行政院國家科學委員會專題研究計畫 成果報告

以服務為導向之合工與認知軟性網路理論架構、效能優化及技術應用的研究與開發--子計畫一:叢集式隨意點對點多媒體會議服務系統及資源最佳化管理之研究與開發

研究成果報告(精簡版)

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中文摘要

多媒體會議是一個實現多方即時互動並且以多媒體應用為主的通訊服務，例如影像、語音、文字等傳輸。由於網路技術成熟和網路使用普遍，使得以網路為基礎的多媒體會議服務受到重視。傳統網路多媒體會議大多以集中式架構建構，卻存在單點錯誤和負載集中等問題。因此，本計畫提出一個叢集式隨意點對點系統架構，提供多媒體會議服務。在 Peer to Peer 分散式架構以 DHT Chord 作為結構，並改良 Chord 在結構形成時會打散實體網路與疊層網路的對應關係，造成多餘的路由，服務搜尋時間延遲，影響會議品質。因此以 IP 位址為叢集的基礎，提出階層式疊層網路，降低實體網路和疊層網路的落差，減少多餘路由，並設計會議服務初始化、服務註冊、服務搜尋等機制，提供隨意式多媒體會議服務。最後透過模擬分析，驗證改良之後的成效。

關鍵字：多媒體會議、叢集式隨意點對點會議服務、階層式疊層網路

英文摘要

The multimedia conference is a communication service to approach multi-party, real-time, interactive and multimedia applications. Due to the mature Internet technology and the ubiquitous network access, the multimedia conference over Internet is attended. The traditional multimedia conference is mostly based on centralize framework, but exists single failure and overloads on single point. Therefore, this project proposes a cluster-based peer to peer framework to provide multimedia Ad-hoc conferencing service. In this framework, using DHT Chord as structure, and improving Chord that forming structure occurs mismatching between physical network and overlay network. Since the mismatching, increase redundancy routing, service discovery latency, affect the quality of conference. For this reason, this project using IP address-clustering propose hierarchical overlay network to improve the mismatching between physical network and overlay network, reduce redundancy routing. Besides, design conference service initial、service register、service discovery, and such mechanisms. The last simulation analysis validates improvement results.

Keywords: Multimedia Conference, cluster-based peer to peer conference service, hierarchical overlay network

1. 前言

多媒體會議是一個實現多方即時互動並且以多媒體應用為主的通訊服務，例如影像、語音、文字等傳輸。傳統以電信傳輸模式所提供的遠距視訊會議服務是使用事先所部署的昂貴設備和專線的方式來達成，雖然會議傳輸品質較穩定、可靠，但是也因為成本高以及應用不易開發等原因，使用的情況並不普及。由於網路技術快速地發展及普遍地使用於一般的家庭使用者，並且受惠於網路資源共享及無所不在的特性，期望能夠藉由建構在以 IP 網路為傳輸媒介的方式來提供廉價、易取得的多媒體會議服務。

目前在網路多媒體會議服務的系統架構設計已有相當多的討論及研究。IETF 將其區分為三種模型[1]:

- 緊密耦合模型
- 鬆散耦合模型
- 完全分散式模型

在緊密藕合模型中，服務元件以網路上的伺服器來呈現，這些伺服器用來提供會議服務並且處理相關的訊息及影音串流。因為資訊和資源的集中式管理，緊密耦合的系統具有高穩定性和高可靠度。然而，集中式的環境造成伺服器的負擔重及依賴性高，這樣的架構模型不利於大型的服務規模。此問題可以經由部署多台伺服器並且以階層式管理的方式來解決，但是卻也同時提高了系統成本。

鬆散耦合模型通常建立在以網路層的群播技術來達到資料傳輸。在這種架構下，會議成員之間不具有信令協調的關係，而是經由加入、離開一個群播位址群組來參與會議。鬆散耦合模型適合大型的服務規模，然而它的缺點則是使用者端必須具備串流處理的功能以及尚未普遍地具備群播技術的問題。

完全分散式模型尚未有較明確的系統架構雛形，目前 IETF 提出應用點對點(Peer-to-Peer) 技術的 P2PSIP 多媒體應用服務架構[2]，我們認為此架構為發展完全分散式模型的基礎。此系統模型最主要的概念是透過部份使用者建立一個服務網絡來提供並且維護管理整個服務，也就是說某些設備能力充足的使用者能夠分享其資源並且提供部份服務及功能。

基於前文所說明三種模型的特性，我們提出一個以 SIP[3]為基礎的叢集式隨意點對點系統架構來提供多媒體會議服務。這個系統架構整合了緊密耦合模型和完全分散式模型的概念，透過分享使用者資源來降低系統的通訊成本。此外也因為其優點互補，能提供一個穩定、可靠且適合大型服務規模的多媒體會議服務系統。

我們系統架構的特性是引入 P2P 概念，把部分的會議功能由使用者來提供，甚至不需要業者建置會議伺服器的協助，也能彼此協調進行會議，進而降低業者建置成本。然而一般 P2P 架構的形成將使用者打散平均分佈到疊層網路上，造成實體網路位置相近的使用者卻在疊層網路中卻是距離遙遠，因此不利於多媒體會議的進行，也增加了 P2P 搜尋的時間，因此我們提出了結合 IP 位址和階層式疊層網路特性的多媒體會議系統架構，利用 IP 位址將實體網路相近的使用者分到同一個叢集，使用者可以在叢集內部存取並分享會議資源，進而降低了實體網路和疊層網路間的落差性，也降低了會議資源搜尋的時間，提升會議的服務品質，相關的設計於第三章會進一步說明。

本計畫貢獻主要在於提出一個能夠兼具穩定、可靠且擴展性高的多媒體會議系統架構，又同時藉由分享使用者資源來降低系統的通訊成本。在我們的系統中，一個資源充裕的叢集內部會議將不會佔用系統資源，藉此能夠降低系統的使用成本。如能有效地管理使用者所提供的資源，進而讓系統的負擔最小化，定能提高業者的競爭能力。

2. 相關工作

目前針對多媒體會議應用的研究以 IETF 所發展的規範為主，其中 MMUSIC、SIPPING 和 XCON 等研究團隊分別負責針對多媒體通訊協定、SIP 多媒體應用和緊密耦合模型協定的開發工作[4][5][6]並於 2007 年新成立 P2PSIP[2]研究團隊，致力於研究 SIP 應用於 Peer to Peer 相關規範。表 1 列出相關的 RFC 與 drafts 規範。

我們底下簡單介紹緊密耦合模型，依據 RFC4353 所定義的緊密耦合模型[1]，有四個主要的服務元件:

(1) Focus: Focus 是一個會議控制管理中心，它主要處理會議協調、引導的訊息。當使用者與 Focus 接觸時會建立一個描述此通訊關係的識別資訊 (dialog)，藉由維護此資訊並且與使用者溝通，Focus 能夠為參與會議的使用者們建立及設定一個會議。

(2) Mixer: 此服務元件負責處理媒體串流，將多個串流整合成單一的串流格式並且分送給所有會議參與者。在這裡的 Mixer，廣義認定為可處理影像及語音等串流。

(3) 會議策略伺服器 (Conference Policy Server): 會議策略伺服器提供修改及儲存管理會議的策略。

(4) 會議伺服器 (Conference Server): 在此模型中，會議伺服器是一個實體元件，它必須最少包含 Focus，其實際的服務元件配置乃視系統架構所考量的伺服器部署而有所不同。

表 1 多媒體會議應用的相關 RFCs 與 Drafts

WG	*RFC*	*Title*
MMUSIC	RFC3264	An Offer/Answer Model with the SDP
	RFC4566	Session Description Protocol
SIPPING	RFC3725	Best Current Practices for Third Party Call Control in the SIP
	RFC4245	High-Level Requirements for Tightly Coupled SIP Conferencing
	RFC4353	A Framework for Conferencing with the SIP
	RFC4575	A SIP Event Package for Conference State
	RFC4579	SIP Call Control - Conferencing for User Agents
XCON	RFC4376	Requirements for Floor Control Protocols
	RFC4597	Conferencing Scenarios
	RFC4582	The Binary Floor Control Protocol
	RFC5018	Connection Establishment in the Binary Floor Control Protocol
	RFC5239	A Framework for Centralized Conferencing
P2PSIP	Draft	REsource LOcation And Discovery(RELOAD) Base Protocol
	Draft	P2PSIP Overlay Diagnostics
	Draft	A SIP Usage for RELOAD
	Draft	Concepts and Terminology for Peer to Peer SIP
	Draft	REsource LOcation And Discovery (RELOAD)

IETF 提出五種實際可實現的系統架構[1]，這些架構主要針對多種環境需求而發展的服務元件配置及伺服器部署。然而，它們仍然受限於集中式而負載過重以及服務規模不大

的問題。由此可見，如能經由使用者的資源分享來提供部份的系統功能或會議服務，就能達到分散系統負載、減少通訊成本以及突破服務規模等效用。

3. 系統架構

3.1. 系統架構

我們整體的系統環境如圖一，我們扮演一個多媒體會議服務提供者，使用者會向會議服務提供者購買多媒體服務，同時使用者也會依據實體網路位置，各自形成區域性的疊層網路，並可以互相分享和使用多媒體會議資源，而各區域之間則是選派出 Super Node 加入到共同的疊層網路中，使用者若要聯繫其他區域疊層網路的使用者，可以透過 Super Node 幫忙將訊息導向目的地。因此，當使用者加入了區域疊層網路後即可使用區域疊層網路的相關會議資源，使用電話和會議服務。然而，若使用者的區域疊層網路沒有足夠的資源可以發起會議或是無法加入到區域疊層網路的使用者，可使用骨幹網路的會議資源，因此我們在骨幹網路中也會部署一些會議伺服器來輔助資源不足的區域疊層網路。此外，我們將節點大致分為三種型態：

Peer：組成區域疊層網路的主要成員，設備的能力中等，在區域疊層網路中會幫助其他節點轉送訊息，也會貢獻自己的計算能力和儲存空間。

Superior：具有更高階處理能力和網路頻寬的節點，除了有 Peer 的能力外，還能擔任 P2P Focus，負責會議的建立、修改和終止；亦或擔任 P2P Mixer，進行影像和聲音的串流混合處理。

Client：具較差的處理能力和儲存空間，不幫助訊息轉送和貢獻儲存空間，而是透過 Peer 或 Superior 使用疊層網路的資源。

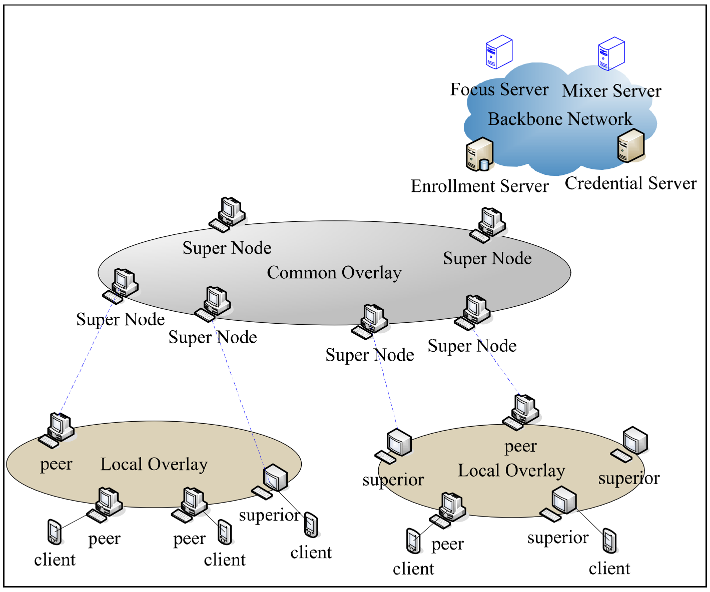

圖 1 系統環境

3.2. 階層式 Node-ID 指派

Node ID 一般是透過將使用者的 IP 位址和 Port 經過一個 Hash Function 而生成的字串，即使兩個相近的輸入值，其輸出的結果也會差異很大，這在檔案分享的環境中資源分散儲存是合適的，但在多媒體會議的環境下，需要考量到使用者真實位置的時候，Node ID 的差異可能會造成多餘的路由，反而會增加會議的延遲時間，因此 Node ID 的指派勢必要做一些改變。例如我們欲邀請鄰近使用者加入會議，邀請訊息可能轉送到國外遠處節點，再查詢回來。

因上述等問題存在，如圖二(a)，我們先將 IP 切成兩個部份，以 IPv4 為例子，我們將 140.123.107.9:5050 切割成 140.123 和 107.9:5050 兩個部分，然後再分別經過 SHA-1 雜湊函數獲得兩個 160 bits 長度的字串，在各別截取 64bits 來組合成疊層網路中的 Node ID。

(a) 140.123.107.9:5050
->H("140.123") = 986e02a28d6012d478fa3bbdcff5a0f8a3f7014d
->H("107.9:5050") = 688bcc96ebd2684abff93d7fe8be45f5138b9b5f
(b) H("140.123")+H("107.9:5050")
= 986e02a28d6012d4 688bcc96ebd2684a
H("140.123")+H("107.8:5050")
= 986e02a28d6012d4 0a187b7f1d747ac0

圖 2 Node ID 切割雜湊：(a) Hash 處理方法(b) 完整 Node ID

改變 ID 的分派後，如圖二(b)，當相同網域的兩個節點經過雜湊函數後，前面的 64bits 依舊會相同，我們把這 64bits 稱為 Overlay-Prefix，而後面的 64 bits 則稱為 Local-ID。圖三所示，使用者加入區域疊層網路時，僅拿後面的 Local-ID 做為區域的識別碼，只有當使用者同時也擔任超級節點時，才會拿 Overlay-Prefix 和 Local-ID 的前 4 bits 來當作共同疊層網路的識別碼，詳細運作於第四章進一步說明。

3.3. Super Node 選取

Super Node 是同時加入區域疊層網路和共同疊層網路的節點。除了幫忙儲存區域資料和轉送區域訊息外，還需要處理區域疊層網路和共同疊層網路間的訊息轉換，讓區域疊層網路的使用者可以透過 Super Node 將區域內訊息導到其他區域疊層網路中，以便查詢想要的資訊。

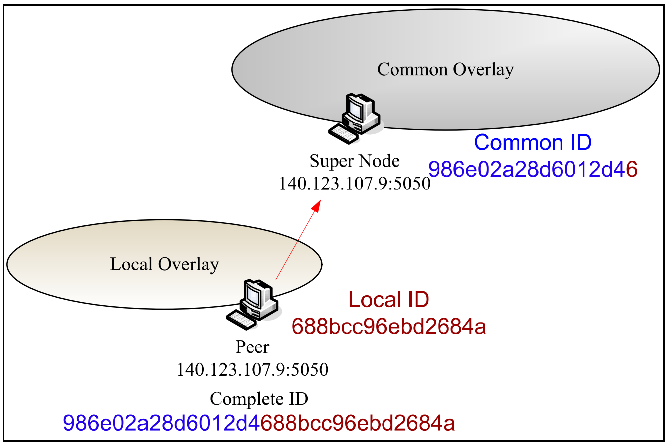

圖 3 使用者在區域疊層網路和共同疊層網路的 ID 對照關係

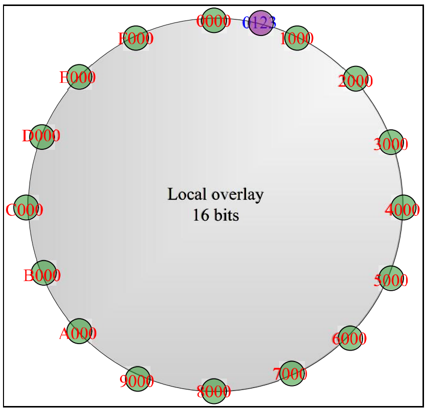

圖 4 選定 Super Node 的特殊 ID

利用 Chord[7]疊層網路的特性，每一個節點都會知道上一個節點的位置(Node ID)，藉此來得知自己應該負責哪些儲存區塊。圖四說明，這 16 個特殊 ID 分別為 Local ID 第 1 到第 4 個 bit，節點 0123 負責這些特殊 ID，就會擔任 Super Node，加入到共同疊層網路中幫助轉送訊息。這種機制可以選出區域疊層網路中 1~16 個節點來擔任此區域疊層網路中的 Super Node，若 Super Node 離開後，其負責的儲存區塊會給其他節點繼承，Super Node 就由新的節點擔任，如此可以不需一直維持 Super Node 是否可用，也不需複雜的替補機制。

4. 系統運作

4.1. 使用者登入系統

使用者使用本系統時，一定得先通過認證並完成登入的程序，如圖五說明，(1)使用者會判斷自己的使用者型態(Peer 或 Superior)後並向 Credential Server 進行登入認證的動作，(2)通過認證後，Server 會發送憑證給使用者，憑證上包含專屬的完整 Node ID 及 User Name 給使用者在疊層網路上使用。(3)認證完後使用者會向 Enrollment Server 索取同網域的疊層網路組態檔，(4)Enrollment Server 判斷使用者的型態和資料後，找尋此使用者同網域的 Local XML 檔案，若此檔案存在，且使用者型態不為 client，會將使用者 IP 和 Port 填入 Local XML 檔案中並連同 Common XML 傳給使用者。(5)使用者判別 Local XML 檔案中，是否有其他使用者 IP，若有則發送 Join Request，加入 Local Overlay；若沒有則創造 Local Overlay 等待其他使用者加入。(6)使用者加入 Local Overlay 後，定期檢查自己是否為 Super Node，若是則檢查 Common XML 是否有其他使用者 IP，若有則發送 Join Request，加入 Common Overlay；沒有則創造新 Common Overlay。當使用者加入 Common Overlay 時，需回報給 Enrollment Server 以更新 Common XML 檔案 Bootstrap Peer 的資訊。

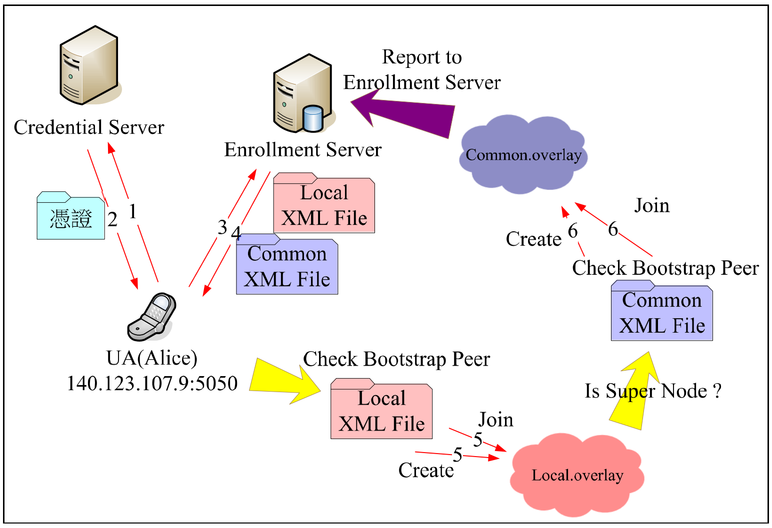

圖 5 使用者登入及加入疊層網路的示意圖

4.2. 使用者註冊

使用者加入區域疊層網路後，要將自己的資訊註冊到區域疊層網路中，如此一來，當有使用者要找尋另一個使用者時，也才能利用疊層網路將使用者的資料搜尋出來，因此我們將服務註冊大致分為下列三種:

(1)SIP URI 註冊

(2)會議資源註冊

(3)Overlay-Prefix 註冊

SIP URI 註冊是將節點的 SIP URI 和節點 Node ID 的對應關係儲存在區域疊層網路上，只要節點得到其他節點的 SIP URI，即能在區域疊層網路上取得對應的 Node ID，有了節點的 Node ID 就能透過 RELOAD[8][9]和其節點建立 Tunnel，即可以傳送 SIP 訊息。

會議資源註冊則是為了讓區域疊層網路的使用者得知，哪些節點具有 P2PFocus 或 P2PMixer 的能力，也才能找到這些節點，請求幫忙創建會議和處理影音串流。

Overlay-Prefix 註冊是將區域疊層網路的 Overlay-Prefix 註冊到共同疊層網路上，如圖二所示，兩個節點的都有共同的 986e02a28d6012d，外部疊層網路要找到這兩個節點，就必須知道此 Overlay-Prefix，因此必須把疊層網路的 domain name 和 Overlay-Prefix 的對應關係儲存在共同疊層網路上，其他區域疊層網路只要擁有此網路的 domain name，即可在共同網路上找尋對應的 Overlay-Prefix，進而傳送訊息給此區域疊層網路中的節點。

以下我們將 Node ID 精簡為 24 bits，以方便說明。圖六為會議資源註冊，P2PFocus 或 P2PMixer 會各別在 H("focus;no=k")或 H("mixer;no=k") (k=1,2,..,m，m 為系統參數)這 m 個特定 ID 上儲存自己的 Node ID，這些 Node ID 即形成 Focus-list 或 Mixer-list 等陣列的資料結構，每一個值都是標示區域疊層網路中具有會議資源的 Node ID。

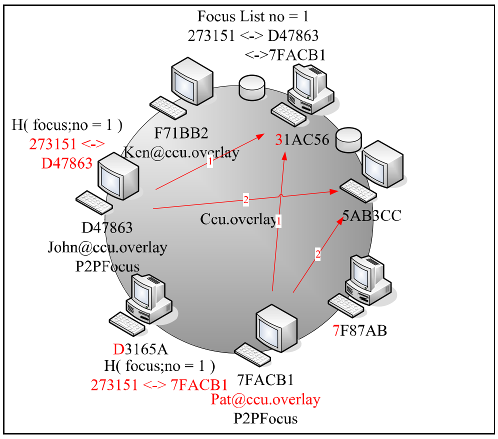

圖 6 會議資源註冊(m=2)

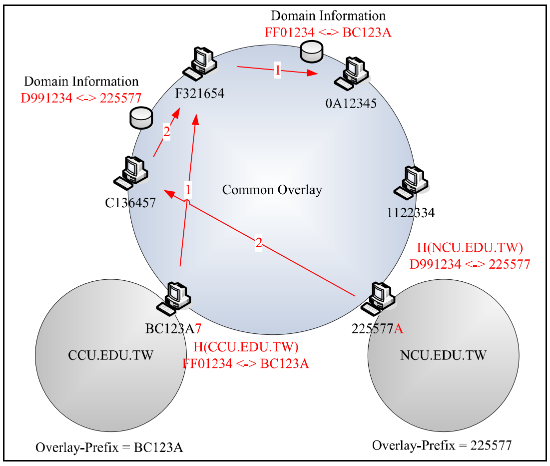

圖 7 Overlay-Prefix 註冊在共同疊層網路上

圖七所示，兩個區域疊層網路其 Overlay-Prefix 分別為 BC123A 和 225577，而 BC123A7 和 225577A 分別為這兩個區域疊層網路的超級節點，當 BC123A7 和 225577A 加入到共同疊層網路時，需要將自己區域疊層網路的 Overlay-Prefix 也就是 Domain 資訊儲存到共同疊層網路中，以提供其他 Domain 的使用者搜尋之用。

4.3. 服務搜尋與運作

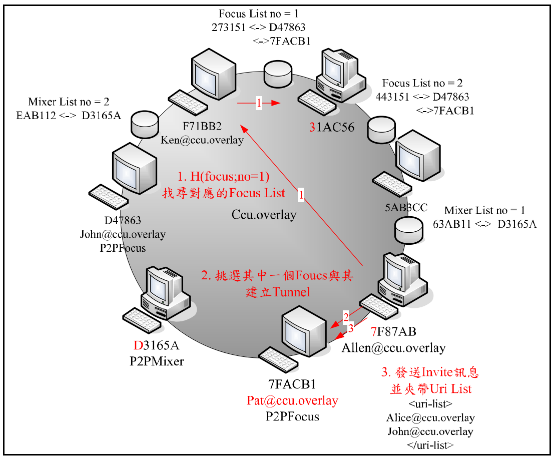

圖 8 區域疊層網路會議服務搜尋

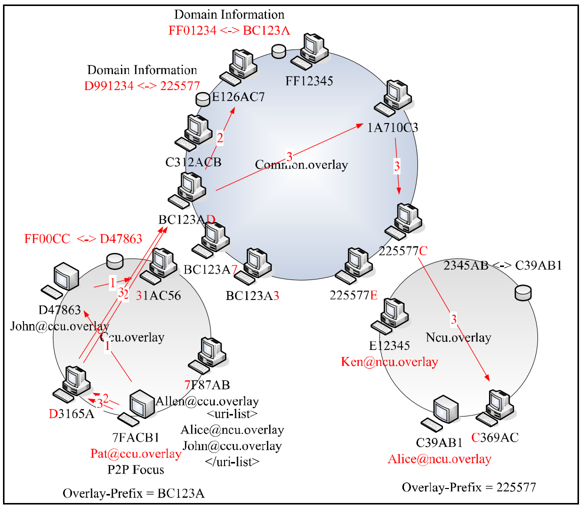

圖 9 跨區域疊層網路的會議搜尋

若節點 A 想搜尋節點 B 的 Node ID，必先得到節點 B 的 SIP URI，依據節點 B 的 SIP URI domain 是否和節點 A 相同，可以分為兩種搜尋狀況：

(1) 若其 URI 的 domain 和節點 A 相同，表示 A 和 B 位於同區域疊層網路，A 只要把 URI 的 User Name 經過雜湊(key 的長度為 64bits)後在區域疊層網路上尋找對應的資料即可。

(2)若 domain 不相同，則節點 A 必須先查節點 B 所在區域疊層網路的 Overlay-Prefix，節點 A 將 domain name 經過雜湊(key 的長度為 68bits)後，丟給最近的 Super Node(可能是自己)，透過 Super Node 在共同疊層網路上搜尋 domain information，

當節點 A 拿到節點 B 的 Overlay-Prefix(64bits)後，將 B 的 SIP URI 的 User Name 經過雜湊後，附加在 Overlay-Prefix 之後組合新的 Key(128bits)，再透過 Super Node 將此訊息導到共同疊層網路上路由，路由期間取前 68bit 作為路由的依據，若有 Super Node 收到此訊息，比對目的地所處的 Overlay-Prefix，若和 Super Node 相同時，即把此訊息轉到內部區域疊層網路上搜尋，在內部路由則取後 64bits 作為依據，進而找到節點 B 的 Node ID。

當使用者想進行會議時，便需搜尋 P2PFocus，使用者可以用 H("focus;no=k")(k=1,2,…,m)，基於[8]搜尋 Focus-list，並於 Focus-list 隨機選取一個 P2PFocus 節點，和其節點建立 Tunnel，傳送 SIP Invite 訊息。如圖八所示，(1)Allen 利用 H("focus;no=1")向 31AC56 取得 Focus-list，(2)隨機選取 list 上的 P2PFocus，如圖八選到 Pat，並和 Pat 建立 Tunnel，(3)向 Pat 發送 SIP Invite 訊息並夾帶 uri-list 使用者名單。

5. 模擬與分析

我們使用 Oversim[10]這套模擬軟體，增加模擬 SIP 訊息的模組及我們的設計，以模擬搜尋時的行為，參考 RELOAD[8]協定，添加 Attach 訊息建立 Tunnel。

我們模擬系統實際一天的時間，假設一天內平均一個使用者會有 4 通電話和 2 通會議，因此系統若總共有 N 個節點，則電話的取樣數為 4*N，會議取樣數則是 2*N，而一個區域疊層網路內約有 10 個使用者，因此會有 N/10 個區域疊層網路的存在。當 Focus 收到會議建立請求後，會隨機邀請 2~10 個人進來開會，我們也依據這 2~10 個人的選取狀況區分為隨機選取(Random)、全選同網域(All Intra Cluster)和全選不同網域(All Inter Cluster)三種情況。

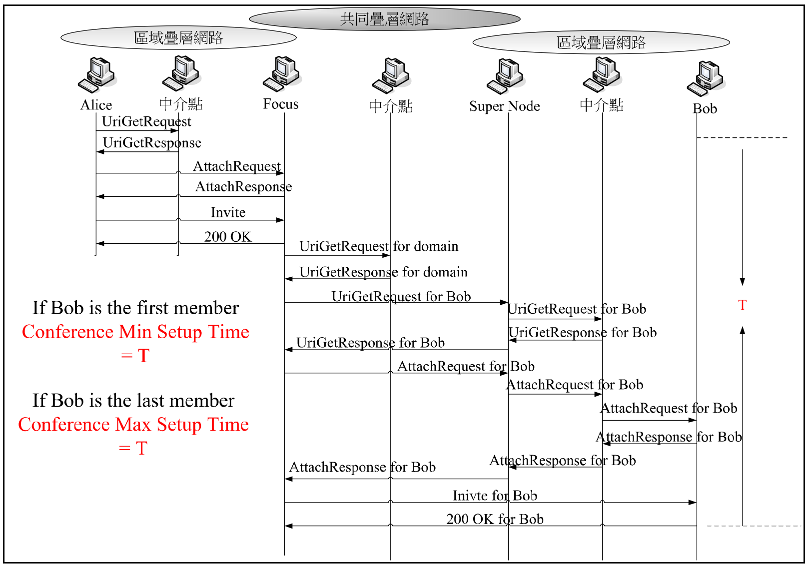

圖 10 會議服務搜尋模擬順序圖

圖十為我們系統模擬的訊息順序圖，Alice 找 Focus 會先利用 UriGetRequest 找尋到 Focus 的 Node ID，並對 Focus 發送 AttachRequest 建立 Tunnel，成功建立通道後再發送 Invite 訊息請求建立會議。當 Focus 邀請會議成員時，第一個回覆 200OK 的時間差為 T，

我們定義為 Min Setup Time，最後一個回覆的時間差則為 Max Setup Time，兩者取平均則為 Average Setup Time。

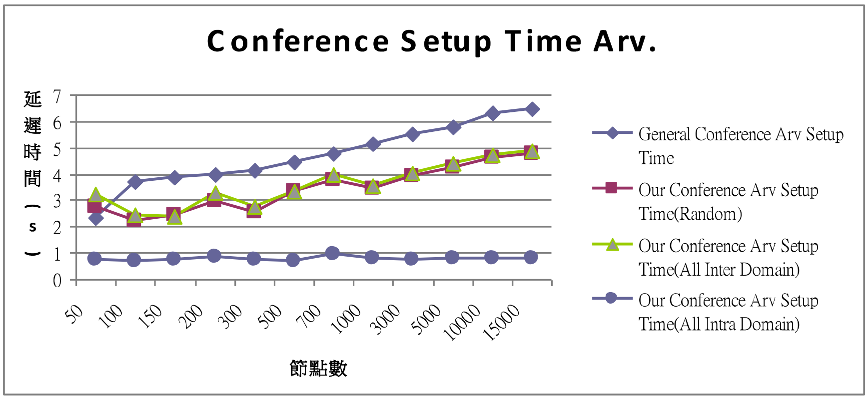

圖 11 Conference Average Setup Time

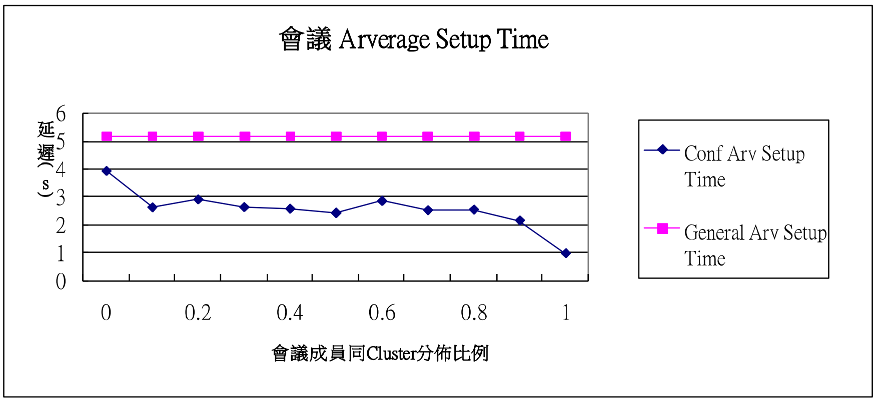

圖 12 會議成員 cluster 分布比例之會議 Average Setup Time

圖十一為 Average Setup Time，X 軸為節點數，Y 軸為會議建立的延遲時間，第一條曲線為基於[8]實現一般疊層網路之會議搜尋，其 Node ID 分配不作修改；第二條為本系統隨機挑選 2~10 個使用者的曲線分佈之會議建立延遲時間；第三條為挑選 2~10 個外部疊層網路使用者的分佈曲線；第四條為挑選 2~10 個內部疊層網路使用者的分佈曲線。由此可發現本系統的會議建立延遲比一般疊層網路短，若邀請同區域疊層網路的使用者，則效能會更好。

圖十二為會議成員(10 人)分布在同區域疊層網路的比例之會議 Average Setup Time。X 軸為使用者和 Focus 在同區域疊層網路的分佈比例；Y 軸為會議建立的平均延遲時間。第一條曲線為一般疊層網路的會議建立之延遲時間，和使用者分佈比例無關；第二條為本系統的會議建立之延遲時間，隨著使用者在同區域疊層網路的比例越高時，Setup Time 即呈現明顯的下降，和一般疊層網路的搜尋的時間差距也就越明顯。

6. 結論

本系統提出階層式的疊層網路結構，利用使用者的 IP 位址將實體網路相近的使用者分在同一個區域疊層網路；在區域疊層網路內，使用者可彼此分享會議資源，建立隨意式會議。由於實體網路位置拉近，進而在疊層網路搜尋時能加快搜尋速度，減少會議建立時間。

Mixer 是多媒體會議開會重要的元件，其受實體位置影響更為嚴重，且負載往往很高，目前 Mixer 透過疊層網路搜尋到後，多媒體串流皆以實體網路傳輸，但實體網路卻可能有某些節點頻寬甚小，導致影像傳輸的延遲增加，這反而會降低多媒體會議的品質。因此，未來將可以考慮利用疊層網路來決定串流的傳輸路徑，如此可避免卡在頻寬甚小的節點上，能有更好的會議品質。

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[3] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, and E. Schooler, “SIP: Session initiation protocol,” RFC 3261, Jun. 2002.

[4] MMUSIC, http://www.ietf.org/html.charters/wg-dir.html

[5] SIPPING, http://www.ietf.org/html.charters/wg-dir.html

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[7] Stoica, I., Morris, R., Liben-Nowell, D., Karger, D.,Kaashoek, M., Dabek, F., and H. Balakrishnan, "Chord: A Scalable Peer-to-peer Lookup Protocol for Internet Applications", IEEE/ACM Transactions on Networking Volume 11, Issue 1, 17-32, Feb 2003.

[8] C. Jennings, B. Lowekamp, E. Rescorla, S. Baset and H. Schulzrinne, “REsource LOcation And Discovery (RELOAD) Base Protocol”, IETF Draft, draft-ietf-p2psip-base-03.txt, July 13, 2009.

[9] C. Jennings, B. Lowekamp, E. Rescorla, S. Baset and H. Schulzrinne, “A SIP Usage for RELOAD,” IETF Draft, draft-ietf-p2psip-sip-01.txt, March 3, 2009.

[10] Ingmar Baumgart, Bernhard Heep, Stephan Krause, OverSim: A Flexible Overlay Network Simulation Framework, Proceedings of 10th IEEE Global Internet Symposium (GI '07) in conjunction with IEEE INFOCOM 2007, Anchorage, AK, USA, May 2007.

計畫成果自評

- 原計畫與已達成之研究成果
  1. 原計畫為三年延續型計畫，第一年預計完成 Cluster-Based P2P Ad-Hoc Multimedia Conferencing 需求分析、系統架構設計與運作機制設計；第二年預計完成 Intra-Cluster P2P Ad-Hoc Multimedia Conferencing 資源最佳化管理方法研究與 Inter-Cluster P2P Ad-Hoc Multimedia Conferencing 運作機制設計；第三年預計完成 Inter-Cluster P2P Ad-Hoc Multimedia Conferencing 資源最佳化管理方法研究、系統整合與測試，以及服務改良與演進。
  2. 由於本計畫僅核定一年，因此本計畫已完成原訂計畫中第一年研究內容，共三大研究成果，(1) 在 Cluster-Based P2P Ad-Hoc Multimedia Conferencing 需求分析中，我們發現會議建立時間之重要性，以及 P2P 結構與 IP Address 結構不匹配問題，造成服務搜尋速度影響；(2)因此針對服務搜尋速度問題，完成叢集式點對點多媒體會議系統設計；(3) 最後透過 OverSim 模擬工具，完成本系統之效能驗證。
  3. 因此，本研究成果與原計畫完全相符，本研究成果將作為未來延續性研究之重要基礎，探討叢集式隨意點對點多媒體會議服務系統資源分配與管理最佳化等研究議題。

- 研究成果之學術與應用價值
  1. 本研究初步成果與相關延伸之研究成果已發表於國內外學術研討會。
     - ◆ NCM 2009 國際研討會論文一篇 (EI)
     - ◆ NISS 2009 國際研討會論文一篇 (EI)
     - ◆ ICUMT 2009 國際研討會論文一篇 (EI)
     - ◆ 2009 民生電子研討會 (WCE 2008) 論文一篇
     - ◆ 2009 全國電信研討會 (NST 2009) 論文二篇
  2. 本研究設計與發展之叢集式多媒體會議服務系統，目前已透過 OverSim 模擬軟體，驗證可行性與預期效能，未來可作為延續研究基礎。

可供推廣之研發成果資料表

■可申請專利　□可技術移轉　日期：98 年 10 月 21 日

國科會補助計畫	計畫名稱：叢集式隨意點對點多媒體會議服務系統及資源最佳化管理之研究與開發 計畫主持人：蘇暉凱 計畫編號：NSC97-2221-E-150-079　學門領域：電信學門(網路)
技術/創作名稱	以網際網路位址為基礎點對點服務快速搜尋方法
發明人/創作人	蘇暉凱
技術說明	中文： 本叢集式隨意點對點多媒體會議系統，以 IP Address 結構為基礎，將鄰近之 peer node 組成區域性叢集，最後透過階層叢集組成一點對點多媒體會議服務網域。此系統具有 peer node 不正常離開之容錯設計，並且對於區域性點對點通訊，可以提供快速服務搜尋，尤其對於隨意點對點多媒體會議應用，可以大大縮短整體會議建立時間。 英文： The cluster-based ad-hoc P2P multimedia conferencing system groups near peer nodes into a regional cluster according to IP Address architecture. Finally, several hierarchical clusters are constructed to a P2P multimedia conferencing domain. The system provides fault tolerance to deal with abnormal leave; moreover, the system can provide fast service discovery to local P2P communications. Especially for ad-hoc P2P multimedia conferencing service, the call-setup time of conferences can be shorten.
可利用之產業及可開發之產品	未來此技術，將可以應用於 P2P 網路視訊會議、P2P 網路電話服務與 P2P 相關應用產業。
技術特點	本系統採用 IP Address 結構特性，提出叢集式隨意點對點多媒體會議系統架構；本系統具有 peer node 不正常離開/加入之容錯設計，並且對於區域性點對點通訊，提供快速服務搜尋效能。
推廣及運用的價值	此架構，可提供 P2P 相關應用參考，以增加 P2P 服務搜尋效能。

※　1.每項研發成果請填寫一式二份，一份隨成果報告送繳本會，一份送　貴單位研發成果推廣單位（如技術移轉中心）。

※　**2.本項研發成果若尚未申請專利，請勿揭露可申請專利之主要內容。**

※　3.本表若不敷使用，請自行影印使用。

出席國際學術會議心得報告

1.

計畫編號	NSC97-2221-E-150-079
計畫名稱	以服務為導向之合工與認知軟性網路理論架構、效能優化及技術應用的研究與開發－子計畫一:叢集式隨意點對點多媒體會議服務系統及資源最佳化管理之研究與開發
出國人員姓名 服務機關及職稱	蘇暉凱, 國立虎尾科技大學 電機工程系, 助理教授
會議時間地點	會議時間：2009 年 6 月 30 – 7 月 2 日 。會議地點： 中國 ，北京 。
會議名稱	2009 International Conference on New Trends in Information and Service Science (NISS 2009)
發表論文題目	Optimal Quality Adaptation for SLA-Based VoIP Services over DiffServ/MPLS Networks

一、參加會議經過

2009 International Conference on New Trends in Information and Service Science (NISS 2009) 今年舉辦於中國北京市，友誼大飯店 (Beijing Friendship Hotel)，由 AICIT 組織主辦，會議相關資料請參考 http://www.aicit.org/niss2009/；AICIT 是近五年由南韓發起的國際組織，致力於研究、技術與服務之理論、發展、應用及實務；該組織每年於亞洲地區舉辦多場研討會，主要以南韓與中國兩地點為主。NISS2009 會議時間從 2009/6/30 至 2009/7/2，期間共三天，正式論文研討會議從 2009/6/30 下午開始發表，第一天有 4 個 session 同時進行，第二天有 6 個 session 同時進行，第三天有 2 個 session 同時進行。NISS 2009 今年論文被接受率僅 29.4%，投稿超過 40 個國家參與；發表之文章，被收錄於 IEEE CS Digital Library 與 EI 索引。

本人搭乘之飛機於 6/28 傍晚抵達中國北京，6/29 先處理個人住宿與生活事宜，並且熟悉周遭環境，6/30 即至大會會場註冊，圖 1為 NISS2009 研討會會場入口。6/30 除了註冊外，本人亦參與一場 Invited Speech，與數篇研討會論文發表。

本人於當地時間 7/1 14:40，以投影片方式口頭發表『Optimal Quality Adaptation for SLA-Based VoIP Services over DiffServ/MPLS Networks』論文，論文發表現場如圖 2。論文中，我們提出 Optimal Quality Adaptation 參考模型，在 DiffServ/MPLS Network 中提供 VoIP 服務品質動態調整，以達到 VoIP 服務提供者最佳獲利。與會學者們也提供幾個參考意見，建議可以引用 game theorem 方法求解。

由於本人正研究 IP 網路保護相關議題，因此剩餘時間亦參與有關網路錯誤回復之論文發表，其中探討乙太網路、無線網路之網路錯誤處理之相關議題。

圖 1：NISS2009 研討會會場入口

圖 2：NISS2009 研討會論文口頭發表 (報告者為蘇暉凱助理教授)

二、與會心得

這次參加 NISS 2009 研討會收穫良多，從技術論文發表中我們可以觀察到幾個重要研究主題與研究方向，如：P2P 議題、Mobile 議題，以及品質管理與控制相關議題。除此之外，也觀察到主辦國北京大學與清華大學學生，及研究人員之積極參與；未來若有機會，也希望能鼓勵自己的學生多參與國際學術活動，給予更多體驗與培養國際觀。

AICIT 組織每年在亞洲地區舉辦數場研討會，並且與 IEEE 組織合作，所接受之文章將收錄於 EI 與 IEEE CS Digital Library 索引，如果需要 AICIT 組織相關資訊，也非常歡迎與本人聯繫。最後，再次感謝國科會補助本人參加 NISS 2009 研討會。

2.

計畫編號	NSC97-2221-E-150-079
計畫名稱	以服務為導向之合工與認知軟性網路理論架構、效能優化及技術應用的研究與開發－子計畫一:叢集式隨意點對點多媒體會議服務系統及資源最佳化管理之研究與開發
出國人員姓名 服務機關及職稱	蘇暉凱, 國立虎尾科技大學 電機工程系, 助理教授
會議時間地點	會議時間：2009 年 8 月 25－27 日。會議地點：南韓，首爾。
會議名稱	5th International Joint Conference on INC, IMS and IDC (NCM 2009)
發表論文題目	Design of Cluster-based System Framework for SIP-based Multimedia Conferencing Services

一、參加會議經過

5th International Joint Conference on INC, IMS and IDC (NCM 2009) 是一個聯合研討會，其中包含三個子研討會，INC2009: International Conference on Networked Computing、IMS2009: International Conference on Advanced Information Management and Service 與 IDC2009: International Conference on Digital Content, Multimedia Technology and its Applications。NCM 2009 今年舉辦於南韓首爾市，Grand Hilton Seoul 飯店，由 AICIT 組織主辦，會議相關資料請參考 http://www.aicit.org/ncm2009/；AICIT 是近五年由南韓發起的國際組織，致力於研究、技術與服務之理論、發展、應用及實務；該組織每年於亞洲地區舉辦多場研討會，主要以南韓與中國兩地點為主。會議時間從 2009/8/25 至 2009/8/27，期間共三天。NCM 2009 今年共收到 1348 篇論文，來自於 33 個國家，最後接受 399 篇論文發表，接受率為 29.6%；除此之外，發表之文章，被收錄於 IEEE CS Digital Library 與 EI 索引。

參加會議經過簡述如下：

1. NCM 2009 大會會場位於 Grand Hilton Seoul 飯店二樓整層，圖 3為大會註冊區，服務人員協助註冊作業以及會場環境介紹；圖 4為大會壁報論文發表區。
2. 本人於當地時間 8/26 早上 9:45，以投影片方式口頭發表『Design of Cluster-based System Framework for SIP-based Multimedia Conferencing Services』論文，論文發表現場如圖 5。論文中，我們針對 SIP 多媒體會議提出叢集式系統架構，以改善傳統集中式多媒體會議系統覆載集中之問題；以及延伸 SIP 基本功能，詳細設計協定通訊流程與內容。圖 6為口頭報告後，與與會先進互動情況。
3. 在會議期間巧遇亞洲大學資工系陳興忠助理教授，陳教授在此研討會議中擔任『2009 International Workshop on Mobile E-commerce, Mobile Payment, Mobile Content Service Systems and Technologies, Security Issues and Applications』Workshop 召集人，並且協助研討會論文優良論文評選事宜，優秀論文將推薦至 Journal of Networks 之 Special Issue；因此，本人也受邀參與同儕推薦。圖 7為本人與亞洲大學資工系陳興忠助理教授、國立中正大學電機系吳承崧教授合影。

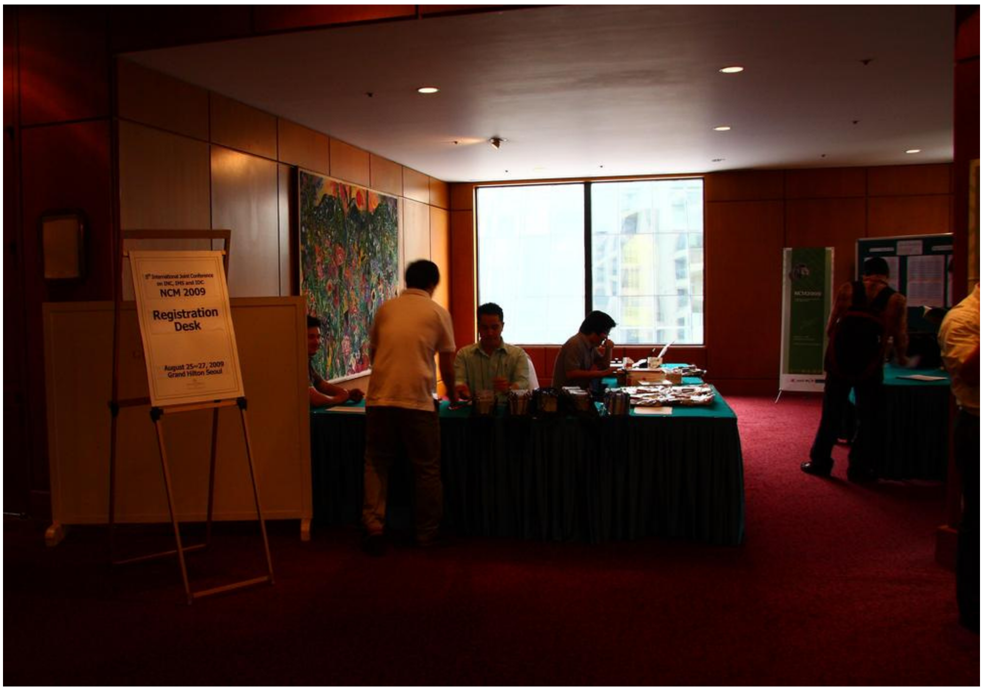

圖 3：NCM2009 研討會註冊區

圖 4：NCM2009 研討會壁報論文區

圖 5：NCM2009 研討會論文口頭發表 (報告者為蘇暉凱助理教授)

圖 6：NCM2009 研討會論文口頭發表，與現場先進互動情況

圖 7：亞洲大學資工系陳興忠助理教授(左)、國立中正大學電機系吳承崧教授(中)、
國立虎尾科技大學電機系蘇暉凱助理教授(右)

二、與會心得

這次參加 NCM 2009 研討會受益良多，在此會議本人瞭解近年 Multimedia Communication, WiMAX, Ad Hoc Network, Wireless Mesh Network, Wireless Sensor Network 相關研究議題，以及未來研究主題趨勢。除此之外，本人在 NCM 2009 中擔任 Program Committee，參與論文審稿，在聆聽論文發表過程中，也體會大家認同之研究貢獻與論文品質。

AICIT 組織每年在亞洲地區舉辦數場研討會，NCM 2009 是本人參與 AICIT 組織的第二場研討會；因此，如果需要 AICIT 組織相關資訊，也非常歡迎與本人聯繫。最後，非常感謝國科會補助本人參加 NCM 2009 國際研討會。

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The 2009 International Conference on
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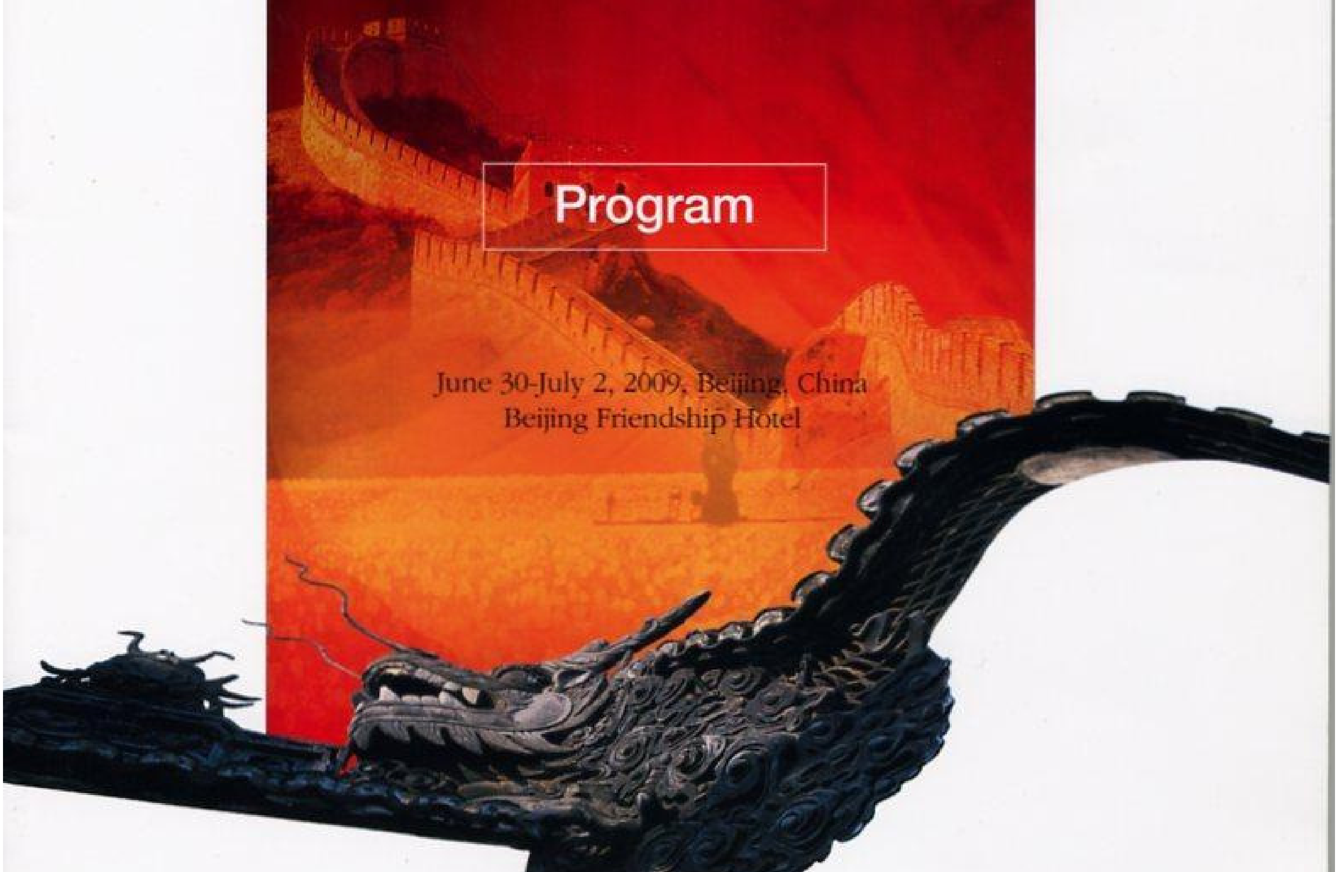

Optimal Quality Adaptation for SLA-Based VoIP

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***Abstract*—This work proposes an optimal quality level adaptation mechanism based on contracted SLAs (Service Level Agreement), and defines an optimal service policy for a VoIP (Voice over IP) application provider over DiffServ/MPLS networks. A VoIP application provider between VoIP users and network service providers faces the challenge of SLA mapping to maximize its profit and satisfy users' requirements. The numerical results show that the proposed mechanism adapts the quality level of each active call to new network conditions well, and maximizes the profit under the constraints on the contracted SLAs. The contributions would help application providers develop CAC (Call Admission Control) schemes and proprietary service policies efficiently.**

I. Introduction

A service level agreement (SLA) is a formal contract of the relationship between a service provider and its customers. The SLA also specifies what the customer can expect from his service provider, including service quantity and service quality, i.e., the services provided by the service provider, and the penalties paid by the service provider if it cannot meet the committed goals. Many publications on SLAs can be found in [1]–[3] and references therein.

In a VoIP service environment, people can be divided into three roles, namely *VoIP users*, *VoIP application providers* and *network service providers*. The VoIP users are concerned about what VoIP service price they must pay and what user-level QoS (Quality of Service) they may receive. The VoIP application providers care about the balance of revenue and investment, and providing a good VoIP service to users. Similarly, the network service providers are concerned about the balance of revenue and investment to provide a good transport service. Thus, the VoIP users and the VoIP applications provider would sign session-level SLAs (S-SLAs), while the VoIP application providers and the network service providers would also sign network-level SLAs (N-SLAs) between the application and transport service domains.

VoIP application providers can partition the framework of SLA management into resource planning and dynamic on-demand processing. The resource evaluation and planning are not discussed in this paper, and interested readers are referred to [4], [5]. For the dynamic processing, VoIP users first stipulate the acceptable quality profiles in S-SLAs with their application provider, and the VoIP application provider purchases the network transport service signed in N-SLAs to support the VoIP application. After contracting S-SLAs and N-SLAs, the VoIP application provider has to adapt the VoIP quality level to maximize the profit and fulfill user requirements in any network condition according to the purchased network resource.

This work proposes a SLA-based quality level adaptation mechanism, and defines an optimal service model for a VoIP application provider under a broad range of traffic conditions. We assume that the transport networks have deployed QoS technologies to provide QoS guarantee, i.e., DiffServ/MPLS networks. The problem of adapting the VoIP quality level in the dynamic on-demand processing is formulated as a binary linear programming problem. Finally, the numerical results show that the proposed mechanism adapts the quality level of each active call effectively to any new network condition, and maximizes the profit for the VoIP application provider under the constraints of the contracted S-SLAs and N-SLAs.

II. Related Works

A lot of works on the issues of SLA definition, architecture, and SLA management can be found in [6]–[8]. The structure of SLA real-time management for multi-service packet networks and network admission control are discussed in [2]. From the view of business point, the maximizing benefit under users' requirements for service overlay network (SON) operator is discussed in [9].

Another study that uses an utility model to formulate the adaptive QoS management problem and maximize the profit of a network service provider can be found in [3]. To maximize the VoIP provider's utility, it dynamically adapts the operating quality of each VoIP session among a set of acceptable operating qualities under the resource constraint. Although the utility model contains illuminating discussions about QoS adaptation, it is driven by some prior off-line evaluation results. Consequently, it overlooks some fundamental questions. For example, it did not consider the influence of network condition, i.e., delay, loss, and jitter, on VoIP operating quality, or the derivation of network-level resource requirements from user-level QoS requirements.

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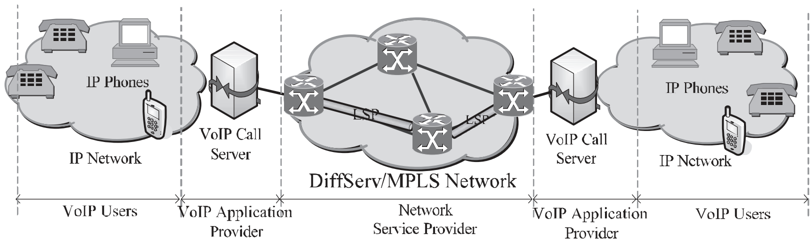

Fig. 1. VoIP services over DiffServ/MPLS networks.

Based on a realistic VoIP service environment, this paper is to play the role of a VoIP service provider that is a SON operator. The SLA-based quality level adaptation problem for on-line phase (or operation phase) is extended from [4]. Additionally, this problem is modeled to an utility model and formulated to an optimization problem. The dynamic changes of network condition and call arrival/leave are considered in this paper. The contribution can help a VoIP service provider to maximize the total profit under the contracted S-SLAs and N-SLAs in any realistic environment condition (i.e., any VoIP traffic condition and any network condition).

III. System Environment

Actually, VoIP service environment may be a mesh topology constructed with several VoIP call servers. In order to discuss easily, Figure 1 illustrates a part of the enterprise VoIP service environment. Unlike the conventional best-effort VoIP service on Internet, the VoIP application provider provides a QoS-oriented VoIP service via the QoS-enabled DiffServ/MPLS network. The VoIP application provider provides several quality profiles for its customers, e.g., high and low quality video/audio calls. Additionally, the network service provider offers a DiffServ/MPLS network to provide differentiated service (DiffServ) or simple class of service (COS) and bandwidth reservation to the VoIP application provider. For instance, high-quality call traffic is aggregated into a virtual trunk, called an LSP (Label Switched Path) in MPLS networks, and treated as the highest priority traffic in the whole network to satisfy the transport requirements. To reduce the call setup time, the DiffServ-enabled LSPs should be established by the network provider in advance after contracting N-SLAs. Finally, the VoIP traffic of each call is aggregated into the proper LSP using session classification [10] or other technologies.

The VoIP application provider in this environment not only provides the connectivity and interworking of VoIP services as well as the general VoIP application providers, but also offers the QoS-oriented VoIP service by integrating VoIPoMPLS (Voice over IP over MPLS) or VoMPLS (Voice over MPLS) technologies. The infrastructure of VoIP call servers, such as SIP proxy, SIP registrar, H.323 gatekeeper, signaling gateway and media gateway, must be established. Additionally, the VoIP application provider knows the extent of the dedicated network resources. Therefore, he can periodically probe the network quality and help user agents to determine an appropriate quality level to any new network condition during the call proceeding and their conversation.

Thus, to maximize the total profit and satisfy the contracted S-SLAs under the contracted N-SLAs, the VoIP application provider has to help the VoIP users to decide the proper quality level dynamically and adapt their quality level in a new VoIP traffic and network condition.

IV. Optimal SLA-Based Quality Adaptation Mechanism

To provide optimal service policies, this study models the behavior of the VoIP quality level adaptation in Fig. 2. The internal mapping functions (F_1,F_2, F_3 and F_4) are defined as below.

- *Quality-to-resource mapping* ($F_1(i,k)$): the mechanism maps session operation quality to the resource requirement. It means how much kth class network resource is needed to provide ith quality call. The measurement results of transport network indicate that the VoIP application provider is aware of the network abilities, such as the quality statistics of each network differentiated-service class in any particular network condition. Hence, the function F_1 can be defined by the VoIP application provider based on the quality statistics and the QoS requirements of each service quality. Briefly, F_1 is dynamically configured according to the measured network conditions.
- *Quality-to-revenue mapping* ($F_2(i)$): the mechanism maps session operating quality to the generated revenue. It indicates how much revenue is received by a ith quality call.
- *Violation-to-penalty mapping* ($F_3(i,j)$): the mechanism maps SLA violations to the incurred penalty. If the quality of jth call is downgraded to ith quality under the contracted S-SLA, the violation penalty is incurred.
- *Inefficiency-to-penalty mapping* ($F_4(k, L'_k/\eta_k R_k)$): the mechanism maps resource inefficiency to the incurred penalty. $L'_k/\eta_k R_k$ indicates the network target utilization of kth-class network resource. Thus, F_4 function can control the network inefficiency penalty. Either the expensive

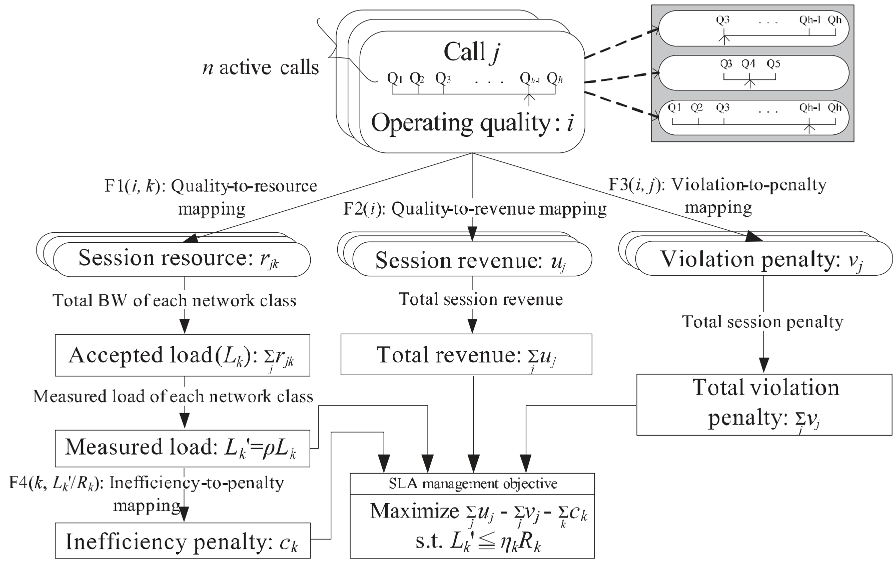

Fig. 2. The utility model of VoIP quality level adaptation.

network resource or the cheap network resource would be used first.

The VoIP application provider should configure the above internal mapping functions to enforce the service policy. The route selection policy plays a significant role in avoiding S-SLA violation, and is implied in F_1. Additionally, the VoIP application provider should also specify the target utilization level of each network differentiated-service class (η_k). To regulate the value of function F_1 in various network conditions, the network ability should be measured periodically. The accuracy of the network measurement is not discussed in this study; however, several network measurement techniques can be found in [11]–[14].

All parameters are listed in Table I. Besides the above internal mapping functions and parameters, to decide the optimal service policy, the VoIP application provider has to give the following parameters: the set of acceptable operating qualities (quality profile: Q_i) of each call, the number of quality profiles (h), the amount of current active calls (n), the number of network differentiated-service classes (m), the reserved bandwidth of each network service class (R_k), the target utilization level of each network service class (η_k) and the average load effective ratio of each call (ρ) are acquired.

To maximize the total profit, the appropriate quality of each call can be determined by solving the binary linear programming problem in Eq. (1) - (4). Let u_j, v_j and c_k denote the revenue of call j, the violation penalty of call j and the inefficiency penalty of network service class k, respectively. The total profit is equal to the total revenue decreased by the total violation penalty and the network inefficiency penalty. Finally, the decision variable x_{ijk} can be resolved. If $x_{ijk} = 1$, the call j is operated in the quality profile i and treated as the network service class k. Otherwise, $x_{ijk} = 0$. The optimization formulations are shown below.

TABLE I
PARAMETER LIST

Notation	Description
h	The number of quality profiles, e.g., 1st VoIP quality profile, 2nd VoIP quality profile, ..., hth VoIP quality profile.
m	The number of network differentiated-service classes, e.g., 1st network class, 2nd network class, ..., mth network class.
n	The amount of current active calls, e.g., 1st call, 2nd call, ..., hth call.
Q_i	The quality profile i where $i = 1, 2, ..., h$, e.g., Q_1, Q_2, ..., and Q_h.
R_k	The reserved bandwidth of each network service class where $k = 1, 2, ..., m$, e.g., R_1, R_2, ..., and R_m
η_k	The target utilization level of network service class k where $k = 1, 2, ..., m$.
ρ	The average load effective ratio of each call.
u_j	The revenue of call j where $j = 1, 2, ..., n$.
v_j	The violation penalty of call j where $j = 1, 2, ..., n$..
c_k	The inefficiency penalty of network class k where $k = 1, 2, ..., m$.
x_{ijk}	The decision variable. If $x_{ijk} = 1$, then call j is operated in the quality profile i and treated as the network class k. Otherwise, $x_{ijk} = 0$.

Objective:

$$\text{Maximize} \sum_{j=1}^{n} u_j - \sum_{j=1}^{n} v_j - \sum_{k=1}^{m} c_k \qquad (1)$$

TABLE II

AN EXAMPLE OF F_2 FUNCTION FOR 3-MINUTE CALL

Qualities (Q_i)	Call revenue
Gold (Q_1)	\$0.17
Silver (Q_2)	\$0.1
Copper (Q_3)	\$0.06

Subject to:

$$L'_k \leq \eta_k R_k,\ k = 1, 2, ..., m \quad (2)$$

$$\sum_{k=1}^{m} \sum_{i=1}^{h} x_{ijk} = 1,\ j = 1, 2, ..., n \quad (3)$$

$$x_{ijk} = 1/0,\ i = 1, 2, ..., h,\ j = 1, 2, ..., n \text{ and } k = 1, 2, ..., m \quad (4)$$

Where:

$$L'_k = \rho \sum_{j=1}^{n} r_{jk}$$

$$r_{jk} = \sum_{i=1}^{h} F_1(i, k) \times x_{ijk}$$

$$u_j = \sum_{k=1}^{m} \sum_{i=1}^{h} F_2(i) \times x_{ijk}$$

$$v_j = \sum_{k=1}^{m} \sum_{i=1}^{h} F_3(i, j) \times x_{ijk}$$

$$c_k = F_4(k, L'_k/\eta_k R_k)$$

In order to find the maximal profit, this problem can be resolved simply by brute-force search. However, the calculating complexity is $O((h \times m)^n)$. This is a NP-hard (nondeterministic polynomial-time hard) problem. With the increasing amount of active VoIP calls, the computing time is very significant.

V. NUMERICAL RESULTS

A VoIP application provider is assumed to provide a QoS-oriented VoIP service between both enterprise networks. In Table III, the VoIP application provider provides three quality profiles ($h = 3$) to their customers, denoted as Q_1, Q_2 and Q_3. The customers contract these *three quality profiles* and accept *dynamic quality level adaptation* during communication. Additionally, the average call holding time is set to 3 minutes. The F_2 function is listed in Table II. $F_2(Q_1)$ $F_2(Q_2)$ and $F_2(Q_3)$ of each 3-minute call equal \$0.17, \$0.1 and \$0.06, respectively. After beforehand SLA evaluation, the VoIP application provider purchases the network bandwidths of three differentiated service classes ($m = 3$), namely EF class, AF1 (Assured Forwarding) class and AF2 class, where $R_1 = 768$ Kbps, $R_2 = 1280$ Kbps and $R_3 = 256$ Kbps.

The network ability is assumed to be measured periodically by the VoIP application provider. Table III shows the F_1 functions of both network conditions, namely low network loading and high network loading. In the low network loading,

TABLE III

AN EXAMPLE OF QUALITY PROFILES AND F_1 FUNCTION

Qualities (Q_i)	Network req.	$F_1^L(i,k)$	$F_1^H(i,k)$
Gold (Q_1)	BW $\geq$ 384 Kbps delay $\leq$ 50 ms jitter $\leq$ 40 ms loss $\leq$ 1 %	EF: 384 Kbps AF1: 384 Kbps AF2: ∞	EF: 384 Kbps AF1: ∞ AF2: ∞
Silver (Q_2)	BW $\geq$ 256 Kbps delay $\leq$ 100 ms jitter $\leq$ 60 ms loss $\leq$ 3 %	EF: 256 Kbps AF1: 256 Kbps AF2: 256 Kbps	EF: 256 Kbps AF1: 256 Kbps AF2: ∞
Copper (Q_3)	BW $\geq$ 128 Kbps delay $\leq$ 200 ms jitter $\leq$ 80 ms loss $\leq$ 5 %	EF: 128 Kbps AF1: 128 Kbps AF2: 128 Kbps	EF: 128 Kbps AF1: 128 Kbps AF2: 128 Kbps

TABLE IV

AN EXAMPLE OF F_4 FUNCTION

Network Class	Inefficiency penalty
EF (k=1)	$(1 - L'_1/\eta_1 R_1) \times 10^{-3}$
AF1 (k=2)	$(1 - L'_2/\eta_2 R_2) \times 10^{-4}$
AF2 (k=3)	$(1 - L'_3/\eta_3 R_3) \times 10^{-5}$

the network requirements of the Q_1 VoIP service, in terms of bandwidth, delay, loss and jitter, can be satisfied by the EF class and AF1 class bandwidths (F_1^L). However, because the high network loading may have heavy network congestion, it can only be satisfied by the EF class bandwidth (F_1^H) in the high network loading.

To increase the network usage of the expensive network resource, the network inefficiency penalty is considered. The F_4 function is listed in Table IV. $F_4(k, L'_k/\eta_k R_k)$ functions of EF class, AF1 class and AF2 class are defined as $(1 - L'_k/\eta_k R_k)\times 10^{-3}$, $(1 - L'_k/\eta_k R_k)\times 10^{-4}$ and $(1 - L'_k/\eta_k R_k)\times 10^{-5}$, respectively. ρ and η_k for each differentiated service class k are set to 1. Since the customers set all quality profiles and accept dynamic quality level adaptation, no SLA violation occurs here, and the F_3 function is set to 0.

The problem is calculated by a high-performance optimization software (i.e., ILOG CPLEX [15]). Figures 3 (a) and (b) show the quality level distributions of the active calls in low and high network loading conditions, individually. The matrix Y shown below the bars is defined as the optimal operation status. The y_{ik} calls are operated in Q_i, and transmitted at network class k. For instance, while $n = 7$, the optimal solution in the low network loading is $\#Q_1 = 5$, $\#Q_2 = 1$ and $\#Q_3 = 1$. Furthermore, two of the five Q_1 calls are transmitted at EF class, and the other three are transmitted at AF1 class.

Experimental results demonstrate that each call with the optimal service policy can be successfully adapted to a new traffic and network condition. In the condition of low active calls and low network loading, the higher quality profile is selected first, and the calls are transmitted using the cheapest network bandwidth to maximize the total profit. For example, several Q_1 calls are treated as AF1 class between $n = 3$ and $n = 14$. However, in the condition of high network loading, because the F_1^H constraints are stricter than that of F_1^L, the

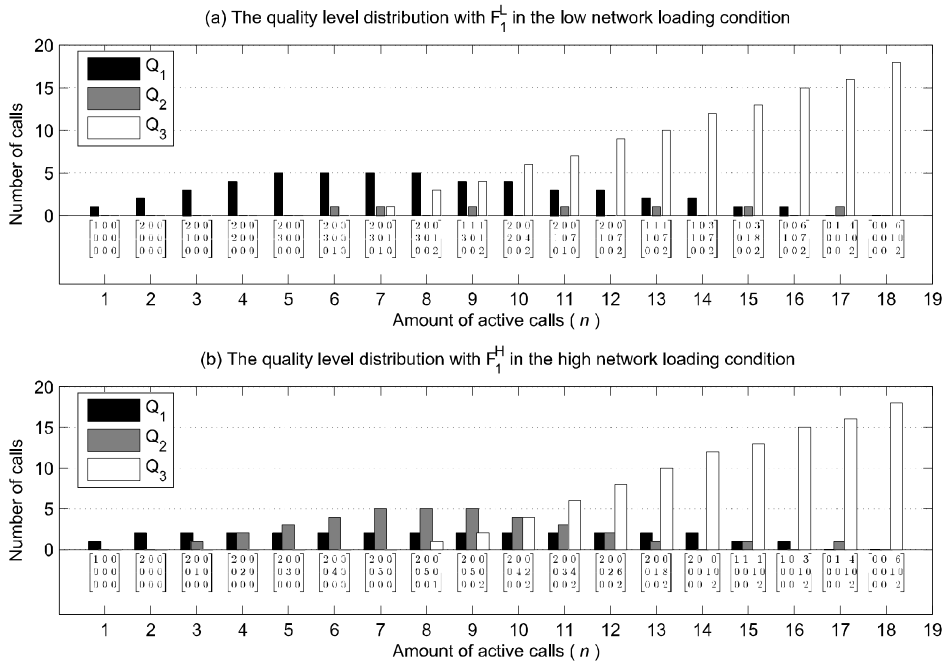

Fig. 3. The quality level distributions.

number of Q_1 calls is limited, while the available EF class resource is empty. Additionally, regardless of network loading, the quality levels in low and high network loading conditions are degraded to maximize the total capacity when the number of active calls is increasing.

Figure 4 shows the total optimal profits. Since the constraints of the higher quality profile requirements are relaxed in the low network loading, the total profit in low network loading is observably higher than the profit in high network loading. Thus, the results would help the VoIP application provider develop CAC (Call Admission Control) policies. The CAC should either degrade the quality levels of the existent active calls and accept the new call to maximize the capacity, or reject it to keep the maximal profit.

VI. Conclusion

This work proposes an optimal SLA-based quality adaptation mechanism, and defines an optimal service model for a VoIP application provider over DiffServ/MPLS networks between two VoIP call servers. The problem of the VoIP quality level adaptation is formulated in the dynamic on-demand processing as a binary linear programming problem. The numerical results show that the proposed mechanism can adapt the quality level of each active call to the variety of traffic and network conditions well, and maximize the total profit under the constraints of the contracted S-SLAs and N-SLAs. Thus, suchcontributions would help a VoIP applications provider to develop CAC and proprietary service policies efficiently. In the future, we are also going to develop a heuristic mechanism to improve the calculating speed for a large-scale environment.

Acknowledgement

We would like to thank the National Science Council (NSC) in Taiwan. This research was supported in part by the NSC under the grant number NSC-97-2221-E-343-003 and NSC-96-2221-E-343-001. Moreover, we would also thank Dr. Kim-Joan Chen, Department of Electrical Engineering, National Chung-Cheng University. He offered his personal research resources to support this research.

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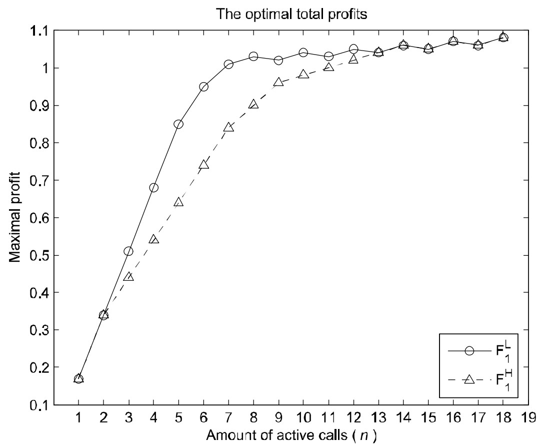

Fig. 4. The optimal total profits.

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Design of Cluster-based System Framework for SIP-based Multimedia Conferencing Services

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***Abstract*—Multimedia conferencing is a communication service to approach multi-party, real-time, interactive and multimedia applications. Due to mature Internet technologies and ubiquitous network access services, more and more Internet applications about real-time multimedia conferencing are used. In the current conferencing systems, the media processing in centralized mixer servers is the system bottleneck. The communication quality will be decreased because of the central system loading. This paper proposes a cluster-based framework to provide a SIP-based multimedia conferencing service. In this framework, the cluster members are self-organizing and self-maintaining to share the resources of mixers within the cluster. Additionally, the extended SIP messages and SIP-based clustering mechanism are designed in this paper. Thus, if the resources of the mixers can be allocated in an optimal planning, the mixer loading can be distributed well to each mixer in the cluster. The system requirements of super-performance mixer servers can be avoided. In the framework, not only the system loading of conference servers can be distributed well, but also the infrastructure cost can be reduced.**

I. Introduction

Multimedia conferencing is a multi-party interactive real-time communication service, which connects two or more endpoints over IP networks. The users can communicate to all participants with multimedia applications in the same conference, such as audio/video, whiteboard, file transfer, etc. In order to support the service reliability, availability and security, a multimedia conferencing server, i.e., Focus, helps users to handle the conference management and control. The multimedia conferencing system realizes a real-time interactive conference as sharing a virtual working space through Internet, the users can join and leave a conference remotely, and they feel like the same in a real conference. Additionally, due to the Internet characteristics of resource sharing and ubiquity, the multimedia conferencing provides lower cost and higher flexibility than the traditional video conferencing in telecommunications.

Nowadays, several researches and standards about multimedia conferencing architectures were proposed [1], [2]. Their conference models can be separated as follows:

- Loosely coupled model,
- Tightly coupled model, and
- Fully distributed multiparty model

The loosely coupled model is usually based on IP multicast, and there is no central point of control or conference server. This model is suitable for applying in a large-scale environment, but it is lack of reliability.

In the tightly coupled model, several entities are implemented as servers in the network, such as Focus and Mixer. These entities are deployed to enforce the conference management and control. The reliability and stability can be guaranteed well, since the states and memberships of conferences are maintained in the centralized servers. However, the model should deploy more centralized servers and apply hierarchical management to a large-scale environment, but this needs many powerful servers and may increase higher communication cost.

The fully distributed multiparty model so-called a peer-to-peer conference model is discussed recently. Each participant contacts to other participants in order to negotiate in a conference. There is no central point of control in this model.

In this paper, we propose a cluster-based conferencing architecture to provide multimedia conferencing services over IP networks. Our motivations are to share users' resource and reduce the cost of infrastructure establishment. In our framework, a clustering mechanism is design to achieve the user collaboration. Except for the low-ability peers, i.e. clients, the high-ability peers in the same cluster can share their computing and audio/video mixing resources to support intra-cluster multimedia conferences.

However, the quality of long-distance conferences in a large-scale IP networks is very difficult to control well. In our framework, a centralized multimedia conferencing backbone network are deployed to deal with long-distance conferences in the inter-cluster. Therefore, due to the user collaboration, the system loading of the multimedia conferencing backbone network would be reduced and distributed to high-ability peers. The system requirements of super-performance mixer servers can be avoided. Thus, if the minimal resource utilization of each cluster can be maximized, the infrastructure cost of the multimedia conferencing backbone network will be minimized.

This paper is organized below. In Section II, we introduce the related works about multimedia conferencing system architecture. In Section III, we propose a cluster-based multimedia

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conferencing system architecture. The clustering mechanism is shown in Section IV. And, a use case for multimedia conferencing service is explained in Section V. Finally, we give some conclusions in Section VI.

II. Related Works

The conferencing models can be separated into loosely coupled model, tightly coupled model and fully distributed multiparty model. Because of the approachability, several working groups in IETF are focus on the tightly coupled model, sucn as MMUSIC [3] and XCON [4] working groups. The related works about the tightly coupled model can be found in [1] and [5].

In the tightly coupled model, three logical entities, i.e. Conference Server, Focus and Mixer, are implemented to provide the conference management and audio/video mixing for the tightly coupled conferences. Additionally, the functionalities of Participants Management, Floor Control and Media Management are performed in the Conference Server. Focus is a center of conference control, and its primary assignment is to handle the signaling of conferences. Mixer is an entity that mixes the audio/video streams of a conference and redistributes them to each participant in the same conference. Mixers are managed by the Focus. Additionally, RFC 4353 [1] still proposed five physical instantiations based on the tightly coupled model, these instantiations are developed to meet the different requirements for conferencing.

In the above frameworks, the reliability and the stability of conferences can be controlled well, since the states and memberships of conferences are maintained in the centralized servers. However, all these frameworks are suffered by the overweighted system loading and the limited scalability. The overall system loading would be aggregated into some critical points. If one critical server is crashed, the conference service would be disrupted. Besides, the infrastructure cost is raised rapidly with the service scale increasing. Therefore, if the system loading can be distributed to user agents, the scalability and system performance can be improved.

III. Multimedia Conferencing System Architecture

A. *System Architecture*

Our system framework is shown in Fig. 1. The system has two kinds of networks, the multimedia conferencing backbone network (MCBN) and the cluster. The MCBN is a conference virtual overlay network over IP network, and it provides a tightly coupled multimedia conferencing service. The MCBN consists of the main centralized servers, such as Focuses, Mixers and signaling servers. The deployment of these servers is based on the cascaded mixers model [1].

A cluster is a group of user agents, the cluster members are self-organizing and self-maintaining. A cluster head is selected to perform group management, i.e., cluster maintenance, heartbeat notifications and messages forwarding. Additionally, the clusters are managed by the signaling server.

B. *The Multimedia Conferencing Backbone Network*

The multimedia conferencing backbone network consists of three logical entities that are implemented on different servers. These entities are Focuses, Mixers and signaling servers.

The functionalities of Focus are introduced in Section II. A least one Focus should be constructed in our architecture. The number of Focuses may be increased to support a large-scale environment. The Focus contains three management functions and a signaling handler, i.e., participant management, mixer management and floor control. The participant management deals with the affiliation and departure of a conference participant. In addition, it generates notifications after the states of the participants are changed, such as join and leave. The mixer management handles the mixer configuration and the media capability negotiation. The third function, floor control, manages the preaudience and the conference resources. The signaling handler receives or generates SIP messages, and it parses the SIP messages and represents its content.

The functionalities of Mixer are introduced in Section II. In our architecture, the Mixers are deployed near to the specified domains. However, several specified domains may share the resource of the same mixer. The mixers' topology in our system is arranged to a hierarchical architecture that is the same as the cascaded mixers model defined in [1], if the distribution of the participants is wide. Besides, the mixer also mixes the media streams which belong to the same conference from the other mixers.

The signaling servers deal with user registration, destination querying and SIP messages forwarding, those functionalities are handled by three logical entities in SIP, i.e. Registrar, Redirect and Proxy, respectively, and implemented in the signaling server [6]. A new logical entity named *Cluster Manager* is defined, its functional block is shown in Fig. 2. The functions in the Cluster Manager are explained below:

- *Member Management:*
  This function is different from the participant management in the Focus. It maintains the profile of each user in our system. User authentication, user authorization and user accounting are included.
- *Cluster Maintenance:*
  Cluster formation and cluster head selection are handled by this function. The function also detects cluster head failure and provides failure recovery by using a heartbeat mechanism, which was designed in this paper.

C. *The Clusters*

Before introducing the concept of cluster, two types of user agents are defined. The user agents are classified into the peer type due to their low capability of hardware and network bandwidth. The hardware means the CPU, RAM and storage of the user agent's device. A threshold, like a minimum requirement, is given to determine whether a device resource can be shared or not. It is defined by application developers. The user agents that satisfy the requirements are classified into the superior type.

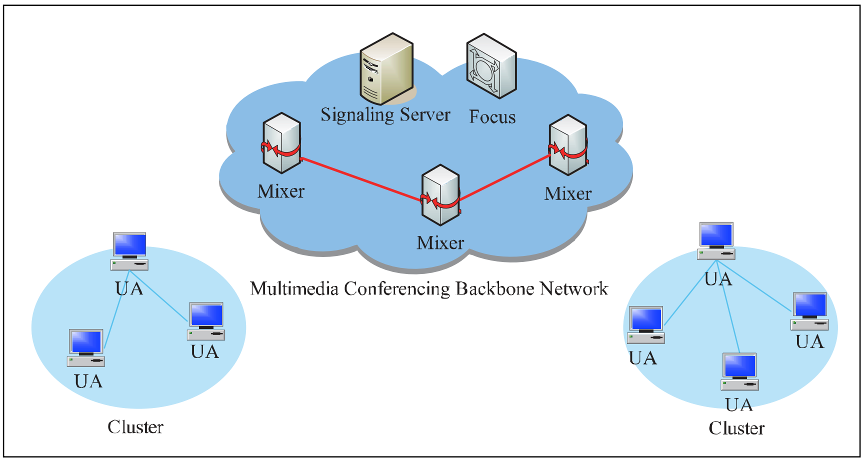

Fig. 1. The system framework for cluster-based multimedia conferencing services over IP networks.

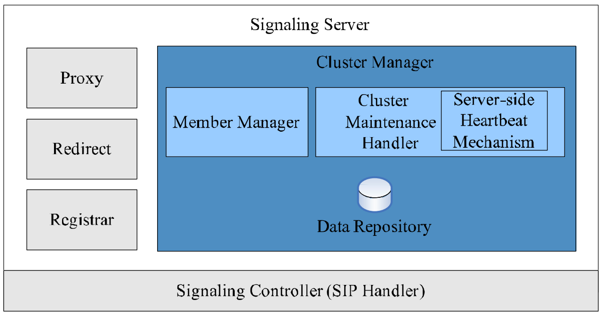

Fig. 2. The functional blocks of signaling server.

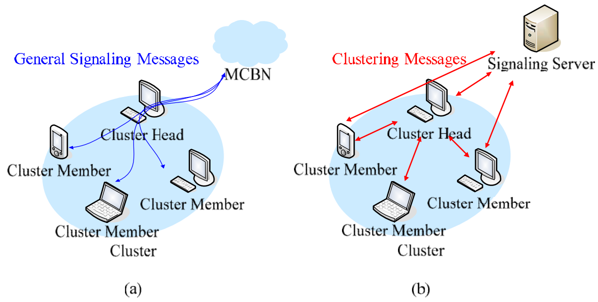

Fig. 3. (a) The data flow of general signaling messages, and (b) the data flow of clustering messages.

Users are grouped into clusters, and a cluster head is selected from the superiors in the same cluster. The remain users in the cluster are called cluster members. The cluster head maintains the state of cluster relationship for each cluster member, and it notifies every cluster member when the relationship is changed. The registration status is also maintained for the purpose of authorization for each cluster member in the cluster. The third information, the mixing capacity of each superior in the cluster, is collected in advance in order to perform mixing resource management.

Signaling flows can divide into general signaling messages and clustering messages. The signaling messages are relayed to the MCBN by cluster heads, and the clustering messages are exchanged with the cluster head. The clustering messages are defined in the extended heartbeat mechanism which is proposed in our system framework. The illustrations of signaling flows are shown in Fig. 3.

IV. Clustering Mechanism

The cluster mechanism includes the user registration, the heartbeat mechanism and the cluster maintenance. The related cluster mechanisms in Mobile Ad Hoc Networks (MANET) can be found in [7] and [8].

A. User Registration

The user registration in our system is separated into two modes. One of the modes is that a user agent registers at the signaling server directly. This situation is often occurred when no cluster exists or the new peer cannot find an available cluster head. The other mode is that a user agent registers at the signaling server through a cluster head, i.e. the register messages are relayed by a cluster head. The register messages include the SIP REGISTER request and the relative response. Two SIP header fields in the REGISTER request are extended. The *UA-Type* indicates the type of a user agent, and the *TimeStamp* indicates whether a superior list is kept up to date or not. The superior list records the IP address, port number and estimated availability of a superior. The signaling server possesses the latest superior list for each specific domain. The superior list is used in the Cluster Head Discovery (CHD). An example is shown in Fig. 4.

The superior with the highest estimated availability in a cluster will be selected as the cluster head. A cluster head selection is occurred when a cluster membership joins and

```
<?xml version="1.0"?>
...
 <config timstamp="20080323133036">
   <superior_list>
     <superior>
       <address>140.123.107.21</address>
       <port>12345</port>
       <availability>0.75</availability>
    </superior>
    <superior>
       <address>140.123.107.125</address>
       <port>12345</port>
      <availability>0.60</availability>
    </superior>
   </superior_list>
   ...
 </config>
...
```

Fig. 4. An example of superior list.

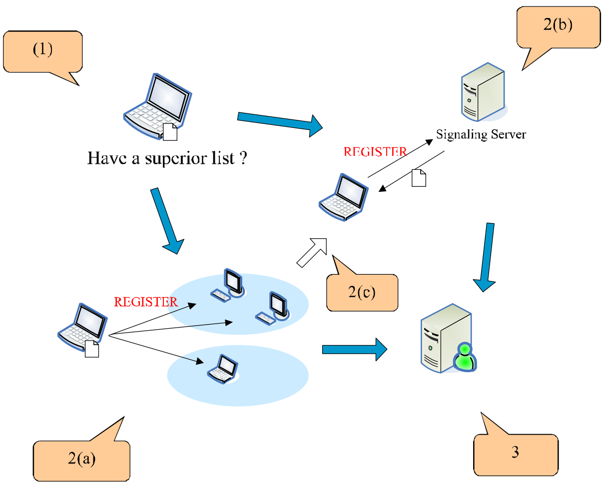

Fig. 5. The work flow of user registration.

leaves. The estimated availability of each superior is calculated after registering. If a user logs in the system for the first time, the estimated availability is given to a default value. The estimated availability is an exponentially weighted moving average. The formulation of estimated availability is shown in Eq. 1.

$$E = (1 - \alpha) \times E' + \alpha \times a \qquad (1)$$

where E' denotes the previous estimated availability, and α is the sample availability. The characteristic of this average is that it takes the historical behavior into account. We can manipulate the ratio of the historical average by configuring the coefficient α in Eq. 1. The α is recommended to set for 0.125 since it gives the average a better smoothly adjustment. In the equation, a denotes the sample availability of a superior. The sample availability is a measured availability from sampling.

The illustration of registration workflow is shown in Fig. 5. It is described below:

- **Step 1:** While a user agent starts to log in the system, it first examines the existence of superior list on its local system. If a superior list exists, the progress will go to step 2(a), otherwise, will go to 2(b).
- **Step 2(a):** The progress is named the Cluster Head Discovery (CHD). The user agent sends the register messages to all superiors listed in the superior list. If there exits at least one superior that is playing the role of cluster head, the superior, i.e. the cluster head, will return the acknowledged response immediately. The user agent then registers at the signaling server through the cluster head after receiving the acknowledged response from the cluster head. In addition, a user agent may receive several acknowledged responses. Once it occurs, the user agent will choose the cluster head with a rapid response.
- **Step 2(b):** Because there is no superior list on the local system, the user agent then registers at the signaling server directly. During the registration, the response that returns to the user agent carries a latest superior list.
- **Step 2(c):** The CHD may fail due to the nonexistence of cluster or the out-of-date superior list. In this case, the progress turns to step 2(b).
- **Step 3:** The registration is successfully completed.

B. The Enhanced SIP Heartbeat Mechanism

The heartbeat mechanism is an integrated model of the SIP registration framework and the SIP event notification framework [9]. Two events are extended in this mechanism. One is called *Refer* event which is defined in RFC 3515 [10], and this event provides a mechanism to refer a remote node to the resource indicated in the request. The other event named *Cluster* is proposed in our system architecture. This event defines the generation of notifications for the change in a cluster membership, and it also gives a method to detect a cluster member failure.

The framework of heartbeat mechanism is shown in Fig. 6. Every cluster head is a central notifier. Those clusters will send notifications to their cluster members when the state of cluster membership is changed. The state machine of cluster member is shown in Fig. 7. The state machine of each cluster member is maintained by the cluster head. If there is a change to the states, the cluster head will generate a notification to notify the subscribers [11].

C. Cluster Maintenance

There are seven cluster operations in our system architecture. We describe them as follows:

1) **Joining a cluster:**
When a user wants to join a cluster, it first completes the registration process through the cluster head, and then sends a *SUBSCRIBE* request to the cluster head. The cluster head will check the registration status of the user before approving the subscription. It will reject of the subscription if the user has not yet registered.
2) **Leaving a cluster:**

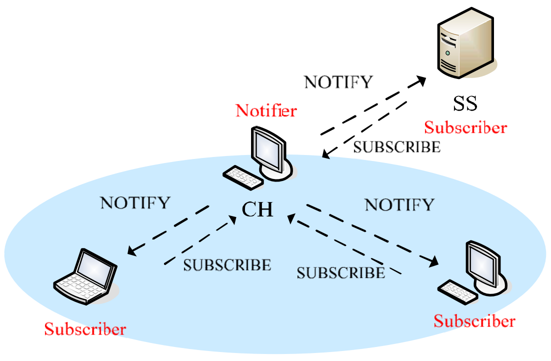

Fig. 6. The framework of heartbeat mechanism.

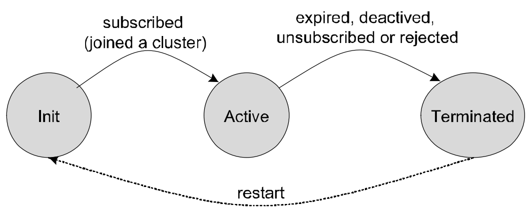

Fig. 7. The state machine of cluster member.

When a user wants to leave a cluster, it unregisters the registration, i.e. sending a *REGISTER* request that contains the *Expires* header field equaled to "0". And, the cluster head will remove the relative subscription and forward the request immediately. Moreover, the cluster head will also set the state of the cluster member to *Terminated* state and will generate a notification for this change.

3) **Cluster head resignation:**
While a new cluster member has successfully joined a cluster, the signaling server will make a comparison on the estimated availabilities of the joiner and the cluster head. If the joiner has a higher estimated availability, the signaling server reassigns it to be the new cluster head. When a cluster head resignation begins, the signaling server sends a SUBSCRIBE request to the new selected cluster head. After subscription, the signaling server informs the old cluster head to terminate its subscription by sending a *SUBSCRIBE* request with the expiration interval of "0". The old cluster head subsequently removes all the subscriptions relative to the *Cluster* event. In the mean time, the new cluster head sends a *REFER* request to each cluster member. Briefly, the *REFER* request advertises the cluster members to establish relationships with the new cluster head. After all the cluster members have connected to the new cluster head, the cluster head resignation is complete.

4) **Cluster splitting:**
Here, we refer to the cluster splitting in peer-to-peer Ad Hoc networks [7]. A split value (Sv) is configured to indicate a threshold for the cluster splitting. When a new cluster member arrives, the signaling server checks the number of members in the cluster. If the number of members achieves the spit value, the signaling server will perform the cluster splitting. First, the signaling server instructs the cluster member with the second highest estimated availability to leave the cluster and sends a *SUBSCRIBE* to it. After subscribed, the signaling server instructs half the rest of the cluster members to join the new cluster.

5) **Cluster merging:**
In the cluster merging procedure, a specified merging value which indicates the trigger threshold is used. The merging procedure is much simpler than the splitting procedure. Briefly, the signaling server sends *REFER* requests to those nodes that are going to be merged. Those requests instruct the nodes to join a merging cluster. The merging cluster is selected by the signaling server.

6) **Abnormal disconnection:**
The abnormal disconnection is divided into two situations. Situation (1) is occurred when the cluster head detects an abnormal event, i.e. the response lost for a notification. In this case, the cluster head becomes a failover controller in the abnormal operation. In Situation (2), a cluster head is out of order. The signaling server detected the cluster head is abnormal, and it then becomes the failover controller.

7) **Inter-cluster Conference and Intra-cluster Conference:**
Two scenarios are presented, the inter-cluster conference and the intra-cluster conference. A conference that takes place across two clusters or more is classified as an inter-cluster conference. The inter-cluster conference is illustrated in Fig. 8. Inter-cluster conferences are handled by the Focus in the MCBN, and the media streams are gathered to the MCBN.
An intra-cluster conference is a conference that takes place inside the cluster. Briefly, all the conference participants are in the same cluster. The illustration of intra-cluster conference is shown in Fig. 9. If there are enough resources in a cluster, the intra-cluster conferences will be handled by the cluster head (or named P2PFocus), and the media streams are mixed by the superiors (or named P2PMixers) in the cluster. Once the resources are consumed, the MCBN will take over the conference.

V. A Use Case of Multimedia Conferencing

A. *User Registration*

In the use case, the scenario of user registration through a cluster head is presented. The user registration is introduced in Section IV-A.

The sequence chart is shown in Fig. 10. First, a cluster is existent in the system, and Bob is the cluster head. Meanwhile,

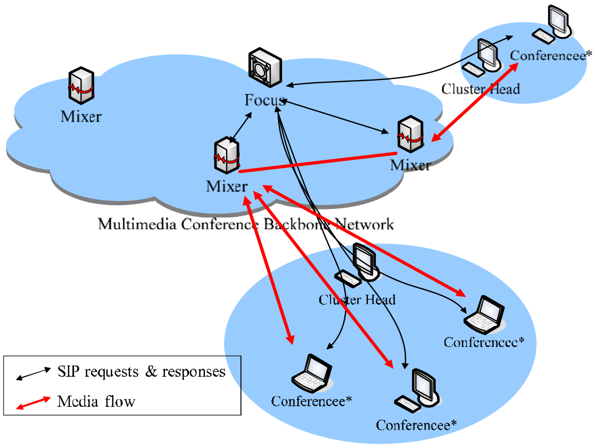

Fig. 8. A scenario of inter-cluster conference.

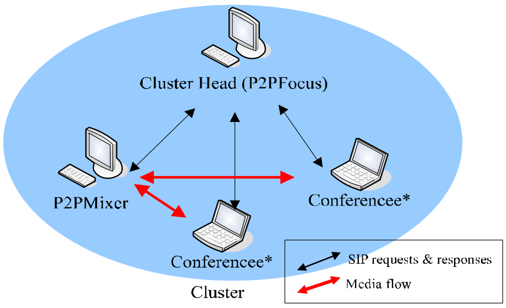

Fig. 9. A scenario of intra-cluster conference.

a user agent, Alice, wants to log in the system. We assume that Alice has a superior list on its local system. Alice finds and contacts Bob through the cluster head discovery explained in Section 4. Alice then registers at the signaling server through Bob, which are shown in messages F1-F4. The messages F5 and F6 show the subscription, and that also represent that Alice joins the cluster. An immediate notification is shown in messages F7 and F8.

B. Intra-cluster Conference

In the following use case, a scenario of intra-cluster conference is presented, and its sequence chart is shown in Fig. 11. An intra-cluster conference is defined to be a conference that takes place inside a cluster that is introduced in Section IV-C.

In this scenario, another two user agents, Joe and Tom, has logged in the system and joined to the cluster. Figure 11 shows that Joe wants to create a conference with Alice and Tom. He sends an INVITE message to Bob to request a conference setup, which is shown in message F11. Bob notices that the request is an intra-cluster conference, so that its user agent initializes the P2PFocus process to handle this conference. The P2PFocus checks the capability of each superior in the cluster for media mixing. In this case, Alice is assigned to be the

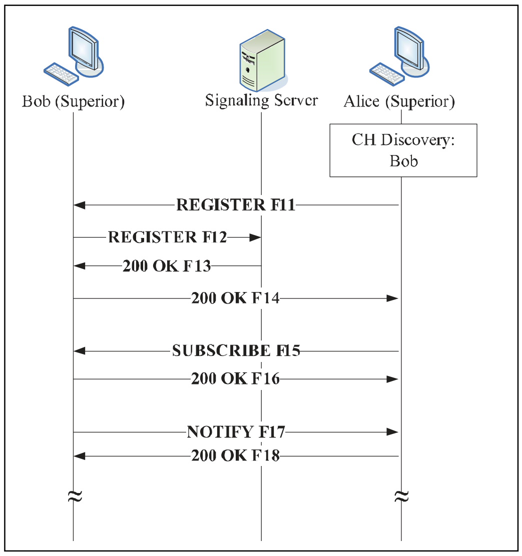

Fig. 10. The sequence chart of user registration while Alice logs in the system.

P2PMixer for this conference, the P2PFocus invites Alice with a INVITE message shown in message F12. After receiving the message, the user agent initializes the P2PMixer process.

The P2PFocus configures the media channel between Joe and the P2PMixer by using third party call control [12] and an offer/answer model with SDP [13], which is shown in messages F11-F16. After a media channel has been setup, Joe can send media streams to the specified P2PMixer. The P2PFocus then invites the other user agents, Tom and Alice, to join the conference shown in messages F17 and F23. The messages F17-F22 indicate the media channel setup between Tom and the P2PMixer. The media channel setup between Alice and the P2PMixer is omitted here. The detail of the media channel setup can be referred to the conference paper [14]. Finally, the conference participants begin the conversation with others.

VI. Conclusion

We propose a cluster-based system architecture for multimedia conferencing services over IP network in this paper. Our system architecture takes the advantage of the user collaboration, not only the system loading of conference servers can be minimized, but also the infrastructure cost can be reduced. Besides, the centralized multimedia conferencing backbone network can improve the overall reliability and the stability of the cluster-based conferencing system. A clustering mechanism was designed in the paper, and the mechanism extends the Session Initial Protocol(SIP) to achieve the cluster operation and the heartbeat mechanism. A use case of our system prototype with sequence charts is explained to show the system is practicable. In our framework, some practical issues are still needed to be deeply investigated, such as conference policy control, floor control, resource allocation

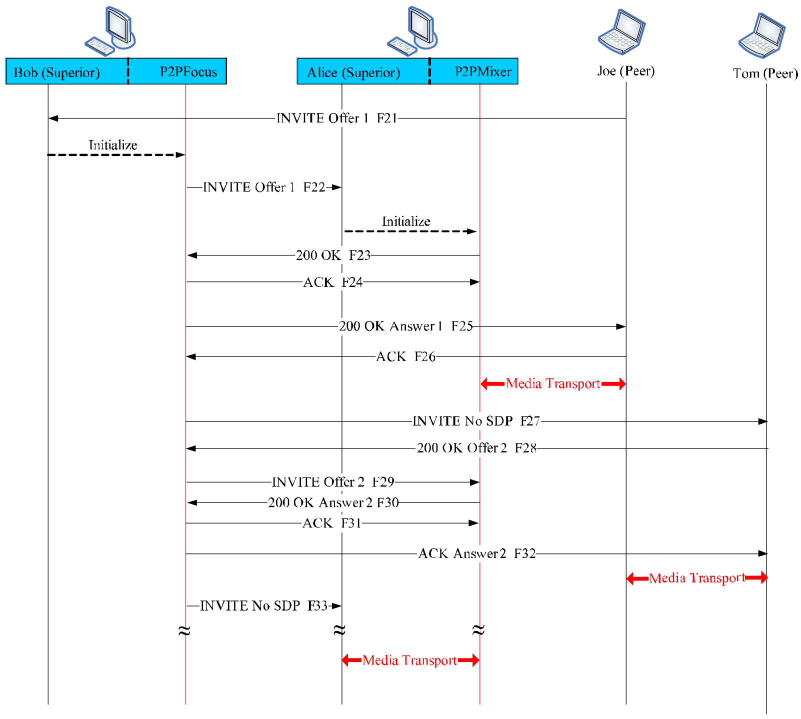

Fig. 11. The sequence chart of intra-cluster conference while Joe creates a conference.

and planning, etc. Moreover, the intra-cluster mixer resource allocation problem in the environment can be modeled to a mathematical optimization problem. If the mixer resource can be allocated in optimal solution, the mixer loading can be distributed well to each mixer within a cluster. According to the optimization model, we will find the optimal solution and apply it to our system framework.

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