Cisco Enhanced Conferencing and Transcoding for Voice Gateway Routers

Cisco® Enhanced Conferencing and Transcoding for Voice Gateway Routers provides conferencing and transcoding capabilities in Cisco IOS® Software-based gateways using the onboard Cisco Packet Voice/Fax Digital Signal Processor Modules (PVDM2 / PVDM3) on the Cisco 2800, 3800, 2900 and 3900 series voice gateway routers. This capability is also supported on Cisco voice gateway router platforms using the Cisco IP Communications Voice/Fax Network Module and the Cisco IP Communications High-Density Digital Voice/Fax Network Module. This feature is delivered in Cisco IOS Software and operates in conjunction with Cisco Unified Communications Manager (CUCM) or Cisco Unified Communications Manager Express (CUCME) or Cisco Unified Border Element (CUBE).

The Cisco Enhanced Conferencing and Transcoding for Voice Gateway Routers feature provides enhanced multiservice support for Cisco routers in a Cisco Unified Communication Network. This is accomplished by enabling audioconferencing and transcoding functions in access routers. This single-package solution simplifies deployment, eases administration, and helps deliver tangible cost savings by locating conference resources in the branch to reduce WAN utilization. This feature further minimizes costs by using transcoding services to reduce bandwidth needs.

An integral part of Cisco Unified Communications solutions, Cisco Systems® IP telephony offers feature-rich telephony services on a fully converged IP network. Using Cisco 2800,3800, 2900 and 3900 series onboard digital signal processor (DSP) capability and also available on the Cisco IP Communications Voice/Fax Network Module (NM-HD) and the Cisco IP Communications High-Density Digital Voice/Fax Network Module (NM-HDV2), the Cisco Enhanced Conferencing and Transcoding for Voice Gateway Routers feature preserves all the WAN, public switched telephone network (PSTN), and private branch exchange (PBX) access capabilities while adding rich-media conferencing and transcoding functions.

Audio Conferencing Services

In a traditional circuit-switched voice network, all voice traffic goes through a central device (such as a PBX system), which provides audioconferencing services as well. Because IP phones transmit voice traffic directly between phones, a network-based conference bridge is required to facilitate multiparty conferences. In an IP telephony network using CUCM or CUCME, the Cisco Enhanced Conferencing and Transcoding for Voice Gateway Routers feature provides the conference bridging service. This feature also supports these services:

- Cisco CUCM and CUCME Meet-Me and ad-hoc conferences with up to 64 participants
- Substantial scalability in the number of conferences supported based on the number of Cisco Packet Voice DSP Fax/Voice Modules (PVDM2s or PVDM3s) employed
- G.711 a/u-law, G.729, G.729a, G.729b, G.729ab, G.722, iLBC participants joined in a single conference
- Easy deployment of conference resources in routers across the network; this provides a local conference resource, reduces WAN utilization, and improves voice network performance

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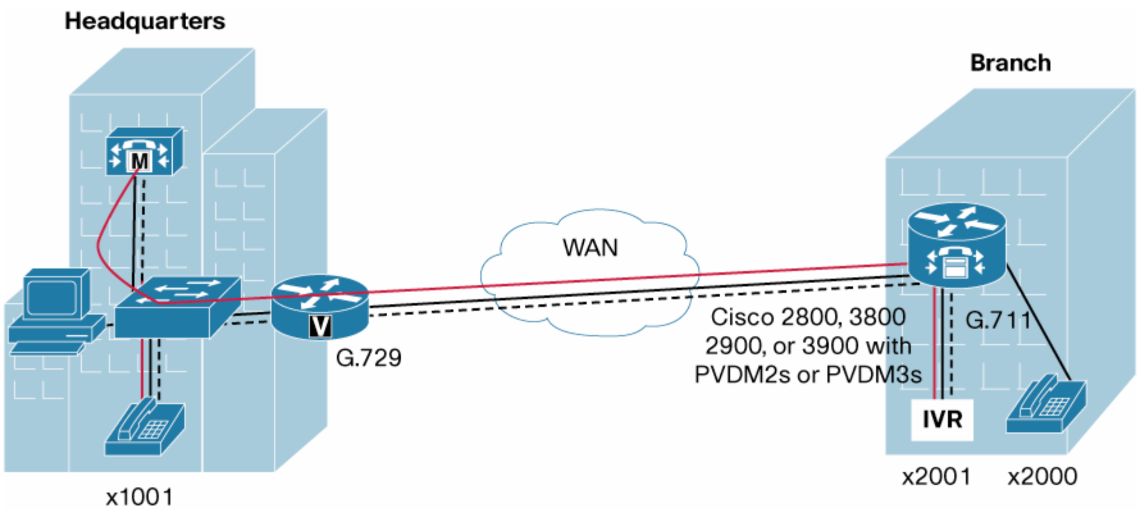
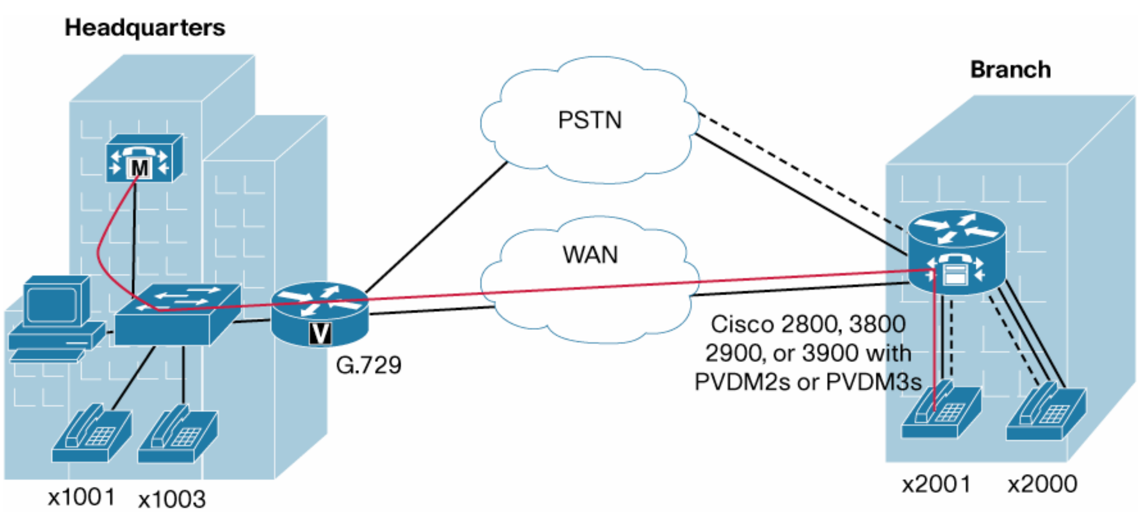
Figure 1. Conferencing



Universal Transcoding Services

Transcoding enables two important functions in an IP telephony network. The first is to save on WAN utilization and attendant costs. This is accomplished by compressing voice traffic across the WAN. The second is to enable communication between different devices that support different codecs. PVDM2 and PVDM3 support universal transcoding from one of the supported codecs to another.

Figure 2. Transcoding



Platform Support and Software Requirements

Cisco Enhanced Conferencing and Transcoding for Voice Gateway Routers provides the following platform support and requires the following software:

- Support is provided onboard on the Cisco 2800,3800, 2900 and 3900 series voice gateway routers. This feature is also supported on platforms using the Cisco IP Communications Voice/Fax Network Module and the Cisco IP Communications High-Density Digital Voice/Fax Network Module.
- Scalable performance using the Cisco PVDM2 and PVDM3.
- Cisco IOS Software Release 12.3(8)T4 with the IP Voice feature set is required for the Cisco 2800 Series voice gateway routers. Cisco IOS Software Release 12.3(11)T with IP Voice is required for the Cisco 3800

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Series voice gateway routers. Cisco IOS Software Release 12.3(8)T with IP Voice is required for the Cisco IP Communications Voice/Fax Network Module and Cisco IP Communications High-Density Digital Voice/Fax Network Module. Conferencing using the PVDM2-8 requires Cisco IOS 12.3(11)T1.

- Cisco IOS Software Release 15.0(1) M with UC image is required for the Cisco 2900 and 3900 Series voice gateway routers.
- Cisco 2801, 2811, 2821, 2851, 3825, and 3845 voice gateway routers require Cisco CallManager Version 4.0(2a) SR1 for full feature support including Media Termination Point (MTP). However, Cisco CallManager Version 3.3(5) may be used on the Cisco 2811, 2821, 2851, 3825, and 3845 voice gateway routers when MTP support is not needed and conferencing and transcoding support is sufficient.
- Cisco IP Communications Voice/Fax Network Module and Cisco IP Communications High-Density Digital Voice/Fax Network Module require Cisco CallManager Version 4.0(1) for full feature support including MTP. However, Cisco CallManager Version 3.3(4) may be used when MTP support is not needed and conferencing and transcoding support is sufficient.

Capacity Planning

DSPs are built directly onto the Cisco IP Communications Voice/Fax Network Module. On the Cisco IP Communications High-Density Digital Voice/Fax Network Module, DSPs are contained on PVDM2s, which are inserted onto the network module. On the Cisco 2800, 3800, 2900 and 3900 series voice gateway routers, DSPs are contained on PVDM2s or PVDM3s, which are inserted directly onto the motherboard. Each PVDM2 DSP is individually configurable to support either conferencing or transcoding and standard voice termination. Each DSP on PVDM3 can be configured to support individual services such as conferencing, transcoding and standard voice termination or a mix of all above services. The total number of conferencing, transcoding, and voice termination sessions is limited by capacity of the entire system, which includes the DSPs, platform, physical voice interface, and CUCM. DSP calculator is a tool for system capacity planning.

Maximum Conferencing Sessions on PVDM2s*[1] and PVDM3s:

PVDM2 family can support:

- Up to 32 participants for G.711 conferences
- Up to 16 participants for G729/G729A/G.722 conferences
- Up to 8 participants for iLBC conferences

PVDM3 family can support:

- Up to 64 participants for G.711 conferences
- Up to 32 participants for G.729/G.729A/G.722 conferences
- Up to 16 participants for iLBC conferences

[1] PVDM2 numbers in the table are based on IOS release 12.4(15)T and above

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