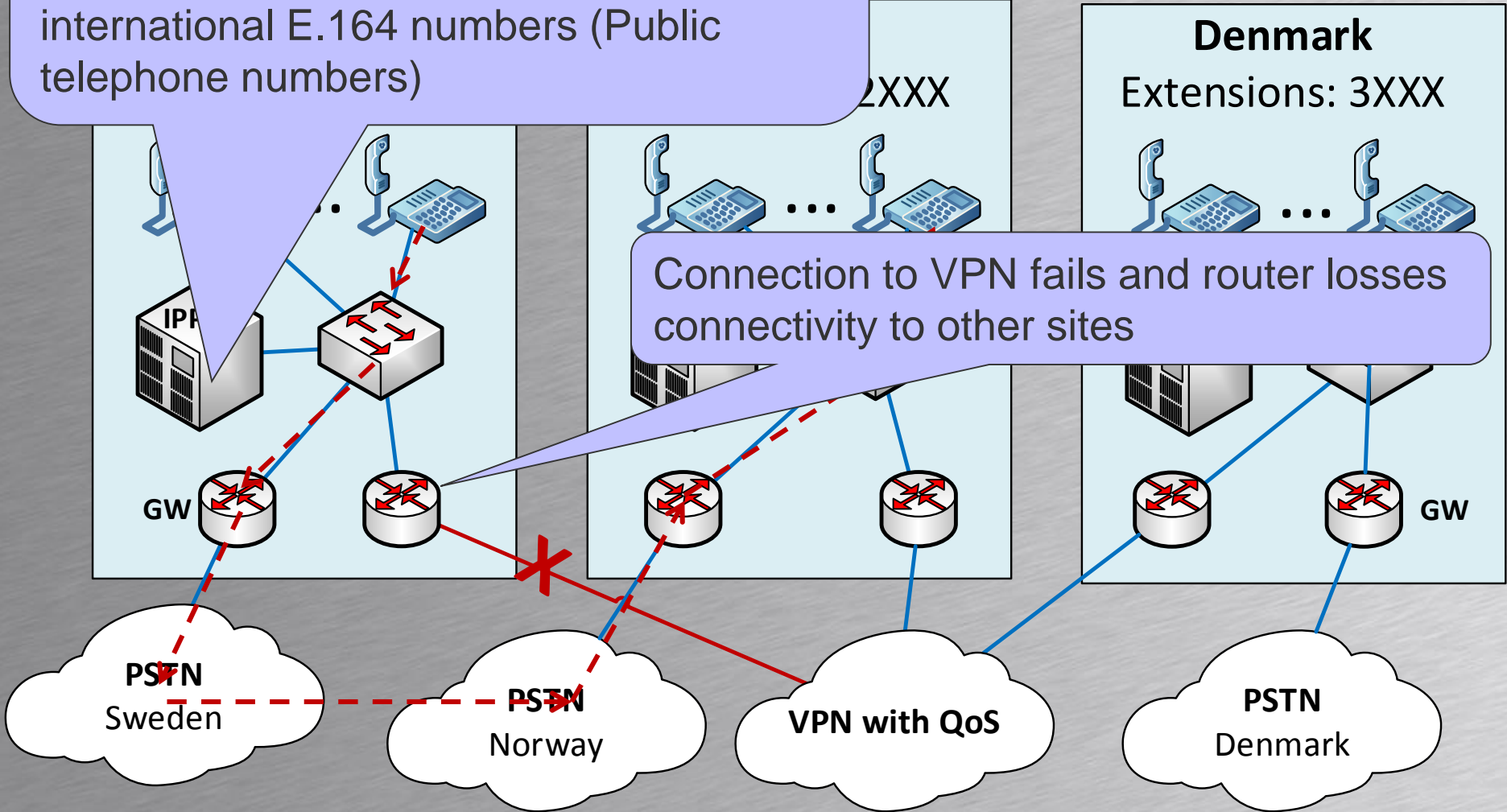
IP PBX discovers failed route to PSTN through local gateway and routes outgoing calls through gateway in other location



VPN failover



IP PBX discovers it cant reach IP PBX or IP phones in the other locations it routes the call through the PSTN using international E.164 numbers (Public telephone numbers)



Connection to VPN fails and router losses connectivity to other sites

Denmark
Extensions: 3XXX

~~PSTN~~
Sweden

~~PSTN~~
Norway

VPN with QoS

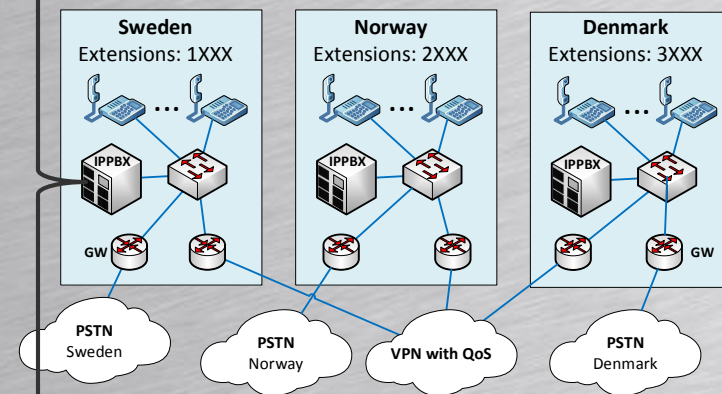
PSTN
Denmark



Route plans

- The individual IP PBX's are programmed with route plans
 - Example of Swedish route plan shown
 - Dialed 1xxx – x meaning any digit
 - Dialed 0. - . Meaning routed through
 - Pri = Priority. Lowest priority best. If unavailable try next

Dialed	Pri	Routed to
1xxx		Not routed processed locally
2xxx	1	The IP address of IP PBX in Norway
2xxx	2	The IP address of IP gateway in Sweden (Failover) Add 0047 for Norway + main-number + 2xxx (DiD)
3xxx	1	The IP address of IP PBX in Denmark
3xxx	2	The IP address of IP gateway in Sweden (Failover) Add 0045 for Denmark + main-number + 3xxx (DiD)
0.	1	The IP address of IP gateway in Sweden Line out – new dial tone from PSTN





VPN failover

ascom

IP PBX discovers it can't reach IP PBX or IP phones in the other locations it routes the call through the PSTN using international E.164 numbers (Public telephone numbers)

Connection to VPN fails and router loses connectivity to other sites

