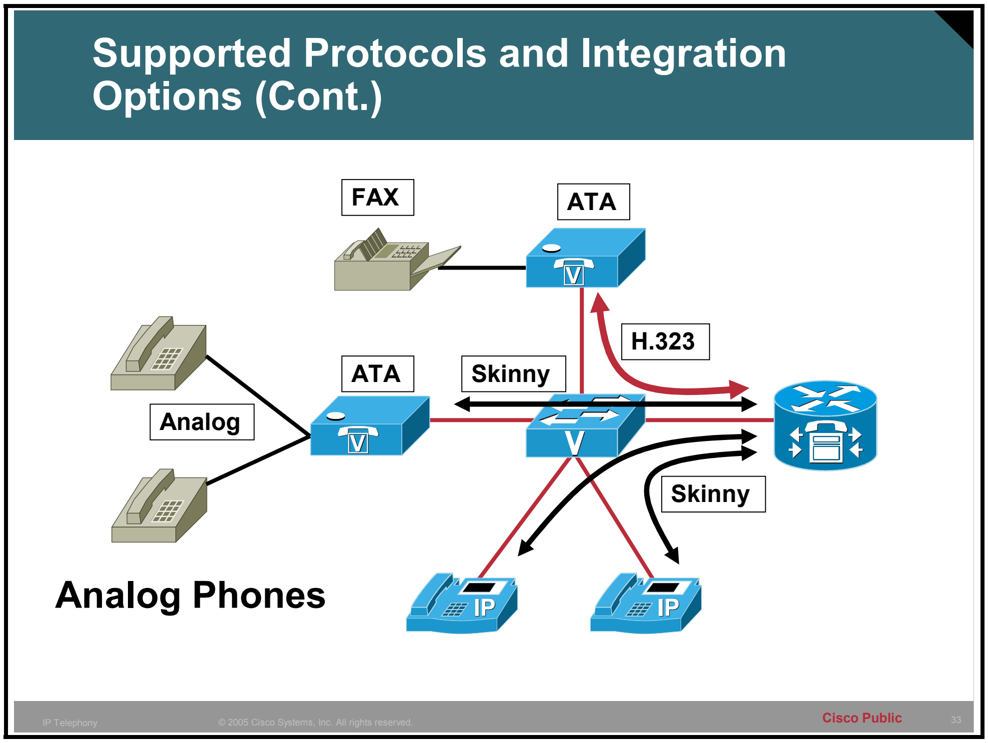
Cisco CME Features and Functionality

Supported Protocols and Integration Options

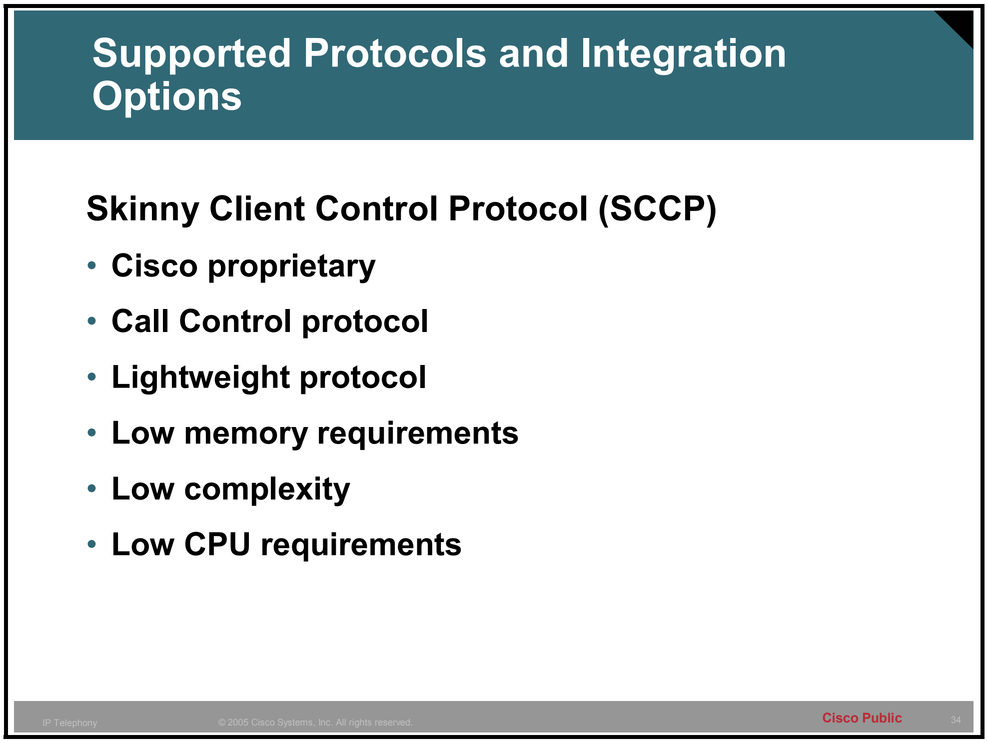
This topic describes the supported protocols and integration options of Cisco CME.



Cisco CME can use both H.323 and the Skinny protocol to control IP phones, analog phones, and faxes.

Supported Protocols and Integration Options: Skinny Client Control Protocol (SCCP)

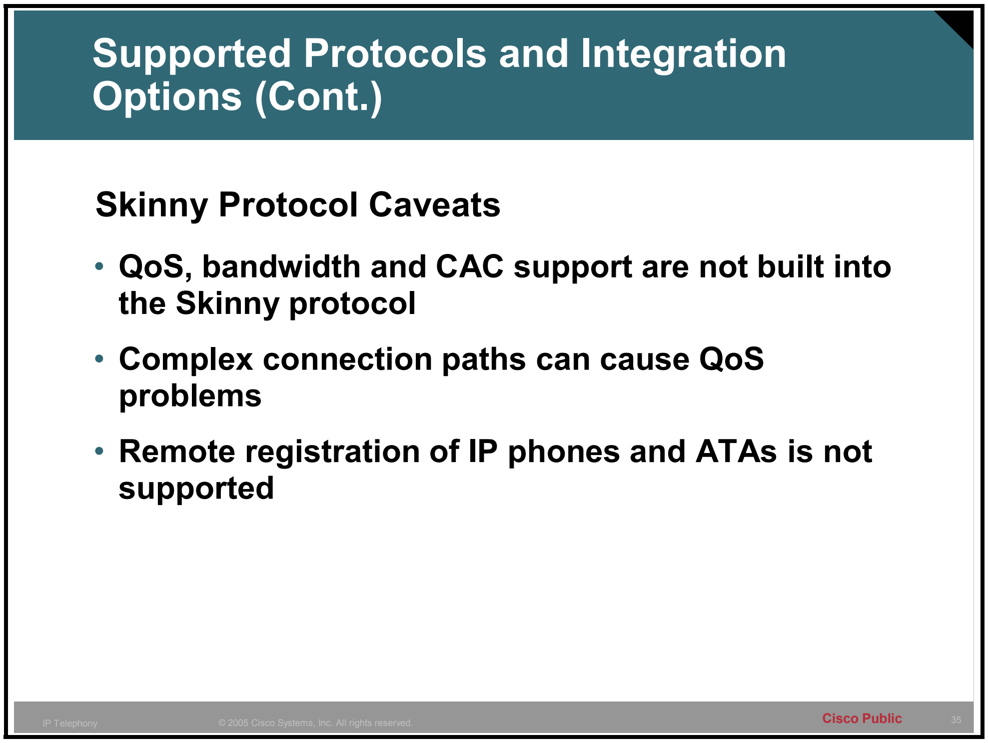
This topic describes the supported protocols and integration options of Cisco CME.



Cisco CME software provides call processing for IP Phones using the Skinny Client Control Protocol (SCCP). SCCP is the Cisco proprietary protocol for real-time calls and conferencing over IP. This generalized messaging set allows Cisco IP Phones to coexist in an H.323 environment. Savings in memory size, processor power, and complexity are benefits of SCCP.

Supported Protocols and Integration Options: Skinny Protocol Caveats

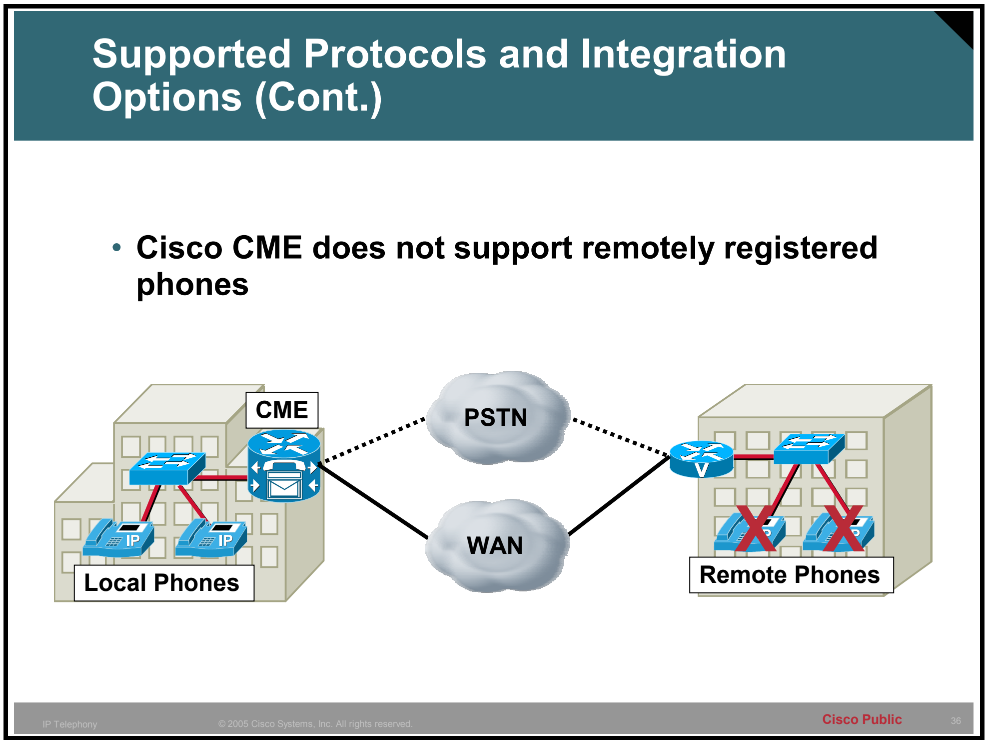
This topic describes the supported protocols and integration options of Cisco CME.



All IP phones must be connected locally to the Cisco CME router because of the factors shown here.

QoS, bandwidth management, and Call Admission Control (CAC) are not supported within the Skinny protocol context on Cisco CME. Complex connection paths could cause QoS problems.

Compressed Real-time Transport Protocol (CRTP) is not supported.

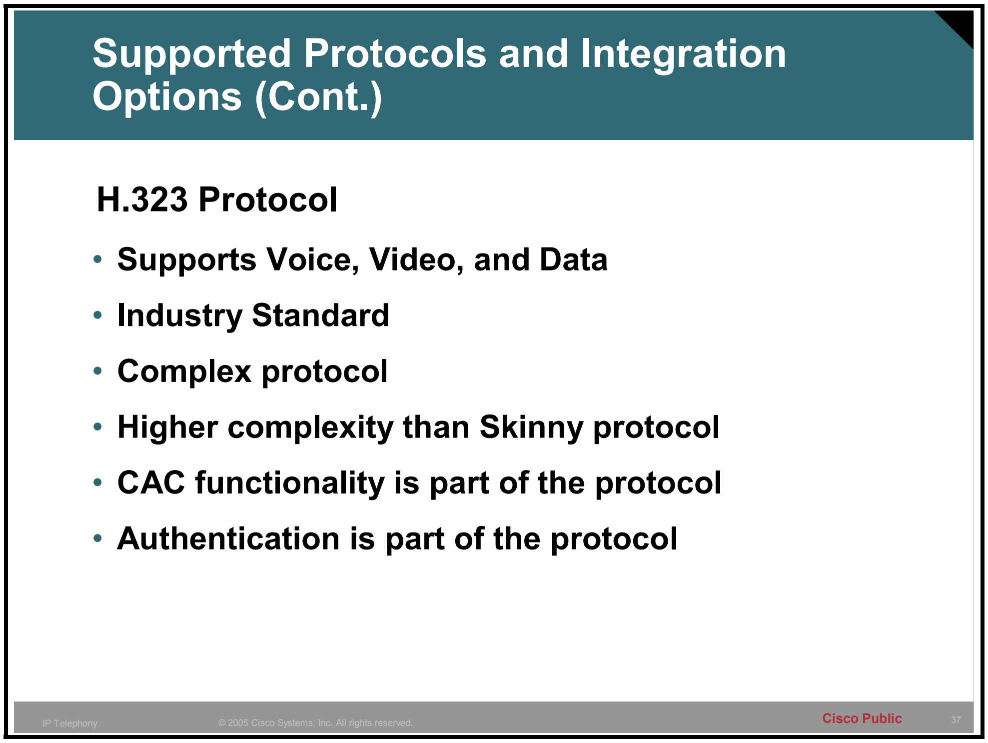


Cisco CME does not support remotely registered phones via a WAN or virtual private network (VPN) connection because the Skinny interface does not have the necessary set of QoS tools; these tools have been built into the H.323/VoIP interface to cope with operating across non-local networks. Cisco CME also does not support bandwidth control or accounting, RSVP, or the max-conn attribute for remotely registered SCCP phones via a WAN or virtual private network (VPN) connection.

Each remote site should have a Cisco CME router so IP phones can register locally. VoIP interworking between multiple Cisco CME routers across the WAN is supported via the H.323 protocol.

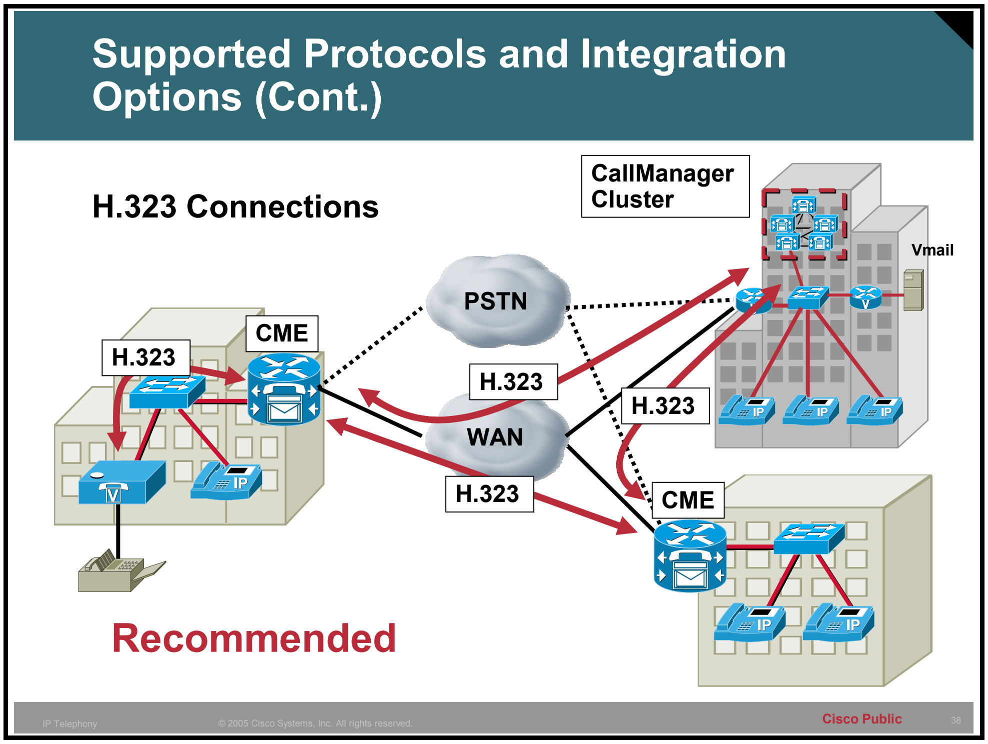
Supported Protocols and Integration Options: H.323 Protocol

This topic describes the supported protocols and integration options of Cisco CME.



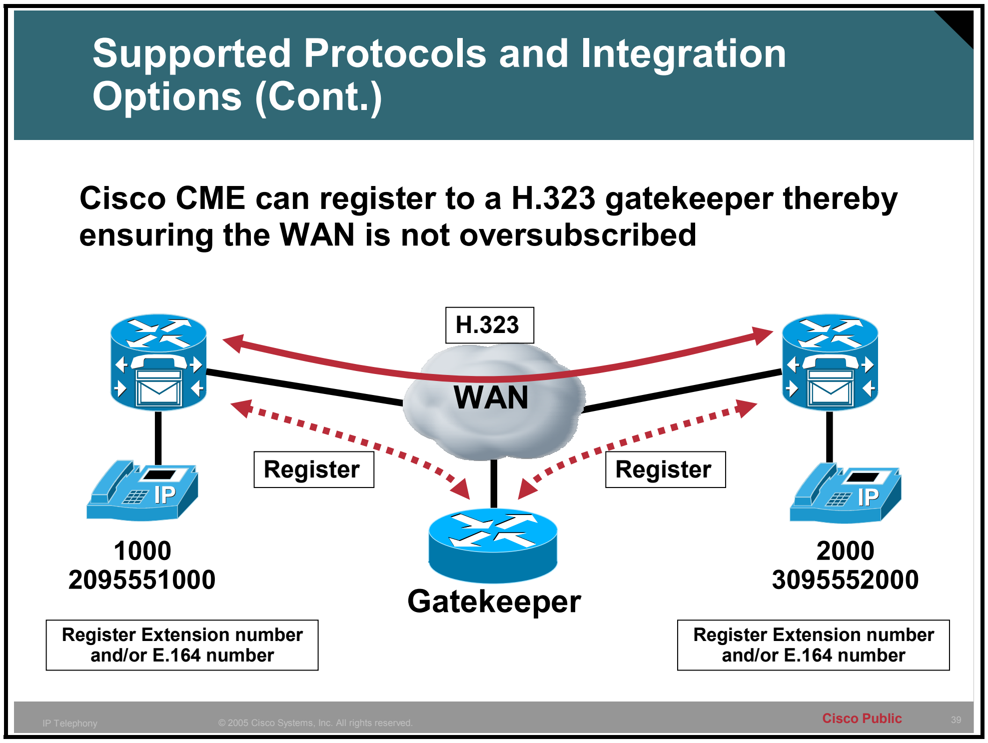
H.323 is a specification for transmitting audio, video, and data across an IP network, including the Internet. H.323 is an extension of the ITU Telecommunication Standardization Sector standard H.320.

Tip	The ATA will need to be configured with H.323 when fax machines are connected to the analog ports.



H.323 is a specification for transmitting audio, video, and data across an IP network, including the Internet. H.323 is an extension of the ITU Telecommunication Standardization Sector standard H.320.

In this slide, the H.323 protocol is used to connect the Cisco CME router together and for controlling the analog fax connected to the ATA.



The Cisco CME system can be configured to register the ephone-dns with a H.323 Gatekeeper. In addition, the IP phone may have both an extension number and an E.164 number defined, and one or both of the numbers may be registered with the H.323 Gatekeeper. H.323 can also be used to allow one Cisco CME to communicate with another Cisco CME or Voice Gateways. A router separate from Cisco CME must be used if gatekeeper is going to be configured.

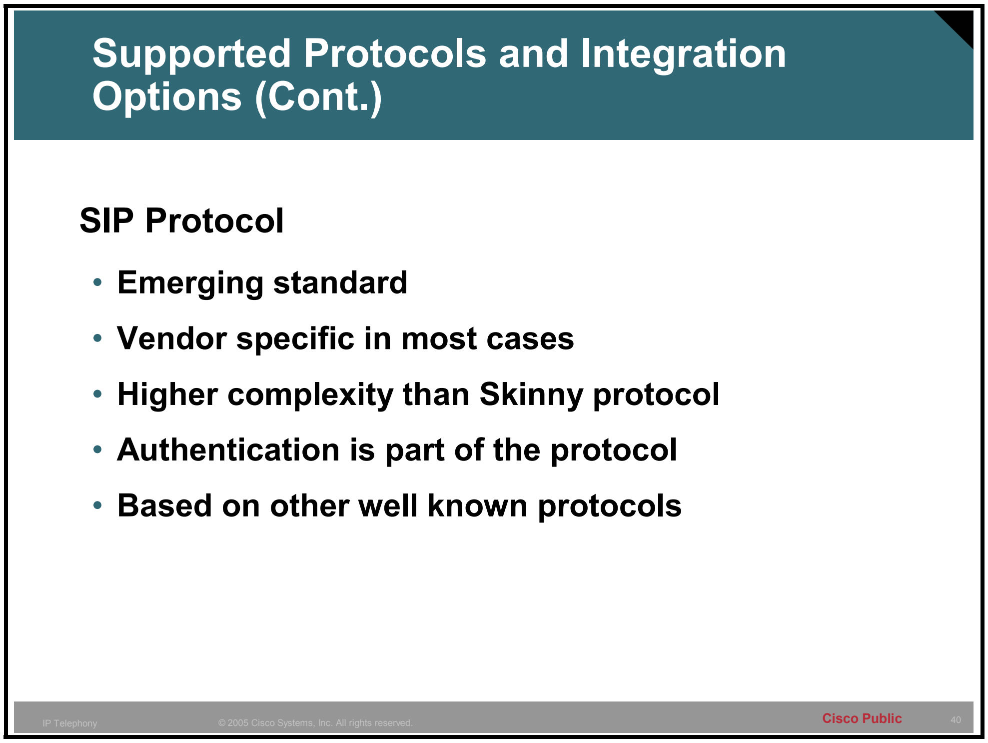
The H.323 Gatekeeper can provide the following functions:

- **CAC –** Call Admission Control over a WAN link to ensure that the WAN link is not oversubscribed
- **Dial plan administration -** Centralizing the dial plan for inter-site numbering
- **IP-to-IP Gateway –** Provides a network to network point for billing, security, and for joining two VoIP call legs together

Please refer to other Cisco documentation for details on Cisco Gatekeepers.

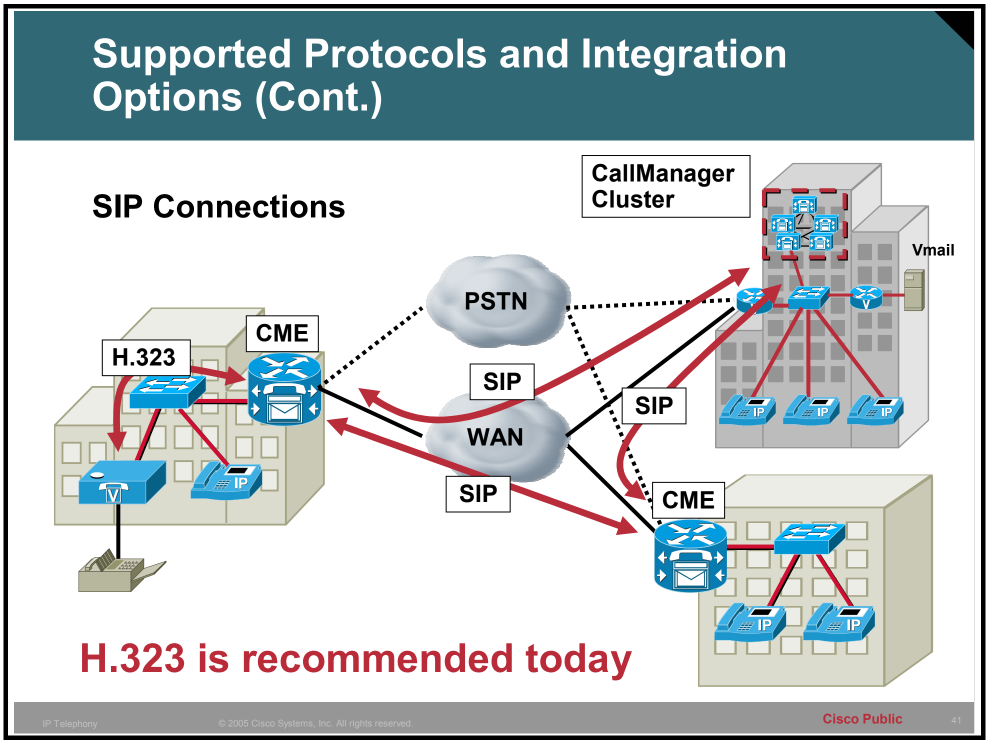
Supported Protocols and Integration Options: SIP Protocol

This topic describes the supported protocols and integration options of Cisco CME.



SIP was designed as a multimedia protocol that could take advantage of the architecture and messages found in popular Internet applications. By using a distributed architecture—with URLs for naming and text-based messaging—SIP attempts to take advantage of the Internet model for building VoIP networks and applications. In addition to VoIP, SIP is used for videoconferencing and instant messaging.

As a protocol, SIP only defines how sessions are to be set up and torn down. It utilizes other IETF protocols to define other aspects of VoIP and multimedia sessions, such as SDP for capabilities exchange, URLs for addressing, Domain Name System (DNS) for service location, and Telephony Routing over IP (TRIP) for call routing.



The SIP protocol can be used to connect calls between two Cisco CME systems. This is currently not the recommended solution and vendor compatibility is problematic.

Note	It is recommended to use H.323 to connect Cisco CME systems together.

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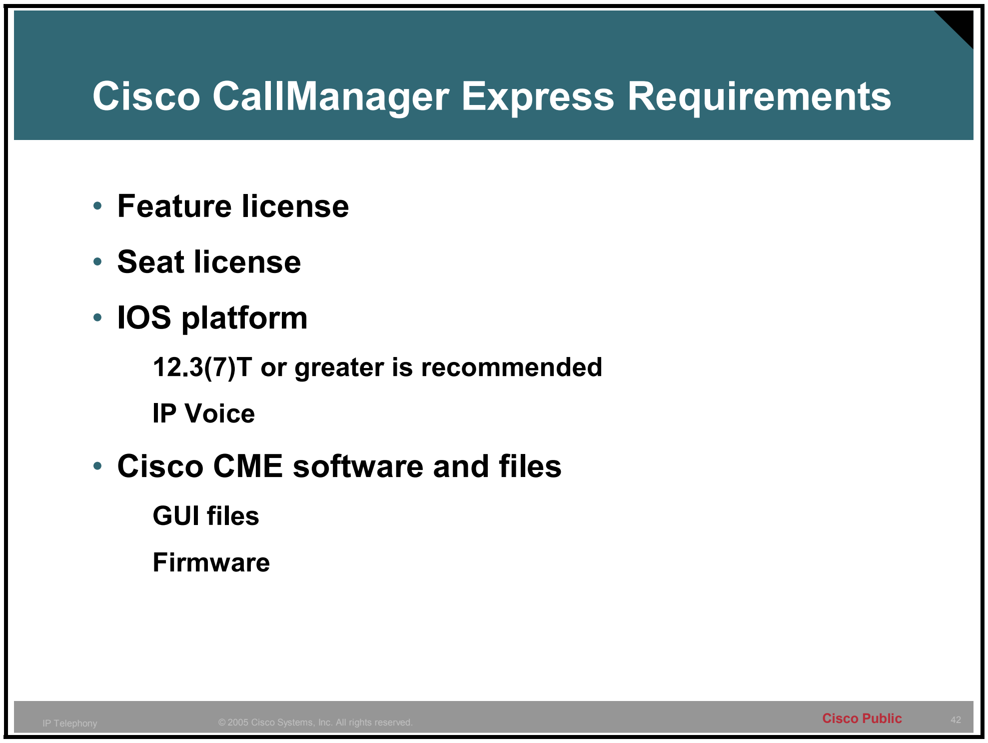
Cisco CallManager Express Requirements

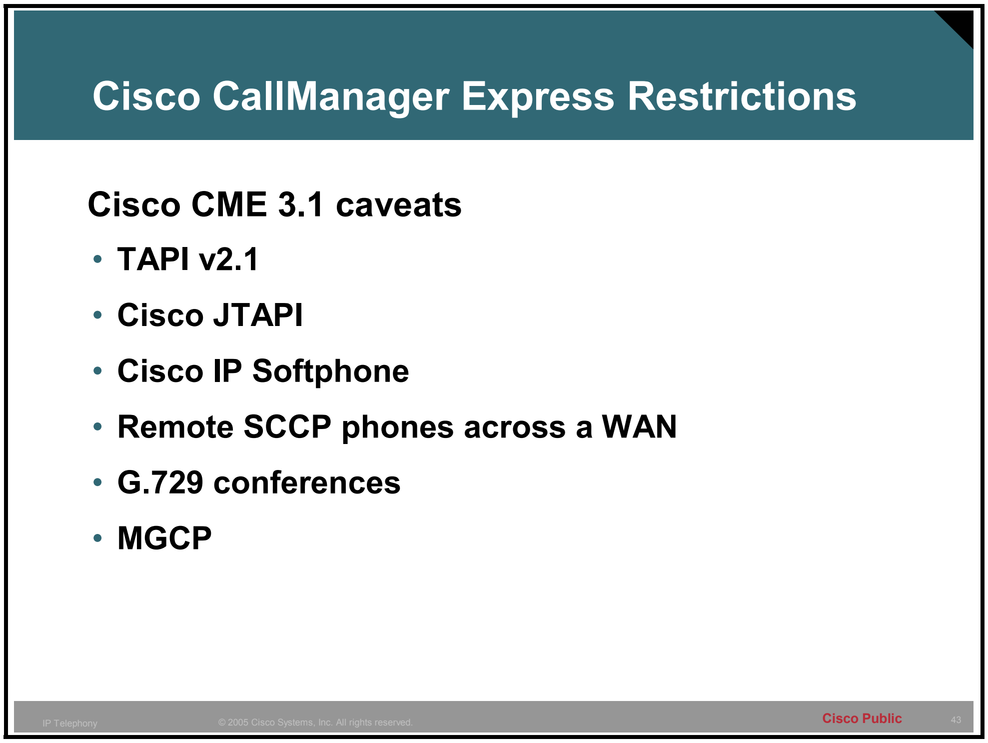
This topic describes Cisco CME requirements.



Cisco CME requires a Cisco CME feature license. This is licensed based on the number of IP phones that will be deployed. The router itself will need to have the correct IOS that is Cisco CME-capable. Each IP Phone or ATA port also requires a Cisco CME seat license, which can be purchased with the IP phone. You also need an account on Cisco.com to download Cisco CME files, such as phone firmware and GUI files and firmware.

Cisco CallManager Express Restrictions

This topic describes Cisco CME restrictions.



There is subset or TAPI 2.1 support in the Cisco CME. This will be covered in detail on the next page. Cisco JTAPI is not currently supported and this limitation restricts the use of a Cisco IP Softphone. The newer softphone called IP communicator is also not currently supported although it may be in future versions. Currently only third party softphones from IP Blue will work with the Cisco CME.

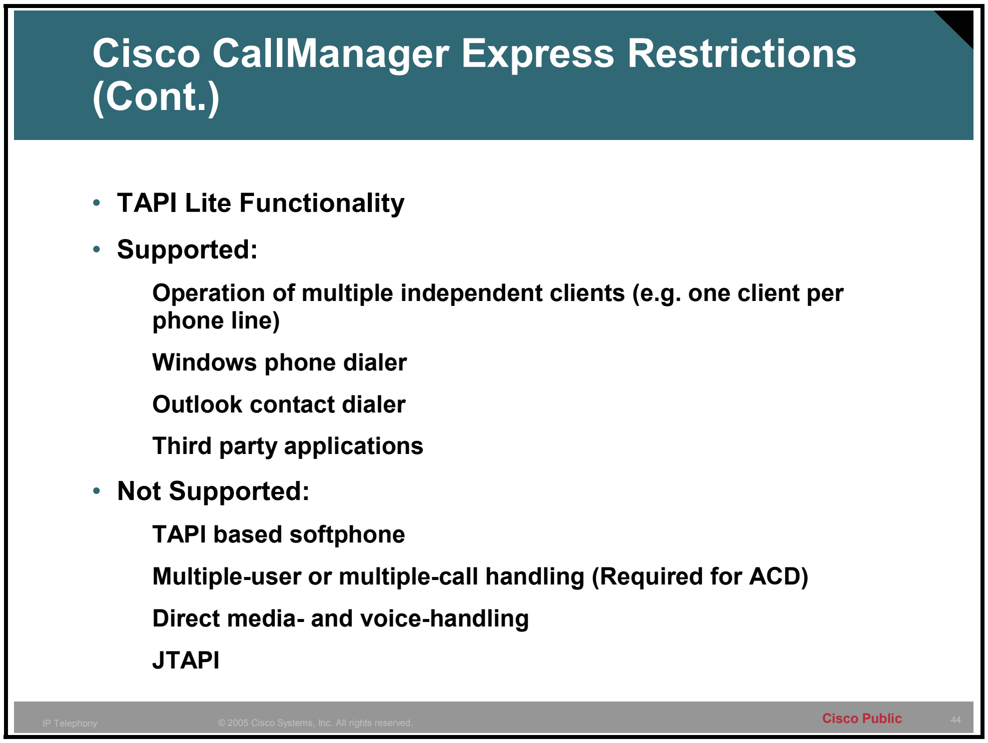
There are some restrictions when working with Cisco CME. The Cisco CME supports only phones that are local to the Cisco CME LAN and does not support remote SCCP phones that are connected across WAN links. The Cisco CME system and IP phones support the G.711 and G.729 codec. However, only the G.711 codec is supported for conferencing. This is due to a lack of support for hardware Digital Signal Processing (DSP)-based transcoding. This should be available in future versions of Cisco CME.

Media Gateway Control Protocol (MGCP) is not supported in Cisco CME.

Note	Upcoming releases of Cisco CME will support transcoding and IP communicator.

Cisco CallManager Express Restrictions: TAPI Lite Functionality

This topic describes Cisco CME restrictions.



Cisco CME does not support TAPI v2.1. Cisco CME TAPI implements only a small subset of TAPI functionality. It does support operation of multiple independent clients (for example, one client per phone line) but not full support for multiple-user or multiple-call handling, which is required for complex features such as automatic call distribution (ACD).

Applications like Windows phone dialer and the Outlook contact dialer can use TAPI Lite to dial, place on hold, transfer, and terminate a call on an associated line on an IP phone. JTAPI is not supported and neither are TAPI-based softphones. TAPI Lite allows for the control of a line on an associated PC but not for the termination of voice on the PC.

Note	Third-party applications can be developed that take advantage of TAPI Lite to control a line.

