行政院及所屬各機關出國報告

（出國類別 ： 實習）

VOIP 交換機維運系統研發設備技術

實習報告

服務機關 ： 中華電信研究所

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出國地區 ： 美國 紐約

出國期間 ：92 年 11 月 29 日至 92 年 12 月 12 日

報告日期 ：93 年 1 月 27 日

摘　要

由於利用網際網路傳送語音通信(VOIP)相關技術之突飛猛進，在話音品質及應用領域將日趨成熟，未來勢必成為通信網路之骨幹。本公司已積極規劃引進及建設 IP 網路應用，本所為研發 VOIP 相關技術之需求，因此有必要至國外電信機構現場實習，有關 VOIP 技術之設計應用與維運管理經驗，以提供本公司 VOIP 規劃建設，維運及研發相關產品之參考。

本次實習係依照本所 92 年度”VOIP 交換機維運系統研發設備”R920288 購案合約辦理，於 92 年 11 月 29 日啟程，赴美國 Intel 公司實習 VOIP 交換機相關技術，並於 92 年 12 月 12 日完成課程順利返國。

此次實習主要是針對 Intel 公司的 IP Media Gateway(IPT 1200C) 及 IP PSTN Gateway(DM/IP481)等系統設備，參與該公司訓練課程、現場實習操作、技術討論、應用實例參訪等，行程緊湊，獲益良多。

經過此次實習，職除了對於 Intel 公司 VOIP 相關產品有深入了解外，對於 VOIP 設計原理及應用趨勢亦有更深確的認識，將可提供本公司 VOIP 相關技術規範制定、建設規劃及本所研發 VOIP 利基產品之參考　。

本報告首先簡述此次實習之目的，接著列出主要行程及實習內容，包括 Intel 公司之 VOIP Media Gateway 相關產品之主要特點及系統架構，硬軟體設計，第三部份描述實習心得，最後提出建議供有關單位參考。

VOIP 交換機維運系統研發設備技術

出國實習報告

目　錄

VOIP 交換機維運系統研發設備技術

出國實習報告

1. 目的

由於利用網際網路傳送語音通信(VOIP)相關技術之突飛猛進，在話音品質及應用領域將日趨成熟，未來勢必成為通信網路之骨幹。本公司已積極規劃引進及建設 IP 網路應用，本所為研發 VOIP 相關技術之需求，因此有必要至國外電信機構現場實習，有關 VOIP 技術之設計應用與維運管理經驗，以提供本公司 VOIP 規劃建設，維運及研發相關產品之參考。

2. 過程

2.1. 行程

日期	地點	工作
92 年 11 月 29~30 日	桃園- 紐約	去程
92 年 12 月 1~10 日	紐約	Intel 公司實習 VOIP 設備參訪
92 年 12 月 11~12 日	紐約- 桃園	回程

2.2. 內容

本次實習主要係針對 Intel 公司 IP Media Gateway(IPT 1200C) 及 IP PSTN Gateway(DM/IP481)等系統設備，參與該公司訓練課程、現場實習操作、技術討論、應用實例參訪等，茲摘要如下 ：

2.2.1 IP Media Gateway(IPT 1200C)中介閘道器

2.2.1.1. 主要特點

IP Media Gateway(IPT 1200C)主要特點如下:

- 具備 Open, standards-based platforms
- 具備 Scalable,carrier-grade solutions
- 支援 672 Port 之 IP 電話路由,以發展具擴充性系統
- 提供乙太網路介面
  - Dual 100Base-TX
  - Dual 100Base-T
  - Provide indicators on the front panel
  - Provide Ethernet interface Link Status
- 傳送編碼器(Transcoding)
  - G.711 A-Law and mu-Law
  - G.723.1 (5.3 Kbps and 6.3 Kbps)
  - G726(Supported via IP Media Gateway)
  - G729 Annex A
- 靜音抑制(Silence Suppression VAD)
  - Ability to detect the absence of audio
- 雙音複頻信號處理 (DTMF Processing)
  - Ability to generate DTMF to and receive notification
  - From IP side via UII alphanumeric message (out of band)
  - From TDM side inband
  - Inband DTMF pass-through
  - RFC 2833 pass-through
- 服務品質(Quality of Service QOS)
  - Support the setting and retrieving of QoS
  - Setting and retrieving of QoS threshold and handling of QoS alarm
  - QoS threshold support : Lost packets, jitter, Roundtrip latency
  - Support the type of services(ToS)
- 支援 IP 傳送語音 (Voice over IP support)
  - RTP 處理

- RTCP 處理
- Jitter 緩衝器管理
- Packet loss concealment

● 傳真功能
  - 經由 UDP 送收傳真信號
  - Notification of audio to Fax and Fax to audio
  - Support T.38 Fax relay

● 具備 H.323 規約 Stack
  - 支援 host-based H.323 Protocol stack
  - Establishing calls over IP network
  - Using the Global Call API for call control

● 具備 SIP 規約 Stack
  - Support the host-based SIP RADVISION
  - Establishing calls over IP network
  - Using the Global Call API for call control

● 具備 SNMP 管理功能
  - Monitoring and control is provided
  - Provide Administration Software

● 具備應用界面(Application interface)
  - Support the R4 Programming environment
  - Global Call API for call control
  - IP Media Library(IPML) API for media management

● 回音消除 (Echo cancellation)
  - Provided on incoming TDM data on a per channel basis
  - The tail length can be set for each channel
  - The following tail length are supported
  - 8 ms
  - 16 ms
  - 32 m

2.2.2 IP PSTN Gateway(DM/IP 481)電話網路閘道器

2.2.2.1. 主要特點

IP PSTN Gateway(DM/IP 481)主要特點如下:

● 具備 Equiped wish a PSTN network front end

● 具備 connected via NetSructure platform to the CT Bus

- 具備 Single board digital IP to PSTN Gateway application
- 具備 Conferenceing Resource
  - Shareable conference resource
  - Run time add ane remove conference
  - Active talker notification
  - Automatic notification on adding and removing parties
  - DTMF clamping
  - Volume control
  - Auromatic gain control
  - Echo Cancellation
  - Monitoring function
- 支援 RFC2833 以發展具擴充性系統
- 支援由 IP 載送傳真信號
  - Support ITU T.38 for real time fax transmission
  - 經由 T.38 閘道器 透過 IP-based Network
- 具備 H.323 規約 Stack
  - Support the host-based H.323 RADVISION
  - Includeed system release software
  - Providing IP signaling for establishing calls over IP network
  - Using Global Call API for call control
- 雙音複頻信號處理 (DTMF Processing)
  - Ability to generate DTMF to and receive notification
  - From IP side via UII alphanumeric message (out of band)
  - From TDM side inband
  - Inband DTMF pass-through
  - RFC 2833 pass-through
- 服務品質(Quality of Service QOS)
  - Support the setting and retrieving of QoS
  - Setting and retrieving of QoS threshold and handling of QoS alarm
  - QoS threshold support : Lost packets, jitter, Roundtrip latency
  - Support the type of services(ToS)
- 支援 IP 傳送語音 (Voice over IP support)
  - RTP 處理
  - RTCP 處理
  - Jitter 緩衝器管理
  - Packet loss concealm
- 具備 SIP 規約 Stack
  - Support the host-based SIP RADVISION

- Establishing calls over IP network
- Using the Global Call API for call control

- 具備 SNMP 管理功能
  - Monitoring and control is provided
  - Provide Administration Software
- 具備應用界面(Application interface)
  - Support the R4 Programming environment
  - Global Call API for call control
  - IP Media Library(IPML) API for media management
- 具備 Call control implemented on the host, RTC on board
- 提供 Split call control via the IP Media Library API
- 支援 Global call API support for IP and PSTN
- 提供 FCD file generation utility
- 支援 Flexing routing (exportable voice resource)
- 具備 Standard Internet protocol including TCP/IP,UDP and RTP/RTCP
- 具備與解碼器全雙工通信功能
- 具備 Coder support including GSM,G.729
- 具備 Microsoft Netmeeting and Vocal tec
- 提供 IP switching bridge two calls with minimum latency
- 提供 Type of service (ToS) byte
- 具備 Packet redundancy : comply with RFC 2189
- 提供 IP media service
- 提供 High reliability configuration
- 具備 IP Voice stream Resource
- 具備 Object ID support when using Non-standard command
- 提供 Voice quality parameter and volume control
- 具備呼叫程控功能
  - Continuous Speech Processing (CSP) support
  - Support NFAS on N12 protocals
  - Support R2MF hot down procotal
  - Support PSTN (4 ESS,5 ESS, DMS,CAS) signaling
- 具備語音特性控制功能
  - Gain control
  - DTMF volume control
  - On board PSTN interfaces
  - Support digital interface
  - Support Media load 1 voice and load 2 enhance voice
  - Basic DTMF and MF detect

2.2.3. 系統發展軟體(Development Software)

2.3.1. 應用程式庫(API Libraries)

2.3.1.1.新事件服務程式(New event service)程式庫

- Provide an interface for registering any application with the Intel Dialogic event notification framk
- Subsystem for sending asynchronous, system administration events to application
- Generated when activities take place in the system
- Single board stop or start , board removal or insertion
- CT bus line failures and T1/E1 network alarm

2.3.1.2. 語音會議電話(Audio conference)程式庫

- Support development of host-based conferenceing application
- Develop customized audio conferencing services
- Including library functions, divice drivers, and firmware
- Provide conference bridging and monitoring
- Support per party basis of DTMF clamping
- Adjust the listening volume of conference
- Echo cancellation for each active talker
- High port density

2.3.1.3. 連續性語音處理(CSP)程式庫

- Host based automatic speech recognition (ASR)
- High performance each cancellation
- Voice energy detection, barge-in, voice event signaling, pre-speech buffering
- Full duplex operation
- Including library functions, device driver, fireware and demonstration
- Streaming echo-cancelled data to the TDM
- Each canceller tap length of up to 512 taps
- Ability to re-arm the VAD
- More powerful Voice Activity Detector

2.3.1.4. FAX API 程式庫

- 支援 Wide variety of Fax applications
- 提供 Intel Dialogic 第六版軟體 for CompactPCI an windows
- 支援 ITU-T G3 ,V.17, 資料速度 14.4K

- 符合 Data transmission encoding scheme with advanced compression
- Polling and turnaround
- 符合 Simple header overlay
- 具備 Image bit order (MSB/LSB) conversion
- Encoding of color fax imagines

2.3.1.5.整體呼叫程式庫

- Uniform call control interface for developing application
- Protocol operation on Intel NetStructure, DM3 and Springware architecures
- Designed to support a variety of protocols
- Provide a consistent application interface
- Use the same input and output parameters at the application level
- 支援 H.323 and SIP host-based stacks
- 符合 Multi-protocol support on Global call devices
- 符合 Register with a Gatekeeper or Registrator
- 支援 Support DTMF Mode
- 支援 Fax transmission and reception

2.3.1.6 IP 中介媒體(Media) 程式庫

- Used to control media on IP devices
- IP Media library (IPML) Provides DialogicR 6.0 & CompactPCI on Windows
- Preferred DTMF Mode, UII Alphanumeric
- Send and receive Fax for T.38
- Using IPML API to build a PSTN-IP gateway
- Media resource management and media resource operation functionality
- Quality of Service(QoS) threshold alarm configuration and status reporting
- Support of Standard Runtime Library(SRL)

2.3.1.7 模組中繼介面應用程式(Modular Station Interface API)

- Provide high density analog station connectivity
- Support up to 120 stations with tone diction and generation
- 支援 FSK caller ID transmission
- Conference Management functions to control conferencing features
- Configuration functions to set and retrieve device parameters
- Device Management functions to set and retrieve device parameters
- Diagnostic functions to test devices
- Routing functions to allow communication between time slots on the CT bus

2.3.1.8. 自然建構管理(Native Configuration Manager)應用程式庫

- 提供 Interface for developing customized system configuration
- 具備 Get the AUID of a board
- 具備 Get the board name from and AUID
- Modifying board-level and system-level configuration
- Starting, stopping and checking the status
- Setting the TDM bus clock master fallback list
- Getting stystem software version information
- Cached prompt management
- Enhancement to Multi Frequency signaling
- Enhancement to DV_TPT termination conditions

2.3.1.9 標準執行(Standard Runtime)應用程式庫

- Provide a common interface for event handling and functionality
- Provide the framework for implementing the programming model
- Events are handled in a standard manner
- Support for alternative variant of the extended asynchronous programming
- Support for the synchronous model
- Support for synchronous with SRL callback model
- Support for polled model
- Asynchronous with non-signal callback model
- Extended asynchronous model
- Device event management
- Device information retrieval using ATDV_prefixed functions

2.3.1.10 語音應用程式(Voice API)

- Building a wide range of high-density call processing application
- Voice messaging, interactive voice response,telemarting/call center
- Including tone signaling,global tone detection and generation
- Call progress analysis and voice encoding algorithm selectable
- Cached prompt management
- Interactive multimedia association with ADPCM algorithm
- Using linear coding ,VOX and WAVE file format
- Speed control and volume control
- Transmit/receive analog display service interface
- Perfect call call progress analysis
- Transaction record
- Bulk data buffer sizing

3. 心得

3.1. 近年來由於 IP 硬體設備及相關規約標準的日趨成熟,國外通信大廠主力產品均以 Internet 應用環境為主.其功能更加完整且價格不斷下降,因此市場競爭非常激烈,在我們研習之 Intel 公司通信部門均感受人事異動/精簡之壓力,每位工作人員需全力以赴,才能存活下來

3.2. VOIP 之遠景看好,語音之重要是永恆的,VOIP 初期利基在廉價之長途及國際電話,但長期利基應在多媒體化,加值化等新服務更具發展空間

3.3. 語音通信著重連續性訊流,須具即時特性,因此 VOIP 新規範 RTP 適時產生

3.4. RTP(Real Time Transport Protocol) 即時傳輸協定,具有 16 比次之標頭序碼,32 位元之時間截記(Time stamp)以配合靜音抑制, 64 位元之 Source ID, 使二個 Port 間可以同時進行傳送數個資訊流, RTP 附屬於應用層, 路由器並不予處理

3.5. RTCP(Real time Transport Control Protocol)係監視 RTP 之運作,並提示報告,即時性(如語音)或交談式的訊務,優先安排於序列之前頭,公平第按序處理,於產生擁塞之前,及早偵出,丟出一些 IP 標頭所註 Class 級較低之封包.

3.6. DiffSer 差別式服務,封包進入 Edge 路由器註以 Class 參數,骨幹網路路由器依封包 Class 施以不同等級處理,訊務量在進入點之節制以防擁塞

3.7. IP Media Gateway(IPT 1200C)主要特點為提供開放及標準的服務平台,並具有高可靠性,可擴充性之 IP 閘道器.IPT 2000 符合下一代網路(NGN)架構需求及具有重大新功能.,在單一 CompactPCI 電路板可以提供 672Channel 之電話服務

3.8. IP PSTN Gateway(DM/IP 481)主要特點為允許語音經由 IP 傳送(VOIP),透過 Intel NetStructure 平台至 CT Bus. 利用 DM/IP Board 接至 PSTN 網路,完成 IP 與 PSTN 介面功能,並可具有會議電話,語音辨識,文字轉語音等多樣化之應用服務.

4. 建議

- VOIP 隨著服務需求及技術進步,由早期特定使用者以節省國際電話為目的,演進為現今以 IP 網路傳送語音信號,並結合多媒體通信服務為需求,以致應用面更為廣泛.然而,隨著IP網路建設日趨普及,與傳統PSTN電路交接更加複雜,對於本公司現有網路建設與維運需求,建議儘早規劃本公司整體 IP 網路發展策略,維持最佳之市場競爭力及滿足客戶需求

- 國外先進電信廠商對於 IP 技術與產品,不斷推陳出新,本所係研發單位,對於IP 新技術之學習與引進,仍須持續進行與積極推展,以技術領先同業,讓公司達到永續經營與成長之目標.