# Text-to-Speech Digital Public Address System based on Internet Telephony Transport Protocol: A Review

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## ABSTRACT
In this paper, we describe the formatting guidelines for IJCA Journal Submission. A digital public address (PA) system which is capable of multi-zone and text-to-speech (TTS) broadcasting functions for campus broadcasting. The digital PA system can achieve environmental broadcasting requirement which means different broadcasts for multiple zones at the same time and it cannot cause noise to other zones. The emergency broadcasting is major considered functionality for the modern digital PA systems. Digital PA system is very suitable for language listening training. ITTP is used to address key Voice over Internet Protocol (VoIP) requirements and solve the problems resulting from Real Time Protocol(RTP) /User Datagram Protocol(UDP). The Comparison shows that Internet Telephony Transport Protocol (ITTP) gives better performance compared with RTP/UDP in terms of packet loss, delay and bandwidth usage.

## Keywords
ITTP, RTP, UDP, VoIP

## 1. INTRODUCTION
A public address (PA) system consists of an electronic sound amplification and distribution system with a microphone, amplifier and loudspeakers. In general, PA systems are classified into two categories: analog PA system and digital PA system, respectively.

Presently, digital PA systems have been widely adopted in many daily public places such as campus, manufacture factory, airport, shopping hall, and intelligent building etc. Compared to analog PA system, generally, the modern digital PA system can be supported multicast- oriented addressing broadcasting, which can be achieved environmental broadcasting requirement.

The emergency broadcasting is also major considered functionality for the modern digital PA systems.

Digital PA system is also very suitable for language listening training in the campus. However, language listening exercises usually use pre-recoded method, but at some point, that we may need to temporarily add a new broadcast sentence immediately, however, we may not find a professional speaking teachers immediately for providing stable and superior oral recordings. Thus, TTS-based broadcasting will be a very convenient and practical tool, which can offer a high quality and stable speech [7]. we review a digital PA system, which is capable of multi-zone environmental broadcasting with TTS ability, campus emergency calls integrated with campus emergency facilities, and so on.

VoIP utilizes network infrastructure to replace current circuit switching telephone networks called the Public Switched Telephone Network (PSTN) with Packet Switching telephone networks [2-4]. Furthermore, VoIP technology employs network protocols to transfer calls around the world.

H.M. Chong and H. S. Matthews worked on Comparative analysis of traditional telephone and voice-over-Internet Protocol (VoIP) systems, 2004. X. Wang, J. Lin, Y. Sun, H. Gan and L. Yao applied feature extraction of speech recognition on VoIP auditing, M. AbuAlhaj, M. S. Kolhar , M. Halaiyqah, O. Abouabdalla and R. Sureswaran , worked on Multiplexing SIP applications voice packets between SWVG gateways. T. Abbasi, S. Prasad, N. Seddigh and I. Lambadaris did a comparative study of the SIP and IAX VoIP protocols. M .M. Abu -Alhaj, M. S. Kolhar, L.V. Chandra, O. Abouabdalla and A . M . Manasrah did Delta-multiplexing: A novel technique to improve VoIP bandwidth utilization between VoIP. Tsung-Hsinh Lin , Liang-Bi Chen, Tung-Lin Lee , Yung-Chang Tseng , Chaio-Hsuan Chuang , Chung-Heng Chuang , Chih-Lin Hunh , and Chao-Wen Wu proposed a Text-to-Speech-based Digital Public Address System for Campus Broadcasting ang Language Listening Training , 2013. M .M. Abu -Alhaj, Manjur SK , R. Sureswaran, Tat-Chee Wan, Imad J. Mohamad and A . M . Manasrah worked on ITTP A new transport protocol for VoIP applications , 2012.

## 2. TEXT-TO-SPEECH PA SYSTEM
The TTS-based digital PA system for campus is shown in Fig. 1. Basic hardware architecture of the TTS-based digital PA system consists of a digital PA control unit, a server with its dedicated control software, and message remote decoders for each zone. The digital PA system for campus will be introduced as follows.

A. Digital PA Control Unit
The digital PA control unit is adopted with digital two-way transmission system in a single cable connection. That reduces the installation time aswell as makes the installing easier and repairing faster. A digital PA control unit can support maximum 64 single zones, 8 grouping zones broadcasting by real hardware implementation. It can be equipped with 8-CH input mixer. Moreover, if the broadcasting zones are more than 64 zones, thus, we can also extend to 192 zones and 24 control groups by using dedicated control software method for reducing the cost of hardware [7]. Moreover, it can support wire-breaking detection also.

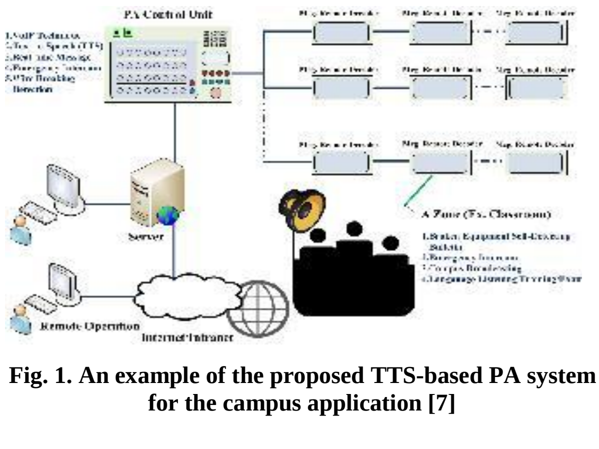


**Fig. 1. An example of the proposed TTS-based PA system for the campus application [7]**

B. Server with its Dedicated Control Software

The server runs user-friendly dedicated control software, which can be handled all functions of the TTS-based PA system. The main execution screen of TTS-based PA system is shown in Fig. 2. It is synchronized with the proposed digital PA control unit, and it can show all the broadcasting status at the same time.

Users can set remote, telephone, scheduling, grouping zones, any zones and all zones broadcasting functions by using dedicated control software. It provides 8 sets of play mode for daily routine, special events music broadcasting, such as festival, exam and work [7].

Fig. 3 shows execution screen of TTS function. The TTS broadcasting function supports settings of male/female speaker selection, rhythm, speed, modulation, and pitch. Currently, 2 languages of TTS are provided which are English and Chinese.

C. Message Remote Decoder

The message remote decoder is two-way designated address broadcasting. It is also built-in two-way intercom module for executing intercom function. The functions of fire emergency priority broadcasting and the broken equipment self-detecting bulletin for server are also provided.

D. VoIP Systems

Two main protocol categories are used in VoIP systems [4,5].The first category comprises the signaling protocols which are used to establish and manage a session between call endpoints [6].There are two standard signaling protocol for VoIP , namely, H.323 and the session Initiation Protocol (SIP). Recently, the InterAsterisk Exchange Protocol (IAX) has been introduced as a new signaling protocol.

The Second Category comprises the media transfer protocols. Typically, media transfer protocols are used for the exchange of media data once a session is established between the call endpoints.

The Real-time Transport Protocol (RTP) is specializes in transferring all types as real - time media data, including VoIP .IAX , specifically IAX mini-frame, can transfer protocols, both RTP and IAX mini-frame, cannot transfer media data by themselves, For this reason, media transfer protocols work atop transport layer protocols.

Typically, the transport layer User Datagram Protocol (UDP) works in conjunction with media transfer protocol to transfer VoIP application data [8].

VoIP technology has started replacing PSTN technology because VoIP provides many advantages for the telecommunication field. The main advantage of VoIP is that it enables calls anywhere around the world at a cheap rate, and sometimes even for free, compared with the conventional PSTN pone system. Second,

VoIP enables other functions in addition to voice call, such as video streaming and text messaging, which make users' communication experience more interactive and meaningful. Third, VoIP provides a higher degree of reliability than PSTN. Finally, unlike PSTN which is a closed system, VoIP has a free and open architecture, which implies that VoIP extends the opportunity for innovation and creativity to everyone. As a result, the VoIP system continues to undergo rapid and further development [8].

However, VoIP applications must provide phone conversations of better or at least similar quality as the current PSTN technology. In recent years, VoIP developers have made every effort to provide VoIP applications that perform excellently considering the global scale of VoIP technology spread. With respect to these efforts, the present work protocol (ITTP), which is dedicated to carrying VoIP application data [8].

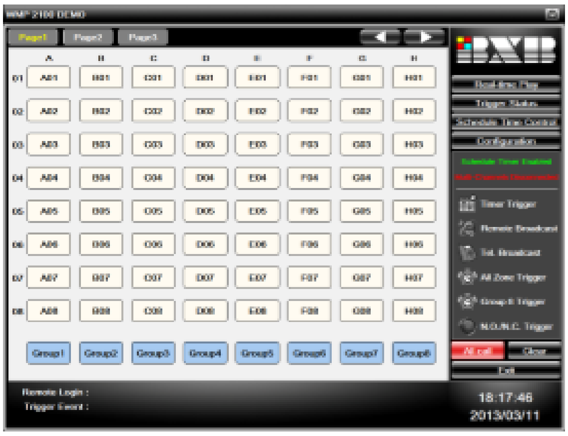

Fig. 2. Main execution screen of the proposed TTS-based PA system.

Fig. 3. Execution screen of TTS function handling.

D1: ITTP

There are many drivers behind the design of ITTP as a new transport layer protocol specialized in carrying VoIP application data. These are some disadvantages of RTP, UDP and IAX.

None of existing transport layer protocols address VoIP application requirements.

RTP and UDP degrade voice quality.

Apart from degrading voice quality, RTP/UDP causes an inefficient use of bandwidth in high - cost Internet links.

RTP/UDP burdens Internet links.

The IAX mini-frame causes the same problems as RTP/UDP.

IAX mini-frame has no chance to spread in the VoIP world.

D2: ITTP Protocol Header

ITTP consists of a 2-byte Source-Port field, 2-byte Destination-Port field, and 2-byte Timestamp field .Figure 4 shows the ITTP header format.

| 0<br>15 | 16<br>31 |
|---|---|
| Source - port | Destination - port |
| Timestamp | |

**Fig. 4. ITTP header format [8]**

Source-Port Number. The source-port number is the transport layer address used to identify the application on the sender endpoint, Usually, the source-port number is used by the receiver endpoint for acknowledgment purposes.

Destination-Port Number. The destination-port number is used to identify the transported application in the receiver endpoint.

Timestamp. The timestamp is a key field in ITTP. The Timestamp represents the number of milliseconds since the first data packet transmission of the call. For each packet, the timestamp increases by a value equal to the packet payload time length. The timestamp field is 16 bits.

The timestamp is used by VoIP application key functions to ensure real-time and smooth delivery:

First, the timestamp is used for timely VoIP packet delivery.

Second, the timestamp is used to overcome the variability of the received bit rate.

Third, the timestamp is used reorders out-of-order packets.

Fourth, the timestamp is used to discards duplicate packets.

Table 1. ITTP performance[8]

| Element | Effect |
|---|---|
| Header overhead | Considerable header overhead reduction |
| Capacity | Increases the number of concurrent calls running in a specific channel |
| Bandwidth Usage Efficiency | Improve the bandwidth utilization |
| Delay | Reduce the delay |
| Packet loss | Reduce the packet loss |
| Voice quality | Improve the overall voice quality |
| Buffer Utilization | Improve the buffer utilization |

# 3. CONCLUSION

In this paper, the TTS-based PA system has been explained for campus broadcasting. The digital PA control unit can support maximum 64 single zones, 8 grouping zones broadcasting by real hardware implementation. It can be equipped with 8-Channel input mixer. Two languages Chinese or English are provided for TTS broadcasting to achieve high quality and stable voice broadcasting to the target zones.

In this paper, a new transport protocol called ITTP, which is dedicated to carrying VoIP application data, has been explained. ITTP has reduced header overhead compared with RTP/UDP, the currently used protocols.

# 4. REFERENCES

[1] Public address, http://en.wikipedia.org/wiki/Public_address.

[2] H.M. Chong and H. S. Matthews, Comparative analysis of traditional telephone and voice-over-Internet Protocol (VoIP) systems, 2004 IEEE International Symposium on Electronics and the Environment, 2004.

[3] X. Wang, J. Lin, Y. Sun, H. Gan and L. Yao, Applying feature extraction of speech recognition on VoIP auditing,

[4] M. AbuAlhaj, M. S. Kolhar , M. Halaiyqah, O. Abouabdalla and R. Sureswaran , Multiplexing SIP applications voice packets between SWVG gateways, Proc. of the 2009 International Conference on Computer Engineering and Applications, pp.237-241,2009.

[5] T. Abbasi, S. Prasad, N. Seddigh and I. Lambadaris, A comparative study of the SIP and IAX VoIP protocols, Canadian Conference in Electrical and Computer Engineering, 2005

[6] M .M. Abu -Alhaj, M. S. Kolhar, L.V. Chandra, O. Abouabdalla and A . M . Manasrah, Delta-multiplexing: A novel technique to improve VoIP bandwidth utilization between VoIP gateways , 2010 IEEE 10th International Conference on Computer and Information Technology, 2010.

[7] Tsung-Hsinh Lin , Liang-Bi Chen, Tung-Lin Lee , Yung-Chang Tseng , Chaio-Hsuan Chuang , Chung-Heng Chuang , Chih-Lin Hunh , and Chao-Wen Wu: A Text-to-Speech-based Digital Public Address System for Campus Broadcasting ang Language Listening Training , 2013 IEEE 17th International Symposium on Consumer Electronics , 2013.

[8] M .M. Abu -Alhaj, Manjur SK , R. Sureswaran, Tat-Chee Wan, Imad J. Mohamad and A . M . Manasrah: ITTP: A new transport protocol for VoIP applications , 2012 International Journal of Innovative Computing, Information and Control, volume 8 , Number 3(A), March 2012.

[9] Li, Chin-Lin Hung, and Chao-Wen Wu, A Hybrid Multi-Function Digital Public Address System with Earthquake Early Warning.”Proceeding of the 2014 Tenth, International Conference On intelligent Information Hiding and Multimedia Signal Processing”,2014.

[10] Tsung-Hsinh Lin, Liang-Bi Chen, Chung-Heng Chuang, Tung-Lin Lee, Chaio-Hsuan Chuang, Yung-Chang Tseng, Chun-Long Chiu, Chin-Lin Hung, and Chao-Wen Wu, A Multi-Functions Digital Public Address System for campus broadcasting and security.”Proceeding of the 2013, Global High Tech Congress On Electronics (GHTCE)”,2013,IEEE .

[11] Broadcast Engineering, Vol 32.pp 64-65,Jun 2005. ] Xu Guanghui, Cheng Dongxu, Huang Ru,The embedded development and application based FPGA. Publishing House of Electronics Industry, Beijing, sep 2006.

[12] J.-S. Kim. “Development of the digital public address system,” Proceedings of the 2012 IEEE International

Conference On Computing Technology and Information Management (ICCM'12), vol. 2, pp.632-638, 2012.

[13] J.-S. Kim. "Development of the digital integrated and minimized public address system with central control," International Journal of Control and Automation, vol.5, no.3, pp.267-276, Sep. 2012.

[14] Y. Fu and B. Tan, "A design of network digital audio public address system ,"Proceeding of the 2010 IEEE International Conference on E-product E-Service and E-Entertainment (ICEEE'10), pp. 1-3,2010.

[15] S.-B. Yu, M.-J. Cho, Y.-s. Park, and J. Hwang, "Emergency broadcasting with ATSC mobile DTV,"Proceeding of the 2012 IEEE International Conference on Consumer Electronics (ICCE'11), pp. 73-74,2011.

[16] Luo Dongshan and Huang Xiaoge, Audio Transmission network based on CobraNet Technology.Radio and TV

[17] Jung-sook Kim, "Development of the digital public address system." Computing Technology and Information Management (ICCM), 2012 8th International Conference",2012

[18] Pei- Yang Lin worked on Earthquake Early Warning Systems,2011. National Center for Research on Earthquake Engineering, National Applied Research Laboratories (NARL), Taiwan 2011.

[19] ] Ayeshabanu Ashraf Kalaniya, Versatile Public Address System," Shri Sant Gadge Baba College of Engineering and Technology, Bhusawal, India, 324-327, PISER12, Vol.02, Issue:02/06, March-April.