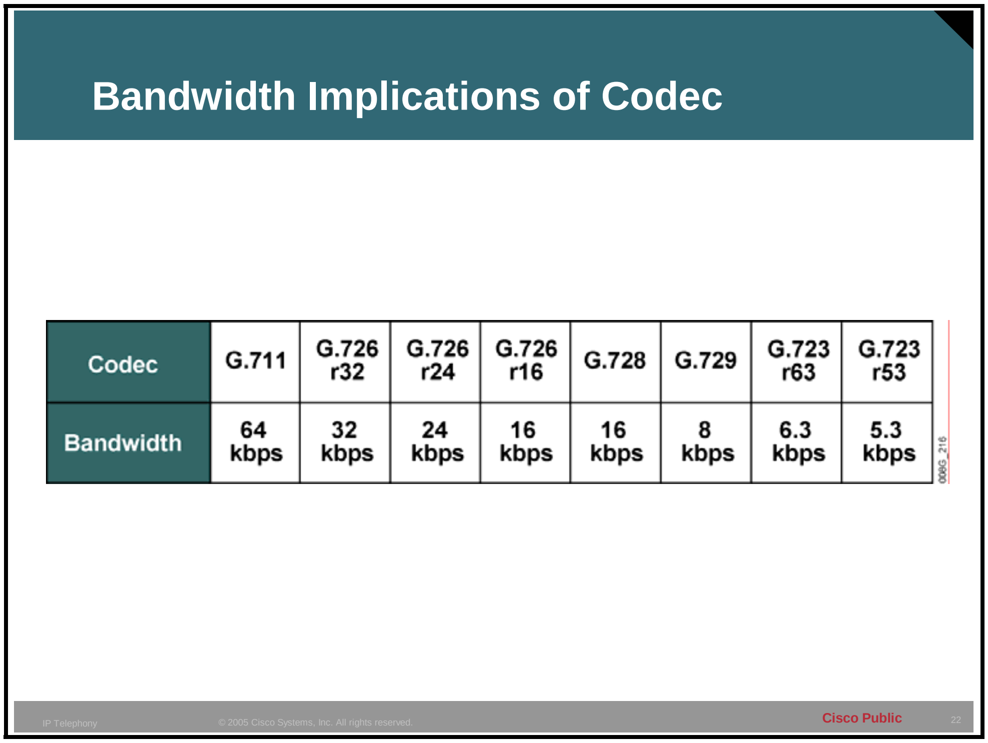
Calculating Bandwidth Requirements

Codec Bandwidths

This topic describes the bandwidth that each codec uses and illustrates its impact on total bandwidth.



Bandwidth Implications of Codec

Codec	G.711	G.726 r32	G.726 r24	G.726 r16	G.728	G.729	G.723 r63	G.723 r53
Bandwidth	64 kbps	32 kbps	24 kbps	16 kbps	16 kbps	8 kbps	6.3 kbps	5.3 kbps

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One of the most important factors for the network administrator to consider while building voice networks is proper capacity planning. Network administrators must understand how much bandwidth is used for each VoIP call. With a thorough understanding of VoIP bandwidth, the network administrator can apply capacity-planning tools.

Following is a list of codecs and their associated bandwidth:

- **G.711:** The G.711 pulse code modulation (PCM) coding scheme uses the most bandwidth. It takes samples 8000 times per second, each of which is 8 bits in length, for a total of 64,000 bps.
- **G.726:** The G.726 adaptive differential pulse code modulation (ADPCM) coding schemes use somewhat less bandwidth. While each coding scheme takes samples 8000 times per second like PCM, it uses 4, 3, or 2 bits for each sample, thereby resulting in total bandwidths of 32,000, 24,000, or 16,000 bps.
- **G.728:** The G.728 low-delay code excited linear prediction (LDCELP) coding scheme compresses PCM samples using codebook technology. It uses a total bandwidth of 16,000 bps.
- **G.729:** The G.729 and G.729A Conjugate Structure Algebraic Code Excited Linear Prediction (CS-ACELP) coding scheme also compresses PCM using advanced codebook technology. It uses 8000 bps total bandwidth.

- **G.723:** The G.723 and G.723A multipulse maximum likelihood quantization (MPMLQ) coding schemes use a look-ahead algorithm. These compression schemes result in 6300 or 5300 bps.

The network administrator should balance the need for voice quality against the cost of bandwidth in the network when choosing codecs. The higher the codec bandwidth, the higher the cost of each call across the network.

Impact of Voice Samples and Packet Size on Bandwidth

This topic illustrates the effect of voice sample size on bandwidth.

Impact of Voice Samples

Codec	Bandwidth	Sample Size	Packets
G.711	64000	240	33
G.711	64000	160	50
G.726r32	32000	120	33
G.726r32	32000	80	50
G.726r24	24000	80	25
G.726r24	24000	60	33
G.726r16	16000	80	25
G.726r16	16000	40	50
G.728	16000	80	13
G.728	16000	40	25
G.729	8000	40	25
G.729	8000	20	50
G.723r63	6300	48	16
G.723r63	6300	24	33
G.723r53	5300	40	17
G.723r53	5300	20	33

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Voice sample size is a variable that can affect total bandwidth used. A voice sample is defined as the digital output from a codec DSP that is encapsulated into a protocol data unit (PDU). Cisco uses DSPs that output samples based on digitization of 10 ms-worth of audio. Cisco voice equipment encapsulates 20 ms of audio in each PDU by default, regardless of the codec used. You can apply an optional configuration command to the dial peer to vary the number of samples encapsulated. When you encapsulate more samples per PDU, total bandwidth is reduced. However, encapsulating more samples per PDU comes at the risk of larger PDUs, which can cause variable delay and severe gaps if PDUs are dropped.

Example: Encapsulated Bytes Calculation

Using a simple formula, it is possible for you to determine the number of bytes encapsulated in a PDU based on the codec bandwidth and the sample size (20 ms is default):

Bytes_per_Sample = (*Sample_Size* ∗ *Codec_Bandwidth*) / 8

If you apply G.711 numbers, the formula reveals the following:

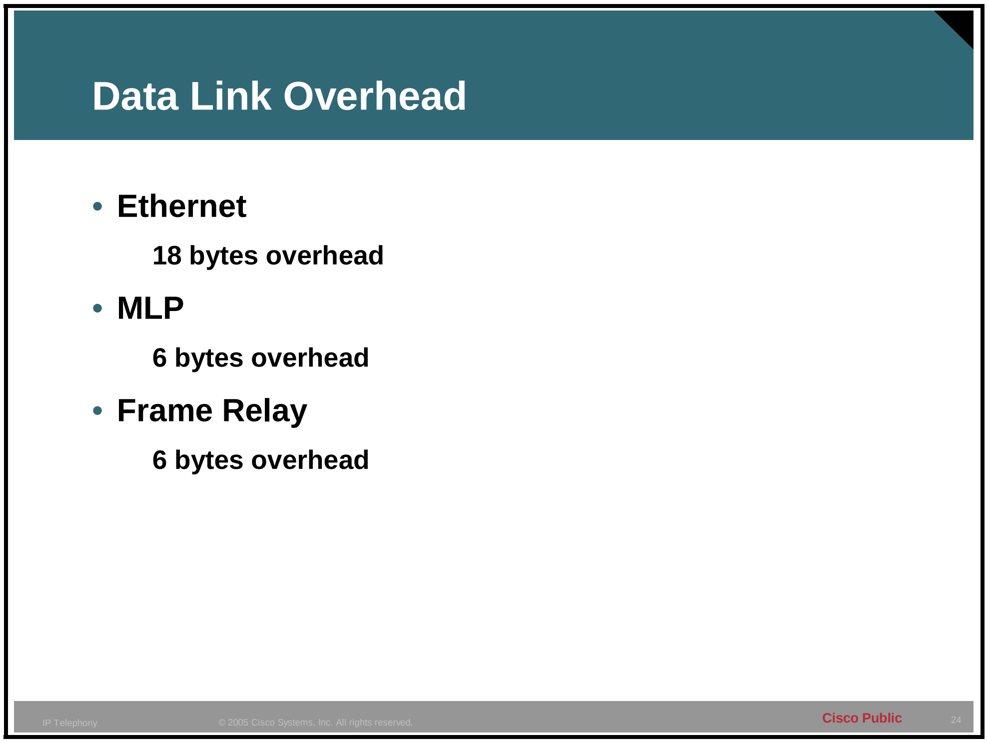
Bytes_per_Sample = (.020 ∗ 64000) / 8

Bytes_per_Sample = 160

The figure illustrates various codecs and sample sizes and the number of packets that are required for VoIP to transmit one second of audio. The larger the sample size, the larger the packet, and the fewer the encapsulated samples that have to be sent (which reduces bandwidth).

Data Link Overhead

This topic lists overhead sizes for various Layer 2 protocols.

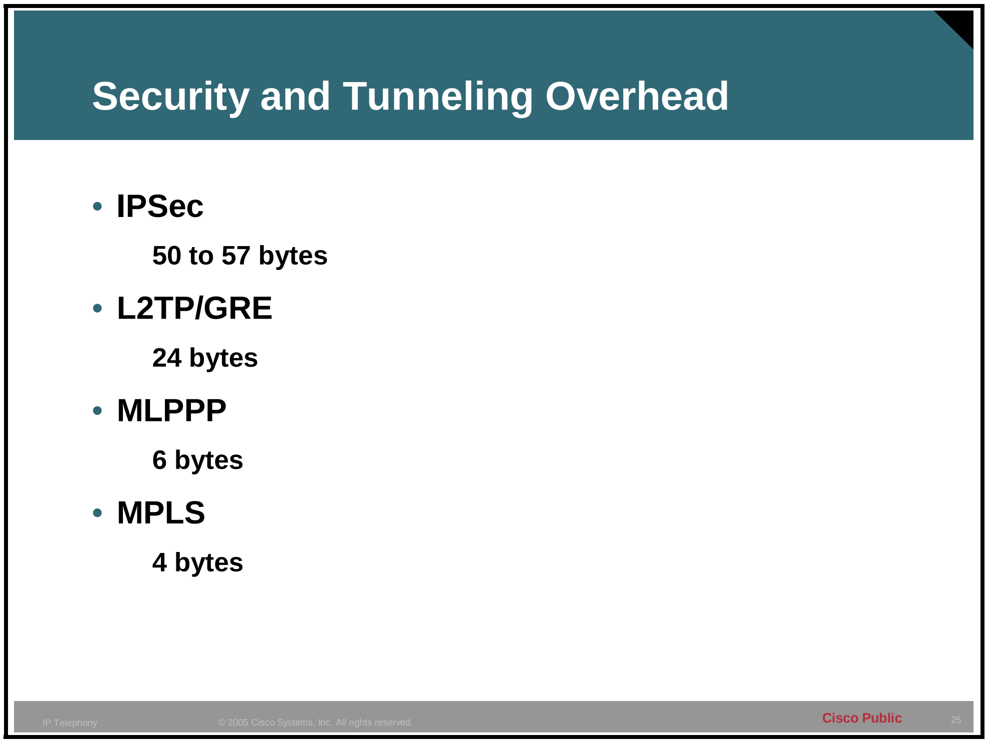


Another contributing factor to bandwidth is the Layer 2 protocol used to transport VoIP. VoIP alone carries a 40-byte IP/UDP/RTP header, assuming uncompressed RTP. Depending on the Layer 2 protocol used, the overhead could grow substantially. The larger the Layer 2 overhead, the more bandwidth required to transport VoIP. The following points illustrate the Layer 2 overhead for various protocols:

- **Ethernet II:** Carries 18 bytes of overhead; 6 bytes for source MAC, 6 bytes for destination MAC, 2 bytes for type, and 4 bytes for cyclic redundancy check (CRC)
- **Multilink Point-to-Point Protocol (MLP):** Carries 6 bytes of overhead; 1 byte for flag, 1 byte for address, 2 bytes for control (or type), and 2 bytes for CRC
- **FRF.12:** Carries 6 bytes of overhead; 2 bytes for data-link connection identifier (DLCI) header, 2 bytes for FRF.12, and 2 bytes for CRC

Security and Tunneling Overhead

This topic describes overhead associated with various security and tunneling protocols.



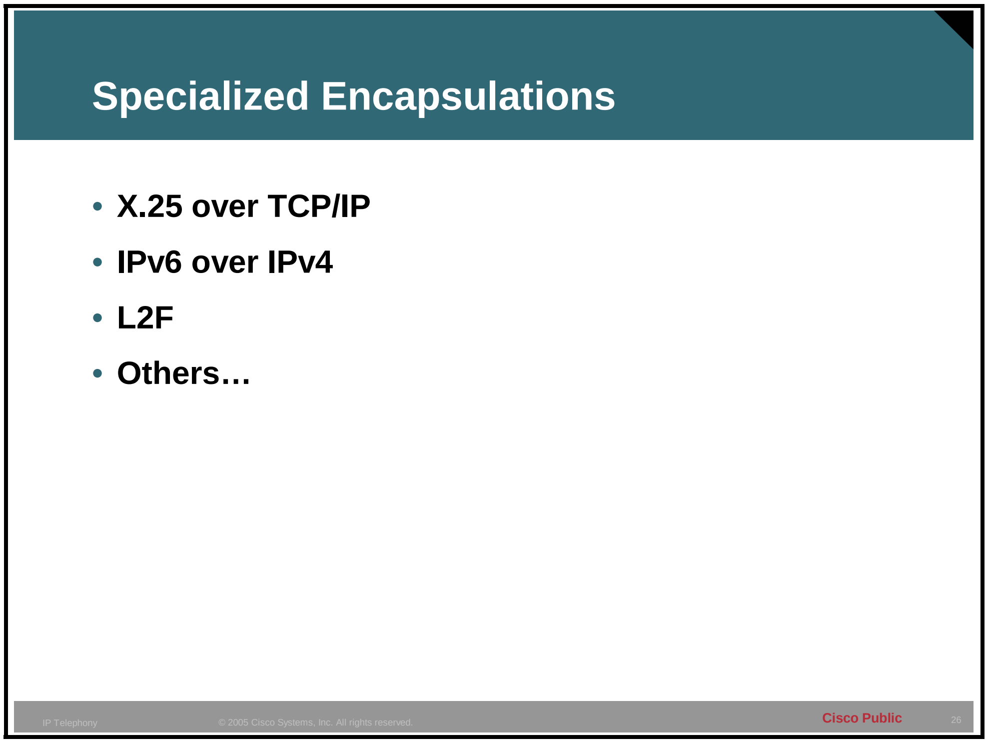
Certain security and tunneling encapsulations will also add overhead to voice packets and should be considered when calculating bandwidth requirements. When using a Virtual Private Network (VPN), IP Security (IPSec) will add 50 to 57 bytes of overhead, a significant amount when considering small voice packets. Layer 2 Tunneling Protocol/generic routing encapsulation (L2TP/GRE) adds 24 bytes. When using MLP, 6 bytes will be added to each packet. Multiprotocol Label Switching (MPLS) adds a 4-byte label to every packet. All of these specialized tunneling and security protocols must be considered when planning for bandwidth demands.

Example: VPN Overhead

Many companies have their employees telecommute from home. These employees initiate a VPN connection into their enterprise for secure Internet transmission. When deploying a remote telephone at the employee's home using a router and a PBX Off-Premises eXtension (OPX), the voice packets will experience additional overhead associated with the VPN.

Specialized Encapsulations

This topic describes considerations for specialized encapsulations for VoIP.



There exist many other encapsulations to consider when transporting VoIP. Specialized encapsulations include protocol-specific encapsulation such as X.25, experimental encapsulations such as IPv6 over IPv4, Layer 2 Forwarding (L2F) Protocol, and other vendor-specific encapsulations. Each must be considered when calculating total bandwidth.

Calculating the Total Bandwidth for a VoIP Call

This topic calculates the total bandwidth required for a VoIP call using codec, data link, and sample size.

Total Bandwidth Required

Codec	Codec Speed	Sample Size	Frame Relay	Frame Relay with CRTP	Ethernet	Ethernet with CRTP
G.711	64000	240	76267	66133	78933	68800
G.711	64000	160	82400	67200	86400	71200
G.726r32	32000	120	44267	34133	46933	36800
G.726r32	32000	80	50400	35200	54400	39200
G.726r24	24000	80	37800	26400	40800	29400
G.726r24	24000	60	42400	27200	46400	31200
G.726r16	16000	80	25200	17600	27200	19600
G.726r16	16000	40	34400	19200	38400	23200
G.728	16000	80	25200	17600	27200	19600
G.728	16000	40	34400	19200	38400	23200
G.729	8000	40	17200	9600	19200	11600
G.729	8000	20	26400	11200	30400	15200
G.723r63	6300	48	12338	7350	13650	8663
G.723r63	6300	24	18375	8400	21000	11025
G.723r53	5300	40	11395	6360	12720	7685
G.723r53	5300	20	17490	7420	20140	10070

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Codec choice, data-link overhead, sample size, and compressed RTP have positive and negative impacts on total bandwidth. To perform the calculations, you must consider these contributing factors as part of the equation:

- More bandwidth required for the codec = more total bandwidth required
- More overhead associated with the data link = more total bandwidth required
- Larger sample size = less total bandwidth required
- Compressed RTP = significantly reduced total bandwidth required

Example: Total Bandwidth Calculation

The following calculation was used to produce the figure:

Total_Bandwidth = ([Layer_2_Overhead + IP_UDP_RTP Overhead + Sample_Size] / Sample_Size) * Codec_Speed

For example, assume a G.729 codec, 20-byte sample size, using Frame Relay without CRTP:

Total_Bandwidth = ([6 + 40 +20]/20) * 8000

Total_Bandwidth = 26,400 bps

Effects of VAD on Bandwidth

This topic describes the effect of voice activity detection (VAD) on total bandwidth.

Effect of VAD

Codec	Codec Speed	Sample Size	Frame Relay	Frame Relay with VAD
G.711	64000	240	76267	49573
G.711	64000	160	82400	53560
G.726r32	32000	120	44267	28773
G.726r32	32000	80	50400	32760
G.726r24	24000	80	37800	24570
G.726r24	24000	60	42400	27560
G.726r16	16000	80	25200	16380
G.726r16	16000	40	34400	22360
G.728	16000	80	25200	16380
G.728	16000	40	34400	22360
G.729	8000	40	17200	11180
G.729	8000	20	26400	17160
G.723r63	6300	48	12338	8019
G.723r63	6300	24	18375	11944
G.723r53	5300	40	11395	7407
G.723r53	5300	20	17490	11369

On average, an aggregate of 24 calls or more may contain 35 percent silence. With traditional telephony voice networks, all voice calls use 64-kbps fixed-bandwidth links regardless of how much of the conversation is speech and how much is silence. In Cisco VoIP networks, all conversations and silences are packetized. VAD suppresses packets of silence. Instead of sending VoIP packets of silence, VoIP gateways interleave data traffic with VoIP conversations to more effectively use network bandwidth.

VAD provides a maximum of 35 percent bandwidth savings based on an average volume of more than 24 calls.

Note Bandwidth savings of 35 percent is an average figure and does not take into account loud background sounds, differences in languages, and other factors.

The savings are not realized on every individual voice call, or on any specific point measurement.

Note For the purposes of network design and bandwidth engineering, VAD should *not* be taken into account, especially on links that will carry fewer than 24 voice calls simultaneously.

Various features, such as music on hold (MOH) and fax, render VAD ineffective. When the network is engineered for the full voice call bandwidth, all savings provided by VAD are available to data applications.

VAD is enabled by default for all VoIP calls. VAD reduces the silence in VoIP conversations but it also provides comfort noise generation (CNG). Because you can mistake silence for a disconnected call, CNG provides locally generated *white noise* to make the call appear normally connected to both parties.

Example: VAD Bandwidth Savings

The figure shows examples of the VAD effect in a Frame Relay VoIP environment. In the example using G.711 with a 160-byte payload, the bandwidth required is 82,400 bps. By turning VAD on, you can reduce the bandwidth utilization to 53,560 bps. This is a bandwidth savings of 35 percent.

