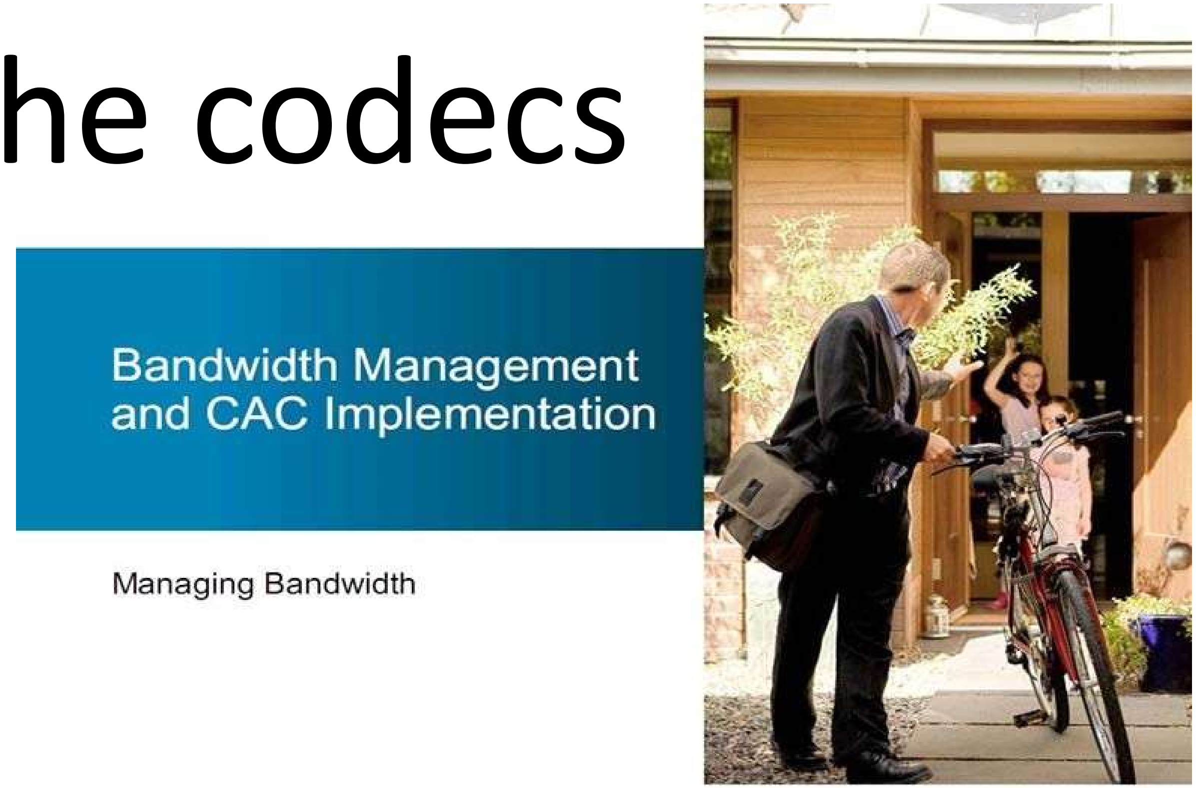
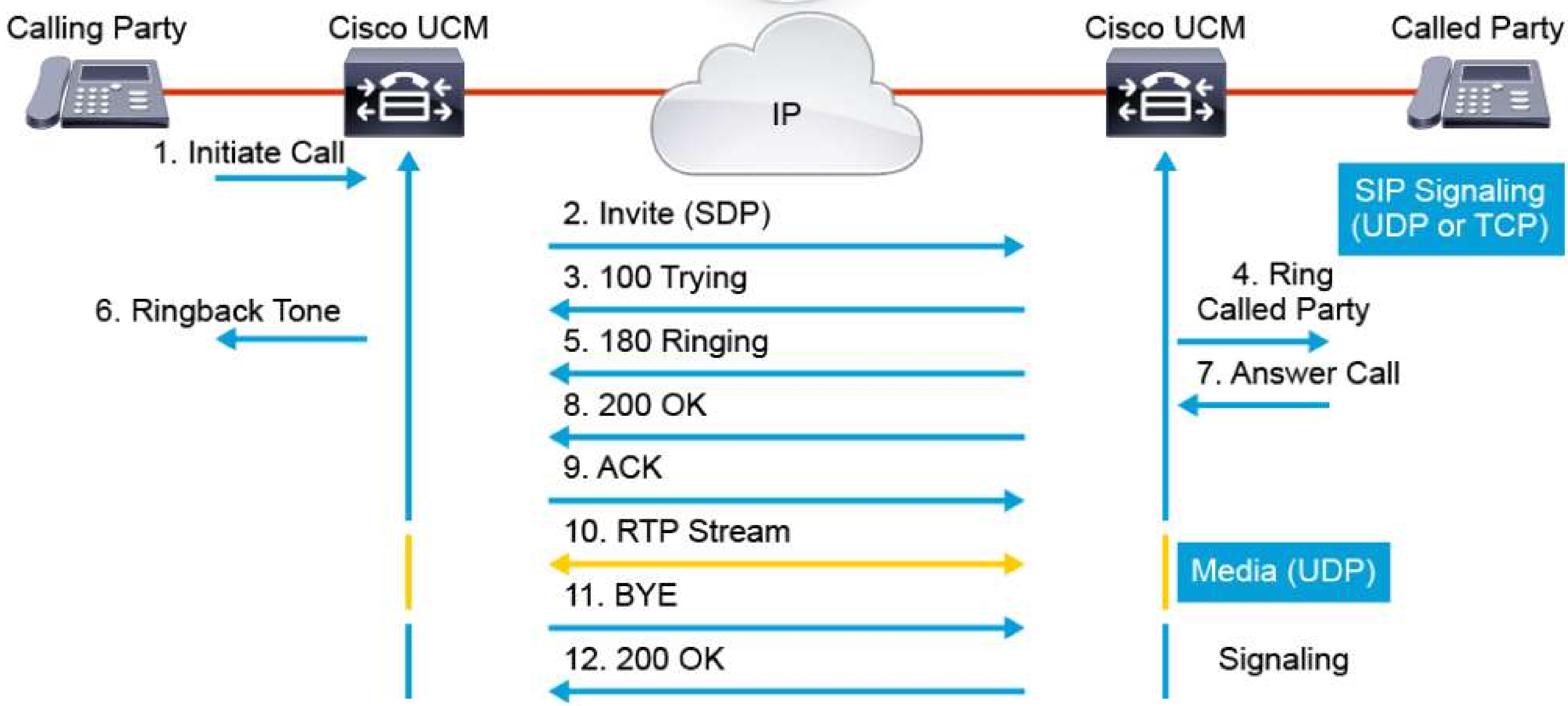
Exploring the codecs



review

Describe SIP Call Signaling for Call Setup and Teardown

Open Transcript



Define Codecs

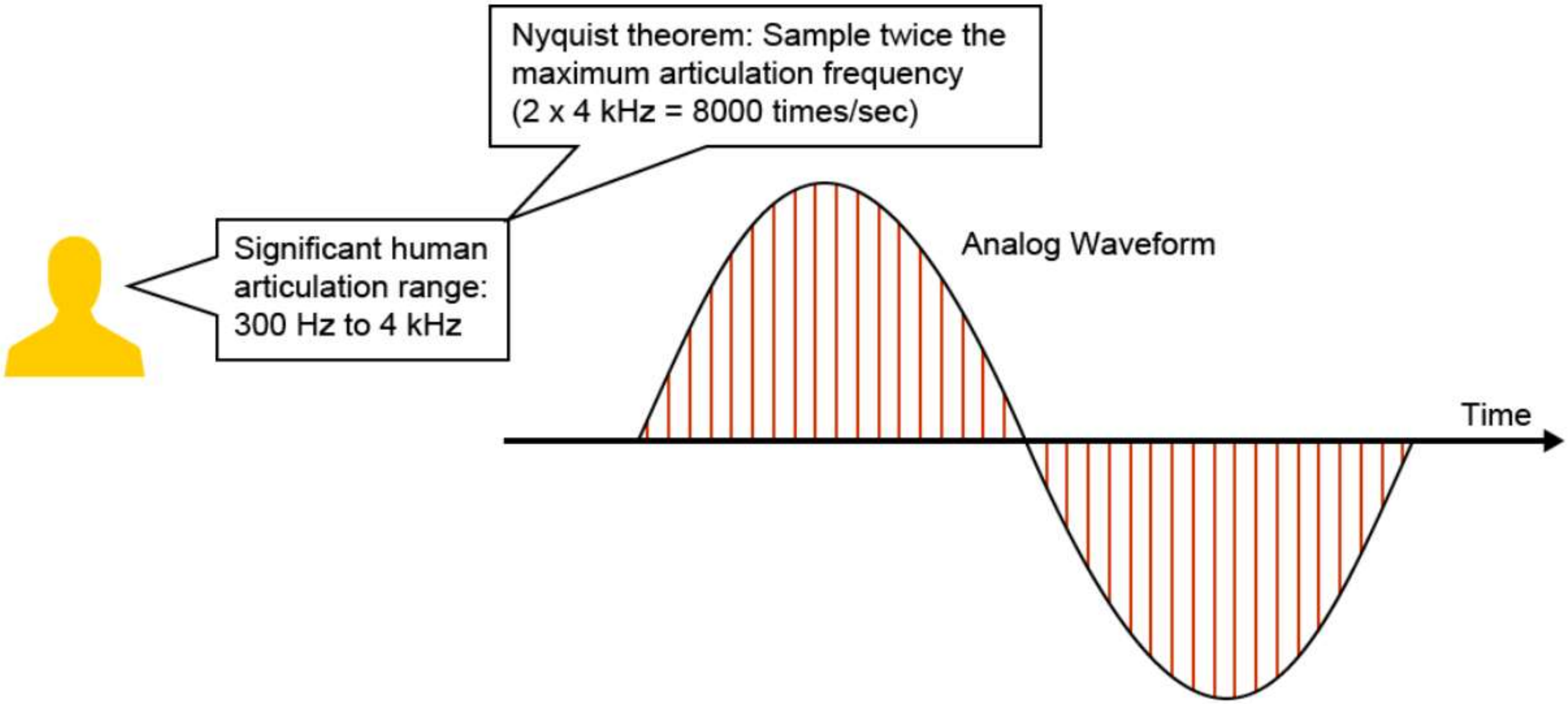
Pulse Code Modulation (PCM)

1. Sample the analog signal regularly.
2. Quantize the sample.
3. Encode the value into a binary expression.

Codec Compression

4. Compress the samples to reduce bandwidth (optional).

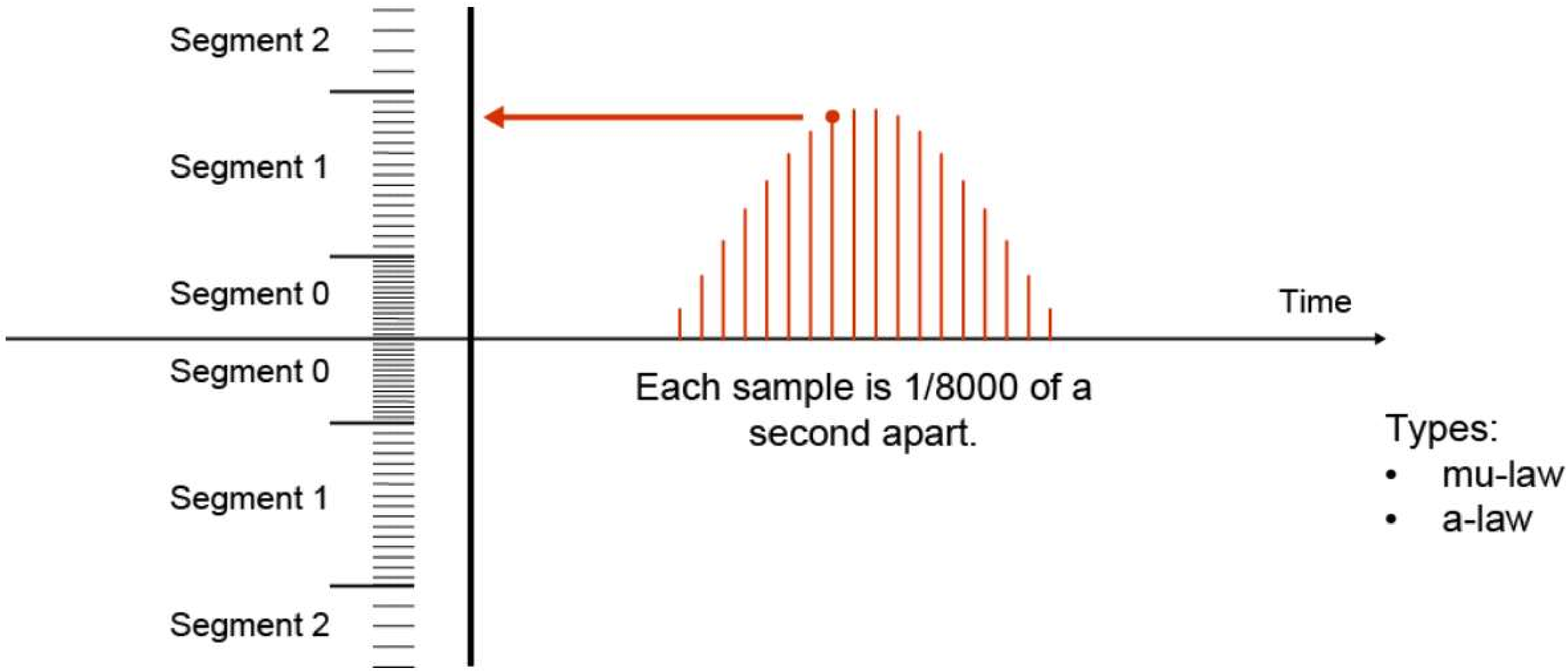
Sampling



1. Sample the analog signal regularly. The sampling rate must be twice the highest frequency to produce playback. The sampling rate is 8000 samples per second (8 kHz), which reflects the fact that the bulk of human voice energy is carried in the spectrum of 0–4 kHz.

Quantization

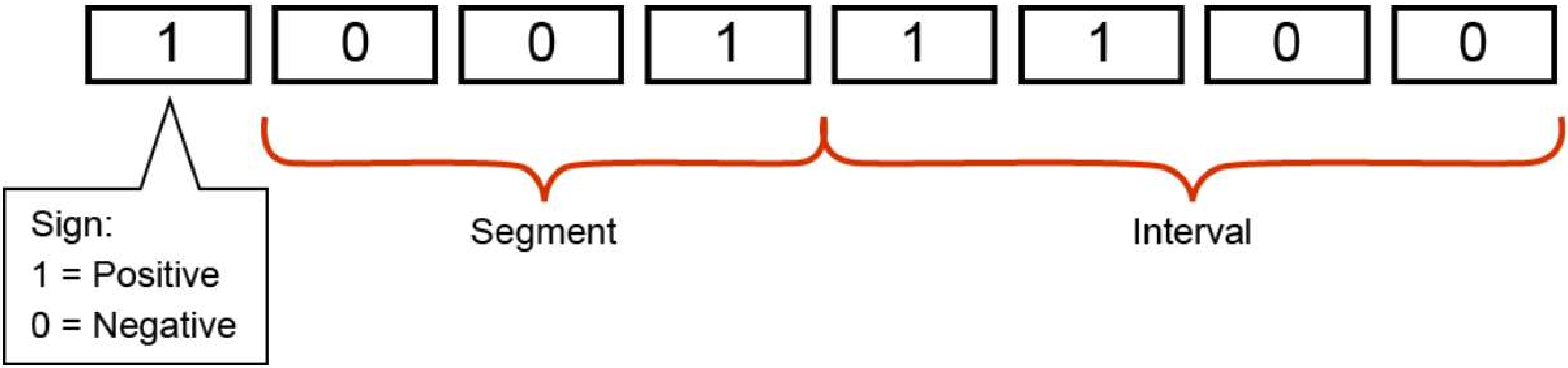
G.711 Operations



2. Quantize the sample. Quantization consists of a scale made up of 8 major segments. Each segment is subdivided into 16 intervals.

Encoding

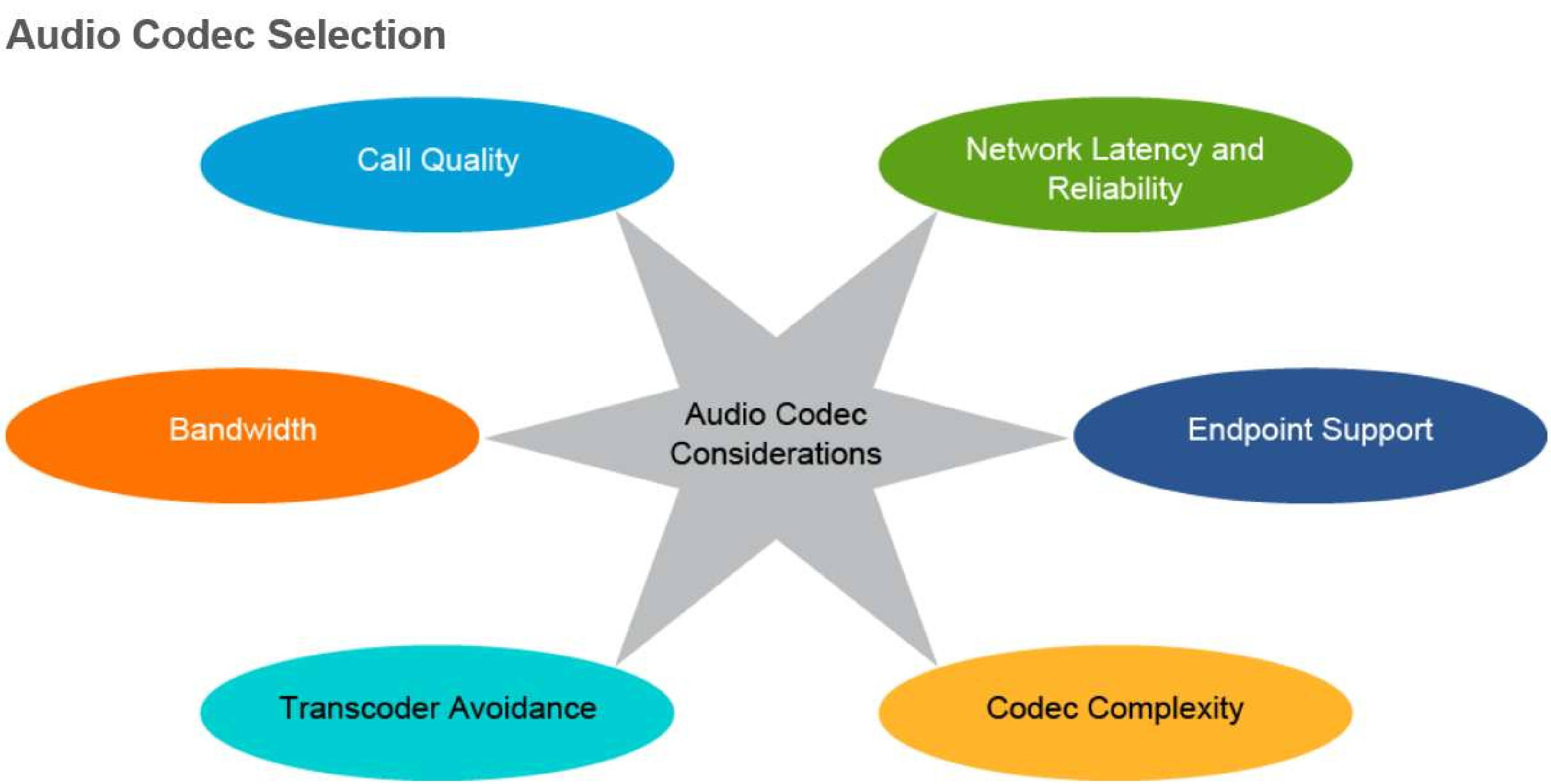
G.711 9-Bit Words



3. Encode the value into an 8-bit digital form (octet).

4. (Optional) Compress the samples to reduce bandwidth. Signal compression is used to reduce the bandwidth usage per call.

Audio Codec Selection



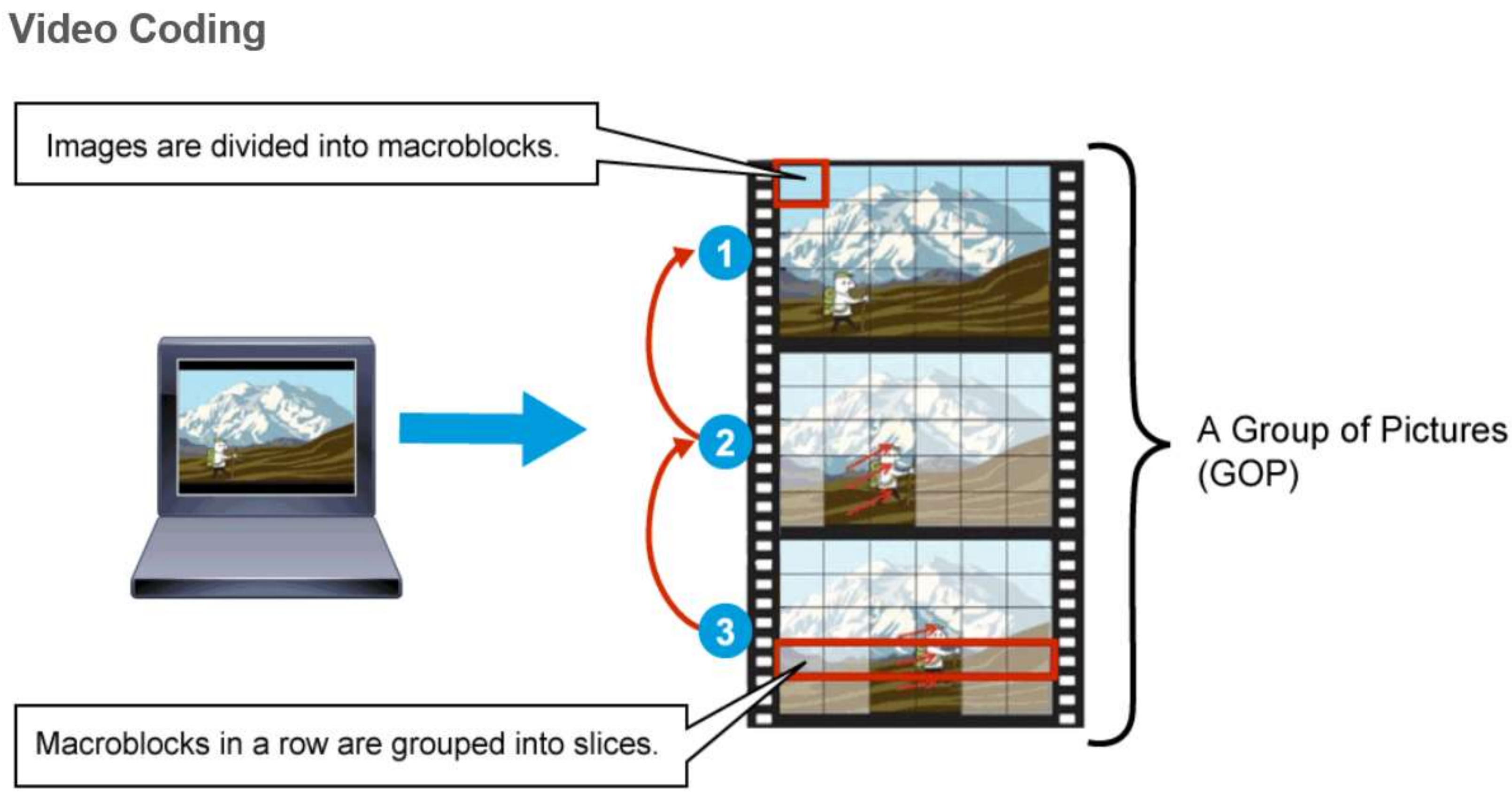
Audio Codecs Compared

Codec	Bandwidth (kbps)	Developed	Band	Sampling (kHz)	License	Released	MOS
G.711	64	ITU-T	Narrowband	8	Free	1972	4,3
G.722	48 / 56 / 64	ITU-T	Wideband	16	Free	1988	---
iLBC	15,2 / 13,3	Global IP Solutions*	Narrowband	8/16	Open-Source	2002	4,14
OPUS	6-510	IETF	Mixed	8/12/16/24/48	Royalty-Free	2012	---
G.729	8	ITU-T	Narrowband	8	Royalty-Free**	1996	3,92

* Acquired by Google in 2011. ** Since January 2017.

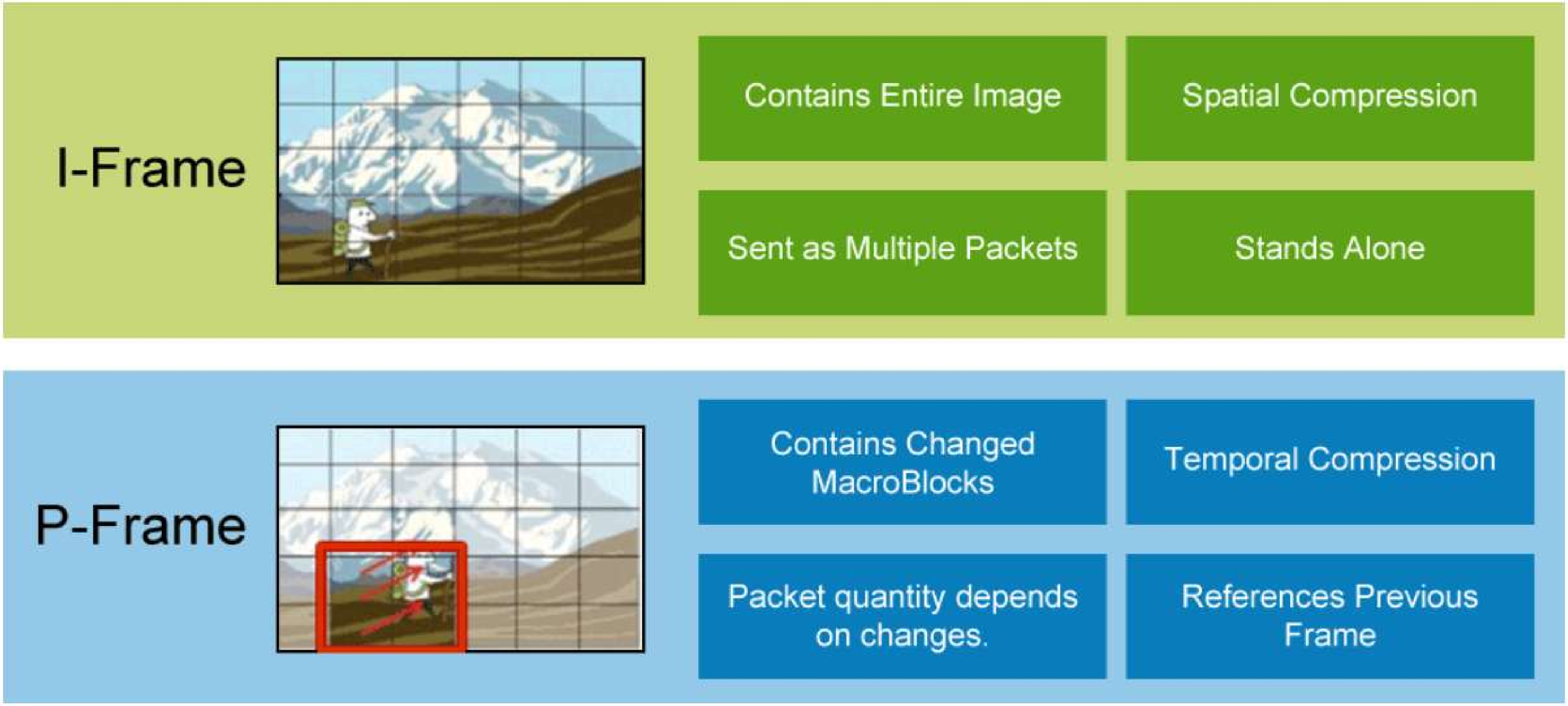
A single frame of the video, group pixels into blocks, then group blocks into Macroblocks.

Macroblocks that are in a contiguous row are grouped into slices (single row of macroblock).

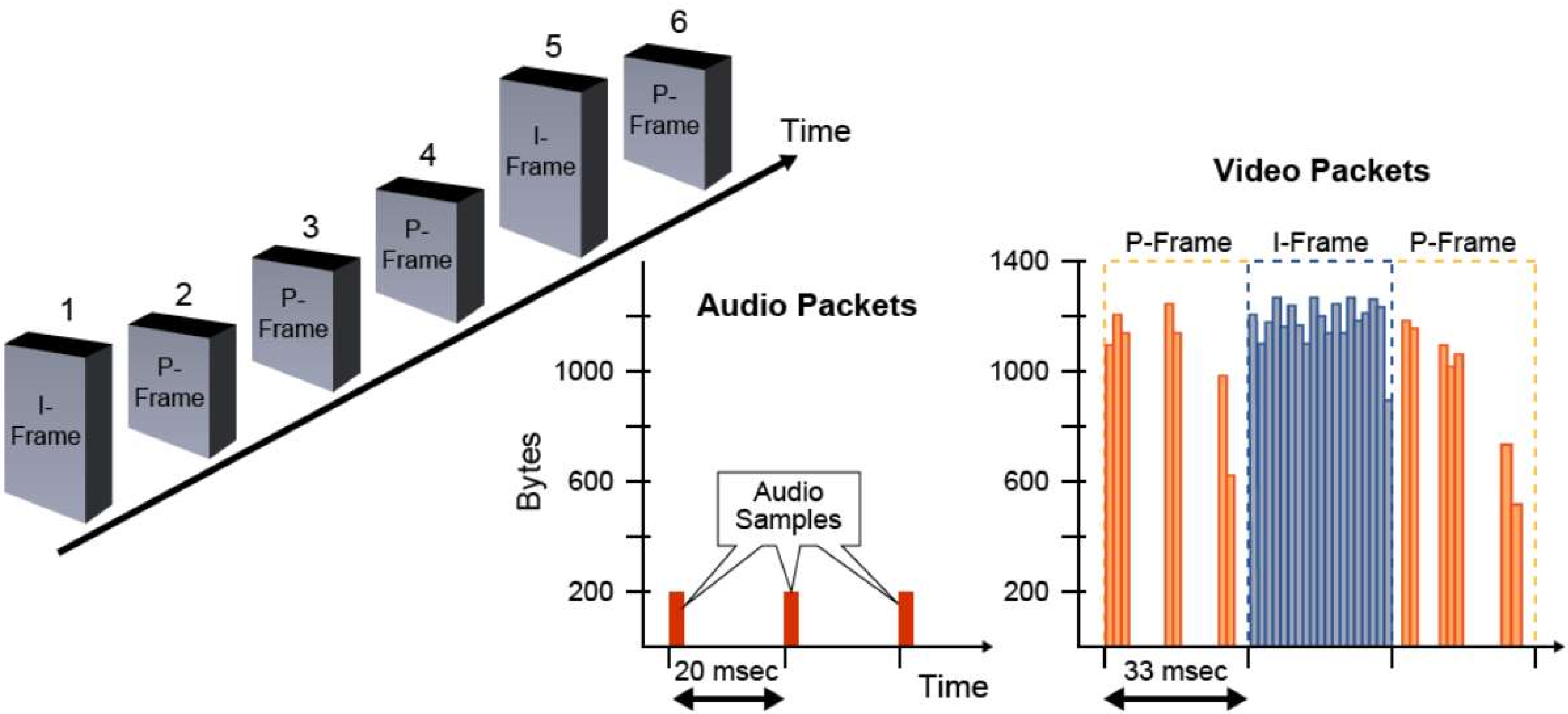


Video codecs are designed to identify changes from one frame to the next and only update the changes.
Initially the process requires a full frame - I-Frame (Intra-coded).
The second frame is compared to the first to identify which macroblocks have changed and only those blocks are updated and transmitted in the form of P-Frames (predictive-coded) or B-frames (bi-directional predictive-coded).

Pictures, or frames are grouped into Group of Pictures (GOPs) with the I-frame as the starting point and P-frames following it to the next I-frame.

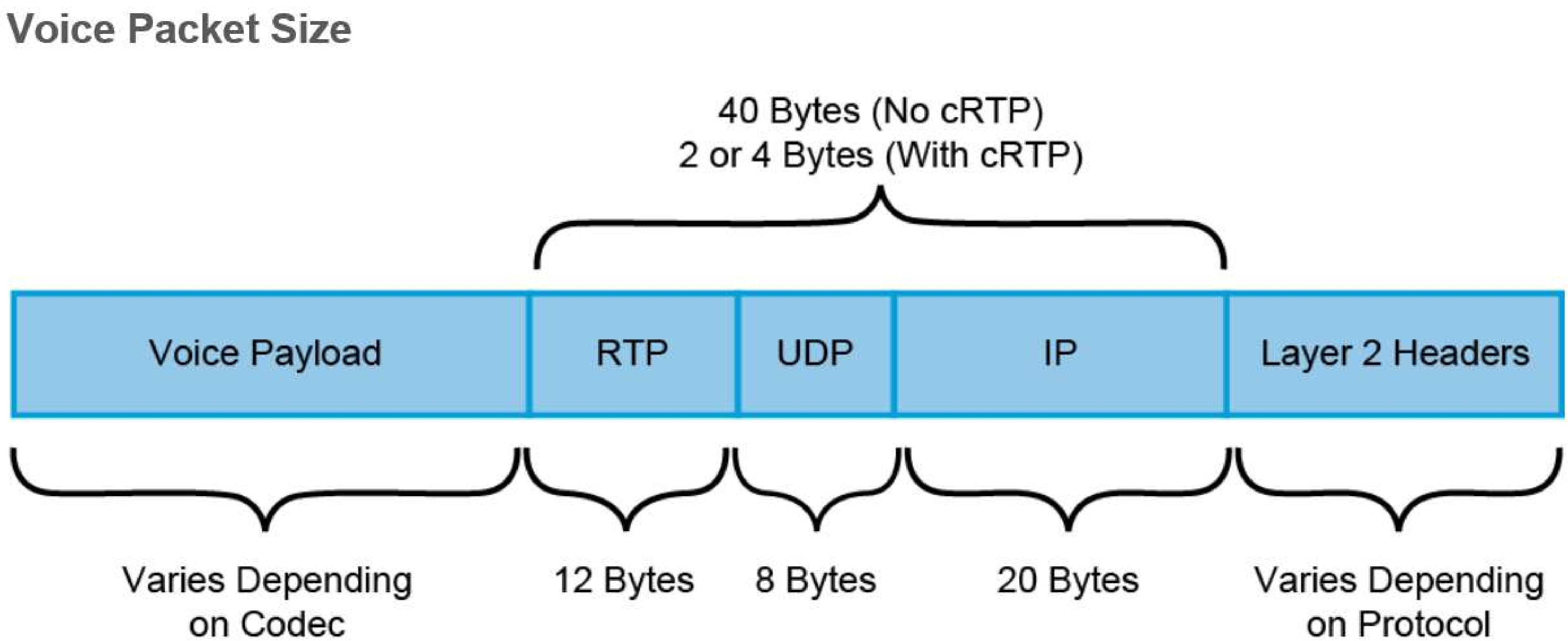


Video codecs, such as MPEG-2, MPEG-4, H.264, and H.265 use temporal compression algorithms. A codec that uses temporal compression does not send a complete frame sample (I-frame) at every sampling interval. Only some of the frames that are sent are I-frames.



Between the full-frames (I-frames), only the differences with the previous frame, represented as motion vectors and prediction errors, are encoded. The frames that carry these frame deltas (P- or B-frames) tend to be much smaller than I-frames, which is how the compression algorithm reduces the bandwidth.

Evaluate the Effects of Encryption on Codecs



- In total, 40 bytes of data are needed per packet.
- To reduce the size of the Layer 3 overhead, enable Compressed Real-time Transport Protocol (cRTP). cRTP will reduce the headers from 40 bytes down to 2 bytes or 4 bytes if a checksum is required.

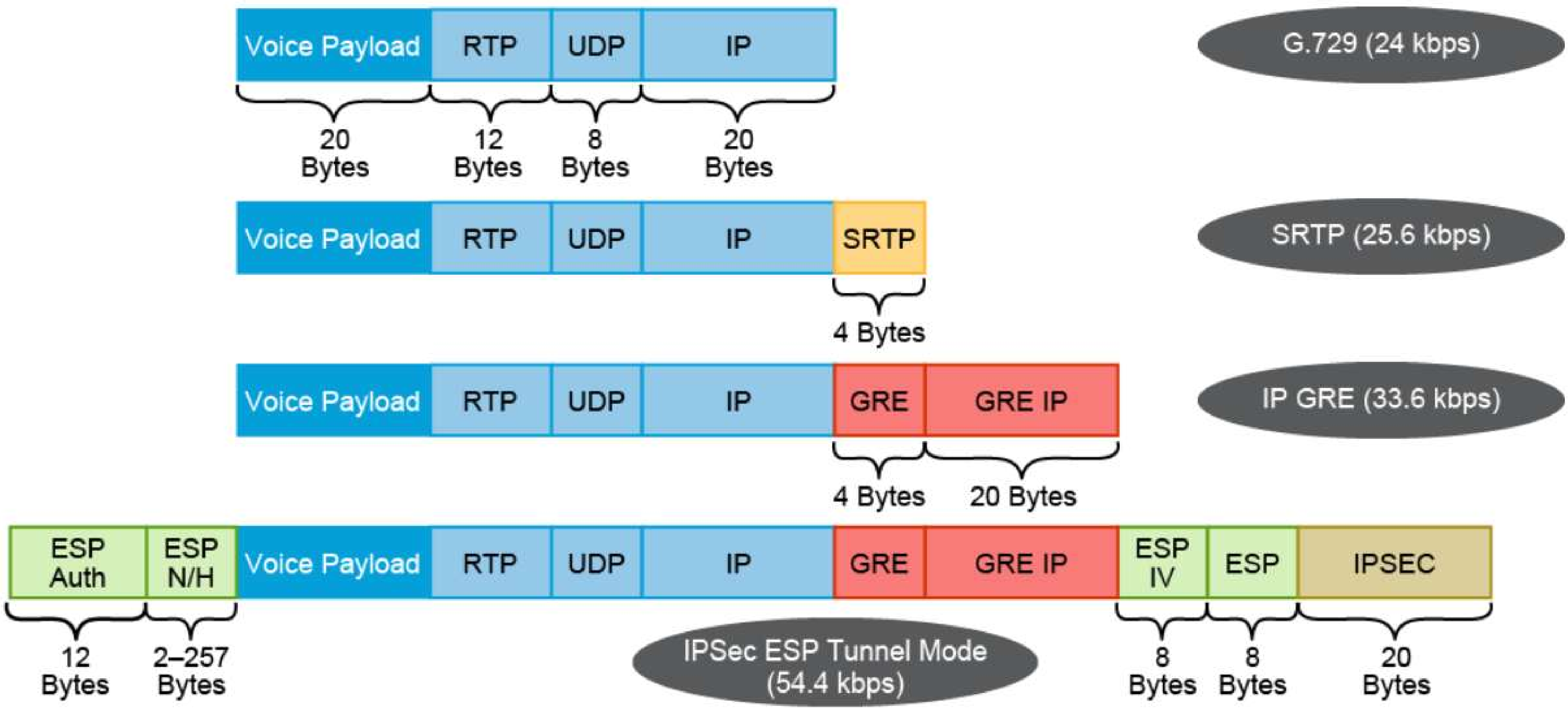
Audio Codec Bandwidth

Codec	Speed	Packetization	Packets per Second	Voice Payload	Bandwidth per Call * (no cRTP)	Bandwidth per Call * (cRTP)**
G.711	64 kbps	20 ms	50	160 bytes	80 kbps	64.8 kbps
G.711	64 kbps	30 ms	33	240 bytes	74.6 kbps	64.5 kbps
G.729	8 kbps	20 ms	50	20 bytes	24 kbps	8.8 kbps
G.729	8 kbps	40 ms	25	40 bytes	16 kbps	8.4 kbps

* Includes Layer 3 headers only. ** cRTP at 2 bytes.

G.711 uses 64000 bits per second, or 64 kbps, to represent one second of human speech, not including the extra overhead of Layer 3 or Layer 2 headers. The next important aspect of bandwidth consumption to understand, is how G.711 transmits the 64000 bits over a one second interval. Cisco Unified Communications Manager allows for the packet sizes to change based on the milliseconds of voice to be added to each packet. The process defines how much voice must be sampled before creating the data for a packet. The smaller the Packetization interval, the more packets are required and the smaller each packet will be.

Security Overhead on Voice Payloads



Examples of Bandwidth Management Options

Bandwidth in the IP WAN can be conserved by:

- Techniques that reduce required bandwidth of voice streams:
  - RTP header compression (as part of QoS link efficiency mechanisms)
  - Low-bandwidth codecs
- Techniques that influence the flow of voice streams:
  - Local media resources such as conference bridges
- Other solutions:
  - Transcoders
  - Multicast MOH from branch router flash

Cisco Unified Communications Manager Codec Configuration

Codec Configuration Considerations

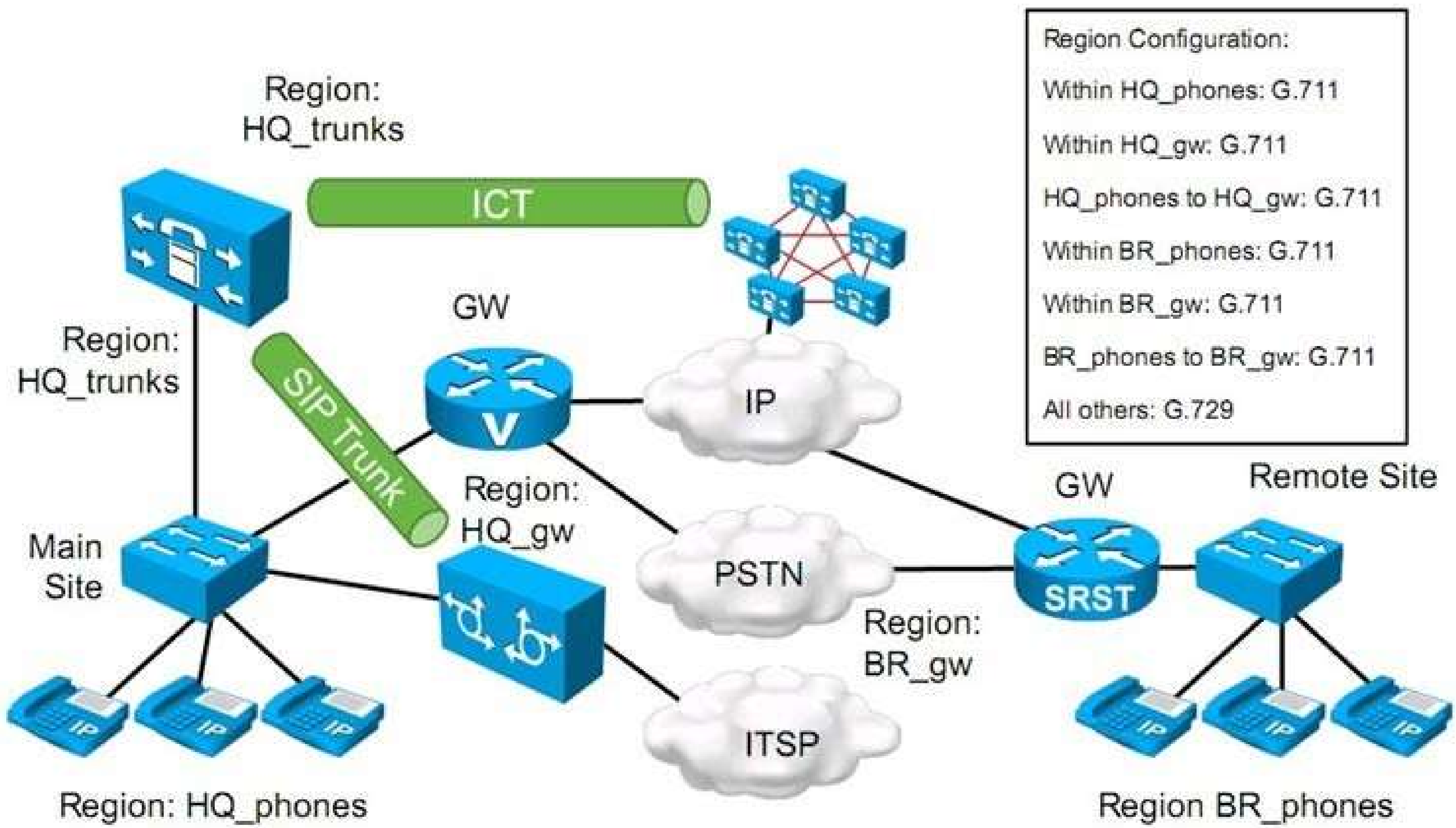
- Use high-bandwidth codecs in LAN environments.
- Use low-bandwidth codecs for IP WAN.
- Low-bandwidth codecs are designed for human speech:
  - Do not work well for other audio streams, such as music.
  - Use alternative methods for MOH:
    - Disable MOH for remote sites.
    - Use multicast MOH from branch router flash.

Review of Cisco Unified Communications Manager Codecs

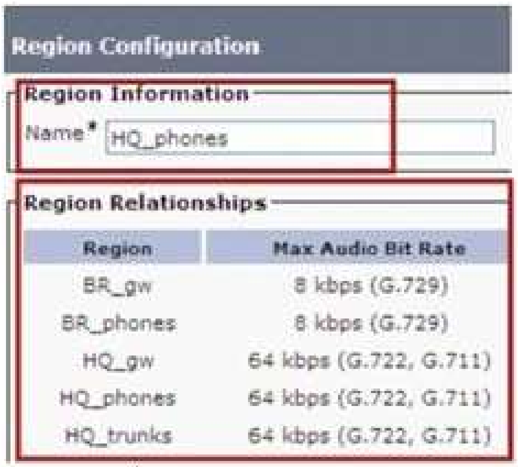
The codec that will be used depends on Cisco Unified Communications Manager region configuration:

- Each region is configured with the highest permitted codec bandwidth:
  - Within the configured region.
  - Toward specific other regions (manually added).
  - Toward all other regions (that have not been manually added)
- Region is assigned to a device pool.
- Device pool is assigned to a device.
- Codec that is actually used depends on capabilities of the two devices:
  - Best codec that is supported by both devices and does not exceed codec bandwidth permitted in region configuration.
  - If devices cannot agree on a codec, a transcoding device is invoked.
  - Loss type (configured in region) is also considered.

Example: Codec Configuration

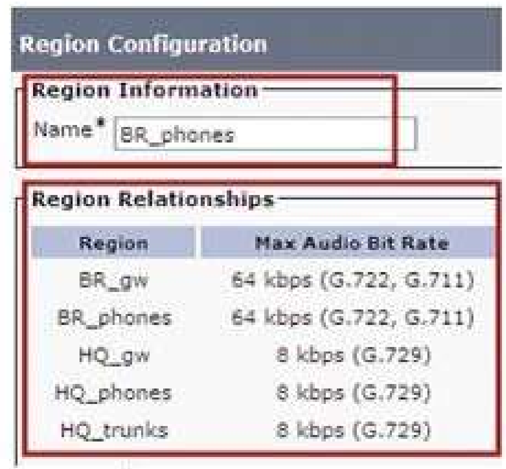


Example: Codec Configuration (Cont.)



Region	Max Audio Bit Rate
BR_gw	8 kbps (G.729)
BR_phones	8 kbps (G.729)
HQ_gw	64 kbps (G.722, G.711)
HQ_phones	64 kbps (G.722, G.711)
HQ_trunks	64 kbps (G.722, G.711)

HQ_phones uses G.711 within its own region and to region HQ_gw; to all other regions, G.729 is used.



Region	Max Audio Bit Rate
BR_gw	64 kbps (G.722, G.711)
BR_phones	64 kbps (G.722, G.711)
HQ_gw	8 kbps (G.729)
HQ_phones	8 kbps (G.729)
HQ_trunks	8 kbps (G.729)

BR_phones uses G.711 within its own region and to region BR_gw; to all other regions, G.729 is used.

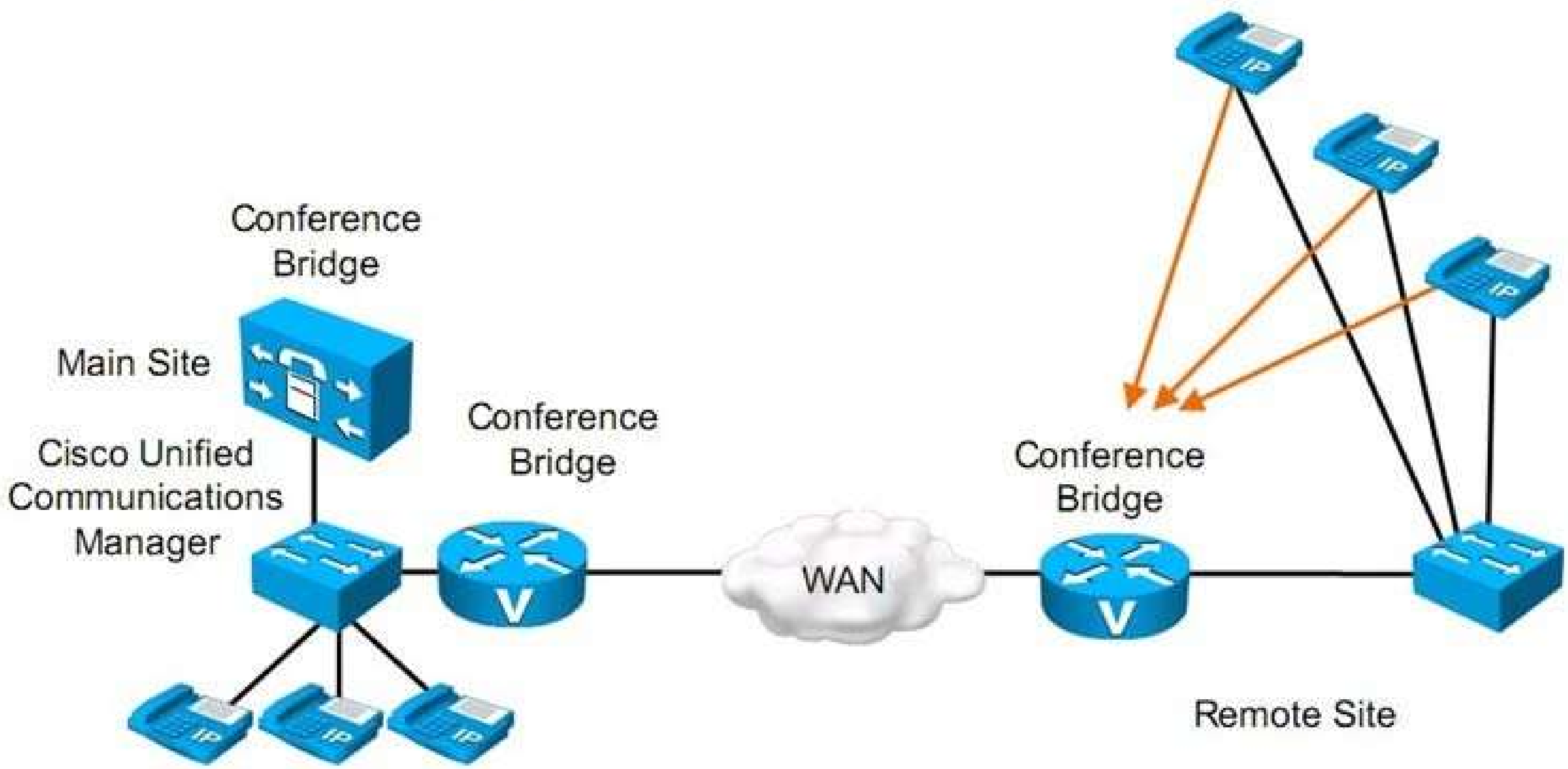
Local Conference Bridge Implementation

Considerations When Deploying Local Media Resources at Remote Sites

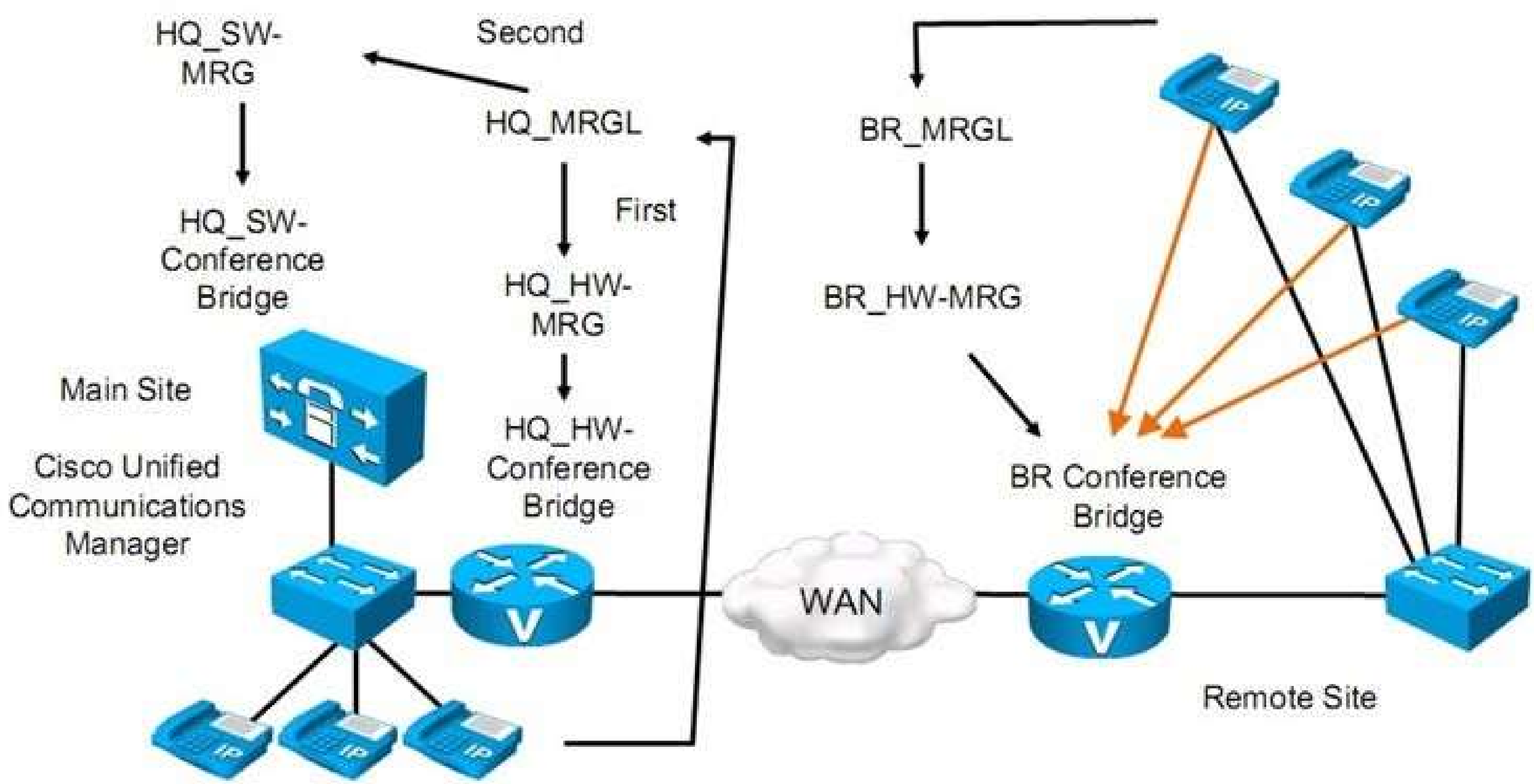
- Traffic stays off the IP WAN when remote site devices require a media resource—for example, conference bridges.
- Local media resources require appropriate hardware at remote site (DSPs in gateway).
- Efficiency depends on several factors:
  - Number of devices at remote site and likelihood of using a feature or application requiring the media resource
  - Ease of adding required hardware resources (DSPs)
  - Available bandwidth toward central site

Example: Implementing Local Conference Bridges at Two Sites

Phones at the remote site should use a local conference bridge when creating a conference.



Example: Implementing Local Conference Bridges at Two Sites (Cont.)



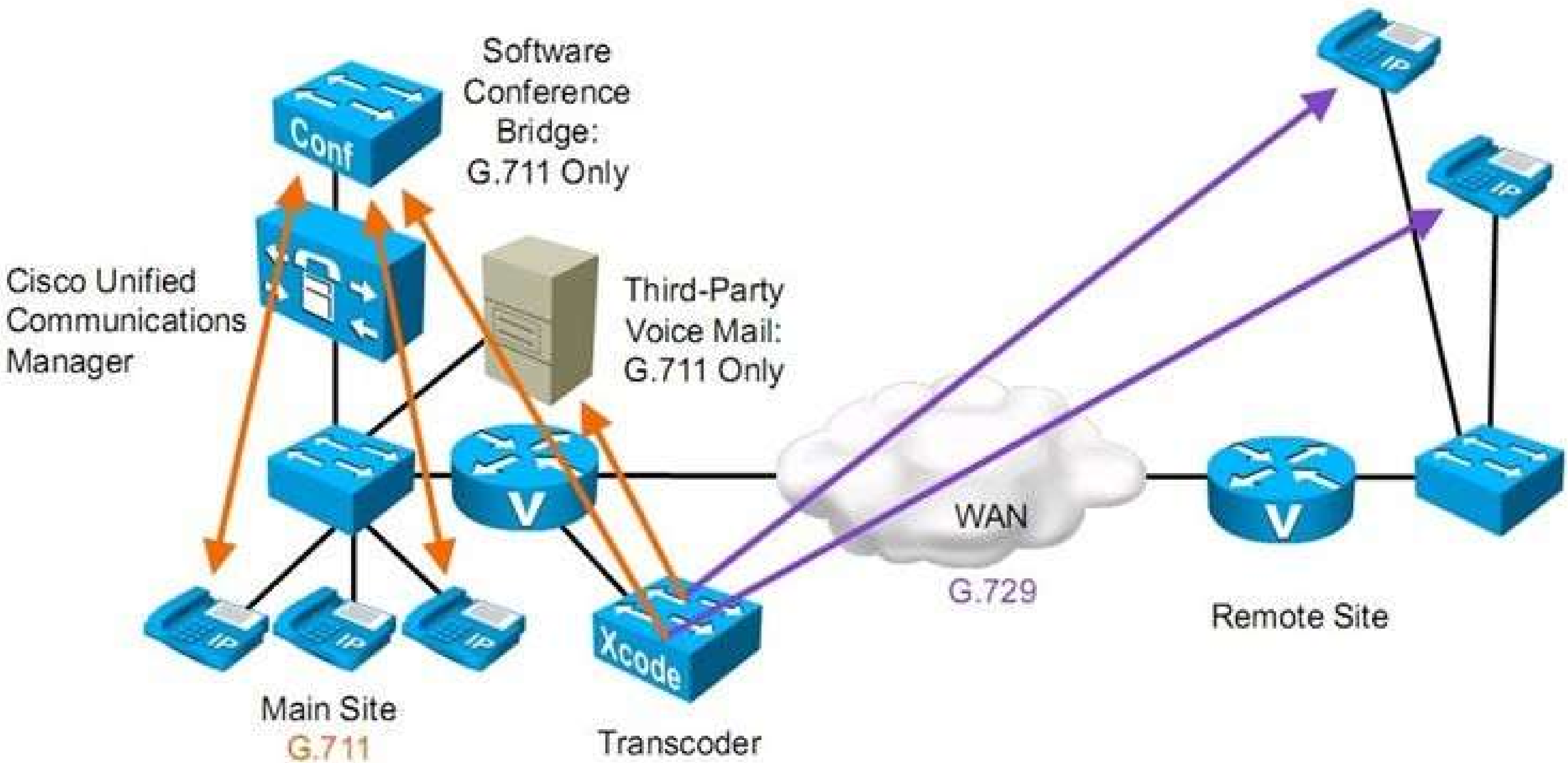
Transcoder Implementation

Considerations When Deploying Transcoders

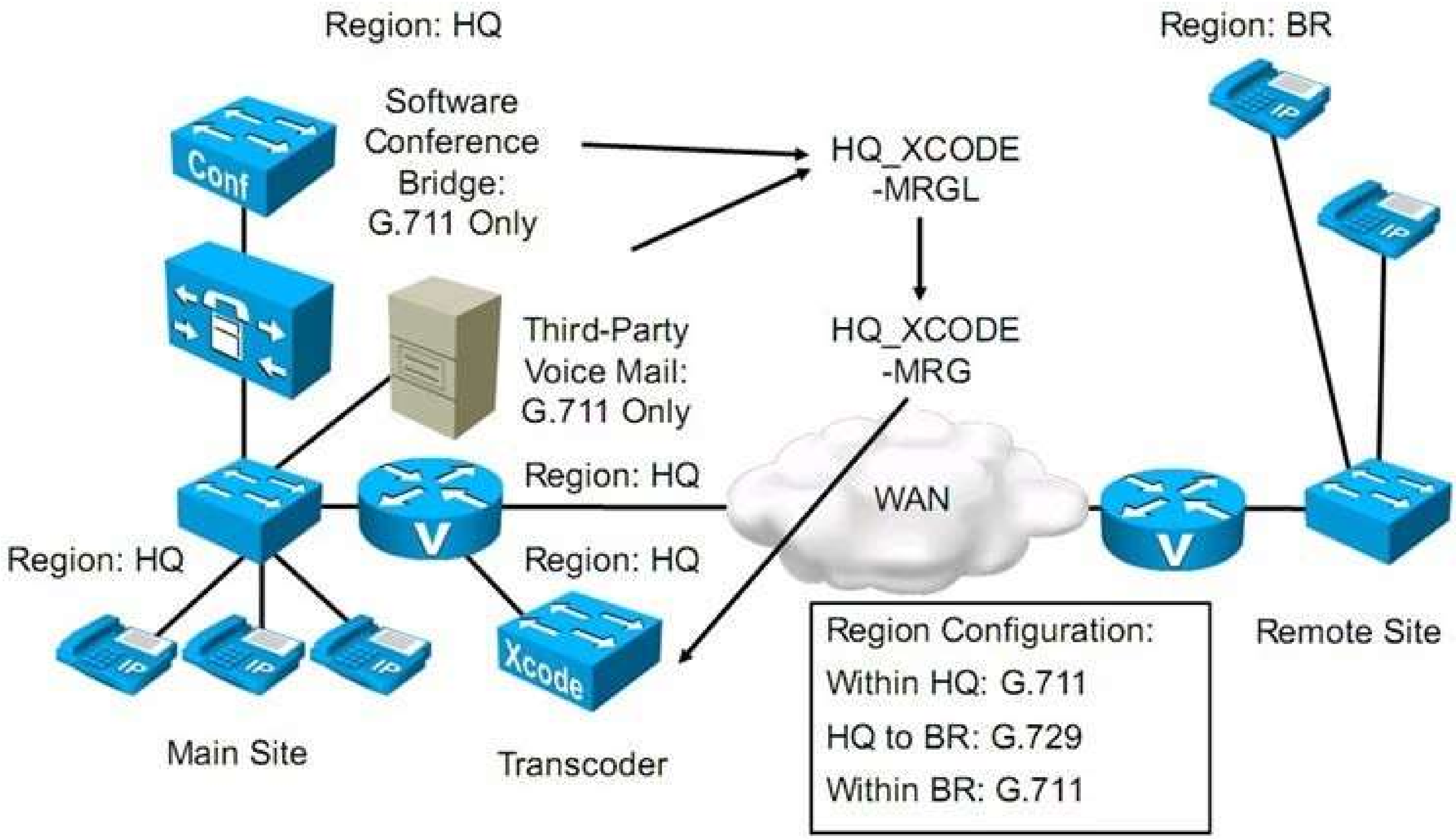
- Allows low-bandwidth codecs to be used in the IP WAN when one or both endpoints do not support low-bandwidth codecs:
  - Affected endpoint uses high-bandwidth codec toward the transcoder.
  - Transcoder changes voice stream from high-bandwidth codec to low-bandwidth codec.
  - Low-bandwidth codec stream is sent to other device (or transcoder) over the IP WAN.
- Efficiency depends on several factors:
  - Number of devices at remote site and likelihood of requiring a transcoder.
  - Ease of adding required hardware resources (DSPs).
  - Available bandwidth toward central site.

Example: Implementing a Transcoder at the Main Site

G.729 audio streams are transcoded at the main site before being passed on to endpoints supporting only G.711.



Example: Implementing a Transcoder at the Main Site (Cont.)



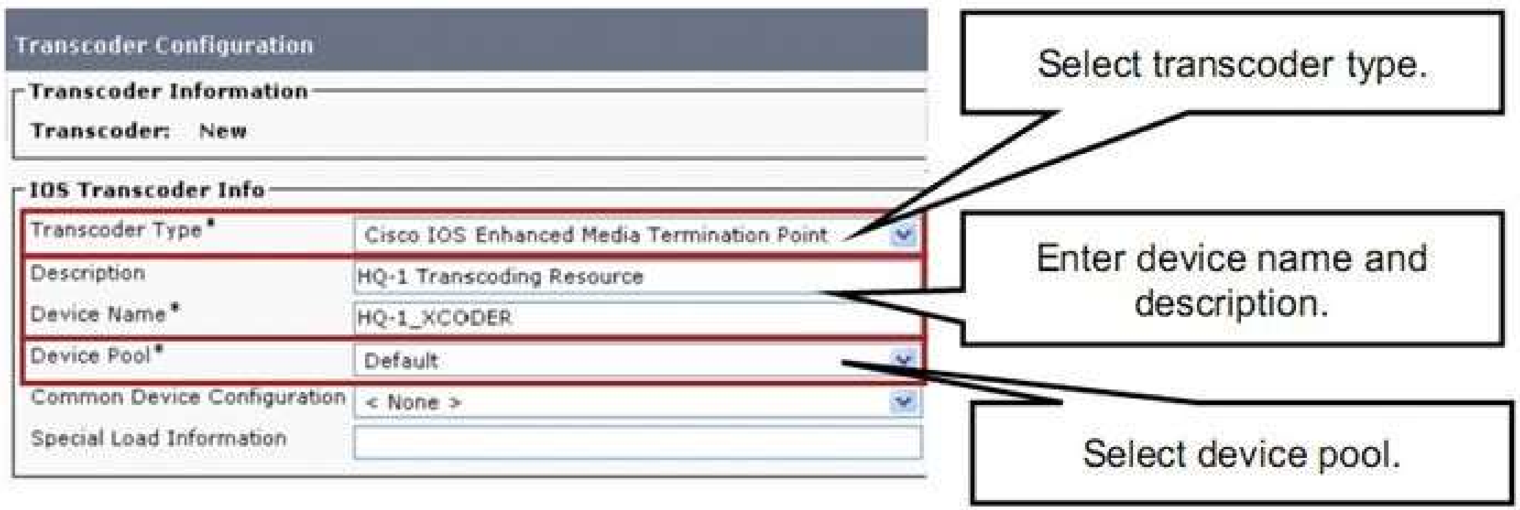
Configuration Procedure for Implementing Transcoders

1. Add transcoder resource in Cisco Unified Communications Manager.
2. Configure transcoder resource in Cisco IOS Software.
3. Configure media resource groups (MRGs).
4. Configure media resource group lists (MRGLs).
5. Assign MRGLs to devices.

Step 1: Add Transcoder Resource in Cisco Unified Communications Manager

You can freely choose the device name when using Cisco IOS Enhanced Media Termination Point hardware.

In all other cases, it must be MTP followed by MAC address.



Step 2: Configure Transcoder Resource in Cisco IOS Software

```
voice-card 0
 dspfarm
 dsp services dspfarm
!
sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 7.0+
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register HQ-1_XCODER
!
dspfarm profile 1 transcode
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 maximum sessions 2
 associate application SCCP
 no shutdown
```

Source Interface for Registration

Cisco Unified Communications Manager to Register With

The ccm ID has to match.

Name to Register With at Cisco Unified Communications Manager

The profile ID has to match.

For verification use:

```
show sccp
show sccp ccm group 1
show dspfarm profile 1
```