

A Preview of the Voice Telco Networks and Studio Interface Systems Course



About the Author

This course is written for SBE by Steve Church of Telos Systems.

Introduction

This course will be of interest to those in both television and radio station facilities. Telco lines and circuits are integral parts of most broadcast installations. We use regular phone lines for talk shows and contests, we use ISDN lines for remote broadcasts and STL backup, and we use T1 lines for STL applications and business telephone service. The broadcast engineer must have a good understanding of the various types of lines and protocols in order to deal with the telephone companies. It is also helpful to understand telco terms and "lingo" so that he or she can "speak the same language" when talking with testers, installers and field repair people.

The course provides a complete overview of all these topics, giving the student a working understanding of how the signal gets from end to end, how to troubleshoot telco problems and how to properly interface broadcast equipment to the telco world.

Course Description

Listener involvement via telephone is an important programming element at many radio and television stations. When we want to create a two-way connection with our listeners, we will probably be using the voice telephone network. Most often, telephone service to our studios is via an old-fashioned analog POTS (Plain Old Telephone Service) line. The vast majority of subscribers interface to the network via an analog technology that would be familiar to Alexander Graham Bell. Digital connections via ISDN and T1 lines offer an important step up in quality and flexibility. Voice over Internet Protocol (VoIP) is sure to be big. Perhaps we'll eventually need a revision of this course that will be devoted to VoIP and include a mention of analog lines as a historic footnote. In any event, even should the service fail to become mainstream for Telco delivery of calls, VoIP will figure prominently within studio facilities.

There are several quiz questions prior to most of the chapters to help the student understand what the chapter covers and confirm to the student how much they already know.

Course Content

1. Introduction to Voice Telco Networks and Studio Interface Systems
2. Overview
3. Analog (POTS) Telephone Service – Part 1
4. Analog (POTS) Telephone Service – Part 2
5. Digital Switched telephone Service
6. Special Services for Broadcast Stations
7. VoIP Interworking with the PSTN (Public Switched Telephone Network)
8. The Cellular Mobile Network
9. Broadcast Interfacing
10. Troubleshooting
11. Summary of Voice Telco Networks and Studio Interface Systems

SBE Recertification Credit

The completion of a course through SBE University qualifies for 1 credit, identified under Category I of the Recertification Schedule for SBE Certifications.

Enrollment Information

SBE Member Price:	\$80
Non-Member Price:	\$115

Analog (POTS) Telephone Service - Part 1

The usual analog POTS circuit is called *2-wire*, because it uses a two-wire pair. The network is internally *4-wire*, so named because in the past, a 4-wire circuit needed two pairs for each of the send and receive transmission directions – four wires altogether.

The lines provided by Telcos are known officially as subscriber loops, trunks or simply CO (Central Office) lines. Trunks are usually lines destined for PBXs and sometimes include special signaling as well.

Because these are 2-wire circuits, the CO uses a 2-to-4-wire converter (also called a *hybrid*) to interface the analog lines to its internal 4-wire system. This process happens on the line card, which is also responsible for digitization, talk battery insertion, off-hook detection, and ring generation.

2.1 Talk Battery and Ringing

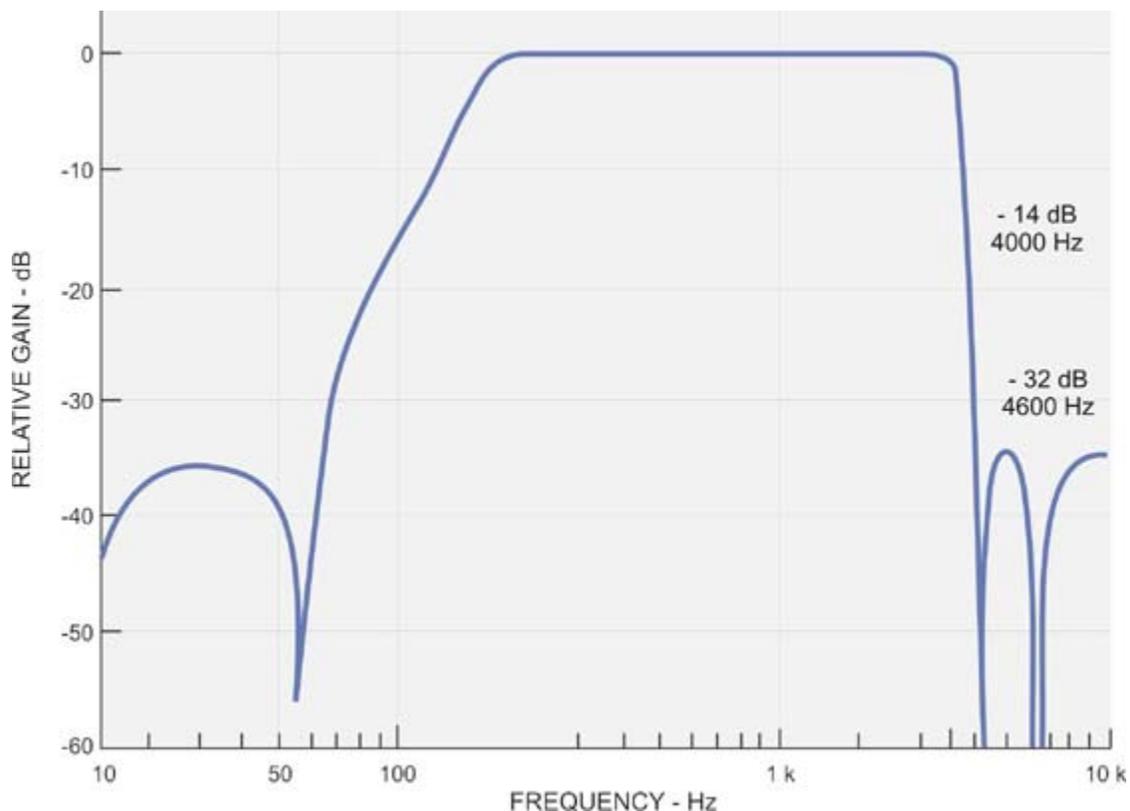
The *talk battery* DC voltage and the conversation audio appear together on the phone pair. The talk battery leaves the exchange at -48V and is limited to 20-100mA by a series resistor and the loop resistance; the value is engineered with the resistance of the loop in mind. The dc resistance of the loop itself varies from a few to 1,300 Ω depending on length. Because of this series resistance, when a line is off-hook, its voltage at the customer equipment drops to around -12V, but this value varies widely.

For ringing, an AC voltage of 90Vrms at 20Hz is superimposed on the line.

Talk signals are AC coupled with nominal impedance of 600 Ω . However, some CO equipment uses complex impedance coupling, and the nature of the telephone network means the actual impedance as presented to the user is rarely a simple 600 Ω . The impedance will almost certainly vary depending upon frequency, and unpredictably so due to many factors. This turns out to be an important issue for broadcast interfacing, as we will see later.

2.2 Frequency Response

Audio bandwidth for POTS calls are strictly limited to a 3.4kHz bandwidth by the sharp low-pass filters required for digitization. The phone network's 8kHz sampling rate permits a theoretical Nyquist frequency of 4kHz, but a 600Hz transition band is necessary for anti-aliasing and reconstruction filtering.



2.3 Noise and Level

Average conversation has a level of around -16dBm. But, as anyone who has wrestled with broadcast-to-telephone interfacing knows, incoming level varies tremendously, with a range of perhaps -40 to -4dBm.

Audio sent into the telephone line must be limited to average -9dBm as specified in FCC rules. Audio loss on any given local loop is limited by tariff to 8dB or less. This loss limit, however, applies only to the line from the CO to the subscriber and does not include the rest of the signal path. The 8dB loss may occur at each end of a conversation path, once at the calling party end and again at the called party end, for a total loss of 16dB.

Telephone engineers measure noise upside-down, defining a reference noise floor and then measuring *up* from there. The reference noise level is one picowatt, which corresponds to -90dBm. Thus, a noise level of -60dB relative to 0dBm would be reported as 30dBm noise (dBm = dB above reference noise). In contrast to professional audio measurements, the higher this number, the worse the noise.

Be aware also that when telephone people measure noise, they are measuring only *idle channel noise*. This is an important difference, since in digital systems idle channel noise is not the same as the traditional (S/N) measurement in analog systems. Noise in a digital system will generally increase when a signal is present. This effect is called *modulation or quantization noise* and is primarily dependent upon the number of bits used for quantization.

A *C-message weight filter* is employed when measuring phone line signal-to-noise ratio (S/N). The C-message curve was developed years ago to simulate the frequency response of an old-style telephone earpiece and, accordingly, it has considerable low-frequency roll-off. This means that a line can have significant hum and other low frequency noise and can still meet the officially mandated noise specs. While this makes life easier for phone company technicians, it can be a problem when you are trying to use phone audio on the air and find yourself plagued with unwanted low-frequency sounds that the C-message meter doesn't register.

Conversion between 2-wire and 4-wire POTS circuits is accomplished by

- A combiner.
- A hybrid.
- Digital interleaving.
- N/A; 2-wire is never converted to 4-wire.

The approximate dynamic range of a POTS line served by a digital CO is:

- 13-bits, corresponding to 78dB.
- 8-bits, corresponding to 64dB.
- 16-bits, corresponding to 96dB.
- Can't say, because quantization noise influences the measurement